Module: 1
Studio Technology

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Module 1 : Studio Technology

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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 2 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 3 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 4 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 1

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 1.1) 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 1.1: 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 1.2 explains how sound enters your ear and how these signals are sent to the brain to process.

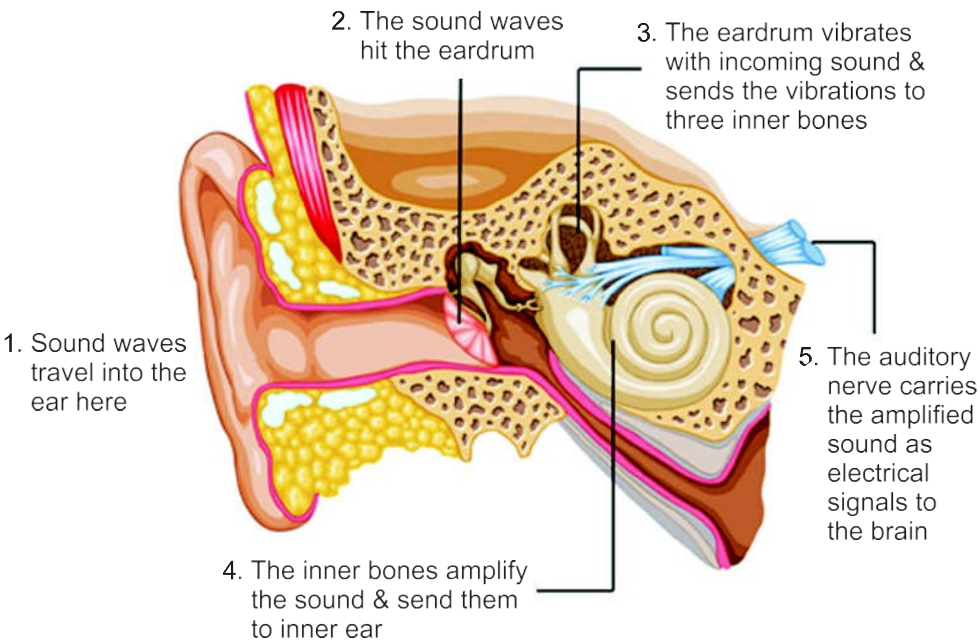


Figure 1.2: Sound waves that enter the ear hit the eardrum and signals from the back-end of the eardrum, which are carried to the brain to process the waves.

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 1.3) 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.

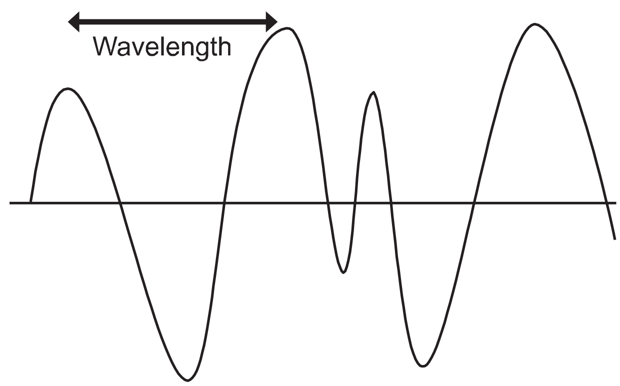


Figure 1.3: Wavelength of sound

9.4.2 Amplitude

Amplitude indicates the height of a sound wave as shown in the figure 1.4 — 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.

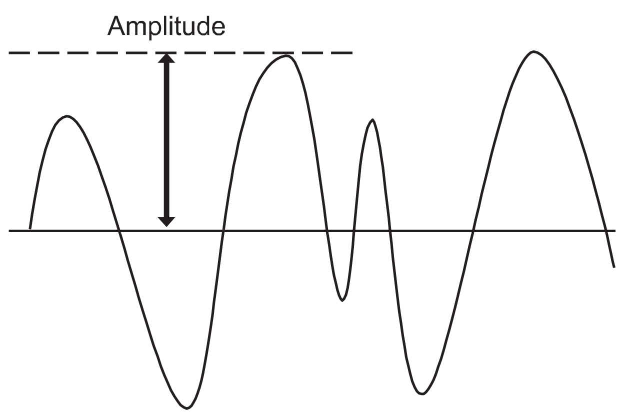


Figure 1.4: Amplitude of a sound wave

9.4.3 Frequency

Frequency is the number of times the wavelength occurs in one second (Figure 1.5). 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.

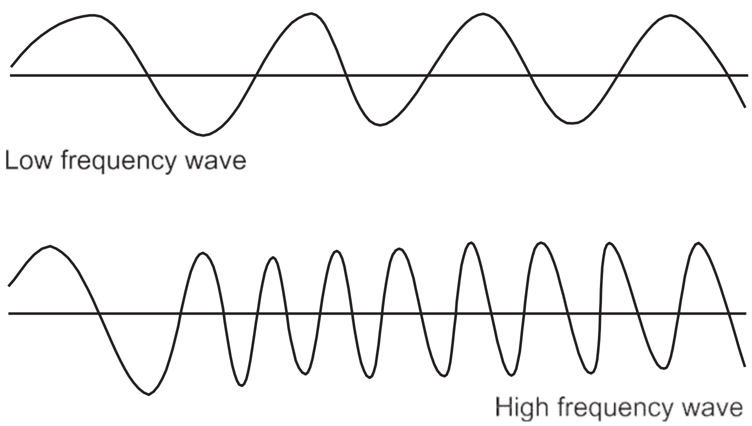


Figure 1.5: Frequency of a sound wave

Activity 1.1

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 1.2

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 1.3

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 1.4

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 1.4).

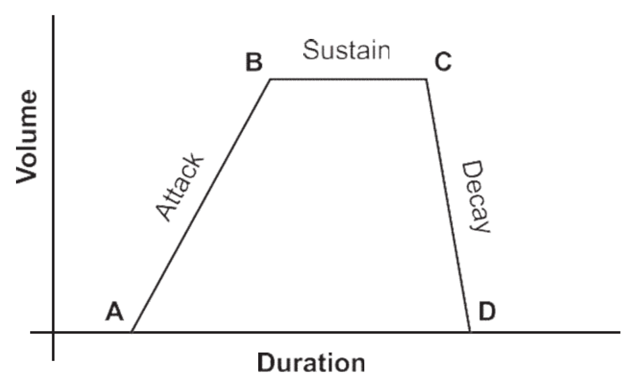


Figure 1.4: Attack, sustain and decay represented in the form of a graph. The volume of sound is represented on the 'x' axis and the duration of attack, sustain and decay is on the 'y' axis

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.

1.9 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.

1.10 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.

1.11 Model Answers to Activities

Activity 1.1

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 1.2

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 1.3

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 1.4

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 2

Analog and Digital Audio

Structure

2.1 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.

2.2 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 2.1.

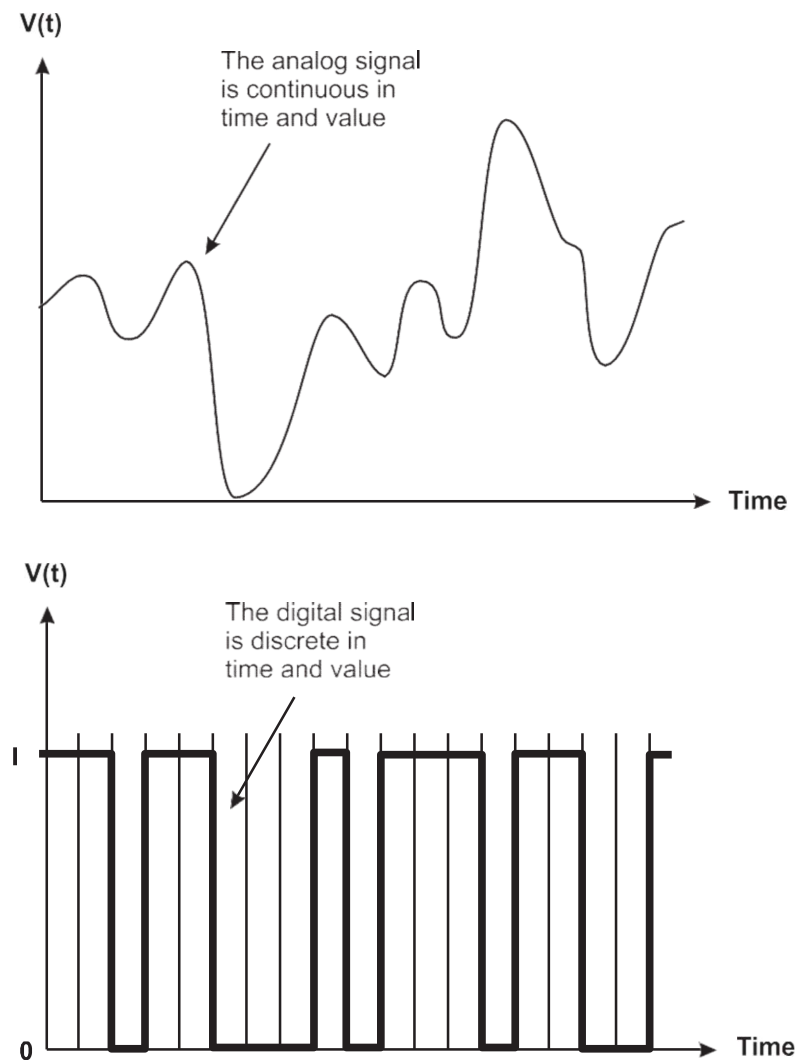


Figure 2.1: Analogue and digital signal

As shown in Figure 2.1, analogue signal varies continuously in amplitude orheight as time progresses, like electrical signal. Analogue signal can assume anyvalue whereas digital signal can take only two definite discrete values, zero andone. 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 2.2, 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.

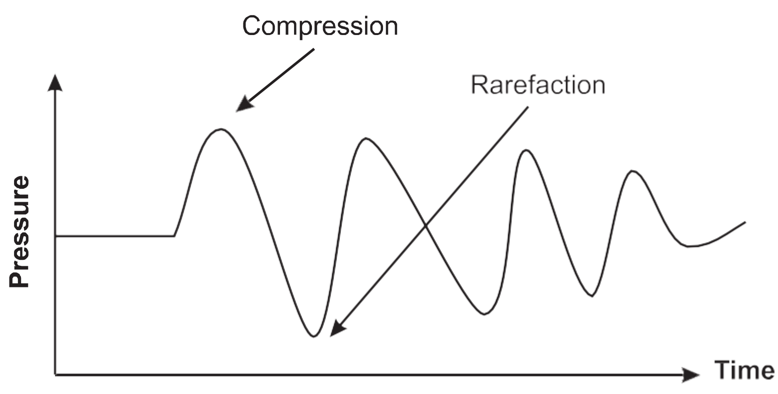


Figure 2.2: Analogue sound wave

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 2.1

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 2.3.

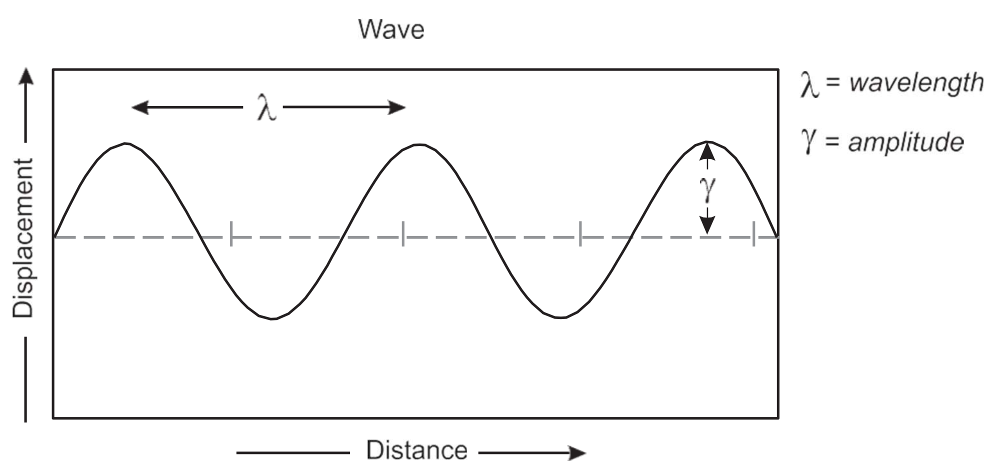


Figure 2.3: Characteristics of analogue audio waveform

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 2.4. This implies that there is a time delay between the two waves.

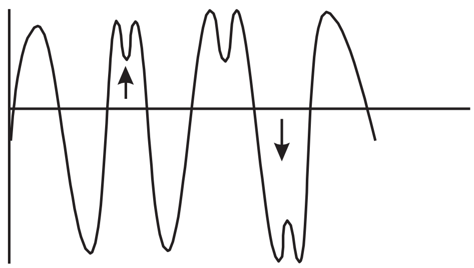


Figure 2.4: Phase shift of waves

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 2.2

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 2.5 shows digitized audio.

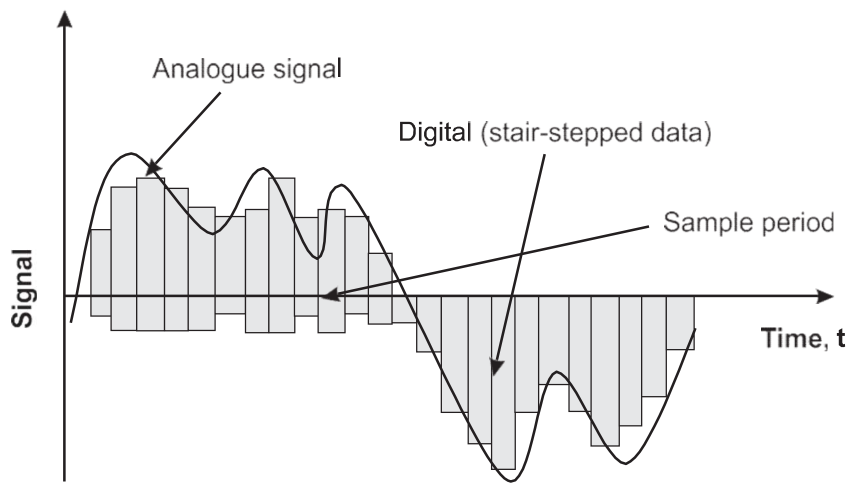


Figure 2.5: A digital signal

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

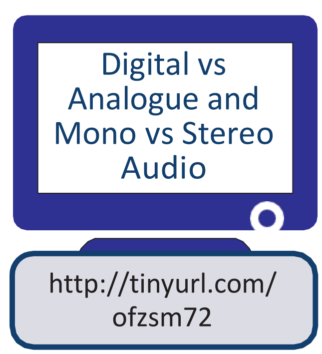
During sampling and quantization, the amplitude of audio signal is sampled at specific rate and rounded off to nearest discrete value. This results in quantisation errors in the form of audible effects of adding low level harmonic distortion in the encoded signal, which is called quantisation noise. 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 2.3

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 2.7.2, 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 2.4

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.

2.10 Model Answers to Activities

Activity 2.1

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 2.2

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:

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

Activity 2.3

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 2.4

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 3

Components of the Audio Chain

Structure

3.1 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).

3.2 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 3.1 shows a simplified block schematic of the audio chain of a broadcast studio.

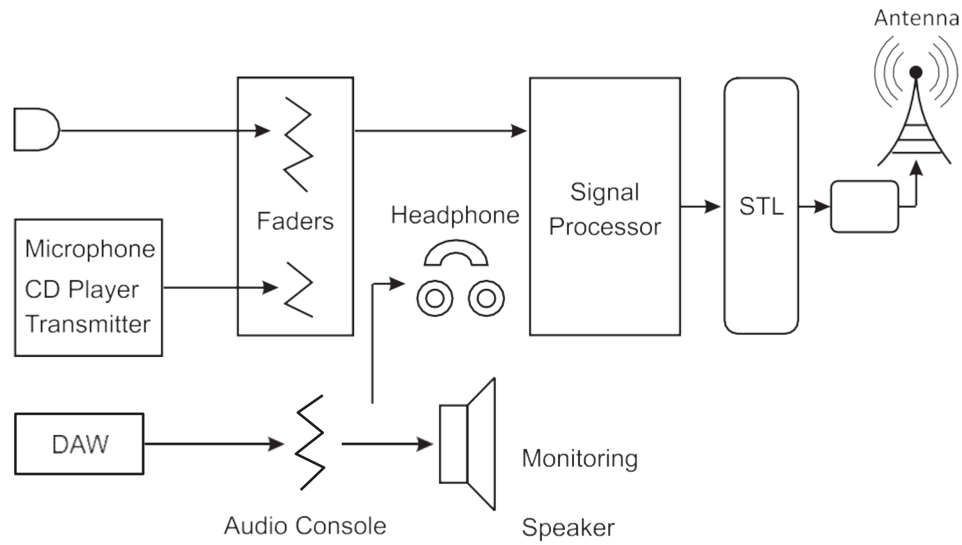


Figure 3.1: Audio Chain of a broadcast studio

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 3.2.

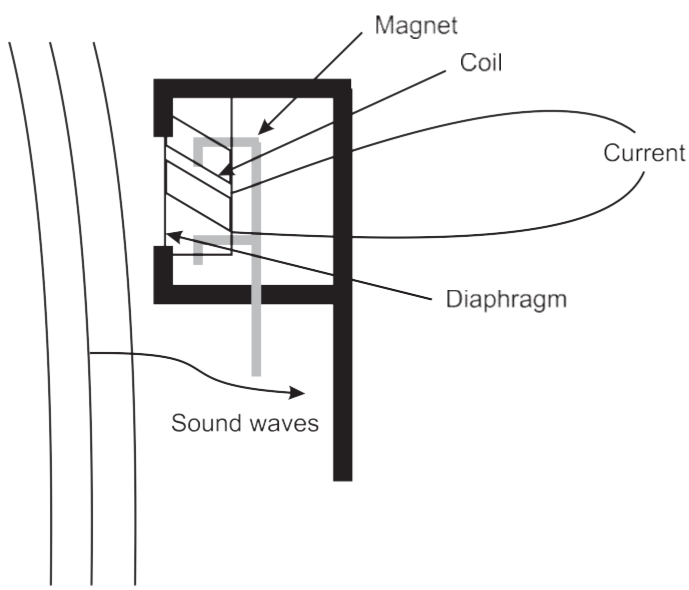


Figure 3.2: Dynamic Microphone - Block Diagram.

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 3.3.

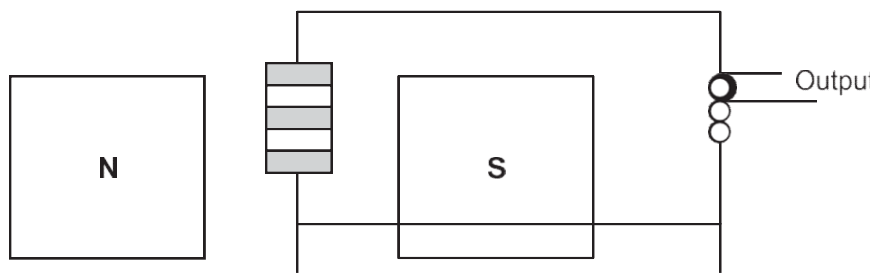


Figure 3.3: A Ribbon Microphone

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 Figure3.4.

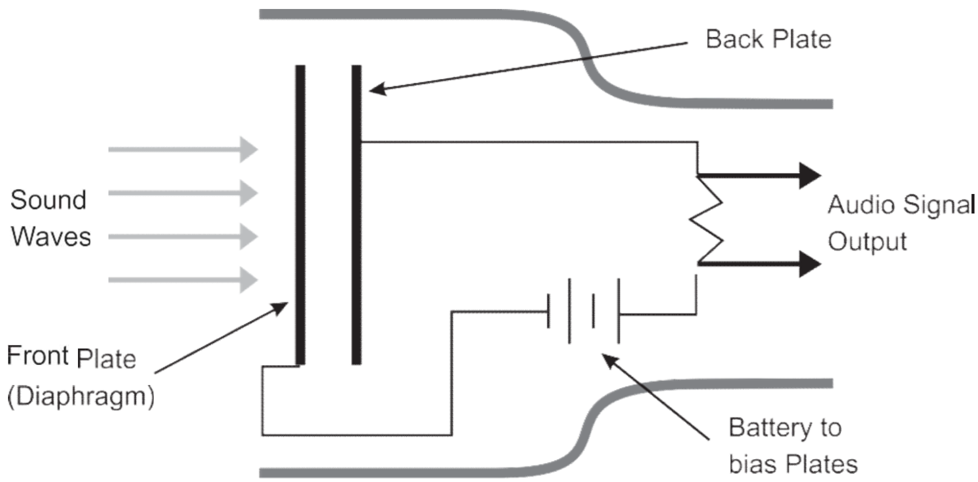


Figure 3.4: A Condenser Microphone

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 3.1

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 3.5. 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.

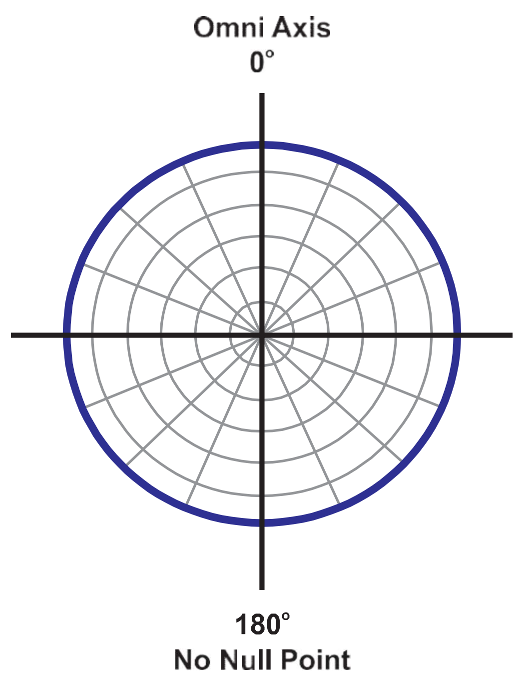


Figure 3.5: Polar Pattern of Omni-directional Microphone

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 3.6. 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.

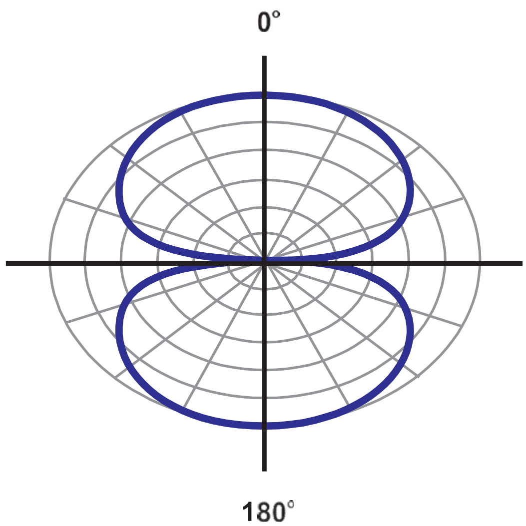


Figure 3.6: Polar Pattern of Bi-directional Microphone

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 3.7.

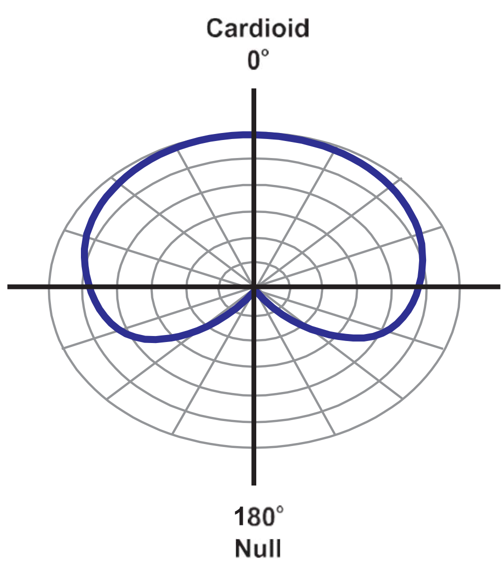


Figure 3.7: Polar Pattern of Cardioid Microphone

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 3.8. Due to the long thin design, these hypercardioids are often referred to as shotgun microphones.

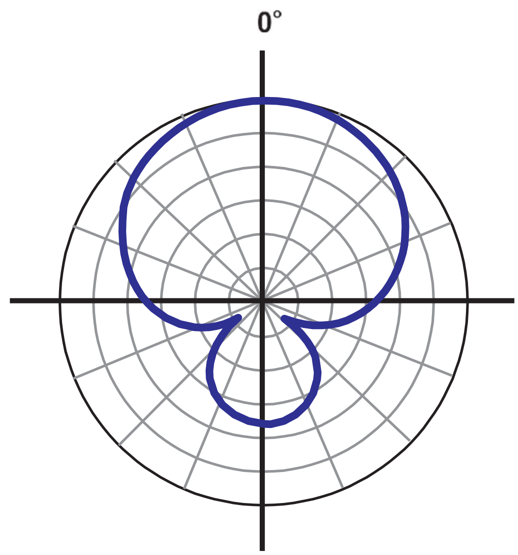


Figure 3.8: Hyper cardioid Polar Pattern.

Activity 3.2

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 3.9 shows an audio mixer with five inputs and outputs.

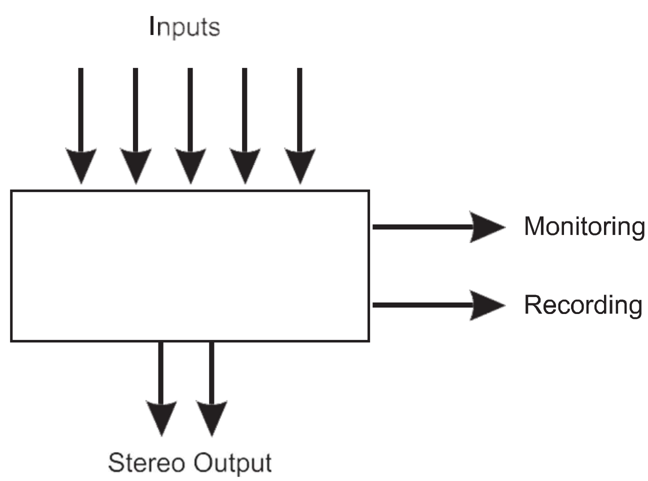


Figure 3.9: Audio Mixer block diagram

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 3.3

i. What are field recorders?

3.7 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.

3.8 Model Answers to Activities

Activity 3.1

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 3.2

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 3.3

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 4

Studio Acoustics

Structure

4.1 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.

4.2 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.

Acoustics & Noise Reduction

http://tinyurl.com/nbyzl64

To understand Studio Acoustics we must know the basic phenomenon of sound waves propagating in a studio. In Unit 1 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 4.1* shows the behaviour of the sound wave when it strikes acoustically treated wall.

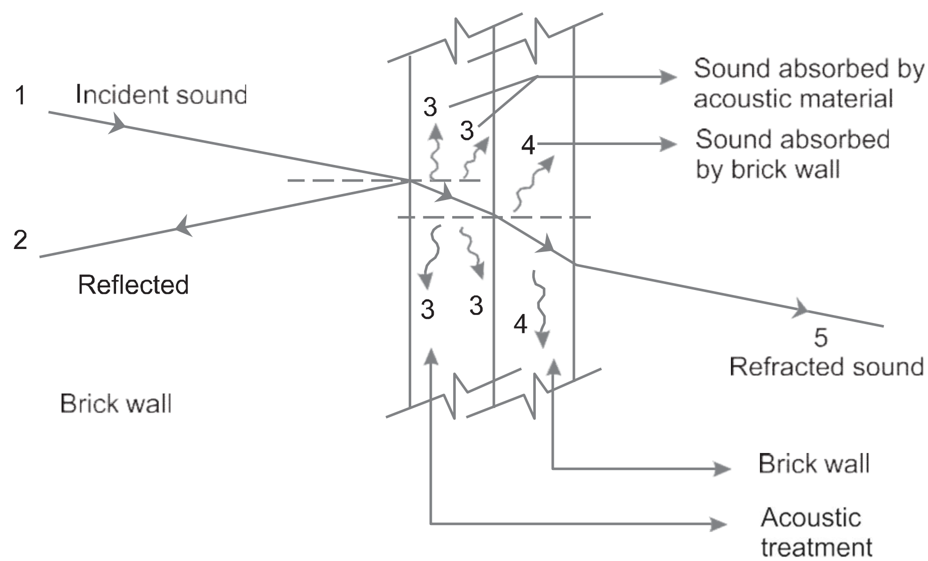


Figure 4.1: Behaviour of Sound Wave on striking Wall

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 4.1 (wave 5)*. The change in the velocitymay 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 4.2*.

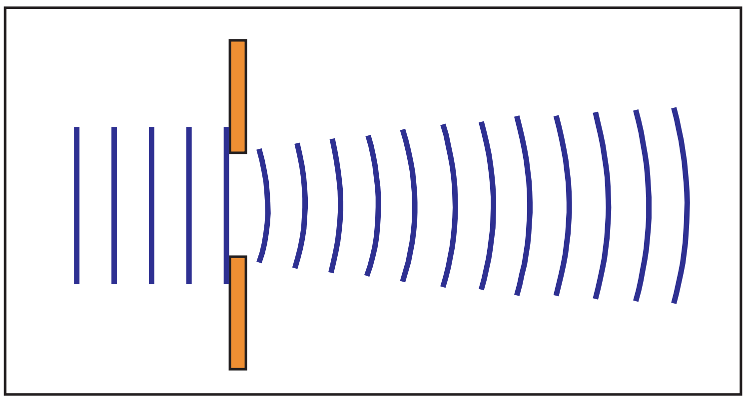


Figure 4.2a: Diffraction of Sound Wave [Wide Gap - Small Diffraction Effect]

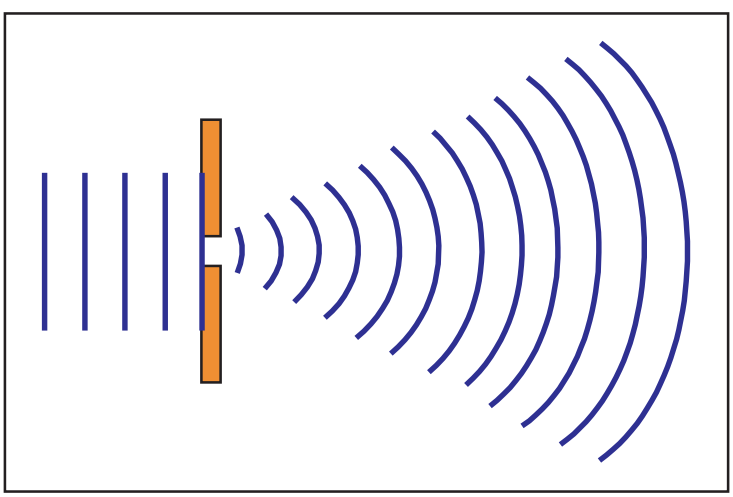


Figure 4.2b: Diffraction of Sound Wave [Narrow Gap - Large Diffraction Gap]

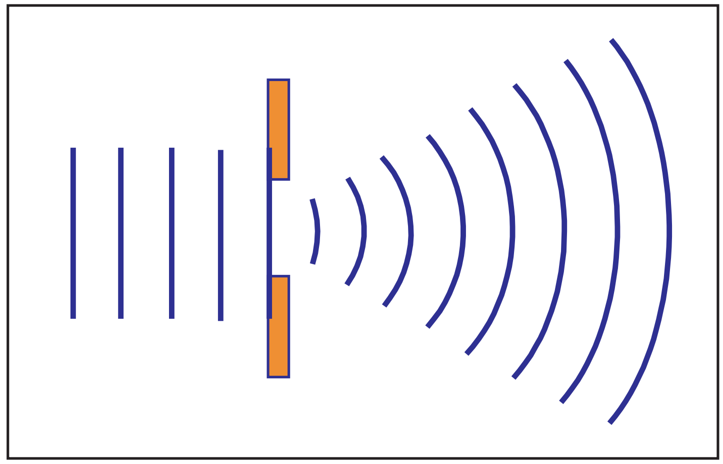


Figure 4.2c: Diffraction of Sound Wave [Large Wavelength - Large Diffraction Effect]

Reverberation

You have already read about reverberation in Unit 1. 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.

$RT60 = 0.161V / S\,\alpha$ (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 4.1* 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 4.1.* 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 4.2.*

Table 4.1: 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 4.2: 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 4.1* 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 4.3*. The last column of the table 4.3 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 4.3: 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 4.4*

Table 4.4: 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 4.1

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.

4.9 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.

4.10 Additional Readings

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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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About the Module

Module Description

In previous modules, you were introduced to the concept of community radio (CR) and its related frameworks. You further received a foundation in the basics of audio (analogueue and digital), electricity, and principles of acoustics.

This module introduces you to the concept of audio recording, that is, the principles and practice of capturing sounds,converting them to a storable form,and storing them in a recording medium for later recovery. You will also understand the various types of hardware that are used to record audio in the field and in the studio, and the concept of using free and open source software (FOSS).

You will also learn about telephony related applications for radio, including the use of hardware and software to create interactive programmes with call-in segments; as well as the use of mobile telephones and systems to connect your listenership with the CR stations.

Module Objectives

- To introduce the concept of recording and storing audio
- To introduce the hardware related to audio recording
- To explain the concept of free and open source software solutions for audio production
- To examine the available options for telephony interfaces for radio

Units in the Module

- Unit 5: Audio Hardware and Field Recording
- Unit 14: Free and Open Source Software
- Unit 7: Telephony for Radio

UNIT 5

Audio Hardware and Field Recording

Structure

- Introduction
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- Audio Cables and Connectors
  - Balanced and Unbalanced Cables
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- Headphones
- Recording Audio in the Field
- Portable Audio Mixers
- Sound Cards
- Digital Audio Workstations (DAWs)
- Let Us Sum Up
- Model

5.1 Introduction

In the preceding units, you were introduced to various pieces of hardware used in an audio studio. You learnt about the differences between analogue and digital audio, as well as the components of the audio chain that enables you to record, produce and broadcast radio programmes. You were also introduced to the importance of acoustic treatment of a studio and the various materials that go into the treatment. It is now time to delve a little deeper into the equipment required for recording in the field and in the studio.

In this unit, you will learn about the tools—microphones, mixers and recorder—that you will use for this purpose. You will also learn about digital audio workstations and sound cards, and the cables and connectors that we use to link the various pieces of equipment together. Finally, you will also receive some tips on recording good audio in the field.

While going through the unit, you will need to do four activities (the model answers of which are given in Section 5.13). You will need approximately four hours to complete this unit.

5.2 Learning Outcomes

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

- identify microphones based on their pickup patterns.
- identify and explain the differences between dynamic and condenser microphones.
- list and describe the features of different types of audio recorders.
- identify and describe the use of different kinds of audio connectors and headphones.
- identify a digital workstation and describe its features.

Hardware for Audio Recording

Depending on whether you are recording audio in a studio or in the field, you require four basic pieces of equipment: (1) a microphone; (2) an audio recorder; (3) a pair of headphones; (4) and an audio mixer. If you are recording in a studio, the audio recorder is usually replaced by a computer capable of recording audio directly into a software recorder. Before going through the various equipment needed for sound recording in the studio and in the outside you may watch a

video on 'Studio Recording Hardware and Field Recording' at http://tinyurl.com/ppkowtp. This video will help you to apprehend the various instruments needed in the recording process, either inside the studio or outside.

Let us look at these in greater detail.

1. *Microphone:* A microphone is basically a transducer that converts sound waves into electrical waves. The electrical waves can then be either amplified using loudspeakers, or recorded and stored. Microphones are either built-in—as in computers, telephones, or mobile phones—or free-standing devices, like those you use in an auditorium or in a studio. They come in various sizes and shapes. It must be remembered that no one microphone serves all purposes. In fact, it is the purpose that decides what kind of microphone one should use.

2. *Audio recorder:* An audio recorder is a device that records the electrical signals emerging from a microphone as audio files. These audio files can then be electronically moved from one device to another (for instance, a computer) and edited. There are several types of audio recorders in the market: some easy to use, some economical and some that combine both these characteristics in addition to being robust. We will examine the different kinds of audio recorders later in this unit.

3. *Headphones:* Headphones are devices that are used to listen to the audio signal being played back by a playback device. In some ways, they are comparable toa small pair of speakers mounted on a band that can be worn over the head. Headphones are also useful when you want to listen to audio without disturbing others around you.

4. *Audio mixer:* An audio mixer is a device that helps connect multiple audio sources to a recorder, while also helping you to refine and improve the quality of the sound. Mixers are often used to individually control the sound from each source independently, in order to achieve a good balance between the various sources.

5. *Computer:* For the purposes of audio recording and editing, the computers we use are called digital audio workstations (DAWs). They are different from the computers we use for day-to-day office work and Internet browsing in that they are more powerful; and also have a variety of audio inputs and outputs. Most will also have a sound card, which is a device that can convert analogue audio to digital audio, and vice versa. Some may be designed to work with specific hardware that can be connected to them to record and edit audio files.

Microphones

You have already learnt about the details of various types of microphones in Section 11.4 of the previous module. You have got to knowtherein that microphones are classified into two main types namely classification by the type of transducers and classification by pickup, that is, directional properties. You have also been made familiar with the polar diagrams of various types of microphones. It is important to know the different kinds of microphones available in the market so that one can put them to appropriate use. The purpose could be recording just speech or music, or evenbird and animal calls. For our purposes, we will limit ourselves to the microphones that radio professionals use all over the world and deal with the key considerations while selecting them along with their basic characteristics and usage.

13.4.1 Key Considerations while Selecting Microphones

Before purchasing or hiring a microphone, one would do well to consider the following five factors:

1. *Impedance:* Impedance, as the word itself suggests, means resistance offered to the flow of the electrical audio signal. You may have noticed that big audio players that come with a microphone (for karaoke) have a very short cable. This is because they have very high impedance, that is, they offer high resistance to the flow of the signal to the recorder. If the cable isvery long, the signal that reaches the recording device would be too weak for any practical purpose. On the other hand, a microphone with low impedance can be used with a long cable, since the signal strength that emerges will still be quite strong. Most professional microphones are of low impedance, making them best suited for audio recording. Impedance is measured in ohms. Professional microphones have an impedance of 200 ohms or lower.

2. *Frequency response:* This is a microphone's capability to receive high and low sounds. A good, professional microphone can receive frequencies ranging from 20 to 25,000 Hz.

3. *Pickup pattern:* Situations may demand requirement of a producer to record sound either from one direction, two directions or all directions. Situations may also require the producer to actively reject sounds from other directions while picking up sounds clearly from one. There are different microphones that meet these requirements. The choice of a microphone is often determined by this requirement. You can read more about this in Section 5.4.2, where we discuss different pickup patterns.

4. *Balanced andUnbalanced microphones:* Microphones can be balanced or unbalanced. Balanced microphones are best used for professional recording purposes. An unbalanced connection uses two wires: The centre conductor carries the audio signal, while the shield or basket carries the ground wire. On the other hand, a balanced connection uses three wires. Two separate signal wires inside the shield carry the plus and minus signals (opposite polarity). The shield is connected to the ground, and protects the signals from external electromagnetic interference. This makes the system more

immune to noise from poor electrical wiring in a room, computers, etc. With an unbalanced microphone, one of the signal wires is the shield and the other is the positive signal. Any noise picked up on the shield will be fed directly into the amplifier or mixer input.

5. *Sensitivity:* The ability of a microphone to pick up very faint sounds is termed as its sensitivity. High-sensitivity microphones are normally used in studios and low-sensitivity microphones on the field, where the chances of the microphone moving or being hit by strong wind are greater.

13.4.2 Microphones Categorized by Directionality

Omni-directional microphones

An omni-directional microphone can pick up sound from all directions. They are very easy to use and most of the basic microphones found in the market are of this type. These microphones are normally the ones that a *tentwallah* (a person who rents out tents) uses during community functions.

