Certificate in Community Radio Technology

Audio Production

4

Module: 4
Audio Production

Commonwealth Educational Media Centre for Asia
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Broadcast Engineering Consultants India Ltd.
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Module 4: Audio Production

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

Video in the Module:

Studio Recording Hardware
by Ramnath Bhat

http://tinyurl.com/ppkowtp

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About the Module

Module Description

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

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

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

Module Objectives

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

Units in the Module

- Unit 13: Audio Hardware and Field Recording
- Unit 14: Free and Open Source Software
- Unit 15: Telephony for Radio

UNIT 13

Audio Hardware and Field Recording

Structure

13.1 Introduction

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

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

While going through the unit, you will need to do four activities (the model answers of which are given in Section 13.13). You will need approximately four hours to complete this unit.

13.2 Learning Outcomes

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

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

13.3 Hardware for Audio Recording

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

Studio Recording Hardware

http://tinyurl.com/ppkowtp

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

Let us look at these in greater detail.

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

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

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

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

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

13.4 Microphones

You have already learnt about the details of various types of microphones in Section 11.4 of the previous module. You have got to knowtherein that

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

13.4.1 Key Considerations while Selecting Microphones

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

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

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

3. Pickup pattern: Situations may demand requirement of a producer to record sound either from one direction, two directions or all directions. Situations may also require the producer to actively reject sounds from other directions while picking up sounds clearly from one. There are different microphones that meet these requirements. The choice of a microphone is often determined by this requirement. You can read more about this in Section 13.4.2, where we discuss different pickup patterns.

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

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

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

13.4.2 Microphones Categorized by Directionality

Omni-directional microphones

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

Uni-directional microphones

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

Bi-directional microphones

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

13.4.3   Microphones Categorized by Construction

Dynamic microphones

Also called moving coil microphones, dynamic microphones are rugged and robust and are mostly used in field situations. They are more resistant to

unwanted vibration and wind rumble. The microphones that are used by rock stars are normally of the dynamic type. Typically, their frequency range lies between 40 and 18,000 Hz. Figure 13.1 below shows a popular dynamic microphone.

Figure 13.1: *A dynamic microphone.*
(Source: Jnan Taranga, KKHSOU, Guwahati)

Condensor microphones

Condensor microphones are much more sensitive microphones, and are generally used for studio applications. Typically, their frequency range covers the entire human hearing range from 20–20,000 Hz. Unlike dynamic microphones, condenser microphones need an external power supply. Some condenser microphones have a battery attachment within the body of the microphone itself. Batteries are usually replaceable, as required. Others need power delivered to them through the microphone cable itself, an arrangement called **Phantom power**, and often denoted by the symbol **+48V** or **P48**. Figure 13.2 shows a popular condensor microphone.

Figure 13.2: *A condensor microphone.*
(Source: Jnan Taranga, KKHSOU, Guwahati)

 ## Activity 13.1

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

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

a. By speaking directly in front of it

b. By speaking next to it

c. By speaking sitting next to it

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

13.5 Audio Recorders

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

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

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

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

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

- Recording format: Should be able to record in .WAV format, 16/24 bit. It is better if it can also record 'mp3' format at varying bit rates.

- Should have built-in stereo microphones, preferably covered with a wind shield

- Should have a provision for line-in stereo, input for external microphone (preferably XLR input)

- Should have a provision to connect headphones to monitor audio

- Should have built-in speakers

- Should have adjustable controls for input audio levels

- Should have an audio level indicator (VU meter)

- Should have a media card slot with a capacity for high data storage (2 GB or more)

- Should have a USB cable and connector slot to enable import of audio files to a computer

- Some recorders are capable of recording from a landline telephone cable. Some can also be connected to a computer and used as a microphone to record directly into the computer using an audio software.

- Some recorders come with built-in batteries that can be charged. Others accept consumer grade batteries of AA or AAA type. For recorders that use the latter, you will require some rechargeable cells of appropriate size, preferably of the Nickel Metal Hydride (NIMH) type. As a general practice, it is always good to carry a couple of extra chargeable batteries, so that you do not run out of power during an extended day's recording.

Figure 13.3: *Zoom, Olympus and Tascam manufacture some of the most popular digital audio recorders.*
(Source: Jnan Taranga, KKHSOU, Guwahati)

The most popular type of recorders available in the market today are **digital flash memory recorders**. These recorders make digital recordings onto flash memory cards, such as **compact flash (CF) cards** and **secure digital (SD) cards**. Usually, they

have an option to save as uncompressed linear PCM or in a compressed format such as MP3. For a given size of memory card, you can record for a longer duration using a compressed format but at a lower quality. Some of these recorders record on a built-in flash memory.

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

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

13.6 Audio Cables and Connectors

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

13.6.1 Balanced and Unbalanced Cables

In an earlier discussion in this unit (Section 13.4.1), we learnt about balanced and unbalanced microphones. It should stand to reason that both these types of microphones will require appropriately matched cables and connectors to carry the signal they generate.

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

Any circuit is complete only when the electrical signal has an incoming and a return path. In the case of the unbalanced cable, the central core provides one of the paths, and the metal shield the other. Unfortunately, this means that any interference that the shield picks up will be added to the audio signal. This can cause disturbances or hums in the recording. Figure 13.4 shows an unbalanced cable.
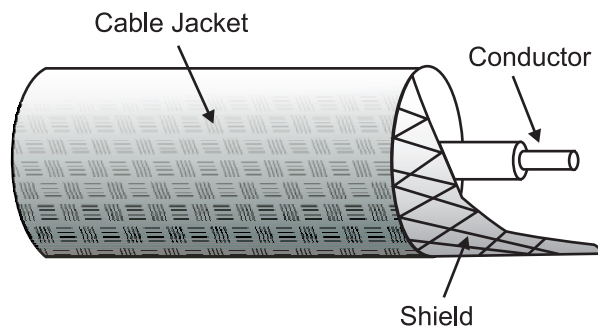
Figure 13.4: *An unbalanced cable. Note the lone central conductor and the two layers of insulation, one of which separates the central core from the shield.*

On the other hand, a **balanced cable** is a cable containing two cores, one positive and the othernegative, which are twisted together and surrounded by an overall shield. Only the two core wires carry the signal and the shield is grounded directly to the frame of the audio equipment. This allows the shield to perform its function of trapping external electromagnetic interference. As a result, the audio signal is completely noise free. Figure 13.5 shows a balanced audio cable.
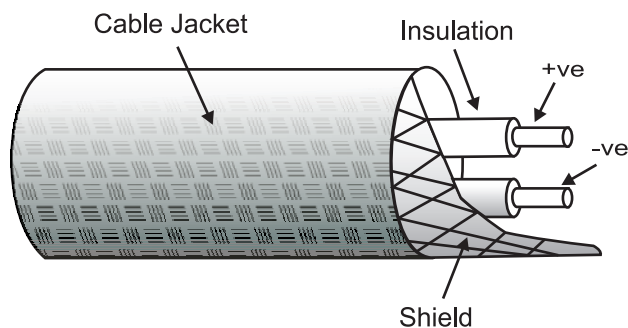
Figure 13.5: *An balanced cable. Note the dual core wires.*

Having dealt with cables, let us discuss connectors now.

