

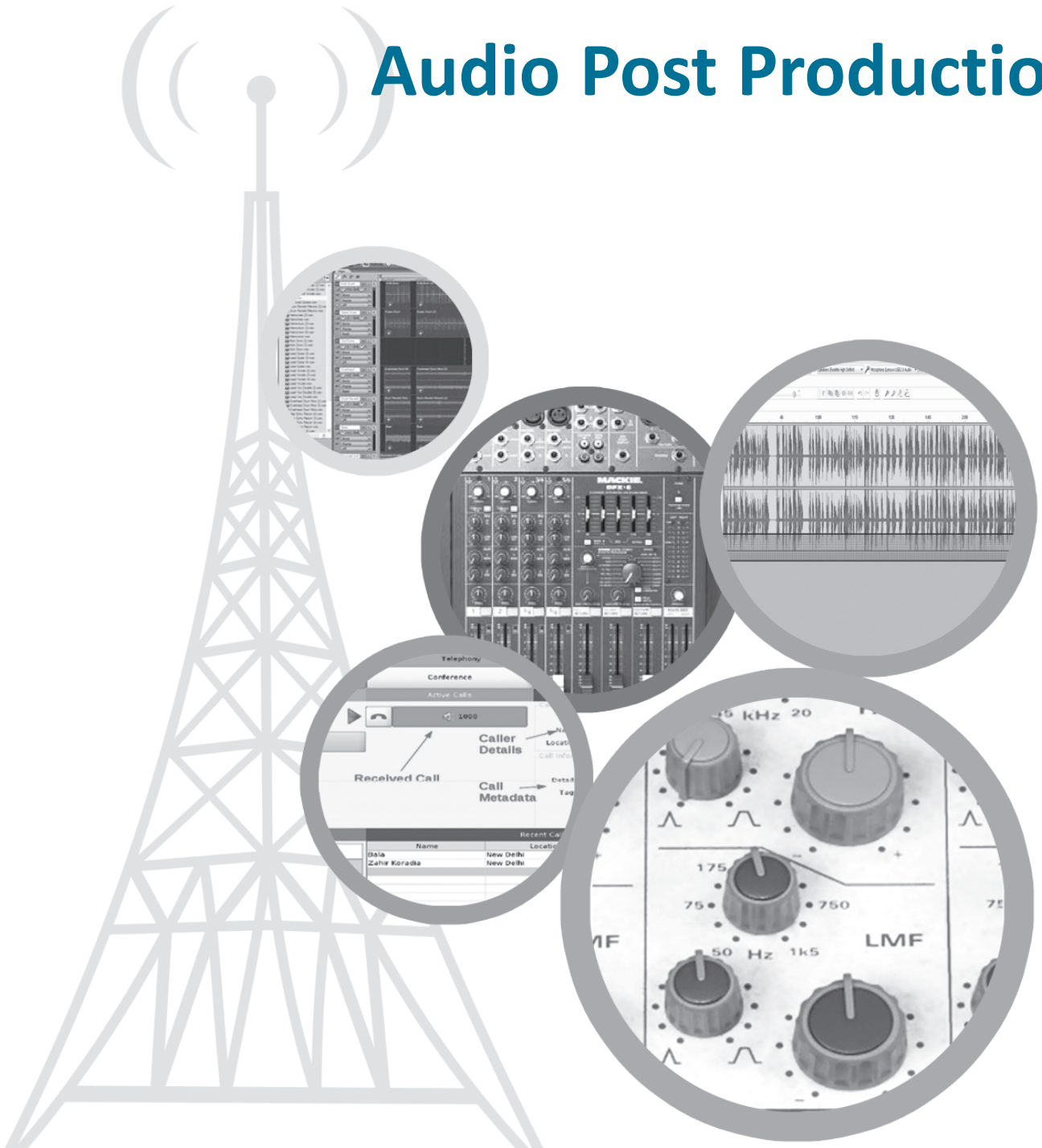
Audio Post Production

5



Module: 5

Audio Post Production



CEMCA

Commonwealth Educational Media Centre for Asia
New Delhi



बेसिल

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Module 5: Audio Post Production

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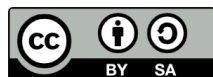
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Certificate in Community Radio Technology

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

Video in the Module:



1. <http://tinyurl.com/ohcffan>
2. <http://tinyurl.com/nhpypz7>

CONTENTS

	Page No.
About the Module	7
Unit 16 : Sound Recording and Editing	8
16.1 Introduction	
16.2 Learning Outcomes	
16.3 What is Sound Recording?	
16.4 Single and Multi-track Recording	
16.5 What is Audio Editing?	
16.6 Equalisation and Digital Audio Processing Equalisation (effects etc)	
16.7 Understanding the DAW and Editing Software (Integrated video)	
16.8 Let Us Sum Up	
16.9 Model Answers to Activities	
Unit 17: Mixing and Mastering	32
17.1 Introduction	
17.2 Learning Outcomes	
17.3 Mixing	
17.4 Mastering and Export	
17.5 Let Us Sum Up	
17.6 Model Answers to Activities	
Unit 18: File Formats and Compression	44
18.1 Introduction	
18.2 Learning Outcomes	
18.3 Types of Audio Formats	
18.4 Need for Compression	
18.5 Compression Techniques	
18.6 Comparison of Formats (tabular)	
18.7 Format Converters	
18.8 Let Us Sum Up	
18.9 Feedback to Check Your Progress	
18.10 Model Answers to Activities	
Unit 19: Storing and Retrieval	55
19.1 Introduction	
19.2 Learning Outcomes	
19.3 Data back-up Techniques	
19.4 Storage Devices	
19.5 Audio Archiving	
19.6 Logging	
19.7 Let Us Sum Up	
19.8 Feedback to Check Your Progress	
19.9 Model Answers to Activities	

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 16

Sound Recording and Editing

Structure

- 16.1 Introduction
- 16.2 Learning Outcomes
- 16.3 What is Sound Recording?
 - 16.3.1 The concept of recording audio
 - 16.3.2 The concept of audio levels
 - 16.3.3 The recording process
- 16.4 Single and Multi-track Recording
 - 16.4.1 Single track recording and multi-track recording
 - 16.4.2 Recording when listening to a reference track
- 16.5 What is Audio Editing?
 - 16.5.1 The concept of editing audio
 - 16.5.2 The process of editing audio
 - 16.5.3 Destructive and non-destructive editing
- 16.6 Equalisation and Digital Audio Processing Equalisation (effects etc)
 - 16.6.1 Equalization
 - 16.6.2 Digital audio processing (DAP) and effects
- 16.7 Understanding the DAW and editing software
- 16.8 Let Us Sum Up
- 16.9 Model Answers to Activities

16.1 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.



16.2 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 (analog-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 16.5. 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 will 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.

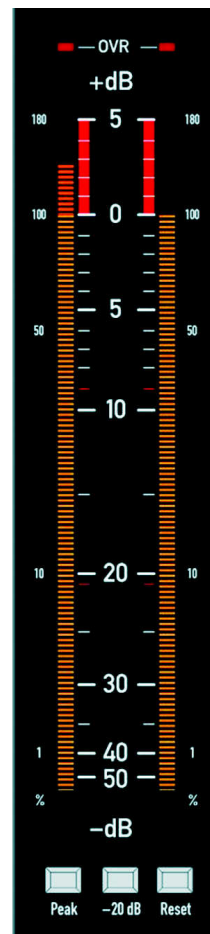


Figure 16.1: A typical digital VU meter. Aim to keep the audio between the -10 dB and the -5 dB mark, with loudest noises not reaching the 0 dB mark. VU meter image;
Source: http://en.wikipedia.org/wiki/File:PPM_IEC_268-10_I_DIN.jpg; CC BY-SA

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 microphones 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.

