Certificate in Community Radio Technology

Studio Technology

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Module: 3
Studio Technology

CEMCA
Commonwealth Educational Media Centre for Asia
New Delhi

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Broadcast Engineering Consultants India Ltd.
Noida, UP

Module 3: Studio Technology

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Certificate in Community Radio Technology

Courses	Modules	Units
Course I: Understanding Community Radio (3 Credits, 90 Hours)	**Module 1** Community Radio: An Introduction	**Unit 1 :** Community Radio: Concept and Evolution **Unit 2:** Context, Access and Equity **Unit 3:** Community Radio: Policy Guidelines **Unit 4:** Technology for CR: Guiding Principles
	Module 2 Setting up of CRS	**Unit 5:** Components of CR Station **Unit 6:** Radio Waves and Spectrum **Unit 7:** Basics of Electricity **Unit 8:** Power Backup and Voltage Stabilisation
Course II: Community Radio Production: System & Technology (5 Credits,150 Hours)	**Module 3** Studio Technology	**Unit 9:** Basics of Sound **Unit 10:** Analog and Digital Audio **Unit 11:** Components of the Audio Chain **Unit 12:** Studio Acoustics
	Module 4 Audio Production	**Unit 13:** Audio Hardware and Field Recording **Unit 14:** Free and Open Source Software **Unit 15:** Telephony for Radio
	Module 5 Audio Post Production	**Unit 16:** Sound Recording and Editing **Unit 17:** Mixing and Mastering **Unit 18:** File Formats and Compression **Unit 19:** Storing and Retrieval
	Module 6 Studio Operations	**Unit 20:** Good Engineering Practices for Studio Setup **Unit 21:** Studio Equipment: Preventive & Corrective Maintenance **Unit 22:** Content Distribution: Alternative Mechanisms
Course III: Community Radio Transmission: System & Technology (2 Credits, 60 Hrs)	**Module 7** Radio Transmission Technology	**Unit 23:** Components of Transmission Chain **Unit 24:** Components of FM Transmitter **Unit 25:** Antenna and Coaxial Cable **Unit 26:** Propagation and Coverage
	Module 8 FM Transmitter Setup	**Unit 27:** Transmitter Setup: Step-by-step **Unit 28:** Transmission System-Preventive and Corrective Maintenance **Unit 29:** Transmission Setup-Good Engineering Practices
Course IV: Technical Internship (2 Credits, 60 Hrs)	**Module 9** Practical Internship Handbook	**Section A:** Introduction **Section B:** Activities to be Conducted During the Practical Internship **Section C:** The Internship Journal and Self-Assessment Paper **Section D:** Assessment of Internship **Section E:** Appendices

Video in the Module:

1. **Characteristics of Sound**
   By Vasuki Belavadi
2. **Digital vs Analogue and Mono vs Stereo Audio**
   By Vasuki Belavadi
3. **Acoustics & Noice Reduction**
   By Hemant Babu

1. http://tinyurl.com/qahdxzx
2. http://tinyurl.com/ofzsm72
3. http://tinyurl.com/nbyzl64

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CONTENTS

About the Module

This module is first in line of the four modules that makes course II viz., “CR Production: System Technology”. As its name suggests, it is all about making of an audio studio. It has four Units with the first one (Unit 9) discussing the basics of sound, its properties and components and how changing any one of the components can change the quality of sound. This Unit also describes the differences between mono and stereo audio.

Unit 10 in this module explains digital audio signals in depth, while summing up the characteristics of analogue audio. It also explains concepts such as sampling, quantisation, bit error rate etc.

Unit 11 in this module concentrates on the Audio Chain and how physical components of recording and reproduction systems i.e. broadcast equipment are connected together in a studio and the signal flow-path taken by the sound from acquisition to its reception by the listener. You will also learn about the hardware required to produce programmes in a studio and outdoors.

Unit 12 deals with Studio Acoustics and introduces you to the basics of acoustic such as reverberation, echo, sound refraction, etc, all of which are crucial to designing a sound proof audio studio. It describes in detail the sources of noise inimical to good radio production and provides tips on how to avoid noise.

Since, there are a variety of sound proofing materials available in the market, it also describes to what extent each material absorbs sound to make the studio noise proof. The crucial elements of an audio studio such as sound proof door, observation window etc are also discussed to provide an idea of how an audio studio could be set up.

Module Objectives

- To discuss properties and components of sound
- To discuss the difference between analogue and digital audio
- To understand the hardware required for field recording and setting up a studio
- To discuss the fundamental principles and the care to be taken, while setting up an audio studio

Units in the Module

- Basics of sound
- Analogue and Digital Audio
- Components of the Audio Chain
- Studio Acoustics

UNIT 9

Basics of Sound

Structure

9.1 Introduction

The earlier Units introduced the concept of Community Radio, the policy guidelines in India, technology and the components of a Community Radio Station (CRS). This Unit will introduce the characteristics of sound, which is a basic ingredient of radio. In order to understand how sound works in different situations, it is equally important to understand the components of sound. This unit will explain the different types of sound that will go into the making of any radio programme.

Some of the concepts might be hard to comprehend. However, this Unit will have activities that ideally require access to the internet, an audio recorder and headphones. The video links provided as a part of this Unit will help to understand more clearly, some of the concepts discussed in this Unit.

9.2 Learning Outcomes

After working through this Unit, you will be able to:

- List and describe the characteristics of sound
- Explain the difference between pitch and volume
- Identify sounds that have different attack, sustain and decay
- Explain the process of propagation of sound waves
- Identify and differentiate between mono and stereo sound

9.3 Understanding Sound

Sound is all pervasive. Right from the time we wake up in the morning and turn on the tap in the bathroom until we go back to sleep. Sounds are of different types, intensities and pitches. One cannot imagine the world without sound. Try to lock yourself up in a room and close the door and all the windows. Sit for about an hour without moving. You will realise what it is to live without sound.

We use sound to communicate with each other. We say “it sounds good” when the sound we hear is pleasing to our ears, but if it gets louder and noisy we feel uncomfortable. You may have noticed that certain sounds are by themselves significant. For example, when we hear the doorbell, we know we need to open the door. When we hear the police siren or the horn of an ambulance, we get alarmed.

What is sound? Sound is created when an object vibrates. Tap on a table and you hear a light sound. Now, thump on the table, the sound gets louder. What happens when you thump on the table is that the molecules in the air, around your fist and the table vibrate at very great speed. It is this vibration that you perceive as sound. When you just tap the table, the speed of displacement of molecules is lesser than when you thump the table. You hear a faint sound when you tap the table and the sound gets louder when you thump it. The sensation of loudness depends on the intensity with which you tap or thump the table.

The intensity of sound is measured in decibel units (dB) and is logarithmic in nature. Therefore, if 10 dB is ten times 1 decibel, 20 dB is 100 times 1 dB (10 × 10=100) and 40 dB is 10,000 times 1 dB (10 × 10 × 10 × 10 = 10,000).

The 'threshold of hearing', that is, when sound is just about audible, is about 0 dB. The following table (Figure 9.1) should give you an idea of sound in different situations:

Sound sources (noise) Examples with distance	Sound Pressure in db
Jet aircraft, 50 m away	140
Threshold of pain	130
Threshold of discomfort	120
Discotheque	100
Diesel truck, 10 m away	90
Footpath of a busy road	80
Vacuum cleaner	70
Conversational speech	60
Average home	50
Quiet library	40
Quiet bedroom at night	30
Background in Radio studio	15
Rustling leaves in the distance	10
Hearing threshold	0

Figure 9.1: *Table explaining the loudness of sound in different situations*

So how do we hear sound? When something makes a noise, it sends vibrations or sound waves through the air.

The human eardrum is a stretched membrane like the skin of a drum. When the sound waves hit the eardrum, it vibrates and the brain interprets these vibrations as sound. However, when the intensity of sound increases beyond a certain level, we try to close our ears! Fig 9.2 explains how sound enters your ear and how these signals are sent to the brain to process.

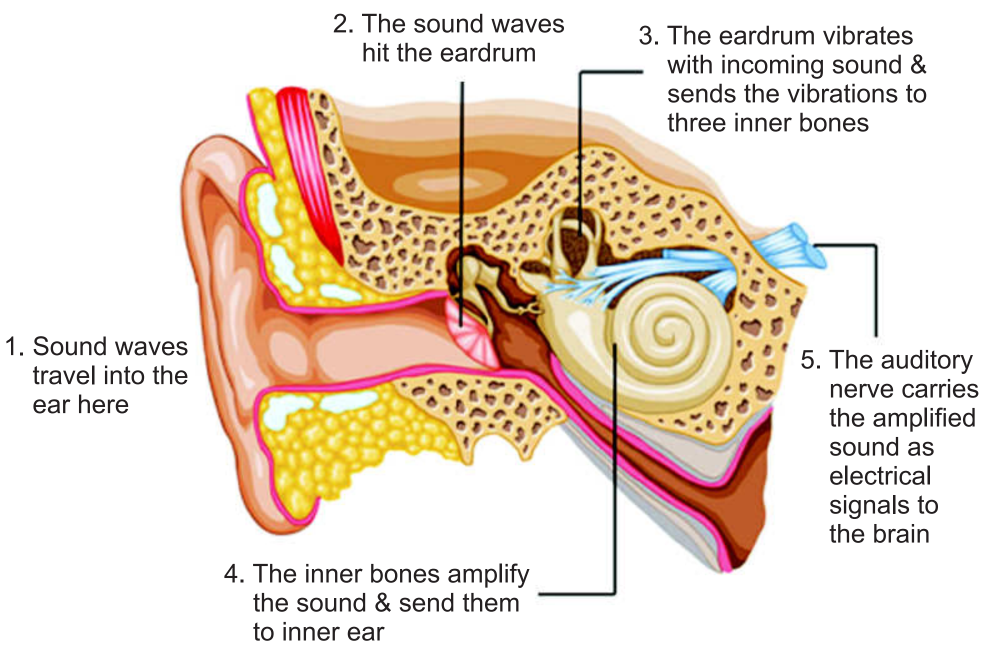


Figure 9.2: *Sound waves that enter the ear hit the eardrum and signals from the back-end of the eardrum, which are carried to the brain to process the waves.*

As the sound level increases, for example, when you get really close to loudspeakers at a festival *mandap* (at around 110 dB), you feel really uncomfortable and close your ears. Constant exposure to any sound above 90 decibels can eventually damage our ears.

9.4 Characteristics of Sound

Characteristics of Sound

http://tinyurl.com/qahdxzx

We have already learnt in our high school physics that sound travels in the form of waves. They are basically mechanical waves. This means that they require a medium to travel. When you speak, your speech travels through air. However, sound cannot travel through vacuum. For your clarity of the concept on characteristics of sound, please go through the video at http://tinyurl.com/qahdxzx. Please read this section of the Unit and watch the video. All types of sound have the following characteristics:

- Wavelength
- Amplitude
- Frequency

9.4.1 Wavelength

Wavelength (Figure 9.3) can be described as the distance between any point on the wave and a corresponding point on the next wave. Sound waves are longitudinal waves; their wavelength can be measured as the distance between two successive compressions (higher pressure and density regions) or two successive rarefaction (lower pressure and density regions). Wavelength is measured in metres.

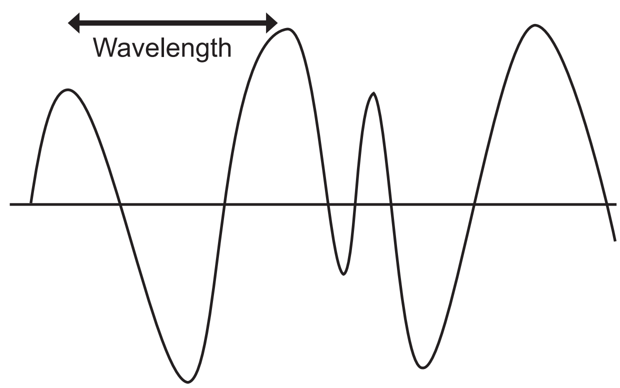


Figure 9.3: *Wavelength of sound*

9.4.2 Amplitude

Amplitude indicates the height of a sound wave as shown in the figure 9.4 — how loud the sound is. The higher the 'height' of the wave, the louder it is and vice-versa. Amplitude also indicates how strong a sound is.

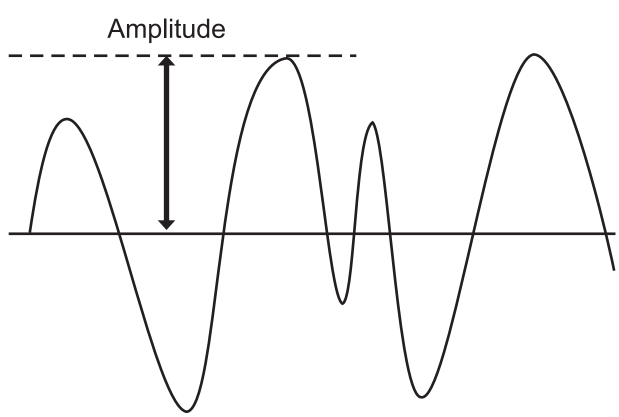


Figure 9.4: *Amplitude of a sound wave*

9.4.3 Frequency

Frequency is the number of times the wavelength occurs in one second (Figure 9.5). Frequency is measured by the number of sound vibrations in one second. If the source of sound vibrates faster, the frequency is higher and vice-versa. For example, if you examine the wave of the strums of a guitar, it will be different from those of the thud on a table. This means the higher the number of vibrations from the sound source, the higher the frequency. In turn, the higher the frequency, the higher the pitch. Frequency is measured in Hertz (Hz). One Hertz = one vibration/ second. The number of times an object vibrates determines its frequency.

