Community Radio

Technical Manual

This Manual Belongs To:

Radio...

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For further information, or permission to reproduce any of the materials in this information file, please contact:

National Community Radio Technical Department
Phone: (011) 4034336
Fax: (011) 4034314
e-mail: ncrf@ncrf.org.za or info@ncrf.org.za

Open Society Foundation for SA Media Support Programme
Phone: (021) 6833489
Fax: (021) 6833550
e-mail: admin@ct.osf.org.za

National Community Radio Forum

The National Community Radio Forum is a national membership-driven association of community radio stations and support service organisations. Radio station members are independent, non-profit community-based organisations, owned and run by diverse local communities who actively participate in the development of programming activities, for sustainable non-discriminatory local development.

Open Society Foundation for South Africa

The Open Society Foundation for South Africa is committed to the values, institutions and practices of an open, non-racial and non-sexist, democratic, civil society. It will work for a vigorous and autonomous civil society in which the rule of law and divergent opinions are respected.

Communication Assistance Foundation

The Communication Assistance Foundation aims to strengthen freedom of expression and media diversity in developing countries through support for free and independent media.

About the Contributors

Martin Sims wrote the text and drew the diagrams. Martin is a former station manager of Matie FM, a community radio station based in Stellenbosch in the Western Cape. He has worked for Open Society Foundation for SA as a consultant to the Media Programme and has trained community broadcasters all over South Africa in operating and maintaining studio equipment. Martin has a degree in Electronic Engineering and currently works as a studio designer and trainer for a broadcast equipment supplier.

Linda Mazibuko contributed text for Section 4 of the manual. Linda is the technical manager of the National Community Radio Forum. He has an extensive background in community radio in South Africa, doing broadcast and production studio design and installation, as well as maintenance and operational training with community radio staff and volunteers. Linda has qualifications in telecommunications management.

Jean Fairbairn edited the text, and with Mabalane Mfundisi, co-ordinated production of the manual. Jean is Director of the Media Programme of the Open Society Foundation for SA. The programme supports the community radio sector in South Africa through grants for studio equipment, training and programme production.

Mabalane Mfundisi is Chief Executive Officer of the National Community Radio Forum. Mabalane has a long background in community radio. He began as a broadcaster and later became manager at Durban Youth Radio in KwaZulu/Natal. In 2000 Mabalane was appointed CEO of the NCRF, which is the largest representative body of community radio stations in South Africa.

Acknowledgements

Contents

Introduction

Many people have initiated or are advocating technical training programmes for community radio. The starting point for many of these programmes is to teach basic electronics. This is not the approach of this manual.

We believe that the most important thing is basic operational training. Many people are able to make good tea or coffee, but most of us don't understand the resistive element that heats the water.

The same applies to the community radio studio. You can make good programmes by using broadcast equipment efficiently and creatively, but you don't have to understand the complex integrated circuits and resistors inside the mixing consoles. An understanding of what the equipment does and how to use it, and the confidence to experiment, is really all you need to start producing good quality sound.

The machines in a radio station rely on people to operate them, and so they can only be as good as the station's staff and volunteers. A community radio station is a complex environment, with many departments, all depending on each other. To make good radio, every department needs to focus on its individual task, while never losing sight of the station's overall mission.

The technical department is no exception. The station depends on the technical department to keep broadcasting sound. At the same time, if the technical department is to succeed in keeping the equipment working, there must be good governance and management, and all staff and volunteers must be committed to the mission of the station.

Aim of the Manual

The National Community Radio Forum (NCRF) and Open Society Foundation for South Africa (OSF-SA) have developed this manual as a resource to help community radio staff and volunteers. It can be used as a reference for technicians as they go about their day-to-day work, cleaning the studios and doing repairs. It can be used to train new people in the programming department - the many producers, newsreaders, presenters and DJs who will operate the studios. It can be used if you need ideas about how to develop your studios further, especially in relation to new digital technologies. We've included prices and costs so that the manager can use the manual to budget for repairs and replacements.

The manual also aims to demystify studio equipment - to help you understand how to use your studios properly and to give you the confidence to experiment. We hope to expand your technical knowledge and vocabulary and - who knows? We might even attract some readers to want to learn more and take up a future in electronics.

The manual provides information about:

- The broadcast system

- Studios at a radio station, and how to link them

- The types of equipment you are likely to find in a community station

- What the equipment in the studios can do

- Commonly used technical language and concepts

- Basic maintenance, fault-finding and problem solving.

How To Use the manual

The manual is divided into a number of sections. Each new section builds on the previous section to give you a more detailed picture of the radio station and how it works.

Different Parts of the Station

The first four sections introduce the broadcast system and different parts of the station. We recommend that everyone who is involved in planning and developing a community radio station; or who is involved in the station on a day-to-day basis; or who has a regular broadcast slot, reads the first five sections.

Fault-finding, Maintenance and Budgeting

Section 5 deals with studio maintenance and fault-finding. It is very important that everyone who uses or works with the studios should read this section. It is especially important for the manager and finance manager to read this section, because of the budgets and other financial information we've included, and because the manager is ultimately responsible for keeping the station on air.

A-Z

The A-Z reference section will be of special use for technicians, but it will also be useful for others wanting more detailed information about the different pieces of equipment found in studios, and how they work. The A-Z will also help you understand many of the words and terms used by broadcasters, sound engineers, equipment suppliers and others involved in the broadcast industry.

Helpful Tips

Throughout the manual, there are helpful tips for using and maintaining equipment. We use this symbol to highlight tips and other advice we think will be useful.

Cross-references

We use this symbol when we refer you to the A-Z for more detailed information.

We use this symbol when we refer you to another part of the manual for more details.

Warnings

We use this symbol to warn you about actions or other things that can damage or destroy your equipment, or in any other way harm you or your station's operations.

Diagrams

There are many diagrams in the manual, to support the text.

Blank Pages

There are a number of blank pages in the manual. We have left these open so that you can write notes and make your own additions to the text. Every station's studios will be different, and you may want to note where our text and your studios differ, or to jot down any other additional information.

Adding to the File

We've used a file so that technicians can slot in their own notes and additional information as needed. As time goes by you will want to add information to the file, to make it more comprehensive and to keep it up to date.

Prices and Costs

We've included prices of some of the key pieces of equipment needed for broadcast and production. Most radio equipment is imported and so prices are subject to changes in the exchange rate. We've calculated as follows:

1 \$ = R10

1 UK £ = R14

This was the exchange rate in about October / November 2001. When you are finding out about new equipment, and see prices quoted in foreign currencies, you must remember to check the exchange rate and work out the latest prices.

The prices provided serve as a guideline for community radio stations wanting to buy good or adequate equipment. Obviously more expensive or cheaper equipment is available, but it is not always necessary or appropriate to buy the most expensive equipment. Buying the cheapest equipment is also not a good idea, as it will wear out faster.

Appendices

There are five appendices:

A. A list of some of the abbreviations used in the text, and their meanings.

B. This appendix explains the prefixes added to units to denote size.

C. A directory of South African broadcast equipment suppliers.

D. A list of useful Internet resources and Web addresses. The Internet was an important resource for us in writing the manual.

E. A list of useful references, including books and resources we drew on to write the manual.

Section 1:

How Does It All Work?

How does the voice of a presenter in the studio reach a listener in the community?

To answer this question, we must look at how all the different parts of the station work together to make radio possible. We'll begin with a broad view, and fill in the details as we go along.

The Broadcast System

Broadcasting is the distribution – or transmission – of sounds and / or pictures using electrical signals, so that they can be heard and seen over a large area. We call the combination of equipment used to make broadcasting happen a broadcast system.

The first task of a radio broadcast system is to change sound into electrical energy.

This electrical energy is called an audio signal. We change sound into electrical audio signals because these signals can be changed in almost unlimited ways by using electronic equipment. In audio electronics, pieces of equipment that change energy from one form to another are called transducers.

The second job of a radio broadcast system is to process or change audio signals so that we can broadcast them. Pieces of equipment that change one or more aspects of an audio signal are called signal processors.

See how we use these terms in the diagram below:

Model of a radio broadcast system
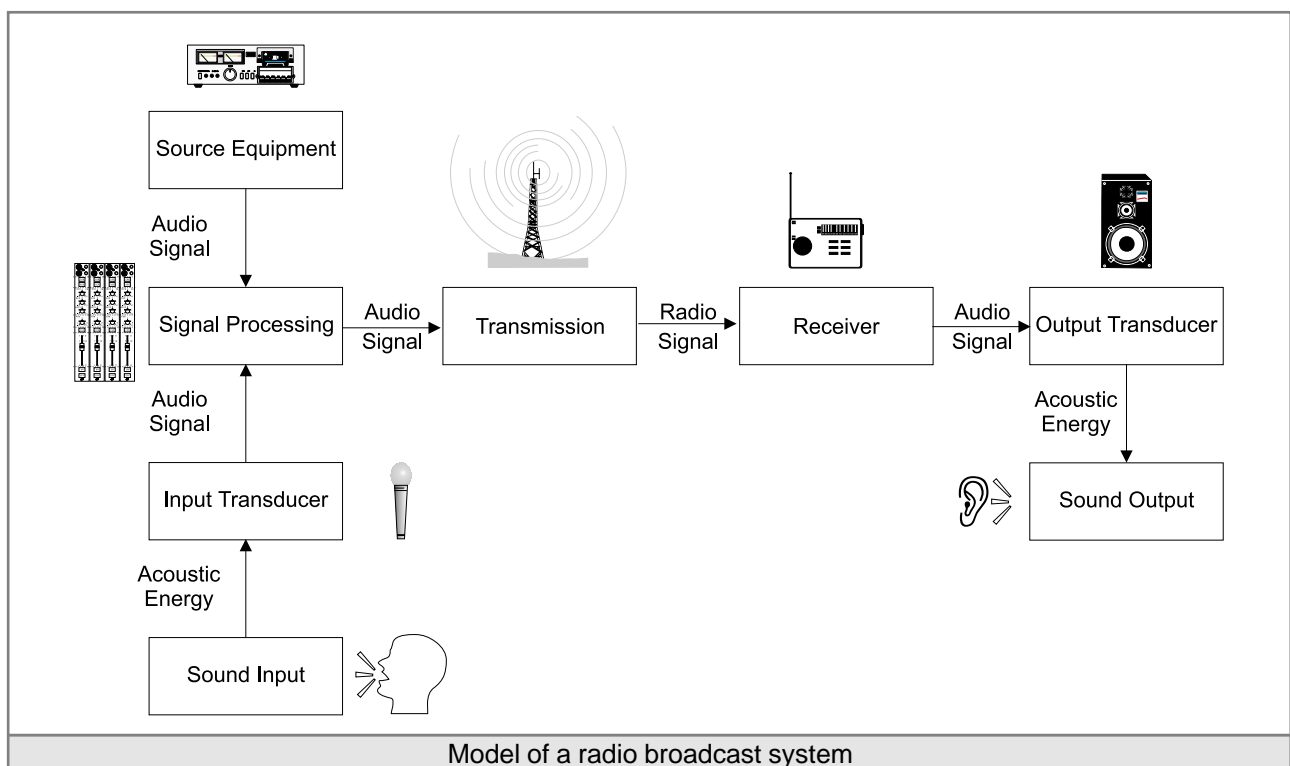
An input transducer (most often a microphone) changes the sound of a voice into a varying electrical signal, a voltage or current that is an exact representation of the sound. This varying voltage or current is called an audio signal.

The microphone is not the only piece of equipment in the broadcast system that produces audio signals. There are many other pieces of equipment, such as compact disc players, minidisc players, tape decks and turntables in the broadcast system, all of which produce audio signals. We call them source equipment. We use signal processors to change one or more characteristics of the audio signals produced by the source equipment..

One of the most common purposes of a signal processor is to change the size of a signal. Most often the signal is made "bigger". This is called amplifying a signal. Once the amplified signal is converted back into sound, it is much louder than before.

In the diagram of a radio broadcast system on the previous page, the block marked signal processing represents several pieces of equipment, such as pre-amplifiers, mixers, power amplifiers, effects units and so on.

In the diagram, the block marked transmission represents the transmitters, antennas and so on that produce the radio signal that your radio set – or receiver – picks up. The processed audio signal is carried to your radio as a radio signal. The process of joining the audio signal and the radio signal is called modulation. This is the origin of the abbreviations FM and AM, which stand for frequency modulation and amplitude modulation.

Your radio, or receiver, receives the radio signals, which have been broadcast by the transmission equipment. The receiver works in the opposite way to the transmission equipment. It turns the radio signal back into an audio signal by demodulation.

The radio set sends the audio signal to an output transducer - loudspeakers or headphones. These output transducers work in the opposite way to the input transducers (microphone and other source equipment) at the start of the system. The output transducers convert electrical audio signals back into sound that we can hear.

The radio station's technical responsibility is to produce a good quality radio signal that listeners can pick up with their receivers. This means delivering the best possible quality audio signal to the transmission block, and ensuring that the transmission block converts it into a radio signal of good enough quality to reach radio receivers in the community.

Section 2:

The Studios and Transmission

In this section, we look at the broadcast system – the studios and transmission equipment – in greater detail.

The Studios

The station's broadcast system can have more than one studio. These are:

A Broadcast Studio or On-air Studio

Every station must have a broadcast studio. This is the studio that produces the sound that is broadcast.

A Production Studio

This is a second studio, where programmes can be recorded, or pre-produced, for broadcasting later. A production studio is not essential, but as the broadcast studio is in constant use while the station is on air, having a second studio where recordings can be made and material can be pre-produced is very useful.

The production studio can also be used as a back-up broadcast studio. This means that the station will not have to go off-air while the broadcast studio is being repaired or maintained.

Connecting Your Studios

When a station has more than one studio, these are connected together so that audio signals can be shared between them.

The Studio Environment

The environment of your studios must be carefully controlled. Broadcast equipment is sensitive and expensive. If it is not properly housed and looked after, it can easily be damaged and will wear out quickly.

Air Conditioning

Studios are air-conditioned to control the temperature, as broadcast equipment works best when the temperature stays constant. Air conditioning will also create a comfortable environment for people working in the studio. Because studios are acoustically treated, there is often no other ventilation.

Acoustic Treatment

To produce quality sound, the studio room is acoustically treated and soundproofed. This means that acoustic panels are mounted on the walls to absorb unwanted sound, and the doors and windows are specially sealed to stop unwanted noise coming into or getting out of the studio.

Studio Furniture

Radio studios usually have specially built furniture. The furniture is designed around the equipment so that all of the equipment and wiring is secure.

Furniture is also designed to take into account the position of the different pieces of equipment, placing them in the most comfortable position for the studio operator.

Good furniture should be able to withstand a lot of wear and tear and still look attractive. The on-air studio is the station's showpiece. To impress visitors and to encourage a sense of pride in what your community has achieved, it is important that the furniture (and the whole studio!) looks good.

Lastly – but most importantly for the technicians - the furniture must allow easy access to the back panels of all the equipment for repairs and maintenance.

Equipment Racks

The furniture will have one or more equipment racks. Most professional equipment is supplied in standard cases. The size of the case is standardised so that many different pieces of equipment can all be mounted in the same standard size equipment rack.

The standard "rack-mount" equipment case is 48.3cm (or 19 inches) across but can be various heights and depths, depending on the type of equipment. Take a look at the diagram below.

The height of the case is specified in rack unit's (U). A 1U case is 4.4cm high; a 2U case is twice that; a 3U case is three times that height, and so on. Because rack units are often the same size and the same colour, the equipment in the studio can all look the same to an unpractised eye.

44 mm

88 mm

483 mm

1U and 2U rack mountable equipment cases
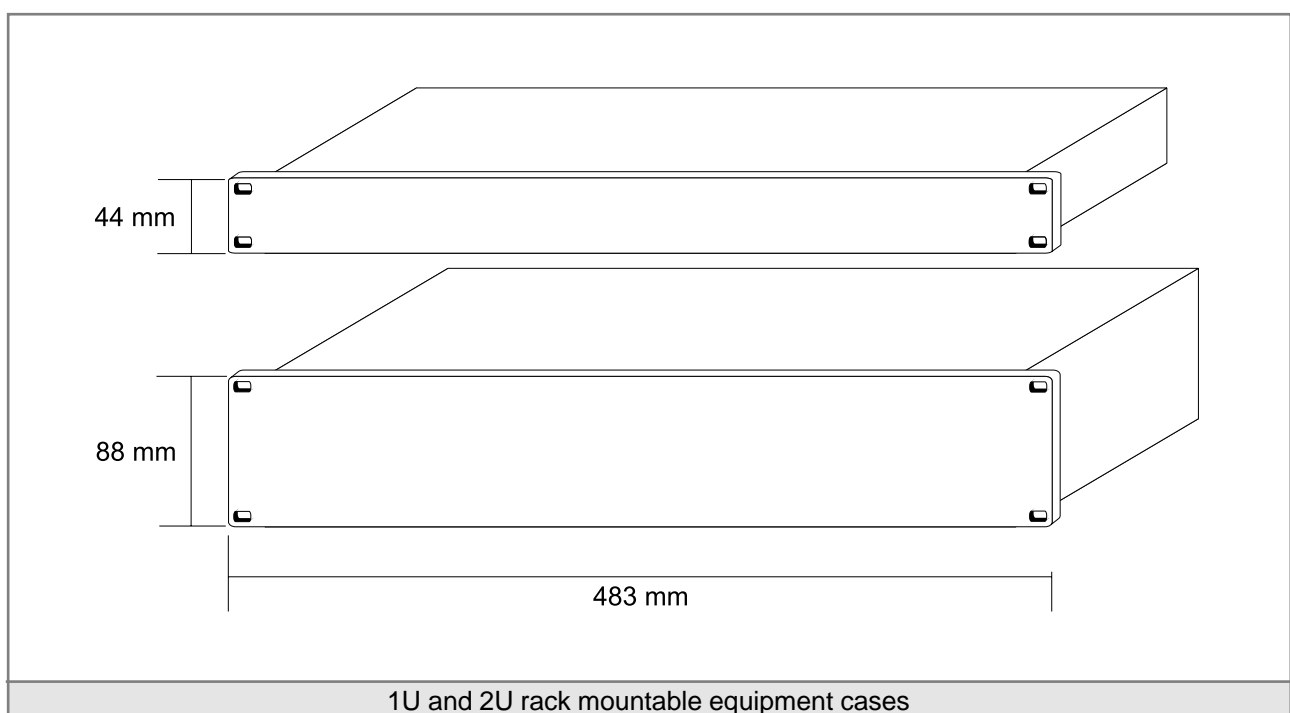
The Broadcast or On-Air Studio

The diagram below shows a model of a typical community radio on-air studio. By following the arrows in the diagram, you will be able to see how sound (as audio signals) moves through the different pieces of equipment, and how the equipment is linked together.