13.6.2 Audio Connectors

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

RCA connectors

RCA connectors are named after the Radio Corporation of America, who invented this type of connector. The male RCA plug consists of a central pin measuring

approximately two millimetres (mm) in diameter, and an outer shell whose inside diameter is approximately six mm. The plug shell is slotted rather than threaded, to facilitate quick insertion and removal from the female jack or receiver. Typically, the cable carries the plug, and the audio device carries the jack. The RCA connector does not provide a balanced audio output. This type of connector is often used in home entertainment systems. You can see a pair of RCA plugs in Figure 13.6 below.
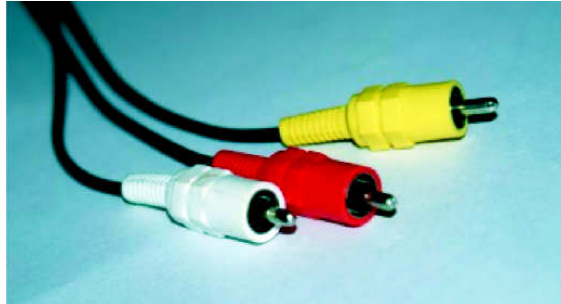
Figure 13.6: *A pair of RCA connectors.*
(Source: Jnan Taranga, KKHSOU, Guwahati)

XLR connectors

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

XLR connectors also come in a mini-XLR variety. These are normally used with field wireless microphones and field recorders. You can see a pair of XLR connectors in Figure 13.7.

Figure 13.7: *A pair of XLR connectors. The one on the left is the female connector; and the one on the right the male connector.*
(Source: Jnan Taranga, KKHSOU, Guwahati)

TRS Connectors

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

A larger variety of TRS connectors is the **phono** or **quarter inch (1/4") jack**, named after its pin that is a quarter-inch wide (see Figure 13.8). The smaller variety, often called the **EP** or **mini-phono jack**, has a 1/8" inch (3.5 mm) wide pin. The former is more commonly found on professional equipment and the smaller one on consumer grade equipment like MP3 players and mobile phones.

Figure 13.8: *1/4" phono jacks. The one on the left is mono, the one on the right is stereo (Source: Jnan Taranga, KKHSOU, Guwahati)*

13.7 Headphones

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

Ear bud headphones

These are probably the most common types of headphones, as they are used with all kinds of portable music players and mobile phones. Also called **earphones**,

they are often provided as free accessories with these devices. While some low-quality ear bud headphones fit loosely within the external ear, there are some that fit into the ear canal itself. While earphones are good for listening to music, they are best avoided to monitor audio while recording. Figure 13.9 shows a pair of earphones.

Figure 13.9: *Two pairs of simple earphones. The pair on the right has an ear canal* adapter
for snug fit.
(Source: WikiMedia Commons, *http://tinyurl.com/4eboaz3*)

On-ear headphones

These are headphones that sit on the ears rather than over them. As a result, they are usually a bit smaller and lighter than over-the-ear models. They tend to have foam or sometimes leatherette pads for extra comfort, and usually have adjustable headbands for a snug fit. These headphones are normally good on treble but not on bass. Since they don't cover the ears, ambient noise tends to enter the ears, making it difficult to monitor audio in critical conditions. They are therefore best used in office situations, for simple listening purposes, or for conducting voice chats over the Internet. Figure 13.10 shows a pair of on-ear headphones.

Figure 13.10: *On-ear headphones*
(Source: Jnan Taranga, KKHSOU, Guwahati)

Over-the-ear headphones

These are traditional-looking headphones, with cushioned pads that enclose and cover the whole ear. This makes them more comfortable to wear over long periods, and they generally deliver good sound quality. Bulkier than other types of headphones, these are best suited for audio monitoring purposes in the studio as well as in the field. Some varieties also cancel out noise, making it easier for the producer/technical personnel to monitor audio. The balanced headphone variety under this category provides the same impression as the sound you would be hearing from two or more speakers. Figure 13.11 shows a pair of professional over-the-ear headphones.

*Figure 13.11: A pair of over-the-ear headphones. Note the cups that cover the entire ear.
(Source: Jnan Taranga, KKHSOU, Guwahati)*

 Activity 13.2

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

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

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

13.8 Recording Audio in the Field

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

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

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

The pre-production phase: Key points to remember

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

1. **Weather:** The ideal weather conditions for radio field recording depends on what exactly you are wanting to record; but for most situations, a dry and calm (less windy) day is ideal. On other days of the year, take care to protect your equipment from rain. A small umbrella comes in handy both in summer and monsoon. If you intend to travel long distances for a field recording, check on the weather conditions at your destination before you leave.

2. **Time:** Even in the best of situations, field recordings take a lot of time. Keep time for walking/driving from one place to another, waiting for people, and other unforeseen delays. Traffic can pose a major problem in urban settings. Planning your time is extremely important, and so is time management. Keep adequate time for non-production activities like lunch. Keep spare time for every activity, and a backup plan ready in case you are unable to complete what you need to within the allocated time.

3. Batteries andstorage cards: Irrespective of the kind of battery your recorder uses, carry an extra set of batteries and a charger. Having to return to your base just to charge batteries can mean losing out on time, energy, money, and most importantly, the people you wish to interview. By the same logic, keep an extra storage card of the kind used in your recorder handy. An interview could go longer than you thought, and you don't want to miss out on recording something just because you ran out of space.

4. Mobility: Do you have your entire production kit neatly packed? Are the audio recorder, batteries, cables and any accessories neatly stored? Do you have a headphone? Are all pieces of equipment working properly? Are the headphone and microphone cables coiled properly? Are non-production items like biscuits or a water bottle in a different bag? Check for all of these before setting out onyour recording trip.

The production phase: Key points to remember

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

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

- Make sure your recorder is working. Check the recorder's display once in a while.

- After pressing the record button, start by saying the subject, location and time of your recording. This may not be always possible but is definitely possible before conducting interviews. In case you lose your notes later on, you will be at least able to understand what you have recorded.

- Use a quiet location to record interviews. Switch off fans and air conditioners while recording interviews. Microphones have a tendency to pick up those sounds.

- Record the room tone (when nobody is talking) for about 3–4 minutes. This tip is also useful when recording interviews outdoors. Whileediting, this tone can be used across the interview on another track to maintain continuity.