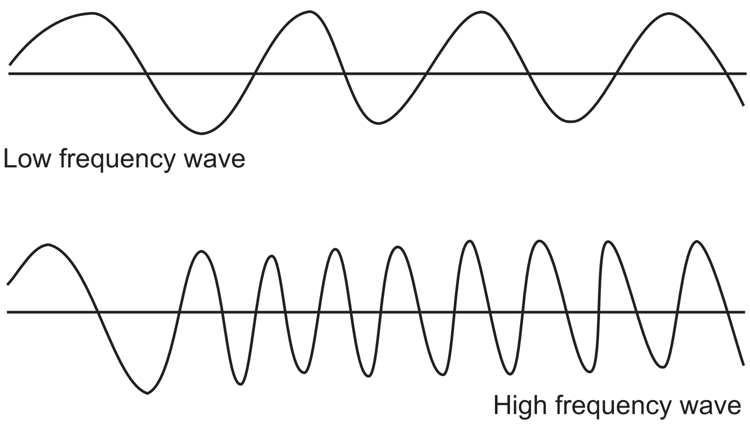


Figure 9.5: *Frequency of a sound wave*

Activity 9.1

If you know the frequency of sound, you can easily measure its wavelength. How does one do it? Just enter the url: http://www.mcsquared.com/wavelength.htm in your browser. Now, enter the required values in the field and hit the 'Wavelength' button. It will provide the answer.

The tool provided above is based on a programmed formula that provides answers to a value you enter in the required field. It automatically calculates the wavelength of a particular sound if you know the frequency.

9.5 Components of Sound

In order to understand how sound works in different situations, it is important that we study its properties. Modification of any of the components of sound can alter its effect in a radio production. Let us proceed to understand the different components of sound.

9.5.1 Pitch and Volume

The frequency of a sound determines its pitch. The higher the frequency, higher will be the pitch and vice-versa. Frequencies are grouped as Low frequencies (bass) - sounds of thunder and gunshots, Midrange frequencies - a telephone ring or normal speech and High frequencies (treble) - small bells or even shrieks. Sounds of lower frequencies are powerful. Sounds of midrange frequencies are energetic. Humans are most sensitive to midrange frequencies. High frequency sounds make their presence felt and add quality to the sound track. Different objects have different frequencies. When you pluck a rubber band, the frequency of sound it generates is less than the thinnest string on a guitar. Therefore, by extension, the pitch of the rubber band is less than the pitch of the thinnest string on a guitar.

Now, let us look at loudness or volume. This is also called amplitude of sound. The loudness of a sound depends on the intensity and distance of the sound stimulus. Obviously, a bomb explosion is louder than the bursting of a tetra-pack, despite both being at the same distance. This is due to the bomb explosion displacing more number of molecules than a tetra-pack. Loudness is relative. A sound that is very loud in a small room cannot even be heard on a busy street. Take a plastic cover and blow air into it. Now, hold its neck tight and burst it in a small room. The sound can be loud. But, if you try the same on a busy street, you might not even hear it properly.

Just a word of caution: one should not confuse pitch with volume or amplitude. They are two different things. For example, one can speak at a high volume, but at a very low pitch.

Activity 9.2

This activity will help you understand the difference between high and low pitch. The same activity can also be used to understand the difference between pitch and volume.

Take four steel cups of equal size from your kitchen. Now, fill one of them with water almost up to the top. Fill the next one about three-fourths full. The third one about half full. Leave the fourth cup empty. Next take a table teaspoon and tap each of the four cups in succession.

9.5.2 Timbre, Harmonics

Timbre and harmonics are that quality of sound that enables you to distinguish them from each other even when they are at the same pitch. To understand this better, let us first discuss harmonics. Say you pluck the string of a guitar or a *sitar.* The string sets off a main frequency at a certain volume but you also faintly hear other frequencies. Now, try the same with a rubber band. The main frequency and the resultant 'child' frequency that it produces are less than the one produced by the string of a guitar and *sitar*. The ability of an object to produce child frequency gives soothing sound. Therefore, the sound of a guitar is more pleasant than that of a rubber band.

Timbre is the combination of a basic frequency, the child frequency and the overtones that a sound produces. It is this combination that enables you to differentiate between two different trumpets, although they are at the same volume. In a way, the pitch of the sound also contributes to the timbre. If actor Amitabh Bachchan and Asrani speak at the same volume, you will be able to instantly recognise their voices because of the timbre. In short, the unique quality and characteristic of every sound can be described as its timbre.

Activity 9.3

This activity will help you understand the difference between timbres of different sounds. Take two cups, one made of steel and another made of glass of equal size from your kitchen. Now, fill one of them with water about half full. Fill the next one half full with sand. Then take a table teaspoon and tap each of the two cups in succession.

9.5.3 Rhythm, Tempo

Rhythm in simple words means, the silences between sounds. Rhythm is everywhere, the way one speaks, the ping-pong on a tennis table, the ticking of a clock, the hoofs of horses, the way it rains, and in clapping. The way sounds and silences are patterned form rhythms.

The rate at which rhythm repeats itself defines tempo. Walking has a particular rhythm. When you walk slowly, you walk at a slow tempo. When you begin to walk fast, the tempo and rhythm both change. In other words, the speed of rhythm is tempo. Musicians normally use the words rhythm and tempo to describe the way their music has been composed.

Activity 9.4

This activity is meant to understand the relationship between Rhythm and Tempo. You have surely done this in your childhood. But let's do it again.

Take two empty coconut shells. Find a flat stone and begin to tap the coconut shells in a way to reproduce the way a horse gallops. Initially, tap the shells in slow succession. Gradually, increase the speed at which you tap the shells on the stone and notice the gaps between the sounds and silences. You will also notice that the tempo, the frequency at which the gaps decrease simultaneously sound increases.

9.5.4 Attack, Sustain, Decay

Different genres and formats of radio production comprise at least 10-15% of sound effects. Understanding attack, sustain and decay of sound is crucial to using sound effects in a radio production. Every sound takes a certain time to rise up to its maximum amplitude, remains there for some time and then dies down. This of course, depends on the kind and nature of sound being used. Study the graph given below (Figure 9.4).

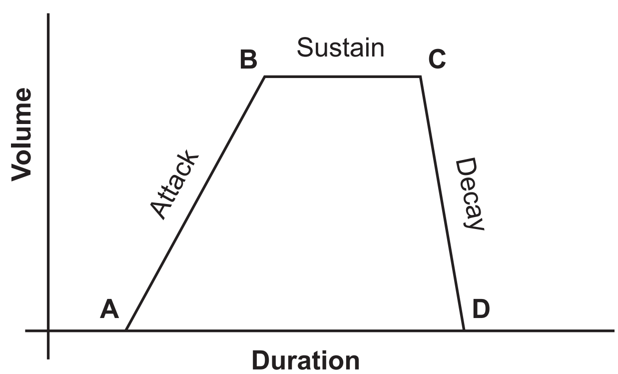


Figure 9.4: *Attack, sustain and decay represented in the form of a graph. The volume of sound is represented on the 'x' axis and the duration of attack, sustain and decay is on the 'y' axis*

The above graph denotes that the sound takes certain time to reach its highest volume starting from A to B. Having reached B, it stays there for some time up to C and then begins to die down from C to D. The duration that a sound takes to go from A to B can be termed attack. The duration from B to C can be termed sustain. Similarly, the duration from C to D can be termed decay.

The time taken by different sounds to reach their highest volume (attack) is different. For example, the sound from a pistol takes a very short time to reach its highest volume. It also dies down almost immediately. On the other hand, if one were to tear a paper slowly, the attack is slow. This sound sustains for a slightly longer duration and also dies down a bit slowly. The sound of waterfalls on the other hand takes longer to reach its peak, sustains for a longer duration and dies too, rather slowly.

One can change the attack, sustain and decay of a sound by changing the speed of sound. Try it out when you work with audio editing software.

9.6 Propagation of Sound

We have already learnt that sound travels in longitudinal waves. We have also learnt the various characteristics and components of sound. We further learnt that sound requires a medium to travel. Sound travels through air, through liquids, solids and gas. But, it cannot traverse through vacuum. This is because there are no particles in vacuum to get compressed and exploded.

Does sound travel with equal speed through all media? No, it travels fastest through solids. Even among solids, it travels the fastest through more tightly packed materials, which means sound travels faster through metals, than it does through wood. For example, it also travels faster in liquids than in gas. This is again because molecules in liquid are more tightly packed than in gas and air.

Does temperature play a role in propagation of sound? Of course it does. When it is hot, molecules in air are more excited and travel with more energy. Therefore, sound also travels faster in higher temperatures than it does in sub-zero temperatures.

Sound not only travels fast or slow depending on the medium, it also gets reflected when it hits a surface. Sound is reflected better from smooth surface than rough and tightly packed materials. Therefore, sound gets better reflected from a glass surface than from a surface that is covered with foam-like material. More porous the material, the absorption will be more. This is the principle behind acoustic treatment in studios.

It is worth noting that larger the surface area, better the reflection. And reflection of sound is manifested in echo. You do not seem to experience echo in a small room. However, when you shout out loud in a hilly area, your voice hits the mountains, gets reflected and returns to your ears, that is why you hear an echo. The number of echoes are dependent on the number of reflections that your shout experiences. Similarly, the roaring of thunder is also due to successive reflections of sound from clouds and the earth surface. Heavier the clouds, more thunderous is the echo.

9.7 Types of Programme Sound

Switch on a radio set and listen to a couple of programmes. You will hear people speaking, some music and also probably some sound effects in some programmes. However, the sound, music, speech used for an educational programme will be different from the one used for a peppy film-based programme. The mood of the programme determines the kind of sound one uses. However, they all use sound of three different types:

Spoken sound, sound effects and music

It must be remembered that sound is not incidental to a programme. It requires conscious planning at the pre-production stage. The nature of a programme, the mood of a scene in a radio drama or documentary or even an educational programme decides the kind of spoken words, music and sound effects. It would be absurd to include 'twangs' in a serious scene of a radio drama.

Spoken sound can be in the form of a narration or a character's dialogue. Obviously, an RJ's introduction to a programme is spoken sound. The manner in which the person speaks affects the effectiveness of the spoken word. The voice itself, emphasis on certain words, the inflection, the pitch and its loudness all contribute immensely to the overall effect. Dialogues in a radio drama cannot be delivered like a news reader's announcement of headlines. Interview speech is normally slow at the beginning. However, when the questioning becomes intense, the volume and sometimes the pitch too go up.

Anything other than music or spoken word is a sound effect. The creak of a chair, bang of a door, a ring tone, falling of books, the wang of a laser gun etc are all examples of sound effects.

Sound effects are also of three types. **Contextual** sounds emerge from a sound source as a consequence of an action. The dialogue in a drama is contextual sound. Let's say a person is in a fit of anger and shoots a person in a radio drama. The sound of the gun shot, although added as a sound effect is contextual sound (also called diegetic sound) because it emerged from within the story. **Descriptive** sounds, as the name itself suggests, adds to the mood of the scene. Say a person is sitting on a rock and throwing pebbles into a pond. The sound of the pebble hitting the water is contextual. However, if one adds the sound of wind, it adds to the feeling of the character's despair. It enhances the person's feeling of sadness. This kind of sound is called descriptive sound. **Commentative** sound is added by the programme producer to add to the overall impact. Say two people begin to fight over their chance near a water-pump. The cackle of hens that are interspersed during the fight is commentative sound.

Music and sound effects serve to provide transition between scenes. Intelligent sound designers use sound as a transition. Say a scene is ending with the heroine's father declaring that the daughter will go abroad to study. The sound of an aircraft faded in at the end of the scene suggests that the heroine has flown abroad without having to show an aircraft. Sound, therefore, provides transition and continuity between scenes linking them.

Sound effects are extensively used in radio programmes to compensate, augment or to add realism. A whole art of *manually creating and recording sound effects* or *foleying* goes into this. For example, walking on dry leaves (by stamping on a few dry leaves), horses galloping (using dry coconut shells) or even munching of biscuits (by actually munching them) etc can all be recorded in the studio using simple techniques. The art of recording these sound effects in the studio is called **foleying**.

Music is an important tool in the hands of a radio producer. Music helps to draw the attention of the listener to the programme through emphasis. Listen to some radio station signature tunes to understand this. The name of the radio station is repeated between two breaks to inform the listeners that they are on a particular radio station. Music is also used in the background to emphasise or intensify action, set pace, unify transition, indicate time or evoke a mood or as foreground music that is diegetic, like someone playing an instrument in shot or miming to a playback.

Music of a particular kind evokes similar feeling universally. Music helps the producer in releasing feelings of disgust, love, hurt, sadness, joy or even restlessness. It is the choice of the music instrument, the rhythm and tempo that create the desired effect. The signature tune of a newscast is totally different from one of a soap opera. The instruments and tempo of music used in newscasts, game shows, and contests is totally different from the one in melodramas and romantic episodes.

9.8 Mono and Stereo Sound

One can record sounds in two ways: mono and stereo. Mono or monophonic sound is recorded or created on one single channel. In effect, what it means is that even if you have two speakers, you listen to the same thing on both the speakers. In this case, the audio signal that is recorded or created is passed through a single channel to both the speakers. Mono recording is best used when making just talk programmes. Mono sound is mostly used in telephony, where the emphasis is on the speech rather than the aesthetics of the conversation. One single microphone is enough to record mono audio.

However, when one is required to produce programmes with various sounds including speech, music and sound effects, stereo recording is preferred. Different sounds are recorded in different tracks on two different channels. Therefore, when you listen to a music track recorded on stereo, you can listen to different instruments on different speakers. For example, the *tabla* plays prominently on one speaker while the violin and *sitar* play on the other. Stereo sound gives one a more natural listening experience than mono sound. To record stereo sound one might require more than one microphone. One advantage with stereo sound is that it provides a spatial illusion. Some sounds may appear to be coming from the right side, from near and some from far and on the left.

Refer to the chapter on Recording Hardware and Field Recording to know more about mono and stereo microphones.

9.9 Let Us Sum Up

In this chapter, you learnt about the basics of sound and how it is propagated. You also learnt about its properties and components. Remember the differences between pitch and volume. The frequency of a particular sound has an impact on its pitch.

We then learnt about the three different kinds of sound that is speech, music and sound effects. Creating sound effects in a studio is called foleying.

We then proceeded to learn about how sound effects and music contribute to creating the overall mood of a radio programme. We also learnt the basic differences between mono and stereo sound.

In the next lesson, we will understand about analogue and digital audio, the differences between them and their applications.

9.10 Feedback to Check Your Progress

1. What are the key characteristics of sound?
2. What is the relationship between Pitch and Volume?
3. Explain attack, sustain and decay of sound. Give examples for fast, medium and slow attack of sound. Similarly, give examples for slow decay of sound.
4. What are different types of programme sound? Explain in detail giving examples?
5. Explain Mono and Stereo sound.