Uni-directional microphones

You may have noticed in music shows that even though instruments are placed close to each other, the microphone placed above one instrument does not pick up sound from another. These microphones preferentially pick up sound from one direction, while rejecting sounds from other directions. Such microphones are called uni-directional microphones. They are widely used for recording in outdoor interviews (where unwanted ambient noise levels are high), and in panel discussions in studio situations. From a pickup pattern point of view, their patterns look like an inverted heart, which is why they are sometimes also called cardioid microphones. Some cardioid microphones are designed to be more directional than others, which are called super- and hyper-cardioid microphones. These are used for recording sounds from far away. Some hypercardioid microphones can pick up sound from as far as 300–500 metres!

Bi-directional microphones

A bi-directional microphone can pick up sounds from the front or the rear of the microphone, but not from the sides. Their polar patterns are usually in the shape of a figure-8. Bi-directional microphones are very sensitive by construction, and need to be handled carefully. When not placed carefully, they can produce pops, and are best avoided for fieldwork. They are very useful for discussions when two persons are seated across a table for an interview.

13.4.3 Microphones Categorized by Construction

Dynamic microphones

Also called moving coil microphones, dynamic microphones are rugged and robust and are mostly used in field situations. They are more resistant to unwanted vibration and wind rumble. The microphones that are used by rock stars are normally of the dynamic type. Typically, their frequency range lies between 40 and 18,000 Hz. Figure 5.1 below shows a popular dynamic microphone.

Figure 5.1: A dynamic microphone.
(Source: Jnan Taranga, KKHSOU, Guwahati)

Condensor microphones

Condensor microphones are much more sensitive microphones, and are generally used for studio applications. Typically, their frequency range covers the entire human hearing range from 20–20,000 Hz. Unlike dynamic microphones, condenser microphones need an external power supply. Some condenser microphones have a battery attachment within the body of the microphone itself. Batteries are usually replaceable, as required. Others need power delivered to them through the microphone cable itself, an arrangement called Phantom power, and often denoted by the symbol +48V or P48. Figure 5.2 shows a popular condensor microphone.

Figure 5.2: A condensor microphone.
(Source: Jnan Taranga, KKHSOU, Guwahati)

Activity 5.1

This activity will require access to different kinds of microphones. If you have access to a community radio station (CRS) or to a studio, this will not be a challenge. If you do not, you may use a mobile phone or a cassette recorder with a built-in microphone or any device on which you can make an audio recording.

Place the microphone on a table and connect it to a recorder, or turn on the recorder. Now record the audio:

a. By speaking directly in front of it
b. By speaking next to it
c. By speaking sitting next to it

Do this one by one with each microphone and notice the differences in the recordings. Note down the results. If you have access to different microphones, repeat the exercise using each type and see if you can identify whether they are uni-, omni- or bi-directional microphones.

Audio Recorders

Audio recorders are devices that can receive an audio signal from a microphone input, and store them in a recording medium. Audio recorders can be designed for use in the studio, or may be compact enough to be used in the field. For practical purposes we will consider the latter, since most studios nowadays tend to record directly on computerized digital audio workstations (DAWs).

The older generation of portable analogue audio recorders included cassette recorders, and reel-to-reel spool tape recorders, both of which record on magnetic tape. The newer generation of audio recorders are digital and record on digital media like secure digital (SD) or compact flash (CF) cards, or hard disks. You have already learnt about analogue and digital audio in Unit 10.

Portable field audio recorders usually come with built-in microphones good enough to record broadcast grade audio. Some of them even come with wind filters, and record on easily available and reusable digital media.

Just search on the Internet for digital audio recorders and you will be surprised by the variety in their kinds and prices, each one of them claiming to be the best. You will also find both expensive and inexpensive recorders. You may have noticed that you can record on a slightly high-end mobile phone too. However, just like microphones, there are considerations when purchasing a professional audio recorder. Check out the specifications before purchasing a recorder to suit your budget.

When purchasing a digital audio recorder, you should check for the following:

- Recording format: Should be able to record in .WAV format, 16/24 bit. It is better if it can also record 'mp3' format at varying bit rates.
- Should have built-in stereo microphones, preferably covered with a wind shield

- Should have a provision for line-in stereo, input for external microphone (preferably XLR input)
- Should have a provision to connect headphones to monitor audio
- Should have built-in speakers
- Should have adjustable controls for input audio levels
- Should have an audio level indicator (VU meter)
- Should have a media card slot with a capacity for high data storage (2 GB or more)
- Should have a USB cable and connector slot to enable import of audio files to a computer
- Some recorders are capable of recording from a landline telephone cable. Some can also be connected to a computer and used as a microphone to record directly into the computer using an audio software.
- Some recorders come with built-in batteries that can be charged. Others accept consumer grade batteries of AA or AAA type. For recorders that use the latter, you will require some rechargeable cells of appropriate size, preferably of the Nickel Metal Hydride (NIMH) type. As a general practice, it is always good to carry a couple of extra chargeable batteries, so that you do not run out of power during an extended day's recording.

Figure 5.3: Zoom, Olympus and Tascam manufacture some of the most popular digital audio recorders.
(Source: Jnan Taranga, KKHSOU, Guwahati)

The most popular type of recorders available in the market today are digital flash memory recorders. These recorders make digital recordings onto flash memory cards, such as compact flash (CF) cards and secure digital (SD) cards. Usually, they have an option to save as uncompressed linear PCM or in a compressed format such as MP3. For a given size of memory card, you can record for a longer duration using a compressed format but at a lower quality. Some of these recorders record on a built-in flash memory.

Such digital recorders are preferred by a wide number of people because they are highly portable (can fit into a pocket) and can be used with very little experience. Transfer of files from such a recorder to a computer for editing is easy. In fact, with a bit of patience, one can also rename the audio files stored on the recorder. Identifying the files after transferring them onto a computer also becomes easy. In fact, some recorders also provide for simple editing within the recorder itself! Yet another important advantage of such recorders is that they make it easy to record and post podcasts on the Internet.

Flash cards are a very useful recording medium since they are robust, use less power supply (particularly when your audio recorder uses replaceable batteries), and enable faster data transfers. They usually come in different capacities—from 512 MB to 64 GB. To record audio, a 2GB or 4GB solid state card is more than adequate. You will learn more about storage media in Unit 19 on Storage and Retrieval.

13.3 Audio Cables and Connectors

Just like any other piece of audio hardware, there are scores of audio cables and connectors available in the market. Every audio cable and connector serves a particular purpose, and it is important that we know the key features of the common ones in order to be able to make the right choices.

13.3.1 Balanced and Unbalanced Cables

In an earlier discussion in this unit (Section 5.4.1), we learnt about balanced and unbalanced microphones. It should stand to reason that both these types of microphones will require appropriately matched cables and connectors to carry the signal they generate.

The simpler type of cable is the unbalanced cable. These are cables with a single central conductor, and a woven metal basket (or shield) on the outside, with the entire combination encasedin plastic insulation. The shield is meant to absorb stray electromagnetic interference from the surroundings.

Any circuit is complete only when the electrical signal has an incoming and a return path. In the case of the unbalanced cable, the central core provides one of the paths, and the metal shield the other. Unfortunately, this means that any interference that the shield picks up will be added to the audio signal. This can cause disturbances or hums in the recording. Figure 5.4 shows an unbalanced cable.

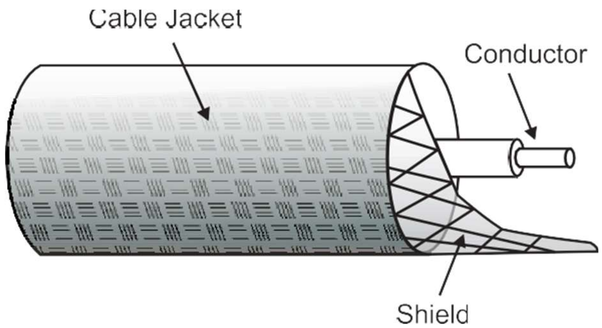

Figure 5.4: An unbalanced cable. Note the lone central conductor and the two layers of insulation, one of which separates the central core from the shield.

On the other hand, a balanced cable is a cable containing two cores, one positive and the othernegative, which are twisted together and surrounded by an overall shield. Only the two core wires carry the signal and the shield is grounded directly to the frame of the audio equipment. This allows the shield to perform its function of trapping external electromagnetic interference. As a result, the audio signal is completely noise free. Figure 5.5 shows a balanced audio cable.

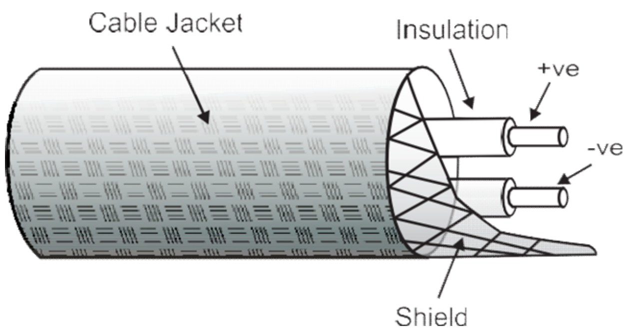

Figure 5.5: An balanced cable. Note the dual core wires.

Having dealt with cables, let us discuss connectors now.

13.3.2 Audio Connectors

Audio connectors are the devices used to connect cables to audio devices or one audio cable to another. There are a large variety of connectors, and they are also available in balanced and unbalanced varieties. Let us look at the most commonly used ones.

RCA connectors

RCA connectors are named after the Radio Corporation of America, who invented this type of connector. The male RCA plug consists of a central pin measuring approximately two millimetres (mm) in diameter, and an outer shell whose inside diameter is approximately six mm. The plug shell is slotted rather than threaded, to facilitate quick insertion and removal from the female jack or receiver. Typically, the cable carries the plug, and the audio device carries the jack. The RCA connector does not provide a balanced audio output. This type of connector is often used in home entertainment systems. You can see a pair of RCA plugs in Figure 5.6 below.

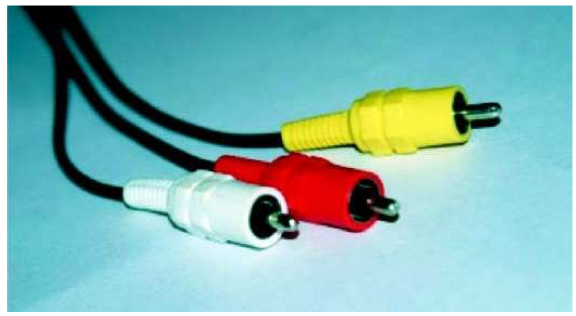

Figure 5.6: A pair of RCA connectors.
(Source: Jnan Taranga, KKHSOU, Guwahati)

XLR connectors

XLR connectors are the global standards for balanced audio output. 'XLR' stands for Ground (X) – Left (L) – Right (R). As its name suggests, it has three pins—two carrying the signal (and connected to the two core wires of a balanced cable) andthe third connected to the shield or ground. XLR connectors also feature a small lock that allows male and female connectors to be connected securely. This allows XLR plugs to be connected to sockets firmly, with no fear that the connector will come out of the socket by accident. They even help to lengthen cables by connecting them with a pair of XLR connectors.

XLR connectors also come in a mini-XLR variety. These are normally used with field wireless microphones and field recorders. You can see a pair of XLR connectors in Figure 5.7.

Figure 5.7: A pair of XLR connectors. The one on the left is the female connector; and the one on the right the male connector.
(Source: Jnan Taranga, KKHSOU, Guwahati)

TRS Connectors

'TRS' stands for tip-ring-sleeve, since this connector uses contact points on a single pin-like connector to convey the separate signals. These connectors usually come in two varieties, mono jacks and stereo jacks. The mono jacks are distinguished by a single black or white insulating ring towards the tip. The stereo jacks have two insulating rings towards the top, with the tip representing the left channel, the first metal ring to the right channel and the lower 'sleeve' connected to the shield. The mono jacks are usually used with instrument amplifiers. The stereo jacks are used with stereo devices, and are often used as connectors for headphones.

A larger variety of TRS connectors is the phono or quarter inch (1/4") jack, named after its pin that is a quarter-inch wide (see Figure 5.8). The smaller variety, often called the EP or mini-phono jack, has a 1/8" inch (3.5 mm) wide pin. The former is more commonly found on professional equipment and the smaller oneon consumer grade equipment like MP3 players and mobile phones.

Figure 5.8: ¼" phono jacks. The one on the left is mono, the one on the right is stereo (Source: Jnan Taranga, KKHSOU, Guwahati)

13.4 Headphones

Headphones are devices that are used to listen to audio, a process also known as monitoring audio. Typically, most units are like small speaker units mounted on a headband that allows them to be worn comfortably on the user's head. Like connectors, there are a variety of headphones, with a number of different uses. In this section, we will look at some of the commonly used types.

Ear bud headphones

These are probably the most common types of headphones, as they are used with all kinds of portable music players and mobile phones. Also called earphones, they are often provided as free accessories with these devices. While some low-quality ear bud headphones fit loosely within the external ear, there are some that fit into the ear canal itself. While earphones are good for listening to music, they are best avoided to monitor audio while recording. Figure 5.9 shows a pairof earphones.

Figure 5.9: Two pairs of simple earphones. The pair on the right has an ear canal adapter for snug fit.
(Source: WikiMedia Commons, *http://tinyurl.com/4eboaz3*)

On-ear headphones

These are headphones that sit on the ears rather than over them. As a result, they are usually a bit smaller and lighter than over-the-ear models. They tend to have foam or sometimes leatherette pads for extra comfort, and usually have adjustable headbands for a snug fit. These headphones are normally good on treble but not on bass. Since they don't cover the ears, ambient noise tends to enter the ears, making it difficult to monitor audio in critical conditions. They are therefore best used in office situations, for simple listening purposes, or for conducting voice chats over the Internet. Figure 5.10 shows a pair of on-ear headphones.

Figure 5.10: On-ear headphones
(Source: Jnan Taranga, KKHSOU, Guwahati)

Over-the-ear headphones

These are traditional-looking headphones, with cushioned pads that enclose and cover the whole ear. This makes them more comfortable to wear over long periods, and they generally deliver good sound quality. Bulkier than other types of headphones, these are best suited for audio monitoring purposes in the studio as well as in the field. Some varieties also cancel out noise, making it easier for the producer/technical personnel to monitor audio. The balanced headphone variety under this category provides the same impression as the sound you would be hearing from two or more speakers. Figure 5.11 shows a pair of professional over-the-ear headphones.

Figure 5.11: A pair of over-the-ear headphones. Note the cups that cover the entire ear. (Source: Jnan Taranga, KKHSOU, Guwahati)

Activity 5.2

This activity will require access to different kinds of connectors and headphones. If you have access to a CRS or to a studio, this will not be a challenge. If you do not, you may collect different kinds of cables and connectors that you have access to at home,or go to a local audio-parts shop and ask for whatever connectors they have. You may have to visit anaudio equipment shop to see some headphones.

Now, draw two boxes one above the other on a piece of paper. Place the different kinds of audio connectors in the first row of boxes. In the second row, identify and name the kind of audio connector. Match your answers with the figures given in this unit. You should be able to identify all the connectors correctly.

Now repeat the exercise with different kinds of headphones. Also use the headphones, if you can, and notice how over-the-ear headphones prevent outside sound from interfering with the audio you are listening to.

13.5 Recording Audio in the Field

Using portable digital audio recorders have made field recordings easy because of more than one reason. Since the audio they record are in the form of files, transferring them onto a computer, renaming them and other storage disks as backups is easy. It is also easy to rename files to avoid confusion if there are multiple files to be used in one single programme. Audio takes very little storage space and therefore optimum quantities can be recorded in order to produce a good programme.

The most common formats that are recorded in the field are voxpops, interviews, features, documentaries, and news segments. Interviews are a common feature among these formats and portable recorders come in very handy to make programmes in these formats.

Let us now look at some of the key processes and procedures to be followed while conducting a field recording.

The pre-production phase: Key points to remember

While making an audio programme and recording audio for it, you must always do your planning and ground work before hand. This means you should ideate and come up with a script for your production first. You must also identify people to be interviewed, and places you will interview them in; fix appointments; and take any necessary permissions to conduct recordings. In addition, you will have to keep the following things in mind:

1. Weather: The ideal weather conditions for radio field recording depends on what exactly you are wanting to record; but for most situations, a dry and calm (less windy) day is ideal. On other days of the year, take care to protect your equipment from rain. A small umbrella comes in handy both in summer and monsoon. If you intend to travel long distances for a field recording, check on the weather conditions at your destination before you leave.
2. Time: Even in the best of situations, field recordings take a lot of time. Keep time for walking/driving from one place to another, waiting for people, and other unforeseen delays. Traffic can pose a major problem in urban settings. Planning your time is extremely important, and so is time management. Keep adequate time for non-production activities like lunch. Keep spare time for every activity, and a backup plan ready in case you are unable to complete what you need to within the allocated time.

3. Batteries andstorage cards: Irrespective of the kind of battery your recorder uses, carry an extra set of batteries and a charger. Having to return to your base just to charge batteries can mean losing out on time, energy, money, and most importantly, the people you wish to interview. By the same logic, keep an extra storage card of the kind used in your recorder handy. An interview could go longer than you thought, and you don't want to miss out on recording something just because you ran out of space.
4. Mobility: Do you have your entire production kit neatly packed? Are the audio recorder, batteries, cables and any accessories neatly stored? Do you have a headphone? Are all pieces of equipment working properly? Are the headphone and microphone cables coiled properly? Are non-production items like biscuits or a water bottle in a different bag? Check for all of these before setting out onyour recording trip.

The production phase: Key points to remember

You make or break your production depending on how you handle this phase. You may have all the equipment—microphones, recorder, batteries, storage cards, headphones—organized and ready. But if you don't use them right when you conduct the recording, no amount of post-production work can make your programme better.

Use the following checklist to ensure that all goes well during the field recording:

- Make sure your recorder is working. Check the recorder's display once in a while.
- After pressing the record button, start by saying the subject, location and time of your recording. This may not be always possible but is definitely possible before conducting interviews. In case you lose your notes later on, you will be at least able to understand what you have recorded.
- Use a quiet location to record interviews. Switch off fans and air conditioners while recording interviews. Microphones have a tendency to pick up those sounds.
- Record the room tone (when nobody is talking) for about 3–4 minutes. This tip is also useful when recording interviews outdoors. Whileediting, this tone can be used across the interview on another track to maintain continuity.
- Always point the microphone to the source of the sound. Place it at least 3 to 6 inches away from the source.
- Microphones need a critical distance to record the right quality of audio. If you take the microphone too close, the audio will be distorted and the recording may have pops of air from the speaker's mouth. On the other hand, taking it far away from the source of the sound is likely to result in a weak and hollow sound quality.
- If you are using handheld microphones, always talk across them. The microphone is best placed at the chest position. However, if you are using a uni-directional microphone, you should keep it close to the mouth.

- Sit down when you have to interview children. This way you can place the microphone at a comfortable position for both of you.
- Do not conduct interviews while walking. People tend to get distracted and there may be long pauses in sentences.
- Do not ask close-ended questions. Asking a question that begins with ‘do you...’ invariably results in a one word ‘yes/no’ answer. Ask questions that begin with ‘what’, ‘how’, ‘why’.
- When recording interviews with more than one person, alert them in advance to speak only when the recorder/microphone is pointed at them.
- Do not place the recorder on the ground/table to record interviews. If you have an external microphone attached, hold the recorder in one hand, where you can see the audio levels on the screen, and the microphone in the other. whileusing abuilt-in microphone, hold the recorder in whichever hand you find it comfortable—but remember to keep an eye on the audio level meter!
- Use microphone stands where possible, to keep the microphones steady.
- When using multiple microphones for a talk show, place the second microphone at least twice the distance away from the user as the first. (For example, if you place one microphone at a distance of one foot from two speakers, place the next microphone at least two to three feet away from the first one.) This helps prevent phase cancellation, a phenomenon where the signal from one microphone may be partially cancelled out by the signal from the other, resulting in a poor recording.
- In a noisy environment, position the interviewee so that his back is against the area with the lowest noise or disturbance. That way, when the microphone is pointed at him or her, the noisiest areas will be behind the microphone, and the person’s voice will be recorded more clearly.
- Nod your head during an interview. Avoid ‘hmm’, ‘uh’ and ‘oh!’, although not at all times. Too many aural responses from you might make it too conversational, and result in audio that is difficult to edit later.
- Use directional microphones wherever necessary.
- Switch off/fade out microphones when not in use.
- Use a windshield when using microphones outdoors, and pop filters when recording indoors. This avoids wind or breath from hitting the microphone.
- Always use headphones/speakers when recording sound outdoors/ indoors.
- Never allow the audio to consistently touch the ‘red’ mark or zero level on your audio recorder’s level meter.
- Record Sound on Tape (SoT) without fail. This is a track containing just the sounds in the environment, without the speakers’ voices. Also known as ambient sound, this lends character to your programme by providing the listener a clue to the environment in which the interview or discussion is being conducted.

- Eye contact is very important when recording interviews. Look the person in his or her eye when asking questions.
- Listening to your interviewee is also extremely important. It helps you ask additional questions that might bring new insights to the programme.
- Always monitor sound from the recorder as you conduct the recording. You will know immediately if there has been any kind of disturbances in the recording.

It is also important to remember some key points with regard to taking care for your microphone units:

- Do not tap or blow into any microphone to test it. Just talk.
- Keep microphones in their pouches when not in use.
- Store microphones safely in an almirah or a box.
- Keep microphones away from moisture, water and fire, as well as physical shocks of any kind.
- Switch off the microphones after use, if they have in-built off/on switches. This is especially important for battery powered condensor microphones.
- Remove the batteries if not in use. Replace them with new but similar type batteries periodically.
- Use high-quality connectors and cables for microphones.
- Do not pull out a connector holding the cable. Always grasp the connector (jack) to disconnect.
- Do not leave heavy microphones dangling from a cable. Do not swing them holding the cable either.
- Do not use any liquid whatsoever to clean microphones.

Activity 5.3

Using the tips mentioned above, record a paragraph from a book using a recorder with a built-in microphone.

Do this exercise in different situations like a small room, big room and outdoors, keeping the microphone once close to you and once a bit far away from you.

Playback the audio you have recorded and notice the differences in each of the situations.

13.6 Portable Audio Mixers

You have earlier learnt about studio audio mixers. Here you will get to know the functions performed by audio mixers:

- Control the volume of signals coming from various inputs (microphones, audio players, etc.) using faders. The faders help in regulating the input levels of sound coming from various sources.
- Combine and balance the inputs coming from various sources, by sliding the faders up and down. For example, in an audio mixer with six inputs, you connect six microphones to record a musical score. Controlling each source by way of its dedicated fader helps one in balancing the input of each instrument, so that we make the combined output pleasant to hear.
- Equalize the audio, that is, manipulate the frequency characteristics of the input sound. Equalizing involves controlling the audio signal by increasing some wanted frequencies and decreasing the unwanted ones. For example, you can reduce hiss in speech by equalizing the input.
- Auxiliary Send ("Aux Send") is used to send the output to other external devices like an effects generator. The output from the Aux Send is again taken back into the audio mixer, mixed and routed through the final output.
- Route the mixed and equalized audio signals to a specific output (a recorder/speaker/headphone).
- Enable monitoring of all the functions mentioned above through speakers/headphones as they are being recorded/broadcast.

Audio mixers come in handy both in live and post-production situations. In a live situation, audio inputs are first balanced and equalized and the recording done.

Professional audio mixers also have a provision to connect a headphone or a pair of speakers to monitor the output. Some audio mixers are also capable of providing the phantom power supply to microphones when needed. It is always better to purchase audio mixers that can provide phantom power supply.

Portable audio mixers come in handy for use in the field when there is more than one source of sound. It allows one to combine the signals from multiple audio sources, be it microphones or other audio players, and mix them all. They work either on direct power supply or batteries. The output from the field audio mixer is then connected to the input of an audio recorder. The audio recorder then records the mixed audio from various sources. Figure 5.12 shows a model of field mixer.

Figure 5.12: Portable audio mixer.
(Source: Jnan Taranga, KKHSOU, Guwahati)

Professional portable field mixers have the following features:

- Can take in any number of audio inputs from 2 to 6
- Provide phantom power for condenser microphones
- Gain and level controls
- Monitor headphones
- Can be low cut filters by cutting out low frequencies while recording outdoors
- Can act as limiters by preventing distortion of audio even when it is very high
- Provide balanced outputs to connect to a recorder

If you have to work with portable audio mixers, ensure the following:

- It has enough audio inputs.
- It has a warning signal for signal overloads.
- Don’t mix or overlap audio and power cables. Keep them as far as possible from each other.

13.7 Sound Cards

A sound card is a peripheral that is connected to a computer. It is sometimes also called a sound board or an audio card.

Most computers come with some form of a basic sound card to edit and playback audio. So how does one identify if a computer has a sound card? If your computer has inputs for a microphone and a headphone on the back of the CPU (Central Processing Unit), which is the box that houses your computer's hardware, your computer has a sound card. It must be noted that while some cards only playback audio, duplex sound cards enable simultaneous input and output of audio, and also enable audio recording.

Some of the motherboards used in computers have good sound cards that are integrated into the motherboards themselves. However, most computers used for professional audio usually need an independent sound card. These are usually plugged into the Peripheral Component Interconnect (PCI) slot on the motherboard. Figure 5.13 shows a sound card of this kind.

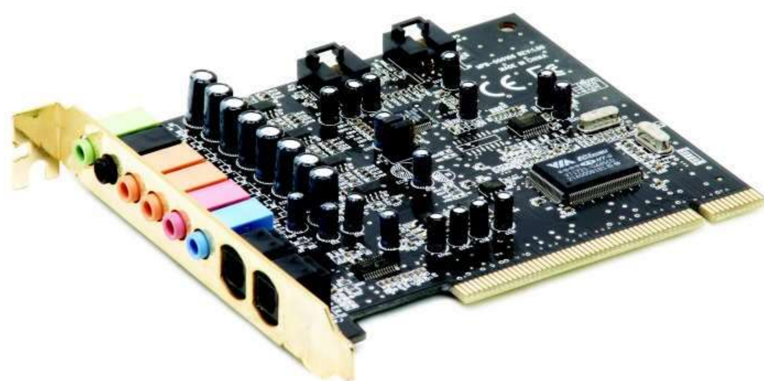

Figure 5.13: An internal sound card. Only the silvered surface at the front and thecoloured connectors are visible at the rear of the CPU cabinet.
(Source: WikiMedia Commons, *http://tinyurl.com/32ssttc*)

A sound card also acts as an analogue to digital convertor (ADC). In other words, it converts analogue signals coming from your microphone to digital bits that can berecorded/stored on the computer. It also has a digital to analogue convertor (DAC) to output the digital signal through speakers, which need an analogue signal.

The purchase of the sound card will depend on your requirement and budget. Budget sound cards can cost anywhere between INR 1,000–5,000. A professional high-end sound card could cost INR 1,00,000or more! For purposes of editing audio, a sound card that is in the range of INR 2,000–3,000 should be quite adequate.

13.8 Digital Audio Workstations (DAWs)

A digital audio workstation (DAW) is nothing but a computer one can use to record audio in a studio or even use for the post-production work of radio programmes.

In the early days of digital audio editing on a computer, one would have to connect hardware like an audio mixer or effects generator to the computer. While the software would help in merely editing together all the audio, all sweetening of audio (including balancing, equalizing, and applying audio effects) would have to be done by the external hardware. Of late, however, computers above a certain grade, with even moderate sound cards, are capable of doing everything from assembly of the programme through to the mixing, balancing, equalization and effects works within the computer's software interface itself.

Today's computer and audio recording/editing software often come with built-in software-based mixing consoles, plugins and effects, which help in completing everything from recording to post-production work right on the computer.

The key things to look for in a computer to be used as a DAW are as follows:

- A high-end processor (the Intel i-series processors work well)
- A power supply (SMPS) of 400W or greater
- Random Access Memory (RAM) anywhere between 2–4GB (more is better)
- A hard disk of a minimum 500 GB capacity, with 1 TB preferred
- As many USB 2.0 (Universal Serial Bus) ports as possible to import and export audio from the system
- A good sound card—one with an external breakout and multiple connector options is always preferred
- A pair of good speakers to monitor audio

There are several choices for digital audio processing software available in the market, ranging from Free and Open Source Software (FOSS) options to proprietary solutions. More about their functions will be discussed in Units 16 and 17 in Module 5.

Activity 5.4

Turn off a computer in your studio, unplug it and try to open it up. If you do not know how to open it, ask for assistance. You will need to open some screws on the case to lift one side of it.

Now try to identify the sound card in the computer's CPU tower. It is normally situated near the slots where you plug in a headphone or an external microphone to the computer. See if it has a (green) slot to connect a headphone or a set of speakers. See if you can identify the make of the sound card. Now switch on the computer and play back some music on the computer with speakers connected to the appropriate socket. You should be able to hear the audio on the speakers.

13.9 Let Us Sum Up

A microphone and an audio recorder are the two minimum pieces of equipment required to record audio on the field.

When recording audio, one needs to take care of the ambience to avoid external noise from getting recorded. Monitoring audio using the right type of headphone is imperative if you have to record good audio.

One must also be able to identify different kinds of audio connectors in order to be able to use them with different pieces of equipment in different situations.

A digital audio workstation is a computer that uses a software to perform edits, sweeten audio and output audio in different formats including .wav and mp3 formats.

5.13 Model Answers to Activities

Activity 5.1

The quality of audio recorded largely depends on the microphone you use, your distance from it and the direction of the source of the sound. When you speak in front of it, the quality of audio is better than when you speak sitting behind it or next to it. In the case of the latter, the audio will sounda little hollow. You will also notice major differences when you record audio using uni-, bi- and omni-directional microphones. You will notice that the audio recorded using an omni-directional microphone has more noise than the ones recorded using uni- and bi-directional microphones.

Activity 5.3

When you record audio in a big hall, and the microphone is far away from the primary audio source, the audio sounds hollow and full of reverberations. This is because sound waves hit nearby objects and are reflected back into the microphone. Similarly, there is considerable noise when you record audio outdoors.

On the other hand, when the recorder is very close to the source, the possibility of eliminating hollow sound is more. Therefore, the distance from the source of the sound and the ambience plays a major role in the quality of audio you record.

UNIT 6

Free and Open Source Software in CR

Structure

6.1 Introduction

There are many misconceptions about the words free software and open source software in the minds of computer users. Many think that free software means you can download it from the Internet without monetary payment. There is also a belief that free software is acceptable for amateur work, but not for professional work. Some people believe that free software is difficult to use, and that there is no support available, that it is good only for computer 'nerds'—people who are expert computer users.

In this unit you will learn the correct meaning of free and open source software, the philosophy behind the free software movement. You will also understand their usability, prevalence and relevance in the community radio (CR) sector. It may be noted here that 'Free Software' and 'Open Source' software have different meanings, though they are quite often used interchangeably. In this unit, you will learn about the difference between the two terms, and some of the open source software used the world over in CR stations.

While going through the unit, you will need to do four activities (the model answers of which are given in Section 6.6). You will need approximately tenhours to complete this unit.

6.2 Learning Outcomes

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

- define open source and free software.
- describe the historic and philosophical background of the open source movement.
- analyse the pros and cons of free software, as compared to its commercial counterparts.
- list available open source software that are useful in CR stations.

14.3 What is Free and Open Source Software?

The free and open source software movement heralded a new era of knowledge building society—one which was non-hierarchical, adhering to the principles of equal access, freedom to use, adapt, modify and redistribute. Both the terms, though quite often used interchangeably, are not really synonyms. Both the terms are underpinned by a similar value structure, even though they are different in their philosophical approaches.

The Free Software Foundation has been promoting the term Free Software, whereas the Open Source Initiative has been using the term Open Source Software.

14.3.1 Definitions

There are no classical definitions available for either of the terms. The Free Software Foundation's explanation published in February 1986 pointed to two fundamental principles to describe the term free software. First and foremost, the term 'free' does not refer to the price or cost of the software. Instead, it refers to the freedom it gives you to use, modify or redistribute the software.

For example, in the sentence "All human beings are free", the word free does not refer to a cost or monetary value, but to the concept of freedom. On the other hand, if I say "My restaurant is going to distribute lunch for free today", the meaning of the word free refers to the cost or monetary value. So, free software is free as in 'freedom' and not as in 'free lunch'!

The second principle that the Free Software Foundation emphasized was the freedom to change a software program, so that the control over the program lay with the user, rather than the original creator. Naturally, this means that the source code must be made available to the user, so that he or she can make changes in it.

These two primary principles were later codified as four 'Levels of Freedom', to which a software had to confirm in order to qualify as free software. They were as follows:

- Freedom 1: The freedom to run the program for any purpose.
- Freedom 2: The freedom to study how the program works, and change it in order to make it do what you wish.
- Freedom 3: The freedom to redistribute copies so you can help your neighbour.
- Freedom 4: The freedom to improve the program, and release your improvements (and modified versions in general) to the public, so that the community benefits as a whole.

In 1997, Bruce Perens, a renowned software programmer, published the Debian Free Software Guidelines, which were subsequently adopted as the core principles behind open source software by the Open Source Initiative, the organization that promotes the term 'open source software'. Though the guidelines were similar to those that governed the term 'free software', they further emphasized the philosophical differences between the two terms.

The Debian Free Software Guidelines referred to the "Cathedral Vs Bazaar", a much talked about article written by Eric Raymond in the same year, which differentiated between two models of knowledge building by comparing them to contrasting architectural styles. The way a cathedral is built is very centralized: very few people or architects have a complete vision of the building, and everyone else is a cog in the wheel, executing a sub-task within the larger purpose of cathedral building. The roles are very well defined and the whole idea is based on hierarchical relationships and superior/inferior roles. Traditionally, the knowledge or commercial software was built in this way. For example, the complete source code for the Microsoft Windows operating system is a highly guarded secret, and very few people have access to it.

On the other hand, the bazaar approach of knowledge building is open, non-hierarchical and user-driven. Like a local bazaar (market), everyone in a society has the freedom to use, adapt, modify and redistribute a given piece of software. There are no superior authorities to decide who does what; roles are not defined; and everyone has the freedom to do what he or she likes to do.

14.3.2 History

The concept of free sharing of technological information is not new. It has been in existence since time immemorial, far before the computer was even invented. Even our mothers and grandmothers have shared information and procedural tricks with friends and family! It is in the same spirit that the twin concepts of free and open software were created.

Even in the field of software, informal sharing of source code was an accepted concept much before the current movement of free and open source software came into existence. In the early days of computers, in the 1950s and 1960s, all software were bundled with their source codes as there was no standardization in hardware. For the software to work on different hardware setups and operating systems, some modifications were absolutely necessary—and this meant each user needed access to the full source code.

By the early 1970s, however, the winds of change had started blowing. Computers came out of the academic and research institutes and into the corporate offices. The growth of the software industry was inevitable, as the corporate houses and individuals thought of diverse ways of using computers. Around the same time AT&T, the global telecom giant, gave the UNIX operating system free of cost to governments and academic institutes. UNIX was free, but not open source, as it did not come with the permission to modify or redistribute it. In the 1980s, AT&T stopped free distribution and started charging for the software. By then, the users were so used to UNIX that most of them paid for the software. This was the first major instance of software licensing. However, there were some exceptions: software like X Window System, which enabled the graphical interfaces, continued to be distributed as an open source software.

The free and open source software movement got a major boost with the advancement of the Internet. The Internet provided a much-needed sharing platform for small and big initiatives towards writing software. With increasing net-based connectivity, many organizations and individuals started websites to share software.

In 1983, Richard Stallman started the GNU Project to develop a free and open source kernel for a new computer operating system similar to the UNIX OS earlier developed by AT&T. However, there was no real kernel as part of GNU Project till the time Linux appeared on the scene.

It was in 1989 that the first version of the GNU General Public License was published. The second version was published in 1991. In the same year, Linus Torvalds shared the first version of the Linux kernel, which drew attention from volunteer communities across the world. In no time, Linux emerged as the first and complete free and open source computer operating system.

The development of Linux demonstrated to the world that the bazaar model of development was viable and efficient. In addition, it made complete business sense to have decentralized, customizable platforms.

Linux is packaged in a format known as a Linux distribution (or distro). They are now available for desktop computers, servers and embedded devices. There are now a variety of popular distributions created by a number of organizations and individuals, which include Debian and its derivatives such as Ubuntu and Linux Mint, Red Hat Enterprise Linux (and its derivatives such as Fedora and CentOS), Mandriva/Mageia, openSUSE (and its commercial derivative SUSE Linux Enterprise Server), and Arch Linux. Linux distributions include the Linux kernel, supporting utilities and libraries, and usually a large amount of application software to fulfil the distribution's intended use.

The Linux distribution for desktops is generally accompanied by the X Window System that provides the graphical interface. GNOME, KDE, Xfce and Unity are some of the prominent desktop environments that a desktop user can choose from. By default all of them use Mozilla Firefox as web browser, Libre Office or Open Office as the office suite, and GIMP as the image editor. There are usually a host of multimedia applications which are bundled as part of the distribution.

The Linux distribution for servers typically includes Apache HTTP server, MySQL database, and PHP5 as web application. This combination is known as LAMP, and is the most popular configuration for the web servers. LAMP usually includes SSH server, an FTP server for the hosting of remote management applications.

Apart from Linux, there are a few more significant kernels for operating systems. FreeBSD and NetBSD, both derived from 386BSD, are noteworthy among them. The development of FreeBSD as well as NetBSD started around 1993. In the last 20 years, it has gained significant ground. Technology giant Yahoo has been using FreeBSD since the year 2000.

14.3.3 Current Scenario

Eric Raymond's famous article, *The Cathedral vs Bazaar*, had an electrifying effect on many software programmers and companies; and many readily adopted the bazaar model of knowledge development. The first among the companies that made a philosophical statement through their actions was Netscape Communications Corporation (NCC). The company released its flagship browser Netscape Navigator as free software. The code for Netscape today forms the basis for Mozilla's Firefox browser and Thunderbird email client.

Thanks to the comparative marketing muscle of proprietary software companies like Microsoft and Apple, open source software's penetration in the consumer market has been very slow. But on mission critical computing, super computers, web servers, DNS servers and similar applications, Linux and other Unix-like operating systems dominate by a significant margin.

The chart above will show you how the most powerful supercomputers in the world, as listed with http://top500.org are using Unix-like or Linux operating system.

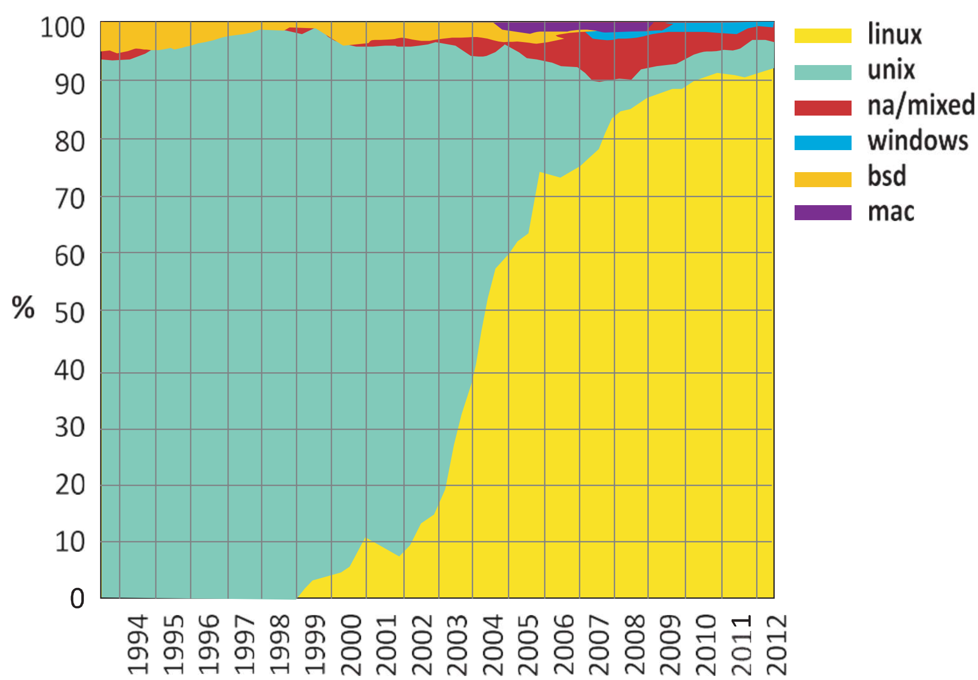

Figure 6.1: A graphical representation of the relative shares of the various popular operating systems on the top 500 supercomputers in the world (1994–2011).
(Source: WikiMedia Commons, *http://tinyurl.com/qxovbma*)

In the broader server industry, Apache HTTP server is the indisputable market leader. Similarly, as per the industry estimate, more than 65 per cent of web servers in the world are running on Linux server distribution. Among desktop computers, Microsoft's Windows operating system is still dominating, though its market share has dropped with every version over the last decade. A large chunk of what Windows desktop version lost has been captured by Apple's iOS.