- Always point the microphone to the source of the sound. Place it at least 3 to 6 inches away from the source.

- Microphones need a critical distance to record the right quality of audio. If you take the microphone too close, the audio will be distorted and the recording may have pops of air from the speaker's mouth. On the other hand, taking it far away from the source of the sound is likely to result in a weak and hollow sound quality.

- If you are using handheld microphones, always talk across them. The microphone is best placed at the chest position. However, if you are using a uni-directional microphone, you should keep it close to the mouth.

- Sit down when you have to interview children. This way you can place the microphone at a comfortable position for both of you.

- Do not conduct interviews while walking. People tend to get distracted and there may be long pauses in sentences.

- Do not ask close-ended questions. Asking a question that begins with 'do you...' invariably results in a one word 'yes/no' answer. Ask questions that begin with 'what', 'how', 'why'.

- When recording interviews with more than one person, alert them in advance to speak only when the recorder/microphone is pointed at them.

- Do not place the recorder on the ground/table to record interviews. If you have an external microphone attached, hold the recorder in one hand, where you can see the audio levels on the screen, and the microphone in the other. whileusing abuilt-in microphone, hold the recorder in whichever hand you find it comfortable—but remember to keep an eye on the audio level meter!

- Use microphone stands where possible, to keep the microphones steady.

- When using multiple microphones for a talk show, place the second microphone at least twice the distance away from the user as the first. (For example, if you place one microphone at a distance of one foot from two speakers, place the next microphone at least two to three feet away from the first one.) This helps prevent phase cancellation, a phenomenon where the signal from one microphone may be partially cancelled out by the signal from the other, resulting in a poor recording.

- In a noisy environment, position the interviewee so that his back is against the area with the lowest noise or disturbance. That way, when the microphone is pointed at him or her, the noisiest areas will be behind the microphone, and the person's voice will be recorded more clearly.

- Nod your head during an interview. Avoid 'hmm', 'uh' and 'oh!', although not at all times. Too many aural responses from you might make it too conversational, and result in audio that is difficult to edit later.

- Use directional microphones wherever necessary.

- Switch off/fade out microphones when not in use.

- Use a windshield when using microphones outdoors, and pop filters when recording indoors. This avoids wind or breath from hitting the microphone.

- Always use headphones/speakers when recording sound outdoors/ indoors.

- Never allow the audio to consistently touch the 'red' mark or zero level on your audio recorder's level meter.

- Record Sound on Tape (SoT) without fail. This is a track containing just the sounds in the environment, without the speakers' voices. Also known as ambient sound, this lends character to your programme by providing the listener a clue to the environment in which the interview or discussion is being conducted.

- Eye contact is very important when recording interviews. Look the person in his or her eye when asking questions.

- Listening to your interviewee is also extremely important. It helps you ask additional questions that might bring new insights to the programme.

- Always monitor sound from the recorder as you conduct the recording. You will know immediately if there has been any kind of disturbances in the recording.

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

- Do not tap or blow into any microphone to test it. Just talk.

- Keep microphones in their pouches when not in use.

- Store microphones safely in an almirah or a box.

- Keep microphones away from moisture, water and fire, as well as physical shocks of any kind.

- Switch off the microphones after use, if they have in-built off/on switches. This is especially important for battery powered condensor microphones.

- Remove the batteries if not in use. Replace them with new but similar type batteries periodically.

- Use high-quality connectors and cables for microphones.

- Do not pull out a connector holding the cable. Always grasp the connector (jack) to disconnect.

- Do not leave heavy microphones dangling from a cable. Do not swing them holding the cable either.

- Do not use any liquid whatsoever to clean microphones.

Activity 13.3

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

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

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

13.9 Portable Audio Mixers

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

- **Control the volume of signals** coming from various inputs (microphones, audio players, etc.) using faders. The faders help in regulating the input levels of sound coming from various sources.

- **Combine and balance the inputs coming from various sources**, by sliding the faders up and down. For example, in an audio mixer with six inputs, you connect six microphones to record a musical score. Controlling each source by way of its dedicated fader helps one in balancing the input of each instrument, so that we make the combined output pleasant to hear.

- **Equalize the audio**, that is, manipulate the frequency characteristics of the input sound. Equalizing involves controlling the audio signal by increasing some wanted frequencies and decreasing the unwanted ones. For example, you can reduce hiss in speech by equalizing the input.

- **Auxiliary Send** ("Aux Send") is used to send the output to other external devices like an effects generator. The output from the Aux Send is again taken back into the audio mixer, mixed and routed through the final output.

- **Route the mixed and equalized audio signals** to a specific output (a recorder/speaker/headphone).

- **Enable monitoring** of all the functions mentioned above through speakers/headphones as they are being recorded/broadcast.

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

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

Portable audio mixers come in handy for use in the field when there is more than one source of sound. It allows one to combine the signals from multiple audio sources, be it microphones or other audio players, and mix them all. They work either on direct power supply or batteries. The output from the field audio mixer is then connected to the input of an audio recorder. The audio recorder then records the mixed audio from various sources. Figure 13.12 shows a model of field mixer.

Figure 13.12: *Portable audio mixer.*
(Source: Jnan Taranga, KKHSOU, Guwahati)

Professional portable field mixers have the following features:

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

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

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

13.10 Sound Cards

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

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

Some of the motherboards used in computers have good sound cards that are integrated into the motherboards themselves. However, most computers used for professional audio usually need an independent sound card. These are usually plugged into the Peripheral Component Interconnect (PCI) slot on the motherboard. Figure 13.13 shows a sound card of this kind.
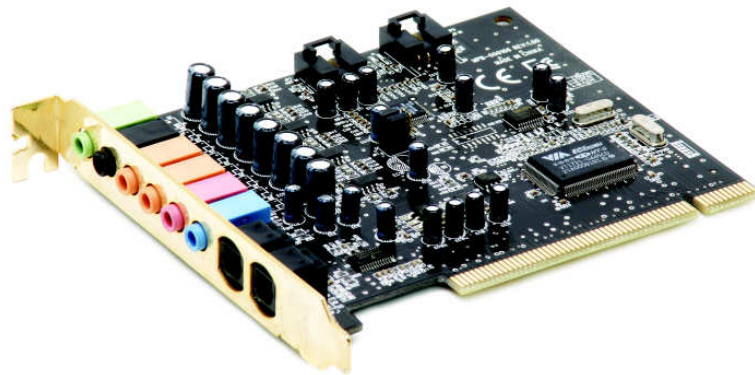
Figure 13.13: *An internal sound card. Only the silvered surface at the front and the coloured connectors are visible at the rear of the CPU cabinet.*
(Source: WikiMedia Commons, *http://tinyurl.com/32ssttc*)

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

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

13.11 Digital Audio Workstations (DAWs)

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

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

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

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

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

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

Activity 13.4

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

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

13.12 Let Us Sum Up

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

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

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

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

13.13 Model Answers to Activities

Activity 13.1

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

Activity 13.3

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

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

UNIT 14

Free and Open Source Software in CR

Structure

14.1 Introduction

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

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

While going through the unit, you will need to do four activities (the model answers of which are given in Section 14.6). You will need approximately ten hours to complete this unit.