9.11 Model Answers to Activities

Activity 9.1

Once you go to the URL www.mcsquared.com/wavelength.htm you will see a field asking you to enter the frequency for which the website will give you the wavelength. Think of a radio station which you listen to often. However, please remember that you will be required to enter the value in Hz, whereas the stations' frequency will be given in Mega Hertz. Remember that Mega is equal to 1000 Kilos and 1 Kilo is equal to 1000 units of whatever you are measuring. Therefore, 90.4 Mega Hertz is equal to 90.4 multiplied by 10 to the power of 6. Try entering in different range of frequencies to see what are the wavelengths, whether they are in inches, feet or meters.

Activity 9.2

Notice the change in the sounds each of the cups creates. The cup that is empty creates sound with the highest pitch, whereas the one completely filled with water creates sound with the lowest pitch. Now, repeat the experiment by tapping the cups a bit harder. You will notice that while the pitch of sound emanating from each cup remains the same, the volume gets louder. You can also try out this experiment with glass bottles.

Activity 9.3

Notice the change in the sounds each of the cups creates. The cup filled with sand produces sound that is uniquely different from the one with water. That is because the harmonics set off by the two cups are different.

Activity 9.4

Try creating sounds with different objects at varying rhythms. Once you have created a rhythm, try increasing and decreasing the tempo. You are expected to notice the difference between rhythm and tempo and how the increase or decrease in tempo can change the rhythm. The same principle is used in editing software.

UNIT 10

Analog and Digital Audio

Structure

10.1 Introduction

In the previous Unit, you have studied 'the basics of sound'. As we now understand, sound is a form of vibration that travels through the air or another medium and can be heard when it reaches our ear. Sound is a form of a wave of compression and rarefactions, similar to the ripples on a pond, when a stone is thrown in it. One of the characteristics of sound is 'frequency', which is the number of vibrations per second. A human being can hear sound of frequency lying in the range 20 Hz to 20,000 Hz. In acoustical terms, audio refers to 'sound' or 'reproduction of sound' and audio work involves the production, recording, manipulation and reproduction of sound waves. In this Unit, we will discuss about analogue and digital audio, their characteristics and compression of audio. About 8 hours should be devoted to understand and learn this Unit.

10.2 Learning Outcomes

After completion of this Unit, you will be able to:

- define analogue and digital audio, their characteristics, sampling and quantization, bandwidth, nyquist criteria
- explain frequency response characteristics, signal-to-noise ratio, audio compression and different file formats, open source and proprietary codec
- calculate audio file size and space required for its storage on hard disk or CD.

10.3 Definition of Analogue and Digital

Analogue means continuous and digital implies discrete or discontinuous. For example, a clock; an analogue clock uses the positions of the hands to describe the time. The hands move smoothly around the clock to describe the time of day. As such, an analogue clock is a continuously flowing representation of the time of day. Whereas, a digital clock uses distinct or individual digits and not hands to describe the time and each digit is a specific numerical value that describes the time of day. So it is not smooth flowing, but is characterized by discrete numbers that tell the time.

An analogue signal is a continuous signal in which the information changes as a response to certain changes in physical phenomenon. In other words, we can say that it is a continuous signal, where the time varying feature of the signal is a representation of some other time varying quantity, for example sound or voice we hear. So, an analogue audio signal is a smooth continuously flowing representation of music or sound.

The word *digital* implies something that uses a digit or number to describe something. Digital signal is a non-continuous signal having discrete values. It has only two values – On or Off or Ones and Zeros, just like a light switch in our house which is either in ON position or OFF position. They change in steps. This is expressed using two digits 0 and 1, called binary numbers. Each '0' or '1' is called a 'bit', which is an abbreviation of the term 'binary digit'. So, digital audio signal is represented by multiple distinct events, also known as digital samples.

As such, any analogue signal is represented by a waveform with continuous range of values whereas digital signals are discrete time signals with discontinuous values to represent information as shown in Figure 10.1.

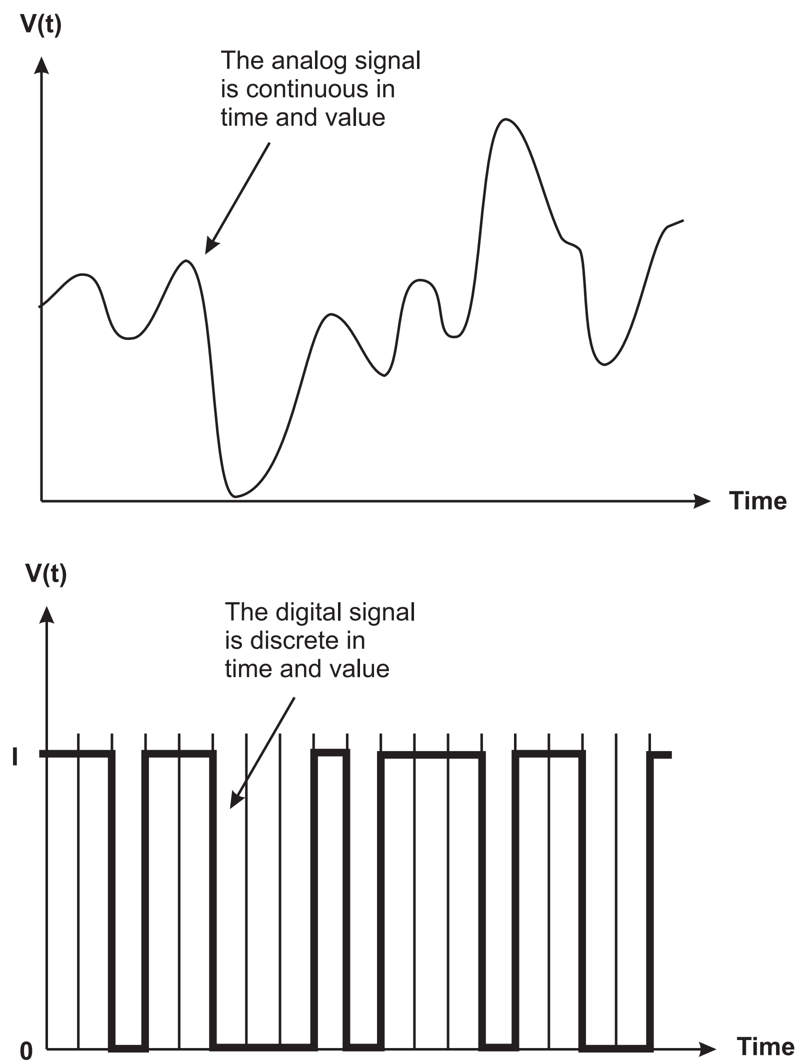


Figure 10.1: *Analogue and digital signal*

As shown in Figure 10.1, analogue signal varies continuously in amplitude or height as time progresses, like electrical signal. Analogue signal can assume any value whereas digital signal can take only two definite discrete values, zero and one. Anything different from these values is discarded.

10.4 Analogue Audio

The dictionary meaning of 'audio' is 'hearing or audible sound'. Sound is nothing but pressure waves of air. Sound is just a vibration. When we beat a drum or pluck a guitar string, it starts vibrating. When it moves outward from its resting position, it squeezes air molecules into a compressed area, away from the sound source. This is called compression. As the vibrating membrane or string moves inward from its normal resting position, an area of lower than normal atmospheric pressure is created called rarefaction. So sound waves are successive areas of air compression or rarefaction. The areas of compressed and rarefied air move out from the sound source in the form of sound wave at the speed of sound, which is nearly 340 meters per second. It arrives at our ears in the form of periodic variations in atmospheric pressure called sound-pressure waves. Our eardrums also vibrate to match the air pressure and this sensation is transmitted to the brain as sound. As such, if there was no air, no sound would be heard. In the waveform shown in Figure 10.2, the horizontal axis represent time and vertical axis represent pressure. The initial high pressure or compression is followed by low pressure or rarefaction, which ultimately dies down.

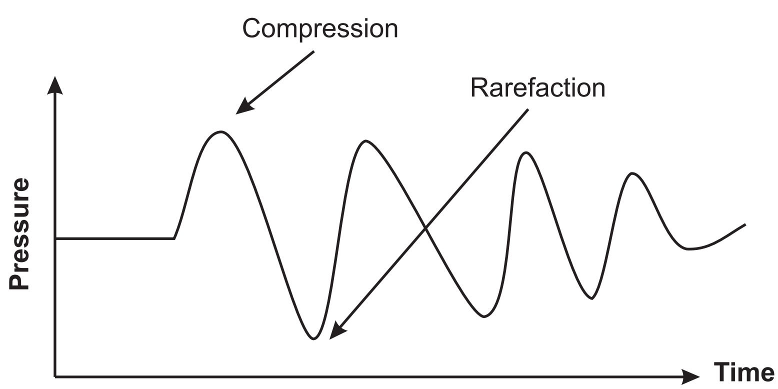


***Figure 10.2**: Analogue sound wave*

As such, analogue audio is a representation of a series of sound signals that change continuously and is analogous to the air pressure waves of the sound. It is a representation of the intensities of those waves in a different form, such as voltage.

Activity 10.1

Explain the differences between Digital Audio and Analogue Audio in about 200 words.

10.5 Characteristics of Analogue Audio

We have already seen that waves have three main characteristics — wavelength, frequency and amplitude. In particular, analogue signals have three main characteristics which define them. These are:

- Amplitude (a measure of how loud they are),
- Frequency or wavelength (a measure of how often they change),
- Phase

Speech is an example of an analogue signal. These characteristics distinguish one waveform or signal from the other. These are as shown in Figure 10.3.

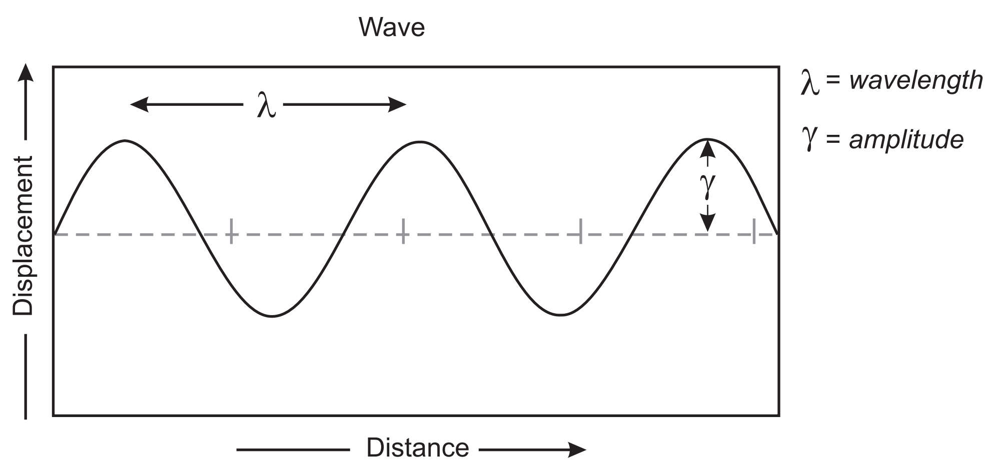


Figure 10.3: *Characteristics of analogue audio waveform*

In the previous Unit, we have already studied what wavelength, frequency and amplitude are. Let us now proceed to understand the term Phase.

10.5.1 Phase

Phase is the rate at which a signal changes its relationship with respect to time. It is expressed as degrees. One complete cycle of a wave begins at a certain point, and continues till the same point is reached. Phase shift occurs when the cycle does not complete, and a new cycle begins before the previous one has fully completed, as shown in Figure 10.4. This implies that there is a time delay between the two waves.

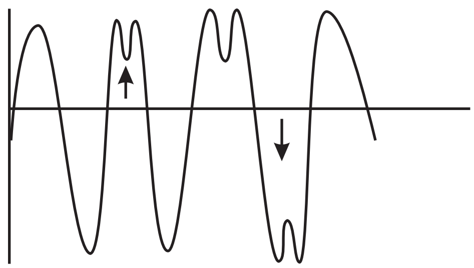


Figure 10.4: *Phase shift of waves*

10.5.2 Frequency Response

Frequency response measures the output level of an audio device at different frequencies 20–20,000 Hz range. It is a series of level measurements made at different frequencies and results displayed on a graph showing level vs. frequency. The graph represents how a device will respond to audio frequencies and how it will affect an audio signal. If the measured signal level is same at all frequencies, the curve will be a flat, straight line from left to right known as a flat frequency response curve. This indicates that the device passes all frequencies equally. So for an audio system, one of the objectives of frequency response analysis is to test the reproduction of the input signal with no distortion.

10.5.3 Mono and Stereo Audio Signal

As explained earlier, we cannot hear sound in vacuum because there is no air to carry the sound waves. The only reason we hear a clap or a bird sing or a guitar play is that those audio sources cause a wave of air to travel to our ear. The sound waves coming from all directions in open air cause the eardrums to vibrate at the same frequency as the sound waves that the source produced. This vibration causes the sound that we hear. In real life, we hear natural sound or audio coming from multiple directions all the time. Our brain converts these sounds into a 3-dimensional audio image that helps in determining from where the sound originated and to judge its distance. This gives rise to the concept of directional or stereo sound and non-directional or mono sound formats.

Mono and Stereo are two classifications of audio signal format, how it is recorded and reproduced. The key difference between the mono and stereo has to do with the use of channels to record or reproduce the sound. Mono describes a sound that is only from one channel while stereo uses two channels. A mono sound signal contains no directional information whereas a stereo signal allows you to distinguish which sound is coming from which direction, which is very similar of being in the same room as the sound was created. Stereo sound provides listeners with a much more natural experience as compared to mono where the sound comes from a single direction. Stereo signal produces a spatial magic by creating the illusion that you are in the middle of a three-dimensional sound source. Stereo audio sounds are clearer than mono, and our brain can detect distance and depth better.

In Mono systems, the signal contains no level and arrival time or phase information that would replicate or simulate directional information. A stereo sound signal contains synchronized directional information from the left and right aural fields. Consequently, true stereophonic sound systems have two independent audio signal channels, one for the left field and one for the right field. The left channel is fed by a mono microphone pointing at the left field and the right channel by a second mono microphone pointing at the right field. The signals that are reproduced have a specific level and phase relationship to each other, so that when played back through a suitable reproduction system, there will be an apparent image of the original sound source.