On the other hand, there is a concerted movement towards greater acceptability of Linux desktop with public institutions. The federal government of Brazil is well known for its support for Linux. The Russian military is also reportedly creating its own Linux distribution for strategic use.

The Indian state of Kerala has made it mandatory for all state-run schools to use Linux on their computers. Spain and Germany in Europe, as well as the continents of Australia and Africa are emerging as the powerhouse of Linux users.

Linux is getting popular in the netbook market as well. Companies like Asus and Acer have started shipping netbooks loaded with specially customized Linux operating systems.

Apart from the operating system domain, the open source software movement is taking centre stage in other areas of computerized activity as well. For example, for computerized EPABX telephony systems, Asterisk is the most advanced system and is used by all leading telephone companies. For audio streaming, Icecast and Shoutcast are the leading types of software developed by the open source community.

The Android mobile OS, used on many mobile phones today, is a classic open source development. Google developed Android based on a Linux kernel, and has kept the source code accessible and revisable. Within a few short years, Android has emerged as the primary challenger to Apple's proprietary iOS, used on its flagship iPhone and iPad devices.

Activity 6.1

For this activity, you will require access to a computer where you have permission to install software. You will also need an installable copy of any Linux distribution, on a CD or downloaded from the Internet and then you will need to burn it to a CD.

Install any distribution of Linux on a computer to work side-by-side with the existing operating system.

14.4 Open Source Software for CR

It has been observed the world over that CR stations require more openness, flexibility and higher levels of customization in their operations. They need to be able to expand their processes and systems organically over a period of time, and even shrink them, under certain circumstances. Also, the technologies used in CR stations need to be cost effective. Given these requirements, the open source and free software movement is a natural fit with CR movement, both of which strive to provide equal access, transparency, openness and scalability to communities.

Across the globe, many community media practitioners have preferred open source software for the following reasons:

- Freedom to use and share the software
- Freedom to modify and adapt the software to their unique situations
- Free support from the global community, which is often seen as far superior to the paid support by a company providing a proprietary software
- Cost effectiveness

In this section, we will briefly examine a sampling of time-tested open source software used in the various functions of a CR station. It should be noted that the software listed in this section constitute a very small part of a large open source software pool. The selection is also subjective and in accordance with the personal preferences of the author. Students should feel encouraged and free to explore other open source software, and not restrict themselves to what is listed here.

14.4.1 Sound Recording, Editing, Mixing and Mastering

Sound recording and editing are core activities in any CR station; and every technician working in a CR must have a better than fair understanding of the software used for the purpose. There are literally hundreds of free and open source software available for these purposes, many of which are well-known. Few of them which are not known are described below.

Audacity

Audacity is one of the most commonly used audio recording and editing software used in CR stations the world over. It is a non-linear, multi-track and feature-rich editing system that works on Windows, Mac or Linux platforms. The main features of the software include the following:

- Recording live audio
- Converting magnetic tape recordings and LP records into digital recordings or CDs
- Editing Ogg Vorbis, MP3, WAV or AIFF sound files
- Cutting, copying, splicing or mixing sounds together
- Changing the speed or pitch of a recording

There are many plugins and add-ons you can find on the Internet to extend the functionality of Audacity, and use the software as you like. Figure 6.2 shows you what the interface looks like.

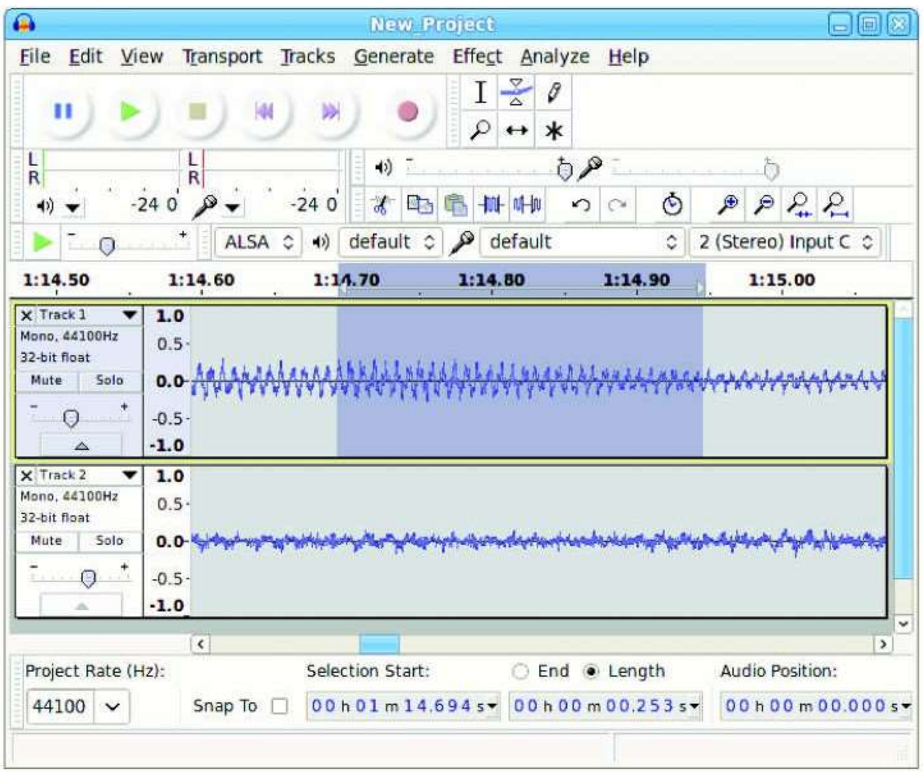

Figure 6.2: The Audacity editing interface.
(The screenshot was captured by the author)

Ardour

Ardour belongs to the JACK Audio Connection Kit environment. It is one of the most efficient and professional audio editing and recording software available to the open source community. It is a complete digital audio workstation, useful for live recordings, concert recordings, and composition. For a complete list of features, visit http://www.ardour.org.

Figure 6.3 shows you what the Ardour interface looks like.

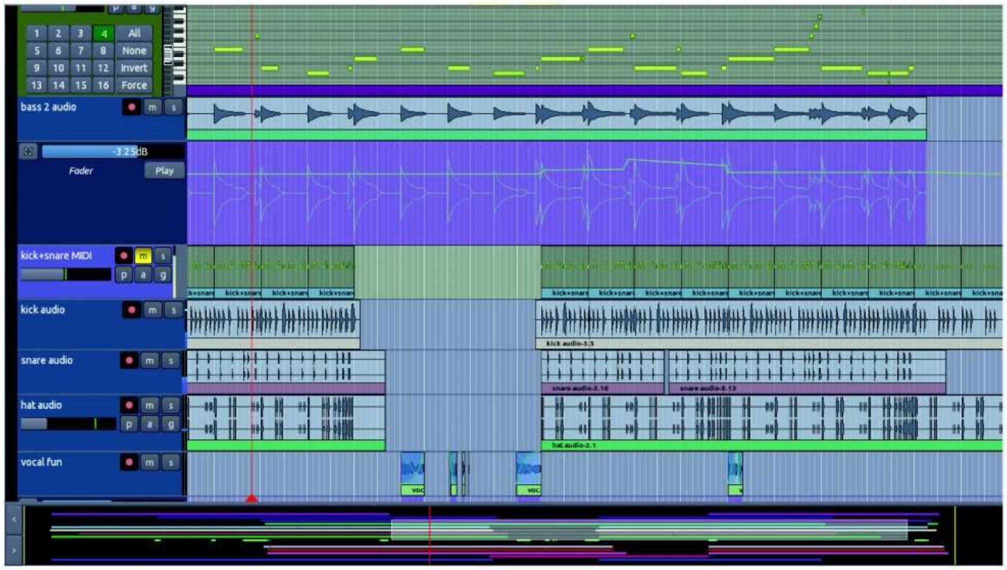

Figure 6.3: The Ardour software interface.
(The screenshot was captured by the author)

JAMin

JAMin is the Jack Audio Connection Kit Audio Mastering interface. JAMin is an open source application designed to perform professional audio mastering of stereo input streams. JAMin is licensed under the Gnu Public License (GPL).

The features of JAMin are as follows:

- Linear filters
- JACK I/O
- 30 band graphic EQ
- Spectrum analyser
- 3 band peak compressor
- Lookahead brickwall limiter
- Multiband stereo processing
- Presets and scenes
- Loudness maximizer

Figure 6.4 shows you the JAMin software interface.

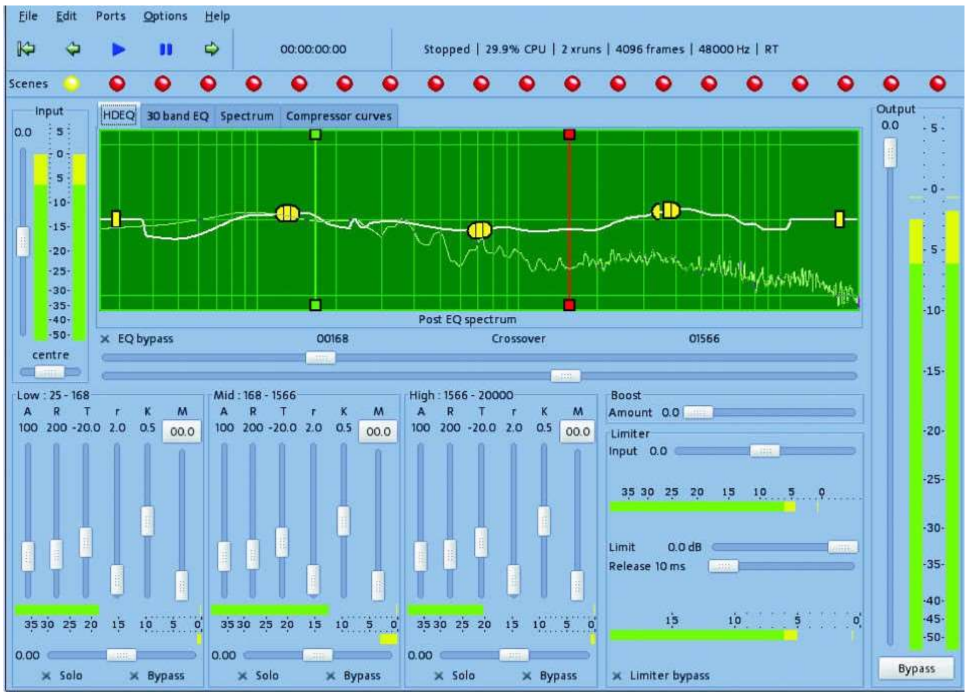

Figure 6.4: The JAMin software interface.
(The screenshot was captured by the author)

Activity 6.2

Carry out a comparative analysis of the software listed below:

1. Audacity vs Adobe Audition
2. Ardour vs Nuendo
3. GIMP vs Adobe Photoshop
4. Mixxx vs WinAmp
5. Mozilla Firefox vs Internet Explorer

List the pros and cons of each type of software as part of your analysis.

14.4.2 Radio Automation

Radio automation software are used for different functions in a radio station. They are used for scheduling (presetting the time when a particular programme plays), logging (listing the order of play of the aired programmes) and playout (the playing of a preset list of programmes). Many radio stations in India use just a simple playlist software like WinAmp as playout system; but the use of a well-configured radio automation software can give a professional touch to a CR station, with pre-defined fades-in and fades-out for programmes, jingle management, advertising management, live assist, telephony and many features that come in handy in a radio station.

There are not many radio automation software in the free and open source domain. There are some free software like Zara Radio Free, but many of those are available with no clear commitment to open source. In this section, you will be introduced to three types of radio automation software, which have clearly stated their commitment to the open source movement.

Airtime

Airtime is one of the most advanced server-based radio automation system, developed by Sourcefabric. It runs on a web server to enable scheduling, logging, playout, live assists, and also streaming to any Internet radio. This software was earlier known as Campcaster.

To be able to install airtime, you need to be proficient in installing web server with PHP, Apache2, mail server and PostgreSQL. The installation requires some practice before you can master it. However, once it is installed, it is an easy software to use. The software does not run on your computer. You need to access it via an intranet or the Internet on your web browser like Mozilla, Chrome or Internet Explorer. As it is server based, the software for the end-user is cross platform. That means you can run any operating system—Windows, Mac OSX or Linux—and the software will be accessible in exactly the same way. Figure 6.5 shows you the Airtime media library and playlist builder interface.

Figure 6.5: Source fabric's Airtime automation software: Media library and playlist builder (The screenshot was captured by the author)

For a complete list of features, please visit http://www.sourcefabric.org. If your CRS has good Internet connectivity, you may also avail the hosted solution offered by Airtime, provided on the professionally managed servers of Sourcefabric, the team that has designed airtime.

Figure 6.6 shows the scheduling interface in Airtime.

Figure 6.6: Source fabric's Airtime automation software: Schedule builder interface(The screenshot was captured by the author)

Rivendell

Rivendell is a complete radio broadcast automation solution, with facilities for the acquisition, management, scheduling and playout of audio content. It is a feature-rich software with further emphasis on touch-based operations, that is, it is optimized for touch screen based computers.

Rivendell uses industry standard components like the GNU/Linux Operating system with Audio Science HPI driver architecture. It also works with the open source JACK Audio Connection Kit and with the MySQL database engine.

One of the limitations of Rivendell is that its documentation is limited to few distributions of Linux, and that it does not have a repository-based installation service. Figure 6.7 shows a view of the Rivendell administrative panel, from which its primary functions can be controlled.

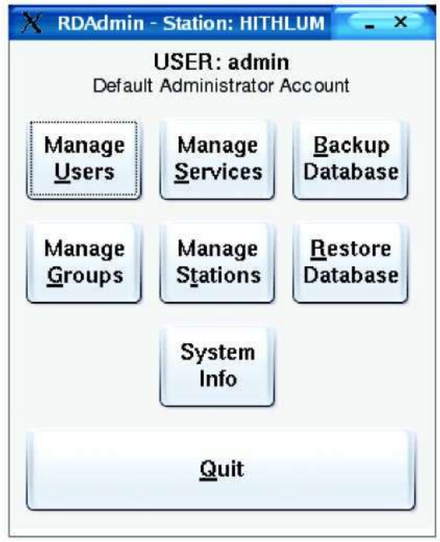

Figure 6.7: Rivendell software Administrative panel(The screenshot was captured by the author)

Figure 6.8 shows a view of the Rivendell's built-in editing capabilities, which allow you to correct audio without moving into a separate software environment.

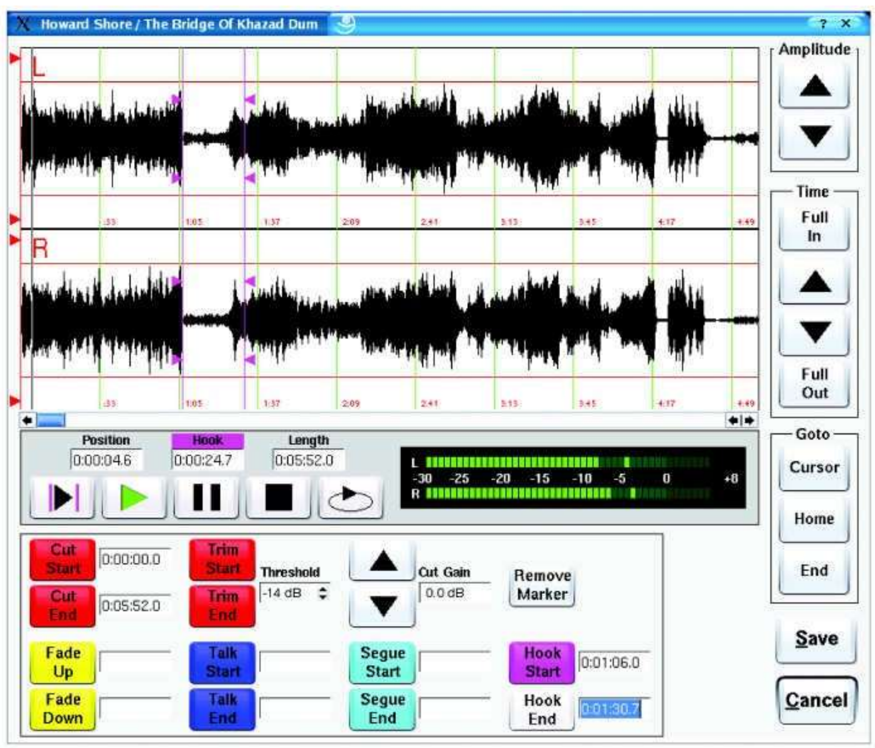

Figure 6.8: Rivendell software's built-in audio editor(The screenshot was captured by the author)

GRINS (Gramin Radio Inter Networking System)

GRINS is one of the most widely used radio automation software in Indian CR stations, and has been developed by the Indian organization Gramvaani. Apart from offering the standard features of a radio automation system (namely scheduling, playout and logging), GRINS also offers complete integration of mobile telephony within its interface, including SMS (Short Messaging Service). Released under the Apache license version II, GRINS currently works only on Ubuntu 12.4 LTS version, which is a Linux-based distribution.

Some of the features of GRINS are as follows:

- Full telephony integration without need of a telephone hybrid or changes to mixer settings. You can receive calls, record calls, put calls on air, have live call-in programmes. You can even maintain a searchable database of your callers within the GRINS interface
- Preview of audio. Since GRINS uses multiple sound cards, you can preview audio over your headphones even while some other programme is playing live.
- Streaming audio support. GRINS supports streaming over IceCast or ShoutCast servers, making it Internet radio ready.

Figure 6.9 shows you the GRINS telephony interface.

Figure 6.9: A screenshot of the GRINS telephony console.
As you can see from the image, there are provisions to make, receive and log calls.
(The screenshot was captured by the author)

Activity 6.3

Using any open source audio editor, edit a three-minute monologue on open source technology. It may be useful for you to write the script first before recording it.

After editing your own speech, take an output in an open source format.

14.4.3 Other Useful Software

The open source domain is a goldmine of highly useful software, with hundreds of options for every possible application need. In this sense, it is worth noting some other key open source software that could be of use in a CR station.

One such application is called IDJC, which is the short form of Internet DJ Console. The software is designed for Internet-based radio jockeying. As a radio announcer or jockey, you can sit anywhere in the world and run either Internet radio or even FM radio. The software has two players and you can use them in tandem to play a programme on one while you cue the next programme on the other. It has auxiliary inputs as well as a complete telephony function built into it; and you can use Skype or a SIP-based telephone to either receive calls or make calls to your listeners. It also lets you stream audio to six different Icecast or Shoutcast servers simultaneously.

The limitation of IDJC is that it works only on Linux and that too specifically with the JACK Audio Connection Kit. To use it, you need to be reasonably well versed with both associated software. Figure 6.10 shows you the IDJC interface.

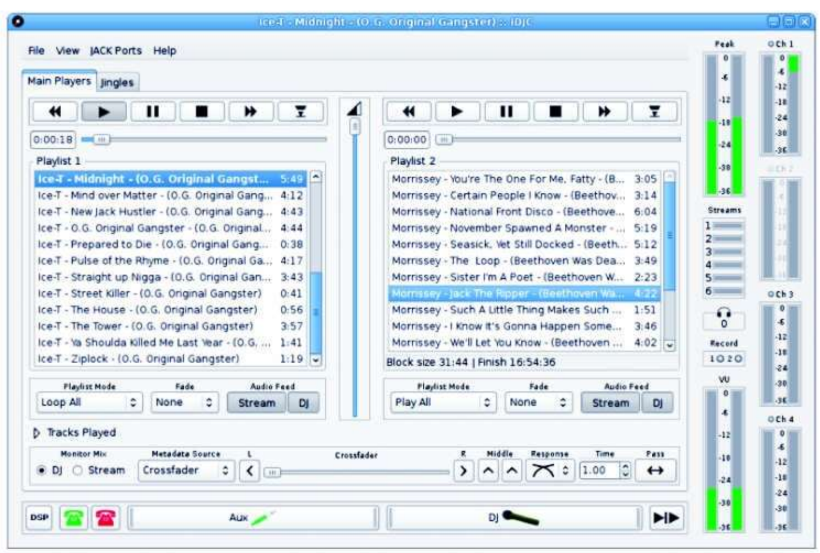

Figure 6.10: A screenshot of the IDJC interface.
You can play items from the two queued lists on the left and right alternately.
(The screenshot was captured by the author)

Another piece of software that is similar to IDJC is Mixxx. This software also has two players and a live assist. It can also stream to an Icecast or Shoutcast server. As compared to IDJC, there are two distinct advantages of Mixxx. The first is that it is a cross platform software, which means it can run on Windows, Linux or Mac. The second is that it can run on Advance Linux Sound Architecture (ALSA) or Jack Audio Connection Kit. Moreover, the look is skinnable, meaning you can customize its look and feel as per your preferences. There are a lot of 'skins' available for the software.

Activity 6.4

Compile a list of open source software that can be used for CR purposes. The list should be divided into two heads: 'Recording, Mixing and Editing'; and 'Radio Automation'.

14.5 Let Us Sum Up

In this unit we have understood that 'Free and Open Source Software' does not necessarily mean software that are available for free, even though many of the open source software are actually available free of cost. Free and open source software are those which come with the freedom to be used, modified and redistributed. There is also no restriction on putting open source software to commercial use. In fact, there are a large number of companies today that earn a tidy profit based on open source technologies.

You have also understood the philosophical basis on which the entire open source software movement rests on: equal access, sharing and community knowledge base are some of the key concepts that are driving the open source movement.

It may be noted here that the proprietary development model has not been able to match the speed at which open source community has been able to develop complex software. Also, many types of commercial software are actually the derivatives of work that has happened in the open source domain. Philosophically, open source environment is more suitable to CR and efforts must be made to avoid using proprietary software.

14.6 Model Answers to Activities

Activity 6.1

While investigating Linux distributions alongside your existing operating system, you must have found Ubuntu 12.4 or Fedora 17 or Mint or Debian. These are all Linux-based operating systems, either based on the Debian core or the RedHat core. You would have also noticed that all Linux distribution packages come with a package called Grub, which allows you to have multiple operating systems on the same computer. The same program also allows you to choose which operating system you want to use at a given point in time.

Activity 6.2

The comparative analysis must have shown you that software in open source domain give more freedom and functionality in some areas, while they fall short of their commercial counterparts in others. There are two reasons for this. The first reason is that in specialized areas like sound editing, proprietary software have had a much longer experience and history, while the open source software alternatives have been in existence for a comparatively shorter period of time. The second reason is that many audio or video codecs are of a proprietary nature, and are therefore not accessible to the open source community, preventing them from creating extended functionality based on such codecs.

Activity 6.3

While recording and editing your own speech on an open source based digital audio workstation, you must have noticed that you can perform most operations as efficiently as you can in any proprietary software commercially available in the market. You may also have noticed that there is no perceptible difference in the final output. At the same time, you may have fumbled or struggled to invoke some of the same functionalities, as some of the commands and terminologies are different in the open source alternatives.

Activity 6.4

While compiling the list of open source software, you must have noticed that in addition to the software listed in this unit, there are many more options available in the open source domain. You may have also come across software that are free and open source but at the same time also have a price tag. This underscores the fact that free and open source software does not necessarily mean 'free of cost'.

14.7 Additional Readings

- Open Source. (April 6, 2013). Retrieved from https://en.wikibooks.org/wiki/Open_Source
- The first project to join the open source movement and a great source of software (2013). Retrieved from http://sourceforge.net
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UNIT 7

Telephony for CR

Structure

- Introduction
- Learning Outcomes
- Conventional Landline Systems
  - Speakerphone–Microphone Combination
  - Transformer-based Interface
  - T elephone Hybrid
- GSM/CDMA Solutions
  - Bluetooth Interface
  - GSM Dongle (Modem)
- Voice Over Internet Protocol (VoIP)
- Use of SMS
- Let Us Sum Up
- Model Answers to Activities
- Additional Readings

7.1 Introduction

In the previous units in this course, you learnt that CR is meant to be a participatory medium of communication. The success of CR can be determined by the extent of participation of the community it is intended to serve. Though there are many ways to ensure and encourage participation, one of the most significant ways is the telephone.

You must have heard many radio programmes on commercial as well as on CR stations that make use of phones to connect the listeners to the station. These are generally known as phone-in programmes. In phone-in programmes, a conversation with a listener is either recorded or incorporated into a 'live' programme on the broadcast system. Many radio stations also use the phone system for conducting polls or recording listener feedback.

For many years, wired landline telephone systems were the only option; and the cost and effort of cable laying meant phone densities even in urban areas was very low. However, over the past two decades, the telephone system in the country has been revolutionized by the advent of mobile (cellular) telephony. Tele-density, which means the availability of telephone system per unit of population, has increased multifold, not only in the cities but also in the rural areas. This has made the use of telephony in broadcast even more important a tool to engage the audience and in making the programmes more participatory.

In this unit, we will discuss various tools to incorporate landline as well as mobile telephones into your broadcast environment. You will also learn about the conventional and modern ways to bring your listeners' voice on the radio. In addition, you will learn how to use SMS and Interactive Voice Response (IVR) systems in a broadcast environment.

It may be noted here that the rate of technological innovation and change is very rapid in the telephony sector. What we are discussing here is a general overview of the available options. This unit is for you to understand the vast possibilities and potential that telephony brings to CR; to understand the principles behind the use of telephony in radio broadcasting and use the available resources to their best potential.

7.2 Learning Outcomes

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

- describe the various tools used to incorporate telephony in radio broadcasts.
- discuss various ways of using telephones for broadcasting.

- describe the uses of computer-based telephony systems.
- analyse the pros and cons of various telephony systems.

15.3 Conventional Landline-based Systems

Before the availability of mobile phones, fixed line or landline phones were the only telephone system available to the world. In technical parlance, the landline phones are known as the PSTN (Public Switched Telephone Network) phones. Essentially, the system is composed of centralized line switching systems called exchanges, with copper wire based cables connecting the exchanges to each individual telephone subscriber's instrument. Each instrument/line is allotted a specific subscriber number, which identifies it uniquely on the network. The lines are energized by a constant electrical voltage supplied by the exchange instrumentation. Dialling a specific number activates the switching system, which then connects the two instruments, with the audio at each end travelling as electrical impulses through the copper wire system.

PSTN lines have been in use by radio stations all over the world including All India Radio and BBC World Service, with which we are quite familiar. In India, however, the use of the telephone system became very extensive with the emergence of private commercial FM stations that started taking listener calls very frequently and broadcasting them live. The commercial sector also saw the emergence of talk radio: Radio Miaow, was launched as a pre-dominantly chat radio, and most of the programmes on Radio Miaow were talk based.

So how do we actually connect in a conversation happening on a telephone landline into our audio process in a studio? Let's look at the most popular ways in which this can be done.

15.3.1 Speakerphone–Microphone Combination

One of the easiest ways to achieve PSTN-based telephony in a radio station is to a have good quality speakerphone in the studio coupled with an external microphone. In this solution, what you do is put incoming and outgoing calls on the speakerphone mode, so that you can hear the caller at the other end over the unit's in-built speaker. You can then place an external microphone to pick up this sound and send it to the mixer. This solution has been tried out in many CR stations the world over and also in India.

The biggest drawback of this solution is that, more often than not, the sound quality is extremely poor. Quite often, the voice is barely audible on the receiving end, or the ratio of the sound to ambient noise or noise on the line is hopelessly adverse. This is mainly because of an impedance mismatch. The external microphone and the speaker of the phone affect each other, as they both have their own magnetic fields.

Figure 7.1 shows a simple speakerphone unit.

Figure 7.1: A common speakerphone unit. Note the speaker grille under the handset. (The screenshot was captured by the author)

15.3.2 Transformer-based Interface

To overcome the interference issue that comes with speaker–microphone combinations, some radio stations have used pickup transformer-based solutions. This method will require you to have a few soldering skills, and the confidence to do some wiring on your own. It may not be very easy; but if it is done well, it can work very well indeed.

In this method, you need to select a speakerphone that has a line-out connector. You will need to plug the appropriate jack into this socket, expose the wires at the other end, and solder the exposed wires at the other end of the connector cable to the inputs of an old time cassette recorder playback head. (If the speakerphone unit does not have a line-out socket, you can solder the lead to the inputs leads connected to the speaker on the unit.) Now take another cassette player head, and align it in close proximity with—but not touching—the first one. The second head is then wired to a cable that is connected into the appropriate input socket on your mixing console. The two cassette player heads together form a combination known as a pickup transformer. By this method you can transfer the sound signal across without having electrical contact, and thereby avoiding the problem known as impedance mismatch.

15.3.3 Telephone Hybrids

In commercial radio stations, the telephone-based programmes are generally conducted through what is known as telephone hybrid instrument. Simply speaking, a telephone hybrid is a sophisticated telephone instrument that is capable of routing the sound signals, incoming as well as outgoing, to a mixer console. It works exactly in the same manner as a normal telephone instrument. You can insert one or more telephone lines in it, and can route the audio to the mixing console without any impedance mismatch, distortion or echo.

The basic technology involved in making a telephone hybrid is the impedance matching circuit. The impedance mismatch happens because the receiving line carries two different signals—for voice as well as the base voltage to make the telephone line operative. When we single out the voice part—since we are only interested in the voice and not the other line information—it creates an impedance mismatch, which in turn leads to distortion and echo. A good telephone hybrid should give you a voice output that is of broadcast quality.

Some of the high-end broadcast consoles have telephone hybrids built in to the mixer console itself. That means you can plug in your telephone line directly to the console, and the console will allow you to receive or make calls. However, these broadcast consoles are far more expensive than a normal sound mixing console. Most CR stations do not prefer to spend large sums of money on such consoles; and would prefer to look for other solutions for their telephony-based programmes.

Activity 7.1

Record a phone conversation with your friend on a computer using a simple speakerphone. All operating systems have a default recording application. You may use that; or if you have access to one, you may use a digital audio workstation.

Use a simple microphone to connect to the 'mic-in' port of your desktop computer. If you are using a laptop, it is possible that it has a built-in microphone that you can use.

15.4 GSM/CDMA Solutions

With the arrival of mobile phones, the communication landscape has changed dramatically. In this era of rapid spread of mobile phones, the CR stations could not afford to ignore the integration of mobile phones with the broadcasting environment. Today, a mobile phone is being used as an instrument to provide live feeds from a distant event, to engage listeners in conversations that can be heard on air, or simply to solicit comments and feedback from listeners on a real-time basis. In addition, the use of SMS as an instrument for conducting surveys has also become very popular. Here, we will examine various methods of integrating mobile phones in your broadcast environment.

15.4.1 Bluetooth Interface

Bluetooth is a short range radio transmission standard developed for use to connect computer peripherals wirelessly, and since adapted for use on mobile phones. One of the tried and tested ways to use mobile phones in your broadcast environment is to use Bluetooth-based interfaces that can help in plugging your mobile phones into your mixing console. The device works like this:

- A Bluetooth-enabled mobile phone pairs up with the Bluetooth sound interface.
- The Bluetooth sound interface has a slot to insert a microphone, and a port to take a line out connection.
- Insert a microphone in the Bluetooth interface.
- Take a line out from Bluetooth interface to your mixing console.

Now when you receive a call on your mobile phone, your voice and the caller's voice will be simultaneously transferred to your mixing console.

The biggest drawback of this solution is that Bluetooth interfaces are not readily available in the Indian market. There are a couple of companies abroad that make such devices professionally, but they are quite expensive. Moreover, the import shipping costs and the custom duties add to the price.

15.4.2 GSM Dongle (Modem)

GSM stands for Global System for Mobile communications, and is the most popular standard worldwide for mobile telephone signals. GSM modems are one of the easiest ways to incorporate telephony into your broadcast.

In this solution, you can purchase any GSM dongle marketed by the mobile phone companies: TATA Docomo, Vodafone, Idea, BSNL or Airtel. Almost all others have a branded dongle which works with their SIM (Subscriber Information Module) cards. You can buy any one of them depending on the availability of the network in your studio area. However, be careful when you choose the dongle: all dongles are not voice enabled. This means that on some of them, you can access the Internet but not use them to make or receive voice calls. When you buy one, ensure that the dongle is voice enabled. Figure 7.2 shows you a typical GSM dongle.

Figure 7.2: A typical GSM dongle unit.
HSDPA stands for High Speed Data Packet Access, a type of high speed mobile connectivity protocol. (The screenshot was captured by the author)

One disadvantage of these dongles is that many of these dongles are locked to their specific service by the mobile service providers. This means that the dongle will not work with any SIM card other than the one from the service provider you have purchased it. For instance, if you buy a dongle from Idea, it will not work with a Vodafone SIM card. You will have to use the SIM card supplied by Idea. You can bypass this by buying a dongle supplied directly by manufacturers like Huawei or Micronet. These dongles are not locked to a specific service, and you can use a SIM from the service provider of your choice in them, and even change the SIM if you are not satisfied with the service. This may also be a cheaper option in terms of cost.

Once you acquire a dongle, using it is very simple. You insert the dongle on your PC or laptop—usually into a computer's USB port—and it will ask you to install a software. Once you have this software installed, you can either make or receive a call. You can even connect to the Internet, but this is not what we will discuss here.

The voice here is routed through the sound card of your PC or laptop. That means you need to have a headphone and microphone attached to your sound card. In a broadcast environment, what you can do is that you can plug the sound card's line out to your mixing console, and connect a dynamic microphone to your sound card's microphone input. When the caller speaks, you will hear it through your mixing console. When you speak into the two microphones, one is connected directly to your computer, and the other to your mixing console. This is the way your listener, as well as your caller, will be able to listen to you on air.

Activity 7.2

Make a list of USB modems (dongles) available in the market, which support voice communication. You need to look at the technical specification of the modem to ascertain if it supports voice.

15.5 Voice Over Internet Protocol (VoIP)

Considered to be the future of telephony, Voice Over Internet Protocol (VoIP) has already found its way into the CR sector. Many radio stations have already started using VoIP-server based telephony in their programming in diverse ways. With the help of VoIP servers, CR stations are using voice, SMS and IVR to make their programmes more participatory and accessible to their listeners through their mobile phones.

There are many ways you can define and explain VoIP. For ease of understanding, let us examine the definition and explanation provided by Wikipedia, which reads as follows:

“Voice over IP (VoIP, abbreviation of Voice over Internet Protocol) is a methodology and broad range of technologies for the delivery of voice communications and multimedia sessions over Internet Protocol (IP) networks, such as the Internet. Other terms commonly associated with VoIP are IP telephony, Internet telephony, voice over broadband (VoBB), broadband telephony, IP communications, and broadband phone service.

Internet telephony refers to communications services—voice, fax, SMS, and/or voice-messaging applications—that are transported via an IP network, rather than the public Switched Network (PSTN).”

This definition makes it clear that VoIP is a mechanism that transports audio and data signals through IP networks like the Internet. It handles most of the multi-media applications, meaning you can distribute voice, text, images and video using the same protocol.

Thus, VoIP opens up a completely new arena for CR stations to reach out to their audience and engage them in programming. A VoIP server can seamlessly integrate with your broadcast environment, so that you can use phone-in and phone-out with ease.

CR stations using GRINS as a play-out software (see Section 6.4.2 in Unit 6 of this module) can make calls directly out of their radio automation system, with the help of a built-in Asterisk server, which uses VoIP protocols. The GRINS usercan also send and receive SMS from the same interface.

You will now learn how you can use VoIP in a real-time broadcast environment. First of all, you will have to configure a VoIP server on your setup. You have a host of proprietary options and quite a few open source options for this: Elastix or FreeSWITCH, for instance. However, the most widely used VoIP server is called Asterisk. You need to know a little about computer programming before you can install or configure Asterisk.

After installing Asterisk, you need to provide the server a GSM or PSTN gateway to start making calls to the outside world. Many hardware options are available in the market today to configure a gateway. Companies like Digium and Cisco are well-known for their gateway products for professional uses.

If this seems difficult, you can use one of the many online VoIP services instead. Skype or Gtalk (now part of Google Hangouts) are also available in India now. You will have to buy credit on these systems, that is, prepay some money so that you can call PSTN networks and mobile numbers from the system to make phone calls. You can also get a permanent Skype-linked telephone number on which your callers can call you.

However, remember that currently VoIP services are more expensive than normal telephony in India. Experts in the field believe that the situation is bound to change, and that VoIP will become cheaper in the future. Many corporate houses who use phones on a large scale are switching over to VoIP services to cut down their telephone bills. Similarly, most call centres have been using VoIP for their operations.

With a VoIP server installed, you can even use Interactive Voice Response (IVR) systems to engage with your listeners, as well as make your programming available ‘on demand’. This means you can have an IVR that prompts any caller to press a number to listen to a pre-selected programme. As you will learn in Unit 22, some of the IVR-based initiatives like Jharkhand Mobile Vaani and CGNet Swara utilize IVR functionality for community-driven media initiative.

Activity 7.3

- Make a phone call using an online telephony service.
- Prepare a list of possible uses of an Asterisk server at home, office and a CRS.

15.6 Use of SMS

Most—if not all—of us are familiar with the term SMS, which stands for Short Messaging Service. It is one of the most widely used communication tools across the globe. Today, all mobile phones support the SMS service, which is provided by all mobile service providers. SMS is derived from the older system of radio telegraphy. In the early years of mobile telephony, short messages used to be sent through devices called pagers, a service which eventually got incorporated into the Global System for Mobile Communication (GSM) standard. Later, SMS services were extended to CDMA (Code Division Multiple Access) networks, as well as some landline networks. As per the global standards, SMS messages are restricted to 160 characters per message.

15.6.1 User Case Scenarios for SMS

In a CR setup, you will see very creative use of SMS. They are generally used for the following activities in a typical CR setup.

- Listeners' feedback: It is important for any CR to know the minds of its listeners. One of the ways to get listener feedback is to encourage them to send SMS to the radio station, which the station can compile and incorporate into a feedback mechanism.
- Live participation: SMS is also a useful tool to encourage listener participation in programmes. One of the important aspects of any CR station is the participatory nature of the medium. In a live programme, a radio jockey or announcer can encourage listeners to participate in the programme with their text messages, which can be read out during live broadcast. You can even do request shows using SMS-based requests.
- Surveys and polling: Many radio stations use SMS as a tool for surveys or to understand public opinion on a variety of subjects. For example, in a particular village, if a road has been newly paved, the CR announcer can ask the members of the community if they are satisfied with the quality of the road by sending an SMS with "YES" or "NO" to the station's number. This feedback could be compiled to present an opinion.
- Competitions: Many CR stations have used SMS as a voting tool for community-based competitions. For example, a CR station called Radio Bundelkhand in Central India ran a competition to showcase singing talent from the local communities. New singers were given a chance to sing on radio, and the community members were asked to vote for the best singer through SMS.