Input/Output
Transducers

Source Equipment

Signal Processing

Final Processing

Turntable

Phono
Preamplifier

Microphones

Microphone
Preamplifier

Antenna

Input
Transducers

Telephone
Hybrid

Receiver

Satellite
Receiver

CD Player

Video
Recorder

Audio from
another studio(s)

Mixer

Control Signals
to "On-Air" Lights

Headphones

Compressor/
Limiter/
Processor

Transmission

Output
Transducers

Recording
Equipment

DAT Player/
Recorder

Distribution
Amplifier

Signal to other
Studio(s)

Cassette Player/
Recorder

Reel Player/
Recorder

Amplifier

Loudspeakers
(Studio Monitors)

Computer
Player/Recorder

MD Player/
Recorder

Recording Output from Mixer

| Sound | Audio | Audio | Audio |
| Waves | Signal | Signal | Signal |

———— Mono Audio Signal

━━━━ Stereo Audio Signal

Model of a typical broadcast studio

Broadcast Studio Equipment

All studio equipment should be rugged (strong) and reliable, with large displays and control buttons. The equipment in a community radio studio is in constant use and is handled by many different people who have different levels of technical experience. Broadcast equipment works hard – often 24 hours per day. Even in a very well maintained studio no piece of equipment will last forever. But if you make the right choices in the beginning, and look after your equipment well, it will last a lot longer.

Now look at the diagram of the broadcast studio on the previous page and follow the flow of audio signals, while reading the text. As you will see, we start with the first input transducer, the microphones.

Microphones and Microphone Stands

The broadcast studio should have at least three high quality microphones, or mics. This provides a microphone for the presenter and two more for guests in the studio.

It is impossible for someone to hold a microphone when they are on-air, so each microphone needs its own microphone stand. The stand will keep the microphone in place, but it must also be adjustable so that its position can be changed to suit different presenters and guests (who may be tall or short and who will have different speaking voices).

A good quality broadcast microphone, with a stand, costs between R3 500 and R8 500.

Under M in the A-Z for more on mics.

CD Players

The broadcast studio should have at least two broadcast quality CD players. Two CD players are necessary to make it possible to mix from a song on a CD played in the first one, to a song on a different CD in the second player.

Broadcast quality refers to some of the extra features that distinguish studio CD players from home CD players. The studio CD player should have a large easy-to-read front display. Studio CD players should be very reliable, as they will be in constant use.

You should be able to control the CD players from the mixing desk, or mixer. Broadcast CD players can be started and stopped from the mixer.

The CD players should also have balanced professional level outputs. This is best suited for connection to a professional mixer.

Broadcast CD players range in price from R4 500 to R7 000 per player.

Under B, Balanced and Unbalanced Wiring, in the A-Z for more about balanced professional level outputs.

Under C in the A-Z for more on CDs and CD players.

Cassette Player/Recorder (Tape Deck)

Most people still use cassette tapes to make and share recordings, so the broadcast studio should have a tape deck to play cassettes.

Broadcast quality tape decks are often hard to distinguish from high-quality home tape decks. The most obvious difference is that the studio tape deck will be built into a rack-mounting case. The studio tape deck might also have balanced inputs and outputs.

A broadcast tape deck costs between R4 500 and R9 000.

Under C in the A-Z for more on cassettes and cassette player/recorders.

Minidisc Player/Recorder

Minidisc (MD) is replacing cassette as a recording medium in broadcasting. MDs are played much like CDs, as the sounds on the discs are divided into tracks. But recordings can be made onto MD just as easily as onto cassettes.

For this reason, MD players are ideal for recording and playing out jingles and adverts. MD like CD, is a digital medium. Recordings on MD are of similar quality to CD and are much better than tape.

A broadcast quality MD player costs between R6 000 and R30 000.

Under M in the A-Z for more on MDs and MD players.

Telephone Hybrid

A telephone hybrid makes it possible to talk to callers on-air. It links the studio to the telephone network, and is an essential item in the community radio broadcast studio as it enables community members from far away to call in and air their views.

You choose your telephone hybrid by thinking about how many callers your station expects to handle at the same time. A single line hybrid allows you to speak to only one caller at a time. A multi-line hybrid enables you to speak to more than one caller.

We recommend a multi-line hybrid for community radio stations to broadcast effective call-in programmes and competitions.

Telephone hybrids range in price from about R4 000 for a single line hybrid, to R25 000 for a high quality multi-line hybrid.

Under T in the A-Z for more about telephone hybrids.

Mixing Desk (Mixer)

The mixing desk – or mixer – is the centre of the broadcast studio. Every piece of sound equipment in the studio is connected to the mixing desk.

The mixing desk is used to control the sound from each piece of source equipment. It is the tool that allows us to combine – or mix – the sounds from microphones, CD players, MD players and other source equipment together.

The choice of mixer will determine how your studio is built and how it will work. Because each piece of equipment in the studio connects to the mixer, it is vital that your mixer has enough connections.

A good quality analog broadcast mixer will cost from R30 000 upwards, depending on the number of connections and other features you want it to have.

Under M in the A-Z for more on the broadcast mixing desk.

Monitor Loudspeakers

In the broadcast studio, we want to hear what we are broadcasting at any time so that we can be sure listeners are receiving good quality sound.

To be able to hear the sound that is being broadcast, the studio must have a set of high quality loudspeakers. These are called monitor loudspeakers or monitors, because we use them to monitor what we are broadcasting.

A set of good quality monitors will cost between R4 000 and R9 000 per pair.

Under L in the A-Z for more about loudspeakers.

Audio Amplifier

The monitor loudspeakers must be fed a high power audio signal to produce sound. We use an audio power amplifier to increase the power of a signal from the mixing desk so that it can be fed to the monitors.

A separate audio amplifier is not always necessary to drive the studio monitors, as some monitors have amplifiers built into them. These monitors are often more expensive.

The price range for an audio power amplifier is between R2 000 and R4 500.

Under A in the A-Z for more about amplifiers.

Video Recorder and Off-air Receiver

It is a legal requirement for all South African radio stations to record their broadcasts. These recordings are sent to the Independent Communications Authority of SA for monitoring.

Most stations record their broadcasts by using a radio receiver (or tuner) and a good quality domestic video recorder. The receiver (tuner) is tuned into the station's broadcast frequency, and the audio from the tuner is fed to a video recorder.

A video recorder is used because the tapes can store much more sound than standard cassette tapes or CDs.

The combination of a video recorder and off-air receiver costs between R2 000 and R6 000. Some equipment suppliers suggest buying professional off-air tuners. While these are ideally suited for use in the studio, they are very much more expensive, and can cost up to R7 000 on their own.

Under V in the A-Z for more on the video logging system.

Distribution Amplifier (DA)

A distribution amplifier (DA) is used to distribute a signal or signals to many pieces of equipment. If you look at the diagram of a broadcast studio you will see how the DA feeds audio signals to the studio's recording equipment (the MD Player / Recorder, the Cassette Player / Recorder and other pieces of equipment).

The DA is a vital piece of equipment. It costs between R3 000 and R5 000.

Under A in the A-Z for more on amplifiers.

Headphones

The broadcast studio should have at least as many pairs of headphones as microphones (usually 3), and perhaps a few more for guests.

Headphones are vital in the studio, because when a microphone is used, the monitor loudspeakers will automatically switch off. They are meant to do this to prevent acoustic feedback (usually a loud whining noise) that can interfere with the broadcast.

Because the monitors switch off, the presenter and any guests must to use headphones so that they can hear themselves. Studio headphones must be particularly hardy, as they generally receive a lot of rough treatment.

A pair of well-made, high-quality studio headphones cost between R600 and R1 200.

Under F in the A-Z for more about feedback.

RAG Light or On-Air Light System

RAG or on-air lights are essential for the studio. They tell people inside the studio – studio operators, guests – that the studio is "live", or on-air. They also warn people outside the studio not to enter as they will interrupt the broadcast. So RAG lights are usually placed above the studio door, both inside and outside the studio.

RAG lights are controlled from the mixing desk and are used to indicate when a microphone in the studio is on. When a microphone comes on, a red light will light up in the studio so that studio guests and others do not make extra noise, and so that people don't suddenly enter the studio. When the mic is off, the red light switches off and in some studios a green light comes on to tell people it is safe to come into the studio.

RAG lights consist of two parts: a control box that handles the switching, and the light fittings.

A RAG light system costs between R1 000 and R5 000, depending on the type and number of lights required.

Optional Equipment

The equipment listed above will give you an effective basic studio. But there are some other pieces that you might like to consider. These are optional – that is, they are not absolutely essential – but will be very useful.

Computer Equipped With Playout and Recording Software

More and more community radio stations are using computers in their studios. The computer is used for playing out sound, in the same way as you would use a CD player or a MD player.

The computer offers a number of benefits. With appropriate playout software a computer can replace the roles of all the other source equipment (except the microphone!) in the studio.

Computers can be programmed – that is, a programme can be recorded onto the computer and can play unattended. This is useful for stations wanting to broadcast for a full 24 hours daily, but that do not have staff who can work through the night.

A playout system consists of computer hardware and playout software. If you are using this system, you will need at least one other computer for production.

A digital playout system, including a computer and software, costs from R30 000 upward, depending on what computer system and software you choose.

Under C in the A-Z for more on computer hardware and software.

Under D in the A-Z for more about digital playout systems.

Patchbay or Jackfield

A patchbay, also called a jackfield, is one of the most useful items in a studio. Regularly used inputs and outputs are wired to a patchbay so that they can be conveniently patched together (connected) with short signal leads.

Including a patchbay with the studio makes it more versatile. It also makes it easier to add new equipment and to find faults.

A patchbay, including patch cords, costs from R4 000 upwards.

Under P in the A-Z for more about the patchbay.

Turntable and Phono Pre-amplifier

Many stations want to be able to play records, either for live DJ mixing, or to play oldies – that is, music that pre-dates tape and CD, and which you can still only find on records. To play records, you need one or more turntables or record players.

The signal from a turntable cannot be fed straight into a standard mixing desk. It must first pass through a phono pre-amplifier. This piece of equipment equalises the signal from the turntable so that it can be fed to the mixer.

A broadcast quality turntable costs from R3 000 to R7 000. For live DJ mixes, two turntables are needed, and a DJ mixer would be used as a pre-amplifier to feed sound to the mixing desk. A DJ mixer costs from R1 200 upwards.

Under A in the A-Z for more on pre-amplifiers.

Under E in the A-Z for more on equalisation.

DAT Players / Recorders

Some stations use Digital Audio Tape (DAT) recorders. DAT is a medium that uses magnetic tape, which is similar to cassette tape, to store digital audio. DAT was initially designed to replace cassettes. It was designed for home use, but did not become popular, except for broadcast and other professional use.

DAT recorders can record large amounts of sound at a slightly higher quality than CD.

A good quality DAT Player / Recorder costs from R9 000 upwards.

Under D in the A-Z for more on DATs and DAT Players / Recorders.

Reel-to-reel Machine

Reel-to-reel tape is fast being replaced by new digital technologies in radio. Reel-to-reel machines use tape to record sound. Reel-to-reels deliver better sound quality than standard cassettes. Reel tape can also be edited by physically cutting out pieces of tape with a blade where you want to remove sound, and joining the cut ends together with splicing tape where you want the sound to continue. This form of editing is called splicing.

A reel-to-reel machine costs about R30 000 second hand, or up to R50 000 new. One reel of tape costs around R300.

Under R in the A-Z for more on reel-to-reel machines.

Satellite Receiver

Many stations use satellite receivers to receive and broadcast programmes. These programmes are broadcast via satellite from a central studio in South Africa or abroad, and re-broadcast on local stations.

Often satellite receivers are provided for free by radio stations and other organisations that want community radio stations to rebroadcast their programmes and information.

Broadcast studio at Radio Graaff Reinet. The broadcast mixing desk, presenter's microphone and playout computer can be clearly seen.

The Production Studio

The production studio is used to produce and package programmes, such as drama, documentaries or features, for broadcast at a later stage. It is also used to produce "jingles" or "stings".

This means that the production studio is geared towards recording, editing and creating sound. Its function is different from that of the broadcast studio, which is mainly designed to play out sound for broadcast.

Although the functions of the broadcast studio and the production studio are different, some of the equipment found in the production studio is the same, or nearly the same, as the equipment in the broadcast studio.

This section discusses typical production studio equipment, but we do not repeat information that is already in the previous section, on the broadcast studio. Where equipment is the same, or similar, we refer you back to the previous section.

The Studio Environment

Like the broadcast studio, the production studio has special acoustic fittings, air-conditioning and furniture. These should meet the same standards of reliability and ruggedness as in the broadcast studio.

The diagram on the next page shows a model of a typical community radio production studio. By following the arrows in the diagram, you will be able to see how sound (as audio signals) moves through the different pieces of equipment, and how the equipment is linked together.

Model of a typical production studio
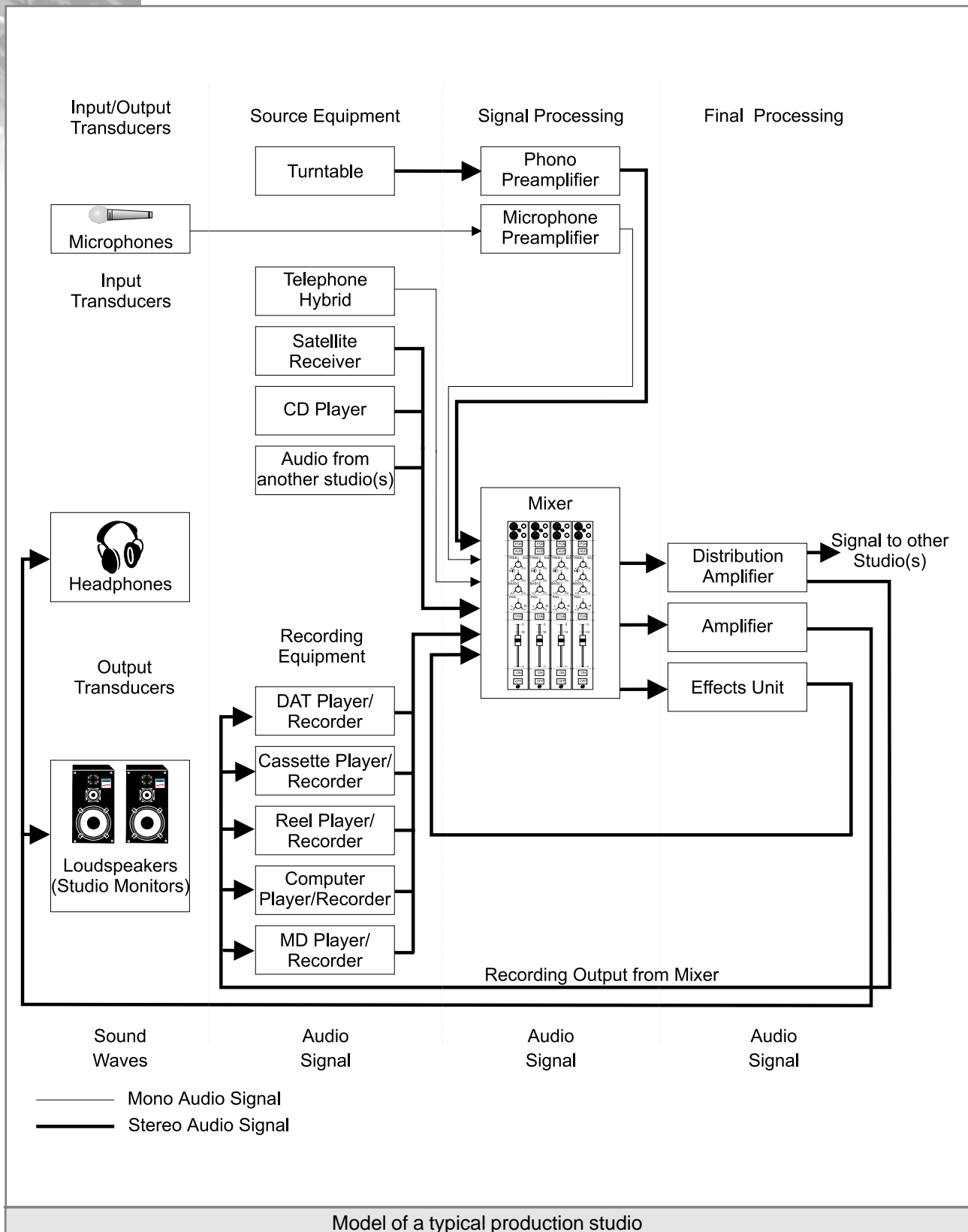
Production Studio Equipment

Microphones and Microphone Stands

The production studio may use microphones that are different to the ones found in the broadcast studio.

For example, if you are recording a panel discussion in the production studio, you might seat guests around a table. The microphone could stand on the table, or hang from the ceiling, but it will be placed in the middle of the guests so that it can record all the speakers at once.

This kind of microphone is called an omnidirectional microphone. Omnidirectional means from every direction, and the microphone will pick up sound from every direction.

The number of microphones being used in the production studio depends on what you are producing.

See Page 6.

Under M in the A-Z for a more about microphones.

Broadcast Quality CD Player

The production studio needs at least one CD player for listening to CDs and for recording from CDs. If your budget allows, you should install two CD players in the studio. This will make it more useful as a back-up broadcast studio.

See Page 6.

Under C in the A-Z for more about CD players.

Cassette Recorder

It might be a good idea to consider having a double cassette recorder (tape deck) in the production studio for dubbing (copying or duplicating) tapes. Copying tapes is useful if members of the community want a copy of a particular programme, or if you want to send a copy of a programme to anyone else for listening or broadcasting. Many stations also share programmes in this way.

See Page 7.

Under C in the A-Z for a more about cassette recorders.

Minidisc Recorder (MD)

If your station plans to use MDs in the broadcast studio or for field recording, then you must have a minidisc recorder in the production studio to record and edit discs for broadcasting at a later stage.

See Page 7.

Under M in the A-Z for a more about MDs.

Telephone Hybrid

Having a telephone hybrid in the production studio is very useful. This makes it possible for your station to pre-record telephone interviews, and enables you to edit the interviews.

It also makes it possible to interview people when THEY are available. One of the drawbacks of not having a telephone hybrid in the production studio is that the guest you want to interview over the phone must be available at the same time as the programme is on air. Remember – people are often not available when you want them to be.

Because the production studio does not have to handle as many calls as the broadcast studio, a single line hybrid is usually enough.

See Page 7.

Under T in the A-Z for more about telephone hybrids.

Production Mixing Desk

The production mixing desk is the central piece of equipment in the production studio. The production mixing desk often has different features to a broadcast mixer. This mixer is less geared to controlling playback of equipment and may lack the control features of a broadcast mixer.

However, the production mixer should have other features, such as equalisers, that allow you to change sound, and panning controls that enable you add stereo effects to sound.

The production mixer should also provide more control over the input and output levels so that you can make high quality productions.

Production mixers can be as costly as on-air mixers, but effective production mixers can be bought for R6 000 upwards.

Under M in the A-Z for more about mixers.

Under E in the A-Z for more on equalisation.

Under S in the A-Z for more about stereo.

Computer Equipped with Editing and Recording software

The computer has become one of the most important parts of the production studio. Even if your station does not use a digital playout system in the broadcast studio, you should think seriously about using a computer for producing and editing sound items that can be recorded to MD, CD or tape.

The production computer uses special production and recording software that has to be bought separately. This computer also usually has special hardware, such as a high quality sound card and a CD writer.

Under C in the A-Z for more on computer hardware and software.

Monitor Loudspeakers and Audio Amplifier

If you can afford it, you could consider using higher quality monitors in the production studio than in the broadcast studio. Producing programmes demands much more careful listening than usually happens in the broadcast studio. However, most community radio stations use the same or similar quality speakers in the broadcast and production studios.

See Page 8.

Under L in the A-Z for more about loudspeakers.

See Page 8 for more about audio amplifiers.

Under A in the A-Z for more about audio amplifiers.

Distribution Amplifier (DA)

See Page 9.