14.2 Learning Outcomes

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

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

14.3 What is Free and Open Source Software?

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

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

14.3.1 Definitions

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

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

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

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

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

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

The Debian Free Software Guidelines referred to the "Cathedral Vs Bazaar", a much talked about article written by Eric Raymond in the same year, which

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

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

14.3.2 History

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

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

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

The free and open source software movement got a major boost with the

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

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

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

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

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

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

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

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

14.3.3 Current Scenario

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

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

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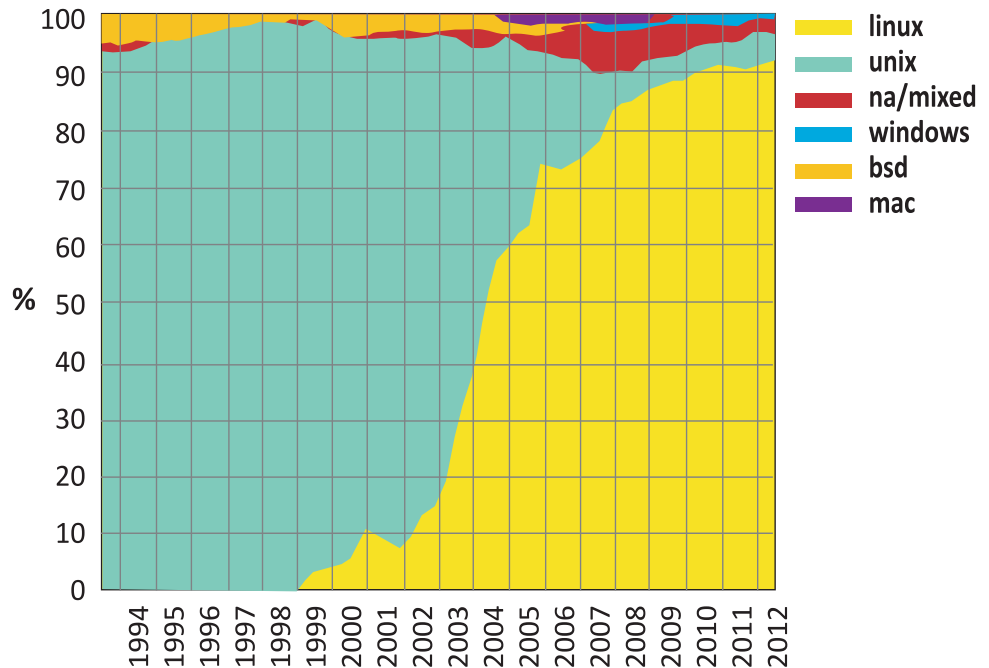

Figure 14.1: A graphical representation of the relative shares of the various popular operating systems on the top 500 supercomputers in the world (1994–2011).
(Source: WikiMedia Commons, *http://tinyurl.com/qxovbma*)

In the broader server industry, **Apache HTTP server** is the indisputable market leader. Similarly, as per the industry estimate, more than 65 per cent of web

servers in the world are running on Linux server distribution. Among desktop computers, **Microsoft's Windows operating system** is still dominating, though its market share has dropped with every version over the last decade. A large chunk of what Windows desktop version lost has been captured by **Apple's iOS**.

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

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

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

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

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

Activity 14.1

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

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

14.4 Open Source Software for CR

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

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

- Freedom to use and share the software

- Freedom to modify and adapt the software to their unique situations

- Free support from the global community, which is often seen as far superior to the paid support by a company providing a proprietary software

- Cost effectiveness

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

14.4.1 Sound Recording, Editing, Mixing and Mastering

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

Audacity

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

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

There are many plugins and add-ons you can find on the Internet to extend the functionality of Audacity, and use the software as you like. Figure 14.2 shows you what the interface looks like.
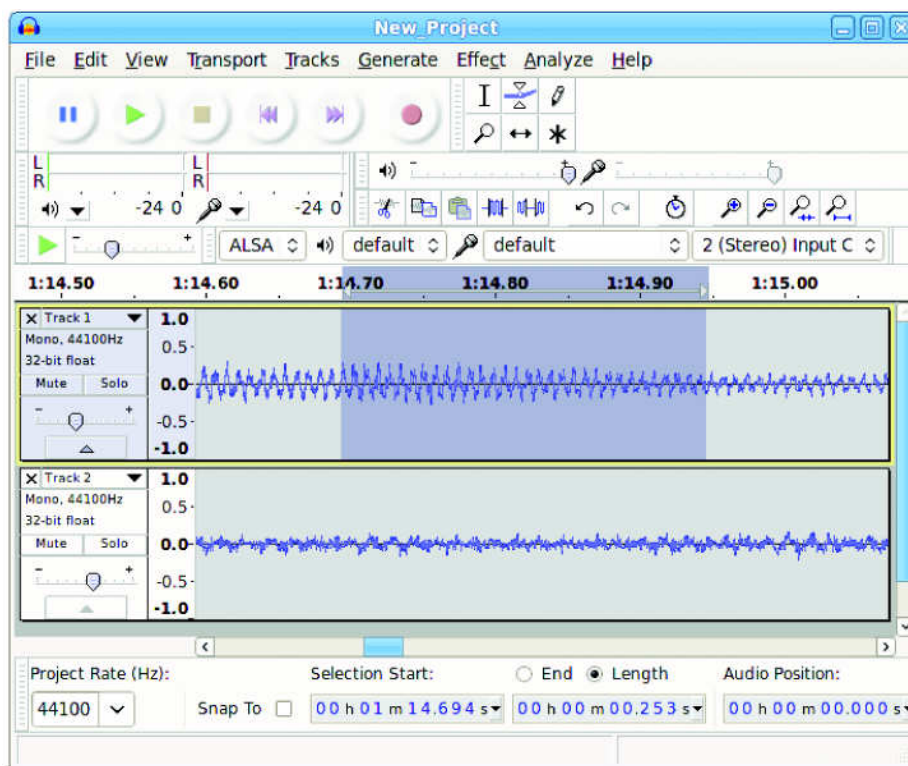

Figure 14.2: The Audacity editing interface.
(The screenshot was captured by the author)