10.5.4 Signal-to-Noise Ratio

An analogue signal is continuous. It can change at any rate. But, its main disadvantage is that these are prone to various kinds of degradations like noise and distortion, which change the signal waveform which might be quite distracting to hear. Signal-to-noise ratio, S/N or SNR is a measure of degradation level of the audio signal. It is a measure of the level of the audio signal compared to the level of noise present in the output signal. It is a measurement that describes how much noise is in the output w.r.t. the signal level. In other words, S/N is a measure of signal strength relative to background noise. The ratio is usually measured in decibels (dB).

If the incoming signal strength V_s is in micro-volts and the noise level V_n is also in micro-volts, then the signal-to-noise ratio, S/N, in decibels is given by the formula

$$S/N = 20 \log_{10}(V_s/V_n)$$

Ideally, V_s is greater than V_n, so S/N is positive. The higher the S/N, better it is. For example, a signal to noise ratio of 100dB means that the level of the audio signal is 100dB higher than the level of the noise and it is better than a signal output with a S/N ratio of 90dB. For reliable communication, signal level should be much higher than the noise level at the point of reception.

The primary disadvantage of analogue signals is the noise i.e. random unwanted variation of the audio signal. As the signal is transmitted or electronically processed, at each step some noise due to electronic circuitry or signal path is introduced. This noise is additive and the signal degrades progressively with the result that the S/N ratio deteriorates and in the extreme case the signal can be overpowered by noise. Noise can show up as 'hiss' and inter-modulation distortion in the audio signal. This degradation is impossible to recover, since there is no way to distinguish the noise from the signal as amplifying the signal to recover attenuated parts of the signal amplifies the noise also. The solution lies in going digital, since digital signals can be transmitted, stored and processed without introducing noise.

Activity 10.2

i. Explain the difference between Mono and Stereo signal.
ii. Define amplitude, frequency, time period and wavelength of an audio signal.
iii. Find time period and wavelength of a 10 kHz (10,000 Hz) audio signal.

10.6 Digital Audio

As mentioned earlier, digital signal is non-continuous with discrete values. It has only two values – On or Off or Ones and Zeros called binary. The analogue audio, which is a continuous signal is measured at specific time intervals and its amplitude at each of these points stored. This results in a string of numbers which depict the waveform in its development over time, rather than representing it by the continously changing property in an analogue recording medium. Digital audio refers to encoding of audio signal in digital form rather than in analogue form. For this, analogue audio signal is passed through an analogue-to-digital (A/D) converter and is then encoded and digitized or converted to a digital signal. The instantaneous voltage level of analogue audio is sampled or measured at an instant and then these samples are encoded into a stream of zeros and ones in binary format that digitally represents the voltage level at that instant. At reception point, the digital signal is converted back to analogue audio with the help of digital-to-analogue converter.

10.7 Characteristics of Digital Audio

Similar to analogue audio which has two characteristic frequency (time component) and amplitude (signal level), digital audio also has two characteristics – sampling (which represents time component) and quantization (represents signal level).

10.7.1 Sampling

Sampling is the method of converting analogue information to digital data. When analogue audio is converted into digital form, samples of changing audio waveform are taken at specific intervals and these samples of signal level are converted into a binary stream based on its voltage level and stored for further processing or reproduction. This process is called digitization, which uses sampling to store data from an analogue waveform. Once a reading is taken, this value is stored and held until the next sample. This is known as sample-and-hold and is common in most digital audio systems. The reading of signal level is done at a rate fixed by a sample clock within the A/D converter and this is known as the sampling frequency or 'Sampling Rate'. Figure 10.5 shows digitized audio.

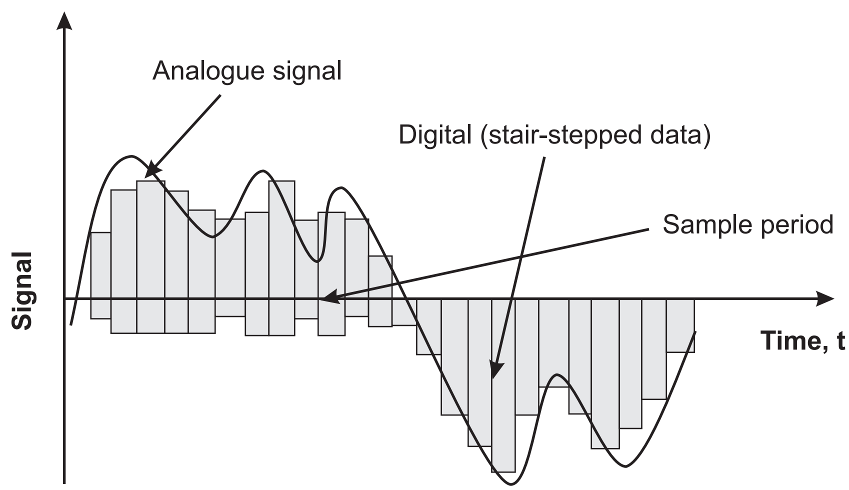


Figure 10.5: *A digital signal*

Here 'Y' axis represents signal voltage and 'X' axis represents time. Samples of analogue signal are taken from time to time and each sample is converted into a number based on its voltage level. How frequently samples are taken or captured is called "Sampling Rate". It is the number of times per second that the analogue signal is measured. For example, if a sampling rate of 20,000 Hz is used, this means in one second 20000 points will be sampled and it corresponds to 1/20000th of a second. During sampling process, the analogue signal is sampled at time intervals determined by the sampling rate and a binary-encoded word is generated equivalent to analogue voltage level at that point. This process is repeated at next sampling interval continuously. During the reverse process i.e. digital-to-analogue conversion, these binary numbers will be converted back into voltages. So, the resulting analogue audio waveform will not be perfect replica of original signal because during analogue-to-digital conversion, all the points of analogue signal were not taken up. Only samples were taken up. Any values that existed between sample points would be suppressed during digital to analogue conversion.

As such, the more samples we take i.e. higher the sampling rate, the more perfect will be the analogue signal produced by digital-to-analogue conversion. But, the sampling is directly related to time component, which subsequently determines the overall bandwidth of the system. The higher the sampling rate, higher is the bandwidth range and more storage space is required to store the resulting digital data.

If the sampling rate is too high, the quality of output will be very close to the original but it will require more storage space. In case the sampling rate is too low, the output quality will be bad. To strike a balance, best sampling rate is decided with the help of the **Nyquist Sampling Theorem,** according to which, "for accurate reproduction of signal, the sampling rate must be at least twice the highest frequency or the bandwidth of the source signal that is to be represented". We know that audible range is 20 Hz to 20 kHz. Therefore, in audio systems, we need to use a minimum sampling rate of at least 40 kHz or 40,000 samples per second. However, due to design considerations, a sample rate of 44.1KHz is used. An audio CD has a sample rate of 44.1 kHz or 44 kHz for short.

In view of this, all high frequencies greater than the sampling frequency must be removed before start of sampling, otherwise some error frequencies would enter into signal path causing harmonic distortion, which is called **Aliasing**. For example, if we take 30,000 samples per second, we can capture frequencies upto 15,000 Hz only. Any frequencies higher than the Nyquist frequency (30 kHz in this case) are perceptually "folded" back down into the range below the Nyquist frequency. This effect is known as Foldover or Aliasing. The approximate new alias frequency can be calculated as:

Alias = sampling frequency - input frequency

For example, if the audio signal contained an input frequency of 22KHz sampled at a rate of 40KHz, the sampling process would misrepresent that frequency as 18 KHz (40KHz – 22 KHz), a frequency that might not have been present at all in the original signal.

So, aliasing can result in addition of frequencies to the digitized audio signal those were not present in the original signal, and unless we know the exact spectrum of the original signal there is no way to know which frequencies actually belong to the digitized sound and which are the result of aliasing.

As such, to eliminate aliasing frequencies, it's essential to use a low-pass filter before the A/D conversion stage to remove any frequencies above the Nyquist frequency. This would allow all the frequencies upto sampling frequency to pass and block all frequency above this. But, it is impractical to design such a tight filter with infinite attenuation above or below a certain cut-off frequency. As such, a sample rate is chosen which is usually above the theoretical requirement e.g. a sampling rate of 44.1 kHz is chosen for audio CDs to accurately reproduce signal in audible range upto 20 kHz.

10.7.2 Quantization

Quantization is the process of converting continuous analogue audio signal to a digital signal with discrete numerical values. It represents the amplitude component of the digital sampling process during analogue-to-digital (A-to-D) conversion of the signal. This is the number of digits in the digital representation of each sample. This process of converting voltages to numbers is known as **Q*uantization*.** These numbers are expressed as a string of binary digits (1 or 0). So the voltage level of analogue signal at discrete sample points (during digitization) is translated into binary digits or bits for digital storage. Thus, the value of each sample will be stored on a fixed-length variable basis – 8 bit or 16 bit. If we use 8 bits, the lowest value will be zero and the highest will be 255 (2^8 = 256 levels). If 16 bits are used, the lowest value will be zero and the highest 65,535 (2^{16} = 65536). The number of bits used to represent the number determines the resolution with which we can measure the amplitude of the signal. The higher the bit size, better the quality but more storage space will be needed. Using a 16 bit variable will require twice the storage space than 8 bit variable as the file size will be doubled, but quality will be far better. Audio CDs use 16 bit resolution. The size of an uncompressed audio file depends on the number of bits processed per second, called the **Bit Rate** and total time duration of the audio signal. The bit rate depends on the sampling rate and bit resolution i.e. no. of bits used. Usually,

Bit rate (Bits per second) = bit resolution x sampling rate, and

File size (in bits) = bit rate x recording time or duration of audio signal

For example, 1 second of mono audio signal with 16 bit resolution will have a file size of approx. 88 Kilobits. (16 bits per sample x 44000 samples per second x 1 second = 704,000 and 704,000 / 8 bits per byte = 88,000 bytes H” 88 KB).

For stereo, total bit-rate will be 88 x 2 = 176KB/second and an hour of CD-quality stereo audio file would be 176 KB/sec x 3600 seconds/hour = 633,600 KB H” 634 MB, which is about the size of a CD.

An analogue signal is continuous and during sampling, each sample is rounded up to the nearest value, which in turn deviates from the original source signal. This results in quantisation errors, which produce audible noise known as quantisation noise in the output. It is unavoidable, but it can be reduced to an acceptable level by using more bits to represent each number.

10.7.3 Bit Error Rate

When a digital signal is transmitted over a communication channel, the bit stream may be effected by distortion and noise of the path. It is denoted in terms of **Bit Errors**. The number of bit errors is number of received data bits that have been altered due to noise and distortion etc. while passing through a communication channel. Bit Error Rate (BER) is the number of bit errors divided by the total number of bits transferred during a fixed time interval. It is a measure of the performance of the communication channel.

10.7.4 Dither

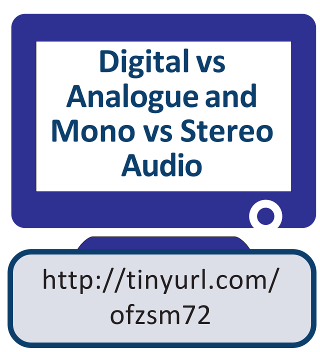
During sampling and quantization, the amplitude of audio signal is sampled at specific rate and rounded off to nearest discrete value. This results in quantisation errors in the form of audible effects of adding low level harmonic distortion in the encoded signal, which is called quantisation noise. It is possible to make this quantization noise more audibly acceptable by adding a small noise-like signal to the original signal before quantization. A common method of reducing quantisation noise is a technique called **Dither.** A very low-level noise is added to the signal prior to analogue-to-digital conversion. This noise forces the quantisation process to jump between adjacent levels at random. This makes the digital system to behave as if it has an analogue noise-floor. This helps the A/D circuit to detect whether the lower-level signal is closer to "0" or "1", thereby making the quantization error independent of signal level. This makes the audible effect far more natural on the human ear. Dither is not generally considered necessary at higher bit depths, such as 24-bit and above, as the human ear cannot hear quantisation errors at this quality. However, when converting to 16-bit and lower, dither becomes necessary to maintain high audible quality.

10.7.5 Jitter

Jitter is the undesired deviation from true periodicity of a periodic waveform, while passing through an electronic circuit. It can be due to several reasons such as error in the clock, which is producing timing pulses or power supply and even data stream itself due to inter modulation. Analogue-to-digital or digital-to-analogue circuits are synchronous digital systems each controlled by a clock which carries timing information. The clock has to be precise and accurate. The two clocks can never be exactly the same and so timing pulse or frequency of the clocks will vary slightly or miniscule. This results in deviation of discrete samples from the precise sample timing intervals. This is called jitter. It produces noise and degrades the sound of a digital audio system.

As such, during the conversion of audio signal from analogue-to-digital and digital-to-analogue, due to jitter, the samples are taken at non-uniform time intervals and the variation of time between them causes error in the signal reproduction.

Hope, you have already got an idea regarding analogue and digital audio and the differences between both of them. Here, you should watch a video on Digital Analogue Audio Mono Stereo Differences. The link for this video is - http://tinyurl.com/ofzsm72. It will help you to understand the topic.



Activity 10.3

i. What is Nyquist criteria for sampling rate? What will happen if sampling rate is too high?

ii. If sampling rate of 44100 Hz is used, what will be the distance between each sampling point?

iii. If a CD uses 44100 Hz sampling rate with 16 bit resolution, how much storage space is required for storing a one hour stereo audio?

10.8 Compression and Audio Codec

As discussed in Section 10.7.2, the size of an uncompressed audio file after quantization depends on the number of bits processed per second and total time duration of the audio signal which in turn depends on the sampling rate and no. of bits used. An hour of CD-quality stereo audio file with 16 bit resolution and 44.1 kHz sample rate would be 44100 *x* 2 *x* 2 *x* 60 *x* 60 = 635 MB size, which is about the size of a CD. Thus, uncompressed digital audio files require a large storage space. It is often required to make an audio file size smaller to optimise the storage capacity and data transfer rate while downloading audio files via the internet.