Under A in the A-Z for more about amplifiers.

Headphones

The production studio should be able to feed sound to many pairs of headphones. This may require using a headphone amplifier.

Having many pairs of headphones will allow you to record large groups of people, for example, a choir or drama group, in the production studio.

See Page 9.

RAG Light or On-Air Light System

RAG (on-air) lights are also important for the production studio. These lights indicate when recording is taking place, so that people outside the studio won't barge in and interrupt the recording.

See Page 9.

Optional Equipment

The equipment listed above will give you a very effective production studio. However there are several other pieces that you might like to consider.

Patchbay or Jackfield

See Page 10.

Under P in the A-Z for more about patchbays or jackfields.

Turntable and Phono Pre-amplifier

A turntable is particularly useful in a production studio, allowing you to record from records onto other mediums such as MD or recordable CD (CD-R).

See Page 10.

DAT Recorder

Many broadcasters use DAT recorders for master copies. CD-R can also be used for this purpose.

See Page 11.

Under D in the A-Z for more about DAT recorders.

Reel-to-reel Machine

For a tape-based (analog) station, having at least two reel-to-reel machines in the production studio is vital to be able to produce sound items. Two machines are needed so that overdubs are possible.

An example of an overdub would be adding background music to a pre-recorded interview. The interview recorded on the first reel would be played through the mixer, and at the same time, you'd add music from a CD player. This combination of music and interview is then recorded on the second reel.

This can be done over and over to build layers of sound. This is not possible with only one reel machine.

See Page 11.

Under R in the A-Z for more about reel-to-reel machines.

Effects Units

Effects units are used to change sounds to create atmosphere. For example, if you want to create the impression that someone in a drama is speaking in a large hall, the effects unit can add echo to the voice.

Different sound effects are normally supplied as part of the editing software used on the production computer. Instead, or in addition to the computer, some stations have dedicated effects units. These are pieces of equipment that can be programmed to process and add effects to sound.

There is a big variety of effects units on the market, and they vary in price.

Under E in the A-Z for more on effects.

Production studio at Radio Graaff Reinet, in the Eastern Cape.
The mixer, digital editing system and equipment racks can be clearly seen.

Voice Booth

A voice booth is a studio room that can be attached to the production studio and / or broadcast studio. The purpose of the voice booth is to record voices in a controlled environment. This means there is very little equipment in the voice booth, which is usually quite small. The most important aspects of the voice booth are the acoustics and the furniture.

Often a table – called a talk table – is part of the voice booth. This is a table for guests to sit around for discussion programmes or debates. There should also be comfortable chairs available.

The only other essential pieces of equipment in the voice booth are the microphones and headphones.

The microphones in the voice booth can be controlled by the mixing desk in the production studio, or by the mixer in the broadcast studio. The studio that controls the microphones in the voice booth is often called the control room.

Having a voice booth that is separate from the recording equipment in the production studio, or the playout equipment in the broadcast studio, improves your sound. It also gives you greater flexibility.

For example, if the broadcast studio is attached to the voice booth, newsreaders can read from the voice booth. While the news is being read from the voice booth, the presenter in the on-air (or broadcast) studio can continue working, searching for CDs or talking to a guest, without disturbing the news reader. If the newsreader reads news from inside the on-air studio, everyone else in the studio has to keep quiet until the news is over.

Under A in the A-Z for more about acoustics and acoustic treatment.

Equipment for the Voice Booth

Follow the arrows in the diagram below to see how the studios are connected to the voice booth.

Diagram of a voice booth, showing connections to other studios
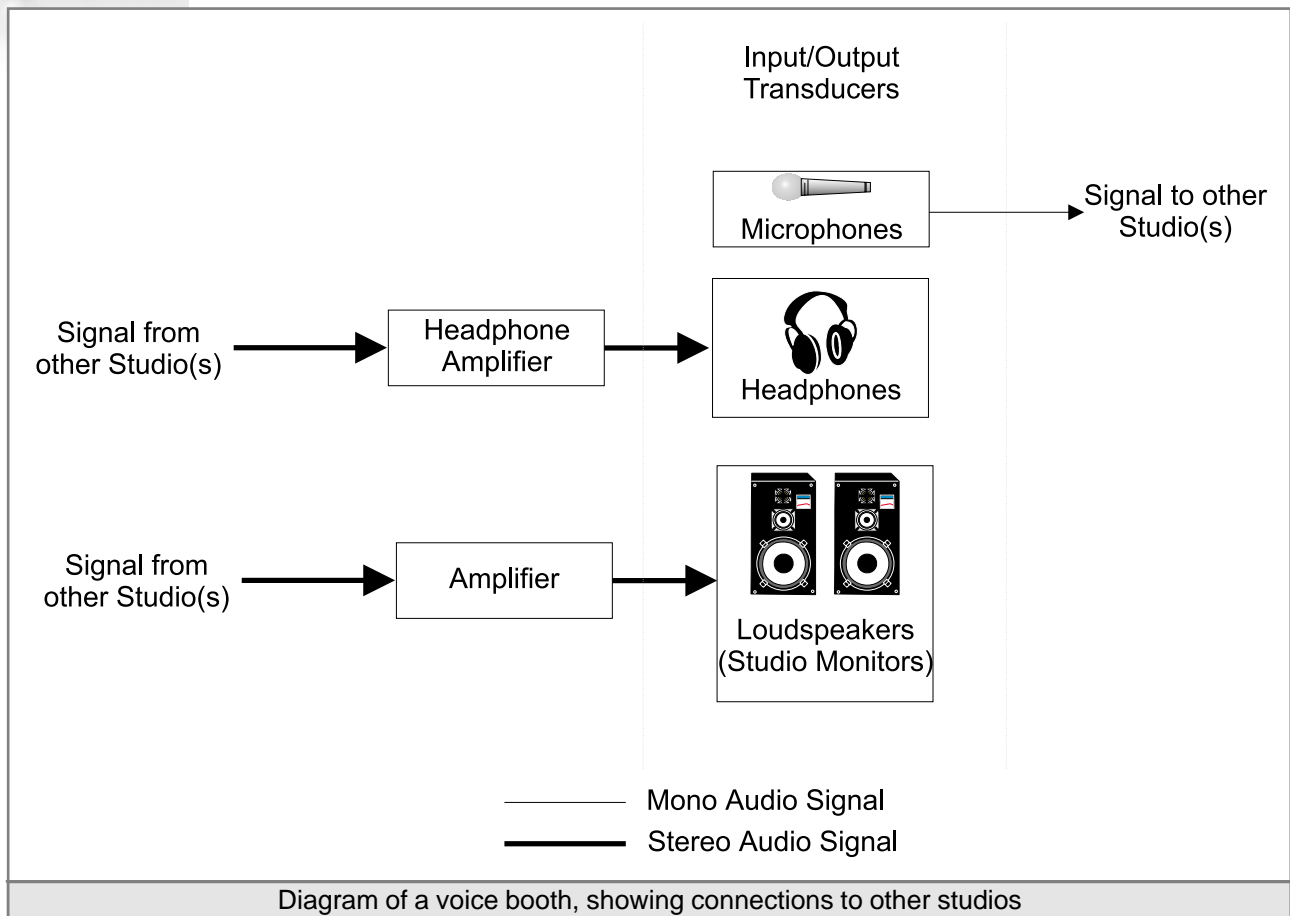
Microphones

The voice booth could have one or more microphones, depending on what you are recording. The microphones are used to record the guest, musician or voice artist in the voice booth. As the voice booth is an extension of the production studio, specialised recording microphones may also be used here.

Under M in the A-Z for more about microphones.

Headphones and Headphone Amplifier

Headphones are a very important item in the voice booth, allowing the people in the booth to hear themselves. The voice booth could also use a headphone amplifier to accommodate several guests by using several pairs of headphones.

See Pages 9 and 17.

Under A in the A-Z for more about amplifiers.

Patchbay or Jackfield

It is useful to have a patchbay between the voice booth and the production studio (or control room). By changing the connections on the patchbay, the voice booth can be connected to either the broadcast studio or the production studio. The patchbay also makes it easier to plug in extra microphones or other equipment.

See Page 10.

Under P in the A-Z for more about patchbays or jackfields.

Options

Studio Monitors

Some voice booths also contain monitors. This allows recorded sound to be played back to the people in the booth. But if you don't have enough money for monitors, headphones will be enough.

Editing Station

Many editing tasks do not need a sophisticated production studio. For example, taking sound bites (short sound clips taken from longer recordings) from a press conference to include in the news needs very simple editing. There are no music levels or other creative issues to think about, and so high quality monitoring, soundproofing and so on are not essential.

An editing station usually has a combination of a computer and a few pieces of audio equipment that can be placed on a desk in an office, or in the newsroom, to do quick edits.

The advantage of an editing station is that it is often much easier to use and understand than a production studio. This reduces the training needed for more people to get started, and also frees up time in the production studio for more complicated production work.

An editing station might consist of

- A small mixer

- A tape deck (or an MD player if your reporters use mini-disc field recorders) to play recordings the reporters bring back

- A computer with basic editing and recording software, and the computer's speakers.

- Some basic monitors or headphones for monitoring.

An even more cost-effective way of doing quick edits is to plug input and output connectors from the computer into your station's field recorders.

See the section on The Production Studio, Page 15, for more about the above equipment.

Field Recorders

Strictly speaking, field recorders are not studio equipment, as they are not often used in the studio (except as described above, with an editing station). But they are essential for community radio stations.

A field recorder is a portable recorder using tape, DAT or MD to record sounds "in the field" – in the community, at news conferences, in the street - anywhere away from the studio.

A field recording kit consists of several pieces of equipment:

• the recorder

• a rugged all-purpose microphone

• a microphone cable

• headphones for monitoring

• a carry bag

Under F in the A-Z for more about field recorders.

Minidisc field recording kit with headphones, a microphone, the necessary cables and a minidisc

Transmission

Transmission is the final stage of the broadcast system, and usually happens outside of the studio. The broadcast and other studios provide the audio signal that the transmission system converts into a radio signal that can be picked up by your listeners' receivers, or radio sets.

The diagram below shows how the audio signal leaves the broadcast studio and reaches listeners.

Model of a radio transmission and reception system
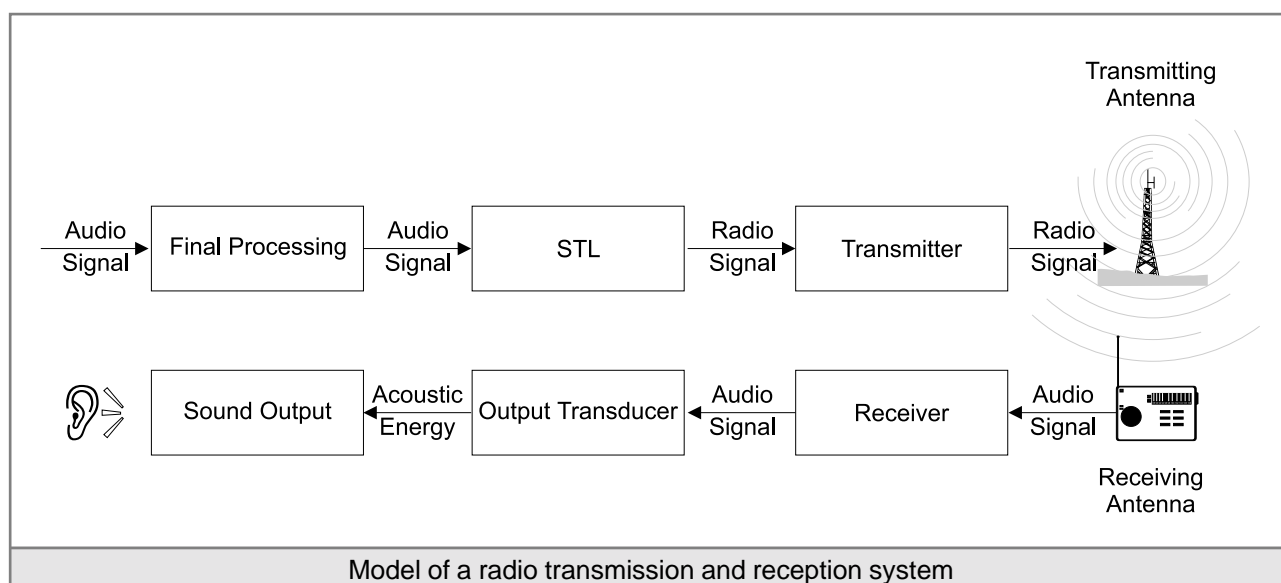
The audio leaving the studio is usually processed before it is broadcast. This processing limits the level of the signal to prevent distortion or damage to the transmission equipment.

Processing can also be used to improve the sound. Most community radio stations use a piece of equipment called a compressor limiter to process sound before it reaches the transmission system.

A Under C in the A-Z for more about compressor limiters.

Once it has been processed, the signal has to travel from the studio to the transmission equipment. Many stations have transmitters far away from the studios and so will need a studio-transmitter link (STL). The STL is needed to carry the signal from the studio to the transmitter.

A Under S in the A-Z for more about STLs.

The transmission equipment combines the audio signal with a radio wave through the process of modulation.

A Under M in the A-Z for more about modulation.

The transmitter feeds the modulated radio signal to an antenna. An antenna is specially designed so that a signal that is fed into it is radiated out into space.

This sends the radio wave from the transmitter out into the air. This is the signal that the antenna on a radio receiver is able to pick up.

The receiver then demodulates the radio signal. This is the opposite of the modulation process that created the radio signal, and separates the audio signal from the carrier radio wave.

The signal picked up by the receiver is often very weak, so the radio receiver amplifies the signal. So that this can happen, the audio signal is passed to an audio amplifier and a loudspeaker that produces the sound we hear.

Under T in the A-Z for more about transmitters and transmission.

Transmission tower at Radio Bushbuckridge in Limpopo. The transmitting antennae can be seen at the top of the tower.

Section 3:

Making a Radio Station With Studios

As we have seen, a radio station often has more than one studio. The studios, with the transmission system, make up the station's broadcast system. The way the studios are connected to each other is one of the things that will define how the station operates technically, and what it is capable of.

The diagram below illustrates the floor plan of an imaginary community radio station's studio complex.

Floor plan of a community radio studio complex
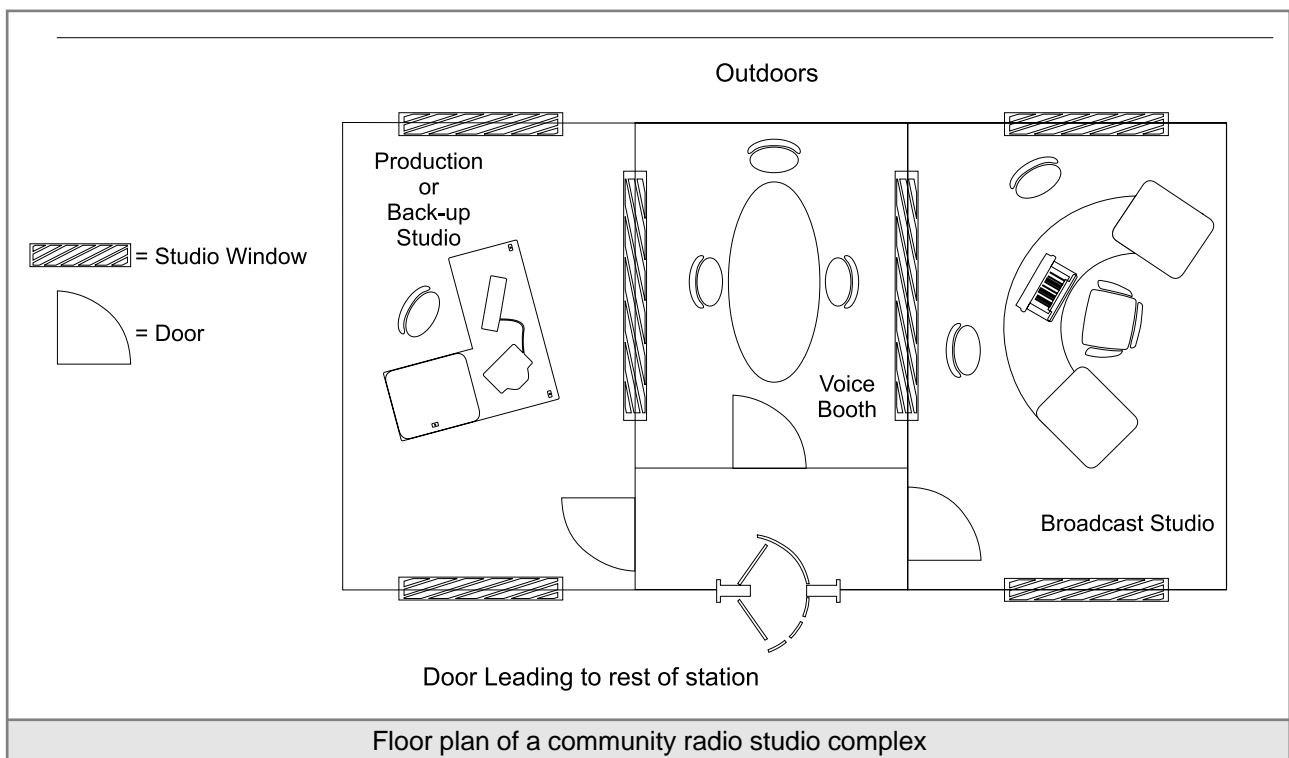
This model has three studios: a broadcast studio, a voice booth and a production studio. The position of the three studio rooms, the positions of the windows and doors and the audio connections between the studio rooms all affect how this station will operate.

As you can see, the studios all have windows. These are important – they let in light and create a pleasant working environment.

But they also have other functions. The windows in the broadcast and production studios allow people outside the studios to see in, and people inside the studio to see out.

The two windows on the outside walls fulfil an important marketing function. They allow visitors and passers-by to look into the studios. Being able to see the radio station in action generates interest and excitement.

The windows facing into the station offices make it possible for other station staff to look into the studios and to communicate with presenters or programme hosts – by waving or holding up notes. Many on-air mistakes or problems have been avoided when someone outside the studio holds a note up to a studio window.

The windows between the studios and the voice booth, allow people in the studios to communicate with one another.

The broadcast studio, the voice booth and the production studio are usually connected to enable the mixer in the broadcast studio to receive audio signals from the voice booth microphones as well as the production studio.

At the same time, the audio signals coming from the broadcast studio and production studio can be fed into headphones or monitors in the voice booth. This allows people in the voice booth to hear what is happening in the broadcast and production studios.

Together, the combination of windows and audio connections allow people to see and hear one another, making two-way communication possible between all three studios.

Connecting Your Studios

The diagram below shows how a voice booth, production studio and broadcast studio can be connected.