15.6.2 Technological Options

In the previous section, you were introduced to a few examples of the creative usage of SMS. You can figure out many more such uses in collaboration with the programming team. Once you understand the usage, it may not be very difficult to select the appropriate technology for the purpose. Let us understand some of the technological options available:

- The simplest method is to buy an average mobile phone with a SIM card of your choice to send and receive SMSs for your radio station. However, it may be noted that it may not be very comfortable or fast to use a normal mobile phone for sending bulk SMSs for your radio station, or even receive large volumes of SMSs in response. However, as a trial or beginning, it may not be a bad idea to use a phone.
- You can even use GSM modems, discussed in Section 7.4.2. These modems are generally provided with a software to send SMS. If your modem has been manufactured by a company like Huawei, it will have bundled software like Mobile Partner or Mobile Connect. Both these types of software can send and receive SMS.

- There are many online SMS services available. Some are free and some are paid. You may check out the following web links for examples and possible services:

    http://www.160by2.com

    http://www.ofsms.in

    http://www.mysms.com

- Most of these services let you send free SMS. Given the fact that they are free of cost, they have certain limitations. Some of these free services have a daily cap on the number of SMSs you can send, while some will let you send only 140 characters instead of the stipulated 160. (The service provider may use 20 characters to piggy-back advertisements along with your messages!)
- There are also specialized software for bulk SMSs, which are generally used by advertisers. These software have multiple abilities to send or receive SMS based on region, time or context. However, most of them are not free, and may cost a lot of money. You may select one of these software if your radio station can afford the cost.
- An excellent option is the free and open source software Frontline SMS, which is being extensively used the world over by many radio stations. This software is browser-based, and works with a large variety of mobile phones and GSM modems. (Go to www.frontlinesms.com to see a complete list of supported phones and GSM modems. It may be a good idea to consult the database before buying a phone or a modem for the purpose or installing the software on your computer.) The software can handle bulk SMSs with ease while keeping a meticulous database of all the SMSs you send or receive, also allowing categorization and time-based search. It also maintains a list of your contacts. Figure 7.3 shows you the main window of the Frontline SMS software.

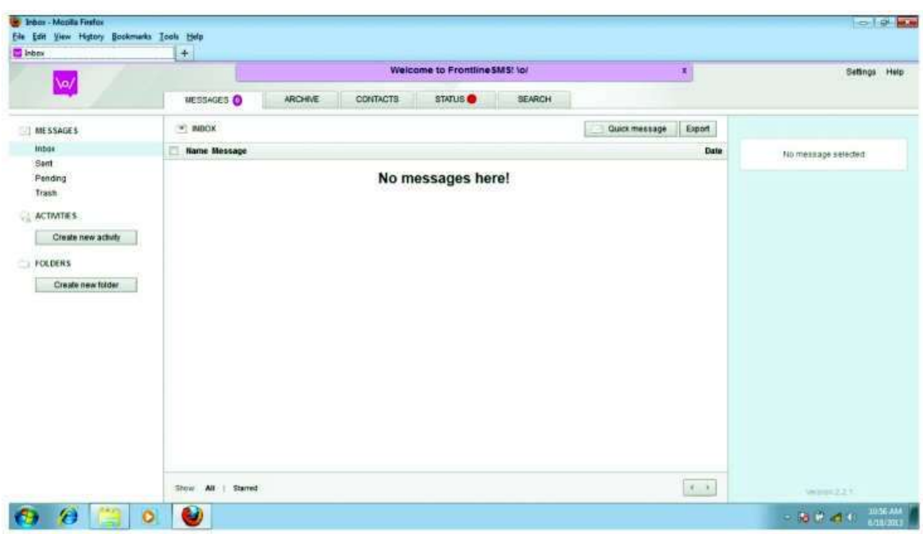

Figure 7.3: The primary interface for Frontline SMS(The screenshot was captured by the author)

Caution

Bulk SMS, which is interpreted as a form of telemarketing, is regulated under the Telecom Commercial Communications Customer Preference Regulation, 2010. Under this regulation, any activity that can be termed as telemarketing can be carried out only after due registration with the authority. The authority also maintains a registry of individual customers, with their preferences regarding the type of unsolicited messages they want to receive or whether they want to receive any messages at all.An unsolicited SMS sent to a number which is registered under the Do Not Call Registry can result in a complaint against the sender of the SMS, who can then be liable for penal action. If your radio station wants to use SMS in a major way, then it may be necessary to register as a telemarketing entity. If you just want to send a few SMSs per day, you can verify if the number you want to send the message to is registered under the No Call registry by simply going to http://www.ndnc.in.

Figure 7.4 shows you the contacts window, through which you can look up the details of the contacts stored within FrontlineSMS.

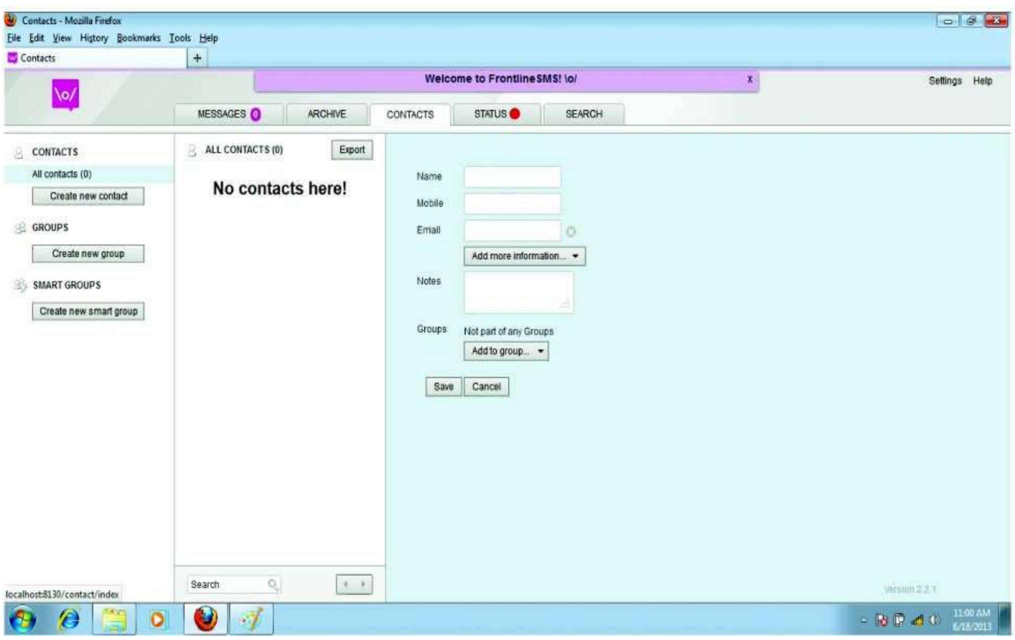

Figure 7.4: The contacts window of Frontline SMS(The screenshot was captured by the author)

Activity 7.4

- Use your mobile phone to send a survey SMS to your friends and colleagues. You should select a question that can be answered in a simple "Yes" or "No", and the question should be one to which all of them can respond. It could be a simple question like "Are you happy with your current job?" You should compile the answers in a spreadsheet, and come up with a conclusion. For example, if 3 out of 10 friends say "no", you could present this as "30 per cent of the respondents said they were not happy with their jobs".
- Use three online SMS services and do a comparative analysis of their pros and cons.

15.7 Let Us Sum Up

In this unit, you learnt about the various options available to a CR station in terms of incorporating telephony in their broadcast environment. You already know that use of telephones is absolutely necessary in a broadcast environment to ensure the participation of your audience, and in making your station more community-driven.

There are simple solutions like using a speakerphone with an external microphone. However, these solutions offer limited functionality and restrict you to just voice communication. Also the resultant quality of voice is not good enough when you use such solutions. The conventional telephone hybrid also falls in the same category. Though it is easy to use and provides better sound quality, one is still dependent on otherwise undependable PSTN lines, many of which are poorly maintained. Additionally, landline phones are not easily available in the hinterlands.

On the other hand, digital telecommunications technologies have not only made it possible to improve audio quality but also placed a plethora of possibilities at your disposal. If you use GSM modems, you can utilize SMS functionality. The availability of GSM networks is also superior to that of landline phones.

However, if you really want to optimise the telephony in CRS, you have to wholeheartedly embrace several of the newer technologies available for this purpose. A telephony server operating in the CRS can change the way members of the community participate in the programming and listen to its programmes. Though telephony servers are not very easy to configure, there are many ready-made solutions available that a CRS can put to use.

15.8 Model Answers to Activities

Activity 7.1

If you used the external mike to record from the speakerphone, you must have noticed as you bring the microphone close to the speakerphone, a humming or a whistling sound ensues. This is due to microphone feedback caused by electromagnetic interference. You will have to put it at a distance where the disturbance is the minimum. This is one of the drawbacks of this method.

Activity 7.2

While making the list of GSM modems, you would have come across some which are marketed by mobile phone companies like Vodafone, Airtel or Idea. These modems, though marketed by individual mobile companies, are usually manufactured by only one or two companies globally. Most of the GSM modems available in India are manufactured by a Chinese company called Huawei. So, it may be easier to look at the product list of Huawei itself, which is available on the company's website.

Activity 7.3

If you are planning to procure a GSM modem, it could be cheaper to look at some of the reselling portals like quickr.com or olx.com. You may find them cheaper there. As mentioned earlier, it is necessary to check if the modem you are buying supports voice communication.

Activity 7.4

While looking for online telephony services, you must have come across services like Skype, GTalk, Akiga, and so on. You must have also noticed that the sound quality of online phones are far superior than your normal landline or even mobile phone, provided your Internet connectivity is good enough. The bandwidth and speed of your Internet connectivity is the key to a good telephony experience.

While researching on the subject of Asterisk applications, you must have realized that there are many uses for Asterisk in a home or office setup. To understand the functionality of Asterisk in particular, and VoIP in general, please have a look at http://www.voip-info.org/.

15.9 Additional Readings

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Glossary

Unit 6: Audio Hardware and Field Recording

Microphone	It is basically a transducer that converts sound waves into electrical waves.
Audio Recorder	It is a device that, with a microphone connected to it, records sound as audio files.
Headphones	These are instruments that help you monitor audio that you are recording or editing.
Audio Mixer	It is a device that helps channel different audio inputs, and helps you balance, sweeten and output to a recording device
Impedance	As the word itself suggests, it means resistance offered to the flow of sound.
Frequency Response	This is a microphone's capability to receive high and low sounds.
Omni-directional Microphone	It is a microphone that can pick up sound from all directions.
Uni-directional Microphone	It is a microphone that can pick up sound only from one direction.
Bi-directional Microphone	It is a microphone that picks up sounds from the front or the rear but does not respond to sounds from the sides.
Solid State Cards	These are chips that store information in EEPROM flash memory embedded inside them.
RCA	Adapted from the name Radio Corporation of America, it is a plug and a jack designed for use with a cable for both very low and very high frequencies.
XLR	Adapted from the Canon plug X, Left and Right, it is the universally used connector for a balanced audio output.
Phono Jack	It is a type of TRS jack. There are two types, namely the mono jack and the stereo jack. Just 'phono' refers to the ¼" size. Mini-phono generally refers to the 1/8" jack.
Sound Card	It is a peripheral that is connected to a computer to edit and playback audio.

DAW	It stands for Digital Audio Workstation. A computer one can use to record audio in a studio or even use for post-production of radio programmes.

Unit 7: Audio Hardware and Field Recording

FOSS	It stands for Free and Open Source Software.
OS	It stands for operating system, which starts up the computer and provides basic functionality.
GNU	It is a UNIX like computer operating system developed by GNU Project.
GPL	It stands for General Public License, one of the most popular software license used by the open source community. There are currently three versions of the license available.
Linux	It is a free and open source computer operating system used extensively in the server segment.
UNIX	It is a free and open source computer operating system based on which many operating systems have been built, including Linux and Apple's OSX.
Proprietary	It is a type of software license which is not free and of which users are liable to pay the license holder for its usage and distribution.
Distro	It is an abbreviation for the word distribution. Linux has many flavours, like Ubuntu, Fedora, Debian, and Mint. All these flavours are known as distros.

Unit 16: Telephony for CR

PSTN	It is the abbreviation for Public Switched Telephone Network—a network that is generally used in landline phones.
GSM	It is the abbreviation for Global System for Mobile Communication.
CDMA	It is the abbreviation for Code Division Multiple Access. The other format is TDMA, which means Time Division Multiple Access.

Telephone Hybrid	It is a device that can route voice from the PSTN line or a landline phone to any sound device like mixing console or a speaker system.
GSM Dongle	It is a USB device that can modulate and de-modulate GSM data.
VoIP	It is the abbreviation for Voice Over Internet Protocol. To make it simple it is name for Internet based telephone system.
IVR	It stands for Interactive Voice Response systems, used generally by the service industry for receiving customer calls.
Asterisk	One of the most popular server software for telephony, it can also handle EPBX functionalities apart from IVR.

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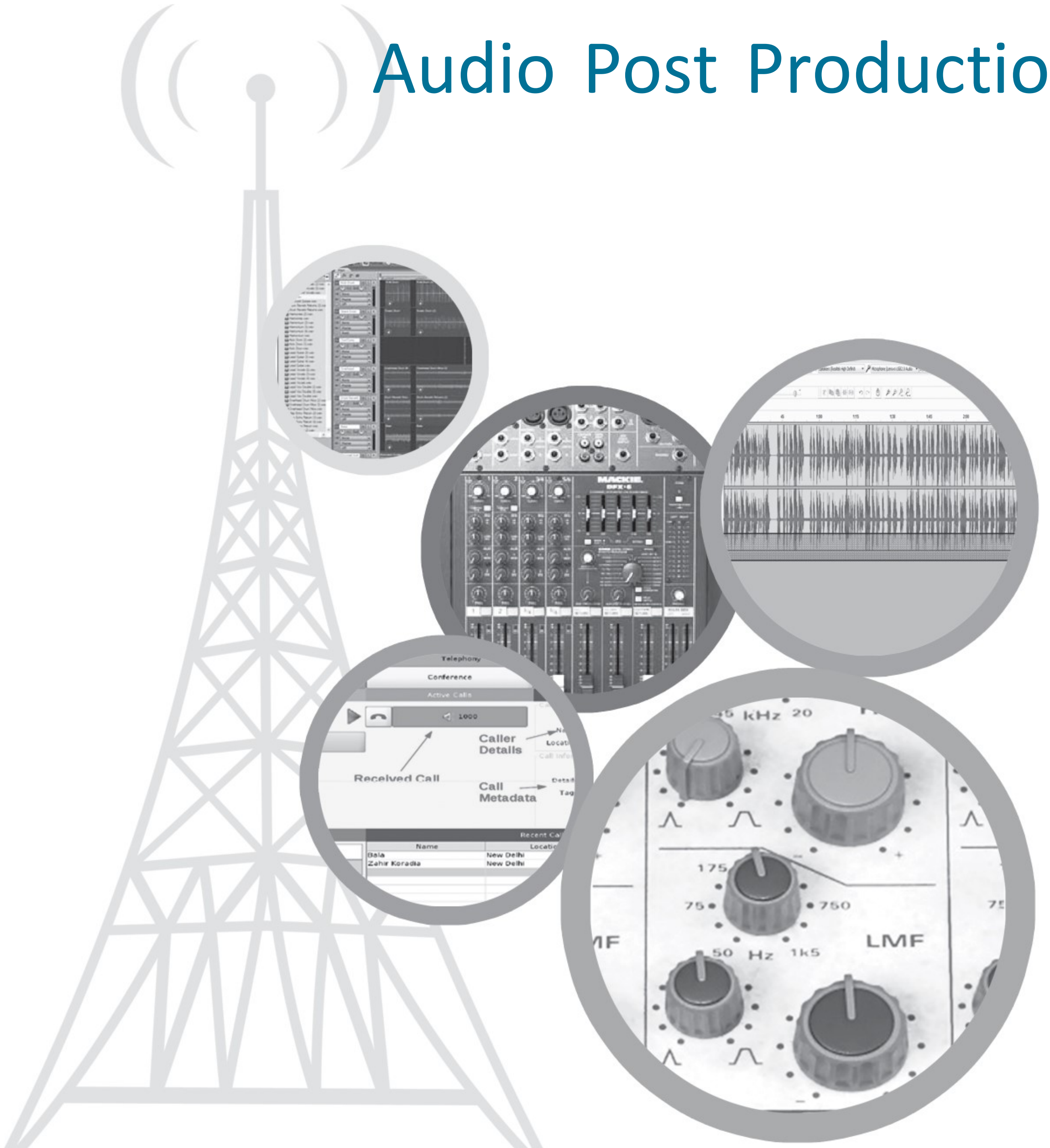
Module: 3
Audio Post Production



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About the Module

Module Description

A radio station is known and identified by its sound design. From signature tune to jingles and from live announcement to recorded programmes, the production of sound is one of the most important aspects of a radio station.

In a typical radio station a sound recordist is as critical and important as a musician, singer or a speaker, even though the listeners hardly know him. The sound recordists have been the unsung heroes of radio stations. But, in the recent years, especially in the community radio sector globally, there is a trend to identify sound recordist and recognise the work put in by them.

From the rapid transition of analog to digital sound technology, the job of a sound recordist has undergone a formidable change in the recent years. Today, sound recordists are accomplishing task in much shorter time as compared with their counterparts in the analogue era. At the same time, the expectations from a sound technician has increased multifold with the ever expanding horizon of sound technology. Currently, the sound technologist need to be on their toes to keep up with rapidly changing concepts and tools.

In a community radio station, the job of a sound technician is an ideal mix of conceptual clarity, practical knowledge of tools and creativity. You will discover this aspect as you go through this module in theory and practice.

Module Objectives

This module is designed to equip a community radio technician with basic sound production concepts, tools and skills. After learning this module a learner should be able to:

- Carry out studio and field recording
- Edit a sound track
- Mixing sound in multiple tracks
- Mastering sound after noise removal, frequency adjustments in a desired audio format
- Convert sound from one format to the other
- Store sound files in different mediums and retrieve them as per requirement

UNIT 8

Sound Recording and Editing

Structure

- Introduction
- Learning Outcomes
- What is Sound Recording?
    - The concept of recording audio
    - The concept of audio levels
    - The recording process
- Single and Multi-track Recording
    - Single track recording and multi-track recording
    - Recording when listening to a reference track
- What is Audio Editing?
    - The concept of editing audio
    - The process of editing audio
    - Destructive and non-destructive editing
- Equalisation and Digital Audio Processing Equalisation (effects etc)
    - Equalization
    - Digital audio processing (DAP) and effects
- Understanding the DAW and editing software
- Let Us Sum Up
- Model Answers to Activities

8.1 Introduction

In previous Units, you have understood the basics of sound, as well as an overview of the field equipment you use for audio recording in the field, as well as in the studio. You have also learnt about file formats, microphones and related equipment, and the concepts of digital and analog audio.

In this Unit, you will build on this understanding by taking the first few steps towards recording audio in the field and the studio, and towards editing and adding effects to the edited content.

An important concern: there are thousands of different kinds of field recorder units and models available, and more keep appearing every single day. Similarly, computer operating system versions keep changing, and the software we use for recording and editing audio on a computer often changes. To give a very specific examples; this unit will be rapidly obsolete, it would become outdated almost as soon as it is published. Therefore, we are going to talk about the key controls and functions that are shared by most field recorders and computer recording and editing systems (often called Digital Audio Workstations, or DAWs). You will need to use this Unit along with user manuals and resources specific to the device or software that you are using in order to understand the specifics of that instrument or software.

8.2 Learning Outcomes

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

- present the recording process in a block diagram
- describe graphical output of recording (in editing software)
- analyse the concepts of single and multi-track recording
- describe the process of audio level adjustments and balancing
- discuss the concept of audio editing and the tools we use to achieve this
- explain the meaning of digital signal processing and related effects as applied to audio

16.3 What is Sound Recording?

When we say we are 'recording audio', we are essentially referring to the act of capturing sound and then storing it on a medium, from which we can recover and play back the original sound at the time of selecting.

In this section, we are going to take a closer look at two types of sound recording: field recording and studio recording. Many of the concepts and basic ideas underlying both types of recording are the same, but as different devices are used, they become slightly different, and the way we adjust some of the settings are likely to be different.

Additionally, in this day and age, virtually all recording is digital. Analog recording has almost vanished. Our discussion, therefore, will proceed on the assumption that our audio is stored and utilized on digital devices and on media that support this.

16.3.1 The concept of recording audio

By the term 'recording audio', essentially we are talking about four sub-processes that together result in a successful and accurate capture of a sound that we can hear. These four processes are:

1. Using a transducer to convert the audio energy generated by a sound into an electrical signal: A transducer is a device that converts one form of energy into another. In our case, the transducer of choice is the microphone, which converts audio into electrical form, an audio signal essentially, the electrical form of the acoustic energy, which corresponds exactly to the frequency and amplitude of the original sound(s).
2. Transferring the audio signal to a recording device: The audio signal is usually routed through cables to a device that houses the recording medium. In this field, the device is the recorder unit itself, which usually has a slot or housing for the recording medium like a hard disk, an SD card, or a similar item, and the cable is the wire that connects the microphone to the recorder unit. Similarly, in the studio, there is a cable connecting the microphone to the computer in addition, in some cases of a mixer unit, multiple mics or source of units are connected.
3. Conversion of the audio signal to a digital stream: As we have previously seen in the Unit on analog and digital audio, modern audio signal processing supposes the conversion of the analog audio signal generated by the microphone to a digital form for storage. This process is called A/D conversion (analaog-digital conversion), and is carried out in a field recorder within the recorder itself. In a DAW, this may be carried out within the sound card installed within the system or in an external breakout unit that comes as part of the sound card, or even in the mixer unit itself, if the mixer is digital.
4. Storage of the digitized audio: This is the final stage of the recording process, and involves storing the binary data generated by the A/D convertor on the storage medium. In a field recorder, this is likely to be flash (or EEPROM) storage of some kind either as built in solid-state memory, or an SD (Secure Digital) card, or a CF (Compact Flash) card. In a

computer, this is likely to be a magnetic storage disk also known as a Hard Disk Drive (HDD) or a Solid State Drive (SSD), which is also a form of flash memory.

Dear learner, by this time you have got some ideas on sound recording and its four processes. Here you can watch a video, which will help you to comprehend the concept in a better way. The video is on 'Sound Recording and Editing'. This video will also help you to explain the concept of audio editing, which is incorporated in this Unit at section 8.5. To go through the video, you may visit http://tinyurl.com/nhpypz7.

16.3.2 The concept of audio levels

Did you ever experience loud sound, that could harm your ears? Loud sound can be very painful and can actually cause physical damage to your ears, including permanent deafness, if you are exposed to it for too long.

Why does this happen? At its simplest level, this is because your ears can take a certain loudness of sound and no more. Beyond this, it overloads the sensory apparatus of your ear and instead of a signal that goes to your brain, it becomes pain.

At the other extreme, if the sound you are hearing is very low, you will not be able to make out the sound at all. Even if you strain your ears, you will get nothing useful that you can understand.

In exactly the same way, every audio apparatus that we use has an upper limit to the sheer volume of sound that it can be exposed to. If sound is beyond this limit, the apparatus will not be able to perform well in its function of transduction or signal processing, resulting in a recorded audio, which wll not be a trustable reproduction of the original sound. And, just as the extremely low sound cannot be heard well by your ears, an extremely low sound will result in a very poor signal that cannot be recorded accurately.

This means, that for every audio apparatus that we use, there is a particular level to which we have to keep adjusting the audio input to, so that we may get the best possible recording. This is achieved by creating a measure of the level of input, which we can see as we record the audio. On most devices, the physical representation of this measure is in the form of a Volume Unit (VU) Meter. On digital devices, this is usually presented as a pair of moving bars against a calibrated scale with the movement of the bars linked to the incoming audio level. The level of the audio is either controlled automatically by the device's circuitry, or there is a provision of a pair of buttons that let you raise or reduce the input level in order to keep the levels at an appropriate setting.

Audio levels are measured in DeciBels (or dB for short), a relative measure based on a logarithmic scale. As a rough measure, each change of Six dB represents a doubling or halving of the input sound; a sound registering at 6 dB is twice as loud as a sound registering at 0 dB, and half as much as one at 12 dB. As a convention, VU meters show 0 dB at one end, usually the right in horizontal meters and the top in vertical meters, and negative dB values before that. The understanding is that 0 dB represents the maximum audio level that the instrument can accept without distorting it , that is, recording or processing it inaccurately, and the levels below that represent a percentage of this maximum value. You will find VU meters on any good field recorder, mixer unit or as part of any audio software that lets you record audio.

As a general rule, on digital VU meters, it is a good practice to keep the levels between -12 dB and -6 dB, with the odd loud sound going up to -3 dB. Never let the level reach 0 dB.

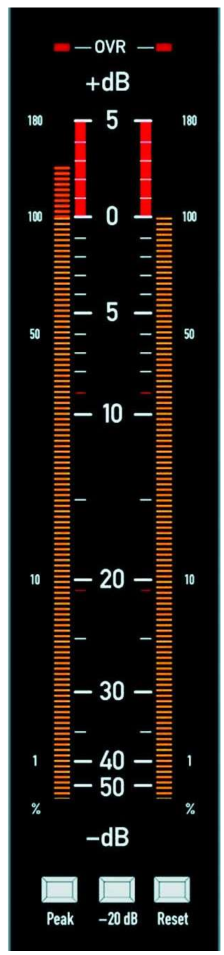


Figure 8.1: A typical digital VU meter. Aim to keep the audio between the -10 dB and the -5 dB mark, with loudest noises not reaching the 0 dB mark. VU meter image; Source: http://en.wikipedia.org/wiki/File:PPM_IEC_268-10_I_DIN.jpg; CC BY-SA

16.3.3 The recording process

We can now look at the series of steps that we follow to make a recording.

Step 1: Connect up the recording equipment

Microphone needs to be connected to the appropriate sockets on the recording equipment, using the appropriate and compatible connectors. Care must be taken to ensure that the output from the microphone is physically compatible with the recording equipment. Some recorders can only accept signals from powered, usually condenser microphones. Others will accept only unpowered, usually dynamic microphones. If there are control surfaces like mixers or sound card breakout boxes in between, they have to be connected and switched on. The recording medium must be in place with adequate storage space to accept the estimated duration of incoming audio. Power sources must be checked to ensure a continuous and assured supply during the recording. In the case of a field recorder, this means inserting and checking battery cells, in the case of a DAW and related equipment, this means power supplies, adaptors and relevant battery cells, as the case may be. An audio monitoring device, preferably headphones should be attached to the recorder units, for the recordist to monitor the audio while recording. Some devices may also have various sensitivity settings for connected microphones. This will need to be set as per the specific microphone connected and should be pre-tested.

Step 2: Position the microphone in front of the audio source or speaker

The audio source may be an instrument or a person. In a previous Unit, you have already seen the principles of good microphone position. The primary concern is to place it, where extraneous noises will not hit it and where streams of wind caused by the instrument or the person's breath will not hit it. If there are multiple sound sources and multiple microphones, it becomes doubly important to position the microophones in a way that they capture only the sound they are dedicated to, so that we may have an easier time of adjusting the relative levels of each of the different sources.

Step 3: Check the audio levels

Most audio equipment, whether in field or studio will have a preview mode, where you can make the device recording ready, but do not actually start recording. In many field recorders, this is achieved by pressing the record button once with a second press activating the recording process. The record button is usually the one marked with a round red circle. In software records on DAWs, there may be a software button marked with the same symbol or a menu option that lets you put the system in a preparatory mode. The preparatory mode lets you view the current levels of audio input on the VU meters. It is good practice to make each speaker or instrumentalist speak or play in turn, so that we can ensure that all of them register at roughly the correct place on the meter for good quality recording. If the level is not correct, this may necessitate changing the position of the microphone or adjusting a control on the mixer or software interface to bring the audio to the correct level. Sometimes, it may also mean asking a speaker to speak more loudly or softly.

Step 4: Commence recording

As noted, this may involve pressing a given physical or software button once or a given number of times. Most recording units will also have a visual indication that recording has started. This may be a black or red colour dot or circle that appears on the screen display or the commencement of a time counter (usually in the hh:mm:ss format); or both.

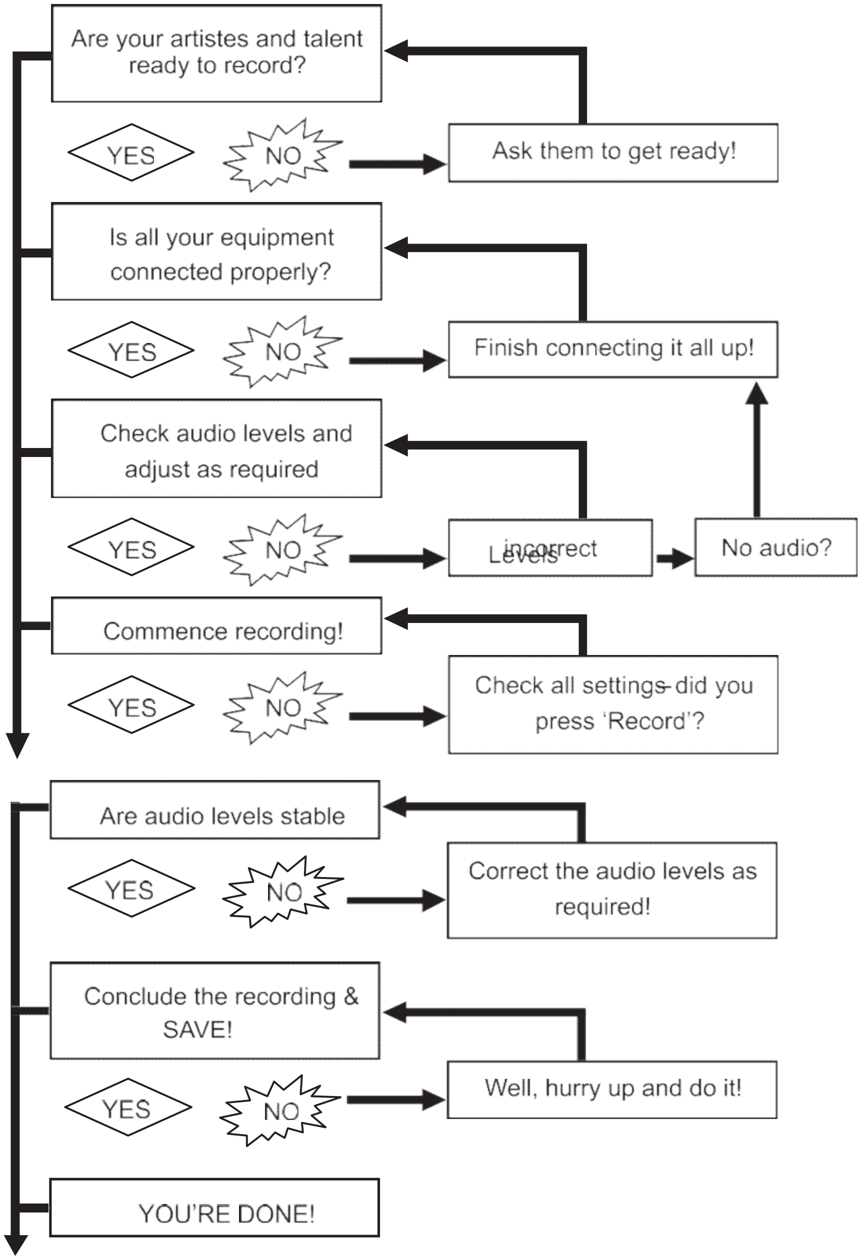
Step 4: Monitoring and adjustments during recording

An experienced recordist will listen over headphones during the recording and keep a hand on the level controls in order to adjust for sudden changes in loudness caused by someone leaning closer to a microphone, or raised voices impulsively. Where there is a mixer, this adjustment may be made using the faders on the mixer. An experienced recordist will be able to adjust on the basis of what he or she is hearing on their headphones, which will be set in advance to a constant level the recordist is used to. Many recordists will then not feel the need to look at the level meters at all. In some cases, where pop or blow has disturbed the audio, or if there is a lack of clarity in enunciation, or an external noise has overshadowed the audio, vigilant monitoring will let you identify whether the recording needs to be interrupted, or the last sentence needs to be spoken again.

Step 5: Conclude the recording and ensure that the recording has been saved

In most cases, stopping the recording involves pressing the STOP button (a button marked with a black square) or the RECORD button once again. Usually, when recordings are concluded, it will take the device a couple of seconds to preserve or 'save' the data onto the recording medium. In computerized DAWs, this may not occur till you physically ask the computer to 'save' the recording: this may be selected from the menu options available or done by a combination of key presses. In most Windows based computers, the key press will be (Control + S), that is, the S letter key is to be depressed with the CTRL key already depressed. Computers may also ask you to give a name to the file and choose where to store it on the computer. In most portable audio recorders, the name will be assigned by default as per the recorder's own naming system. It is always worth playing back the recording briefly to ensure that all is well.

Graphically, this may be represented as a flow chart like this:



Activity 8.1

If you have access to a professional or semi-professional field recording unit, do this activity with that unit. Alternatively, you may conduct this with a recording application on your mobile phone or with a microphone connected to your desktop computer or laptop, using any recording utility on the system.

(For example, Windows Recorder on Windows XP OS.) Decide a simple theme for a short three minute interview. Good examples would be: My dream job or An issue I feel strongly about. Ask a friend to help. Discuss the theme with him or her and develop a set of questions that you can ask. Then, follow the process outlined in the flowchart earlier in this section, and conduct and record a short interview on the selected subject.

16.4 Single and Multi-track Recording

16.4.1 Single track recording and multi-track recording

Let us first understand the difference between a track and a channel. In a previous unit where you have learnt about stereo sound, you must have understood the concept of separate channels for left and right audio streams. A channel in the sense we mean it here, not to be confused with TV channels or the kind that need to be tuned on a radio, is a discrete audio path that denotes audio to be played back over left, right, or both monitors equally or in a specific proportions.

A track on the other hand is the composite net result of a recording from a specific source or sources. In a field recorder, as per our previous examples, we may have many sources but we combine them all by positioning them in front of our microphone or by using a field mixer into a single track.

It must be obvious that when we record on most field recorders, we can only record and hear one track at a time. If we have multiple sources together, we have to put them all on the same track and find a way to position them in front of a single microphone, so that everything sounds reasonably clear, or use a field mixer to mix multiple microphones and give a single output that goes onto that one track. It is very unlikely that you will be able to use the second option.

Figure 8.2: Multiple sources recorded on a single microphone, resulting in a mono track. The squiggles are the graphic representation of the audio also known as the waveform display of the audio

The important thing to remember is that a track may be in mono or in stereo. As a matter of practice, most recordings are made as mono tracks, whether it is in the studio or on the field. It is only after editing and finalization that stereo enters the picture. When we are finished with arranging our various audio sections, we decide what should be heard over the left and right monitors and separate these sounds into two discrete channels, left and right, in the final finished track.

There are, of course, some recorders with in-built microphones which are capable of recording audio in stereo right away. The microphone to the left records the sounds from the left, and the microphone to the right records sounds from the right. But even with such recorders, it is good practice to do mono-recording, if possible; there will be settings that allow you to do so.

In a studio, however, we can have a little more flexibility, since things are more in our control. We can position singers and instrumentalists as we like and we can even build little enclosures to separate them from one another. This means, we can use multiple microphones in the same space and position them so that they pick up different sounds. For example, Microphone 1 for the vocalist and mic 2 for the tabla player accompanying him or her. It is obvious that recording these sounds separately, is logical to next step, so that we can adjust and arrange each sound separately. This is why most audio editing or recording softwares found on DAWs will allow you to provide several discrete sound inputs to the system, each of which can be recorded separately on a different track of the recording software. This is called multi-track recording.

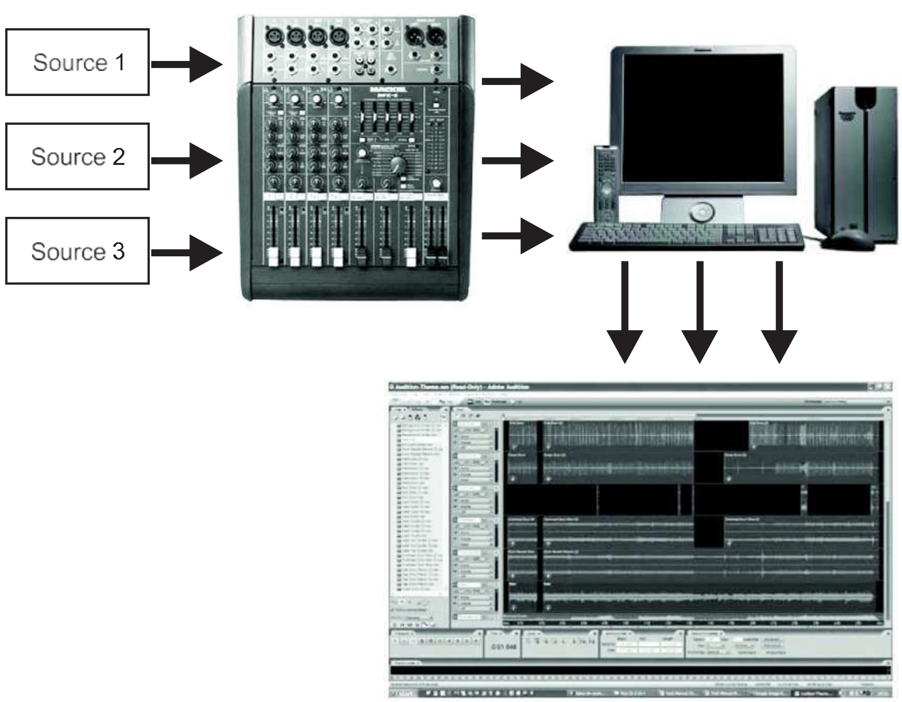


Figure 8.3: Multiple sources recorded through a mixer to a DAW, resultingin multiple mono tracks. This is called multi-track recording.

16.4.2 Recording when listening to a reference track

One of the most important aspects of being able to record on multiple tracks, simultaneously in the studio is to be able to listen to the previously recorded track(s), while recording (or 'laying down') a new track.

Consider a situation where the members of a musical troupe cannot come together at a common time to the studio to record or where the studio is too small to accommodate the whole group together. How do we record their songs?

The solution is provided by multi-track recording, where we can record the musicians one by one with each listening to the previous tracks while recording their own. Naturally, this calls for a considerable amount of practice within the group, so that everyone can play or sing their parts perfectly. The previous tracks are played back to the recording artiste over a pair of headphones as part of what is known as a studio monitor or foldback system. ('Foldback', since the system returns audio to the originating space, or 'folds' it back.)

The usual practice in these cases is to first decide a basic beat or tempo for the song, which is then recorded as a beat on the first track or generated by the recording system as a 'click track' – a track with tones generated at regular intervals, like a ticking clock. The percussion instrument players then come in and record their instruments by listening to the click track; the instrumentalists come in one by one to add their tracks on top and the vocalists come in last, so that they have the benefit of listening to all the background music at once as they sing. There are many, who prefer to record in a different order and that's perfectly okay.

This has obvious benefits in a non-music situation as well. Think of a phone-in programme, where people are calling in on the studio phone line while the guest and anchor are in the studio. Since, the audio is coming from different source devices, they are likely to require different adjustments in order for them to match each other in terms of quality and audio level. This can be achieved by using a mixer unit, where both feeds can be on separate feeder channels. It can also be used by recording them in to two separate tracks on the recording software with the adjustments pre-applied to the respective tracks by the software.

Activity 8.2

Assume you are a recordist in a music studio. You have to set up a recording of an acoustic popular music group that includes the following four individuals, who will come in at different times to record their sections:

1. The vocalist
2. The lead guitarist
3. The drummer
4. The flautist

Can you work out the order in which you will record the four in a multi-track setup?

16.5 What is Audio Editing?

Now, that we have a basic understanding of the process of recording audio, it is time to look at the post-production process called editing.

16.5.1 The concept of editing audio

At its most basic, editing is a process of cutting out what we do not require and joining together the sections that we do in order to create a whole that makes sense and is worth listening to.

It should be understood by now that making a radio programme involves a process of recording several different components like interviews, sound effects, music, narration and so on, at several different points in time. Naturally, at some point we will have to decide what to keep, and in which order. The process of refining and ordering our audios is what we refer to as audio editing.