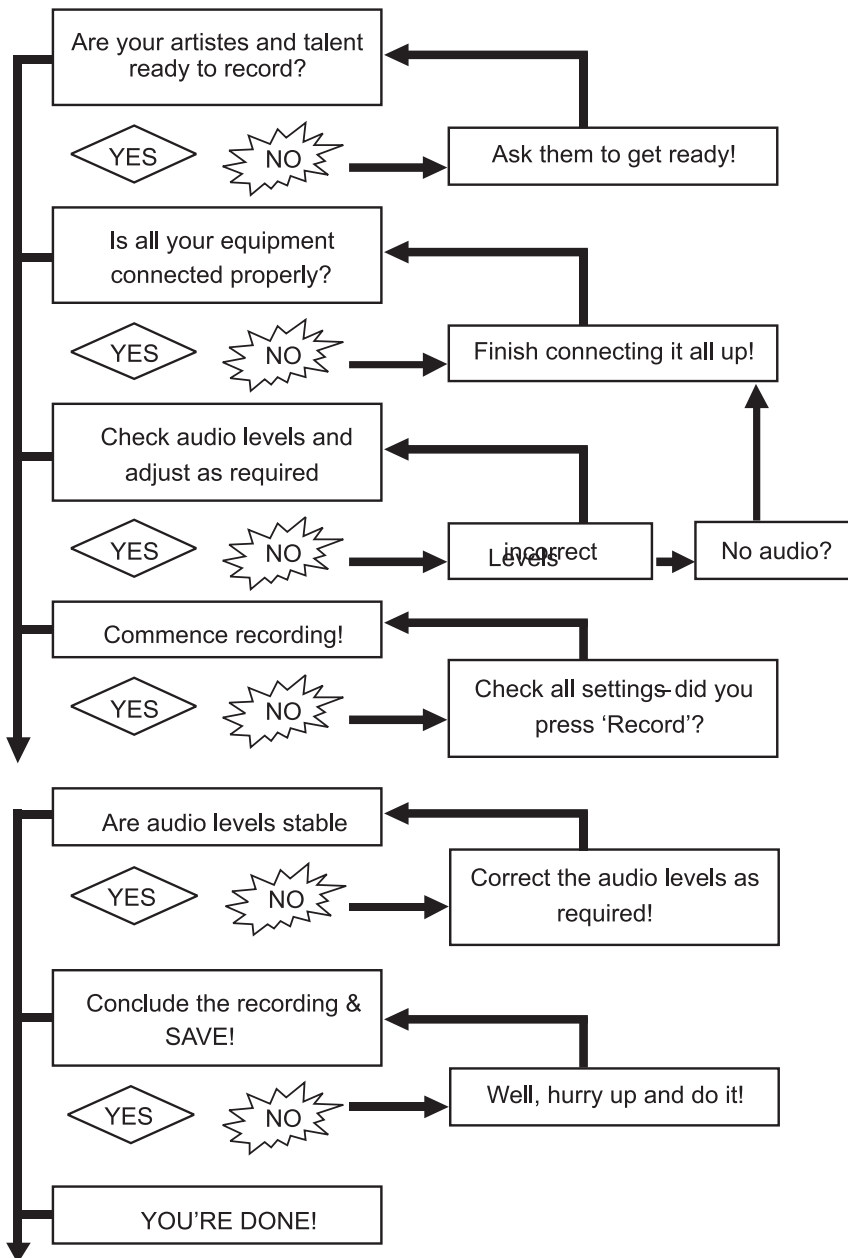
Step 4: Monitoring and adjustments during recording

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 16.1

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 16.2: 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**.