Connections between studios in the broadcast system
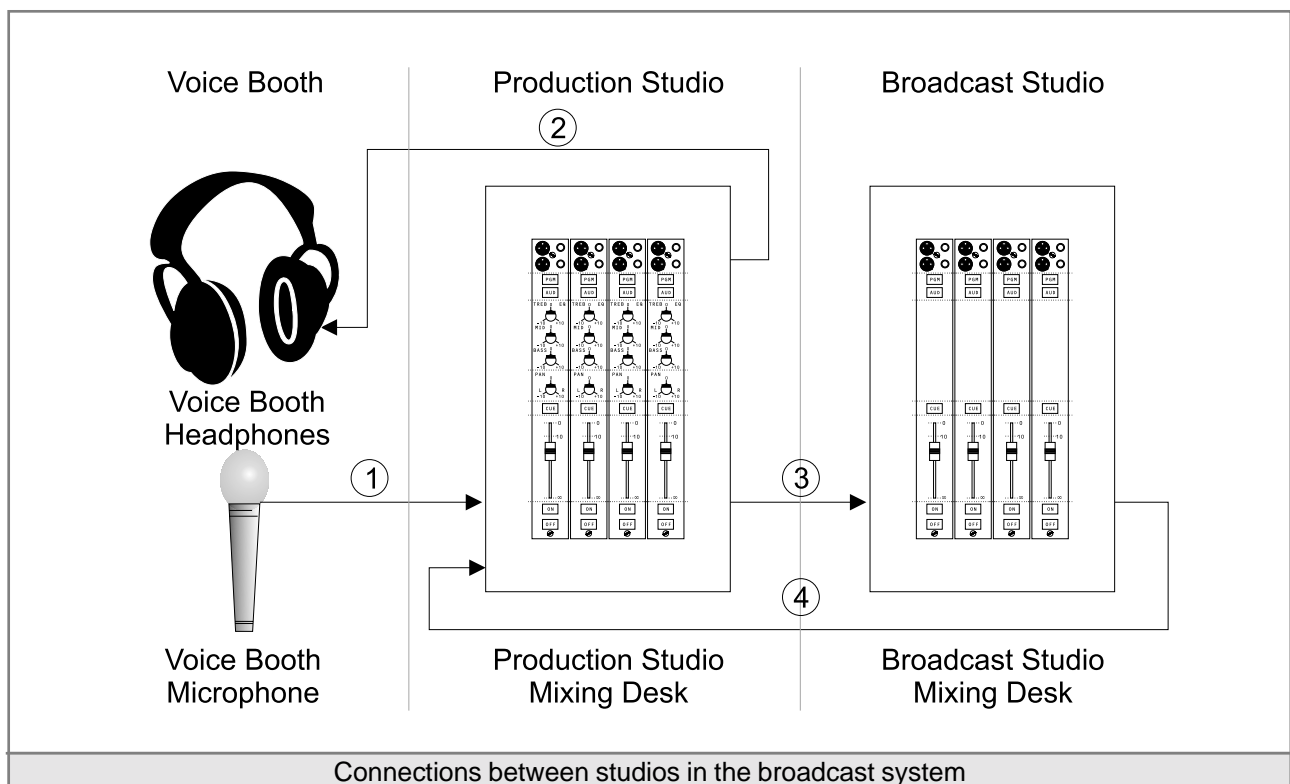
Follow the arrows and numbers in the diagram to track the signals between the studios.

The voice booth microphone(s) are connected to a channel(s) on the production mixer (number 1). This connects the voice booth to the production studio.

The production studio mixer has an output that feeds the headphones in the voice booth (number 2). This allows the audio from the microphones and any other equipment connected to the production mixer to be heard in the voice booth.

The production mixer also feeds audio to the broadcast studio mixer (number 3). This connects the production studio to the broadcast studio, and allows audio from the production studio mixer (including signals from the voice booth microphones, because they are connected to the production mixer) to be broadcast through the broadcast mixer.

The broadcast mixer has an output that is connected back to the production mixer (number 4). This allows audio from the broadcast studio to be heard in the production studio and also in the voice booth, as any signal connected to the production mixer can be fed to the voice booth by the headphone output (number 2).

Models from the Real World

The imaginary floor plan drawn on Page 31 shows only one way of placing and connecting studios.

Different stations will choose different approaches, depending on their premises, their broadcast style, their budgets and other things.

Radio Unique

Radio Unique is a community radio station serving Burgersdorp and surrounding towns, in the Eastern Cape. The diagram on the next page shows how their studios have been laid out.

Radio Unique's studios are in a house that was built late in the last century. At first, the station wanted to put the production studio and voice booth in the space labelled "office". When building started, they found that there were problems with the floor, and the space could not be used for a studio. So they decided to move the production studio and voice booth to the positions shown below.

A disadvantage of the move was that the person or people in the voice booth could no longer see into the broadcast studio. An advantage was that the station gained an extra window in the production studio, looking out into the street. This allowed passers-by to see into both the on-air and production studios, which attracted a lot of public interest.

= Studio Window

= Door

Kitchen
(to be renovated)

Office
(to be renovated)

Voice
Booth

Production
Studio

Reception

Broadcast Studio

Van der Walt Street

Floor plan of Radio Unique
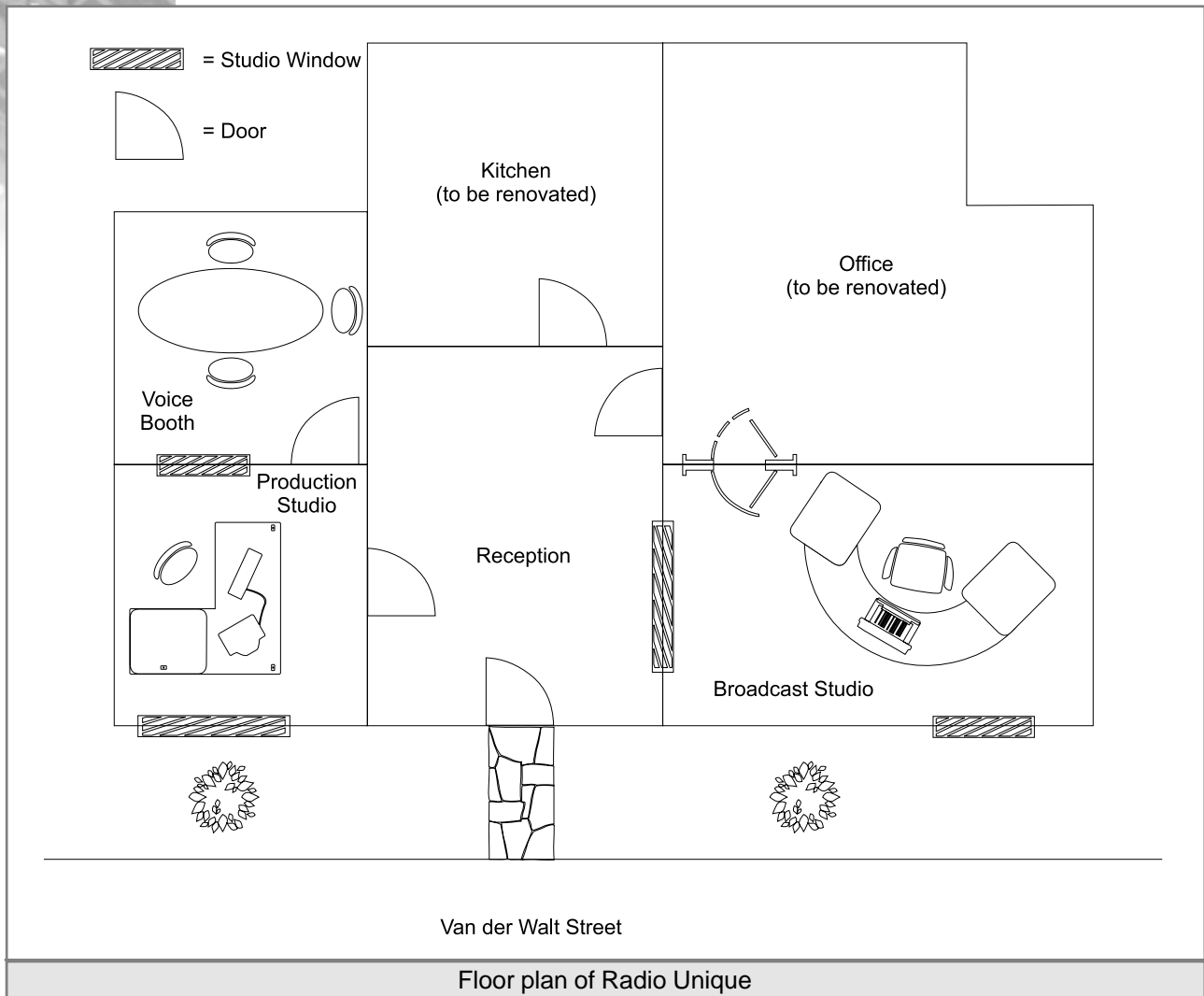
Maputaland Community Radio

Maputaland Community Radio is a station serving Jozini and the surrounding areas of northern KwaZulu / Natal. The station premises are a converted house. The studios occupy rooms that used to be bedrooms.

The station chose their design to allow for the future addition of a production studio.

The rooms are located at one end of the house, with a door separating the studio area from the reception area and the rest of the station. This door can be closed, improving soundproofing and studio security.

The programme manager's office is within the studio area, close to where the action is. The window linking the broadcast studio to the reception area fulfils the important marketing role, as visitors can see into the studio.

WC

Proposed
Production
Studio

Bathroom

Programme
Manager's Office

Reception
Area

Voice
Booth

Broadcast Studio

Outdoors

Floor plan of Maputaland Community Radio
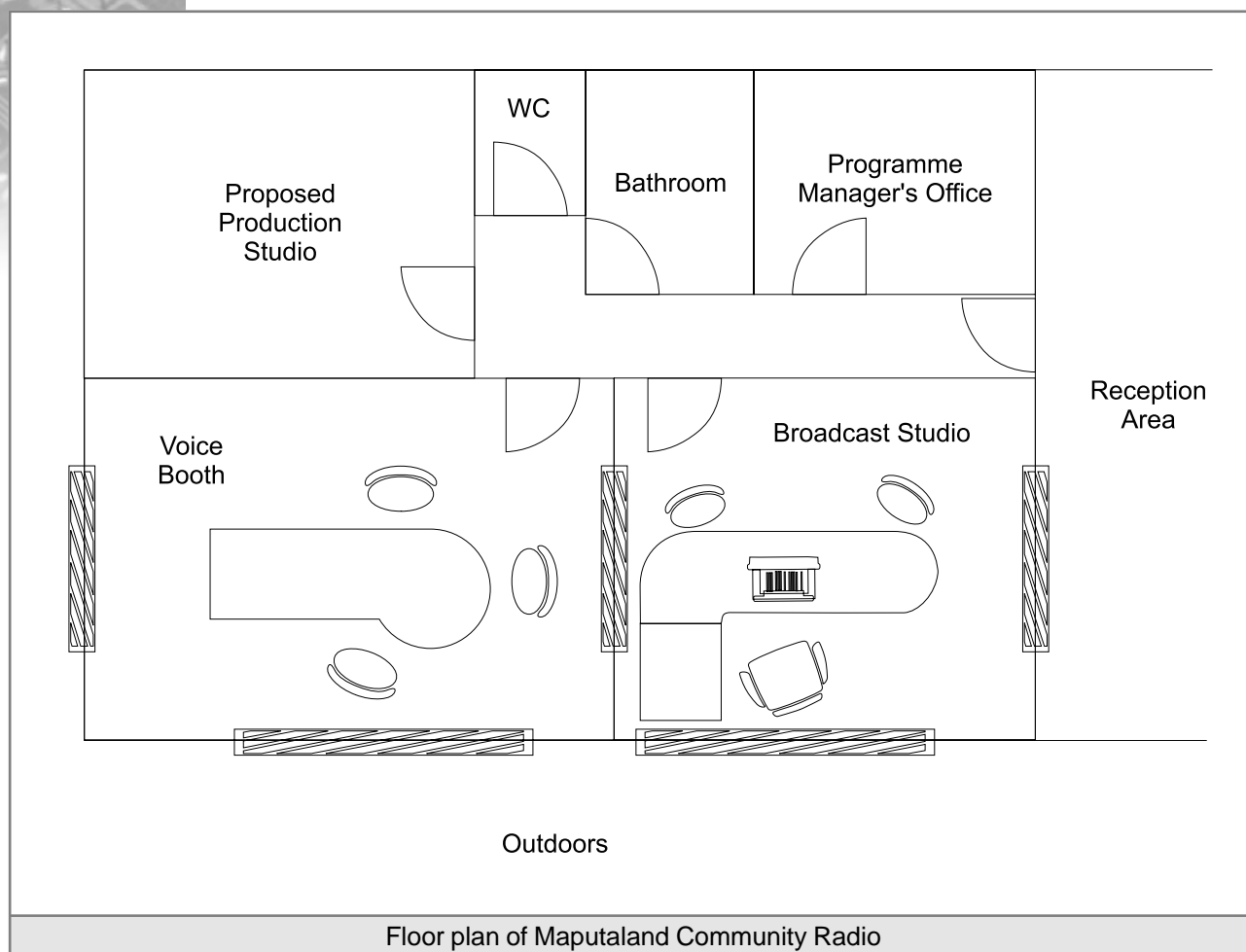
To Sum Up…

The connections between the studios, and the arrangement of studios at your station, will determine how you work. As you can see, there are several aspects of studio layout and connection to consider.

Consult with your supplier when it comes to studio layout to find out what is possible, keeping in mind what is important for your station, based on your mission and how you expect to do your work.

Try to keep the layout as flexible as possible. Broadcasting techniques and technology are changing very fast, and over time the way your station operates could change. As far as possible, the studio layout should allow for this and leave possibilities for additional equipment and future improvements.

Section 4:

Connecting to the Outside World

In sections 1 – 3, we looked at the broadcast system – that is, the studios and transmission system. We also looked at how they are connected.

We now look at the important connections between the station and the outside world. Good connections to the outside world are a way of linking the station to the community, and of linking your community to other communities far away.

The basic tool that most community radio stations use to connect to the outside world is the telephone. But there are other ways – and we will also look at satellite connections, and new Information and Communications Technologies (ICTs).

Telephone Connections

Telephone lines make it possible for members of the community to phone into the studio during call-in programmes. They also enable the presenter or programme host to conduct on-air interviews over the phone.

Telephone lines are also used to send audio from the studio to the transmission site. They can also be used to connect the station to the Internet.

There are many different kinds of telephone connections. The important ones for broadcasters are

- Standard fixed lines

- High quality analog fixed lines

- High quality digital fixed lines

- Cell phones

Standard Fixed Lines

These are the phone lines that we are most familiar with, as most domestic and public phones use them. Standard fixed lines have limited sound quality and are best suited for the transmission of voices. A standard line is not good for carrying music.

A standard fixed line has to be rented from a telephone company, or business. At present Telkom is the only company supplying phones and phone lines in South Africa, but in the next couple of years, other companies will be allowed to enter the phone business and compete with Telkom.

A standard fixed line carries a monthly rental fee in addition to the cost of each call made.

Each line allows one connection to the station. Combined with a telephone hybrid, a standard fixed line can be used to get the voice of an interviewee or a remote community member on air.

The telephone hybrid is a signal processing device. It connects your mixing desk to the telephone network and changes the sound of the voice that comes over the phone into a signal that can fed to the mixing desk for broadcast or recording. It also sends the voice of the presenter to a caller so that the caller can hear what is happening in the studio.

There are single-line and multi-line telephone hybrids. A single-line hybrid can only handle one call at a time. A multi-line hybrid enables you to speak to more than one person at a time.

For example a multi-line hybrid will enable you to have an expert or politician on one line, and listeners will be able to phone on other lines and ask the expert or politician questions.

Many stations use multi-line hybrids, because it is important for community radio stations to be very accessible to community members. But a multi-line hybrid costs more to buy and install. In addition, it needs more incoming telephone lines, each of which carries a monthly rental.

A telephone hybrid can be used for many different purposes. Some examples are:

- Call-ins for dedications, requests and so on.

- Telephone interviews – local, national and international politicians, musicians etc are really only a phone call away.

- News gathering – reporters can call-in with stories from the site of the event they are covering.

- Broadcasting sports matches, civic events and religious ceremonies.

Under T in the A-Z for more on telephone hybrids.

Because telephones are so important, the station should try to afford three standard lines, with different phone numbers as follows:

The Studio Phone

One line is for the studio, which is connected to the telephone hybrid. However, to accommodate call-ins, chat shows and dedications, it is preferable to have at least two lines sharing the studio number. If you have more than one line into the studio, you will need a hunting line system. A hunting system makes it possible for several lines to share the same number. When a community member dials the number, the hunting system "hunts" for an available line. This means that more than one person can dial your studio number at the same time without getting an engaged signal. Only when all the incoming lines are in use will the caller get an engaged signal.

The General Office

This line would be used for all other station business – making appointments; getting advertising; reminding volunteers about their programmes; receiving business and other calls. While at least one line is necessary, this could also be a multi-line system. It could operate through a switchboard that will send calls to different extensions.

The Fax Machine

Many organisations in the community, like the local police station, clinic, municipality, women's groups, schools and so on, may want to fax news and information to the station. Organisations that will want to support your station, like training organisations, advertisers and donors may also want to fax you information. All this makes a phone line dedicated to a fax machine very important.

Warning – your phones can bankrupt you!

It is important to have clear policies about who may use the station phones, when they can use them, what for and for how long.

Some stations are located in communities where there are not many domestic or public phones. So people use the station phones for private and personal business, and run up bills that the station cannot afford. As a result, the phones are cut off and the station loses its primary link to the outside world. The station is then denied phone access to community members as well as advertisers, donors, training opportunities and so on.

Your choice of phone system will depend on what you can afford. The more lines you have, the more rent you will pay. The more complicated your system is, the more it will cost to maintain.

The absolute minimum number of lines for a community radio station is two – that is, a line for the studio and a line for the office. To keep costs down, you can also double up your office line as a fax line by using a phone / fax machine.

Keeping at least one line open is fundamental to your station's sustainability. Clear policies will ensure your bills are affordable and your phones won't be cut off.

Lightning

Phones are also at great risk from lightning strikes. When lightning strikes a phone that is connected to a studio, it can damage and destroy your broadcasting equipment. It is important to take steps to minimise damage that can be caused by lightning.

Under L in the A-Z for more on lightning protection.

High Quality Lines

The shortcoming of the standard phone line is that it can only deliver limited sound quality, and so standard lines are only really suitable for voices. If you want to send a high quality audio signal that combines music, voice and sound effects from one place to another, you need a high quality line.

A high quality line can be analog or digital. These lines are usually used for outside broadcasts and to send signals from the studio to the station's transmitter, when the station and the transmitter are located apart. The link used to send signals from the studio to a remote transmitter is called the studio-transmission link, or STL.

Under S in the A-Z for more on STLs.

The high-quality digital line usually used is called an ISDN line. ISDN stands for **I**ntegrated **S**ervices **D**igital **N**etwork. The advantage of an ISDN line is that it can carry enough digital information to send CD quality sound.

Using an ISDN line involves first converting the signal that you want to send into a digital data stream. To do this you need a piece of equipment called an ISDN codec. (Codec is short for **co**der-**dec**oder). One of these devices is needed at each end of the line.

Under A in the A-Z for more about analog and digital data.

A codec is an expensive piece of equipment, that can cost more than R30 000. Because you need one at each end of the ISDN line, the cost will be R60 000.

Stations using ISDN lines for their STL, or who do many outside broadcasts, may find it cost-effective to buy a codec. Alternately, you can rent a codec with the ISDN line, from Telkom.

Telkom usually requires long notice to provide an ISDN connection. So if you need one, you should contact Telkom at least three or four weeks (or even longer) before the date you'll be needing the line.

ISDN lines can also be used to connect to the Internet. Stations using the Internet extensively for broadcast, and which can afford the rental costs, could use ISDN to provide a high-speed connection to an Internet Service Provider (ISP).