In some ways, editing begins at the time we write our scripts, since we are already beginning to decide what to keep and in what order we are going to present our information or ideas. Some others, however, feel that it actually begins at the point where we have completed most of our recordings and are beginning to log and review our recording to decide what the important bits are. Both are correct, in many ways,. The specific process that we will discuss in this section is the process we undertake once we have our recorded material in hand, and are physically in a position to sit before our Digital Audio Workstation. But, before we go further, let’s understand the four key steps that audio editing is made of.

16.5.2 The process of audio editing

Step 1: Transfer, import and categorization of audio

Above all, audio editing and editing in general is a matter of organization. So the very first process we undertake is the process of bringing all the recorded materials together and organize them. We are, of course, assuming that you have been able to record what you want, keep it safe till now; and keep notes about your recording in a way that you can locate what you want.

When it comes to field recordings, it is selecting the recordings or tracks that we want, and transferring them to the DAW system. It is good practice to keep all the recordings for a given programme in one folder on the computer we are using for the purpose and keep field recordings in one sub-folder, studio recording in another, and music and sound effects in a third. Then we will transfer our field recordings now to the appropriate sub-folder within the one for the programme that we are about to create.

The studio recordings, similarly, will need to be copied from wherever on the system they are or from a different system, if another system has been used for that recording, into the appropriate sub-folder. Music and sound effects, if we have gathered some already, go into the third folder. Needless to say, we may continue adding items to all three folders as we progress.

Once we have all these items in their respective folders, we will need to open a new file in the audio editing software, give it an identifiable name, and save it within the programme folder that we have previously created. This editing file is often termed the session file. We will then tell the editing software where all the relevant clips that we would like to use are – a process usually referred to as importing the audio. In most softwares, this is achieved by the command: File>Import on the main menu. In some cases, this may be done by simply selecting the relevant clips, and click-dragging them to the clips window of the software. (Click-dragging is the process of pressing the left button of the mouse over the clips while all the clips are selected and simply moving the mouse over to the target window we would like to transfer them to, then letting go of the mouse button.)

If you have done this correctly, you should now see the name of your programme file somewhere at the top of the main editing window; and all your audio clips in a window somewhere in the main interface of the editing software.

Step 2: Placing the clips on the timeline

Most audio editing softwares have a part of the interface known as a timeline. The timeline is the space where you may arrange the clips in their chronological order, that is, in you will listen to them in order and layer them in tracks, so that it becomes easier for you to decide which audio overlaps the other portion of the audio. The usual convention is for tracks to be arranged vertically, so that you have a series of tracks from top to bottom and for clips to be arranged horizontally within each track in the order of play. The clips on a given track will usually be arranged one behind the other, edge butted up against edge, so that there are no gaps between them rather they are like the carriages in a train. This process is called assembly.

A vertical moving marker indicates the portion of the timeline that is currently being played out for you to hear. It is often called the play head, after the old convention of the magnetic head over which tapes once used to pass. (It is commonly called as the cursor). Note that all clips across the various tracks that are currently under the play head will be heard simultaneously. The play head usually moves from left to right, with a scale at the bottom or top of the timeline indicating the present position of the playback.

The other convention that you need to know is the distribution of your audio across tracks. Typically, Tracks 1 and 2 will be used for voice, Tracks 3 and 4 for music and Tracks 5 and 6 for sound effects and ambient sound. This arrangement lets you to keep all your material organized, besides letting you make adjustments between clips of a similar nature. You may change this as you choose: what is important is that you have a standard procedure. If your edit is simple, you may like to reduce this to four tracks, with the first for voice, the second for music; the third for sound effects and the fourth for ambient sound.

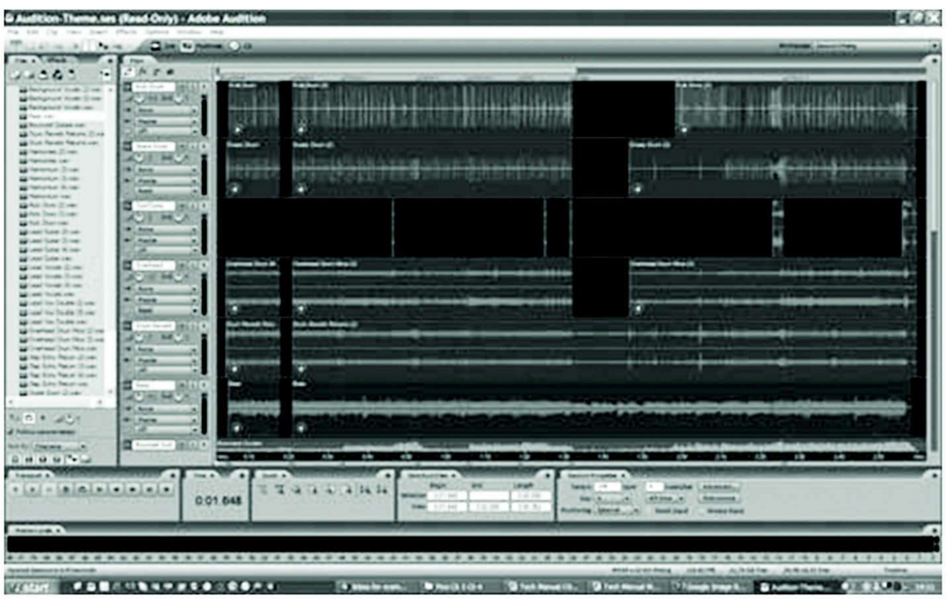


Figure 8.4: The editing interface. Note the source clips in the window to the left; the name of the editing session file at the top; the timeline with the various tracks arranged vertically, and the audio clips placed on the timeline.

Step 3: Cutting and reordering

Now, comes the business of actually removing the sections of audio we do not wish to retain. All editing software gives you the tools to cut audio at a particular point; copy audio from point to point; move audio from one point to another; and paste audio that you have cut or copied from another point. These four commands are usually enough for what we have to accomplish next.

If you have placed your audio on the timeline as per your script, the audio should now represent more or less the flow of ideas that you originally imagine within the programme. The general system is then to listen to the audio from the beginning, while referring to your notes on what to keep out of each clip. Whenever you reach a point, where you need to excise a portion, you place the playhead at the beginning of this portion (this, to the left margin), and make a cut. You then move the playhead to the end of the relevant portion (to the right margin of the section) and make a second cut, like cutting out a section of ribbon in the middle of a larger piece. This will leave a gap in the audio on the timeline. You then simply select all the clips to the right of this gap and drag them leftwards to close the gap. Repeated systematically across the entire timeline you will have a series of ordered clips, which contain precisely the audio you wish to retain.

Before we progress to the next step, it is worth remembering some key concerns that we should keep in mind:

a. Cadence and rhythm: Every person has a way of speaking, including a pattern of breathing when one speaks that gives our speech a characteristic set of pauses and spaces. You may like to cut out irrelevant sections of the audio, unwanted interruptions in the ambient audio (barking dogs, a telephone ringing, a slamming door), and the long silences which interrupt the flow of the audio. But, if you cut too much, the speech will lose its natural rhythm and become staccato and unnatural, not to mention unpleasant to hear. Similarly, the entire programme should present the things you wish to include at a particular pace that is neither too even nor too rushed. In many cases, this pace will wax and ebb, giving the programme itself a character.

b. Cutting order: Typically, one would start by cutting and reordering the voice tracks first; followed by the ambient audio linked to the voice tracks, then the sound effect tracks and finally the music tracks. This is because the core structure of the programme is usually decided by the spoken sections, so it makes sense to cut these first. Music is mainly used for mood and punctuation, so it makes sense to do this last, when the timing and order are more or less finalized. Not doing it in this order often results in wasted effort in re-doing adjustments and cutting over and over across all the tracks.

c. Reinvention: Presumably, the arrangement of your audio clips on the timeline reflects the flow of ideas and concepts presented in your original script. But, as you cut a programme, you will find that new linkages form between the materials and sometimes a whole new presentation of the story begins to appear. Be alert to this, and how it meets the purpose of your original script. Sometimes, being open to such a change can make a programme more dynamic and convey ideas in a better way.

Step 4: Finalizing the cut

If you have successfully made it this far, you should have the entire programme and all the components that you require within it, more or less in their final places. At some points, you would have made some preliminary adjustments to audio and created the relevant overlaps of audio that allow each section to flow smoothly to the next (‘transitions’). This completed stage is called the rough cut of the programme. Your rough cut may be upto half or a third longer than the time envisioned for the final programme, leaving some scope for further refinement.

This is the time to preview the programme with others for suggestions and to test whether the programme can be understood in its current form and whether it is having the impact that you desired. Often, this may be the time to do another round of recording, in case something needs to be fixed or augmented. It may also be at times to strengthen your heart and admit that the programme is not making enough sense and redo the entire programme.

Once you have made these relevant additions or replacements to the rough cut, you are ready to undertake one final round of trimming and refinement, where you will adjust the clips to their final positions and lengths, remove any unwanted material that still remains, and even lose sections or sequences that are not completely relevant to the programme. The last is often a difficult process, as it is the most creative endeavour, which concerns programme length and broadcast schedules and often forces us to take harsh decisions with otherwise excellent material.

The last round of refinement should result in a programme with a timeline where everything left clearly belongs in the programme and where the various components are in the exactly correct positions they should be. This means the music starts at the exact instant it is required, and the sound effects synchronized as they should be. The product at this stage is called the final cut or simply fine cut for short. It now only needs to be mixed and mastered (see Unit 9) for the programme to be broadcast or distributed.

16.5.3 Destructive & non-destructive editing

In the early days of audio editing, working with the raw material on tape often meant that once edited, it became virtually impossible to recover the original audio clips unless you maintained multiple copies of each recording. Such editing is called destructive editing, since each change that you made irretrievable changed the recorded material itself.

This is where editing softwares and working with audio on a DAW are remarkably different. Editing in the older system often meant using the raw material itself, working with digital material means two things. One, that you can make multiple copies without fear of audio loss; and two, that when we place a clip on the timeline and make changes to it, we are not actually affecting the original clip.

Thus, when you make a cut on a clip placed in the timeline, you are not actually slicing off a section of the original clip forever: rather, you are telling the software that you want it to start playing the clip from the point forwards, Similarly, placing the clips in a particular order on the timeline merely instructs the software about the order you want the clips played in.

The safety of the original clips means they can be imported repeatedly into different programes, and retained in their original form indefinitely. Thus, timeline based multitrack editing on most audio editing softwares is also called non-destructive editing, as it does not affect the source materials in any way.

It should be noted here that most editing softwares also have a different mode, where clips may be edited individually rather than as we do on the multitrack setup. This mode is sometimes called the 'edit mode' – is destructive, in this, changes made to the clip are saved permanently, changing it forever. Edit mode or whatever it is referred to in your editing software of choice, should be clearly understood and used only in specific circumstances.

Activity 8.3

Practice these commands on audio editing software of your choice. If you have access to a computer, but do not have an audio editor installed, you can download Audacity from http://audacity.sourceforge.net/.

16.6 Equalization and Digital Audio Processing Equalization (effects etc)

Now that we have understood the basics of audio editing, there is one further concept that we need to be familiar with: the ways in which we can change the quality of audio and add some effects to the sound. (Note that this is not to be confused with the term 'sound effects', which indicates ambient sound items like dogs barking, or the sound of a door closing.). Let us start by looking at the adjustments we can make to audio quality.

16.6.1 Equalization

As you may remember from a previous unit, human beings can hear frequencies between 20 Hz and 20,000 Hz. Within these, the human voice tends to range between 500 Hz and 5,000 Hz. For practical purposes, we divide these audible frequencies into three groups: Low frequencies (LF), which cover the range between 20Hz and 500 Hz; Mid Frequencies (MF), which cover the range between 500 Hz and 5,500 Hz; and High Frequencies (HF), which cover the range between 5,500 Hz and 20,000 Hz.

Low frequencies are also sometimes referred to as bass frequencies and high frequencies as treble. You may already be familiar with the bass and treble controls on music systems. The bass control increases or decreases the level of the bass or low frequencies; and the treble control adjusts the level of the high frequencies. As every sound is usually a complex mix of various frequencies from low to high, adjusting these controls affects the amount you hear of a specific range of frequencies. This process of adjusting the relative levels of the frequencies you hear in a particular sound is called equalization.

In professional equipment, equalization controls are usually found on a mixer unit; or within the editing or recording software (where they may be present as a software controls). Typically, the equalization controls in professional equipment look like the ones in Figure 8.5:

Figure 8.5: Equalization controls on a hardware mixer unit. The two knobs on top control LF and HF; and the collection of knobs below together adjust MF. Equalizer section image Source:http://en.wikipedia.org/wiki/File:Equaliser-section.jpg; CC by BYSA

There are usually two knobs for the low and high frequencies. Since, the mid frequencies cover the human vocal range, there needs to be finer control over this range, and therefore, there are more than one knob. One knob controls the level of the mid frequencies *per se*. The second one called the Q-Control or simply 'Q' for short; controls the area within the 500 – 5500 Hz range that is affected most, by the first knob allowing a finer degree of control.

Between all the knobs it is possible too:

- improve clarity
- add bass to a thin voice and reduce the bass of a heavy voice
- make voices more audible against a noisy background
- emphasize or de-emphasize specific instruments or voices among a cluster of instruments or voices

As a rule, unless one is an advanced user and clearly understands the consequences of changing the equalization settings, one tries to record sound without any adjustments whatsoever except for basic audio levels. Equalization is applied at the editing stage to refine and polish the sound.

16.6.2 Digital audio processing (DAP) and effects

Digital Signal Processing refers to the manipulation of audio in its digital form, in order to introduce several types of variations and corrections to the audio. Where they once required special signal processing units that were distinct pieces of equipment in themselves, they are now routinely found as an integral part of digital mixers, and especially most editing software.

Among the many different types of processing that could be applied are:

1. Filters

Filters refer to a screening process that lets selected frequencies of audio pass through while restricting others. Thus, a low cut filter will sharply reduce the levels of frequencies below a certain limit within a specific sound (usually those under 200 Hz). Similarly, a hi-pass filter will let only high frequencies over a certain limit through. A band-pass filter can let you adjust a precise range of frequencies say, the mid-frequencies to be let through while the HF and LF are sharply diminished.

2. Compressors and limiters

Compressors look for audio that is crossing a certain preset audio level limit, at which point they restrict or 'compress' the excess audio to stay within the prescribed limit. Limiters, on the other hand will sharply cut off the portion that exceeds the preset limit.

3. Clean up

While recording audio, we often record subtle background hums that come from fans, industrial equipment or even electrical wiring related issues. Even the Earth's electromagnetic field sometimes registers as a background noise on unprotected cabling. Clean up processing can remove many of these intrusive presences. For example, the hiss often found on old tape recordings can often be cleaned up in editing software by applying a hiss corrector, which samples the offending frequencies, then subtracts it from the entire audio clip. Properly and skillfully used, you can make ambient noise vanish completely from recordings, but be aware that improper selection of the clean up areas can leave your audio sounding weak and toneless.

4. Effects

While the number of available digital effects now number in the hundreds and even thousands, there are relatively few which are commonly used outside the music industry. The most common ones are:

i. Delay effects: These include reflection effects with delays that mimic the reverberation (‘reverb’) of various spaces ranging from small rooms to concert halls, echoes on mountain sides and even the muffled ‘deadness’ of a padded cell. The effects controller usually allows you to set the precise delay time, allowing you to vary the effect precisely or use a preset that sounds like a given situation.

ii. Chorus effects: These make a single voice sound like several, by repeating the voice with minor delays and variations of tonality. The effect is of a chorus of several voices saying to sing the same thing.

iii. Distortion effects: Distortion refers to a range of sounds that change the shape or feel of the original sound to mimic the broken effect of exceeding peak audio levels, or of making the voice sound dry and robotic. Flanger effects and ‘wah-wahs’, often used for musical instruments, fall within this category as well.

iv. Telephone effect: The technology used in older telephone microphones, as well as their built-in emphasis on mid-frequencies (in the interest of voice quality), give the voice an odd inflection that is characteristic of landline telephones. When rendering telephonic conversations for a radio programme, especially in dramatic situations, this effect mimics the audio quality of the telephone voice and with a subtle reduction in the level of one of the two voices, a realistic sense of someone speaking over a telephone line can be created.

Nowadays, in most cases, the effect is added merely by selecting the relevant portion of the audio and selecting the effect from a drop down menu. A set of controls then lets you adjust the various parameters related to the effect. When applied from a digital mixer, the effects section is generally activated by a specific button, after which the specific effect may be selected from a displayed list.

Lastly, an important concept to remember is the proportion of the original sound to the processed sound that we retain in the final mix that we make. Original audio – the audio as it was before the application of the effect is often referred to as ‘dry’ audio. The audio after being processed is often referred to as the ‘wet’ audio. Sometimes, adjusting the proportion of the two and retaining components of both can make the sound 'feel' much better to the ear.

Activity 8.4

Sound Recording and Editing

http://tinyurl.com/nhpypz7

Using Audacity as a point of reference, make a complete list of the possible list of effects available within the software. Write a short description of each effect.

16.7 Understanding the DAW and Editing Software (Integrated Vedio)

Now that you have a good understanding of the components of a CR station, please watch the *video 'Sound Recording & Editing' once again. You may also watch the video on studio recording hardware for related information*.

The video provides you a detailed step-by-step visual explanation of a Digital Audio Workstation (DAW) and the audio editing software installed on it. In this video, you will see the entire workflow involved in editing and completing a programme till the fine cut stage, including:

- Transferring field recordings into the system and importing them into the editing software
- Conducting a studio recording
- Laying all the audio materials on the timeline and executing an edit

Once you have viewed the video, complete Activity 8.5 below.

Activity 8.5

With reference to the DAW and editing software that you saw in the video, investigate at least three popular audio editing software available currently and compare them on the following parameters:

1. Cost
2. Base operating system required
3. Features
4. Reputation (you may have to read reviews on the internet to understand this)

Which would you select for your CRS, and why?

16.8 Let Us Sum Up

In this Unit, you have been introduced to the concept recorded audio, and understood the procedure by which we capture a recording. We have also understood the process of audio editing, and understood the key steps we have to take in order to successfully select and edit audio into a programme.

We have understood the distinction between tracks and channels, and the difference between destructive and non-destructive editing. We have also seen the primary editing controls that we use to gather and refine audio as part of our editing process from assembly to rough cut to fine cut.

Finally, we have understood the concepts of digital signal processing and the addition of effects to the recorded audio in order to manipulate and correct it as required.

16.9 Model Answers to Activities

Activity 8.1

This activity tests your ability to follow the entire process of setting up and conducting a recording.

For a topic like 'My Dream Job', possible questions could be:

1. Tells us about your dream job.
2. Why do you consider this your dream job?
3. Do you think you will have the opportunity to ever hold a job like the one you dream of?

Activity 8.2

In a mult-track recording situation, the recording preferably starts with the percussionist. The order in which the recordings should happen, therefore, are:

1. The drummer
2. The guitarist
3. The flautist
4. The vocalist

Activity 8.3

In Section 8.5.3, you have seen a list of common commands and shortcuts that can be used as part of your operations on a DAW. It must be noted that these commands are only indicative, and can differ somewhat from softwareto software; and especially from operating system to operating system. For example, the CTRL+X 'cut' command on most Windows based software's becomes a CMD(Command)+X operation on Apple computers. It is worth familiarizing yourself with the specific variation of the command that is applicable to the software and OS that you are working on as part of this activity.

Activity 8.4

The list of internal effects in the Audacity audio editing software is as follows:

Sl. No.	Effect Name	Description of Effect
1.	Amplify	This effect increases or decreases the volume of a track or tracks.
2.	Bass Boost	Boost the low frequencies selectively.
3.	Echo	Introduces a delayed repeat of the original sound
4.	Fade in	Slowly increases level of audio
5.	Fade Out	Slowly decreases level of audio
6.	FFT Filter	A general filter that allows selective amplification of frequencies
7.	Invert	Flips audio samples upside down to allow common sounds to cancel
8.	Noise Removal	Removes selected frequencies from the audio, once selected
9.	Phaser	Combines phase shifted audio with original signal
10.	Reverse	Reverses the audio clip, so that it plays back to front
11.	Wah-Wah	Copies the wah-wah guitar effect

Activity 8.5

As part of this activity, you will realize that there are a large variety of audio editing softwares available in the market, and as FOSS softwares on the internet. Each of them has their own advantages and disadvantages in terms of cost and functionality. Similarly, the specific editing software you select may be available for multiple platforms and operating systems, or may be limited to a specific OS. Audacity, for example, is available for Windows, Linux and Mac OSX. Audition is available only for Windows and Mac OSX.

When taking a call on which one will actually be the best for your process or CRS, you will usually have to consider all these factors, as well as other factors, like the OS that the rest of the team is most familiar with and the OS running on other computers in the CRS before settling on one.

UNIT 9

Mixing and Mastering

Structure

9.1 Introduction

In the previous Unit, you learnt about the concepts of audio recording and editing. You were introduced to the workflows involved in both processes, and the key commands and tools that are used to import, arrange and shape audio using an audio editing software.

In this Unit, you will build on this by understanding the skill of balancing and levelling audio, and getting the edited programme mix ready. You will also learn the process of exporting the completed programme as a single mixed file with a specific set of parameters.

As in the previous Unit, it is important to remember that there are a large variety of operating systems, audio editing software and editing setups/DAWs available, today. Though the specific details and commands may differ between each of them, the basic principles involved in both mixing and mastering are pretty much the same across each of them. Thus, it is more important for you to understand the principles and objectives of each of these processes, rather than the minutiae of the software commands that will execute each process, since those can easily be ascertained from user manuals and while you work with the software itself.

9.2 Learning Outcomes

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

- explain the concept of multi-track audio mixing.
- discuss the need to match clip volumes and balance audio
- explain the concept of mastering and exporting a final output file.

Mixing

If you have successfully done your editing work to the fine cut stage, it means you will now have a session file here:

1. The clips have already been arranged on the timeline in their appropriate tracks (Voice on 1 and 2; Music on 3 & 4; and Sound Effects on 5 & 6 or whatever convention you have decided to follow).
2. The audio clips have been trimmed to their appropriate lengths (that is, we have only retained the portions of the recorded audio that we require in the programme).
3. Each clip is in the precise position on its track, where it is required with

the appropriate breathing space in the edit for the audio playback to sound natural and well paced.

4. The basic transitions have been executed with fade-ins, fade-outs, cross fades and other transitions already in place.

In this section, you will now understand three advanced procedures that need to be executed in order to finalize the audio, and make it ready to listen to: audio levelling, balancing and panning. Together, the three process let you mix the audio: adjust the audio levels of the various components and segments, so that we can hear the audio clearly comfortably, with our attention being drawn to the correct portion of the audio as required. Let us look at each of these processes in turn.

Levelling: The need to adjust clip levels

Audio levelling is the process of raising or lowering the audio levels of individual audio clips in order to make them all sound more or less of the same perceived volume.

But why do this at all? If we have recorded the audio keeping the audio levels in mind, then surely we have sufficiently addressed the issue already?

The answer is not that simple. Recording each individual item at the appropriate recording level may mean the clip itself has been recorded quite well, but that does not guarantee that the clips will sound good when you place them next to each other. Therefore, we have to undergo a second process levelling , in order to adjust the levels of the clips with reference to each other and to our programme level.

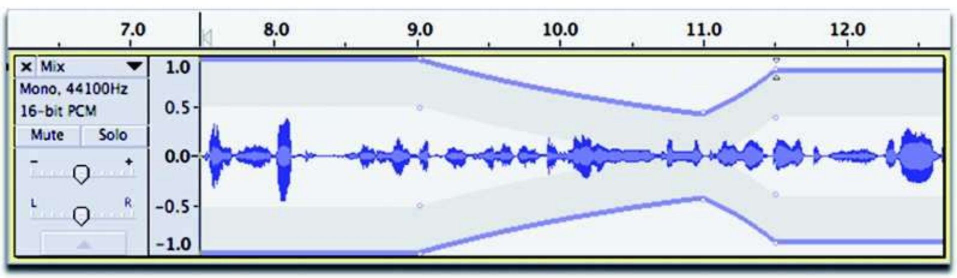
We need to do this so that:

- The clips transition smoothly from one to the other as we hear them, making the programme feel like one smooth whole, rather than the segmented composite it actually is;
- The clips can all stay within a certain audio level range, which will ensure that the audio does not distort by going outside the range, the equipment can handle (this will also ensure that we are able to combine the audio with the radio carrier wave for broadcast with no loss of quality).

Of course, the biggest reason we need to do this is so that our listeners can have an easier time of it, staying at a constant distance from their radio units rather than alternately leaning closer to hear better, and putting their fingers in their ears. If each clip were to play at a different level, it would make for a very awkward listening experience and would not let us concentrate on what was being said at all.

Naturally, it is important for us to still preserve some sections as relatively louder or softer than others. If people are naturally soft spoken, we need to keep their voices boosted, but still a shade less than the average voice on the programme. Similarly, if someone speaks in a booming, loud voice, we need to keep that slightly higher in level than the rest. Only then will the natural differences between the voices show up clearly. If we have to adjust all the voices mechanically to one level just by looking at the VU meter, this would be very unnatural. So one must use judgement and subjective hearing to make some of these calls.

Physically, the act of increasing or decreasing the level of a single clip is achieved in most audio editing software by one of two processes (many software allow you to do it both ways). By opening the property attributes of a given clip and increasing or decreasing the levels on a sliding scale, or by actually typing in the desired dB value in a box or by graphic interface. The graphic interface is usually in the form of a level overlay on the clip itself. Dragging the overlay line towards the bottom reduces the level below its current level. (Note that the same overlay usually lets you assign change points or nodes, and create fade in and fade out. But, what the graphical interface will usually not let you do is boost the levels beyond their current levels).



igure 9.1: A clip on the timeline in the Audacity software. Note the graphical overlay that has been used to adjust the audio levels at various points in the clip in order to even out the perceived audio volume through its duration.

Balancing: The need to adjust track levels

As we have already seen, a good and systematic audio editor tries to bunch similar types of audio into the same track(s), so that it becomes easier to edit a given type of audio. Having all the voice clips in Tracks 1 and 2, for example will save us from the bother of scrolling up and down in the timeline, while we are editing the voice segments of the programme.

There is another reason for this: once we have levelled the clips on a given track to a target value, having the same kind of audio on a track allows us to raise the volume or level of the track as a whole, affecting all the clips on the track together. Let us see why we would want to do this.

On the average, the most important part of an audio (radio) programme is the spoken word, because it is only through the words that we hear on the programme that ideas and concepts become clear. (The exception is probably instrumental music based programmes where words are not required to the listeners to appreciate the programme. But spoken words will be required in some point if not several times to introduce the artiste and the show).

Though, we will include variety of audio besides spoken words, but we will always favour the spoken word in most programmes in terms of audibility and clarity. Therefore, we need to adjust the level of the music track in such a way that the music is heard, but does not overpower the voice. Similarly, sound effects need to be heard in as natural a way as possible, but without obstructing the clarity of the spoken word. A thunderclap may be an important part of setting the scene in a dramatic production, but must be timed to a gap in the dialogue, with rumble of the thunder extending under the next stretch of dialogue without preventing the listener from hearing the dialogue itself.

The ability to adjust track levels as a whole is especially important in multitrack music recordings, where the ability to adjust the percussion (drum/tabla) track against the vocal track and the key instrumental (harmonium, veena, violin/ sarangi) tracks will make a difference between a good sound track and a confused one where you may not be able to hear the lyrics of the song clearly, or hear the soloist play his or her piece.

By combining our individual adjustment of clip levels (levelling) with the raising of lowering of track levels as whole, we will eventually be able to achieve an ideal combination of levels that not only lets us hear everything with the emphasis that we desire. This adjustment levels across tracks is called balancing and as we shall see balancing along with the process of panning, is the core in the act of mixing audio.

As in clip levelling, there are controls that allow you to adjust the level of the track as a whole. These will usually be in the form of a set of controls on the far left or far right of the track display. The tools may include a window, where you can type in a physical value (in dB) for the adjustment required, or a software knob that can be turned left or right to adjust the track level. Some software have a slider in the same place which is used for the same purpose.

Figure 9.2: The track level controls next to the track in a popular software editor. Note that the same controls are repeated for every track, allowing similar adjustments to be made to every track. Note the three individual VU meters marked A, B and C, indicating the audio levels for each individual track

Activity 9.1

For this activity, you will need access to an audio editing software. If you have one already installed on an accessible computer, you may use that. If you need one, download Audacity from http://audacity.sourceforge.net/.

Import any audio file into the software and place it on the timeline. Create a three second fade in one of the audio at the start of the clip, and a five second fade out of the audio on the clip.

Panning: Adjusting spatial distribution

You are already familiar with the concepts of mono and stereo sound. The basic difference is that in mono audio, irrespective of the number of monitor (speaker) units attached to the output, the same audio is fed to all the monitors, giving no spatial distribution to the sound. In stereo audio, different audio channels are fed to the right and left monitors, resulting in a spatially distributed audio experience. Some sounds are heard from just the left speaker, others only from the right speaker, and some from both speakers, resulting in a lifelike sense of distribution of the sound sources across the entire arc in front of us.

This effect is created not at the time of recording the audio, but after the editing is over, at the time of finalizing the audio. Controls for the purpose allow us to decide which channel a specific clip or track will be directed to and by how much. We can thereby create the sense of spatial distribution we have just discussed.

This spatial distribution of audio is called panning the audio. The original control used for this purpose used to be a resistor knob on hardware mixers called the panchromatic potentiometer, shortened to PAN-POT for convenience. This knob would be physically turned left or right of a central point to create the distribution of the audio across the two channels with extreme left being ‘only left’ and extreme right being ‘only right’. Points in-between would result in a partial distribution. The act of using the PAN-POT came, naturally, to be called, ‘panning’; and continues to be the term of choice for the process today.

In modern software audio editors, there are sometimes similar software knobs provided next to each track, close to the level controls to perform the same function. Many software auditors also allow you to achieve this with a graphic interface based tool, usually an overlay line horizontally in the middle of the clip. Raising the line towards the top of the clip pans the audio to the left and lowering it to the bottom of the clip pans the audio to the right speaker. It goes without saying that a lot of people, new to editing on software platforms get confused between the overlays for panning and level adjustment and get the two mixed up, a costly and time consuming mistake to make.

Using the panning function takes experience; a keen ear and a sense of music production that lets you understand how the acoustics of performances works. It is not something that you should experiment with till you consider yourself an advanced user of the system.

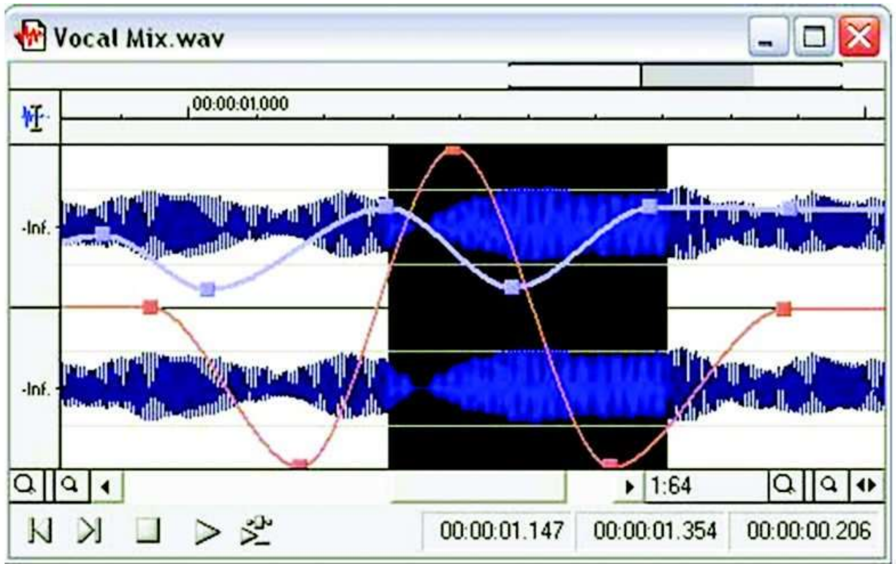


Figure 9.3: A clip showing audio level adjustment as well as PAN adjustments.

Note the graphical PAN overlay on the clip, which lets you pan individual clips.

Mixing audio

We can now summarize the process of mixing like this:

1. Levelling the audio at the individual clip level.
2. Balancing the relative levels of the various tracks.
3. Adjusting the spatial distribution of the audio to create the stereo effect.

Between the three, we will have achieved our purpose of making the audio listening-worthy, so to speak, in terms of its clarity, emphasis and in its ability to make the listener pay attention to the specific part of it we would like him or her to concentrate on.

The most important thing to remember while mixing is that it is important to set a target audio level for the final mix that we can standardize across programmes, so that our station's programming as a whole will be heard at more or less the same volume setting. Each station and broadcast agency has its own regulations regarding where exactly this target level should be, based on its technical setup and its preferences.

The best rule that one can follow is to set a target standard level between -12 dB and -6 dB, with the softest sounds going no lower than -8 dB or -18 dB and the loudest sounds touching no higher than -3 dB. If most of the programme lies around the -9 dB mark, this preserves a reasonable variation in the audio. You should be careful to see that none of the audio goes to the 0 dB mark at all, because that is the limit of what the system will handle safely without distorting the audio.

As noted in a previous section, if we skip the third step and leave all the tracks evenly distributed across both channels called centering the audio, we will end up with what is termed a mono-mix. The output will be mono audio with no difference between the signals going to the left and right speakers, if they are present. If there is a single speaker, as is found on many small transistor radios, this will not make a difference either way.

If we perform the third step, and assign spatial values to each clip and/or track by panning, we will end up with what is known as a stereo mix: an audio mix that will have the audio distributed across an imaginary arc in front of the listener. Such a mix, when played out through a pair of left and right speakers will recreate the sense of stereo listening that we usually feel by virtue of our two ears. Of course, this supposes that we have appropriate equipment to play out the stereo mix to a pair of speakers, the effect does not work with only one speaker.

Again, given that most CR stations have very low power transmission systems, it is wise not to attempt stereo mixing and outputs. These are not only time consuming and require greater expertise, but requires stereo transmission systems if they are to be broadcast successfully. Stereo transmission systems are more expensive than mono transmission systems, and also give a lower transmission range for the same power output, both are serious considerations for a small CRS. The overwhelming majority of CRSs usually prefer a mono transmission system as a result.

As the final product of the mixing process, some editors like to select all the clips and tracks for which they have made adjustments, and generate a fresh combined clip that incorporates all the elements together. This fresh clip is generated on one of the free tracks left after the audio has been laid on the first few tracks. This process is often called mixing down or bouncing the clips/tracks.

However, as you will see in the next section, this is an unnecessary action, since the process of mastering essentially incorporates this process of combining already. However, it is worth bouncing the mix to a fresh track as a sort of trial, if you like to preview the expected end product once for audio levels before mastering.

Activity 9.2

For this activity, you will need access to audio editing software. If you have one already installed on an accessible computer, you may use that. If you need one, download Audacity from http://audacity.sourceforge.net/.

Import at least two audio speech files to the software and place them on the timeline with the first clip on Track 1 and the second clip on Track 2. Ideally, Track 1 should be spoken by one person, and Track 2 should be another person. (If you can record audio that in Track 1 one person is asking some questions and in Track 2 another person is responding that would be ideal.)

Use the graphical volume controls to adjust the relative audio levels and balance the audio.

Now, use the graphical PAN controls available in your software to pan Track 1 fully to the left, and Track 2 fully to the right. Play the file back on a stereo speaker pair to check whether Track 1 (the interviewer) can heard from the left speaker, and Track 2 (the interviewee) from the right speaker.

Mastering and Export

To put it simply, mastering is the process of generating the edited finalized audio programme as a single mixed audio file in a format of one's choice.

During editing, the programme is composed of numerous segments and pieces. In order to manage the programme all these segments have to be put into a single piece, so that it can be lined up for playout and broadcast. If the programme remain as different component pieces then it would become a herculean task to line up the pieces. Mastering allows the generation of this mixed and finalized file with the levels varying as per the mixing settings and adjustments that one has previously made.

Before, we discuss the process of mastering further, let us look at some of the parameters which we have to keep in mind, while generating the audio programme in its final form. You may have to refer to the Unit on analog and digital audio (Unit 10) to brush up on some of these terms and parameters.

5. Mono or stereo: As we have already seen, depending on the mix, and availability of our skills and the infrastructure we have to make a mono or stereo final output.
6. Sampling rate: This is the number of samples each second of audio in your audio is divided into. You would have to set this while setting up the

audio editing software. Typically, CD quality audio is sampled at 44100 hz (or 44.1 kHz). Audio for video use is sampled at 48 kHz.

7. Bit rate: This is the amount of data per second that is transported in the stream of data that gets played out, while you play certain types of audio files. It is usually measured in kilobits per second (kbps). Typical values for FM quality audio are 96 kbps. CD quality audio will range from 128 – 192 kbps. Higher than this is strictly phrohibited for playback use on the system, rather than broadcast. Some softwares allow you to choose between Fixed Bit Rate (FBR) output files and Variable Bit Rate (VBR) files. In VBR files the system decides in which portion of the file, depending on the complexity of the audio, to use a higher bit rate and where to use a lower bit rate. In the interests of interoperability between systems, it is wise to stay with fixed bit rate files, where the entire file is encoded at a standard bit rate, as given above.
8. File format: The final audio can be exported in one of several types of audio formats (or file types) that are commonly used. Generally, while recording and editing, we keep audio uncompressed and therefore record and edit in the WAVE (.wav) format. The final master file can be in WAVE format (larger file size) or MP3 (smaller file size). Depending on the other parameters selected, selecting the MP3 option could create a file several times smaller than the equivalent WAVE file, resulting in enormous space savings. MP3 files are fine for most broadcast purposes, and have the dual advantage of being easy to line up in most audio player softwares on computers.

Thus, a typical output format for a CRS programme would be:

MP3/Mono/44100 Hz/128 kbps

It is best to discuss and settle on a standard file naming system for final master files, as well as common parameters for final master file generation. This will make everyone export their final programme files at the same settings, and will lead to an even sounding broadcast, not to mention make recognizing the master outputs easy.

Where the process of mastering goes, once the editing is complete and the clips have been levelled and balanced, we select all the clips on all the tracks on the timeline simultaneously. This can be achieved by shift-clicking the individual clips (which is more time consuming), or by selecting one clip and then using the CTRL+A command to select all the clips.

In most editing software, the next step is to export the finished file. This is usually achieved by using the menu command: File > Export. Within this command, you may be prompted to select between several choices (selected audio/all audio etc.): select the options that let you export the entire audio. You may then be presented with a window where you can select the other parameters, and name the file that is going to be exported. (In some cases, you will just get a window where you can name the file and decide where it is going to be saved. In such cases, there is usually a button marked ‘Settings’ somewhere within that window, clicking on which will give you the choices noted above. Note that some of the choices may need a specific file type to be selected in order to be activated.)

Once all the settings are set, and all parameters selected, clicking on OK should generate the final master file, which can then be saved in a location of your choice on the DAW. Note that the ‘export’ command actually performs the mixdown or bounce function, saving us from the additional step.

It is good practice to keep the master exported file of the programme in the same programme folder, which also contains the component recordings related to the programme and the connected session files. An additional copy of the file can be stored in a separate folder, from which playouts can be lined up. In fact, as a general principle, it is a good idea to keep two copies of all recordings for a programme, preferably on a different disk on the same computer or on an external backup device. This ensures that even if something gets accidentally deleted on the system, you have what is known as a safety copy of every file.

Activity 9.3

Continuing from the activity presented in Activity 9.2, import two instrumental music pieces into the same session file where you have the interview audio.

Place the first instrumental piece at the beginning of the programme and create a 3-second fade at the start of the music piece. Keep the music on for five seconds, then fade it down, so that it carries on under the first question on Track 1 before fading it completely.