Reduction in size of data or file in order to save space or transmission time is called Compression or audio compression. It helps in storage or transmission of same amount of data in fewer bits, thus making the transmission of the data faster. Compression falls into two main categories: Lossless Compression and Lossy Compression. Compression is lossless, if the received data can be restored as an exact replica of the original. The decompressed file and the original are identical. In Lossy compression, file size is reduced by removing some of the data. This causes a reduction or loss in audio quality during the compression or decompression process. Lossy compression is used mainly for audio and video files because the loss in data quality is not easily recognised by the human ear and is imperceptible.

10.8.1 Audio File Types/Formats

File format is a specific way to encode data or information that is to be saved as a computer file. Without a format specification, a file is just a meaningless string of ones and zeros. The format specifications help the file to be properly interpreted and rendered. A digital audio file is stored in a specific file format or type. These may be compressed or uncompressed formats, which contain waveform data that can be played with audio playback software. There are a number of different types of audio file formats, the most common being Wave and MPEG Layer-3. The type is determined by the file extension (characters after the "." in the file name). Common audio file extensions include .wav, .aiff, .mp3 etc. The encoding for a specific type of file format is done with the help of codecs. Codec is a program or algorithm that encodes and decodes data to convert a file between different formats. For example, ".wav" file can be encoded with the "PCM" codec and ".mp3" file uses "MPEG Layer-3" codec. Thus, a codec performs the encoding and decoding of the raw audio data while data is stored in a file with a specific audio format. The audio file formats can be grouped as:

- Uncompressed audio formats, such as WAV, AIFF. WAV is the standard audio file format used mainly in Windows PC environment, whereas AIFF format (Audio Interchange File Format) is used by Apple for the Mac.
- Formats with Lossless compression such as WMA (Windows Media Audio lossless)
- Formats with Lossy compression such as MP3 (MPEG Layer-3), MP4.

10.8.2 Open and Proprietary Formats

An *Open* format is one where the description of the format is available to all the users. For example ASCII, PDF, .Doc, HTML are the open file formats. As the format definition is freely available, anyone can in principle write software to access data stored using that format.

Whereas, a *Proprietary* format is one that is owned by an individual or a company/corporation. For example AutoCAD's .dwg drawing format, the MP3 MPEG Audio Layer 3 format and Adobe Photoshop's .psg image format. Most proprietary formats are closed, meaning that the definition of the format is not available to the public. This means that data stored in the format can only be accessed using the format owner's software.

Activity 10.4

What are different types of file formats? How do Open source and Proprietary Formats differ?

10.9 Let Us Sum Up

In this Unit, you have learned how to distinguish between analogue and digital audio. Further you have also learned the various characteristics of analogue audio, i.e. amplitude and frequency, as well as the characteristics of digital audio, i.e. sampling and quantization. Further, you have learned with respect to digital audio, how digital audio files can be compressed using a variety of file formats and codecs. You should have a basic understanding about differences between open and proprietary audio file formats as well.

10.10 Model Answers to Activities

Activity 10.1

Analogue signal is a continuous signal which can take any value, whereas a digital signal can take only discrete values such as analogue numbers. Analogue signals are affected by noise and distortion as compared to digital signal. Give comparison.

Activity 10.2

i. Mono signal contains no directional information about sound origination, whereas stereo produces a sense of direction and distance. Stereo adds liveliness and gives a natural experience to hearing music or audio.

ii. Find using the relations:

$$T = 1/f, \qquad \lambda = v/f, \; v = 3 \times 10^8 \text{ m/sec.}$$

Activity 10.3

i. Explain Nyquist Sampling Theorem: Sampling rate must be at least twice the highest frequency of source signal. Also explain Aliasing effect.

ii. In 1 second, 44100 samples will be taken and it corresponds to 1/44100th of a second. Explain Sampling.

iii. Calculate using bit rate and file size:

Bit rate (Bits per second) = bit-resolution x sampling rate,

File size (in bits) = bit rate x recording time or duration of audio signal

Activity 10.4

Explain uncompressed and compressed file formats i.e. .wav, .aiff, mp3 etc. alongwith ASCII, PDF, HTML.

Mono signal contains no directional information about sound origination, whereas stereo produces a sense of direction and distance. Stereo adds liveliness and gives a natural experience to hearing music or audio.

UNIT 11

Components of the Audio Chain

Structure

11.1 Introduction

In the previous Unit, various aspects related to analogue and digital audio were discussed. In this Unit, we will discuss about the 'audio chain' which shows how physical components of recording and reproduction systems i.e. broadcast equipment are connected together in a studio and the signal flow-path taken by the sound from acquisition to its reception by the listener. You will learn about broadcast microphone theory, equipment for programme production, audio mixers, amplifiers, monitoring speakers, digital audio workstations (DAWs) and field recorders. About 10 hours should be devoted to understand and learn this Unit (including working out various activities).

11.2 Learning Outcomes

After completion of this Unit, you will be able to:

- describe different types of microphones based on transducers
- discuss the features and characteristics of microphones and their polar patterns
- describe various equipment for programme production
- explain the typical audio chain in a broadcast studio

11.3 Audio Chain in a Typical Broadcast Studio

Normally, the programme originates from a studio and is then broadcast using a transmitter, which might be Amplitude Modulated (AM) or Frequency Modulated (FM) mode. The broadcast of a programme from source to the listener involves use of a device to pick up sound i.e. microphones, recording/playback and signal processing equipment, routing of audio signal from studio to transmitter location using a studio transmitter link and finally the transmitter. Figure 11.1 shows a simplified block schematic of the audio chain of a broadcast studio.

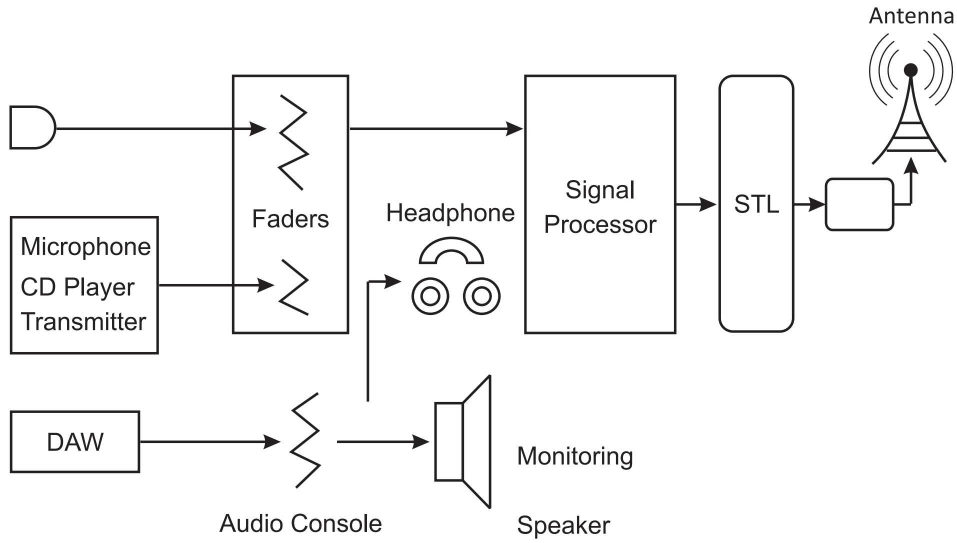


Figure 11.1: *Audio Chain of a broadcast studio*

Audio chain shows linkage of various equipment starting with various sound sources, such as a microphone to pick up announcer's voice, a CD player or a digital audio workstation (DAW), mixer or audio console, signal processing equipment, monitoring speakers and studio transmitter link (STL) for broadcasting. A broadcast studio is an acoustically treated sound proof room for production and broadcast of a programme. The microphone is the first equipment that picks up the voice of announcer or artist and transforms it into an audio signal. All the sound sources like CD players, DAWs and microphones are connected to an audio or mixer console which is used for mixing and controlling the programmes for recording and transmission.

11.4 Microphone

The fundamental purpose of a microphone is to convert sound energy to electrical energy, in which the voltage and current are proportional to the original sound. So a microphone is acoustic to sound transducer that converts sound to an electrical signal. Microphones use electromagnetic induction effect or capacitance effect or piezoelectric effect for converting sound to electrical signal. For this purpose, microphones use a thin membrane known as a diaphragm. The sound waves cause the diaphragm to vibrate and this vibration is converted by a transducer into electrical signals. Microphone output is very low level, which is approximately –70 dBm and it requires a boost to make it usable. There are three major parts of any microphone:

- **A Diaphragm** vibrates in accordance with air pressure of sound waves. It must be lightest possible to accurately reproduce high frequency sounds.
- **A Transducer** converts the vibrations of diaphragm into an equivalent electrical signal.

- **A Casing** provides mechanical support and protection for the diaphragm and the transducer and also helps in controlling the directional response of the microphone.

11.5 Types of Microphones

There are a number of types of microphones in common use. These can be divided or classified into two main types:

(i) Classification by type of Transducer – depending on the internal structure and also the method used to convert sound energy into electrical signal.

(ii) Classification by Pick-Up or depending on Directional Properties.

11.5.1 Classification by Transducer Type or Internal Structure

Depending on the internal configuration, the microphones can be labelled into three types:

(a) Dynamic Microphones

(b) Ribbon Microphones

(c) Condenser microphones

i) ***The Dynamic Microphone***

This is also called 'Moving Coil Microphone'. This type of microphone works on the principle of electromagnetic induction. When a coil of wire moves inside a magnetic field an electrical current is generated in the wire. Sound waves cause movement of a thin metallic diaphragm and an attached coil of wire located inside a permanent magnet. When the diaphragm vibrates in response to the incoming sound waves, the coil moves backwards and forwards in the magnetic field. This causes a current in the coil. The amount of current is determined by the speed of motion of the diaphragm. A common configuration is shown in Figure 11.2.

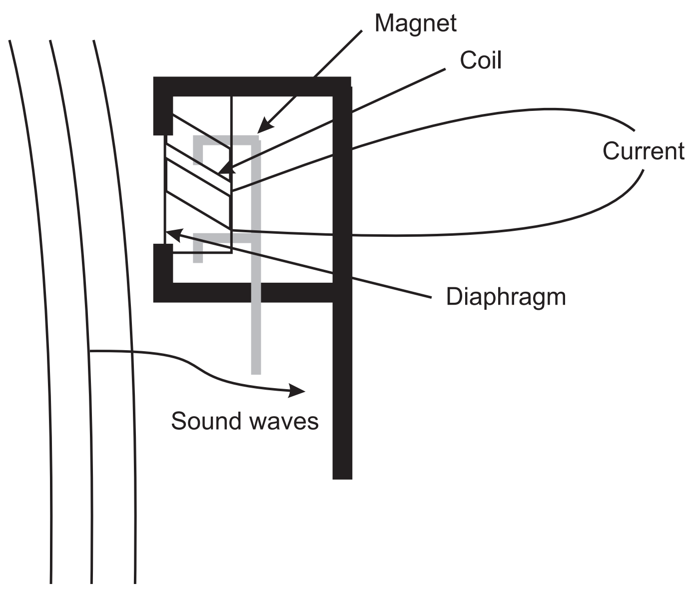


Figure 11.2: *Dynamic Microphone - Block Diagram.*

The response of a microphone to high frequency signals depends upon the moving parts. This type of microphone is relatively heavy, as both the diaphragm and the coil moves. Thus, the frequency response falls off above 10 kHz. However, their resonance peaks around 4–5 kHz and this provides an inbuilt boost that improves speech, singing or vocal intelligence.

ii) ***The Ribbon Microphone***

The Ribbon microphones also operate on the principle of electromagnetic induction to convert sound energy to voltage. In this type of microphone, the transducer is a long thin strip of aluminium foil, which moves within a magnetic field to generate a current and hence voltage. The aluminium foil is also called ribbon, is vibrated directly by the moving molecule of air of the sound waves. As such, no separate diaphragm is required. A common configuration is shown in Figure 11.3.

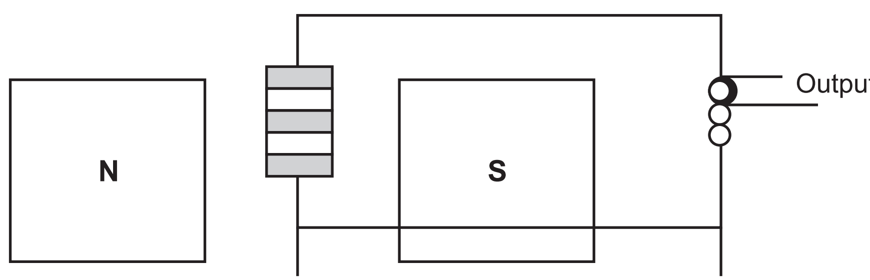


Figure 11.3: *A Ribbon Microphone*

However, this type of microphone has a relatively low output signal, so an output transformer is needed to boost the signal level. The foil's lower weight as compared to a moving coil gives it a smother and higher frequency response to around 14 kHz. As such, such microphones are good for quality studio recording of acoustic instruments.

iii) ***The Condenser Microphone***

The condenser microphone has two electrically charged plates in the form of a thin movable diaphragm and a fixed solid back plate rather than a vibrating wire coil. This makes up an electronic component known as a capacitor or a condenser with positively and negatively charged plates and air in between. The capacitor stores energy in the form of an electrostatic field. Due to sound pressure, the diaphragm moves causing a change in spacing between the two plates and this changes the capacitance resulting in corresponding change in voltage. A voltage is required across the capacitor for this to work. This voltage is either supplied by a battery in the microphone or by an external source called Phantom power. A block of condenser microphone is shown in Figure11.4.