Cell Phones

Cell phones – short for cellular telephones – are a remarkable tool for connecting to the station. Most of South Africa is now covered by cell phone networks, so it is possible for reporters and listeners to call into the studio from nearly anywhere. Reporters equipped with a cell phone can call in from the sidelines of a sports match, the centre of a concert crowd or from the top of a mountain.

At present, cell phones are only suitable for transmitting voice. But this will change in the future, as cell phones are already able to transmit digital information.

Warning – cell phones can bankrupt you!

Calls made on a cell phone cost more than calls made using a land line.

Stations that have "station cell phones" must manage them very carefully, as the costs of cell phones can quickly mount. Also, cell phones are small and portable, and can easily be lost, damaged or stolen.

Managing cell phones means developing policy about how they can be used, and making sure they are insured.

The Internet

The Internet is a global network of computers. By connecting to the Internet you can get access to an enormous amount of information available on these computers.

The Internet is not only a valuable information resource, but also, by using e-mail, the Internet is a quick and cheap communications tool. E-mail allows you to send messages to other people who are connected to the Internet.

To use the Internet and e-mail, you need a computer that can be connected to the global network, or World Wide Web (WWW). Standard phone lines are the most common connection to the Internet.

The computer uses a piece of equipment called a modem to send signals over the phone, and to "speak" to other computers that are connected to the Internet. Modem stands for **mo**dulator-**dem**odulator.

The modem attached to your computer converts digital signals from the computer into analog signals that can be carried on a standard phone line. Another modem on the receiving computer converts the analog signal back to digital data that can be understood by the receiving computer.

An abbreviation that is often used when talking about Internet connections is TCP/IP. TCP/IP stands for Transmission Control Protocol/Internet Protocol. These protocols are the language used by computers on the Internet to "speak" to one another.

In addition to the modem and phone line, the computer needs software that is designed to browse the Internet or send e-mail. Fortunately most computers are supplied with this kind of software

The diagram below shows how you can connect to the Internet using a standard phone line.

An Internet connection using a standard phone line
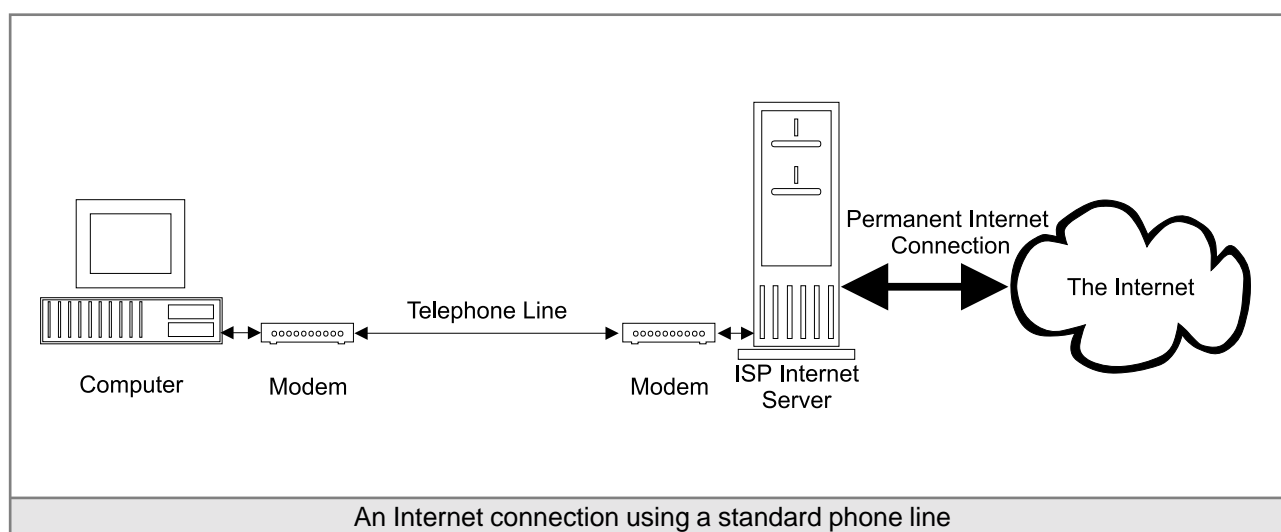
Follow the arrows in the diagram. The modem connects to an Internet Service Provider (ISP). The ISP runs a computer called an Internet server that is permanently connected to the Internet network.

To get information from the Internet, your computer sends a request for that information to the Internet server. The ISP's server receives this request. It then fetches the information you want from the Internet and sends – or "serves" – it to your computer using the modem connection.

South Africa has many large national and smaller regional and local ISPs. Most ISPs run their services on a subscription basis. Apart from the costs of the hardware and software, and making the connection, there are two main costs of using the Internet:

• Payment for use. You can subscribe to the service by paying a monthly fee to the ISP.

• The cost of using the phone line, calculated as units of time, using standard rates.

A basic computer will give you the ability to connect to an ISP. However the more you want to do with the computer, the more advanced it should be.

One of the most common combinations of Internet browsing and e-mail software are Microsoft's Internet Explorer 6 and Outlook Express 5, on a computer using Microsoft Windows 98 or 2000.

Minimum System Requirements for Internet and e-mail:

The following specifications show you the kind of computer you will need to operate to connect to the Internet using this system:

• A Pentium or Pentium compatible class processor with a processor speed upwards of 133MHz

• 32MB of RAM (For Windows 2000, 64MB is recommended)

• At least a 2GB (gigabyte) Hard disk drive

• A modem or other means of connecting to the Internet. Modems operate at different speeds and a "56K" modem is recommended. Slower modems can be found for cheaper prices, but unless it is absolutely unavoidable the faster modem is recommended.

• A CD-ROM to install the software.

See Appendix B for a list of useful Internet sites for broadcasters.

E-Mail

E-mail is a fast and cheap way to communicate via the Internet. E-mail allows users around the world to send and receive messages to each other, using their Internet connection. Part of the service offered by an ISP is an e-mail account, which gives you an e-mail address.

Once this is set up, and you have e-mail software, you are ready to speak to other people in your community who have access to computers with e-mail, or with anyone else in the world.

Your e-mail address is the e-mail equivalent of a phone number. To be able to send messages to someone you need to know their e-mail address.

Typically an e-mail address looks something like this: techguy@theISP.co.za

The first part of the address is the user's name. In this case, the user is called "techguy". The symbol @ means "at", and it tells you which organisation or ISP the user is connected to. In the example, "techguy" is connected to "the ISP".

Instead of using the ISP's name, some companies and organisations put their name as this part of the address. For example, the National Community Radio Forum (NCRF), might have as an address: mabalane@ncrf.org.za.

In the first example, "co" stands for commercial, meaning that "theISP" is a listed commercial organisation, or business. In the NCRF example, "org" stands for organisation, meaning a non-commercial organisation.

The "za" in both examples shows that the e-mail user is based in South Africa. Most countries have a code, for example, Namibia uses "na"; the Netherlands uses "nl". Only the United States does not have a country code, and addresses that end without a country code are likely to be based in the US.

E-mail can be used to send text, pictures and sounds. A problem with e-mail is that it can only send a limited amount of data, or information. Sending big sound files by e-mail can be difficult. ISPs normally limit the size of e-mail messages that can be sent. This stops the e-mail system from becoming jammed with masses of data.

Webcasting

Webcasting, or net-casting, means broadcasting on the Internet. Webcasting is becoming more and more common. Some stations have their programmes available live on the Internet, or have created web-sites containing archives of material that can be listened to by Internet users.

Webcasting is an exciting opportunity, as the reach of webcasting is unlimited. Any Internet user anywhere in the world can hear programmes that have been webcast. But the high cost and difficult technical issues of webcasting limit its scope for community radio.

To listen to a webcast is also demanding on the user. Users who have modem-based Internet connections using standard phone lines often do not receive good quality audio from webcasts.

If you think your station can benefit from exposure on the Internet, building an Internet site is probably a good first step. It is relatively easy to place archived sound items on the site for users to listen to.

There are still some remote rural areas in South Africa that do not have phone connections, and many people in rural and urban areas cannot afford phones, let alone computers. At the same time, compared to many other developing countries, South Africa's telecommunications system is quite advanced. More and more homes, community-based organisations and NGOs are getting connected to the Internet.

This makes it important for community radio stations to connect too: you will be able to communicate with organisations in your community through e-mail, and get access to vast amounts of news and information that is available on the Web.

Satellite Broadcasting

Satellite broadcasting, that is, audio beamed by satellites located in space, is a vital part of broadcasting today. Radio and television stations all over the world make extensive use of satellite transmission to broadcast to far larger areas than is possible with terrestrial (earth-based) transmission.

Satellite transmission places the broadcaster's transmitting antenna on a satellite that is high above the earth. Because the satellite is so high the signals it transmits can be received across a vast area. Some satellites can cover entire continents with their transmission.

The diagram below shows how signals are transmitted from outer space to receivers on earth.

Satellite transmission, with various uplinks and downlinks
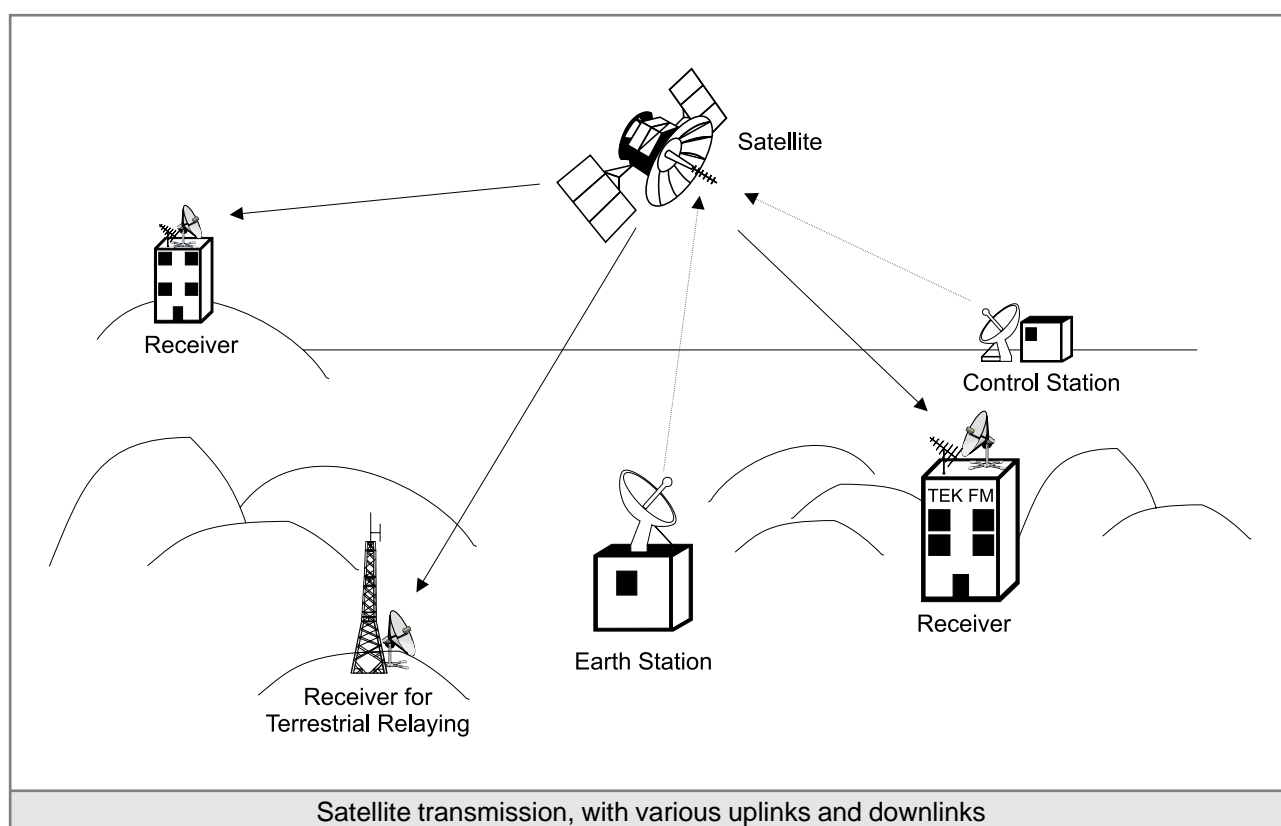
A satellite is a small spacecraft, launched by rocket into space and put into orbit round the earth. There are currently more than 2 500 satellites in orbit.

Communications satellites are often placed in geostationary orbit, about 37 000km above the equator. Satellites in this orbit travel at the same speed as the earth and so appear to be stationary in the sky, allowing them to cover a fixed area with their transmissions.

Now follow the diagram.

All satellites are monitored from earth and can be operated by remote control, from the control station.

The earth station sends the signal that is to be broadcast up to the satellite. This is referred to as an "uplink". The uplink to a satellite is a lot like the STL, or studio transmitter link in the broadcast system. Both are links that send a signal intended for broadcast to a transmitting antenna. An "uplink" satellite dish is about 10 times the size of a domestic satellite dish. Most communications satellites transmit and receive signals in the Gigahertz (thousands of Megahertz) frequency range, and can carry hundreds of channels, such as telephone signals, data, television, radio broadcasts and so on. These satellites are operated commercially and the operators lease channel time on the satellite.

Once the signal has reached the satellite, the satellite rebroadcasts it. We call the transmission from the satellite a "downlink". The signal transmitted through the downlink is what we are able to pick up using a satellite dish and decoder.

In South Africa, international broadcasters, news services and organisations are able to make programmes available to community radio stations throughout the country via satellite.

To receive satellite transmission, we need a satellite dish and a decoder. The decoder and the dish combine to provide a receiver, in the same way as a radio receiver and an antenna work. The decoder decodes the (most often) digital satellite signal to provide an audio output that can be connected to the mixing desk in the studio.

The decoder, like a radio set, needs to be tuned in to the station you want. Tuning in a satellite decoder is a little different from tuning a radio set.

For example, for some satellite decoders, you type in the frequency of the transmission you want, rather than turning a dial to find the station as you would with radio. The typed frequency can then be stored as a channel on the decoder, in the same way as a television set remembers channels. In the future you can return to that station by choosing that channel.

There are also other systems of tuning satellite decoders.

The diagram also shows a terrestrial relay station. These are used by the public and commercial national broadcasters. The satellite link is used to connect the broadcaster's studio with their many transmitters all over the country. At the transmitter, a satellite receiver picks up the signal sent from the broadcaster's studio and feeds it to a conventional earth-based FM/AM transmitter. This transmitter then re-broadcasts the signal received from the satellite. This allows listeners to hear the station using standard FM/AM radio receivers. As the broadcaster needs to link the studio to so many transmitters all over the country, satellite is much easier to install and maintain than other types of STLs.

Section 5:

Basic Maintenance

At the time we were writing this manual, the cost of high quality studio equipment and fittings for community radio studios was around R400 000. This amount excludes transmission and furnishings and fittings for the station's offices (reception area, meeting rooms, music library, etc.).

For about R400 000, you could

- Do some necessary alterations to your premises.

- Buy and install soundproof doors and windows, and acoustic panels.

- Buy and install air conditioning units.

- Buy and install equipment for a broadcast (on-air) studio.

- Buy and install equipment for a production studio.

- Buy and install equipment for a voice booth.

Most community radio stations in developing countries do not have enough money to build studios and install equipment, and so seek help from donors or sponsors to do this. It could be several years before stations are able to buy their own equipment. At many stations, salaries and running costs like phone bills, electricity, rent and water, or the costs of activities, such as marketing campaigns, take priority over spending money on equipment.

This means that to avoid extra costs, stations must maintain their equipment very carefully.

Maintenance should not be a special event performed by technical staff once a year. Maintenance must be an ongoing activity that is part of how everyone at the station uses the studios.

There are three key aims of a good maintenance programme:

- To ensure you consistently broadcast good quality sound.

- To save the station money.

- To anticipate and plan for problems before they arise.

Broadcast equipment is expensive to repair, and many stations struggle on with broken equipment because they can't afford repairs. Sound quality drops, and listeners become frustrated and tune into other stations.

Your maintenance programme must plan for future technical problems before they arise. If plan properly, when a repair or replacement is unavoidable, you will have the money to pay for it.

Basic Studio Rules

Community radio is a participatory activity. Stations have many different kinds of people – young, old, experienced, inexperienced, skilled, unskilled, staff and volunteers – using the studios. Everyone who uses the studios has a big role to play in maintaining the equipment.

Anyone who uses the equipment can cause technical problems at a station. One poorly trained, undisciplined staff member can wreck a studio's equipment in a few minutes. On the other hand, trained staff and volunteers who understand that they are jointly responsible for keeping the studios in good working order can help equipment last for many years.

Maintenance starts with the people using the equipment. Certain basic studio rules that apply to everyone at the station, from the youngest volunteer to the chairperson of the board, are essential to keep your studio working.

Dirty studios, incorrect operation and rough handling will cause studio equipment to fail before its time.

If you stick to the rules below, your equipment will last longer. Everyone at the station should understand these rules. It is a good idea to draft a set of your own rules and pin them to the studio door or stick them onto the wall outside the studio.

Keep the Studio Clean

This means never eating, drinking or smoking in the studio.

Personal hygiene is very important. People using the studios have to share headphones, seats, mics and other equipment. So

• When handling equipment, your hands must be clean.

• Wipe your feet on a mat before entering – mud on your shoes will turn to dust and get into equipment.

• Wear clean clothes in the studio.

Keep a waste-paper basket in the studio so that programme hosts can throw away old notes and scripts. But be sure it is a small one and empty it every day. Lots of crumpled paper gathers dust.

Never allow animals into the studio.

Clean the studio regularly. Someone must be trained to clean, dust and vacuum the studio thoroughly at least once a week.

Ask your supplier which sprays and polishes to use on the studio furniture.

Protect Your Equipment and Handle it with Care

Never allow untrained staff and unsupervised guests to handle equipment. People unfamiliar with the studio equipment can damage it without knowing that they are doing harm. The way to get the most from the equipment is to train staff to use it properly and safely.

Children should always be supervised.

Never Use Force to Get Equipment to Work

You never need to use force with broadcasting equipment. If you find you need to give a button, a connector or a slider anything more than a gentle touch, there is a problem. Stop, check your machine and think about it very carefully.

Report Faults as They Happen

Encourage users to report technical problems to the responsible person as soon as they happen. This means that everyone who uses the equipment must know who is responsible for technical management and maintenance. No-one except the technician should ever open equipment racks and try to repair faults.

Manuals, Wiring Diagrams and Other Documents

Manuals

Almost every piece of equipment in your studio will have a manual written by the manufacturer, explaining how it works and giving basic maintenance instructions. Even if you follow all of the basic studio rules, sooner or later a piece of equipment will need attention. The manufacturer's manual is your first and most important source of information about what to do.

For this reason, equipment manuals are VITAL. Keep them in a safe and accessible place and NEVER let them be taken out of the station. Lost manuals can only be replaced at very great cost.

Wiring Diagrams

When the suppliers have completed installation of the equipment, you must make sure they leave behind wiring diagrams. Wiring diagrams explain how the studio is wired. Each cable drawn in the diagram has a number, and the diagram will help you track the flow of power and audio signals.

Wiring diagrams are essential to help you locate faults and problems in the wiring.

Create Your Own Documents

Make your own notes about your studios and transmission system, and keep them with the manuals and wiring diagrams. It is important to have your own notes because not every studio is the same.