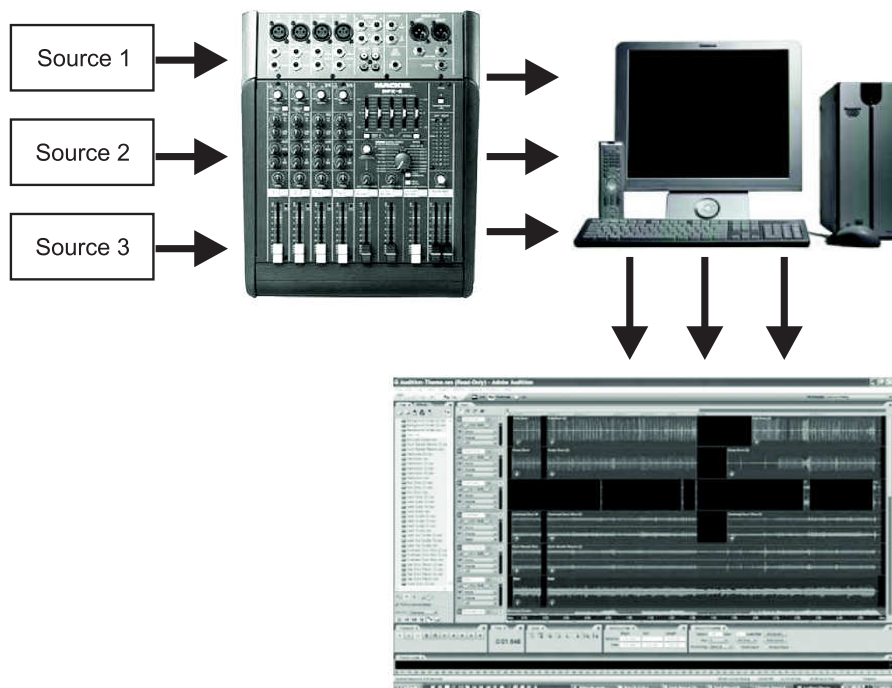


Figure 16.3: Multiple sources recorded through a mixer to a DAW, resulting in multiple mono tracks. This is called multi-track recording.

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 16.2

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.

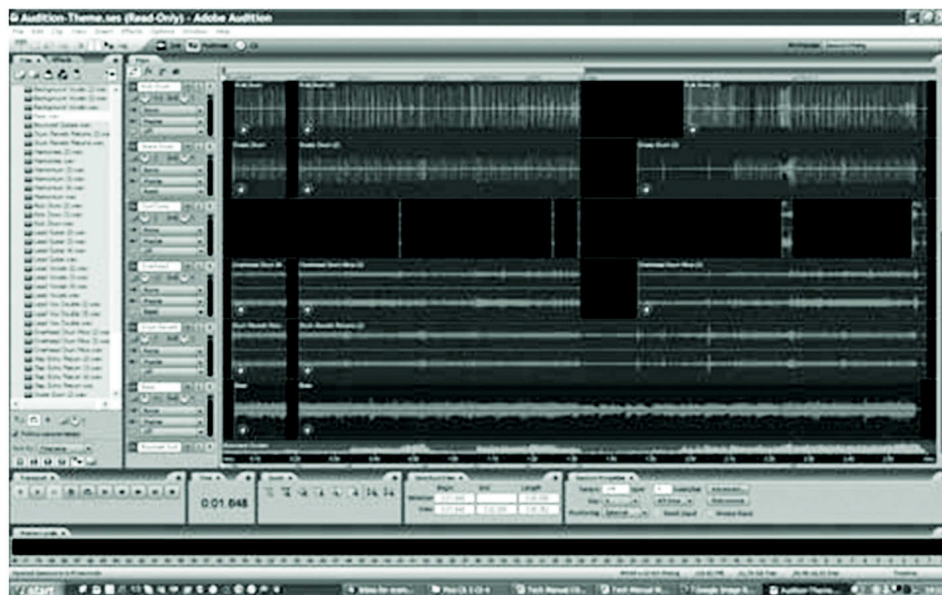


Figure 16.4: The editing interface. Note the source clips in the window to the left; the name of the editing session file at the top; the timeline with the various tracks arranged vertically, and the audio clips placed on the timeline.

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 up to 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 17) 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 irretrievably 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 programmes, 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 16.3

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 16.5:



Figure 16.5: 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 16.4



<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 16.5 below.



Activity 16.5

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 16.1

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 16.2

In a multi-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 16.3

In Section 16.5.3, 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 software to 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 16.4

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 16.5

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 17

Mixing and Mastering

Structure

- 17.1 Introduction
- 17.2 Learning Outcomes
- 17.3 Mixing
 - 17.3.1 Levelling: The need to adjust clip levels
 - 17.3.2 Balancing: The need to adjust track levels
 - 17.3.3 Panning: Adjusting spatial distribution
 - 17.3.4 Mixing audio
- 17.4 Mastering and Export
- 17.5 Let Us Sum Up
- 17.6 Model Answers to Activities

17.1 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.



17.2 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.

17.3 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.

17.3.1 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).

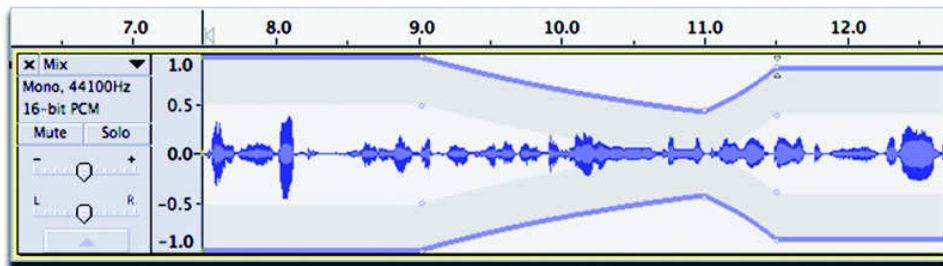


figure 17.1: A clip on the timeline in the Audacity software. Note the graphical overlay that has been used to adjust the audio levels at various points in the clip in order to even out the perceived audio volume through its duration.