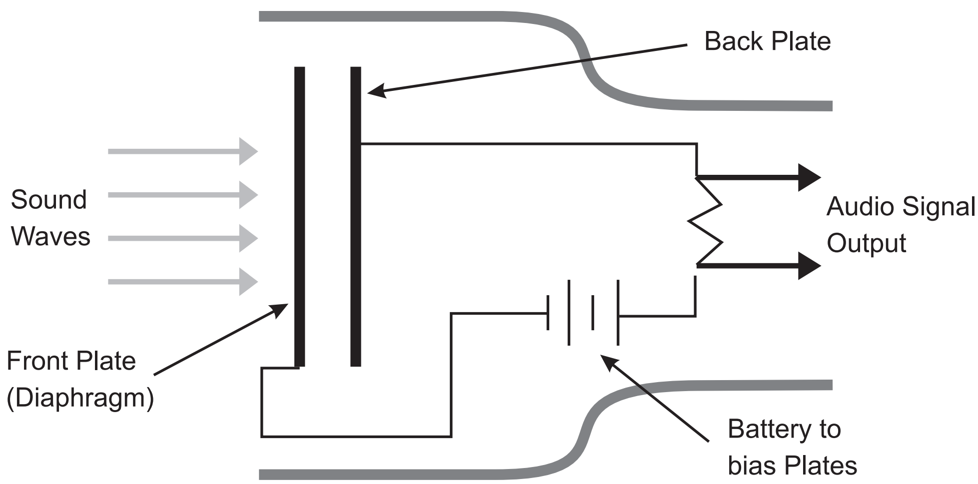


Figure 11.4: *A Condenser Microphone*

A condenser microphone has an omni-directional pattern. To make it directional, small holes are made in the back plate, which delays the arrival of sound at the rear of the diaphragm to coincide with the same sound at the front, which then cancels the sound out. These microphones have very good high as well as low frequency response.

Activity 11.1

i. What is an audio chain?
ii. What is Phantom Power?
iii. What are the major parts of a microphone?

Answer each question in about 50-100 words.

11.5.2 Classification by Pick-Up or Directional Properties

Some microphones pick up sound equally from all directions. Others pick up sound only from one direction or a particular direction. Every microphone has a property of directionality. The factor which determines the directional response of a microphone is the way the diaphragm is exposed to sound. The pickup pattern of microphones describes a three dimensional orientation in space relative to sound sources. The directional response of a microphone is represented graphically. This graph is called a Polar Pattern. The polar pattern shows the level of signal pick-up from all angles and at different frequencies.

In general, pickup patterns fall into following three categories:

(a) Omnidirectional
(b) Bi-directional
(c) Unidirectional

i) Omnidirectional Microphone

An omnidirectional microphone picks up sound equally from all directions. The diaphragm is exposed to the open air on one side only. The air on the other side is enclosed by an airtight structure, so that it is unaffected by the sound. The outside pressure determines the activating force. As the microphone is small compared with the wavelength of the sound being received, it will not obstruct the pressure waves coming from the sides or back of the casing and as such the response of microphone will be the same in all directions. The polar pattern will be as shown in Figure 11.5. These microphones are mostly used for recording of vocal groups or choirs. The major drawback of omnidirectional microphones is their sensitivity to feedback. Due to this, proper placement is necessary for covering a live programme.

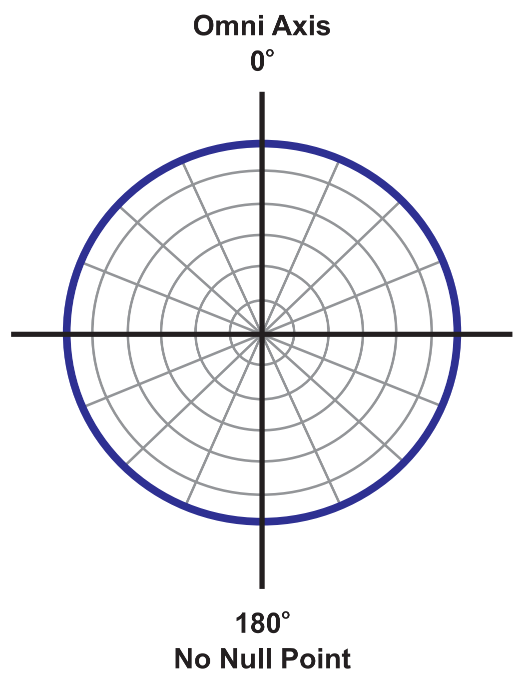


Figure 11.5: *Polar Pattern of Omni-directional Microphone*

ii) The Bi-directional

These microphones pick up sound from the front and the back side of the microphone, but rejects sounds from the left and right sides of the microphone. The sensitivity of the microphone on the sides is low. The polar pattern of such microphones is as shown in Figure 11.6. The frequency response is better on the front side of the microphone. Bi-directional microphones are excellent for capturing a vocal or instrumental duet, and face-to-face interviews using a stationary single microphone. Normally, these types of microphones are optimally positioned above a sound source.

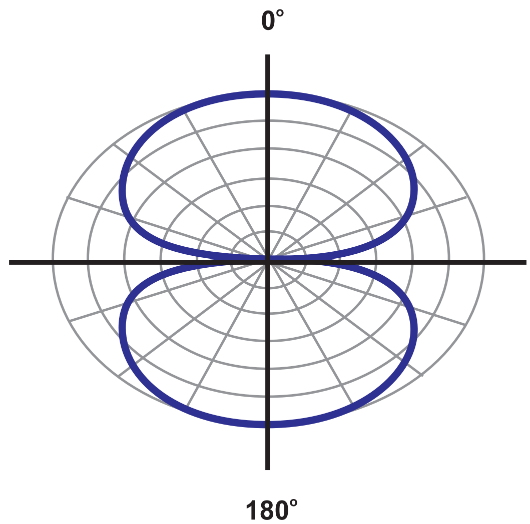


Figure 11.6: *Polar Pattern of Bi-directional Microphone*

iii) The Unidirectional or Cardioid Microphone

This is the most popular configuration for microphones. The Cardioid microphone picks up the sound primarily from one direction i.e. in the front and reduced pickup from the side and the back. This helps in isolating the signal source from the background noise or from other sound sources on the side. They are named cardioid because the polar pattern is heart-shaped as shown in Figure 11.7.

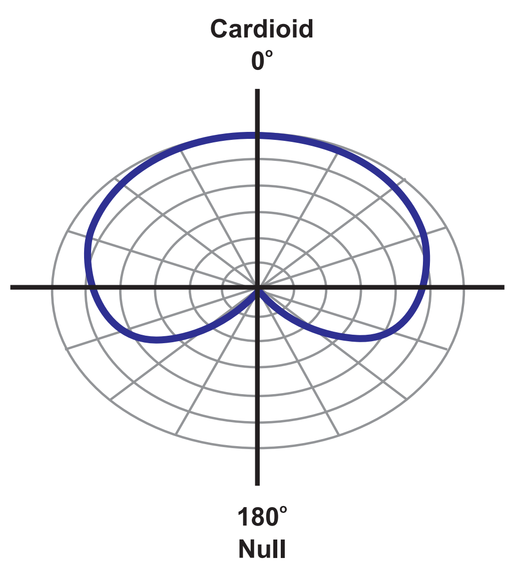


Figure 11.7: *Polar Pattern of Cardioid Microphone*

The cardioid microphone is ideal for general use. Handheld microphones are usually cardioid. Cardioid microphones have different frequency response near to the sound source and at a distance from the sound source. This is called **'Proximity Effect'**. There is a boost in low to mid frequencies as the distance between the sound source and the microphone decreases. Omni-directional microphones do not exhibit the effect. There are many variations of the cardioid pattern such as the hypercardioid as explained below.

iv) The Hypercardioid Microphone

By changing the number and size of the openings on the case of the microphone, the directional characteristics of the microphone can be increased, so that there is even less sensitivity to sounds on the back sides. These microphones are very directional and eliminates sound from the sides and the rear. The polar pattern is as shown in Figure 11.8. Due to the long thin design, these hypercardioids are often referred to as shotgun microphones.

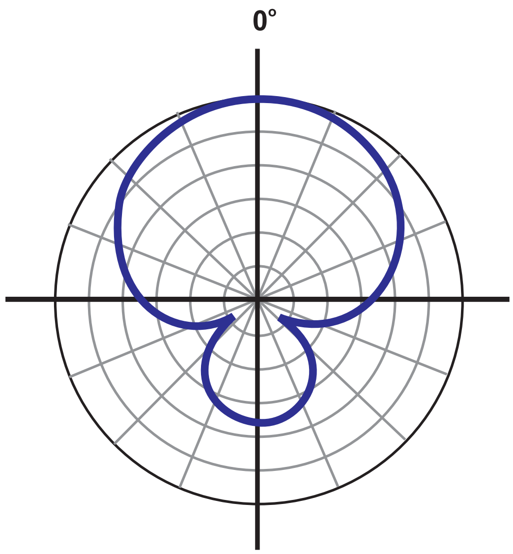


Figure 11.8: *Hyper cardioid Polar Pattern.*

Activity 11.2

i. What are the different types of microphone based on pick-up?
ii. What do you understand by the proximity effect?

Answer each question in about 100-150 words.

11.6 Equipment for Programme Production

Various equipment are available in a studio for programme production and broadcast. The broadcast of a programme from source to listener involves use of studios, audio mixer, playback equipment etc. The equipment meets the requirement of recording, editing, storage and playback. A digital audio workstation with suitable software is utilized for recording and playback facility. The programme may either be live, recorded or field based from OB spot. Basic studio equipment are briefly discussed below.

11.6.1 Audio Mixer

A mixer combines an array of inputs into a few controllable outputs. It is a device which takes two or more audio signals as inputs, mixes them together and provides one or more output signals. It is used for mixing and controlling the programmes. The input signals can be of different levels starting from the microphone. It has the facility to adjust the audio levels of input and output. The mixer provides additional outputs for monitoring, recording and broadcast purposes. Figure 11.9 shows an audio mixer with five inputs and outputs.

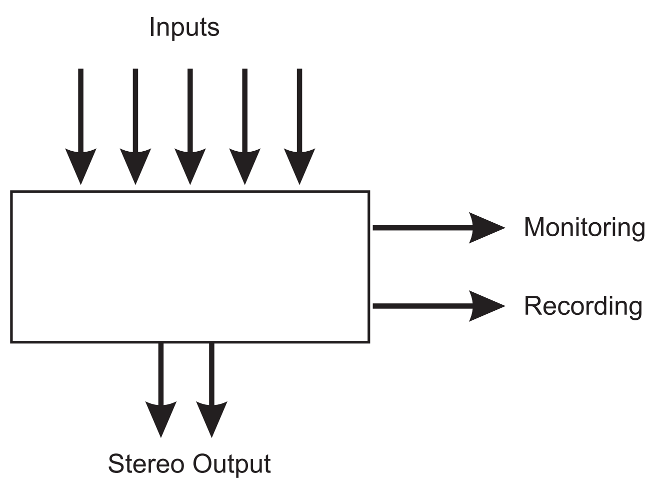


Figure 11.9: *Audio Mixer block diagram*

11.6.2 Amplifiers

Signal amplifiers are the electronic devices that have the ability to boost a relatively small input signal. Amplification is necessary because the desired signal is usually too weak to be directly useful. Amplifiers can be thought of as a black box containing an active device, such as transistors or integrated circuits (ICs), which has two input terminals and two output terminals. Audio amplifiers and audio pre-amplifiers are used to increase the amplitude of sound signals. As explained earlier, the output from a microphone is at very low level (-70 dBm). Pre-amplifiers are used to boost low-level signal from a microphone. The microphone output forms one of the input signals to audio mixer, where it is amplified by a pre-amplifier module and then amplified by a programme amplifier module of the audio mixer, which can be used for recording, monitoring to drive speakers and broadcast purposes. Audio amplifiers used in a broadcasting studio have almost a flat frequency response within a variation of 1 dB with a harmonic distortion of less than 1 %.

11.6.3 Monitoring Speakers and Headphones

A monitoring speaker performs an opposite function to a microphone, i.e. it converts electrical signal into sound wave. Monitoring speakers are provided to monitor the sound quality being recorded and broadcast. The studio monitor should have flat frequency response, so that it reproduces the audio exactly as it sounds, without boosting or reducing any frequencies in the process. Studio monitors may be passive or active. Passive monitors need a separate monitoring power amplifier, whereas active monitors have a built-in amplifier.

Passive monitors are driven by a monitoring amplifier. The monitor output from the mixer is fed to monitoring amplifier which drives the loudspeakers provided in studio and control room. Normal input level to the monitoring amplifier is about -12 dBm in matching condition. The monitoring amplifier and speakers are designed to have a flat frequency response without any sound enhancing effect, so as to faithfully reproduce all the audible frequencies at the same level. Active monitors have built to in power amplifier which drives the speakers. However, placement of speakers also effects the reproduction of sound.

Headphones are also used for monitoring of programmes in the studios. These help in placement of sound source, so that it is not affected by environmental interference from the room. However, monitoring through headphones emphasize low-level sounds. Headphones work on the same principles, which are applicable to loudspeakers.

11.6.4 Digital Audio Work Station

Nowadays, computers are being used for recording and playback of audio. Thus a computer used is called a Digital Audio Workstation (DAW). DAW is a computer system designed to record, edit and playback digital audio. DAW consists of a host computer hardware, professional audio interface hardware, which performs both analogue to digital (A/D) and digital to analogue (D/A) conversion, an audio processing software such as Cubase, Audacity, Adobe Audition etc. Data is stored on hard drives. The computer acts as a host for the sound card and software provides processing power for audio editing. The software controls the two hardware components and provides a user interface to allow for recording and editing. There are no. of advantages in using DAWs for audio production:

- It is easy to handle, longer audio files for recording and editing.
- The audio is recorded on computer hard disk, like audio can be accessed at any point within the recorded file i.e. random access of audio content is possible.
- Non-destructive editing of audio file helps in keeping the original file without any changes.

11.6.5 Field Recorders

Field recording implies recording of audio or other events outside the studio environment. Field recorders are designed for OB recordings and electronic news gathering. A field recorder is a portable digital recorder that runs on batteries, is light weight and creates high-quality audio recording on removable digital media like flash card or internal drive. These can have an inbuilt internal stereo mike and also provision for connecting an external mike on USB port, which can also be used to connect to a DAW in order to transfer the field recording for editing and playback. It is possible to select the recording format like WAV or MP3 and name the file. Some of the field recorders have also in-built facility for editing the file.

Activity 11.3

i. What are field recorders?

11.7 Let Us Sum Up

In this Unit, we have discussed regarding audio chain in a typical broadcast studio. You have also learned the various types of microphones. Further, we have also discussed the equipment needed for programme production, such as audio mixer, amplifier, monitoring speakers and headphones, digital audio workstation, field recorders etc.