Note Down Special Pieces of Equipment Installed

This is important. Your supplier may have introduced extra, or more complicated equipment to make sure you can broadcast the best possible sound. Noting these differences will remind you of what they are and help you to find faults.

Make Notes of Equipment Settings

It is important to note down the settings of different pieces of equipment.

Sometimes getting the right sound, or fixing a fault, can just be a matter of adjusting a setting.

Document Operating Procedures

Write down step-by-step operating procedures of different pieces of equipment. For example, write down a set of step-by-step instructions that explain how to use the mixer, and the CD player, and / or any other pieces of equipment.

Make sure all new staff and volunteers have copies of these notes, and learn from them. This will save a lot of time and trouble in the long run.

Filing Your Manuals, Diagrams and Notes

All of your equipment manuals and technical documents should be kept in one secure place and carefully filed. Your filing system should make it possible for you to easily find the information you need.

Filing all your equipment manuals in lever arch files, using a separate file for each studio and another for transmission and other equipment is a good idea.

Maintenance

The two most important things when considering how to maintain your studios are

• The level of technical skill at the station

• Your financial position and budget

A Little Knowledge Can Be a Dangerous Thing

People who are skilled broadcast studio operators should be able to perform routine maintenance tasks. If you have people with good technical skills at the station, or volunteers with radio technical skills living in the community, you will be able to do a lot of the maintenance work yourself.

But unless you employ a skilled technician, or have easy access to one, you will not be able to perform anything other than routine maintenance and basic repairs.

It is important to know how much skill the station has, and to be honest about this. Remember, a little knowledge can go a long way – but it can also be a very dangerous thing.

Costs of Maintenance

If you have limited technical know-how at the station or in the community, your financial position becomes important.

Professional repairs and maintenance services are expensive, and your

equipment suppliers can generally supply these services. Usually, the suppliers will expect you to send the broken piece of equipment to their offices. Or they will expect to visit the station.

Both these options cost money. Expenses will include

Courier costs: Generally, the equipment will have to be couriered to the suppliers or agents, who are most often based in Gauteng.

Transport: If the suppliers have to travel to the station, they will charge transport and accommodation.

Replacements: Parts are usually imported.

Labour: This is charged by the hour.

Rental: You may have to rent equipment to replace the broken piece while it is being fixed.

The high cost of repairs has meant that stations often just disconnect the broken piece of equipment and put it in a cupboard, where it is forgotten. This is wasteful, and can compromise your broadcast quality.

Planning for Maintenance

Some stations ask their suppliers to draft a maintenance plan for the studio, as part of the installation. This is a good idea. The plan will take the form of a contract with the supplier. For an annual or monthly fee the plan would include:

*   reasonable support by telephone

*   at least two site visits per year for maintenance and servicing

*   organising courier and repair of broken equipment

*   replacement of equipment, as necessary.

Maintenance Budget

While most major repair jobs are best left to your equipment supplier, there are many basic maintenance and cleaning tasks that you should learn to perform on your own. To do these, you'll need a budget. This means the station will have to set aside money on a monthly basis for maintenance.

Monthly Savings

The guideline is to set aside 10% of the full cost of the studios to cover maintenance over the first three years. Following this guideline, if your studio cost R400 000, you should set aside a total of R40 000 over three years for maintenance. This means about R13 000 a year, or just over R1 100 a month.

It is very important to be disciplined about this and to set aside an amount for maintenance from your income each month.

Even if you don't spend the money each month, you must save it. Remember, a CD player costs around R6 000 – it will take you several months to save this amount. Your station cannot operate without a CD player – if yours breaks down, you will have to repair or replace it immediately. You should always be prepared for things like this.

This amount that you save each month will increase as your studios get older, and wear and tear leads to more repairs and replacements being needed.

Example of a Monthly Maintenance Budget

The table on the next page gives you an idea of what monthly maintenance of your studios will cost. As you will see, in addition to the actual monthly costs, the budget allows for a saving calculated at 10% of R400 000 worth of equipment over three years.

Obviously, if your equipment has cost less, this amount will be less – for example, the monthly saving for equipment valued at R250 000 will be 10% of R250 000 = R25 000. Divide the R25 000 by 36 months (3 years) = a saving of R694 per month.

As you will see, there are three categories of items: once-off expenses; monthly expenses and monthly savings. We have used them as follows:

Once-off expenses: These are items that will last for a few years if you use them carefully and look after them.

Monthly expenses: These are items that you will have to replace regularly. We have taken the average annual cost, and divided by 12 to give you a monthly saving towards monthly expenses. You will not always spend this money every month, as you will not necessarily need a new duster, or light bulb every month. But you should save the money so that you have it when the need arises.

Monthly maintenance savings: This is the equipment service and maintenance budget calculated at 10% of your equipment value. This amount is over and above the monthly savings for items you will have to replace regularly.

Example of a Maintenance Budget for Studios Worth R400 000		
Item	**Price**	**Notes**
Once-off expenses	**Immediate expenditure**	**Items you buy and then keep for a few years.**
Vacuum cleaner	R700	A medium-priced vacuum cleaner will last long and you will be able to use it for the whole station.
Broom	R50	
Dustpan and brush	R35	For a set.
Total once-off expenses	**R785**	
Monthly expenses	Monthly saving	These are items that you will use up and have to replace regularly.
Vacuum cleaner bags	R5.40	For a pack of 5: R65 The number of bags you use a year will depend on dust and dirt levels in your station. We are estimating 1 bag a year.
Duster	R1.70	Budget for around 4 new dusters a year, @ R 5 each = R20.
Furniture polish	R2.35	Around 4 bottles a year @ R7 each = R28 / year.
RAG Light Bulbs	R7.50	Each one costs R15. Always keep spares on hand, budgeting for 6 / year = R90
Studio/Office Light bulbs / Fluorescent Tubes	R11.70	Always keep spares on hand. A pack of 5 100 Watt costs R20. Studios use a lot of light because they can be in operation 24 hours a day. At around 7 packs a year (including lights for general office use) = R140. Some studios have fluorescent lights, which are more expensive but need to be replaced less often.
Alcohol / Cleaning Fluid (for cleaning tape heads as per manual instructions)	R2	2 x 200ml bottles per year @ R12 each = R24 / year. You buy this from a chemist. Ask for denaturated alcohol surgical spirits.
Ear buds (for cleaning dust and dirt from tape heads and other equipment)	R1.70	4 packets per year @ R20 each.
Connectors (spares for re-placing faulty cables or making new ones as necessary)	R12.50	The average price of a connector will be around R25 (some are more expensive than others). Budget for about 4 new connectors a year, and come cable = about R150 / year, or R12.50 month.
Spare MD's/Tapes/CD-R	R25	Minidiscs cost around R40 each; a CD-R costs around R10. Cassette tape is around R10. The amount of recording media you use will depend on how much fieldwork and production you do. You may also use recording media to reproduce and distribute programmes to organisations in your community or other community radio stations. You may also want to archive some of your programmes. We estimate you will need around R300 / year, or a saving of R25 / month.
Tools	R20	Setting aside a small amount each month will make it easier to buy tools when needed.
Needles for Turntables	R25	These cost over R300 each, and should be replaced at least twice a year if your station uses turntables frequently.
Batteries	R20	For Remote Controls and Field Recorders. This will depend on use of these devices, but you will always need to have spares on hand.
Headphones	R70	All studio headphones will break eventually! If you save a small amount each month you will be able to afford a good quality set when the time comes.
Mouse	R10	The computer mouse usually offers a year or two of good service before it needs replacement. Mice cost around R120.
Total monthly expenses	**R214.85**	
Monthly maintenance savings	**R1 100**	Monthly amount for major repairs and replacements (calculated at 10% Of R400 000 over 3 years. This will cover repairs, maintenance and improvements for the first three years of your station's life. After that, you will need to increase this amount as breakdowns of the older equipment will be more frequent.
Total:	**R1 314.85**	**To be set aside each month.**

Maintenance is a Management Issue

Because maintenance is as much a financial issue as a technical issue, it is important that the station manager and the financial officer know and understand what is needed, and why.

Budgeting means putting the money aside. Even if you don't spend it each month – you should keep it, because you will eventually need it.

Maintenance Equipment

Cleaning Equipment

Because keeping the studio clean and dust free is so vital to maintenance, the most important maintenance tools are a vacuum cleaner and the everyday cleaning materials like dusters and furniture polish listed in the table above.

Many studios have failed simply because equipment has become clogged with dirt and dust. Studios must be regularly dusted and vacuumed, daily if possible, but not less than twice a week, to prevent a build-up of dust that can gather in equipment. Only once the studio can be kept clean should you consider buying more tools or maintenance equipment.

Basic Tools

Assuming your studios are cleaned regularly, a suggested basic toolkit for the studio would be as follows:

Suggested Toolkit	
Item	**Estimated cost**
Lockable Tool case	R300
Soldering Iron w/ stand, fitted with three prong plug	R50
Roll of Rosin Core Solder	R30
Solder Remover	R30
Spanner set (Sizes 6, 7, 8, 9, 10mm)	R50
Shifting Spanner	R50
Stanley Knife / NT Cutter	R50
Allen Key set (1.27, 1.5, 2, 2.5, 3, 4, 5, 6, 8, 10mm)	R50
Phillips Screwdriver Set (0,1,2)	R60
Flat Screwdriver Set (3.5, 4, 5.5, 6.5, 8mm)	R60
Precision Screwdriver Set	R20
Paint Brush (20 - 25mm)	R5
Side Cutters	R50
Long Nosed Pliers	R50
Wire Cutter	R50
Pliers	R50
Wire Stripper	R30
File	R20
Hack Saw	R40
Hammer – Ball Pin 100g	R60
Nut Driver Set (6, 7, 8mm)	R60
Tweezers	R40
Inspection Torch	R40
15m Extension Cord	R50
300mm Metal Ruler	R15
3m Tape Measure	R20
Multimeter that can measure continuity, resistance and AC/DC voltage.	R500
Total	**R1 830**
Consumables	
150mm Cable Ties	R5.00

This toolkit contains all the tools you will need for removing equipment from racks, making audio cables, general fault finding and basic repairs and maintenance.

Manage Your Tool Kit

The tool kit should be kept in a secure place and should only be used at the station. The technical manager should be responsible for the toolkit and ensure that tools are not lost or damaged.

Faults and Repairs

Sooner or later, a piece of equipment in the studio will fail. When this happens you need to get it repaired, as quickly as possible, so that the station can continue operating smoothly.

The first part of dealing with a fault is making sure that it is reported to the person responsible for maintaining equipment. Reporting faults as soon as they happen and taking action before they become too serious will save your station time and money.

The station's technical staff are not on duty all the time. This means a problem that arises late at night or over the weekend may go unattended for some time if nobody tells the technical staff.

To solve this problem, it is useful to keep a studio log book to note and list technical problems. The log book should stay in the studio and staff members on duty should write down any technical problems that they experience in the studio in the book.

This leaves a record to guide the technical department in its work. The log book will also provide a useful guide to the kind of training staff members and volunteers need – often people who struggle to get a particular piece of equipment to work just don't know how to operate it!

Repair Procedures

Can We Do It Ourselves?

Firstly, check to see if the problem can be solved by one of the station's technical staff. To do this you will have to check the manuals – and remember, be honest about your abilities. If you don't feel you can do the repair, then don't even try.

Involving the Supplier

If you can't fix it, contact your equipment supplier as soon as possible.

Guarantees or Warranties

Good suppliers provide guarantees or warranties for their installation work. This will normally cover the equipment and wiring for at least 12 months from the date of installation. Your equipment may still be under guarantee when it breaks or when something goes wrong. The sooner you contact them the better.

Describe the Problem in Detail

Be sure that when you contact the supplier, you can describe the problem in detail. For example, just saying that the CD player doesn't work is not enough information for them to help you. Be clear and exact in describing the problem. Say, for example, that the CD player does not always open and close properly

Being specific about the problem will save you and your supplier a lot of time and frustration. Also, if your supplier charges by the hour the less time they spend working on your problem, the cheaper it is for you.

Sending the Equipment Away

If the suppliers can't help you over the phone, you will have to send the equipment to their offices.

Ask for a Replacement

Always remember to ask for a temporary replacement to cover the period while your equipment is being fixed.

If the equipment has broken within the warranty or guarantee period, you won't have to pay rental for the replacement. But you will have to pay for transport.

Disconnecting the Broken Equipment

To send the equipment away for repair, you will need to disconnect it and remove it from its rack. If you aren't sure about how to do this, ask your supplier to talk you through the process of disconnection and removal on the telephone.

Always make sure that any cables you disconnect from the equipment are labelled, and make a note of exactly where they plug into the system to make connecting the replacement easier.

Make a Note of the Serial Number

Before packing the equipment and sending it away, make a note of its serial number. This number is unique to your equipment. When discussing the equipment with your supplier, you will refer to the serial number. When the piece is returned check the serial number to make sure that you have your piece of equipment back.

Packaging

Packaging the equipment is very important to prevent damage during transport. If possible, pack the equipment in its original packaging. This is the safest way to transport your equipment.

If the original box is not available, you will need to find a sturdy cardboard box of the right size, and bubble wrap to protect and cover the machine. At least two layers of bubble wrap are necessary to absorb the shock if the equipment gets bumped about in transit.

The equipment should be packed so that it does not move around in the box. If necessary, add in more bubble wrap and other packing material or newspaper to secure it.

Always seal the box completely, using packaging tape.

Never Send Manuals Away

Never send manuals away with equipment, unless the supplier especially asks for them. If you do send one away, make a note of it and be sure it is returned.

Courier

Always deal with a reputable courier company. The cost of loss or further damage to your equipment is far greater than the cost of paying a courier

Dealing with Service Providers

A vital part of your maintenance strategy will be maintaining a good relationship with your equipment supplier and other service providers, such as Telkom, Eskom, Sentech and local technicians.

Cultivating good working relationships with these providers and being able to contact them for advice and assistance will go a long way to easing the burden of maintenance.

But do not be afraid to demand good service of these providers. You are the client. You will be paying them – and so you can make demands for quality and speed of service.

Follow up with the providers if they do not deliver – it is your community that will suffer if service providers let you down.

Inventory and Markings

A key aspect of good maintenance is knowing what equipment your station owns and making sure it is secure.

An inventory – that is a full list of equipment, plus serial numbers - is an essential item.

All equipment must be insured. Your insurer will demand an inventory of your equipment before selling you insurance.

A typical inventory should contain detailed information about all of your equipment. Normally you would have an inventory of all the equipment in each room of the station. So you would have an inventory for the broadcast studio, the production studio and the voice booth.

Look at the example of an inventory on the next page.

Inventory of equipment in the broadcast studio				
Item	Brand or model; model number	Serial number	Value	Any other comment
CD player	Denon DNC630	D45320/0	R5 600	Bought in 2001
Minidisc	Sony MDSE11	MD43562	R3 500	Second hand, bought 1999

In many large organisations every piece of equipment is marked with its inventory number. It may not necessary to go this far in the community radio station. But permanently marking your equipment with your station's name may discourage theft and help identify an item should it be stolen.

Tips for General Maintenance

The following points are found on the front pages of virtually all equipment manuals. Often people are in a hurry to open the manuals – and don't bother to read them. So here they are again.

1. Read the Instruction manuals – All the safety and operating instructions should be read before equipment is operated.

2. Keep the Instruction manuals – The safety and operating instructions must be kept for future reference.

3. Heed Warnings – All warnings on equipment and in the operating instructions should be obeyed.

4. Follow Instructions – All operating and use instructions should be followed.

5. Cleaning – Unplug equipment from the wall power outlet before cleaning. Do not use liquid cleaners or aerosol cleaners. Use a soft, damp cloth for cleaning the exterior of equipment.

6. Attachments – Do not use attachments not recommended by the equipment manufacturer, as they may be hazardous.

7. Water and Moisture – Do not use equipment near water – for example, near a bath tub, wash bowl, kitchen sink, or laundry tub; in a wet basement; or near a swimming pool or any similar wet or damp place.

8. Stability – Do not place equipment on an unstable surface. The equipment may fall, causing serious injury to yourself or others, and serious damage to the equipment. Preferably all broadcast equipment should be mounted in a rack or stand, recommended by the supplier, or sold with the equipment.

9. Equipment should be moved and carried with care. Rough handling, quick stops, excessive force, and moving over uneven surfaces may cause the equipment to be dropped or damaged.

10. Ventilation – Slots and openings in equipment cabinets are provided for ventilation and to ensure reliable operation of the equipment and to protect

it from overheating. These openings must not be blocked or covered. The openings should never be blocked by placing the equipment on a bed, sofa, rug or other soft surface. Equipment should not be placed in a built-in installation, such as a bookcase or rack, unless proper ventilation is provided or the manufacturer's instructions have been obeyed.

11. Power Sources – This product should be operated only from the type of power source indicated on the marking label. If you are not sure of the type of power supply you need for the equipment, consult the supplier or local power company. For products intended to operate from battery power, or other sources, refer to the operating instructions.

12. Power-Cord Protection – Power-supply cords should be routed so that they are not likely to be walked on or pinched by items placed upon or against them, paying particular attention to cords at plugs, convenience receptacles, and the point where they exit from the equipment.

13. Outdoor Antenna Grounding – If an outside antenna or cable system is connected to the equipment, be sure the antenna or cable system is grounded so as to provide some protection against voltage surges and built- up static charges.

14. Lightning – For added protection for your equipment during a lightning storm, or when it is left unattended and unused for long periods of time, unplug it from the wall outlet and disconnect any antenna or cable system. This will prevent damage to the equipment due to lightning and power-line surges.

15. Power Lines – An outside antenna system should not be located near overhead power lines or other electric light or power circuits, or where it can fall into such power lines or circuits. When installing an outside antenna system, extreme care should be taken to keep from touching such power lines or circuits, as contact with them can kill you.

16. Overloading – Do not overload wall sockets or extension cords, as this can result in fire or electric shock.

17. Object and Liquid Entry – Never push objects of any kind into equipment through openings as they may touch dangerous voltage points or short-out parts that could result in damaged equipment, fire or electric shock. Never spill liquid of any kind on the equipment.

18. Servicing – Do not attempt to service equipment yourself. Opening or removing covers may expose you to dangerous voltage or other hazards. Refer all servicing to qualified service personnel.

19. Damage Requiring Service – Unplug this product from the wall outlet and refer servicing to qualified service personnel under the following conditions:

a) When the power-supply cord or plug is damaged.

b) If liquid has been spilled, or objects have fallen into the equipment.

c) If the equipment has been exposed to rain or water.

d) If the equipment does not operate normally by following the operating instructions.

e) If the equipment has been dropped or damaged in any way.

f) When the equipment exhibits a distinct change in performance – this indicates a need for service.

20. Adjust only those controls that are covered by the operating instructions. Improper adjustment of other controls may result in damage and will often require extensive work by a qualified technician to restore the equipment to its normal operation.

21. Replacement Parts – When replacement parts are required be sure the service technician has used replacement parts specified by the manufacturer or that have the same characteristics as the original part. Unauthorised substitutions may result in fire, electric shock, or other hazards.

22. Safety Check – Upon completion of any service or repairs to this product, ask the service technician to perform safety checks to determine that the equipment is in proper operating condition.

23. Wall or Ceiling Mounting – The equipment should be mounted to a wall or ceiling only as recommended by the manufacturer.