Ardour

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

Figure 14.3 shows you what the Ardour interface looks like.
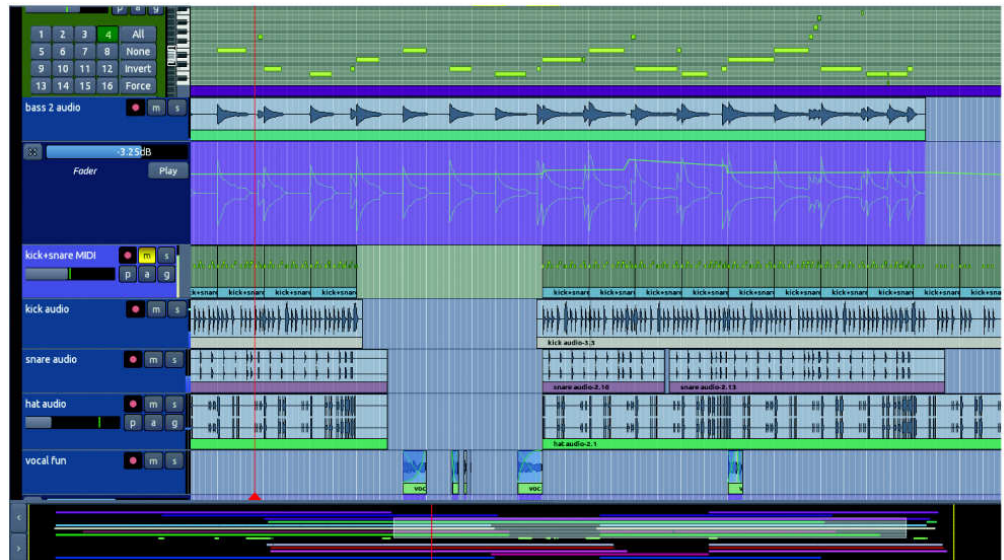
Figure 14.3: The Ardour software interface.
(The screenshot was captured by the author)

JAMin

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

The features of JAMin are as follows:

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

Figure 14.4 shows you the JAMin software interface.
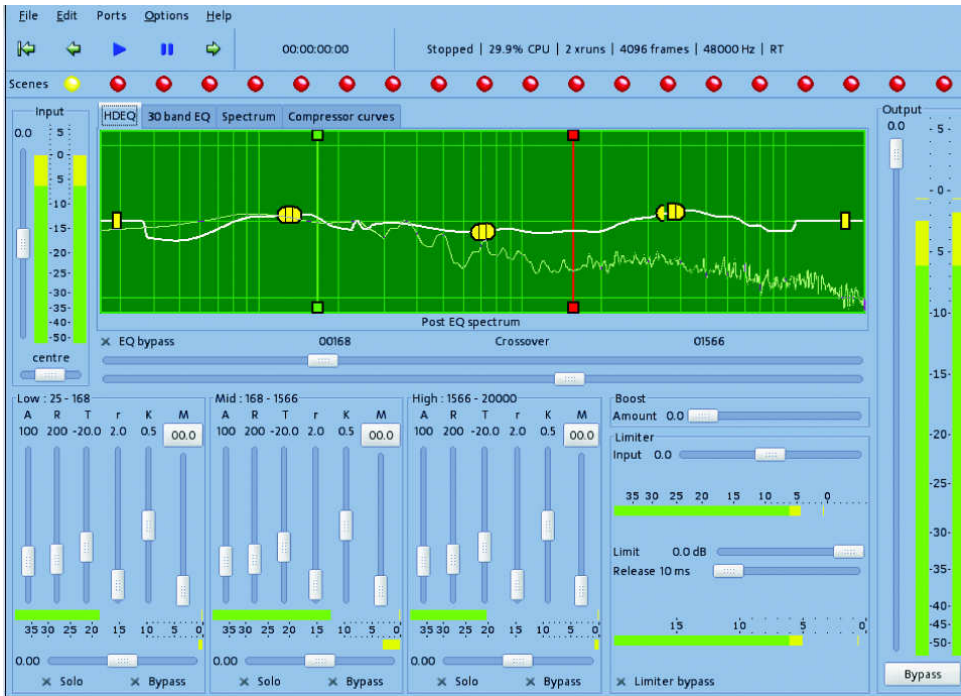
Figure 14.4: *The JAMin software interface.*
(The screenshot was captured by the author)

 Activity 14.2

Carry out a comparative analysis of the software listed below:

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

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

14.4.2 Radio Automation

Radio automation software are used for different functions in a radio station. They are used for **scheduling** (presetting the time when a particular programme plays), **logging** (listing the order of play of the aired programmes) and **playout**

(the playing of a preset list of programmes). Many radio stations in India use just a simple playlist software like WinAmp as playout system; but the use of a well-configured radio automation software can give a professional touch to a CR station, with pre-defined fades-in and fades-out for programmes, jingle management, advertising management, live assist, telephony and many features that come in handy in a radio station.

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

Airtime

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

To be able to install airtime, you need to be proficient in installing web server with PHP, Apache2, mail server and PostgreSQL. The installation requires some practice before you can master it. However, once it is installed, it is an easy software to use. The software does not run on your computer. You need to access it via an intranet or the Internet on your web browser like Mozilla, Chrome or Internet Explorer. As it is server based, the software for the end-user is cross platform. That means you can run any operating system—Windows, Mac OSX or Linux—and the software will be accessible in exactly the same way. Figure 14.5 shows you the Airtime media library and playlist builder interface.
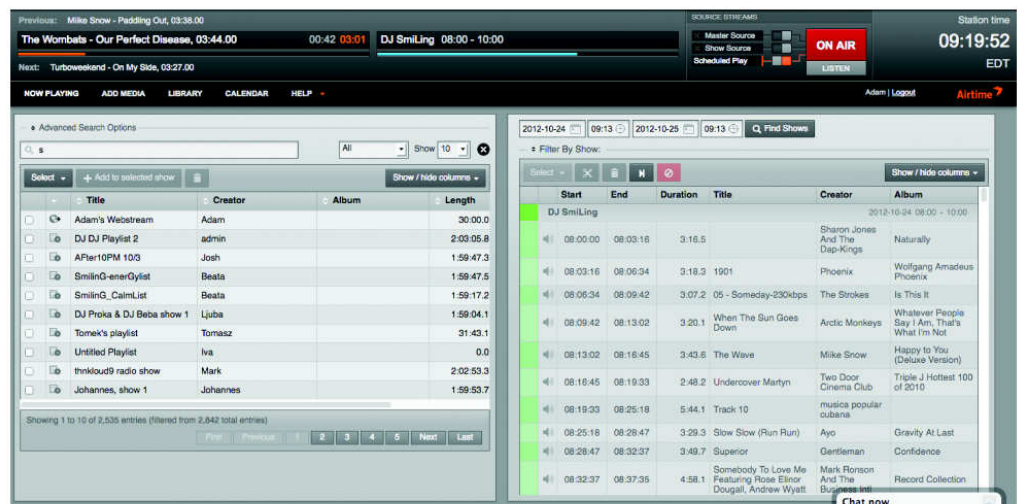

Figure 14.5: Source fabric's Airtime automation software: Media library and playlist builder (The screenshot was captured by the author)

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

Figure 14.6 shows the scheduling interface in Airtime.