Take the second instrumental clip, and place it at the end of the programme, so that it overlaps the end of the last response on Track 2 by about five seconds. Fade the end music up slowly under this last response, such that it reaches full level about two seconds after the last word is spoken. Keep the end audio up for five seconds, before fading it out over four seconds.

Rebalance the audio for all the clips till you are satisfied with the relative levels.

Now use the ‘export’ command on your software to export a single combined file with the following parameters: Mp3/128 kBps/44.1 kHz sampling. Name this file ‘Test.mp3’.

Let Us Sum Up

In this Unit, you learnt the final steps of the programme post-production process: namely, mixing and mastering.

Mixing is the process of setting the relative levels of all the components of audio sections, so that we have a listening experience that is both smooth and even and also directs the attention of the listener correctly.

Mastering is the process of combining all the constituent component files into one final file, whose file parameters and formats are standardized. This final file will have audio levels that vary as per the mixing related adjustments that has been made.

Mixing and mastering are done as per several parameters, which require a considerable amount of thought before selection (mono/stereo; FBR/MBR; file format; etc.).

Model Answers to Activities

Activity 9.1, 9.2, 9.3

The activities in this Unit are oriented around actually executing certain tasks on a software editor. For this, you will require access to a computer, and an audio editing software.

Your successful execution of these activities is based on the amount of time you will spend working on the software, and your willingness to experiment.

Try to follow the instructions closely, and use your judgement to assess what could be improved in the audio you are hearing as a result of your activity work.

UNIT 10

File Format and Compression

Structure

- Introduction
- Learning Outcomes
- Types of Audio Formats
- Need for Compression
- Compression Techniques
- Comparison of Formats (tabular)
- Format Converters
- Let Us Sum Up
- Feedback to Check Your Progress
- Model Answers to Activities

10.1 Introduction

Having learnt about the techniques of production and post-production in Unit 9, it is now time for us to proceed to learn the different formats in which audio files exist and how they can be used. We will also learn the various compression techniques. Scores of audio formats exist; some are software dependent and some are computer platform dependent.

In addition to audio formats, we will also learn about audio compression and its effects on audio.

It is important to know the different audio formats and audio compression techniques, since each file format has a specific application. For example, at the end of this chapter, you will realise that mp3 file format is a compressed file that occupies less space and can be easily used for broadcast and podcasting purposes. Also understanding open file formats versus proprietary ones helps one in deciding which file format to use in a particular instance.

You will need about four hours to complete this Unit, including completing the activities mentioned in this Unit.

10.2 Learning Outcomes

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

- discuss different types of audio formats.
- explain compression techniques., compress audio into different formats.
- compare and identify different file formats.

Types of Audio Formats

There are a number of file formats in the world of audio. When we talk of file formats, we are basically discussing digital files. Remember, digital information is only '1s' and '0s'. How these '1s' and '0s' are organised, determines the kind of file format. It is the difference in organisation of this information that differentiates an image file from an audio file.

File formats are nothing, but the way information is organised so that equipment or software understand them easily to work with them. For example, when you save a Microsoft Word file, the file format is ".doc". This means that all information in that file is organised in a way to enable software such as Microsoft Word to interact with (open, edit and save) it. Some file formats can interact only with specific software/equipment, whereas some can interact with more than one software/equipment. For example, an audio file such as mp3 can be opened by more than one software solutions.

Following are examples of file formats that you might encounter sometime at work:

- Images : BMP, JPG, SVG, GIF, PNG
- Text document : DOC, ODT,
- Sound : MP3, WAV, OGG
- Video : WMV, QuickTime, h264, mp4

The number of file formats has only increased over the years and technicians around the world have been constantly working on reducing file sizes, while trying to retain the originality of sound.

However, depending on the software one uses or the computer operating system (Windows, Mac, Linux etc), audio formats are either open standard or proprietary in nature. Open standard formats are those that will play across software and computer operating systems. On the other hand, proprietary formats will only work with specific software and on specific operating systems.

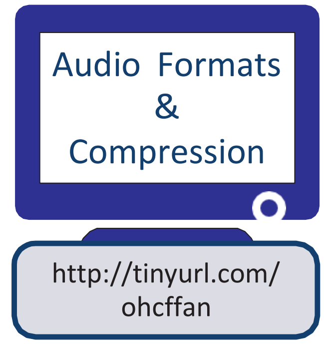
Before we proceed to understanding file formats, it is important to understand the difference between file format and codec.

A codec (short for Compression-Decompression or Coder-Decoder) means the way the audio is compressed and stored is called the *codec.* It is a piece of software that compresses an audio file and later decompresses it to be heard properly. Some file types always use a particular codec. For example, '.mp3' files always use the MPEG Layer-3 codec. Other files like '.wav' support can use different codecs like 'PCM', MPEG3 and many other codecs.

A file format is normally associated with a media player. Certain media players can play multiple file formats. Some can only play particular formats.

In short, a file format contains the content, the audio, and the codec is like a container. Let us draw an analogy here. Say you have a book with text and colourful images. You can print it on normal paper or on glossy paper. Printing it on glossy paper will make the colourful images stand out. The text and colourful images are the contents the file format. The normal paper or glossy paper is the codec. Let's take the example of an audio file itself. Say you have an audio file that is about one hour long. If you wish to stream it over the internet, you will need a certain codec such as MPEG Layer-3. This way, the file size is optimised for internet and streaming becomes easy.

Before going into the details of the various concepts of audio formats and compression, you can watch the video available at the CEMCA YouTube site - http://tinyurl.com/ohcffan. This video will help you to enumerate the concepts of audio formats and compression.



The most popular file formats, both open standard and proprietary are as follows:

- *wav* - standard audio file format used mainly in Windows PCs. Commonly used for storing uncompressed (PCM), CD-quality sound files, which means that they can be large in size around 10MB per minute of music. It is less well known that wav files can also be encoded with a variety of codecs to reduce the file size (for example the GSM or mp3 codecs).
- *mp3* - the MPEG Layer-3 format is the most popular format for downloading and storing music. By eliminating portions of the audio file that are essentially inaudible, mp3 files are compressed to roughly one-tenth the size of an equivalent PCM file while maintaining decent audio quality. This reduces the file size to a great extent. As a result, Mp3 files are popular for streaming on the internet and for storage purposes.
- *ogg* - a free, open source container format supporting a variety of codecs, the most popular of which is the audio codec Vorbis. Vorbis files are often compared to MP3 files in terms of quality.
- *flac* - a lossless compression codec. Lossless compression is like zipping a file but for audio. If you compress a PCM file to flac and then restore it again it will be a perfect copy of the original. (All the other codecs discussed here are lossy which means a small part of the quality is lost). The cost of this loss is that the compression ratio is not good.
- *au* - the standard audio file format used by Sun, Unix and Java. The audio in au files can be PCM or compressed with suitable codecs.
- *aiff* - the standard audio file format used by Apple. It is like a wav file for the Mac operating system.
- *wma* - The popular Windows Media Audio format is owned by Microsoft. Designed with Digital Rights Management (DRM) abilities for copy protection.
- *aac* - the Advanced Audio Coding format is based on the MPEG4 audio standard owned by Dolby.
- *ra* - a Real Audio format designed for streaming audio over the Internet. The .ra format allows files to be stored in a self-contained fashion on a computer, with all of the audio data contained inside the file itself. *.ram* is a text file that contains a link to the Internet address where the Real Audio file is stored. The .ram file itself does not contain any audio data.

Activity 10.1

Search the internet for the following file formats: mp3, ogg, au, mpeg2 and ra. Now try opening each of these files using Windows Media Player (WMP) on your computer.

You will notice that the player can only open certain file formats. This is because, while WMP interacts with some of them, it cannot with the others.

Need for Compression

When talking of compression, one should avoid getting confused between compression during recording and compressing audio files.

Voice has a dynamic range and so is the case with some instruments. As a result of this dynamic range, while recording, sound goes through several highs and lows. Compression during recording reduces such extreme shifts, reduces dynamic range and polishes the sound by controlling maximum levels and maintaining higher average loudness. Compression using hardware or software can also be used to slightly tweak an audio track to make it sound natural without distorting it in any manner. On the other hand, compressing beyond a limit can destroy the audio.

The compression that we are discussing here is *data compression*. But what exactly happens during compression? Compression of data essentially reduces the number of bits by removing redundant information thereby converting it to a file size of lesser size. All the wonderful pictures that you see on the internet were originally large files. Compression to web resolution reduces the file size by removing a lot of redundant information (mostly required for printing purposes) without affecting the way the picture looks. Images are optimised for web so that they load faster when you visit a web page.

Similarly, a smaller, compressed file reduces the amount of storage space required making it possible to store more music or video on a portable music player or hard drive and files can be transferred more quickly via the Internet or between storage devices. This means that when compressed the size of the file reduces.

Activity 10.2

This activity is meant to understand the meaning of data compression. Transfer a photograph from a camera that is about 1.5 MB or more to in a computer. Open the image using the Microsoft Office Picture Manager on your computer. Now zoom out the picture. You will still see a clear image. Next hit the Edit Pictures button. On the right side, you will see a number of options. Go down to the last option that says Compress Pictures. Click it. You will see the actual size of the original image.

Next, select the Web pages option. You will notice that the size of the image on the screen is reduced. You will also notice that the file size of the compressed size has reduced. Click the Ok button and save the file.

Now, open the file in the same Picture Manager. You will notice that while the picture looks almost original, however, when you zoom in, you will notice that the picture got a bit softer.

Compression Techniques

Compression is the process of reducing the size of a file by arranging data contained in it in a more efficient manner. By doing this, one is effectively removing the number of bits used to store any information. For example, in a Microsoft Word file, if you have entered text providing more space or you have redundant and repetitive words and phrases, it is naturally going to occupy more space. However, when you remove the unwanted spaces and redundant words and phrases, the size of the file is reduced.

Similarly, in the same Word file if you have used high resolution images, the size of the file would be very high. On the other hand, if you use smaller images (with smaller file sizes) the total size of the file will significantly decrease.

One step further, if you use a software to convert it into a .zip file, its size is again reduced. What you have basically done is to remove all redundant information to reduce the file size. It is now easy for you to email the document.

Compression can happen in two ways: Lossless and Lossy. Say you have an uncompressed .wav file. You can compress it to another file format using the lossless and lossy method.

In the lossless method, the file size is reduced but the quality of the audio is not compromised. Lossless compression is normally used when the quality of the audio is critical, say for example, on a music CD.

The lossy method of compression uses data compression methods where the file size is reduced but retains information that is just about useful. The mp3 files that we download from the internet are not of great quality but just about useful to store on our portable players and mobiles to listen to them on the go.

The amount of data an audio file format retains is measured in Kilobits per second (Kbps) (the bitrate). The higher the bitrate, the more data stored and the higher the audio quality.

You will know how to compress the file using reduced bitrates in the chapter on editing.

Comparison of Formats (tabular)

The following table (Figure 18.6.1) should give you a fair idea of the quality of audio compressed at different bitrates and their file sizes. All file sizes mentioned below are approximates only.

Quality (estimate)	Format (compression type)	Bitrate (Kbps)	Filesize (KB/min)	Compression Type
CD Quality	Uncompressed WAV	1411	105,000	None
	MP3	128	960	Lossy
	RA	96	720	Lossy
	WMA	92	690	Lossy but lossless in WMA9
	OGG	112	840	Lossy
	AAC	80	600	Lossy
FM Radio	MP3	96	720	None
	RA	64	480	Lossy
	WMA	56	420	Lossy
	OGG	67	500	Lossy but lossless in WMA9
	AAC	56	420	Lossy
AM Radio	MP3	64	480	None
	RA	32	240	Lossy
	WMA	20	150	Lossy
	OGG	32	240	Lossy but lossless in WMA9
	AAC	20	150	Lossy

Figure 18.6.1: A comparative table showing the approximate kilobits per second and the corresponding audio quality

Activity 10.3

Record a programme in an audio studio. Save the recording as a .wav file. Now, import the same audio file into an audio editing software such as Audacity and export it as an

- Mp3 file
- Ogg Vorbis file
- Mp2 file

Now check the size of each of the files you have converted and notice how their sizes will be different. Now repeat this exercise using different Bitrates (you can do it by choosing 'Options' before exporting the file) and then notice how their sizes differ. Also playback each of these files and notice the difference in quality.

Format Converters

Choosing the right audio compression for a given purpose is important. What is even more important is the format one chooses to convert the audio file into. There are two ways in which audio file formats can be converted. One method is by using hardware and another, using software.

Using hardware: The method is used while converting audio either from digital to analog or vice-versa. For example, analog to digital audio convertors (from tape/vinyl turntable to digital audio files). The convertor uses output from the analog audio player and feeds it into a piece of hardware that converts and outputs the file as a digital file through a USB interface that can be saved using software on a computer. The USB interface (the output from the convertor) is connected to the computer, which serves as an input. The software on the computer then digitises the file and outputs the same as a digital file.

This conversion method is used when archival material is in analog format (on tapes) and need to be converted to digital format for sharing purposes.

Activity 10.4

Through this activity we will attempt to convert analog audio from tape to a digital audio file. You will need:

1. A cassette tape recorder with an audio output
2. An audio cassette with some recording on it
3. An audio cable that connects output from the cassette recorder and serves as an input to your computer microphone
4. Audacity, a free and open source audio editing software

On your computer, launch Audacity. Load the cassette into your audio recorder and press the play button. On Audacity, press the record button. The audio playing on the cassette recorder will be 'captured'. Next export the file to a file format of your choice. You now have converted analog audio into digital audio.

Using software: Using software basically means that you already have a digital audio file in one format and you wish to convert it into another format to suit a purpose. Conversion of the file format means that one is converting an audio file into a lossless or lossy format. An uncompressed audio file can be compressed into another file format using lossless or lossy mode. On the other hand, a file that is already compressed in lossy format cannot gain anything even if converted into a seemingly uncompressed format.

Conversion from uncompressed to a lossy format will result in reduction of file size and quality, while conversion to a lossless format will compress the bit slightly in order without losing out on the audio quality of the original file.

There are several audio conversion software solutions, both free and licensed. One only needs to download them from the internet and use it on a computer. The audio recording/editing software Audacity too serves as an audio file format convertor. Using this format, you can convert a .wav file into an mp3 file and several other formats as provided by the software.

10.8 Let Us Sum Up

In this chapter, we discussed the different audio formats and how some are open standard and some proprietary. The most popular audio formats in vogue and their uses have also been detailed in this chapter.

We then moved on to understanding lossy and lossless compression, the hardware method and software methods of file conversions.

In the next Unit, you will learn about the different file storage and archiving options for audio. We will also discuss audio archiving and its importance in community radio.

Feedback to Check Your Progress

1. What is a file format? What use is it when it comes to dealing with digital files?
2. What are the most popular audio file formats?
3. What is compression? What actually happens when you compress a file?
4. Explain the terms lossy and lossless compression.
5. Explain the techniques behind audio file compression.

Model Answers to Activities

Activity 10.1

You would have noticed that a software player like Windows Media Player recognises only proprietary file formats such as mp3 or wma. A player like VLC Media Player will recognize both proprietary and open source file formats. You should remember that if you are producing a digital audio file that will be sent to many people who will be using different computers with different audio systems and with different media players, it would be better to have the same file saved under different formats so that all people can access the file easily.

Activity 10.2

What the Picture Manager has essentially done is to remove all the redundant data (the finer details, actually) available in the image file and compress it for the internet. However, the picture still looks almost like the original although smaller in size. Therefore, when you zoom in, you will see that the compressed picture is drastically different from the original file. It has become slightly fuzzy and the details available in the original would be missing from the compressed image. In short, the quality has also undergone a change.

Activity 10.3

.wav is an uncompressed file format and is of good quality. However, mp3, ogg and mp2 are compressed formats and the quality also suffers a bit. However, the file size is reduced significantly thanks to compression. Compression is also dependent on the Bitrate. Lower the bitrate, lesser the file size and more the compression.

Activity 10.4

You would have noticed that when you record the output from a audio cassette on to a digital audio file on Audacity, there is a difference between analog audio and digital audio. You should pay attention to the process of how an audio file (converted from analog) increases the mobility of the file. This means that you can import and export the digital audio file on a variety of softwares as compared to audio on the cassette.

UNIT 11

Storing and Retrieval

Structure

- Introduction
- Learning Outcomes
- Data Back-up Techniques
- Storage Devices
- Audio Archiving
  - Factors for archiving audio
  - Meta data tagging
- Logging
- Let Us Sum Up
- Feedback to Check Your Progress
- Model Answers to Activities

11.1 Introduction

In the previous Units you learnt about digital audio editing, mixing and mastering using software, compression of files and file formats.

In this Unit, you will learn about the various data storage devices, their characteristics and uses. You will also learn about audio archiving techniques and meta-data tagging.

You will proceed from learning the various back up techniques to types of storage devices like CDs, DVDs, magnetic and solid state devices and their characteristics.

Thereafter, you will learn the significance of archiving in a radio station, the factors that go into archiving and the reasons thereof.

Searching for files when you have thousands of them, can be an uphill task. Meta data tagging is an effective way of narrowing down to the file you are searching. This Unit will also explain meta data tagging and methods of adding such tags to files.

File sizes (both uncompressed and compressed) have an impact on the way one stores them and the space required to store them. Decisions need to be taken on whether to use uncompressed file formats or otherwise to store them or archive them for future use. This chapter will help you decide on the storage devices and teach you back-up techniques to maintain redundancy in terms of file maintenance. Redundance (meaning having multiple copies of the same file) is not a sin when it comes to radio stations.

You will need at least six hours to successfully complete this Unit, if you undertake all the activities seriously.

11.2 Learning Outcomes

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

- describe various data back-up techniques.
- discuss storage devices.
- explain audio archiving.
- use meta data tagging and logging.

Data Back-up Techniques

Audio files are at the heart of any radio station. Losing these crucial files can mean stoppage of broadcast. Important files always have the habit of being deleted, most likely by accident. However, a file once lost will take a long time, energy and money to recreate. Thankfully, one can create a backup of all the files in use. A back-up is like making a copy of the existing files and storing them away for future use.

Backing up files is like having an insurance plan in place. Back up files always come handy when one's system crashes or someone deletes a file by accident.

For a radio station that is meant to be on air all the time, having a backup plan is crucial. Every radio station worth its name will necessarily have a backup plan. A backup plan is dependent on the following factors:

- *Importance of data:* It cannot be stressed enough that audio files are central to the working of a radio station. All audio files are important. However, there are some that are more important than others in terms of priority. Music loops and sound effects are important (and need back-up too), but they could be less important than the programmes themselves that can be used over a period of time. For example, if you have a programme on the biography of a national leader, you might want to back up that file for future use too.
- *The data itself:* What kind of data is important to you? Are the project files from your editing software, or just the output files in uncompressed format or just the mp3 files that you finally use for broadcast? Account for the data itself while drawing up a backup plan.
- *Frequency of change:* How often does the data change? For example, the CR policy requires that you backup all audio files broadcast for the last three months. This means, you will need to backup all the audio files that you have broadcast for each of the 90 days.
- *Backup equipment:* Your backup plan also depends a lot on the kind of equipment you have. What kind of media does your radio station have in order to execute the backup plans?
- *Backup schedules:* How often will you need to do a backup? Daily, twice a week, weekly, fortnightly or just monthly? Radio stations would do well to backup daily.

Broadly, backups can be classified as:

- Daily backups: As the name itself suggests, one could backup either the project file or even the outputs on a daily basis.
- Incremental backups: If the same file/programme undergoes a change over a period of time, the latest version replaces the earlier version. For

example, if you were to make a programme on Right to Information (RTI) today and back it up, the changes made to the law and the programme four years later will be replaced by the earlier file. This means that you will now be broadcasting a more current version of the programme.

- Copy backups: Irrespective of whether a project/file have undergone changes or not, they are backed up for future use. Take the earlier example of the programme on RTI. You may have produced a programme three years ago and backed up a file. While you make a new programme, you might use the same project file but save it under a different name. In this case, you will have a backup for programmes you produced three years back and the most recent one too.

Radio stations would do well to follow the Daily and Copy backup plans to be on the safer side.

One can backup files either on the computer itself in a different folder or on different media. While backing up files in a different folder on the computer itself can be convenient, if the hard disk of the computer crashes, one loses all data. Backing up files on external media storage would be wisest thing to do at all time.

So how does one determine a backup solution? Consider the following factors:

- Capacity: What is the amount of data you need to backup on a daily basis? How much space would one need for a month or a year?
- Frequency: How often would you want to create backups?
- Retrieval speed: How fast can you retrieve the data? This decides the storage device on which you backup.
- Cost: Ofcourse, this, is a decisive factor, particularly for community radio stations.

In the next section, we will discuss the different kinds of storage devices.

Activity 11.1

Visit the nearest radio station and find out:

a) If they have back up plans mentioned above?

b) If yes, how frequently do they back up audio?

c) Do they back up audio manually (transferring each file to a storage device) or does a computer software do it for them?

d) How much space is required to store a day's audio files that have been broadcast?

Storage Devices

Storage devices are those that can store information and support a system for accessing and retrieving the information using hardware interface. A storage device is a physical piece of equipment that can hold data/information. The information can be anything that can be stored electronically: software programs, source code, images, audio or video files, office documents, spreadsheet numbers, and a host of other file types. Mass-storage devices typically store information in files. A file system defines how the files are organized in the storage media.

A storage device is used when large amounts of data needs to be stored/ transferred from one place and retrieved at another place and at the time of using a hardware interface.

There are basically three types of materials that serve as storage devices:

- Magnetic: All the hard disks, storage tapes etc are all magnetic types.
- Optical: CDs, DVDs, Blu-ray discs etc that run and retrieve data using an optical interface are called optical storage devices.
- Solid state: All the memory sticks (pen drives), flash cards, memory cards etc. are all solid state storage devices.

Magnetic storage devices: These devices store all information/data in the form of magnetised dots. Small electromagnets in the drive create these dots, enable them to read them and also erase them. Magnetic storage devices can either be hard disks (fixed or removable) or even magnetic tapes.

Fixed magnetic storage disks are the ones that are found in computers. They act as the main storage device for a computer. Storing, accessing and retrieving data from these disks can be done at amazing speeds. They also have amazing capacities. For example, most computers today come with hard disks that range from 100 GB to even 2 Terra bytes (2000 GB)!

One of the advantages of such drives is that they make storing of files and backing up very easy. However, the disadvantage is that if they happen to crash all the data is likely to be lost forever.

Portable hard disks are removable types and can be carried from one place to another. They are mostly connected to the computer using a USB cable or a Firewire cable. They too come in large capacities and can be used to store a variety of data.

One of the major advantages with hard disks is that information can be stored on it and it will store the data even when the computer is switched off. When the computer is switched on or when a portable type is connected to a computer, the latter recognises it and helps us retrieve the files.

Magnetic tape: This type of storage, often called the tertiary storage types are used in large servers when the data is very huge in size.

Rotation per minute (rpm) is a major factor that determines the performance of a hard disk. Hard disk spins can range between 3600 rpm and 7200 rpm. The rigidity of the disk and the high-speed rotation allows more data to be recorded on the surface. The disk that spins faster can use smaller magnetic charge to make current flow to the read/write head. The drives' heads can also use a lower density current to record data on the disk.

Optical storage disks*:* An optical storage device is one in which laser technology is used to write and read data from the disk. The most popular storage devices next to the magnetic disk drives are the optical drives. These include CDs, DVDs and Blu-ray disks.

CD stands for compact disc and is still the most universally compatible format for wide audio distribution. A standard CD-ROM can hold about 700 megabytes of data and can be played back on a computer CD drive and most DVD players.

DVD stands for digital versatile disc. A standard VCD stores video and audio data in MPEG-1 format. A standard DVD stores its data in MPEG-2 format. A DVD player or a computer equipped with a DVD drive is required to play DVDs. A DVD is a very high-density optical storage medium. It holds almost thrice the data VCDs can hold. A typical movie of 2½ hr requires two VCDs. The same movie requires only one DVD. You can now imagine how much audio a DVD can hold.

The following table should give you an idea of the kinds of CDs and DVDs available in the market.

Disc	Disc Types	Data Capacity	Mp3 audio
CD	CD-ROM, CD-R, CD-RW	700 MB	80 mins
DVD	DVD-ROM, DVD+R, DVD-R, DVD RW	4.7 GB	72 hours
	Single layer, double sided	9.4 GB	140+ hours
	Dual layer, single sided	8.5 GB	120+ hours
	Dual layer, double sided	17 GB	240+ hours

Although optical disks are widely available in the market and nearly all computers come with optical disk drives and software to read and write data, these are soon losing out to hard disk drives. The main reason for the fading popularity of optical disks is that they are very delicate and any scratch on the disk means that the data cannot be retrieved. Hard disks on the other hand are very robust and can retain data for a much longer period of time.

Solid state storages*:* These new storage devices do not have any moving parts. Flash memory and pen drives are very common today. Flash memory is usually found in digital cameras, digital camcorders and mobile phones. A suitable drive is needed to read/write on flash memory. Pen drives use the same technology to read and write data. Currently, pen drives are capable of storing more than 32 GB of information. Recent pen drives are coming out with an inbuilt mp3 player too. A USB drive is required to connect a pen drive to your computer.

Major advantages with pen drives is that they are compact and portable, removable, faster to use, comes with high storage capacities and are more reliable, since they do not have any moving parts.

Other types of solid state storage devices, include memory cards. There are several different types of memory cards but for purposes of our study we will examine the following:

	SD Cards: The Secure Digital (SD) cards. They are the most popular cards used in audio and video applications. They are mostly the size of a postage stamp and even have a lock switch to protect the contents from being erased. Any piece of equipment that has SD written on its package will accept these cards. They come in two varieties — SDHC and SDXC. These cards are available in capacities varying from 4 to 64 GB.
	The Compact Flash cards are used by some audio recorders. They are slightly thick, bigger and even more reliable. What’s more, they are even cheaper than the SD cards. They too come in the range of 2-4GB capacities.
	Micro SD cards work on the same principle as SD cards. They are much smaller and are mostly used in mobile phones etc. Although very small, they can hold a large amount of data ranging from 1-64GB. However, one drawback is that they are too very small and run the risk of getting lost in the melee of production.
	Some other consumer end audio recorders also use xD cards. They too work similar to SD cards but these days are more limited to being used by Olympus products. Yet another drawback is that their capacities are limited.

Activity 11.2

Note:

This activity will need some persuasion since not all human beings are cooperative. Visit a shop that sells CDs, DVDs and solid state cards. Request them to show all the varieties they have. Now, examine the boxes and compare the contents with the notes you have with you. In all probabilities, you have used some of these storage devices, but may not have taken the time out to notice the differences between/among them. Check how their size, capacities specifications, particularly in the case of DVDs, affect their usage.

Audio Archiving

CR stations produce a variety of programmes ranging from health, sanitation, education, agriculture and entertainment. They also have a bank of music and sound effects that they can use over a period of time. However, as time passes, managing the astounding number of files can be an uphill task. Maintaining a searchable archive of files that can be used, shared and distributed later can be a daunting task, if proper archiving mechanisms are not put in place. Audio archiving is crucial to the functioning of a community radio station for the following reasons:

- Policy requirement: The Community Radio Policy of India requires all CR stations to maintain archives of three months. The policy guidelines state:
- The Permission Holder shall provide such information to the Government on intervals, as may be required. In this connection, the Permission Holder is required to preserve recording of programmes broadcast during the previous three months failing which Permission Agreement is liable to be revoked.
- Smooth functioning: As mentioned earlier, archiving (or backing up programmes) is also useful for the smooth functioning of the radio station. A programme that one may want to use later or share it with other radio stations or even distribute among community members for a small fee will be easier if proper archiving mechanisms are in place.
- Reflect: CR stations can look back once in, maybe two years to look at the kind of programming they have been doing and make changes to it.
- Celebrate: If your station has completed a milestone (say, five years) you might want to celebrate it by playing back select programmes from the past.

- Audio heritage: Community radio stations do a lot of oral history. These histories are important to look back on change. They help us understand how people experienced lives in a certain socio-political, cultural context and importantly also point us to what has not changed, since then.

Factors for archiving audio

So what factors does one consider when archiving audio?

- The storage device
- The file format
- The space available/required
- Searchability (remember, we want to be able to locate the right file from among the thousands)
- Speed of retrieval

One of the simplest ways of archiving is to use hard drives. One can probably allocate one hard drive for each year and label them, accordingly. Another way of doing it is by storing all the files on the internet, which is expensive and time consuming.

Yet another way of archiving audio is to use a storage system but use software to label the audio files properly so that they can be searched easily using the software. While there are a number of licensed software solutions, use of open source is desirable. For example, UNESCO recommends using WINISIS and Greenstone Software (http://unesdoc.unesco.org/images/0018/001808/180855e.pdf). Yet another software solution that can be explored is Avalon (http: //www.avalonmediasystem.org/software) that claims be open source.

In any case, the purpose of using such software solutions is to ensure that they meet all the requirements of archiving.

For an audio archive to be searchable, it will need to have all the pertinent data available for each of the audio files. This now takes us to Meta data tagging.

Meta data tagging

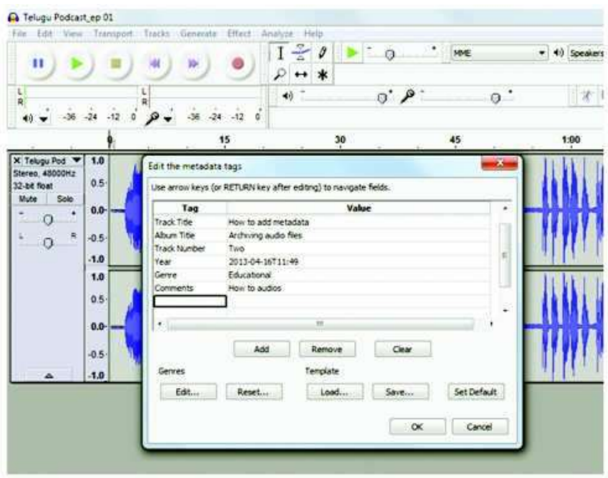
What is data tagging? We all give names to our audio files. However, when there are audio files on the same topic (say there are 200 programmes on environment or 2000 songs of a local folk artiste), how does one narrow down the search to a particular file? The solution is meta data tagging.

So what is meta data tagging? Simply, it is data that describes other data. Metadata can be used to describe digital files such as videos, images, documents, audio files and more. Digital files and assets are given value by the metadata that describes them, not only by making it possible to locate and repurpose them, but also to ensure others who wish to use or license an asset can find out where it has originated from.

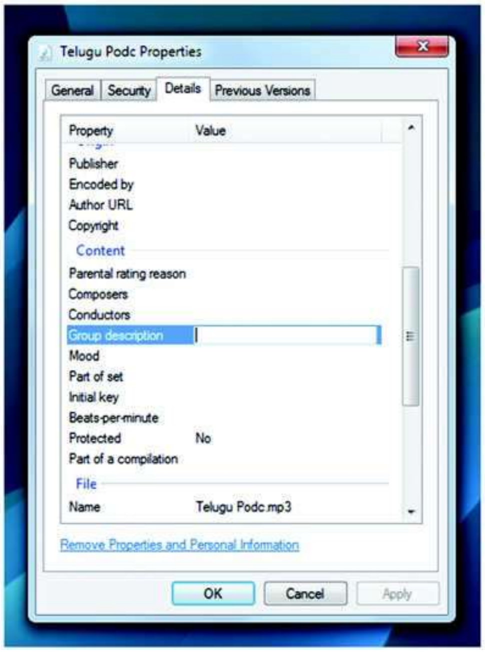
Most of the times the metadata entries that your audio editor provides are good enough and are searchable. It is searchable more easily when the metadata itself can be exported as a separate file. For example, how does one go about adding metadata to an audio file, while using Audacity?

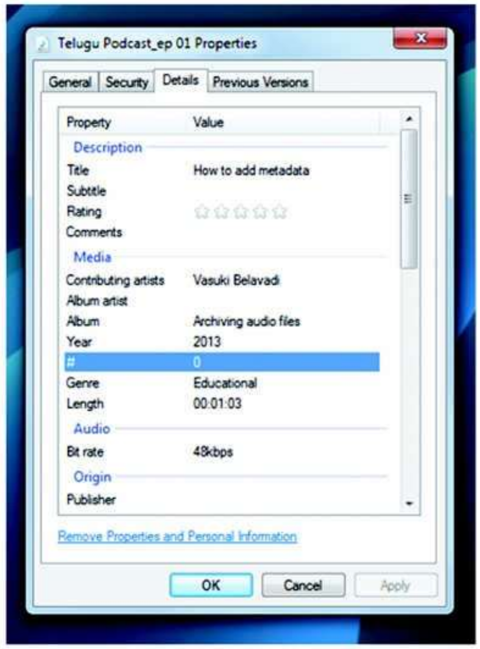
Here are some simple steps to follow:

- Import your audio into Audacity
- Click 'File' and then 'Open Metadata Editor'
- Enter the metadata. One can also add other fields to customise the data to be available.



Export the file. The file is then exported along with the metadata. So, how does one know the metadata has been included in the file? One way of finding out is to right click on the file and clicking properties. Then click on the 'Details' tab in the properties window. If the metadata has been entered, you will find it in there.





When such and more detailed information about an audio file are available, searching for it becomes easier. Although, there might be more than one file on the topic, it is easier to narrow down to the file at least.

It is worth to add as many details as possible soon after editing in the audio editor itself, so that the file becomes more searchable.

The typical details one would want to enter as meta data are mentioned in the table below:

Name of the file	Year of production
Name of the creator	Copyright details (Organisation, year etc)
Name of the publisher (the organisation)	Whether file is protected or not
The genre of programme (feature, drama, interview etc)	Composers, Conductors, Beats per minute, Whether it is part of a compilation (in the case of music)
Date of production	Additional comments (Ex: during 2004 floods)

The more details the more searchable it becomes. If you are using GRINS (Gramvaani) to schedule & broadcast your programmes, you will see why these details are significant

Activity 11.3

There is yet another way of adding metadata tags to a file. Try this activity to see if you are successful in entering metadata. Yet another way of adding meta data to an already existing audio file is to do it to the file itself. Follow these simple steps to do so:

1. Right click the audio file and click Properties
2. In the Properties window, click the Details tab
3. Slowly scroll down and click next to each of the single items that appear under the Content section. You can enter a number of details there.
4. Click the OK button.
5. Now right click the file and click Properties.
6. In the Properties window, click the Details tab. You will see all the details you had entered in there.

Logging

In radio, the word logging is used in two ways. One, before editing, where you make a list of audio files you have recorded, select portions of the audio clip to edit in etc. The other time you hear the word logging is during transmission.

If your radio station is using a broadcast automation software (that helps one in creating playlists, scheduling, taking phone-calls and broadcasting), it would also probably come with a programme logging option. Programme logging simply means creating a list of all the programmes (including the on-air announcements, PSAs, commercials, jingles, break bumpers etc).

Earlier, logging of all programmes used to be done manually. One would have to actually sit and listen to all programmes one after the other and enter them into a register. No longer. The software that automates your broadcast in all probabilities creates a programme log too. A programme log is necessary because it is:

- A legal requirement: If you are playing songs that require you to pay a license fee, the licensor, in all probability will require you to keep a log of all the songs you have played. This is to ensure that you have not infringed copyright.

- Beneficial to the RJs: Keeping a programme log is beneficial to radio jockeys; it lets them know which files/songs have been played/not played. This will help them create an interesting playlist.

Another important logging feature on broadcast automation software is the phone calls you receive at your station. This feature not only lists out all the numbers, but if you store the names of the callers (Figure 19.5.1), they will help you give a personal touch to the call when you receive a call from the same number. You can probably identify the caller by name to her surprise. This goes a long way in carrying forward a good conversation on radio.

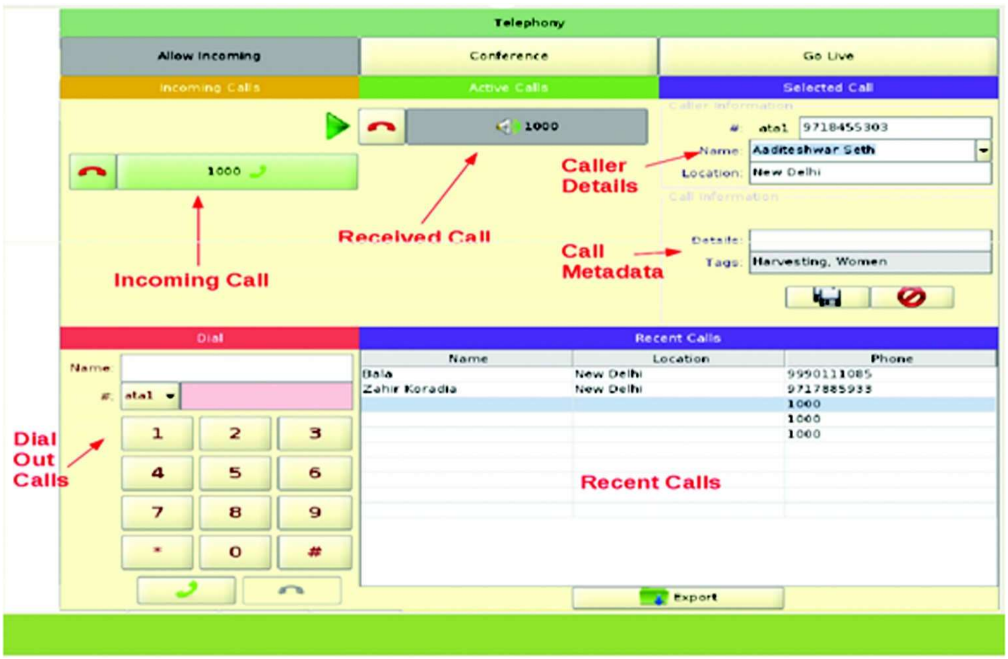


Figure 19.5.1: Screenshot of the telephony component of GRINS broadcast automation software. This allows you to store the names of the callers and save a list too.

11.7 Let Us Sum Up

In this Unit, we have learnt about the various data back-up techniques and the hardware required for the same. We also emphasised on the back-ups by classification. We learnt that there are magnetic and optical storage devices and which ones to use for longevity.

The importance of audio archiving for a community radio station was discussed in detail and the various methods by which archiving is done. Links have also been provided to open source software that can be used for archiving.

All audio files must be searchable from an archive. Meta data tagging is the best proven method for this purpose. We learnt how to add meta data to files while editing and already existing audio files in the bank.

In the next couple of Units, we will concentrate on best practices in studio operations, basic maintenance of radio studio and field equipment and alternative methods of content distribution.

Feedback to Check Your Progress

1. What is back up? What are the different back up techniques?
2. Explain in detail the different kinds of storage devices available in the market today.
3. What is meta data tagging? What is its significance in archiving?
4. How does one add meta data tags to an audio file? Explain the different methods to do so.

Model Answers to Activities

Activity 11.1

When you visit the radio station, you are expected to observe the methods through which the staff are backing up their programmes, if at all. If they are storing the audio files on the same computer which they use on a day-to-day basis, then it is an unsafe backup option as the computer could crash anytime due to extensive usage. If they are backing up their programmes on cassettes and/or Compact Discs, it is an expensive and unsafe option since these physical media are expensive to procure on a continual basis and are also prone to scratches or other kinds of damage. The preferable options for backup are hard drives which are reserved purely for backup or if they are backing up their progammes on a secure web server which is independent of physical machines. However, please note that both these latter options are also expensive and limited and the radio station may have to take a balanced decision depending on the monetary resources at their disposal.

You may also observe the frequency of their backing up procedures. Ideally, if they are producing programmes on a daily basis, then the backup should also take place daily. However, due to limited number of people working at their radio station, and due to lack of bandwidth, they may take longer to backup their programmes. Finally, also please observe whether the back up is done manually or through a software like GRINS (on Linux) or Time Machine (on Apple). Obviously, manual back up is time consuming and prone to minor errors. Please also make a note of the amount of space that is required by the radio station to create back up. If the radio station is saving files in mp3 format, then lesser space will be required while saving backup files as wav files will need exponentially more space.