11.8 Model Answers to Activities

Activity 11.1

i. An audio studio chain shows linkage of various equipment starting with various sound sources, such as a microphone to pick up announcer's voice, a CD player or a digital audio workstation (DAWs), mixer or audio console, signal processing equipment, monitoring speakers and studio transmitter link (STL) for broadcasting. The microphone is the first equipment that picks up the voice of announcer or artist and transforms it into an audio signal. All the sound sources like CD players, DAWs and microphones are connected to an audio or mixer console which is used for mixing and controlling the programmes for recording and transmission.

ii. Phantom Power is used with condenser microphones. It is a DC voltage (48V), which is connected to a condenser microphone as its diaphragm needs an electric current for it to function like a capacitor.

iii. There are three major parts of any microphone:

(a) A Diaphragm vibrates in accordance with air pressure of sound waves. It must be the lightest possible to accurately reproduce high frequency sounds.

(b) A Transducer converts the vibrations of diaphragm into an equivalent electrical signal.

(c) A Casing provides mechanical support and protection for the diaphragm and the transducer and also helps in controlling the directional response of the microphone.

Activity 11.2

i. Every microphone has a property of directionality. Some microphones pick up sound equally from all directions, while others pick up sound only from one direction or a particular direction. The directional response of a microphone is represented graphically by polar pattern, which shows the level of signal pick-up from all angles and at different frequencies. Based on the pickup patterns, microphones fall into following three categories:

(a) Omnidirectional

(b) Bi-directional,

(c) Unidirectional

ii. Directional microphones (like cardioid microphones) have different frequency response near to the sound source and at a distance from the sound source. This is called 'Proximity Effect' that produces a boost in low to mid frequencies as the distance between the sound source and the microphone decreases.

Activity 11.3

A field recorder is a portable digital recorder that runs on batteries, is light weight and creates high-quality audio recording on removable digital media like flash card or internal drive. These are used for OB recordings and electronic news gathering. These can have an inbuilt internal stereo mike and also provision for connecting an external mike on USB port.

UNIT 12

Studio Acoustics

Structure

12.1 Introduction

For Community Radio Stations, studios are built for programme production, post production and broadcast. These studios are acoustically designed rooms. For achieving best recording and broadcast quality from these studios, all undesirable sound terms as noise must be kept under control. Noise from air-conditioners, outside environment, human-made noise from traffic movement and industrial activities are inappropriate for programme production and recording. Construction of studio with reference to the *noise control* is an important aspect for realising high quality of sound.

For ensuring the recordings to be exact reproduction of original we must understand as to how sound behaves in an enclosed room. *The science of sound is called as* ***Acoustics***. It is important to understand the fundamental of acoustics for understanding concepts of studio acoustics and for controlling noise in the sound studios. In this Unit, you will learn about acoustics of the studios, sound isolation, sound absorption, noise sources, and acoustic treatment.

This Unit will require about five hours of study.

12.2 Learning Outcomes

After going through this Unit, you will be able to:

- describe Acoustics.
- explain the necessity of noise control.
- analyse noise sources.
- discuss isolation and absorption.
- describe and categorise different types of acoustic treatment.

12.3 Studio Acoustics

Acoustics is a science that deals with the study of sound waves. The application of acoustics is involved in sound and noise control. In broadcast industry, acoustics is a very important aspect while planning for studio building and studio area within a building. ***Studio Acoustics*** administers the principles and a process of how sound behaves in an enclosed space. Studio Acoustics mainly deals with the enhancing of sound quality generated inside the studio and minimizing the outside noise entering the studio. A video is produced on the '**Studio Acoustics'** and it is available at http://tinyurl.com/nbyzl64. You may first watch this video and then read the following details on the topic.

To understand Studio Acoustics we must know the basic phenomenon of sound waves propagating in a studio. In Unit 9 we have read about sound wave propagation mechanism. Like any other wave, sound waves also show wave characteristics when they encounter an obstruction in the room. These are reflection, refraction and diffraction. When sound wave strikes the wall surface a portion of it is *reflected,* portion of it is *refracted* and portion of it is *diffracted*. In addition to these phenomena behavioural characteristics like Sound Absorption and Sound Diffusion are made use of in designing the interior of studios. Extent to which each of these phenomena takes place depends upon the structure and shape of the obstacle, and also on the frequency of sound waves. *Figure 12.1* shows the behaviour of the sound wave when it strikes acoustically treated wall.

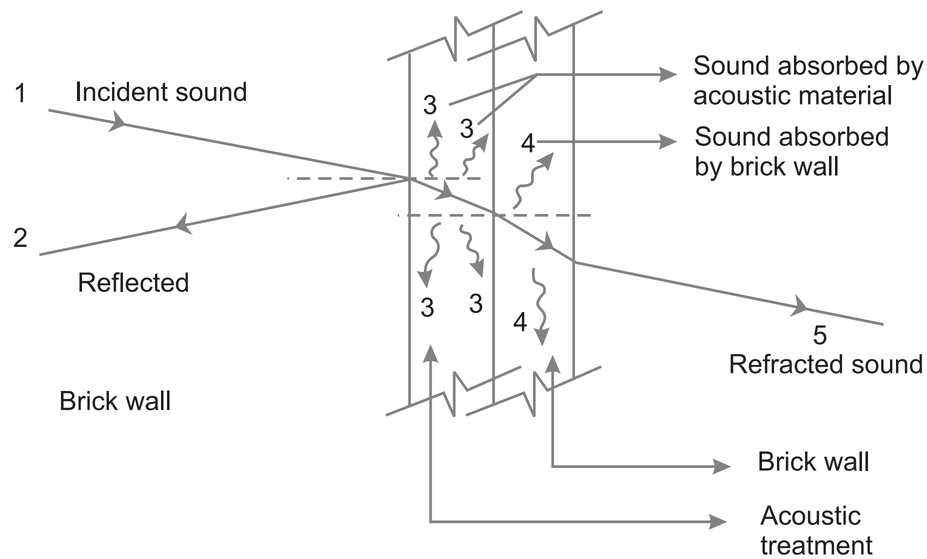


Figure 12.1: *Behaviour of Sound Wave on striking Wall*

Phenomena of sound propagation

Reflection

Bouncing back of sound after striking a flat and hard surface is known as reflection of sound. In close rooms, the sound will reflect and re-reflect till its intensity weakens and it dies completely. A single strong reflection can be heard as an *echo.* When lots of reflections combine, it gives rise to reverberation effect. ***Reverberation Time*** *is the amount of time it takes a loud and short sound to die away by 60 dB drop in loudness.* The reverberation time desired in a room depends on the activity for which room or studio it is designed.

Refraction

Refraction is the change in the direction of travel of the sound by differences in the velocity of propagation. The refraction of wave due to difference in the density of medium is shown in *Figure 12.1 (wave 5)*. The change in the velocity may be due to:

- Density of the mediums,
- Temperature gradient in atmosphere.

Diffraction

The diffraction phenomenon is described as the apparent bending of waves around small obstacles (compared to wavelength) and the spreading out of waves past small openings (compared to wavelength). The diffraction of a wave is shown in *Figure 12.2*.

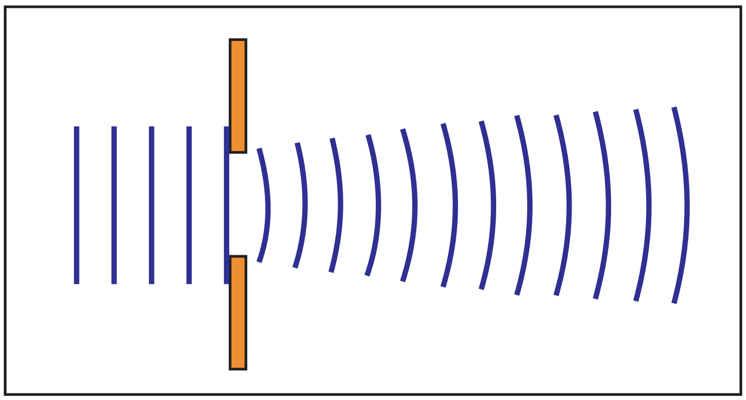


Figure 12.2a: *Diffraction of Sound Wave*
[Wide Gap - Small Diffraction Effect]

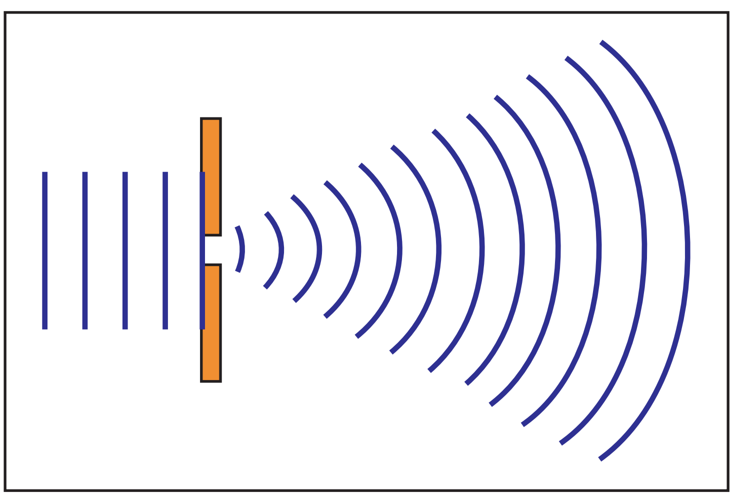


Figure 12.2b: *Diffraction of Sound Wave*
[Narrow Gap - Large Diffraction Gap]

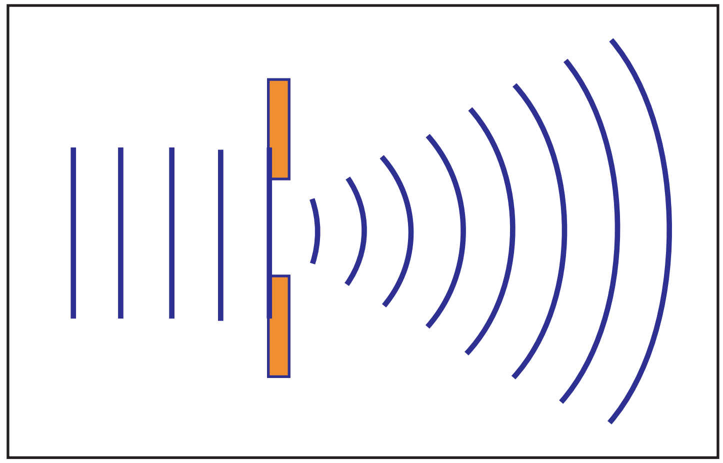


***Figure 12.2c:** Diffraction of Sound Wave*
[Large Wavelength - Large Diffraction Effect]

Reverberation

You have already read about reverberation in Unit 9. It is the collection of reflected sounds from the surfaces in an enclosure like studio. In an auditorium, reverberation is helpful in overcoming the loss of sound waves. However, if reverberation is more, it makes the sound un-intelligible and leads to loss of articulation. Reverberation time is dependent on the volume of the room. Reverberation is due to the sound wave reflections from hard floor, wall or ceiling surfaces. It can be reduced by replacing some of the hard and reflective parts of the walls with soft and absorptive sections of acoustic material.

RT60 = 0.161V / S α (Metric System)

Where

V = Total Volume

S = Surface Area

α = Absorption Coefficient

Note It

Acoustics in broadest sense, is the science of sound.

Studio Acoustics is defined as those qualities of a space that affect the production, transmission and perception of music or speech.

12.4 Noise Sources

Noise can be classified into two types depending upon the medium used by noise for propagation. Noise which propagates from the source via air as a medium is known as ***Air-Borne Noise***. Noise that travels a part of its journey by means of vibration of a solid structure is known as ***Structure-Borne Noise.*** Whenever there is a physical connection between vibrating machine and supporting structure, vibration energy is transmitted into the supporting structure from which it may be radiated as audible sound or felt as vibration.

Noise in a studio can originate from:

- Outside the studio
- Inside the studio itself

Note It

Structure borne noise is attenuated by isolation, while airborne noise is reduced by absorption.

Noise originated from outside the building

Main sources of noise from outside of studio are:

- Heavy industries in nearby areas
- Noisy streets
- Unwanted sound from adjacent rooms
- Airplanes, road and rail traffic movements

During planning phase of the studio, these noises can be avoided and minimised by selecting the studio site in a quiet environment. Setting up of studios should be avoided near railway lines, highways, airports and industrial areas. For avoiding noise from the busy street, studios are located at the back side of the building, so that front portion of the building acts as a sound barrier for the studio.

Noise from inside the studio

Noise from inside the studio consists of:

- Air-conditioning noise due to air flow
- Noise from illumination lights
- Noise from cooling fans in Audio Work Stations and other electronic equipments etc.

By providing absorbers and diffusers in the AC duct, we can reduce noise due to airflow. Also always try to use the electronic equipment with low noise type to avoid noise from the illumination lights and cooling fans of equipment.

12.5 Sound Isolation

Sound Isolation is an acoustic treatment for reducing the effects of exterior noise. Main motive of isolation is to reduce the level of sound entering an enclosed space and preventing the transmission of sound energy into adjoining air space. During studio designing, Sound Isolation techniques are used for controlling the sound and noise to the acceptable levels to ensure that the programme recording is free from unwanted noise. Sound Proof Doors, Observation Windows and walls are, therefore, designed after proper calculation of the *Transmission loss* for the material used in studio construction. The actual process of sound isolation involves inserting insulating material into the walls, as well as above the ceiling and below the floor.

Isolation of the studio is required mainly from:

- Footfall, dragging of furniture,
- Adjacent room, corridor noise,
- Air-Conditioner, Diesel Generator and lift noise and vibrations

Sound Isolation from footfall, dragging of furniture etc.

Noise due to footfall, dragging of furniture, falling of object is classified as structure borne noise. Such a noise travels in the framed structure building to long distances. Steel and concrete frame buildings provide a path for such noise and spoil the programme production. For controlling such type of noise, studios are generally constructed as a load bearing building structures.

Sound Isolation from adjacent room or corridor noise

Monitoring in control room and conversation nearby corridors may cause leakage of this sound in a studio. Poor isolation of the partitions and thin acoustic treatment leads to leakage of adjacent room noise and this leakage of sound or noise from such areas may disturb the recording in the studio.