Figure 14.6: *Source fabric's Airtime automation software: Schedule builder interface (The screenshot was captured by the author)*

Rivendell

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

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

One of the limitations of Rivendell is that its documentation is limited to few distributions of Linux, and that it does not have a repository-based installation service. Figure 14.7 shows a view of the Rivendell administrative panel, from which its primary functions can be controlled.
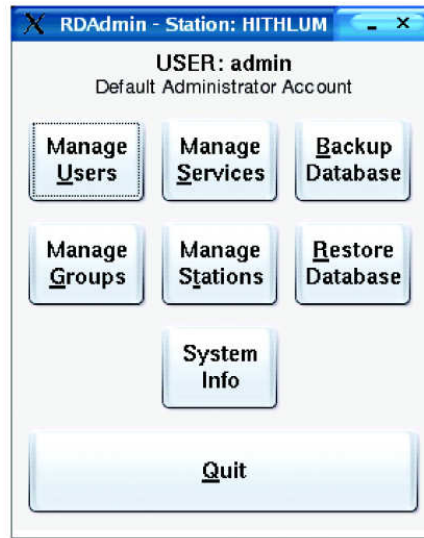
Figure 14.7: Rivendell software Administrative panel
(The screenshot was captured by the author)

Figure 14.8 shows a view of the Rivendell's built-in editing capabilities, which allow you to correct audio without moving into a separate software environment.
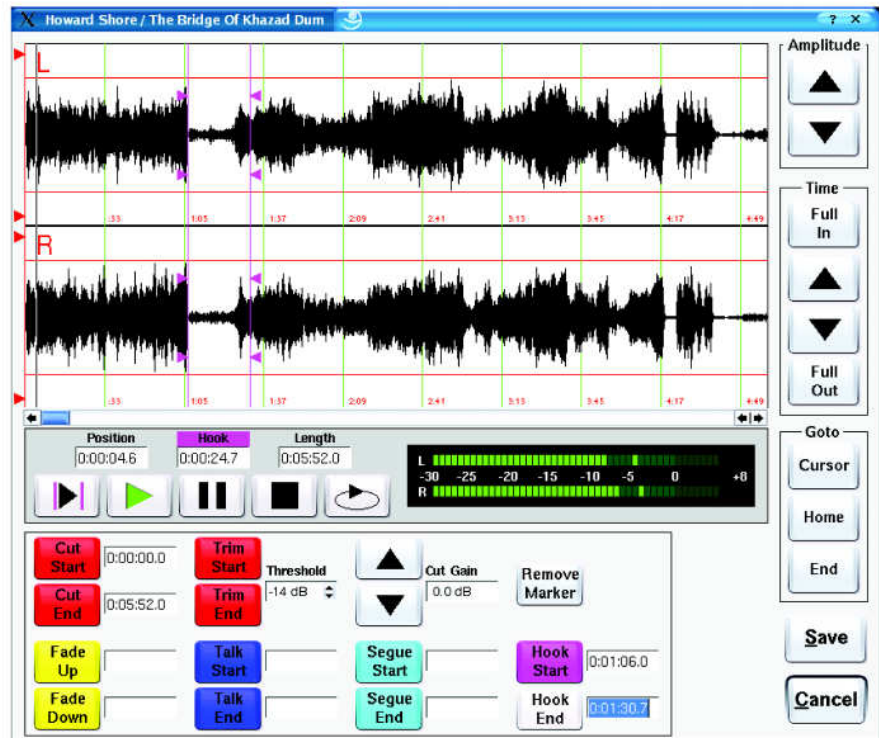

Figure 14.8: Rivendell software's built-in audio editor
(The screenshot was captured by the author)

GRINS (Gramin Radio Inter Networking System)

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

Some of the features of GRINS are as follows:

- **Full telephony integration** without need of a telephone hybrid or changes to mixer settings. You can receive calls, record calls, put calls on air, have live call-in programmes. You can even maintain a searchable database of your callers within the GRINS interface

- **Preview of audio**. Since GRINS uses multiple sound cards, you can preview audio over your headphones even while some other programme is playing live.

- **Streaming audio support**. GRINS supports streaming over IceCast or ShoutCast servers, making it Internet radio ready.

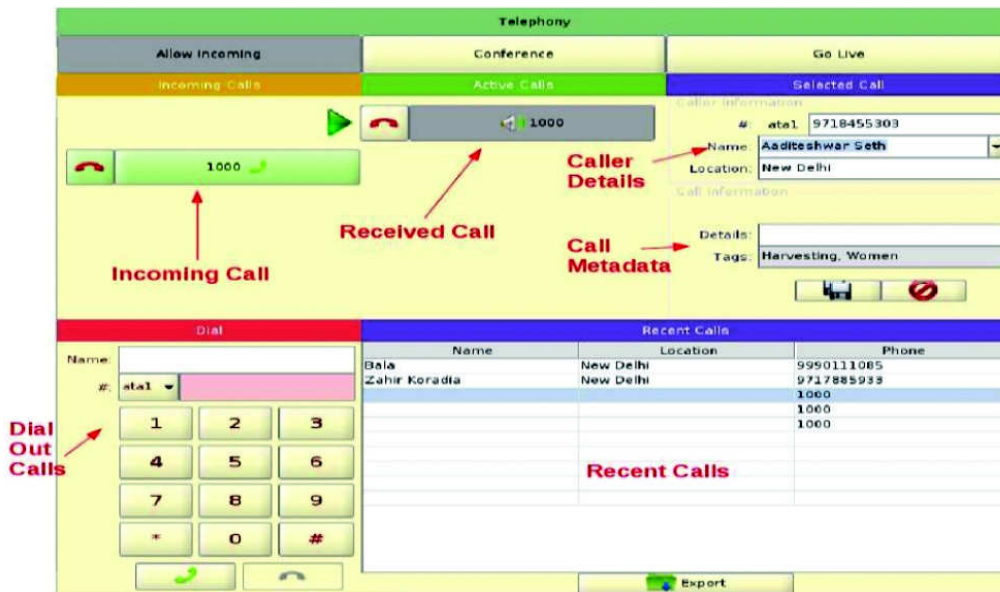Figure 14.9 shows you the GRINS telephony interface.