17.3.2 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).

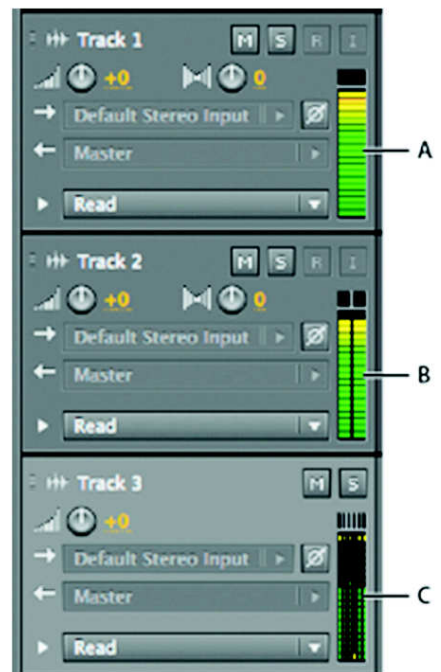
Though, we will include variety of audio besides spoken words, but we will always favour the spoken word in most programmes in terms of audibility and clarity. Therefore, we need to adjust the level of the music track in such a way that the music is heard, but does not overpower the voice. Similarly, 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 or 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 17.2: 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 17.1

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.

17.3.3 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.

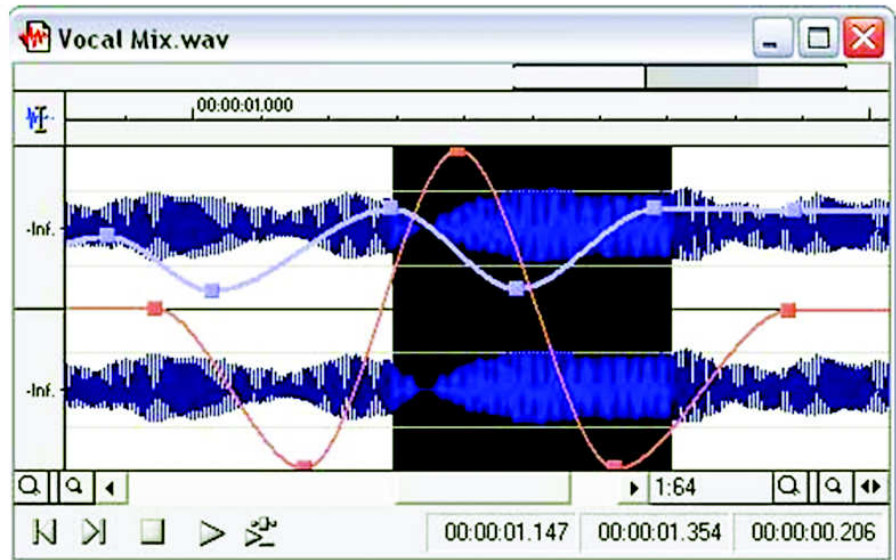


Figure 17.3: A clip showing audio level adjustment as well as PAN adjustments.

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

17.3.4 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 -16 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 17.2

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 be heard from the left speaker, and Track 2 (the interviewee) from the right speaker.

17.4 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 remains 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.

1. **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.
2. **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.

- 3. 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 prohibited 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.
- 4. 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 17.3

Continuing from the activity presented in Activity 17.2, 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'.



17.5 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.).



17.6 Model Answers to Activities

Activity 17.1, 17.2, 17.3

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 18

File Format and Compression

Structure

- 18.1 Introduction
- 18.2 Learning Outcomes
- 18.3 Types of Audio Formats
- 18.4 Need for Compression
- 18.5 Compression Techniques
- 18.6 Comparison of Formats (tabular)
- 18.7 Format Converters
- 18.8 Let Us Sum Up
- 18.9 Feedback to Check Your Progress
- 18.10 Model Answers to Activities

18.1 Introduction

Having learnt about the techniques of production and post-production in Unit 17, 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.