Activity 11.2

Once you have persuaded a shopkeeper to show you various storage devices at his shop, you are expected to observe how different devices have different storage capacities. You will observe that different devices are of different sizes too. The storage devices in the past, like CDs and DVDs are larger in size because they are optical drives which are meant to be read on computers or CD players. The recent devices like SD cards, micro SD cards etc are meant for more compact devices like cameras and mobile phones. The fact that some SD cards are restricted to some readers is because of a phenomenon called Digital Rights Management (DRM). This is also the reason why different phones have different charging cables and ports with different voltage and power readings.

Activity 11.3

Yet another way of adding meta data to an already existing audio file is to do it to the file itself. Follow these simple steps to do so:

1. Right click the audio file and click Properties
2. In the Properties window, click the Details tab
3. Slowly scroll down and click next to each of the single items that appear under the Content section. You can enter a number of details there.
4. Click the OK button.
5. Now right click the file and click Properties.
6. In the Properties window, click the Details tab. You will see all the details you had entered in there.

Glossary

A/D Conversion	Conversion of an audio signal from analog to digital. In a field recorder, this usually happens within the field recorder itself. In the studio, this happens within a sound card or a digital mixer.
Assembly	The process of arranging audio clips with reference to each other in order to create an audio programme.
Audio editing	The process of removing unnecessary audio from recordings, as well as the process of reordering and arranging audio.
Audio levelling	The process of adjusting the mutual levels of all the clips of audio in a programme to make it possible to listen to the programme at approximately the same level setting.
Audio levels	A measurement of the amplitude of the audio signal.
Audio recording	The process of capturing and storing audio on a medium.
Backup	Back up is like making a copy of the existing files and storing them away for future use.
Balancing	The process of adjusting audio levels between various segments of the programme.
Band pass filter	A filter that removes a collective set of frequencies from an audio clip.
Bouncing the audio	The process of mixing multiple tracks down to a single track. Sometimes done in order to control the number of tracks being used, or as a preliminary step to exporting.
Cadence and rhythm	The pace and feel of the edited programme.
Centering the audio	Removing any spatial distribution adjustments to audio in a programme to result in audio that can be heard evenly across left and right speakers.
Chorus effect	A type of multiplier effect where a single voice or instrument can be made to sound like a group of voices or instruments.
Compact Flash	Another commonly used type of flash storage medium. Often referred to as CF.
Cutting	The process of editing and refining audio.

Cutting order	The order in which the audio is going to be edited.
DeciBels (dB)	A standard unit used to measure the loudness, and thereby the related amplitude of a given audio signal.
Destructive & non-destructive editing	Destructive editing is a process where the act of editing changes the source material permanently. Non-destructive editing is a process where the act of editing only changes references to the source material, without affecting their source material itself.
Digital Audio Workstation (DAW):	A computerized system to manage, edit and process audio.
Digital Signal Processing (DSP):	The process of manipulating and adjusting digital audio content.
Distortion effects	Effects like the Flanger – wah-wah which make the original sound feel more unnatural by deliberately distorting the audio.
Dry and wet audio	The raw audio before any effects or processing is called 'dry'audio. The audio once processed or with effects applied is called 'wet' audio.
Editing tools	The tools available within an audio editing software in order to trim, refine, and rearrange audio content within the software.
Equalization	The process of adjusting the relative mix of frequencies in a given piece of audio.
Exporting the final file	Sending the final mixed audio file out of a software editor as a single file in a format and a quality of choice.
Field Recording	A recording conducted outside the studio.
File formats	The specific way in which the audio file stores the digital information. Different file formats employ different algorithms to store the relevant audio information, ranging from comparatively lossy processes to comparatively loss free methods. Common file formats include WAV, MP3, FLAG, OGG.
Final cut	The finished programme that includes all the relevant audio and the related transitions.
Fixed Bit Rate (FBR)	A setting employed while exporting to certain file formats (especially MP3), where the data rate is constant throughout the file.

Flash storage	A type of reusable digital storage medium that is based on electronic transistor gates. Common types are SSDs, SD cards and CF cards.
Graphic overlay tools	Visual tools provided by many software editing tools to allow easy adjustment of parameters like clip volume, stereo panning etc.
Hi pass filter	A filter that allows only high frequency audio to pass.
High Frequency (HF)	Frequencies in the range between 5500 Hz and 20000 Hz. Also known as treble.
Hiss corrector	A noise filter that removes high frequency hiss noise from an audio clip
Importing audio	The process of listing audio references to the clips within an audio software, to allow the placement and manipulation of the audio within the software.
Logging	A list of all the programmes/phone calls received by the radio station.
Low cut filter	A filter that removes low frequencies from an audio clip.
Low Frequency (LF)	Frequencies in the range between 20Hz and 500 Hz. Also known as bass.
Mastering	The process of creating a final mixed down file in an audio format and quality of choice for storage, playback and/or broadcast.
Metadata	Data that describes other data. Metadata can be used to describe digital files such as videos, images, documents, audio files and more.
Mid frequency (MF)	Frequencies in the range between 500 Hz and 5500 Hz.
Mixdown	A final composite track or file which combines all the audio segments of a given audio programme, with all audio level adjustments completed.
Mixing audio	The process of raising and lowering audio levels of different clips in a strategic manner in order to result in smooth transitions between the various segments of the programme.
Mono & Stereo audio	Audio recorded on a single channel without any determination of spatial spread is called Mono audio. Information recorded on two channels which together offer a spatial distribution of the audio is called Stereo audio.

Mono-mix	A final mix that results in audio with no spatial definition, that can be heard evenly across left and right speakers.
Optical storage device	A disk in which laser technology is used to write and read data from the disk. Examples: CD/DVD/Blue-ray disk.
PAN-POT	Short for Panchromatic potentiometer, a tool to adjust the spatial distribution of audio in a stereo space.
Panning	The function provided by the PAN-POT tool, referring to the distribution of audio signals in a stereo space.
Playhead/ Cursor	A tool available in most audio editing softwares to represent the actual point in the audio placed on the timeline that is being played back.
Q-control	A control to set the mid point of an adjustment affecting the mid-frequencies on an equalizer.
Reference track	A track recorded to provide a constant point of reference when recording subsequent tracks as part of a multi-track recording. Also known as a Click track, for the rhythmic clicks often recorded as part of the reference.
Reordering	The process of re-arranging audio clips.
Reverberation/ Echo effect	A type of delay effect that creates a repetition of the original audio after a pause. If the pause is short, the result is a perceived extension of the original sound, called reverb. If the delay results in a distinct gap between original and repeat, the repeat is called an echo
Rough cut	A first assembly of the programme content that approximates the programme.
Safety copy	A backup copy of a file, in order to avoid unforseen losses due to data corruption or system failure.
SD Card	Secure Digital Cards are a type of commonly used flash storage.
Session file	A file opened within an audio editing software in order to conduct the business of editing, mixing and mastering.
Single track & multi-track recording	When a recording places a single or a composite signal on one track of a recording unit, it is called a single track recording. When multiple discrete signals are placed on different tracks on a recording unit, it is called a multi-track recording.
Solid State Devices	Chips that store data. These chips do not have any moving parts but are cost effective.

Spatial distribution of audio	The division of sounds across left and right channels in order to create a listener experience that duplicates the human experience of listening with two ears.
SSD	Solid State Drive. A large capacity drive based on flash storage.
Stereo mix	A final mix with audio distributed across the left and right channels/speakers, in order to create a listening experience that mimics human being by natural hearing.
Stereo transmission system	A transmission system that is capable of broadcasting stereo audio.
Storage device	Storage devices are those that can store an information and support a system for accessing and retrieving the information using a hardware interface.
Studio Recording	A recording conducted within the studio.
Target audio level	A reference level selected for a programme to which all the audio clips will be adjusted in order to achieve a final mix.
Telephone effect	An effect to copy the effect of a voice heard over a telephone instrument, typically by reducing levels, removing the low frequencies, and emphasizing the mid-tones.
Timeline	A tool within most common audio editing softwares which allows one to arrange sections of audio with reference to each other in time.
Transferring audio	The process of moving recorded audio from one device to another, typically from a field recorder to an editing system.
Transitions	The changeover of one audio clip to another, and the effects applied to make that change smooth. (For example, a cross fade.)
Variable Bit Rate (VBR)	A setting employed while exporting to certain file formats (especially MP3), where the data rate varies based on the complexity of the content at a given point in the audio.
Volume Unit (VU) meter	A standard meter used to measure audio levels on a device.

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Studio Operations

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CONTENTS

About the Module

Module Description

Module 6 on studio operations is the last module of Course II: CR Production System and Technology. Module 6 deals with practical aspects of studio installation, operations and maintenance and content distribution using alternative mechanisms. After studying modules 3, 4 and 5 covering studio technology, production and post-production, it is important to learn the practical aspects of the studio operations. This module covers good engineering practices to be followed in installation and operation of the studios, preventive and corrective maintenance aspects to ensure that studio equipment give trouble-free service. This module also covers the study of alternative mechanisms for content distribution. It involves 37 hours of learning and has assignment and video to help your learning. It has three units. With this you will complete studying Course II and will be ready to take up Course III on Transmission System and Technology.

Module Objectives

After completion of the module, the learner should be able to:

- Demonstrate proper techniques of wiring, fixing of connectors, soldering and use of tools and equipment for studio work.
- Carry out preventive and corrective maintenance of studio and equipment installed therein.
- Describe different methods of content sharing using alternative platforms.

UNIT 12

Good Engineering Practices for Studio Setup

Structure

- ✓ Introduction
- ✓ Learning Outcomes
- ✓ Tools and Equipment
- ✓ Techniques of Handling Various Tools
    - Soldering tools
    - Stripping tools
    - Crimping tools
- ✓ Types of Connectors
    - Audio Connectors
        - 25.5.2RF Connectors
- ✓ Types of Audio Cables
- ✓ Let Us Sum Up
- ✓ Model Answers to Activities
- ✓ Additional Readings

Introduction

In Unit 11, you learnt about the working and use of the components of an audio chain, starting from microphones to programme production equipment such as audio mixers and digital work stations. However, in order to get the best performance from these equipment, it is necessary to follow certain good engineering practices during the installation stage itself. “Good engineering practice” is a term applied to all the activities related to the quality setup of a studio project. It means that each activity must be completed with perfection and precision. It is a skill which comes with practice. In this unit, you will learn about the good engineering practices involved in the installation, wiring, fixing of connectors, use of tools and equipment, and day-to-day operations in a community radio station especially with reference to studio equipment. In Unit 29, you will learn about the good engineering practices in respect of transmitter, RF cable and antenna system. In this unit, we shall focus on the application of good engineering practices in respect of the following topics:

- Tools and equipment
- Techniques of handling various tools
- Types of connectors
- Types of audio cables

In the video on “Working with Tools”, you will get a chance to see the demonstration on the techniques of use of various tools for wiring, soldering, stripping and crimping. It will show various types of audio connectors along with the process and precautions required while fixing of these connectors. The video will also demonstrate the use of test and measuring equipment for checking the performance of the studio equipment. This will definitely help you in understanding the use of a right tool for a specific job.

The glossary at the end of the module shall be helpful in understanding the content of this unit.

You may need about 8 hours to complete this unit including answering the questions given in the Activities.

Key words: Audio/Line cable, Mic cable, RF cable, Connectors, Soldering, Tools

12.2 Learning Outcomes

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

- list, identify and describe various types of cables and their usage.
- list, identify and describe various types of connectors and their usage.

- describe the techniques of using soldering, stripping and crimping tools for wiring and interconnecting of cables in a studio setup.
- describe the process and precautions to be observed while fixing the cable end connectors.
- describe the use of various test and measuring equipment for checking the performance measurements of the studio equipment.

Let us begin with tools and equipment.

Tools and Equipment

For installation and testing of a CRS, it is essential to follow certain engineering practices in respect of wiring and fixing of connectors. Proper use of tools and equipment helps in achieving quality results. In this section, you are going to study about the list of various tools and equipment required at a community radio station along with the purpose and function of each item. Table 20.1 gives a list of the tools and equipment commonly used at a CRS.

Table 20.1: List of Tools and Equipment along with Field of Application

Sl. No.	Tools and Equipment (including test and measurements)	Field of Application/Use
1	Soldering irons of 40W and 100W with tips of various sizes	For soldering pins of audio cables and connectors.
2	Temperature controlled soldering station with accessories	For soldering joints on printed circuit boards etc. where controlling of temperature is important.
3	Set of screw drivers of assorted sizes	For opening the covers of equipment, accessories and connectors etc.
.4	Set of watchmaker's screw drivers	For opening and tightening of mounting screws of PCBs, miniature connectors etc.
5	Spanner set	For opening and tightening of nuts and bolts.
6	Hand drill with assorted sizes of drill bits	For drilling holes on PCBs and mounting plates.
7	Crimping tools	For fixing the lugs and connectors requiring pressure fitting.
8	Set of pliers (long nose, wire stripper, cutter etc.)	For cutting cables, stripping of cable insulations and fixing of connectors.

9	Multi meter (digital)	For checking voltages, currents and resistances of circuits.
10	Continuity tester	For checking continuity of wires and cables.
11	Phase tester	For checking the availability of phase voltage.
12	Light duty blower/ suction	For removing the dust from racks and cubicles
13	Light duty vacuum cleaner	For cleaning the delicate units such as PCBs.
14	Earth tester	For measurement of Earth resistance.
15	Megger (insulation tester)	For checking the insulation resistance between the live and earth wires.
16	Tong tester (Clip-on-meter)	For measuring the currents flowing through phase wires.
17	Sound level meter	For acoustic measurements of studios.
18	Audio Generator	For feeding audio frequencies at the required frequencies and levels to Equipment while doing the measurements.
19	Audio Analyser	For checking the performance measurements of audio equipment such as frequency response, distortion and signal-to-noise ratio.
20	Cathode Ray Oscilloscope	For studying and analysing the waveforms during repairs and trouble shooting.

In Table 20.1, you can see the function of each tool and equipment and their requirement for the installation of a studio set up in a CRS. The video on "Working with Tools" showing the use of these tools and equipment will make the concepts more clear. Good engineering practices here mean using these items with precision and perfection. However, you will be able to develop the necessary skills only by practice. Before going into the details, you should watch the video, which is available at http://tinyurl.com/nr5rtpc .

Working with Tools

http://tinyurl.com/nr5rtpc

Activity 12.1

While viewing the video, you should note and identify the tools and equipment (including the test and measurements) along with their use. Write down your observations on list of tools and their use as shown in the video and others described in the text of Section 20.3 above in about 100 words in the space provided. To do this activity, you may need about 15 to 20 minutes. This activity will help you appreciate in identification of various types of tools and equipment. This activity may also help you in understanding the necessity of proper tools for different types of applications.

Having learnt the identification and field of use of various tools, we will now proceed to discuss the techniques required while using these tools in the following section.

Techniques of Handling Various Tools

In this section and the subsections that follow, you will learn about the techniques of handling various tools required for soldering, stripping, crimping and fixing of connectors.

Soldering tools

In this subsection, you will learn about the various types of soldering tools, accessories and techniques required for making a good solder joint. There is a huge range of solder joints to be made in an installation starting from fixing of tiny chip resistors on circuit boards to large size VHF connectors. A large variety of soldering irons, tips and solder wire (metal) are available in the market. You have to choose a right tool for each specific job. In this section, we will focus our discussion on soldering and fixing of connectors on audio cables and soldering joints especially applicable only to CRS setups.

Figure 20.1 shows tools and accessories required for soldering.

Figure 20.1(A): Soldering station

Figure 20.1(B): Soldering tools

As can be seen in Figure 20.1(A), the list of tools consists of a soldering iron, solder wire, a side-cutter and a few soldering tips of different sizes. Different soldering jobs will need different tools and different temperatures too. For example, for replacing a resistor on a printed circuit board, you will need a fine tip, a lower temperature and finer grade solder. You may also require a magnifying glass to see the fine tracks on the PCB. On the other hand, for fixing an XLR connector, you will require a larger tip, higher temperature and thicker solder. Use of clamps and holders are also handy when you are soldering audio cables. Figure 20.1 (B) shows a temperature controlled soldering station with which you can do good soldering jobs at the required temperature such as on printed circuit boards.

While choosing a soldering iron for a particular application, you may have to consider the following important points:

- Wattage of a soldering iron.
- Adjustable or fixed temperature control settings on a soldering station.
- Portable or bench use type of soldering iron.
- Size of the soldering tip.

Soldering Accessories

In the process of soldering, you may require a large number of accessories which are helpful in doing professional jobs. Some of these accessories with their functions in described brief are below:

(i) Solder

It is the soldering metal which is used for making solder-joints. The most commonly used type of solder is rosin core. The rosin is a flux, which cleans as you solder. Rosin core solder comes in three main types – 50/50, 60/40 and 63/37. These numbers represent the percentage of tin and lead present in the solder as shown in Table 20.2.

Table 20.2: Composition of different types of soldering metal

Solder Type	% Tin	% Lead	Melting Temp (°F)
50/50	50	50	425
60/40	60	40	371
63/37	63	37	361

As can be seen in Table 20.2, the type of solder metal to be selected depends on the percentage of tin and lead used in them. Also note the melting temperature shown against each type of solder metal. Higher the percentage of lead in a solder, higher is the melting temperature.

(ii)Soldering Iron Tips

Tips of different sizes are available. Try to use the right sized tip for the specific job.

(iii) Soldering Iron Stand

It is a heat resistant cradle for your iron to sit in, so that you may not have to put it down on the bench while it is hot.

(iv) Magnifying glass

If you are doing work on PCBs (printed circuit boards), you may need to get a magnifying glass. This will help you see the fine tracks on the PCB. Soldering of small chip resistors are pretty difficult without a magnifying glass.

(v) Solder suckers

These are spring loaded devices that suck the melted solder out of the joint. They help in making a smooth and perfect solder joint.

Soldering Techniques

In order to have a perfectly soldered joint, it is necessary to learn soldering techniques. Soldering technique involves the following four steps:

Step 1: Preparation

Whatever you want to solder, it is necessary to make preliminary preparations such as opening the parts of a connector, cleaning the contact surface area with use of proper tools. Select a proper size tip. An oversized solder tip may even spoil the connector. Take necessary precautions to put the sleeves first on to the cable side before soldering so that these can be fitted to the connector after soldering.

Step 2: Stripping

Strip the insulation of the cable wire up to the length required for making the connection by use of a wire stripper or a knife. Be sure to cut the insulation up to the exact length required otherwise, the connections may create problems. The inner conductor may touch the outer while bending or pulling the cables.

Step 3: Tinning

After stripping the wires to a required length, you should 'tin' the wires and connector pins before you attempt to solder them. This process of tinning coats or fills the wires or connector-contacts with solder so that you can easily make a quick and smooth solder joint. Be careful not to overheat the wire, otherwise cable insulation will also start to melt. The larger the copper core, the longer it will take to heat up enough to draw the solder in. Hence, it is advisable to use a higher temperature soldering iron for larger cables. Once you have tinned both parts, you are ready to solder them together.

Step 4: Soldering

Once you have tinned the stripped strands of the conductor, soldering job becomes easy. You simply need to place your soldering iron onto the contact to melt the solder. When the solder in the contact melts, slide the wire into the connector pin. Remove the iron and hold the wire still while the solder solidifies again. You will see the solder 'set' as it goes hard. This should all take around 1-3 seconds.

If it does not go so well, you may find that either the insulation has melted or the extra solder has made the joint thick enough which may not fit in the connector. If this is the case, you should de-solder the joint and start again. See Box 1 for important soldering tips.

Note It

Box1

Soldering Tips

1. Don't move the joint until the solder has cooled.
2. Keep your iron tip clean.
3. Use the proper type of iron and tip size.
4. Clean the contact-surfaces to be soldered thoroughly.
5. Don't overheat the contacts otherwise the cable insulation or the connector will get damaged.
6. Don't use excess solder metal; otherwise, spike or ball will be formed.

Stripping tools

In this subsection, you will learn the techniques of using stripping tools. Stripping means removing the insulation from the end of the cable and exposing the portion of copper core that is to be soldered. You can either use a wire stripper, side cutter, or a knife to do this. There are many types of wire strippers, and most of them work on the same line. Figure 20.2 (A) and Figure 20.2 (B) show the pictures of a wire stripper and a side cutter respectively.

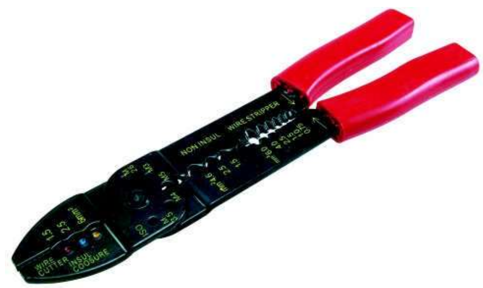

Figure 20.2 (A): *W*ire stripper

Figure 20.2 (B): Side cutter

As seen in Figure 20.2 (A), a wire stripper has got pre-set notches for different sizes of cables. Using a right notch will cut the insulation up to the preset depth only and not the strands of the conductors. Be sure to use a proper size stripper, otherwise, you are likely to cut some of the strands of the cable conductor. On the other hand, some people prefer to use a knife or side cutter as shown in Figure 20.2 (B). If you are using a side cutter, you have to be extra careful to cut only the insulation and not the copper conductor.

Crimping Tools

In some of the cases, we require to fix the lugs/connectors on the wires and cables, which may or may not require soldering. In such cases crimping tools are used. Crimping tools are the tools which are used to fix the lugs or connectors by putting a large pressure or a force on them. With a large force/pressure, the strands of a conductor and the sleeve of a lug are compressed to such an extent that the connection becomes a perfect joint.

For example, in case of larger size aluminium core power cables, use of crimping tools is essential as soldering of such cables is difficult. After stripping the insulation of a conductor, use the exact size of lug which just fits on it.

Use of crimping tool however, needs a caution. The lug size must not be more than the size of a bare conductor. The lug fitted on a cable must sit in the jaws of a crimping tool. The pressure applied by the lever-operated handle should be enough to compress the lug to a required pressure. It should not be too high to crush the lug or connector. For fixing lugs on power cables of higher sizes, usually a large lever-operated or even hydraulic crimping tools are used. However, in case of community radio stations, most of the connectors are fitted by soldered joints or by a light duty crimping tools wherever applicable.

Activity 12.2

To do this activity, you may need about 10 minutes to write down the answers in the space provided. This activity will help you in understanding the techniques of using the right tool for each specific job.

Question 1: Why are soldering guns not recommended for printed circuit boards and audio cables?

Question 2: What is the difference between a normal soldering iron and a temperature controlled soldering station?

Question 3: What does a 60/40 type of solder metal mean?

Question 4: What precaution must be observed while using a wire stripper?

Question 5: State a case where crimping tool is necessary.

Now let us move to the next topic, which is on the various types of connectors used in a CRS.

Types of Connectors

In the preceding sections of this unit, you learnt about the various tools and techniques used for fixing these connectors on cables. In this section and the subsections that follow, you will learn the following types of connectors that are used in a CRS setup:

- Audio connectors
- RF connectors

Let us discuss these connector types one by one.

Audio Connectors

A large variety of audio connectors are available in the market. Of these, the following 3 types of audio connectors are commonly used with audio equipment:

1. XLR connectors

XLR connectors are the most commonly used industry standard connectors. They are also called cannon conductors. The connectors are circular in design and available in 3 to 7-pin types. However, 3-pin XLR connectors are mostly used for feeding balanced mono or unbalanced stereo signals. Even the majority of professional microphones use the XLR connector.

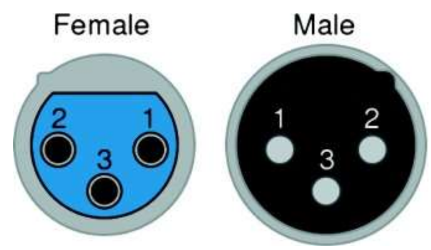

Figure 20.3 (A): 3-Pin XLR cable connectors *Figure20.3 (B): 3- Pin XLR chassis connectors*

As shown in Figure 20.3 (A) and (B), XLR connectors are available in male and female versions in both cable and chassis mounting designs, comprising a total of four styles. A special feature with XLR type of connector is that the ground connection is established before the signal lines are connected. The pin details of a 3-pin XLR connector are given below:

- Pin 1 is the earth (or shield)
- Pin 2 is the +ve (or 'hot')
- Pin 3 is the -ve (or 'cold)

These pin details are necessary while fixing the connector on an audio cable. Shield is used to connect Pin 1 to the earth wire, whereas red wire is connected to Pin 2 and white /black wire to Pin 1.

2. RCA connectors

The second type of connector, commonly used in audio circuits, is called RCA connector. Such types of connectors are mostly used in consumer-level audio and video systems including home stereos, videos, DVDs etc. Figure 20.4 shows different varieties of RCA connectors commonly used in commercial audio equipment.

Figure 20.4: *RCA connectors*

As you may see in Figure 20.4, the design of these connectors is a simple non-locking male/female connection. The male plug has a centre pin surrounded by a ring, whereas the female socket has a corresponding hole for the pin and a slightly smaller surrounding ring. The connection is made by simply pushing the plug into the socket. It is wired the same way as a mono jack: the centre pin is the +ve, and the outer ring is the -ve or shield.

The most common colour convention used in this type of connectors is as follows:

- Yellow: For video signals.
- Red: For audio (right channel).
- White or black: For audio (left channel).

A common problem with RCA connectors is that the male centre pin can easily touch the female shield ring when making the connection. Also, being a non-locking connector, the connection can fall apart which sometimes causes the centre pin to stay in contact with the ring or other objects. This results in a nasty hum or buzz.

3. TRS (Tip-Ring-Sleeve) connectors

The third type of commonly used audio connector is called a TRS or a phone connector. TRS (Tip-Ring-Sleeve) connectors are known by many different names, such as phone plug/jack, headphone jack or audio jack. This nomenclature is mostly based on their use. The term "jack" is particularly common for this type of connectors. Figure 20.5 shows various types of TRS connectors.

Figure 20.5: TRS (Phone) male connectors

Note the size of TRS connectors as shown in Figure 20.5. The length, thickness, shapes and the three parts (tip, ring and sleeve) of each connector may be noted. Even though they may differ in size, shape and length, they are functionally the same. These connectors are very common in audio equipment. The original 1/4" size was used in early telephone switchboards and has since become a standard connector for musical and other audio equipment. The jack is available in three sizes: 2.5mm (3/32"), 3.5mm (1/8") and 6.3mm (1/4") but the wiring for all of them is the same.

Connectors can be either mono (tip/sleeve) or stereo (tip/ring/sleeve). Some plugs are able to carry more signals for use with camcorders, laptops and other applications.

Stereo plugs can also be used to feed a single balanced audio signal instead of unbalanced stereo. Possible configurations are as given here:

	Unbalanced Mono	Balanced Mono	Stereo
Tip	Signal	Positive / Hot	Left channel
Ring	(Not connected)	Negative / Cold	Right channel
Sleeve	Ground / Return	Ground	Ground

20.3.1 RF (Radio Frequency) Connectors

In this sub-section, we will discuss various types of RF connectors used in a community radio station. The connectors that are used for carrying radio frequency (RF) signals are called RF connectors. The size and type of a RF connector typically depends on the size of cable to which it is connected, which further depends on the frequency of operation and the RF power passing through it. Various types of RF connectors commonly used are:

- N type (Neill) Connector
- EIA type (Electronic Industries Alliance)
- BNC type (Bayonet Neill Councelman)
- SMC (Sub Miniature Connector)
- TNC (Threaded Neill Connector)

For low power FM transmitters and antenna systems, usually 'N' type of connectors are used. For high power transmitters (above 1kW) and antenna systems, 'EIA' types of connectors are used. BNC types of connectors are common for radio and test equipment and upto exciter stage where the power is 10 to 30 watts. SMC and TNC types of connectors are commonly used for interconnecting the RF stages/modules within the transmitters.

All the above RF connectors are available in two types, namely chassis type and cable end type. Both types are further subdivided into two categories, namely male type and female type. Chassis types (usually female types) are used for mounting on panels of equipment. In field mostly, cable end connectors (usually male type) are used for connecting them on cable ends. Figure 20.6 (A) and 20.6 (B) show N (male) and BNC (male) types of RF connectors.

Figure 20.6 (A): N (male) type connector

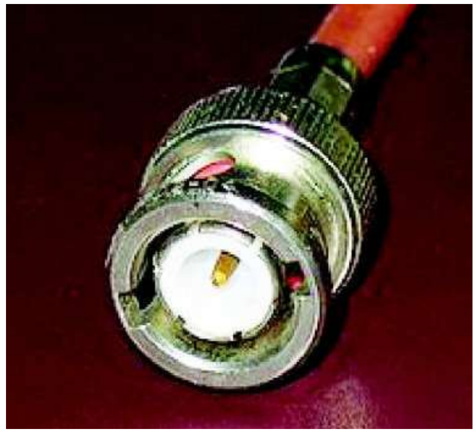

Figure 20.6 (B): BNC (male) type connector

As may be seen in Figure 20.6 (A) and (B), RF connectors are typically used with coaxial cables and are designed to maintain the shielding that the coaxial design offers. The ends of coaxial cables usually terminate with connectors. Coaxial connectors are designed to maintain a coaxial form across the connection.

Another important point about RF connectors is that they must have the same impedance as that of coaxial cables; otherwise there will be signal reflection and losses. In case of FM transmitters, almost all the RF connectors used are of 50 ohm impedance.

Activity 20.3

Imagine that you are working in a community radio station. From the details learnt so far in this unit, previous units, and from the various videos which you have seen, prepare a list of the different types of connectors used, starting from microphone to the antenna system and complete the table given below. Visualizing the audio chain in your mind may help you in answering this activity.

To do this activity, you may need about 20 minutes including writing down the answers briefly in the space provided.

This activity will help you in identifying and understanding the purpose of the use of different types of connectors required in a community radio station.

Sl. No.	Type of connector	Location	Purpose/Function

20.4 Types of Audio Cables

In this section, you will learn about different types of audio cables commonly used in community radio stations. We will also discuss special types of audio cables which are used as microphone cables.

There are two main types of audio cables that are most commonly used in interconnecting of different equipment in an audio chain:

1. *Single core shielded cable*

In a single core shielded cable, the single core is used as the +ve, or 'hot' and the shield is used as the -ve, or 'cold' line. The constructional details of a single-core shielded cable are illustrated in Figure 20.7.

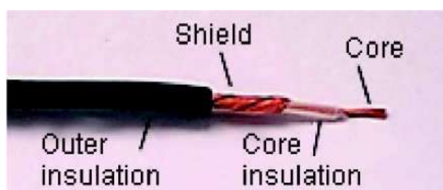

Figure 20.7: Single-core shielded cable

As shown in Figure 20.7, a single core cable consists of one inner conductor and a shield separated by an insulating material. The shield works as a second or a return conductor. The audio signal travels between the inner line called a hot line and an earth line. Unbalanced audio cables are commonly associated with the 1/4" TS and the RCA connectors and are used for unbalanced audio systems.

2. *Two core shielded cable*

A 2-core shielded cable has one core as the +ve line, and the other core as a –ve line. The shield is earthed. Figure 20.8 illustrates the constructional details of a 2-core shielded cable.

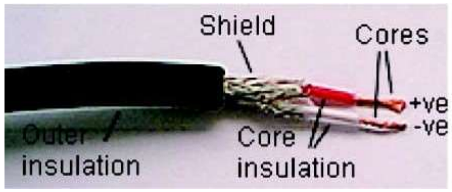

Figure 20.8: Two-core shielded cable

Figure 20.8 shows that a 2-core shielded cable has two inner conductors and a shield; a *hot* line (positive), *cold* line (negative) and an *earth*. This type of cable is used mostly for the balanced audio circuits. The audio signal is transmitted on both the hot and cold lines, but the voltage in the cold line is inverted (i.e. the polarity is changed). Hence, it is negative when the hot signal is positive. It minimizes unwanted noise and interference in audio cables. This type of connection is very important in sound recording and production because it allows for the use of long cables while reducing susceptibility to external noise.

For example, if the power amplifiers of a public address system are located at a distance from the mixing console, it is a normal practice to use balanced lines for the signal paths from the mixer to the amplifiers. Many other components, such as graphic equalizers and effects units, have balanced inputs and outputs.

Balanced circuits use 3-pin connectors, usually the XLR or TRS phone connector. XLR connectors, for instance, are usually used with microphones whereas TRS jack plugs are usually used for mixer inputs and outputs.

Microphone cables

Microphone cables connect microphones to mixers (desk, consoles etc). Most professional mixers' microphone inputs are designed with "balanced" circuits to help decrease or eliminate noise and unwanted radio frequency interference (RFI).

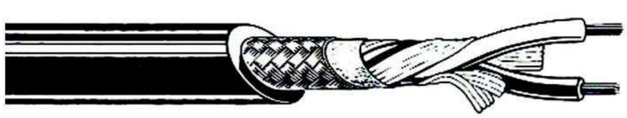

Figure 20.9: Twisted pair microphone cable

As seen in Figure 20.9, a microphone cable consists of a twisted pair of copper conductors (typically 6 mm in diameter). These conductors are covered with one of three types of shielding: braided, spiral, and foil shielding. Braided shield is best for microphone cables whereas spiral shield is a little more flexible and less expensive than braided. Foil shields are unreliable in cables and are designed for portable use.

Good shielding in microphone cables help in preventing electromagnetic interference. Microphone cables are used to carry stereo audio signal of frequency range 20 Hz to 20 kHz having 50 ohm impedance from the microphone to mixing console.

Choosing the right microphone cable

Most professional Low-Z (low impedance) microphone outputs can easily be run up to 500 feet. However, Hi-Z (high impedance) microphones have the same roll off problems that guitar cables have and their lengths should be limited to 20 feet or less to avoid high frequency attenuation.

Microphone cables come in a wide variety of diameters. Nature enthusiasts need for their sound recordings small cables that will roll up into the compartment. To conserve space, Tape recorders use microphone cables about the diameter of a normal pencil (1/4"). Balanced mic cables are quieter than unbalanced mic cables because 1/2 of the signal travels on one of the two conductors and they tend to cancel out extraneous signals that jump on both conductors.

Activity 20.4

To do this activity, you may need about 15 minutes including writing down the answers in the space provided. This activity will help you in identifying different types of audio cables in respect of their constructional and functional details.

Question 1: What is the difference between the unbalanced and balanced audio systems?

Question 2: Why is working with balanced audio system better than unbalanced one?

Question 3: Why is shielding necessary in microphone cables?

20.5 Let Us Sum Up

In this unit on 'Good Engineering Practices', you have learnt that:

- For any project, certain good engineering practices are necessary to get the best quality and performance out of the Equipment installed in that project. Doing the right job in the right manner is a skill that comes only by practice.

- A variety of tools and equipment such as soldering iron, wire stripper and an audio analyser are used for installation, wiring and testing the performance of the Equipment. The video demonstration on use of tools and measuring equipment will further help you in developing the skill and confidence.
- Soldering, stripping and crimping are the techniques which are learnt mainly by practice. You have also learnt and understood certain precautions which are to be taken while soldering and fixing connectors. A little overheating can melt the insulation of the cable connector or the track of the printed circuit board thereby damaging the connector or the PCB.
- Various types of audio and RF connectors are used in the CRS. You have seen that selection and use of the right type of connector is very important.
- Various types of audio cables are used for the interconnection of components of the transmission chain. Both balanced and unbalanced cables are used depending upon the requirements of the circuit. Shielded audio cables help in minimizing the interference in audio signals due to RF pick up, noise and hum.

20.6 Model Answers to Activities

Activity 20.2

1: Soldering guns don't have temperature control and can get too hot easily. This can result in damage to circuit boards, melting of cable insulation, and can even cause damage to connectors.

2: In normal soldering iron, there is no method to control the temperature, whereas in the temperature controlled soldering station, the temperature can be preset to any desired temperature.

3: It means that soldering metal consists of 60% of tin and 40% of lead.

4: A proper size wire stripper must be used, which may only cut the insulation and not the strands of the conductor.

5: In case of power supply cables having aluminium conductor where soldering is not practically possible, lugs are fitted by use of proper size crimping tools.

Activity 20.4

1: In unbalanced systems the audio signal flows between one line and the earth, whereas in balanced circuits, the signal flows between two lines independent of the earth wire.

2: Working with balanced systems is better as they are less sensitive to noise and interference.

3: Since the output of a microphone is of very low level, shielding of the cable protects the signal from interference due to noise and hum.

20.7 Additional Readings

- *Electrical connector – Wikipedia, the free encyclopedia*. (n.d.). Retrieved March 3, 2014, from http://en.wikipedia.org/wiki/Electrical_connector
- *Florida Sound Engineering Co., Inc. | Welcome! | Home Page*. (n.d.). Retrieved from http://www.floridasound.com
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UNIT 13

Studio Equipment: Preventive and Corrective Maintenance

Structure

- Introduction
- Learning Outcomes
- Cleanliness and Dust-free Environment
- Dressing of Cabling
- Earthing Connection
- Handling of Microphones
- Ventilation and Fresh Air
- Preventive Maintenance of Digital Audio Workstations
- Let Us Sum Up
- Model Answers to Activities
- Additional Reading

Introduction

In Unit 11, you learnt about the functioning of equipment used in audio chain of a CRS studio. In Unit 20, you learnt about the good engineering practices used in installation, wiring and testing of studio equipment. In this unit, you will learn about the preventive and corrective maintenance aspects of studio equipment, which are essential to keep them in perfect working condition. This helps in timely detection and forewarning about the likelihood of occurrence of any major fault or malfunctioning of studio equipment which may result in service breakdown. Preventive maintenance implies regular cleaning, checking, testing and measurement of equipment to keep them in working order as per accepted norms and standards, which may help in preventing a major breakdown. Corrective maintenance can be defined as the maintenance which is required to bring the equipment back to the working order once it has failed or worn out. A regular preventive maintenance schedule such as weekly, monthly, quarterly etc. helps in achieving this purpose. In this unit, we will discuss the following issues:

- Importance of cleanliness and dust-free environment in studios
- Dressing of cabling
- Earthing of equipment
- Handling of microphones
- Ventilation and fresh air in the studio
- Maintenance of Digital Audio Work Stations (DAWs)

You may require about 6 hours of study to learn this unit including answering the questions given in the Activities.

The glossary given at the end of the module will help you in understanding the content of this unit.

13.2 Learning Outcomes

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

- describe the importance of cleanliness and dust-free environment in the studios.
- explain the purpose of dressing of cables.
- underline the importance of earthing of equipment and check earth connectivity.
- describe the process involved in handling of microphones.

- discuss the issues related to the maintenance of AC plants and ventilation in the studios.
- describe the maintenance of DAWs.

Let us begin with cleanliness and dust-free environment.

Cleanliness and Dust-free Environment

We all know cleanliness is next to godliness. The studio environment is to be kept clean and dust-free because dust is one of the main enemies of broadcast media (including CDs/tapes) and playback/recording equipment. Cleanliness is absolutely necessary. No eating, drinking or smoking is to be permitted inside the studio premises. All rubbish must be thrown in the dustbins. Dust can damage console faders, switches, CD Players, DAWs and other switch gear. Dust is the enemy of electrical contacts and faders. Dust and dirt prevents the intimate contact of replay heads to media which is essential for the accurate retrieval of information. Dust can cause “head crashes” of computer hard discs which may lead to irretrievable loss of data. The effective prevention of dust and other kinds of dirt is, therefore, an indispensable measure for a broadcast environment. For this purpose, routine dusting with anti-static dusting should be done daily. A head cleaner or alcohol can be used for cleaning of fader contacts and switches. A schedule should be prepared for keeping the equipment clean and dust-free. For this purpose, the following guidelines may be followed:

- Use a light duty blower for suction of dust from equipment and racks.
- Clean all rack-mounted equipment in control room and studio.
- Clean all the tag blocks, patch cords and patch panels.
- Clean recording and playback equipment and all PCBs.