Figure 14.9: A screenshot of the GRINS telephony console.
As you can see from the image, there are provisions to make, receive and log calls.
(The screenshot was captured by the author)

Activity 14.3

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

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

14.4.3 Other Useful Software

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

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

The limitation of IDJC is that it works only on Linux and that too specifically with the JACK Audio Connection Kit. To use it, you need to be reasonably well versed with both associated software. Figure 14.10 shows you the IDJC interface.
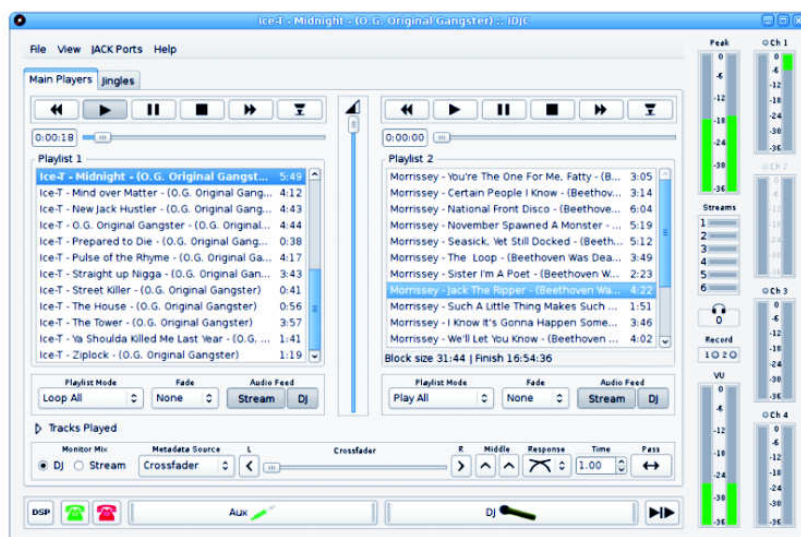

Figure 14.10: A screenshot of the IDJC interface.
You can play items from the two queued lists on the left and right alternately.
(The screenshot was captured by the author)

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

Activity 14.4

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

14.5 Let Us Sum Up

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

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

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

14.6 Model Answers to Activities

Activity 14.1

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

Activity 14.2

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

Activity 14.3

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

Activity 14.4

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

14.7 Additional Readings

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UNIT 15

Telephony for CR

Structure

15.1 Introduction

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

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

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

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

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

15.2 Learning Outcomes

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

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

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

15.3 Conventional Landline-based Systems

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

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

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

15.3.1 Speakerphone–Microphone Combination

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

The biggest drawback of this solution is that, more often than not, the sound quality is extremely poor. Quite often, the voice is barely audible on the receiving end, or the ratio of the sound to ambient noise or noise on the line is hopelessly adverse. This is mainly because of an impedance mismatch. The external

microphone and the speaker of the phone affect each other, as they both have their own magnetic fields.

Figure 15.1 shows a simple speakerphone unit.

Figure 15.1: *A common speakerphone unit. Note the speaker grille under the handset. (The screenshot was captured by the author)*

15.3.2 Transformer-based Interface

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

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

15.3.3 Telephone Hybrids

In commercial radio stations, the telephone-based programmes are generally conducted through what is known as telephone hybrid instrument. Simply

speaking, a telephone hybrid is a sophisticated telephone instrument that is capable of routing the sound signals, incoming as well as outgoing, to a mixer console. It works exactly in the same manner as a normal telephone instrument. You can insert one or more telephone lines in it, and can route the audio to the mixing console without any impedance mismatch, distortion or echo.

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

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

 Activity 15.1

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

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

15.4 GSM/CDMA Solutions

With the arrival of mobile phones, the communication landscape has changed dramatically. In this era of rapid spread of mobile phones, the CR stations could not afford to ignore the integration of mobile phones with the broadcasting environment. Today, a mobile phone is being used as an instrument to provide live feeds from a distant event, to engage listeners in conversations that can be

heard on air, or simply to solicit comments and feedback from listeners on a real-time basis. In addition, the use of SMS as an instrument for conducting surveys has also become very popular. Here, we will examine various methods of integrating mobile phones in your broadcast environment.

15.4.1 Bluetooth Interface

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

- A Bluetooth-enabled mobile phone pairs up with the Bluetooth sound interface.

- The Bluetooth sound interface has a slot to insert a microphone, and a port to take a line out connection.

- Insert a microphone in the Bluetooth interface.

- Take a line out from Bluetooth interface to your mixing console.

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

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

15.4.2 GSM Dongle (Modem)

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

In this solution, you can purchase any GSM dongle marketed by the mobile phone companies: TATA Docomo, Vodafone, Idea, BSNL or Airtel. Almost all others have a branded dongle which works with their SIM (Subscriber Information Module) cards. You can buy any one of them depending on the availability of the network in your studio area. However, be careful when you choose the dongle: all dongles are not voice enabled. This means that on some of them, you can access the Internet but not use them to make or receive voice calls. When you buy one, ensure that the dongle is voice enabled. Figure 15.2 shows you a typical GSM dongle.

Figure 15.2: *A typical GSM dongle unit.*
HSDPA stands for High Speed Data Packet Access, a type of high speed mobile connectivity protocol. (The screenshot was captured by the author)

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

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

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

Activity 15.2

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

15.5 Voice Over Internet Protocol (VoIP)

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

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

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

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

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

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

CR stations using GRINS as a play-out software (see Section 14.4.2 in Unit 14 of this module) can make calls directly out of their radio automation system, with the help of a built-in Asterisk server, which uses VoIP protocols. The GRINS user can also send and receive SMS from the same interface.

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

After installing Asterisk, you need to provide the server a GSM or PSTN gateway to start making calls to the outside world. Many hardware options are available in

the market today to configure a gateway. Companies like Digium and Cisco are well-known for their gateway products for professional uses.

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

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

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

Activity 15.3

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

15.6 Use of SMS

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

15.6.1 User Case Scenarios for SMS

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

- **Listeners' feedback:** It is important for any CR to know the minds of its listeners. One of the ways to get listener feedback is to encourage them to send SMS to the radio station, which the station can compile and incorporate into a feedback mechanism.

- **Live participation:** SMS is also a useful tool to encourage listener participation in programmes. One of the important aspects of any CR station is the participatory nature of the medium. In a live programme, a radio jockey or announcer can encourage listeners to participate in the programme with their text messages, which can be read out during live broadcast. You can even do request shows using SMS-based requests.

- **Surveys and polling:** Many radio stations use SMS as a tool for surveys or to understand public opinion on a variety of subjects. For example, in a particular village, if a road has been newly paved, the CR announcer can ask the members of the community if they are satisfied with the quality of the road by sending an SMS with "YES" or "NO" to the station's number. This feedback could be compiled to present an opinion.