24. Heat – The equipment should be situated away from heat sources such as radiators, heat registers, stoves, or other products (including amplifiers) that produce heat.

A-Z

In this section we explain how many key pieces of studio equipment work, and some important audio terms and concepts. The A-Z will be especially useful for technicians, but it may also be useful for others wanting more detailed information.

We've arranged information in alphabetical order to allow quick access to topics of interest.

Contents

Acoustics and Acoustic Materials

All studios must have some form of acoustic treatment. Acoustic treatment stops sound from leaking into and out of the studio. Acoustic treatment takes the form of double doors, special walls, double-glazed windows and acoustic panels.

We could write a whole separate manual on acoustics and studio design. The mathematics of acoustics are quite advanced, and many sound experts have different opinions about acoustic treatment. But these things are important, and often misunderstood. So in the next few paragraphs we look at some of the very basics of acoustics.

Acoustics and Studio Design

There are two essential aspects to studio design:

Firstly, studios should be soundproof. This means sound from the outside must not leak into the studio, and sound from the studio must not leak out.

Secondly the studio should sound good. This means that when building a studio, you must to keep the room acoustically neutral. This means that sounds reflected off the walls, ceilings and windows are controlled in such a way that they do not influence the sounds heard by the person working in the studio.

Soundproofing Your Studio

Soundproofing is the easier part of acoustic treatment.

Firstly, you have to keep unwanted sounds out of the studio. When the studio is being used for broadcasting or recording, all noise from outside is undesirable. Any sound picked up by a live microphone, but which is not meant to be part the programme, will have a negative influence on the sound that is being broadcast or recorded. Ideally, if the equipment is not switched on, the inside of a studio should be absolutely quiet, with no sounds coming in from outside.

Secondly, you have to keep sound inside the studio, so that it will not be disturbing and distracting for others at the station.

How Soundproofing Works

The job of soundproofing is to restrict the paths available for sound to travel through.

Under S in the A-Z for more on sound and audio.

Sound travels through air. This means sound will get through any small opening in a wall, window or door in the studio. The first thing studio designers try to do is to make studios airtight, so that there is no direct contact between air inside and outside the studio. This means that sound travelling through the air outside the studio cannot get into the studio, and sound travelling through air inside the studio cannot get out. This is why doors and windows are sealed, or – even better – double doors and windows are used.

Using Mass and Dead Air to Stop the Movement of Sound

There are two ways to stop sound being transmitted from one space to another: dead air and mass.

Mass is solid material, like concrete or brickwork. The more solid or dense the material is, the better it will block the sound.

Dead air is trapped air, or air that cannot move from one place to another. The panes of glass used in double-glazed studio windows are usually sealed with foam rubber, silicon and/or expanding foam. Sealing both panes of glass means that the air is trapped between the panes, and so the air is "dead". Sound cannot travel through that air, as it cannot enter the sealed space.

This principle is also used when building studio walls and ceilings. These walls are often built with a sealed air gap inside them.

Many studios use "dropped ceilings" to stop sound from coming in through the roof. A dropped ceiling is a second ceiling built below the original ceiling. Both the original ceiling and the dropped ceiling are sealed, creating a volume of dead air above the studio that sound cannot travel through.

Vibrations

Loud sounds can travel through solid materials such as walls and floors, as vibrations. This means the studio must be isolated from sound sources that could cause walls and floors to vibrate.

An example is a studio near a road. A heavy truck passing by can cause the walls of the building to vibrate. This vibration will be carried into the studio as a rumbling sound through the walls, even though the room is airtight.

Studio designers resolve this problem by "floating" walls and windows and, if the budget allows, even whole rooms. For example, you can isolate a studio wall from the floor by placing thick rubber between them. While rubber is solid, it is also soft enough to absorb vibrations. The wall "floats" on the rubber, and makes it more difficult for the vibration to pass from the floor to the wall or vice versa.

Making Your Studio Sound Good

Sounds travel as waves, outwards from the sound source. When the sound wave comes to something like a wall, it will bounce off the wall, hit another wall, bounce off that one, hit another one, and so on. Sound that bounces off the walls, ceiling or floor of the studio is called reflected sound, or ambient sound. Making a studio "sound good" is about minimising ambient or reflected sound.

If you are putting up a completely new building for your station, you can use mathematical equations to calculate the best shape of the studio room. However, most studios are located in buildings that are already built, and most community radio stations can't afford to employ teams of acoustic scientists.

So it is more likely that you will ensure existing walls and ceilings are acoustically treated. This usually means placing acoustic foam tiles or acoustic panels on the walls and ceilings. These absorb the reflected sound and help the room sound better.

Another kind of acoustic treatment is called a "bass trap". "Bass" is low frequency sounds. Low frequencies tend to gather or accumulate in corners of studios. To absorb this low frequency sound energy, the corners in studios are often "flattened" by placing panels across them. You will often see perforated board and absorbent foam across the corners of studios – these are bass traps.
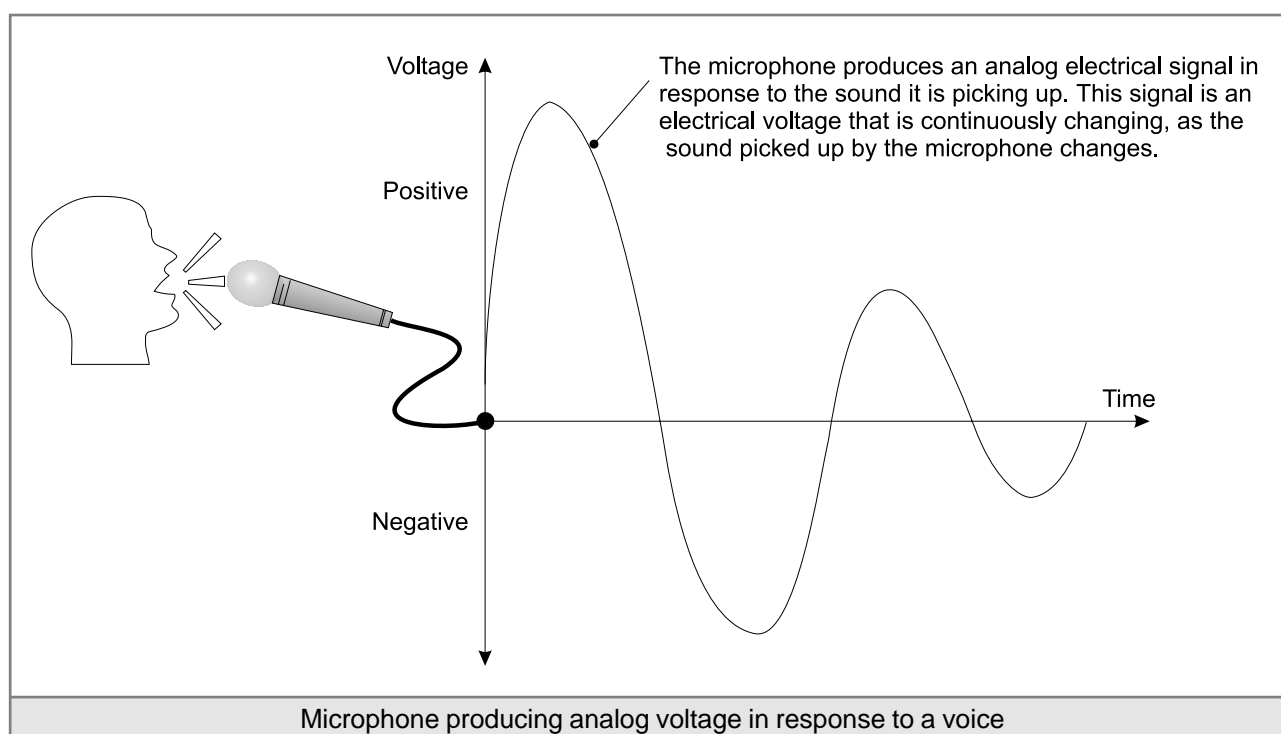
Analog and Digital

We explain on Page 149 how a microphone produces an electrical signal when someone speaks into it. It may be useful for you to read this section before getting into the details of Analog and Digital sound.

Under M in the A-Z for more on microphones.

Analog Signals

The electrical signal produced by a microphone is created by the vibrations of the diaphragm of the microphone. The diagram below shows how the sound of a voice will create a signal with varying amplitude (loudness), as the voice changes volume (gets louder and softer), and of varying frequency as the voice changes pitch (gets higher and lower).

The microphone produces an analog electrical signal in response to the sound it is picking up. This signal is an electrical voltage that is continuously changing, as the sound picked up by the microphone changes.

Microphone producing analog voltage in response to a voice

This signal is continuous or continuously varying (changing). A signal produced by a microphone in response to a voice is described as analogous to (the same as, or like) the voice. This continuous signal is an analog signal.

Converting from Analog to Digital

It is often useful to convert analog signals into a digital form. This conversion process is called analog to digital (A/D) conversion. When an analog signal is passed through an A/D converter, a digital signal is created. The digital signal represents the original analog signal as a stream of numbers.

Advantages of A/D Conversion

When signals are represented as digital data, or information (that is, data in the form of a stream of numbers), it is much easier to do calculations on the data using a computer.

It is also much easier to store large quantities of data as numbers.

One of the most powerful aspects of digital signals is that they can be transmitted with very little degradation (deterioration). An analog audio signal will always pick up noise of some kind during transmission by cable or radio. This noise cannot be removed. But if the signal is converted to a series of numbers that represent the signal at successive instants of time, and these numbers are transmitted as a digital signal, the analog signal can be reconstructed at the receiving end without any noise.

To summarise - converting to digital allows data to be stored as numbers. This allows us to change the data with ease using computers and other digital devices. Digital signals are also much less likely to be degraded or affected by noise. So conversion to digital is a good idea.

Community radio stations are using more and more digital equipment. For example, CDs store music digitally; so do MDs and DATs. All sound on computers is stored digitally, and more and more stations are using computers to edit and for playing out sound.

Many commercial and public broadcasters nowadays use exclusively digital equipment in their studios. So having skills and understandings around digital sound, and A/D conversion, is very useful.

A/D Conversion

The diagram below illustrates the A/D conversion process.

The signal voltage is sampled at regular intervals so that the original waveform can be reconstructed by effectively connecting the dots. The level of each sample is then conveted to a binary number for digital storage.

original waveform

Sampling an analog signal
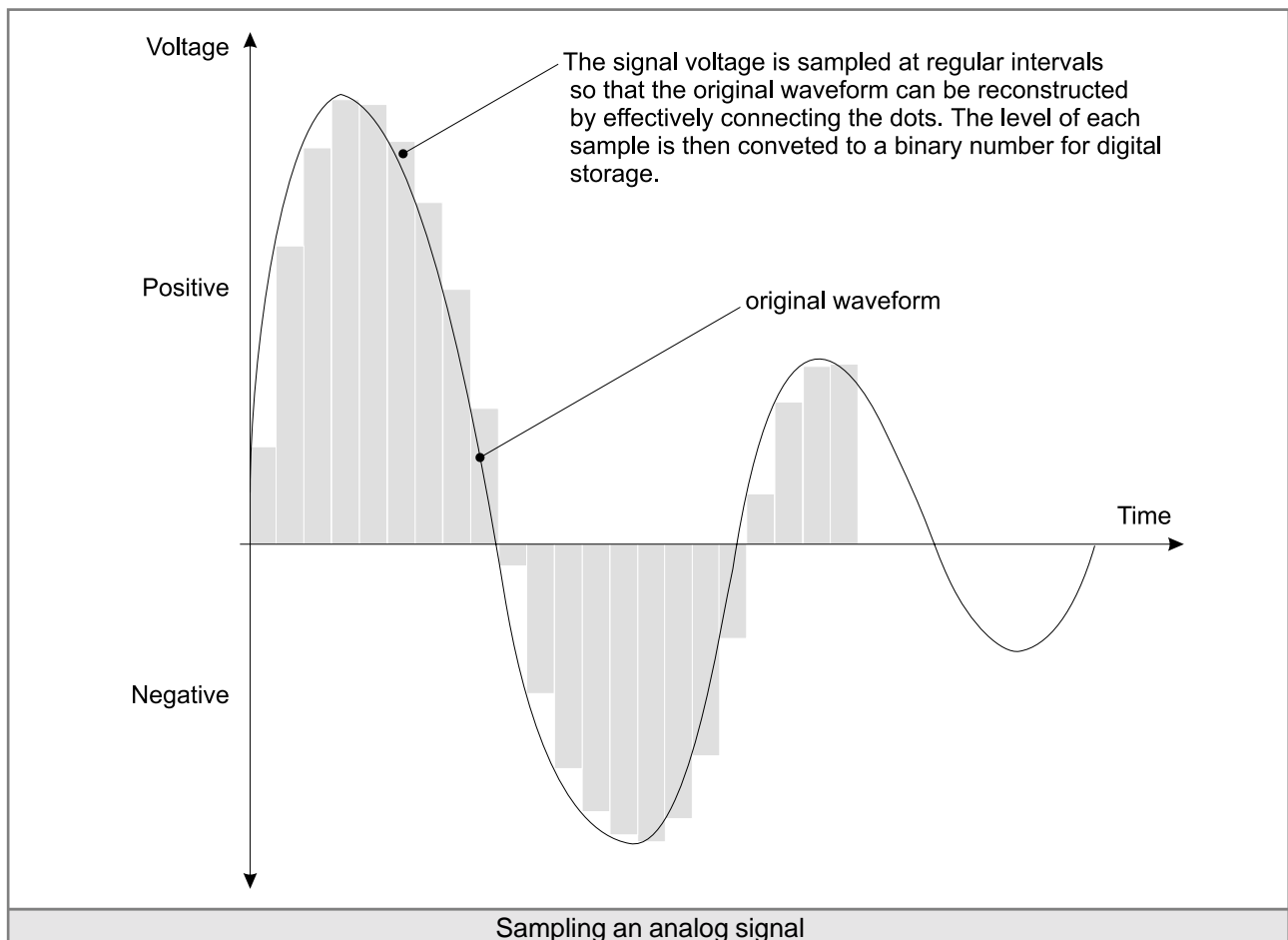
The A/D converter is fed an analog signal. The converter cuts the signal up into slices or samples, much like a loaf of sliced bread. This is called sampling. The converter then measures and records the level of each slice or sample as a stream of binary numbers. Digital systems use binary numbers – ones (1) and zeroes (0). In the stream, each 1 and each 0 is called a "bit".

Looking at the diagram, you will see that the more often the signal is sampled, or the thinner the slices, the more accurate the digital signal will be. This is more or less the case: if the A/D converter measures the original signal enough times per second, then the original sound can be accurately recreated.

Sampling theory states that if a signal is sampled at, at least twice the frequency of the highest frequency present in the signal, then the signal can be accurately reconstructed. The highest audio frequency is considered to be 20kHz. This means that if an audio signal is sampled at, at least twice that frequency, it can be accurately reproduced.

This is why the digital audio signal stored on a CD is sampled at 44,1kHz. A DAT, at normal speed, samples at a slightly higher frequency of 48kHz, resulting in slightly improved reproduction. Many broadcasters sample audio at 32kHz for FM broadcast. This is because most of the sound above 16kHz is lost in transmission and the lower sampling rate allows more digital audio to be stored on whatever digital medium is being used.

The audio quality is also affected by the accuracy with which samples are measured. The samples are represented as digital numbers. The greater the number of digital bits you use to represent each sample, the more accurate the measurement will be. CDs and DATs use 16-bit sampling, which is more than adequate for most audio purposes. However many multi-trackers and audio processors now use 20-bit or 24-bit conversion. Again, the more bits used to represent the samples the more accurate the reproduction.

Converting Back to Analog – or D/A Conversion

The process of converting digital audio back to analog audio is called digital-to-analog or D/A conversion. A conventional CD player is basically a specialised D/A converter. The CD player reads the 16-bit, 44,1kHz digital samples recorded on the compact disc, and uses them to reconstruct an analog signal that can be fed to an analog audio amplifier and loudspeakers, and heard as sound.

Amplifier

An amplifier is a device that amplifies the magnitude (size) of signals – that is, it makes signals bigger. To do this, the amplifier introduces "gain".

Gain is a term that many engineers will use in a studio. It is a measure of the ability of an electronic device to increase the magnitude of a given electrical input. In a power amplifier, gain is the ratio of input power to output power and is measured in decibels (dB).

It may be useful for you to find out more about decibels before reading further.

Under D in the A-Z for more on decibels

The diagram below shows how the amplifier works, by introducing gain.

Input Signal   Amplifier   Output Signal

Gain = 2

How the amplifier increases the magnitude of an electrical signal
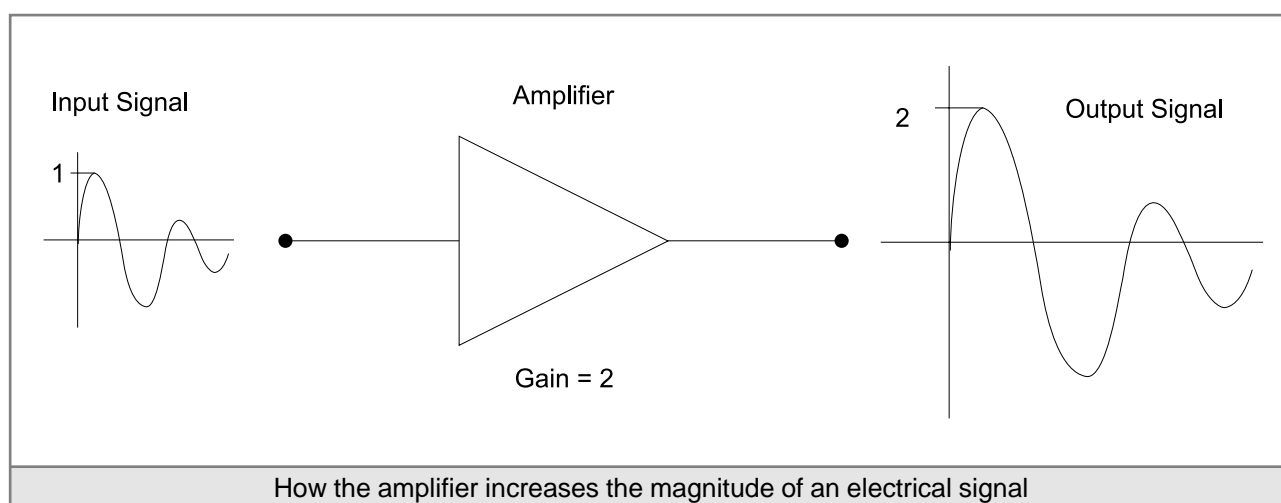
From the diagram it can be seen that for a gain of 2 (3 decibels, or 3dB) the amplitude of the output signal of an amplifier is double that of the input signal.

In radio, there are many occasions and instances where we need to increase the magnitude of signals. There are many different types of amplifiers to suit these needs.

There are some of the amplifiers most commonly used:

Audio Power Amplifier

An audio power amplifier is used to increase the power of an audio signal to a level that can be used to drive loudspeakers. In the studio the amplifier is fed a signal from the mixing desk. The amplifier amplifies this signal to drive the studio monitors.

Power amplifiers play an important role in live sound work, for example at concerts, or in churches and discos, or for outside broadcasts. For live sound, high power amplifier(s) are used to drive the large loudspeakers that are needed.