18.2 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.

18.3 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.

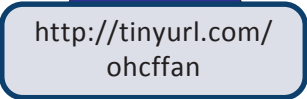
Before going into the details of the various concepts of audio formats and compression, you can watch the video available at the CEMCA YouTube site - <http://tinyurl.com/ohcffan>. This video will help you to enumerate the concepts of audio formats and compression.

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.



Audio Formats & Compression



<http://tinyurl.com/ohcffan>



Activity 18.1

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.

18.4 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 18.2

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.

18.5 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.

18.6 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 18.3

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.

18.7 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 18.4

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.



18.8 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.



18.9 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.



18.10 Model Answers to Activities

Activity 18.1

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 18.2

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 18.3

.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 18.4

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 19

Storing and Retrieval

Structure

- 19.1 Introduction
- 19.2 Learning Outcomes
- 19.3 Data Back-up Techniques
- 19.4 Storage Devices
- 19.5 Audio Archiving
 - 19.5.1 Factors for archiving audio
 - 19.5.2 Meta data tagging
- 19.6 Logging
- 19.7 Let Us Sum Up
- 19.8 Feedback to Check Your Progress
- 19.9 Model Answers to Activities

19.1 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.



19.2 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.

19.3 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 19.1

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?

19.4 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:

	<p>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.</p>
	<p>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.</p>
	<p>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.</p>
	<p>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.</p>



Activity 19.2

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.

19.5 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.

19.5.1 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 to 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.

19.5.2 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.

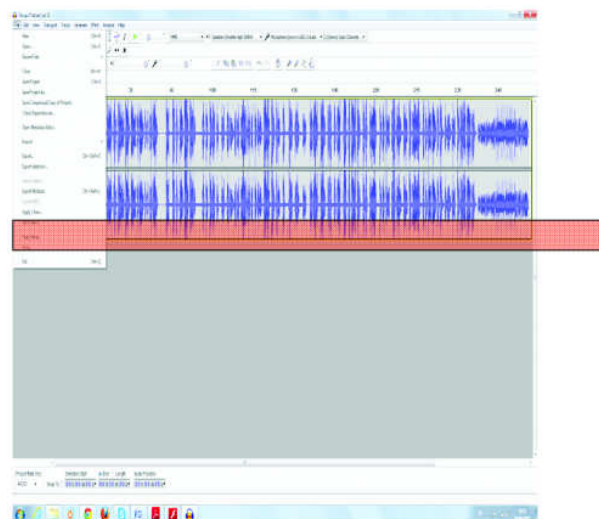
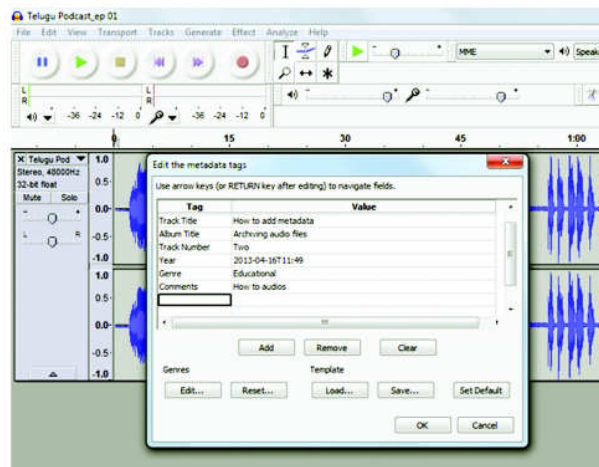
So what is meta data tagging? Simply, it is data that describes other data. Metadata can be used to describe digital files such as videos, images, documents,

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.