- **Competitions:** Many CR stations have used SMS as a voting tool for community-based competitions. For example, a CR station called Radio Bundelkhand in Central India ran a competition to showcase singing talent from the local communities. New singers were given a chance to sing on radio, and the community members were asked to vote for the best singer through SMS.

15.6.2 Technological Options

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

- The simplest method is to buy an **average mobile phone with a SIM card** of your choice to send and receive SMSs for your radio station. However, it may be noted that it may not be very comfortable or fast to use a normal mobile phone for sending bulk SMSs for your radio station, or even receive large volumes of SMSs in response. However, as a trial or beginning, it may not be a bad idea to use a phone.

- You can even use **GSM modems**, discussed in Section 15.4.2. These modems are generally provided with a software to send SMS. If your modem has been manufactured by a company like Huawei, it will have bundled software like Mobile Partner or Mobile Connect. Both these types of software can send and receive SMS.

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

  http://www.160by2.com

  http://www.ofsms.in

  http://www.mysms.com

- Most of these services let you send free SMS. Given the fact that they are free of cost, they have certain limitations. Some of these free services have a daily cap on the number of SMSs you can send, while some will let you send only 140 characters instead of the stipulated 160. (The service provider may use 20 characters to piggy-back advertisements along with your messages!)

- There are also specialized **software for bulk SMSs,** which are generally used by advertisers. These software have multiple abilities to send or receive SMS based on region, time or context. However, most of them are not free, and may cost a lot of money. You may select one of these software if your radio station can afford the cost.

- An excellent option is the free and open source software **Frontline SMS**, which is being extensively used the world over by many radio stations. This software is browser-based, and works with a large variety of mobile phones and GSM modems. (Go to www.frontlinesms.com to see a complete list of supported phones and GSM modems. It may be a good idea to consult the database before buying a phone or a modem for the purpose or installing the software on your computer.) The software can handle bulk SMSs with ease while keeping a meticulous database of all the SMSs you send or receive, also allowing categorization and time-based search. It also maintains a list of your contacts. Figure 15.3 shows you the main window of the Frontline SMS software.
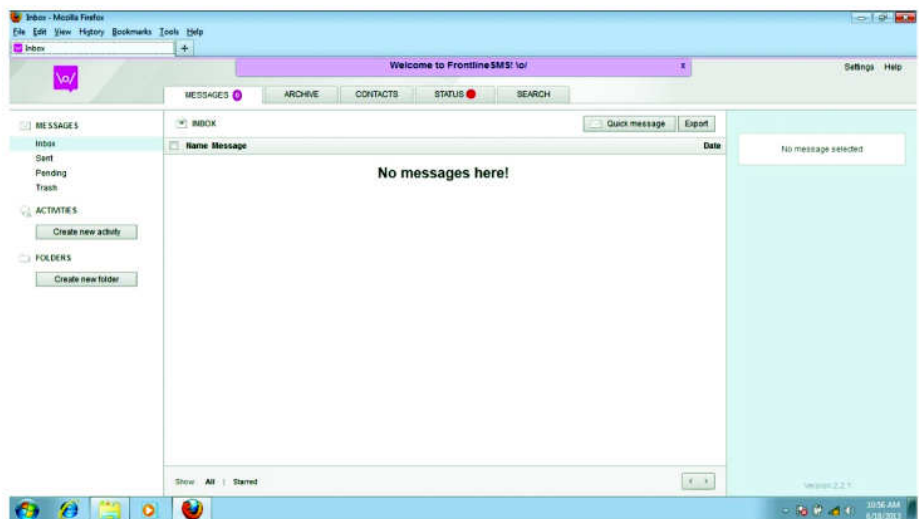

Figure 15.3: *The primary interface for Frontline SMS*
(The screenshot was captured by the author)

Caution

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

Figure 15.4 shows you the contacts window, through which you can look up the details of the contacts stored within FrontlineSMS.
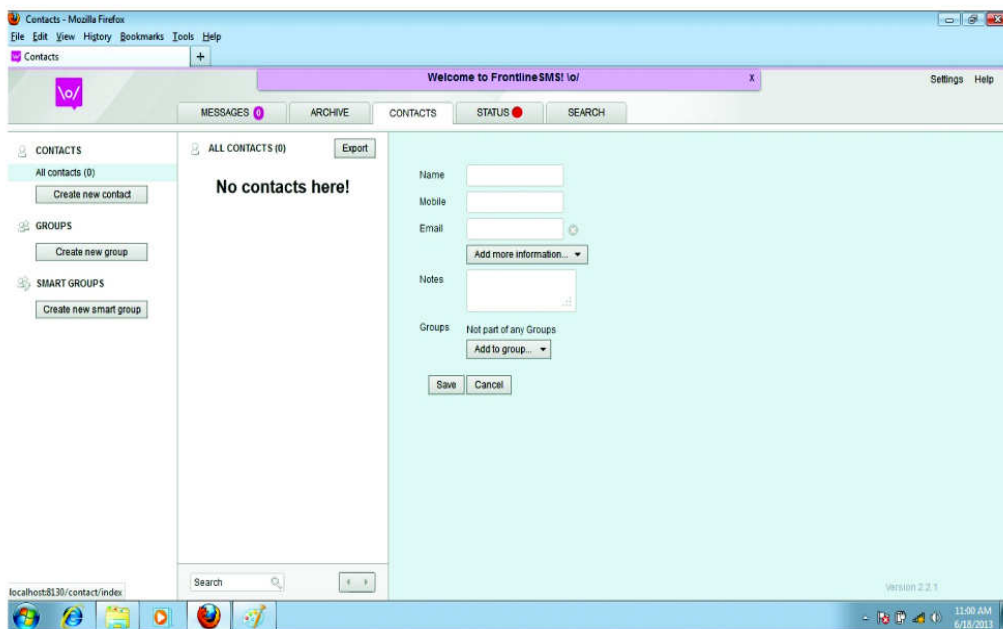

Figure 15.4: *The contacts window of Frontline SMS*
(The screenshot was captured by the author)

Activity 15.4

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

- Use three online SMS services and do a comparative analysis of their pros and cons.

15.7 Let Us Sum Up

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

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

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

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

15.8 Model Answers to Activities

Activity 15.1

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

Activity 15.2

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

Activity 15.3

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

Activity 15.4

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

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

 15.9 Additional Readings

- International Engineering Consortium. (2007). Voice over Internet Protocol. Definition and Overview. Retrieved from http://www.teliqo.com/voip/

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 Glossary

Unit 14: Audio Hardware and Field Recording

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

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

Unit 15: Audio Hardware and Field Recording

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

Unit 16: Telephony for CR

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

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