Activity 13.1

To do this activity, you may need about 10 minutes including writing down the answers in the space provided. This activity will help you in understanding the importance of cleanliness and dust-free environment in respect of preventive maintenance.

Question 1: Why is a clean and dust-free environment important for broadcast equipment?

Question 2: What steps should be followed for keeping the equipment dust-free?

Having learnt the importance of cleanliness and dust-free environment, we will now discuss dressing of cabling.

Dressing of Cabling

In this section, you will learn the significance of dressing of cables. Dressing of cables plays a very important role in preventive maintenance. Cable dressing means properly aligning and positioning the cables in a neat and orderly manner in the studio trenches and audio racks. Proper cable termination practices should be followed for the complete and accurate transfer of both analogue and digital information signals. All the cables should be properly identified and labelled. Extra length of cables should be avoided and cables should not be too long. However, enough slack in length should be left in case it needs to be re-terminated or rerouted for any reason. All the signal cables, except power cables, should be laid parallel and orderly bunched. Tie wraps or hook and loop straps should be used to secure the cables and they should be evenly spaced throughout the dressed length. Cable dressing ensures that cables used are neatly arranged and easy to trace, when required. All the audio wiring and power wiring should be laid separately; otherwise there may be hum generation. For this purpose, cable documentation should be done to keep track of types of cables laid, their destination, numbering for identification, the path followed and termination details. The following guidelines may be followed for cabling:

- Use separate paths for power and signal/audio cables so as to reduce the electromagnetic induction effect (EMI).
- Wherever the power and audio cables must cross, these should be laid at right angles to neutralize EMI effect.
- Cables should be properly tagged, strapped, identified and laid in trenches/racks.
- All signal/audio cables should enter from the left side of the rack at the back and power cables from the right side of the rack.
- Use contact cleaner for cleaning the contacts to ensure good contact.

Activity 13.2

To do this activity, you may need about 10 minutes including writing down the answers in the space provided. This activity will help you in understanding the necessity of dressing of cables.

Question 1: Why is dressing of cables needed?

Question 2: What guidelines should be followed while laying power and audio cables in a studio?

Now let us proceed to discuss the next section which is on earthing connections.

Earthing Connections

In this section, you will learn about the significance of earthing connection in a studio set up.

As a safety measure, earthing is essential and mandatory. Earthing connects the body of the equipment to the earth electrode. It is a physical connection between the exposed metallic parts of an electrical equipment/appliance and the earth, which is known to have zero potential. Proper earthing provides an alternative and easy path for leakage or faulty current to flow. It ensures that any exposed conductive part of the appliance does not reach a dangerous level of potential or voltage that endangers the user's life. For any electrical system to be safe, proper insulation and earthing must be provided for protection and safety of staff and equipment operations. Earthing is done to provide a conducting path to the ground for the fault current which may flow due to a short circuit. The body of the equipment is earthed so that fault current flows to the ground and the operator remains safe; otherwise the operator may get a fatal shock. For this purpose, a pit is formed by digging the earth and putting some charcoal and salt at the bottom and placing a metallic plate with an electrode over it, to which earthing copper or aluminium metal strip is connected. As a precautionary measure, there should be at least two independent earthing connection paths for safety of equipment and personnel. A proper earthing system should have least electrical resistance. Lower the earth resistance, better it is for the safety of equipment and personnel. The earth resistance can be measured using an instrument called 'Megger'.

Failure of earthing can result in:

- Danger to equipment and operating staff due to short circuit and malfunctioning of electrical switch gear.
- RF pick up from nearby radio frequency sources leading to noise and distortion in sound recording and broadcast.
- Following steps can ensure prevention of faults due to bad earthing:
- Regular inspection of earthing strip or cables.
- Watering of earth pits regularly to keep earth resistance within specified limits.
- Regular checking of continuity of earthing connection from equipment to earth pits.
- Measuring earth resistance every quarterly.

Activity 13.

To do this activity, you may need about 10 minutes including writing down the answers in the space provided. This activity will help you in understanding the necessity of earthing in a studio setup.

Question 1: What is the importance of equipment earthing?

Question 2: What is the importance of checking continuity between the equipment and earth pits?

Having learnt the importance of earthing in studios, we will now discuss the next topic, which is on handling of microphones.

Handling of Microphones

In this section, you will learn about the practices to be adopted in handling of microphones. Microphones are costly equipment and extremely fragile. Microphones are very stable over long periods of time, provided that they are handled properly. The components of the microphone are fragile and can get damaged by misuse. The following practices should be adopted for proper functioning of microphones:

- Keep it clean. The microphone screen can get dusty. It should be cleaned regularly.
- All the microphones have a diaphragm in one form or the other. The diaphragm is exposed to the air. So it should be protected and handled carefully.
- High-sound pressure levels (SPL) can damage the diaphragm.
- Most of the microphones are made of metal, which can rust easily. To prevent rusting, a bag of silicone gel should be kept inside the microphone case.
- The ribbon microphones should be stored vertically because the ribbon inside the microphone is installed vertically. This prevents any slack in the ribbon if stored horizontally.
- Keep condenser microphones out of direct sunlight because high heat can damage the diaphragm.
- Phantom power, i.e., + 48V should be switched off or disabled before plugging in/out ribbon microphone.
- Microphones should not be installed too close to a loud/blaring instrument as this will damage the diaphragm.

Activity 13.4

To do this activity, you may need about 10 minutes including writing down the answers in the space provided. This activity will help you in understanding the precautions and instructions required while handling the microphones.

Question 1: What precautions should be followed while handling various types of microphones?

Ventilation and Fresh Air

In this section, you will learn that by maintaining the ventilation equipment properly, occurrence of a number of faults in studio equipment can be averted. An efficient ventilation system and its proper maintenance are essential for the broadcast studio environment. For this purpose, an air conditioning system is provided for cooling and dehumidification in the studios. This involves control of temperature, humidity, ventilation and movement of fresh air. Ventilation is a process by which stale air is removed and fresh air is supplied to the studios. Regular preventative maintenance of AC plants is, therefore, essential to ensure trouble-free operation of the studio equipment. It helps in improving the quality of programme production as well. Pre-season maintenance is also important. It can help to avoid a system failure in severe hot or cold conditions. While the system is in operation, the following monitoring checks should be carried out regularly as part of routine maintenance:

- Measure indoor dry and wet bulb temperature.
- Measure and adjust air flow.
- Check vent system for proper operation.
- Listen for any abnormal noise.
- Inspect, clean and change air filters.
- Inspect and clean blower assembly.
- Inspect for any gas leakage.
- Check thermostat settings.
- Check electrical connections.
- Lubricate moving parts.

Activity 13.5

To do this activity, you may need about 10 minutes including writing down the answers in about 50 words in the space provided. This activity will help you in understanding the techniques of maintenance of ventilation system including air-conditioning of studios.

Question 1: What are the monitoring checks to be followed for maintenance of AC plants and ventilation system at the studios?

Preventive Maintenance of Digital Audio Workstations

In the preceding section you learnt about the technique of maintaining ventilation system in the studios. In this section, you will learn the steps to be taken for maintaining the digital audio workstations.

As mentioned in earlier units, Digital Audio Workstations (DAWs) are used for recording and playback in broadcast studios. It is an high-end computer having large RAM and hard disk capacity with professional audio cards and professional audio software for programme production and playback. Preventive maintenance of DAWs is essential for reliable functioning. It helps to detect serious problems, prevent system crashes and reduce equipment down time. A regular periodic maintenance schedule should be drawn up to take care of the following issues involved in maintenance of DAWs:

- Scan the memory and hard disk of the DAW regularly to protect the system from any virus. For this, use a genuine professional antivirus software.
- Always keep the backup of important data and program files.
- Clean CPU fan, CPU, keyboard, mouse etc. to make the system dust-free.
- De-fragment the hard disk.
- Run system performance diagnostics to check the health status of the DAW.
- Always maintain system software and networking updates.
- Do head cleaning of CD drives periodically.
- Do cleaning of cards, subassembly units and other components at regular intervals.
- Check and maintain UPS and its batteries. Always make a habit to put the load on battery to ensure that the system is capable of working on battery backup during mains failure. Overcharges and deep discharges reduce the life of batteries.

Activity 13.6

To do this activity, you may need about 10 minutes including writing down the answers in the space provided. This activity will help you in understanding the techniques of doing the preventive maintenance of DAWs.

Question 1: How can you prevent virus attack to your digital audio workstation?

Question 2: Why should you take a regular backup of data?

Let Us Sum Up

In this unit, we have discussed preventive and corrective maintenance of studio equipment and system. In this process, you have learnt that:

- A dust-free and clean environment is very much essential in the broadcast studios. It helps in curtailing the occurrence of faults due to accumulation of dust.
- Dressing of cables with proper identification and termination helps in ease of cable tracing in case of cable faults.
- Maintenance of ventilation equipment and uninterrupted connectivity reduces the breaks in transmissions.
- Earth conductivity and continuity of earth connections to studio equipment and racks is important for protection of equipment and staff in case of short circuit and lightning. Earth resistance should be very low for this purpose.
- Microphones are delicate equipment. These are essential for recording and broadcasting of programmes and should be handled with care to prevent any damage.
- Regular preventive maintenance of AC plants and other ventilation equipment is to be carried out. This helps in controlling rise in temperature, humidity and improving ventilation in studios.
- Nowadays high-end computers, called Digital Audio Workstations, are used for recording and playback of programmes in broadcast studios. Regular maintenance of DAWs should be done to prevent their failure and loss of data stored on hard disks. Use of properly rated UPS systems and maintenance of batteries help in curtailing faults due to failure or variations in mains supply.

Model Answers to Activities

Model answers to questions given in Activities 21.1 to 21.6.

Activity 13.1:

1: Dust can damage console faders, disk heads of CD players and audio workstations.

2: Do not allow eating and drinking in the studios. Do routine dusting daily with anti-static duster. Use a head cleaner for cleaning of fader contacts and switches.

Activity 13.2:

1: All the cables should be properly identified and labelled as this helps in ease of cable tracing in case of cable faults and relaying.

2: All the audio and power wiring should be laid separately otherwise there may be hum generation due to electromagnetic induction.

Activity 13.3:

1: Proper earthing provides an alternative and easy path for leakage or faulty current to flow. This prevents danger to equipment and operating staff due to short circuit and mal-functioning of electrical switch gear.

2: To protect equipment and staff in case of short circuit and lightening, it is essential to regularly check continuity between earth connections and studio equipment.

Activity 13.4:

1: Microphones are delicate equipment. These are essential for recording and broadcasting of programmes and should be handled with care to prevent any damage.

Activity 13.5:

1: The following checks will be done:

- Measure indoor dry and wet bulb temperature.
- Measure and adjust air flow.
- Check vent system for proper operation.

- Inspect, clean and change air filters.
- Inspect and clean blower assembly.
- Check for any gas leakage.
- Check thermostat settings.
- Check electrical connections.

Activity 13.6:

1: Install anauthenticated anti-virus software and regularly update for latest virus definition database.

2: Regular data backup should be undertaken to prevent any loss of data due to hard disk failure.

Additional Readings

- Gookin, D. (2009). *Troubleshooting & maintaining your PC all-in-one for dummies: [6 books in one ; hardware, software, laptops, internet, networking, maintenance]*. Hoboken [NJ: Wiley.
- McCartney, T. (2003). *Recording studio technology, maintenance, and repairs*. New York: McGraw-Hill.

UNIT 14

Content Distribution: Alternative Mechanisms

Structure

- ✓ Introduction
- ✓ Learning Outcomes
- ✓ Internet
  - Podcasting or audio blogging
  - Streaming
  - Social networking sites
  - Content sharing sites
- ✓ Wireless Mesh Networking (WiMesh)
- ✓ Mobile Telephony
- ✓ Let Us Sum Up
- ✓ Model Answers to Activities
- ✓ Additional Readings

Introduction

In unit 14, you learnt about the open source software used for recording, editing and playback of programmes in production studios. In unit 15, you learnt about the telephony application software for phone-in, SMS, IVRS etc. In Units 16 to 19, you learnt about the sound recording and editing, mixing and mastering, file formats and compression, and storing and retrieval of programmes in studios only. However, in this unit, you will learn about distribution and sharing of content using alternate platforms.

In an era of convergence of media, it may feel very restrictive to use only FM medium to reach to your audience. As we know the FM radio is bound by the geographical boundaries based on the transmission power. In the case of community radio in India we are bound by 50 Watts of power that roughly translate to a coverage radius of about 10 km. On the other hand, the technology is surpassing the geographical limitations. An email sent from a desktop can reach anywhere in the world in a matter of seconds, whereas a voice on FM radio may not reach beyond 10 km. As the technology advances you may realize that there are many ways to reach out to your target audience. Hence it would be useful for a community radio technologist to understand alternative mechanisms to broadcast in addition to the FM radio.

Apart from understanding different technologies for communication, it is critically important to understand how we can forge a convergence of media. In other words, we need to understand how different technologies can feed into each other and make use of strategic advantages of available mediums like the Internet, mobile etc.

In addition to the use of alternative mediums, there is also a vast potential to sharing programmes, information and skills with other radio stations. With the growth of the Internet, sharing has become easier and there are quite a few ways of sharing content. In this Unit, you will learn about how to broadcast on Internet, share content and even create community owned communication infrastructure based on wireless technology. Therefore, in the process of learning of this unit, we will discuss the following topics one by one as given below:

- Internet
- Wireless Mesh Networking (WiMesh)
- Mobile Telephony

To complete this unit, you may need about 6 hours of study. The glossary at the end of the module will help you in understanding the contents of this unit.

Learning Outcomes

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

- describe various alternative platforms available for publishing and broadcasting.
- discuss strategic advantages and disadvantages of alternative platforms.
- analyse various methods of convergence of tools.
- list out and describe various content sharing methods.
- describe the fundamentals of wireless networking.
- discuss the use of mobile telephony as a broadcasting mechanism.

Let's now begin with the Internet.

Internet

All of you are more than familiar with the Internet. However, what we are going to learn here is the use of Internet as an alternative publishing medium or a medium to share content across the world and other community radio stations. It may be noted here that though Internet is a very powerful tool, its reach in the rural parts of India is still not upto the mark. The access of Internet is far less than what is desirable. Hence in the present scenario, it cannot be seen as an alternative to terrestrial radio but can be used as a supplement. In this section and the sub-sections that follow, we will discuss some of the more popular and technically advanced methods of sharing content over internet which include:

- Podcasting
- Streaming
- Social networking
- Content sharing sites

Podcasting or Audio Blogging

Podcasting is a method of publishing in which an audio or video file is stored on a web server and listeners can listen to the file either through a browser or by virtue of being a podcast client. This is a good example of push technology which works on publisher and subscriber model. There are many podcast clients available on the Internet for listeners as well as publishers. In addition, there are plenty of sites that allow you to create a podcast account. Using such podcast account, you can publish your audio or digital content on the Internet. You can even embed a podcast link in your website so that a listener can listen to your audio on your website rather than going on to the third website.

The term podcast is a combination of two words Ipod — which is a popular audio device by Apple Inc — and Broadcast. Many users of the technology have opposed the term podcast as they say it gives undue credit to Apple which had very little to do with the development of the technology. Those who are opposed to the term podcast refer it as Audio Blogging.

The Audio Blogging or Podcast uses the unicast protocol. Unicast protocols send a separate copy of the media stream from the server to each recipient. This model, however, does not scale well when many users want to hear the same audio programme concurrently.

Podcast should not be confused with webcast which is a concept that refers to streaming technology. There is a basic difference between Podcast and Streaming. We will learn about it in subsequent paragraphs.

Podcast require a podcast server which could be similar to a web server. However, a specific programme should be running for any web server to act as a podcast server as well. It is possible to run a podcast server out of your own desktop computer but it may not be a very feasible option since you have to keep your desktop machine power on all the time. Also there are security risks if the protection provided along with podcast is not up to date.

The other alternative is to use one out of many hosted solutions. Many groups and companies provide hosted solution for podcast. Some of them charge monthly or yearly fees, whereas some are completely free of cost. You may choose one of them to suit your requirement.

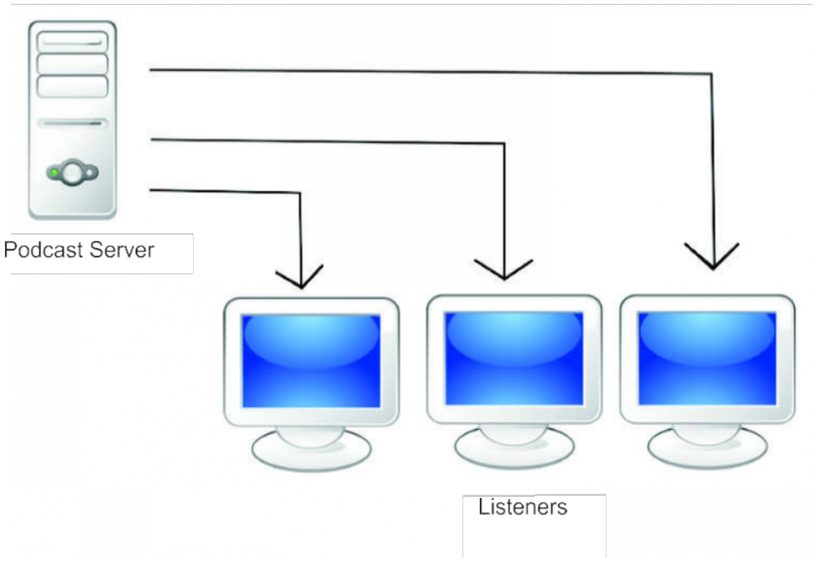

Figure 22.1: Unicast network

Figure 22.1 shows a typical example of Unicast Network. As you can see in the figure, the server is putting out different stream of data for each listener.

Streaming

Streaming is the closest counterpart on Internet to FM broadcast. If you are doing a live broadcast on your FM based community radio, the listeners at the other end can receive your voice almost simultaneously. They can listen as you speak. In other words, they are listening on real-time basis. Similarly, in streaming over Internet, the audio content is delivered to the listeners on a real-time basis.

Unlike audio blogging, the streaming content does not need to be placed on the server but the listener can stream directly out of the publisher's desktop. Most audio streams use multicast protocol. This protocol sends a single stream from the server to all the connected recipients. This protocol was developed to reduce the server/network loads resulting from duplicate data streams that occur when many recipients receive unicast content streams independently.

Use of multicast protocol has certain limitations since the recipient does not have an audio-on-demand facility. For example, if an audio programme starts at 12 noon and a listener joins the stream at 12.30, he or she would lose half an hour of programme, just like in an FM radio. Also you cannot play back a content that is streaming. However, there are ways to mitigate this limitation by deploying caching servers or buffered media players.

Streaming requires a streaming server and a streaming client. Again, you have a choice of hosting your own server. What you need to consider here is that what kind of Internet bandwidth is available to you. Remember that a streaming server hosted at your own location would consume a lot of bandwidth and you may find it difficult to carry out your normal Internet work like checking emails or surfing a website. Also a low bandwidth will restrict the number of simultaneous listeners you can have. Needless to say, lower the bandwidth, lower the number of listeners. However, you can increase the number of listeners on a lower bandwidth if you are willing to compromise with audio quality. Alternatively, you may consider a hosted solution. There are many companies around the world which provide hosted solution. However, practically none of them are free. They charge nominal fees for a year or month.

There are several types of streaming servers. The prominent among them are Windows Media Services, developed by Microsoft and Darwin Streaming server developed by Apple Inc. Both these servers are proprietary and they do not come for free. ShoutCast is another streaming server developed by Nullsoft, a company that is known for its popular Winamp media player. ShoutCast is available free of cost but it is not an open source application. The Nullsoft has kept it as a proprietary solution. One of the most popular streaming server is called IceCast. The IceCast is the most versatile and is free of cost and also available in open source domain.

The server applications mentioned in the previous paragraph are to be installed on a server. Also, you will need a desktop software that will encode your audio and stream up to the server which in turn will stream down to the listeners.

It is noteworthy that, with growing number of smart phones in the market, the streaming audio has become one of the critical ways of distributing content. Any mobile phone that is running Android, Blackberry, Windows or IOX can receive a stream generated by servers.

Following is the list of clients available for Icecast server. Note that this is just an indicative list. You will find many more sophisticated and versatile clients across various operating systems. GRINS, a radio automation software promoted by Gram Vaani, also allows simultaneous streaming to Icecast server.

Source clients

The source clients that are known to work with Icecast 2 are given in Table 22.1 and Media players that support Icecast 2 are shown in Table 22.2.

Table 22.1: Source clients

Application	Platform	Download Link
IceS	Linux/Unix	http://www.icecast.org/ices.php
Oddcast/Edcast	Windows	Formerly at http://www.oddsock.org/tools
Edcast reborn	Windows	http://code.google.com/p/edcast-reborn/
Muse	Linux/Unix	http://muse.dyne.org
Darkice	Linux/Unix	http://darkice.sourceforge.net/
SAM2	Windows	http://www.spacialaudio.com
ezstream	Windows	http://www.icecast.org/ezstream.php
Nicecast	Mac OSX	http://www.rogueamoeba.com/nicecast/
IceGenerator	Linux/Unix	http://sourceforge.net/projects/icegenerator
Orban Opticodec-PC	Windows	http://www.orban.com/
freej	Linux/Unix	http://freej.org/
Traktor DJ Studio 3	MacOS X, Windows	Native Instruments
Savonet/Liquidsoap	Linux/Unix, Windows	http://savonet.sourceforge.net/
DeeFuzzer	Linux/Unix	http://pypi.python.org/pypi/DeeFuzzer/

RoarAudio	Linux/Unix, Windows	http://roaraudio.keep-cool.org/ roaraudio.html
RoarAudio PlayList Daemon	Linux/Unix, Windows	http://roaraudio.keep-cool.org/rpld.html
butt - broadcast using this tool	Linux/Unix, Mac OSX, Windows	http://butt.sourceforge.net/
Mixxx	Linux/Unix, Mac OSX, Windows	http://mixxx.org/
iCast	iOS	http://icast.anthonymyatt.net
MPD - Music Daemon	Linux/Unix, Mac OSX	http://mpd.wikia.com/wiki/Music PlayerDaemon_Wiki
KRADradio	iOS	Linux/Unixhttp://kradradio.com/

Table 14.1 gives a list of source clients using application with platform. Download link of each application is also shown along with them for easy reference.

Table 14.2: Media players that support Icecast streaming

Application	Platform	Download Link
foobar2000 (mp3 + ogg vorbis)	Windows	http://www.foobar2000.org
winamp 2.x, 5.x (Not 3.x) (mp3 + ogg vorbis)	Windows	http://www.winamp.com
XMMS(mp3 + ogg vorbis)	Linux/Unix	http://www.xmms.org
Zinf(mp3 + ogg vorbis)	Linux/Unix, Windows	http://zinf.sourceforge.net
MPlayer	Linux/Unix, Windows, Mac OSX	http://www.mplayerhq.hu
Xine	Linux/Unix	http://www.xine-project.org/home
VLC	Linux/Unix, Windows, Mac OSX	http://www.videolan.org

Figure 14.2 is an example of Multicast Network. As you can see in the figure, the media server is putting out one stream and the same stream is being tapped by multiple users

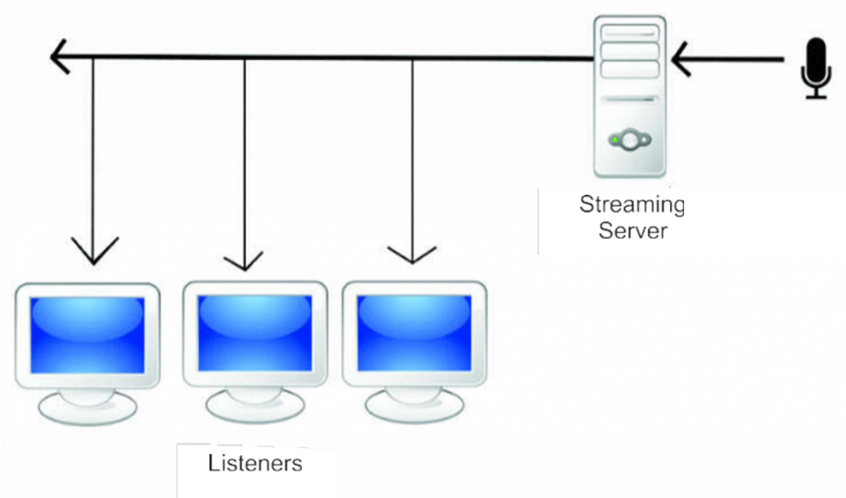

Figure 22.2: Multicast network

Activity 14.1

Search for an audio blogging portal and note one blog on the community media course you have undertaken. You can search for free blogging sites on your search engine and choose the one you like. Also write the name of the blog writer and the website searched.

To do this activity, you may need about 15 minutes including writing down the answer in the space provided.

This activity will help you in understanding and using the alternative platforms available for content distribution.

Social Networking Sites

Social networking sites have become very popular in the past few years. Sites like Facebook, Youtube, LinkedIn or Myspace have millions of users who can turn out to be your listeners if you know how to use the social networking sites effectively. Though there is no single formula to be effective on these social networking sites which are very dynamic and evolving, some suggestions are given below to help you in making use of these sites.

- Popularise your radio station by having a dedicated page on social networking sites like Facebook, Twitter, Pinterest or MySpace. You can even publish your schedule on these sites and write about the programmes your station is broadcasting.

- Put a direct link to your podcast or live stream (you will learn about live stream later on in this unit).
- Use these sites as a feedback mechanism if your radio station is broadcasting in the areas where your listeners have Internet access.
- You can even upload audio or video giving glimpse of your programme on these sites.

Content Sharing Sites

Content sharing in a multi-lingual and diverse country like India is a very complex concept where not only the language but even the dialect matters. Even otherwise the core concept of the community radio is community-specific programming which limits the use of content sharing among the community radio stations.

Sharing of content goes a long way in enhancing the scope of learning. It is not only about sharing content but it is also about sharing ideas, concepts, formats and style. Internet plays an important role in exchange of programmes among the community radio stations.

There has been a slow start to the idea of content sharing in India. The first portal that came up was EK duniya anEK awaz. Available at http://www.edaa.in, it is now publishing user generated content in 26 languages. You can upload the content for free as all the content uploaded on the site is available for free under creative common license. Yet another initiative called Manch has been launched in 2013 for encouraging content sharing and collaboration among community media practitioners. The site is available at http://manch.net.in.

Activity 14.2

Write a blog on the current important event of the day. Publish your blog on a social networking site and see your popularity by seeing the comments of friends and users.

To do this activity, you may need about 15 minutes including writing down the blog in about 50 words in the space provided.

This activity will help you in understanding the method of sharing the content of the blog written by you with your friends and listeners.

Wireless Mesh Networking (WiMesh)

Many believe that Wi-Fi Mesh Networking, which is also known as WiMesh, can make our world more connected than it is today. This emerging technology can seamlessly connect vast geographical area in the most humane way. Some experts also assert that WiMesh is more a sociological project than a technical one.

A wireless mesh network, in very simple terms, can be described as a network of hundreds or even thousands of wireless devices which are capable of transmitting as well as receiving RF signals. The most commonly used device in a mesh network is the commonly available Wi-Fi routers, which are used in offices or homes for distributing Internet connectivity. These routers in a mesh network are called nodes that talk to each other. These nodes not only transmit data generated by its owner but also help passage of data generated by other nodes. Nodes use the common Wi-Fi standards known as 802.11a, b and g to communicate wirelessly with users, and, more importantly, with each other. Radio signals generated and received by the Wi-Fi devices are at a higher frequency of 2.4 GHz and 5.8 GHz as against 88 MHz to 108 MHz in FM.

There are multiple uses of such networks. Imagine your office or a home Local Area Network that is expandable to your entire village or even city. In that case you can easily run a community radio over Internet or you can even run a community TV. You can have a telephone system that does not cost you when you receive or make a call. You can easily give Internet access to most remote areas or run tele-education programme in a school or even run a tele-medicine programme in your village. The usage of such a network is of immense value but the cost is negligible.

The most important advantage of such network is the fact that you do not require a license from the Government since frequency spectrum of 2.4 GHZ and 5.8 GHz have been de-licensed almost in all the countries including India.

Figure 22.3 shows how a small village can form a wireless network. All the houses in a village are wirelessly linked to each other. Each house is having at least two wireless access points. Note that there is also Internet gateway. Similarly, the second image in Figure 22.3 shows how many villages which have their own wireless network can be linked to each other.

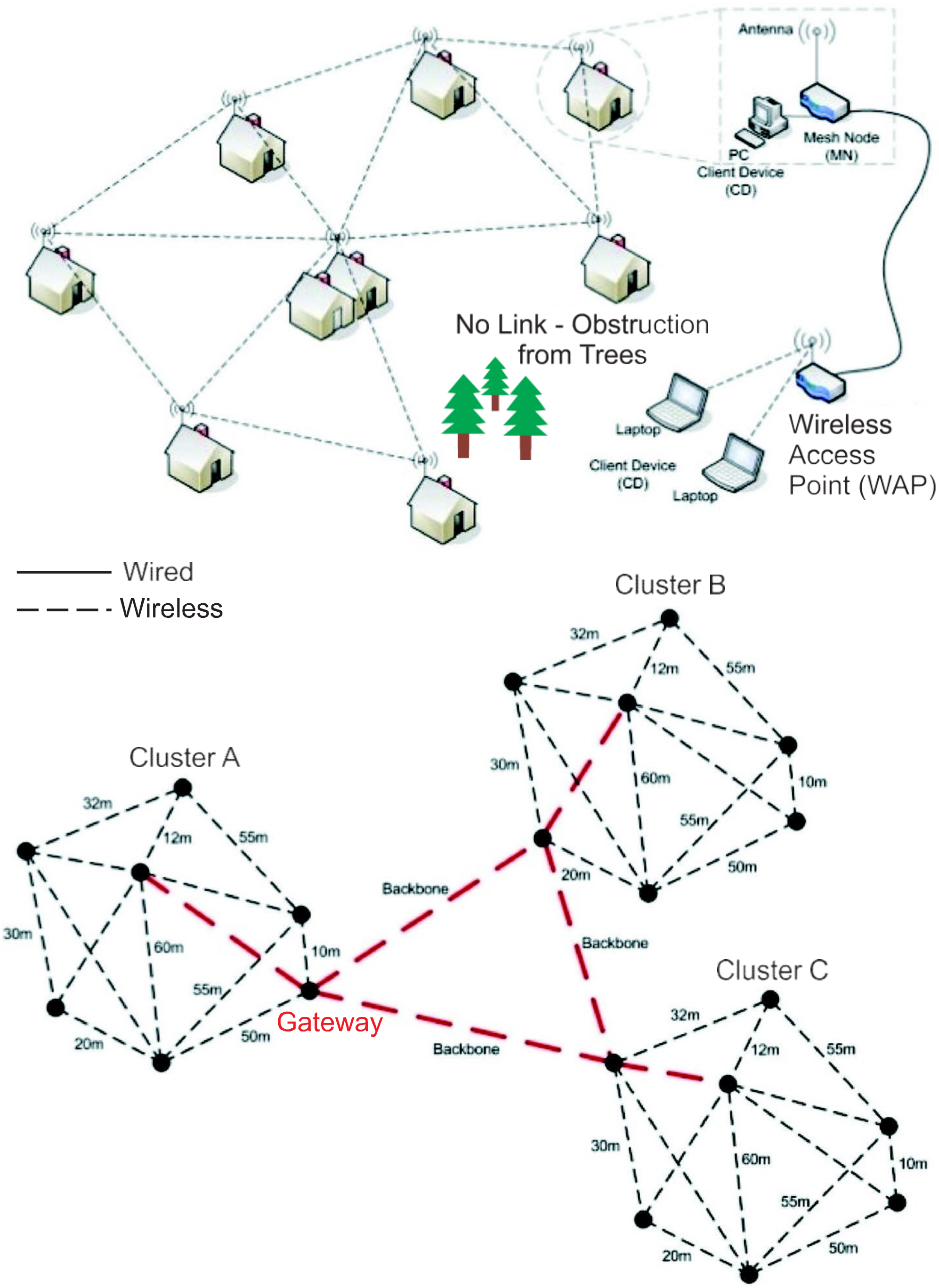

Figure 22.3: Clustered mesh with backbone

Activity 14.3

Prepare a small map, using any mapping site like http://maps.google.com for a possible mesh network in your neighbourhood. You will have to find out your location on the google map and other locations which you want to cover under your network. Once you have identified the locations, draw a line linking them.

Label them on the lines similar to that shown in Figure 22.3.

To do this activity, you may need about 30 minutes including writing down the answers in the space provided.

This activity will help you in understanding the use of mapping sites for creating a mesh network in your neighbourhood.

Mobile Telephony

In this section, you will learn how a mobile phone can easily be used for distribution of digital content. It is a well known fact that if there is any communication medium in the hands of the largest number of people in India, it is the mobile phone. Using mobile phone for distribution of digital content or even broadcasting is a reality that a radio technician today cannot afford to ignore. Though there are many possibilities, a few tried and tested ways of distributing content on mobile phone include:

- Caller tunes
- IVR based audio repository

We will briefly discuss them here.

Caller Tunes

If you have a short but very catchy audio clip which you may want to convert into a caller tune for mobile phones, you can create a page on your website to list such caller tunes for listeners to download on their mobile phones. You can even use social networking sites like Facebook or Twitter to draw listeners to your caller tunes. Film industry has been using caller tunes to popularise songs and dialogues of movies for quite some time now. It is one of the low cost options of content distribution. The caller tune eco-system spreads on its own. Having downloaded once, a user can easily transfer it to friends and colleagues via bluetooth or USB.

IVR Based Audio Repository

IVR stands for Automatic Voice Response. When you call a bank or your mobile company, you must have heard an automatic voice prompting you to select a number for the kind of services you require, for example dial 1 for banking service, dial 2 for credit card services etc.

This technology is in use for many years to give information to listeners. However, in the community media domain, it is also used for collecting information from listeners and playing it back to other listeners. An open source technology, called Asterisk, has made it very easy to configure an IVR system. Using Asterisk, you can configure an IVR prompting listeners to record their message on to the server. In the next stage you can review or edit the message recorded by a listener and play back to other callers. Asterisk is a software that can be installed on Linux or Unix based system.

The advantage of this is the fact that you have millions of potential listeners because of deep penetration of mobile in the society. On the other hand, the disadvantage is that you are dependent on telecom service provider. Also each call to the IVR server will cost the caller unless the cost is absorbed by the host organisation.

In India, there are two such experiments currently in operation and we will briefly discuss them. CGNet Swara was launched in the predominantly tribal region of Chhattisgarh and it acquired considerable popularity at least in the minds of media thinkers. CGNet Swara publicized a mobile number. A listener is expected to give a missed call to that number and the number will call you back. Once a listener gets a call, he or she can record a message, news or even a song. This recording would be reviewed by community editors and put into a bulletin for the day. At a specific allotted time of the day, a listener can give a missed call to the number again and get a call back from the server. In this call, the day's bulletin is read out to the listener. You can get more information on CGNet Swara, which is now supported by Microsoft, at http://cgnetswara.org.

A similar experiment was initiated by Gramvaani Community Media Pvt Ltd called Mobile Vaani. Mobile Vaani also uses Asterisk based IVR system to collect and publish information in rural Jharkhand. You can see more details about Mobile Vaani at http://gramvaani.org.

Activity 14.4

Using the Internet sites, prepare a list of five community radio stations in India, which are streaming on Internet. Also note the type of media server being used by them.

To do this activity, you may need about 15 minutes including writing down the answers in the space provided.

This activity will help you in learning the advantages of using streaming on the Internet.

Sl. No.	Name of the community radio station using streaming on the Internet.	Details of websites	Type of media server used
1			
2			
3			
4			
5			

Let Us Sum Up

In this unit, we looked beyond the conventional FM transmission as a method of broadcasting and sharing content. It is believed that with the growing trend towards digitalisation, the analogue FM transmission will become obsolete and digital transmission of data and voice will take over. Though there are many impediments in the path of complete digitalisation, it is imperative for a radio technician to understand the trend and undercurrents.

We understood the importance of Internet in community media in general and community radio in particular. As a community radio technician you can help your station in garnering maximum listenership by making effective use of social networking sites to the content sharing sites. Do not forget audio blogging. It may appear to be a relatively non-exciting mode of communication but has a huge potential.

On the other hand, streaming is closest to FM broadcasting in terms of effort and methodology. From the listener’s point of view, listening to FM radio and Internet radio is almost the same barring the difference in technology, receiver and cost. While FM radio is cheap to listen but restrictive in terms of its reach, the Internet radio is comparatively expensive to listen but has no geographical restriction. It is always beneficial to run the Internet radio and FM radio in parallel.

Wireless Mesh Networking is one of the emerging areas and holds a great promise for the future. The model, which is the most participatory in nature, is being adopted by many communities across the world. In this model, the community is in the centre and its members own and operate the communication infrastructure. It requires a very high level of community motivation, mobilisation and capacity building. The advantage of WiMesh is that you are not restricted to radio alone. You can do many things, such as run a community TV station, run a local telephone exchange, provide access to Internet, facilitate tele-medicine and tele-education to rural population, etc where these are needed the most. And most important of all, it brings a sense of connectedcommunity.

On the other hand, the mobile-based communication model is also a fast emerging solution. It is riding on rapidly growing number of mobile users across the world among rich and poor communities even though based on the current experiments in India, it may be stated that it has a limited functionality and it has a recurring cost factor.

Model Answers to Activities

Activity 14.1

While searching for free blogging sites, you may have come across sites like blogger.com, wordpress.com or weebly.com. These are all good sites and you can choose whichever you like. But it would be useful to choose one which has topics similar to the one you want to write on.

Activity 14.2

Once you have written your blog, you inform your friends and colleagues and encourage them to have a look at it. Remember, every click on your blog will be recorded by the blogging server and would help you in gaining popularity.

Activity 14.3

While trying to make a map, you might have noticed that there are differences in geo-locations as recorded by google map and the actual. One of the better ways of finding accurate geo-location is to use Global Positioning System device which are available in the market. These devices are expensive though. You can probably use your mobile phone if it has a GPS sensor or even assisted GPS system. Look at the technical specification of your phone to determine if you could use your phone.

Activity 14.4

While looking for the streaming community radio station, you would notice that most of them are using Icecast or Shoutcast servers. These are the most popular media server.

Additional Readings

- *Building a Rural Wiremesh Network, A Do-It-Yourself guide to planning and building Freifunk based mesh network Meraka Institute*. (2007). Retrieved from http://wirelessafrica.meraka.org.za
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Additional Activity

In order to improve your knowledge and skill further on the use of alternative mechanisms for content distribution, write a comparative analysis of various content sharing sites. It would be useful to have your thoughts also recorded in the analysis.

Glossary

Audio	means the audio frequency signals lying in the range of 20 Hz to 20 kHz.
Audio Blogging or Podcasting	is a term used for publishing digital content on the Internet.
Audio Cable	means cable used for carrying audio frequency signals.
Audio Connector	means connector used in the audio frequency range.
Content Sharing Sites	are the Internet sites that are designed specifically for sharing content of community radio.
IVR	stands for Interactive Voice Response.
Line Cable	means an audio cable used for connecting the input channels to the mixer.
Microphone Cable	means a cable used for connecting output of the microphone to the mixer or any other recording equipment.
RF	means radio frequency used for transmitting of programmes. In our case, it means frequencies transmitted by stations in FM band (88MHz to 108MHz).
RF Connector	means connector used in radio frequency range.
Social Networking Sites	are Internet sites that run on user-generated contents.
Solder	means a solder metal (wire) used for soldering the joints.
Streaming	is a term used for live transmission of data over the Internet.
WiMesh (Wireless Mesh Networking)	is a community-owned network of Wi-Fi nodes.

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