During planning of the studio and associated rooms, proper sound isolation can be designed by keeping the two different sound sources or studios at a distance, so as to minimize sound transmission from adjacent rooms. Also additional isolation is achieved by providing an acoustically treated ***buffer room (Sound Lock),*** at the entrance of the studio, so that corridor noise does not leak to the studio through the entrance door. To avoid leakage through corridors, all the partition walls in the studios should be constructed up to the real ceiling height of the studio.

Control of air-conditioning, diesel generator and lift noise

Noise due to Air-Conditioning Plants can get transferred to the studios as structural borne noise as well as air borne noise. Measures to be taken to minimize such a transfer of noise are explained below:

- Generally, AC plants are installed in a separate building to reduce the transfer of structure borne noise. Apart from their installation in different building, such machines are installed on rubber pads which act as a damper for vibration. AC plant and Ducts are connected via flexible connection to minimize the structure borne noise. Water pipes for condensers are also isolated from walls with proper packing material to avoid transmission of vibrations.
- The supply and return duct of AC plant act as the path for Air borne noise from AC plant. To reduce such noise, entire length of supply and return duct is treated with sound absorbing materials e.g. glass wool and mineral wool.
- Lastly, the speed of the blower is also kept low for controlling the airborne noise from air flow of the AC plant.

For controlling the **Diesel Generator and Lift machine room** noise, designing and planning should be done in such a way, that these machines are either installed in structurally isolated block or in a separate building away from the studio. For reducing the vibration footprint, the Generator and Lift Machine Room is mounted on anti-vibration mounting, such as rubber pads.

As you are aware that sound can travel through any medium, but sound intensity is reduced in the transition from one material to another. The amount of reduction known as ***Transmission Loss*** is related to the density of the wall. The difference of the sound level of sound wave 1 to that of refracted wave 5 as shown in *Figure 12.1* is the transmission loss of sound energy, while travelling from the room to outside environment. Sound isolation generally known as *Transmission Loss* against airborne noise is determined by its mass per unit area.

Transmission Loss (TL) = 20 Log f + 20 Log w

Where

f = Frequency

w = Surface Mass of Barrier

Sound Transmission Class (STC)

The Sound Transmission Class is a rating of the effectiveness of a material to reduce the transmission of airborne sound. Some of the STC values along with their rating are provided in *Table 12.1.* The condition of the room is also explained in the table for the corresponding STC value. The STC rating and Transmission

Losses at different frequencies of different construction material are provided in the *Table 12.2.*

Table 12.1: *Subjective Equivalent of different STCs for Studio*

STC	Conditions	Subjective Rating
< 30	Normal speech heard and understood	Poor
30-35	Loud speech heard and understood; normal speech heard but not understood	Fair
35-40	Loud speech heard but not understood; normal speech faint	Good
40-45	Loud speech faint; normal speech in-audible	Very good - minimum required for studios
> 45	Loud sounds faint	Excellent - design goal for most professional studios

Table 12.2: *STC and Transmission Loss*

Sl. No.	Material	Transmission Loss (in dB)				STC Rating (in dB)		
		125 Hz	250 Hz	500 Hz	1000 Hz	2000 Hz	4000 Hz	
1	Gypsum Board - 12.5 mm	14	20	24	30	30	27	27
2	Brick Wall - 100 mm	31	33	39	47	55	61	45
3	Solid Wood Door - 50 mm with airtight casketing and drop seal	29	31	31	31	39	43	35
4	Laminated Glass - 12.5 mm	34	35	36	37	40	51	39
5	Mineral Fibre acoustic ceiling tile - 12.5 mm	6	10	12	16	21	21	17

12.6 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 from the absorption phenomenon of sound wave in *Figure 12.1* that when sound wave strikes the acoustically treated surface, some of the sound wave penetrate 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 re-transmission or reflection of sound wave from the surface. The absorbing material is required to be selected on the basis of frequency distribution of noise and the purpose of the use of 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 material 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 absorbers 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 act as a trap which is 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).*

Sound Absorption Co-efficient

Sound Absorption Co-efficient is defined as the ratio of sound energy absorbed to that arriving at a surface or medium.

The sound absorption co-efficient indicates how much of the sound is absorbed in the actual material. The absorption co-efficient can be expressed as:

$\alpha = I_a / I_i$

Where

I_a = Sound Intensity Absorbed (W/m^2)

I_i = Incident Sound Intensity (W/m^2)

Total Sound Absorption

The total sound absorption in a room can be expressed as:

$A = S_1\ \alpha_1 + S_2\ \alpha_2 + \ ..\ + S_n\ \alpha_n = \Sigma\ S_i\ \alpha_i$

Where

A = Absorption of the room (m^2)

S_n = Area of the Actual Surface (m^2)

α_n = Absorption Coefficient of the Actual Surface

The Absorption Coefficients of various construction materials at different frequencies are provided in *Table 12.3*. The last column of the table 12.3 provides the *Noise Reduction Coefficient* of the respective material. NRC is the scalar representation of the amount of the sound energy absorbed upon striking a particular surface.

Table 12.3: *Absorption Coefficients*

Sl. No.	Material	Sound Absorption Coefficient				NCR Number		
		125 Hz	250 Hz	500 Hz	1000 Hz	2000 Hz	4000 Hz	
1	Painted Masonry Wall	0.08	0.05	0.05	0.07	0.08	0.08	0.06
2	Gypsum Board -12.5 mm	0.27	0.10	0.05	0.04	0.07	0.08	0.07
3	Window Glass	0.30	0.22	0.17	0.13	0.07	0.03	0.15
4	Fabric on the wall	0.08	0.06	0.10	0.16	0.25	0.32	0.14
5	Linoleum Flooring	0.02	0.03	0.03	0.03	0.03	0.03	0.03
6	Wooden Flooring	0.15	0.12	0.10	0.06	0.06	0.06	0.09
7	Thin Carpet Flooring	0.03	0.06	0.10	0.20	0.43	0.63	0.20
8	Thick Carpet Flooring with under padding	0.08	0.28	0.38	0.40	0.48	0.70	0.39
9	Mineral Fibre acoustic ceiling tile - 12.5 mm	0.45	0.50	0.53	0.69	0.85	0.93	0.64

12.7 Noise Control

Good acoustics is a main requirement of high quality broadcasting or recording studio. Noise control measures are provided in studios, control rooms, and other technical areas in order to achieve the acoustic conditions desirable for the various types of programmes.

Some basic approaches for controlling noise in a studio are:

- Locating the studio in a quiet environment
- Reducing the noise energy within the room
- Reducing the noise output of the source
- Placing an insulating barrier between the noise and the room
- Using Sound Lock area before studio entry

12.7.1 Acoustic Treatment

Acoustic treatment refers to placing a suitable material on the wall surface, ceiling and floor that will have a direct effect on the sound quality. For reduction of echo, dead spots, reverberation points, reflection points and unnecessary sound magnification in the enclosed space acoustic treatment is applied. For limiting the unwanted noise acoustic treatment uses noise control measures, which reduces the noise level to the extent which is inaudible for human ears. The two distinct requirements may need to be considered when designing acoustic treatments are:

- *Improvement of sound within a room*, and
- *Reduction in sound leakage* to and from adjacent rooms or outdoors.

Design of Acoustic Treatment

In previous sections of this Unit we have learnt about how sound behaves in an enclosed room. For designing an acoustic treatment for a particular room some of the basic concepts you must keep in mind are:

- On hitting a surface, some of sound is absorbed, some of sound wave is reflected and some of it is transmitted through the surface. Dense surfaces will isolate sound well, but reflect sound back into the room whereas porous surfaces will absorb sound well, but will not isolate the room.
- The main way to minimize sound transmission from one space to another is adding mass and decoupling the spaces.
- Sound bounces back and forth between hard, parallel surfaces.

- The best way to stop sound transmission through a building structure is to isolate the sound source from the structure.
- Every object and material used in acoustics has a resonant frequency at which it is virtually an open window to sound. Different materials have different resonant frequencies.
- Trapped air (Air Gap) is a very good de-coupler.
- Airtight construction is a key concept. Sound, like air and water will leak through any small gap and will result in diffraction.

Note It

Acoustics treatment is required so that musical qualities of intimacy, timbre, balance, dynamic range, fullness of tone, loudness etc. should be preserved.

12.7.2 Technical requirements for construction of studio

Height:

During designing phase of a studio sufficient height is planned for providing space for acoustic treatment of the ceiling.

Wall Thickness:

For achieving better sound isolation outer walls must be kept thick (more than the normal wall thickness) to provide better transmission loss to the noise.

No Pillar/Column:

No pillar or column (for clear and obstacle free working, and to minimize structure borne noise).

Observation Window:

Provision of Observation Window between recording booth and recording studio and between Control Room and Transmission Room for visual continuity. Observation window is constructed with double glass and are fitted at an angle.

Sound Proof Door:

The transmission loss depends upon the density of the SP Door. Sound Proof Door is provided for better sound insulation. A door leaf with magnetic seal and gasket provides good sound isolation.

Structural Isolation:

Structural isolation between machine block and studio, and between office block and studio reduces the structure borne noise. A structural isolation gap of 75 mm width right from foundation level up to the roof height is provided between the two blocks. Wherever required, only flexible connections are used for linking these blocks for running electrical cables, duct etc.

Shape:

a. Avoid circular shape to avoid acoustic defects such as sound foci.

b. Avoid cubical shape.

c. Fairly rectangular as per aspect ratio given in *Table 12.4*

Table 12.4: *Aspect ratio of Studio as per Volume*

S.No.	Volume	Aspect Ratio		
	(Cu. Mtrs)	**Length**	**Width**	**Height**
1	Up to 250	1.6	1.3	1
2	650 to 1250	2.5	1.5	1
3	2000 to 4000	3	2	1
4	4000 Upwards	3.3	2.2	1

Volume of Studio:

a. The volume of an enclosure for music recording is related to the number of musicians.

b. An empirical formula establishes the following relation between the number of performers and the volume of the studio

v = 21 n + 55

Where

v = Volume in cubic meters

n = Number of performers

Activity 12.1

During your visit to a Community Radio Station, have a look on acoustic treatment of different studio rooms. Note down the various materials used for the treatment studied in this Unit. Fill in the details in the proforma given below. This will help you identify the types of acoustic treatment and visualize their significance.

A. Primary use of the rooms

S. No.	Item	Utilization
1	Room 1	
2	Room 2	
3	Room 3	
4	Room 4	
5	Room 5	

B. What are the overall studio dimensions?

S. No.	Item	Dimension
1	Length	
2	Width	
3	False Ceiling Height (FCH)	
4	Real Ceiling Height (RCH)	

C. What are the finishes on the ceiling?

S. No.	Material	Thickness of Material
	Real Ceiling	
1		
2		
	False Ceiling	
1		
2		

D. What are the finishes on the floor?

S. No.	Material	Thickness of Material
1		
2		

E. What are finishes on each wall?

S. No.	Material	Specification
	East wall	
	West wall	
	North wall	
	South wall	

F. Sound Proof Door

S. No.	Parameter	Specifications
1	Height	
2	Width	
3	Thickness	

G. Observation Window

S. No.	Parameter	Specifications
1	Height	
2	Width	
3	Thickness	
4	No. of glasses	

12.8 Let Us Sum Up

As we have learnt in the beginning of this Unit, the science of acoustics can be wide-ranging and confusing. We have seen how sound waves behave in an enclosed space like studio and effects of objects within the studio. In this Unit, we have learnt about the necessity of noise control and how we can achieve good acoustic to ensure the most advantageous flow of sound. We have discussed about the different noise sources and different ways to diminish the noise from these sources. Phenomena of Sound Isolation and Absorption were discussed, which are very important for understanding acoustic requirement and design. We have learnt different characteristics of construction materials towards sound isolation and absorption. We have discussed technical requirements for studio construction, like studio height, wall thickness and shape and volume.

It is a known fact that broadcasting studios should be free from noise and be designed for optimum R/T requirements. These requirements are duly taken care of at the design and installation stage. However, sufficient precautions should be taken during maintenance i.e. painting etc. and at the stage of making any additions or changes in the studios, so that the characteristics are not altered.

12.9 Model Answers to Activities

The information gathered in the activity presented in this module should be your own experiences. The activity is hands-on activity here.

12.10 Additional Readings

Basics of Sound Waves [Video file]. (n.d.). Retrieved from http://www.youtube.com/watch?v=12uJ7Jzcr_4

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Glossary

Acoustics:	The science that deals with the study of sound waves.
Amplitude:	It is signal strength or height of signal. The loudness of sound. Commonly called volume. Measured in decibels.
Attack:	The time sound takes to reach its peak.
Audio File size:	Space required for storage of an audio file.
Bandwidth:	It is the difference between upper and lowest frequency of an audio signal.
Bit Error Rate:	Error introduced by communication channel in transmission of a digital signal.
DAW:	Digital Audio Work station. A computer that allows one to record, edit and add effects to an audio programme, most of the times independent of hardware.
Decay:	The time sound takes to die down.
Diaphragm:	A small membrane in a microphone which vibrates in accordance with air pressure of sounds wave.
Dither:	Method used for reducing quantization noise.
File Formats:	WAV, MP3, PCM, AIFF
Foleying:	Creating sound effects in a studio
Frequency:	Rate at which a signal changes per second or number of cycles per second. Measured in Hertz.
Hyper-cardioid:	Very directional microphone, which eliminates sound from the sides and the back.
Jitter:	Deviation of discrete samples from precise sample timing intervals due to error in clocks.
Mono sound:	Audio recorded and heard on just one channel. Both speakers reproduce the same sound.
Quantization and Quantization Noise:	Conversion of continuous signal to digital signal.
Reflection:	Bouncing back of sound after striking a flat and hard surface.

Refraction:	Changes in the direction of travel of the sound by differences in the velocity of propagation.
Sampling:	Method of converting analogue signal into digital signal.
Sound absorption:	The process by which we can reduce the reflection of the sound energy by the surfaces.
Sound Isolation:	Acoustic treatment of studio space to reduce the effects of exterior noise.
Stereo:	Audio recorded on more than one channel. Sound is heard differently on right and left speakers.
Sustain:	The time sound remains at its peak.
Transducer:	Converts vibrations of diaphragm into an equivalent electrical signal.
Unidirectional microphone:	Which picks up the sound primarily from one direction i.e. in the front and reduced pickup from the side and the back.
Wavelength:	The distance between any point on the wave and a corresponding point on the next wave. Measured in metres.

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