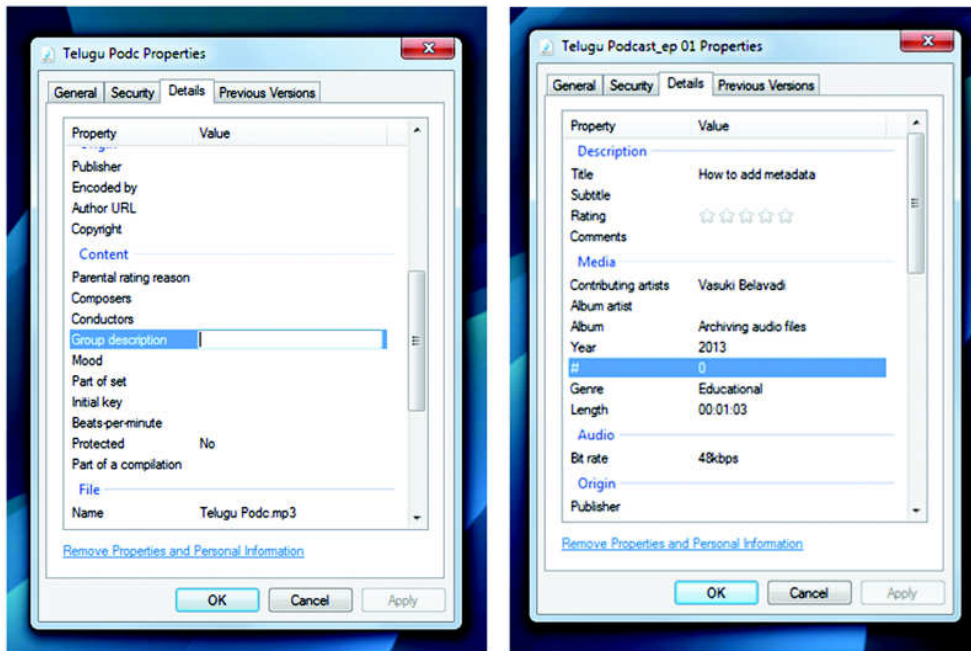
Most of the times the metadata entries that your audio editor provides are good enough and are searchable. It is searchable more easily when the metadata itself can be exported as a separate file. For example, how does one go about adding metadata to an audio file, while using Audacity?

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 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 19.3

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.

19.6 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.

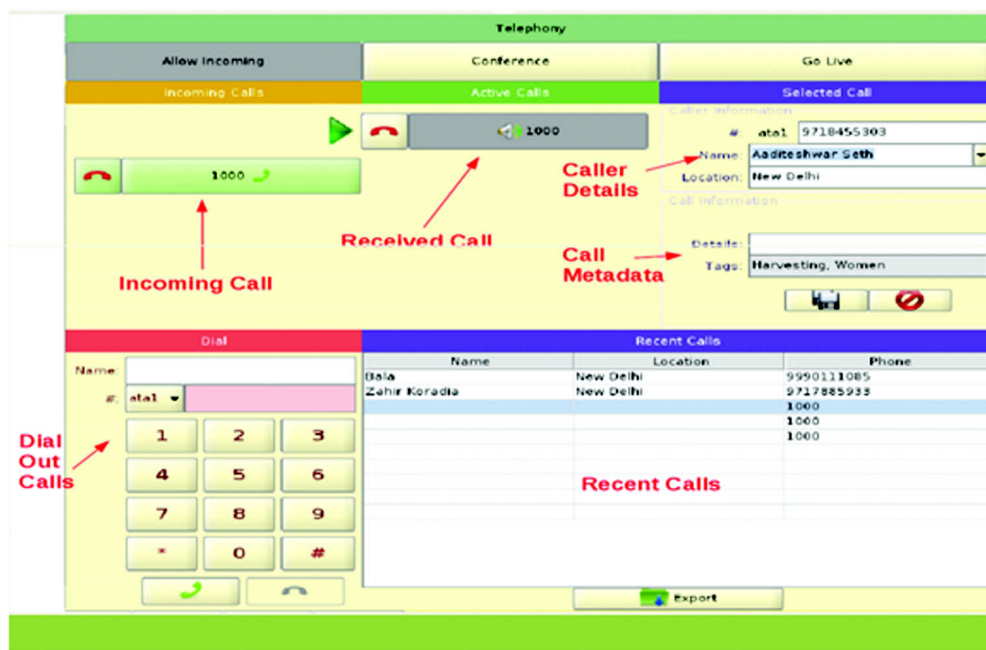


Figure 19.5.1: Screenshot of the telephony component of GRINS broadcast automation software. This allows you to store the names of the callers and save a list too.



19.7 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.



19.8 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.



19.9 Model Answers to Activities

Activity 19.1

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 programmes 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 19.2

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 19.3

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 unforeseen 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.



CEMCA

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