The controls and connections of a power amplifier are very simple. The amplifier usually has an ON/OFF switch and sometimes it has a gain (volume) control, although often an external device controls the volume. For example, the studio monitor volume is controlled from the studio's mixing desk.

Connections to the amplifier usually consist of the right and left channel input signals, and corresponding left and right channel outputs that are connected to the loudspeakers. The inputs use audio connectors such as RCA, Jack or XLR connectors. The outputs often use screw terminals. The bare speaker cable is often screwed into these terminals.

Under C in the A-Z for more on Connectors.

Distribution Amplifier (DA)

A distribution amplifier (DA) is a vital piece of equipment in any recording studio. A DA is used to distribute a signal or signals to many pieces of equipment.

A typical use of a DA in the studio would be to feed audio to all the recording devices. For example, a broadcast studio contains an MD recorder, a DAT recorder, a computer, a tape deck and a reel-to-reel recorder. Sound from the broadcast studio will also need to be fed to the production studio, voice booth and the station's background music system. This means that there are 8 devices that require a feed from the studio mixer's output.

It is not possible to connect all eight devices to the mixer, as this would overload the mixer's output. This is where the DA comes in. The mixer's recording output is fed to a DA. The DA splits and amplifies the signal from the mixing desk providing a feed of the signal to all 8 devices that are connected to the DAs outputs.

DAs can be used in different ways, depending on your needs. In the radio studio, a DA typically has a signal stereo output, taken from the programme or audition bus of the studio mixer. It then provides several stereo outputs that are connected to recording devices, effect processors and sometimes the transmission system.

Headphone Amplifier

A headphone amplifier works in a similar way to a distribution amplifier. Often we want to use more headphones in a studio than we have headphone outputs on the mixing desk. This is when we need a headphone amplifier.

The amplifier's input is normally fed from the headphone output of a mixing desk. It boosts and splits this signal to be able to drive 4, 6 or more pairs of headphones at the same time. Headphone amplifiers often allow individual volume control for each pair of headphones connected to the amplifier. They are commonly used in on-air studios and voice booths to drive the many pairs of guest headphones that might be needed.

Pre-amplifier

Pre-amplifiers are used to boost the weak output signals from microphones and phonograph (turntable) cartridges to the higher –10dBv or +4dBm signal level used by audio equipment.

The pre-amplifier is the first active stage – that is, the first electronic circuit that processes a microphone or turntable signal connected to a mixer. A microphone pre-amplifier is part of the input portion of a mixer's microphone channel. Microphone pre-amplifier's are also available as external units that are used to boost the microphone signal before connecting to a mixer's line input.

The phonograph pre-amplifier is a specific type of pre-amplifier. The pre-amplifier provides a special equalisation characteristic known as R.I.A.A. (Record Industry Association of America Equalisation) equalisation.

Under E in the A-Z for more about equalisation.

Because of the limitations of vinyl and record cutting heads, the R.I.A.A. standardised a curve that provides a 15dB cut at 20Hz and 20dB of boost at 20kHz, relative to a flat response at 1kHz when the records are made. When the record is played the, phono pre-amp applies a reciprocal equalisation curve. Because this special equalisation is required, a standard mixer input will not do a satisfactory job playing back audio from a turntable. Similarly playing through a phono pre-amp will severely colour the sound of a microphone or a CD player.

Balanced and Unbalanced Wiring

There are two types of output and input types used on audio equipment. These are balanced and unbalanced outputs and inputs.

Balanced inputs, outputs and wiring are expected on professional equipment, while unbalanced inputs and outputs are normally found on domestic audio equipment.

What Balanced Wiring Does

Balanced wiring helps eliminate external noise that can be picked up by an audio cable.

Balanced wiring uses three wires for a given audio signal, two signal conductors (hot and cold) and a common screen. The two signal wires of a balanced cable carry the same signal, but the signals in each wire are of opposite polarity. The diagram below represents this type of system.

Balanced wiring
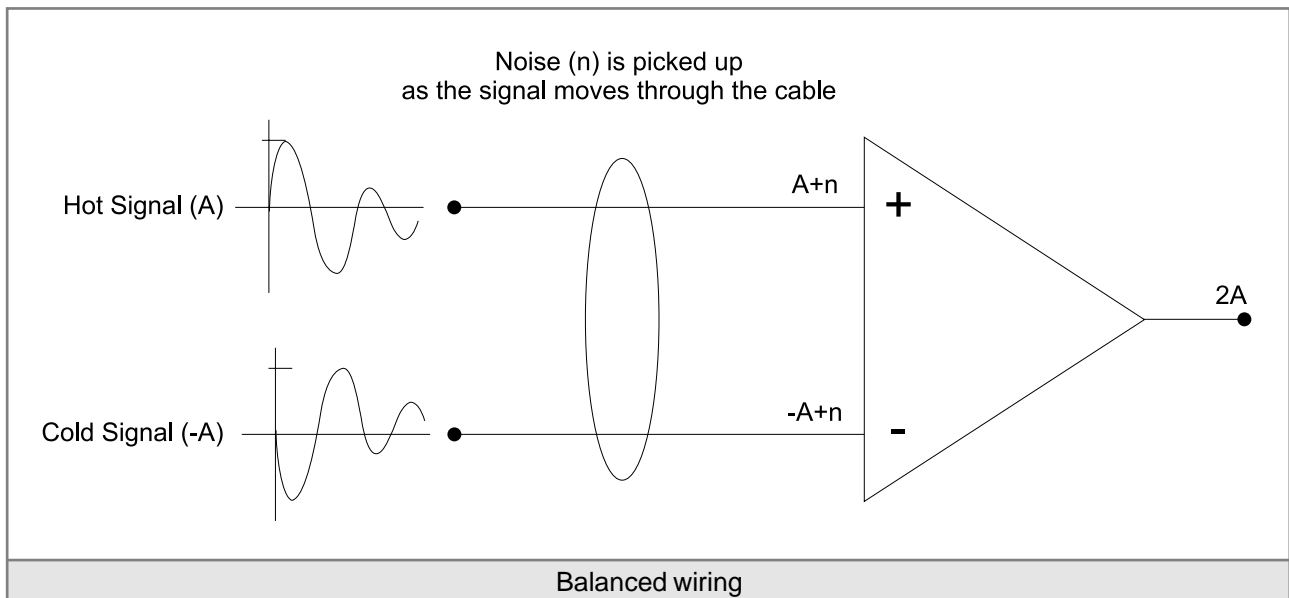
The two signals on the right represent signals leaving a piece of equipment with balanced outputs. As you see, the hot signal, labelled A, is of opposite polarity to the cold signal labelled as –A.

As the signals move through a cable to the next piece of equipment they pick up noise. The signals that have passed through the cable and arrive at the balanced input of the next piece of equipment can now be described as the original signals with noise added to them, hence the hot signal is now A+n and the cold signal is –A+n.

The balanced input is designed to calculate the difference between the two inputs, and this is the signal that the equipment then processes. This can be written mathematically as follows:

A+n – (-A+n) = A+n+A-n= 2A

Once the signal has passed through the balanced input, the noise component *n* is rejected. All that is left is an amplified version of the original signal.

For balancing to be effective, both the sending and receiving machines must have balanced output and input stages.

An unbalanced wiring system uses a two wire electrical connection in which a shield outer conductor forming a screen against interference usually surrounds the inner "hot" conductor. Any noise picked up by unbalanced wiring is not rejected.

The signal level in balanced systems is higher than in unbalanced systems. Balanced levels are referenced to +4dBm.

Unbalanced equipment uses -10dBv (-7.8dBm) levels, which are much lower.

Under D in the A-Z for more about decibels, dBm, dBV, etc).

The higher signals levels combined with the balanced inputs and outputs mean that balanced cables can be made as much as 500 metres long without a loss of quality. Long unbalanced cables are more likely to pick up hum and other noise.

It is important to be careful when connecting unbalanced equipment to balanced equipment. Most inputs and outputs can handle both signal signal levels. However, when feeding a +4dBm signal into a -10dBv input you need to be aware that the input signal level is much higher than the system expects. So care needs to be taken not to amplify the signal anymore in case you overload some part of the system.

If you feed a -10dBv signal into a +4dBm input, you might not have enough level and may have to amplify the signal at some point in the system. Connecting unbalanced and balanced inputs and outputs together is best done using a balancing amplifier. A balancing amplifier can convert balanced signals to unbalanced signals, and vice versa.

The diagram below illustrates a variety of balanced and unbalanced connections and their connectors.

Refer to C in the A-Z for more on connectors.

Balanced Connections:
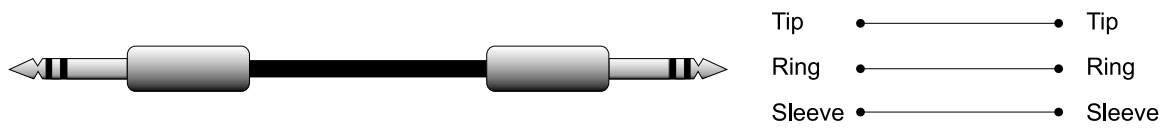
Tip ●————————● Tip
Ring ●————————● Ring
Sleeve ●————————● Sleeve

Balanced ¼"Jack plug to balanced ¼" Jack plug

Tip ●————————● 2 Hot
Ring ●————————● 3 Ring
Sleeve ●————————● 1 Screen

Balanced ¼"Jack plug to balanced XLR

1 ●————————● 1Screen
2 ●————————● 2 Hot
3 ●————————● 3 Cold
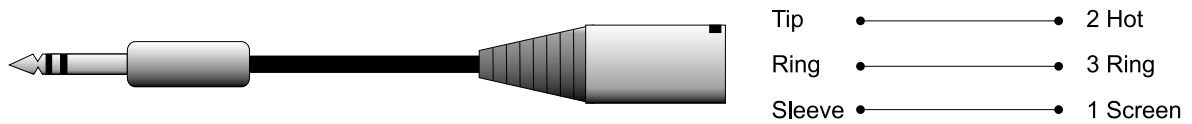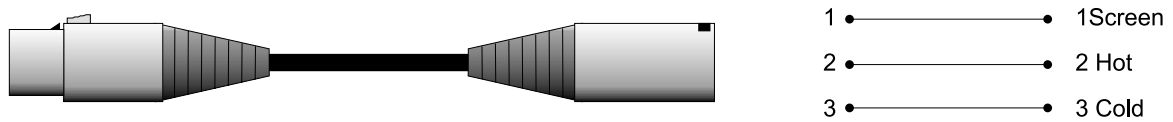
Balanced XLR to Balanced XLR

Wiring of different types of balanced connections

Unbalanced Connections:

Tip —————— Tip

Sleeve —————— Sleeve

Unbalanced ¼"Jack plug to unbalanced ¼" Jack plug

Tip —————— Tip

Ring

Sleeve —————— Sleeve

Sleeve left unconnected at unbalanced end

Balanced ¼"Jack plug to unbalanced¼"Jack plug

Screen left unconnected at unbalanced end

1 ——————

2 —————— Tip

3 —————— Screen

For some equipment pin 1 must be connected
to pin 3 for unbalanced operation.
Check this in your equipment manual.

Balanced XLR to unbalanced¼"Jack plug

1

Tip —————— 2

Sleeve —————— 3

Unbalanced ¼"Jack plug to balanced XLR

1

Tip —————— 2

Sleeve —————— 3

Unbalanced RCA to balanced XLR

Wiring of different types of unbalanced connections

Cassette Tape

The cassette tape, or compact cassette, is one of the easiest media for quickly recording and distributing sound. However, new media such as minidiscs and recordable CD (CD-R) is replacing cassette in the studio. The cassette is an analog medium that uses thin magnetic tape to store audio information.

The cassette comes from a long line of magnetic recording media.

- The original format was not tape at all, but thin steel wire, invented by Valdemar Poulsen in 1900.

- German engineers at BASF and AEG perfected the first form of magnetic tape, and the *magnetophon,* the first tape recorder in the 1930s. Tapes originally appeared in a reel-to-reel format.

- In 1948 the first radio programme was broadcast from magnetic tape. It was a music programme, "The Bing Crosby Show".

- Reel-to-reel tapes were common until the compact cassette or "cassette tape" was invented and patented by Philips in 1964. These tapes eventually beat other formats to become the dominant tape format in the consumer audio industry.

Both reel-to-reel recorders and cassette tapes are magnetic recording systems. The underlying principles of the two are exactly the same, but they do differ in operation, appearance and performance.

There are two parts to any audio magnetic recording system:

- the tape it uses as the storage medium.

- the recorder, which also acts as the playback device

Under C in the A-Z for more about the cassette/tape recorder

Under R in the A-Z for more on reel-to-reel players.

Features of the cassette tape

The three main features of the cassette tape are the box, the tape and the plastic cassette.

The tape cover

This is a plastic box, and has a hinged top that you open to remove the tape. The box also usually has a labelling sheet.

It is very important to label tapes that you are working with. People finding unlabelled cassettes lying around in the studio may record over them, and you may lose important sound that you will never be able to record again. Another reason is that you may lose your cassette and it will take time to listen to a whole lot of unmarked or poorly marked tapes to find the sound you need.

The tape

There are cassettes of various lengths. The ones most commonly sold in shops are 60-minute or 90-minute cassettes. A 90-minute cassette has 135 meters of tape inside it.

The tape is usually made of a thin plastic material. This plastic layer is coated with a ferric oxide powder. This oxide is a ferromagnetic material, meaning that if you expose it to a magnetic field, it is permanently magnetised by the field.

This ability gives magnetic tape two of its most appealing features:

• Firstly you can record audio instantly and the tape will "remember" what was recorded for playback.

• Secondly you can erase the tape and record something else on it.

The cassette

The cassette is a fairly simple protective covering and case for the tape. It has two spools around which the tape is wound, two rollers and the plastic outer shell. The outer shell has holes that hook the cassette into the tape player.

There is also a small felt pad that acts as a backstop for the record/playback head in the tape player.

The cassette also has two protective tabs on top. When these tabs are broken off, you can no longer record on the tape. Use a pen or something similar to gently break the tabs when necessary. If you have broken these tabs, but want to record onto the cassette again, you can cover the holes you have made by breaking the tabs with two small pieces of sticky tape. Be sure that the tape is securely stuck down, so that it cannot come loose while inside the cassette recorder.

Sound quality

Cassette tapes do not deliver the same quality as reel-to-reels. The wider tape used in reel-to-reels provides better audio performance. Reel-to-reel tape can also be easily edited by splicing (a system of editing sound by cutting out sections of tape with unwanted sound, and rejoining the ends where you want the sound to continue).

The relatively poor quality compared to CD and reel-to-reel, and lack of editing facility, mean many broadcasters believe cassettes are not appropriate for regular use. Nevertheless, the cassette tape's small size and the almost universal access to cassette tape decks in studios make it a very useful medium.

Looking after cassette tape

Because the tape is contained inside the cassette, compact cassettes are very robust. However to protect your recordings, you should handle tapes with care.

• Keep cassettes in their cases when not in use.

- Keep tapes away from magnetic fields. The tape is a magnetic medium, so exposing the tape to a strong magnetic field will affect the stored sound. In the studio, computer screens and loudspeakers often have strong magnetic fields. So leaving tapes near the computer, or on top of a loudspeaker is not good.

- Always label the cassettes you are using.

- Ensure that labels and stickers identifying tapes are not loose or peeling off. The labels can cause the tape to get stuck in the tape deck and will damage the machine.

Cassette/Tape Recorder

All cassette tape recorders – for example, field recorders, studio cassette decks, or the tape deck in a car - operate on the same basic principles.

Recording onto tape involves an electromagnet that applies a magnetic field to the oxide coating on the tape. The oxide "remembers" this magnetic field. The device that does this is the tape recorder's record head.

Playback involves reading the magnetic field back off the tape. The device that does this is the tape recorder's playback head.

The diagram below shows a tape recorder's record head.

Tape recording head
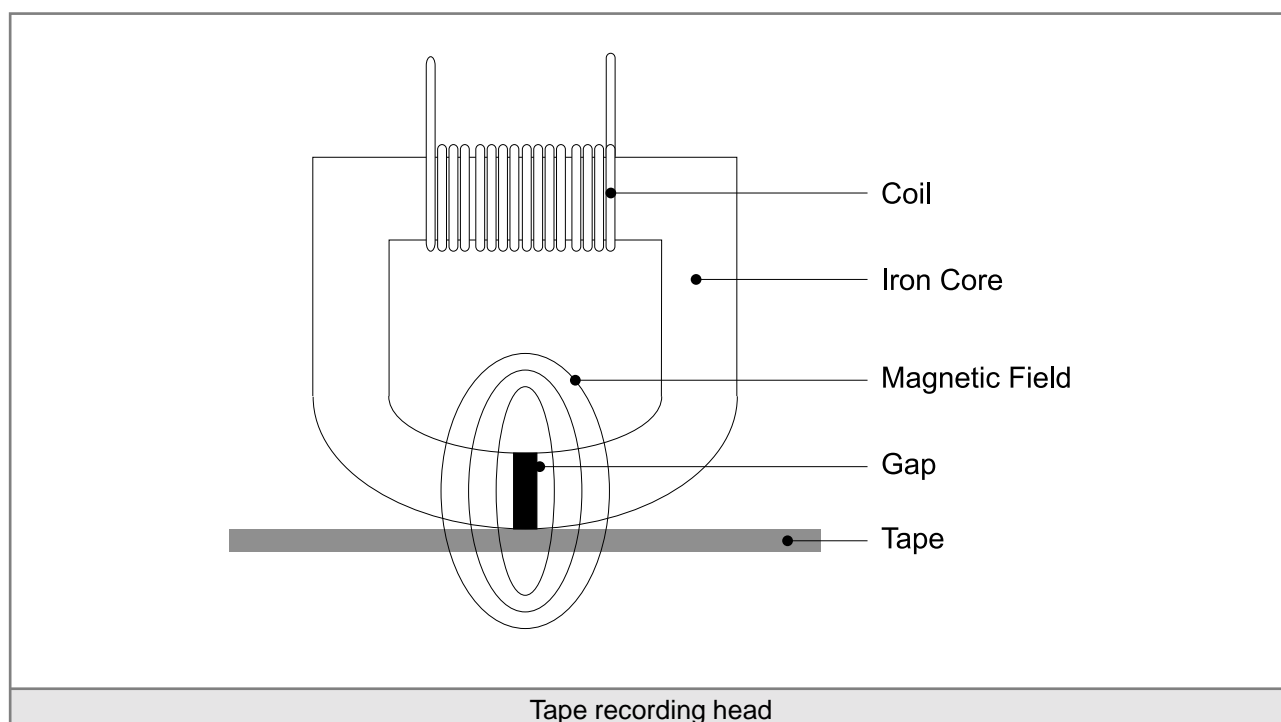
The tape recording head is a very small electromagnet with a tiny gap in it.

The electromagnet can be easily seen inside any tape deck. It consists of an iron core wrapped with several coils of wire. During recording, the audio signal is passed through the coil of wire. This creates a magnetic field in the core. At the gap, the magnetic field magnetises the oxide on the tape.

The process is reversed during playback. The motion of the magnetised tape across the gap creates a varying magnetic field in the core, and an audio signal in the coil.

In a normal cassette deck there are actually two electromagnets side-by-side, each equal in size to half of the tape width. This allows for recording and playing stereo.

When you look inside a tape recorder, you generally see something like this:

Capstan   Sprockets   Capstan

Pinch roller   Record and playback head   Pinch roller
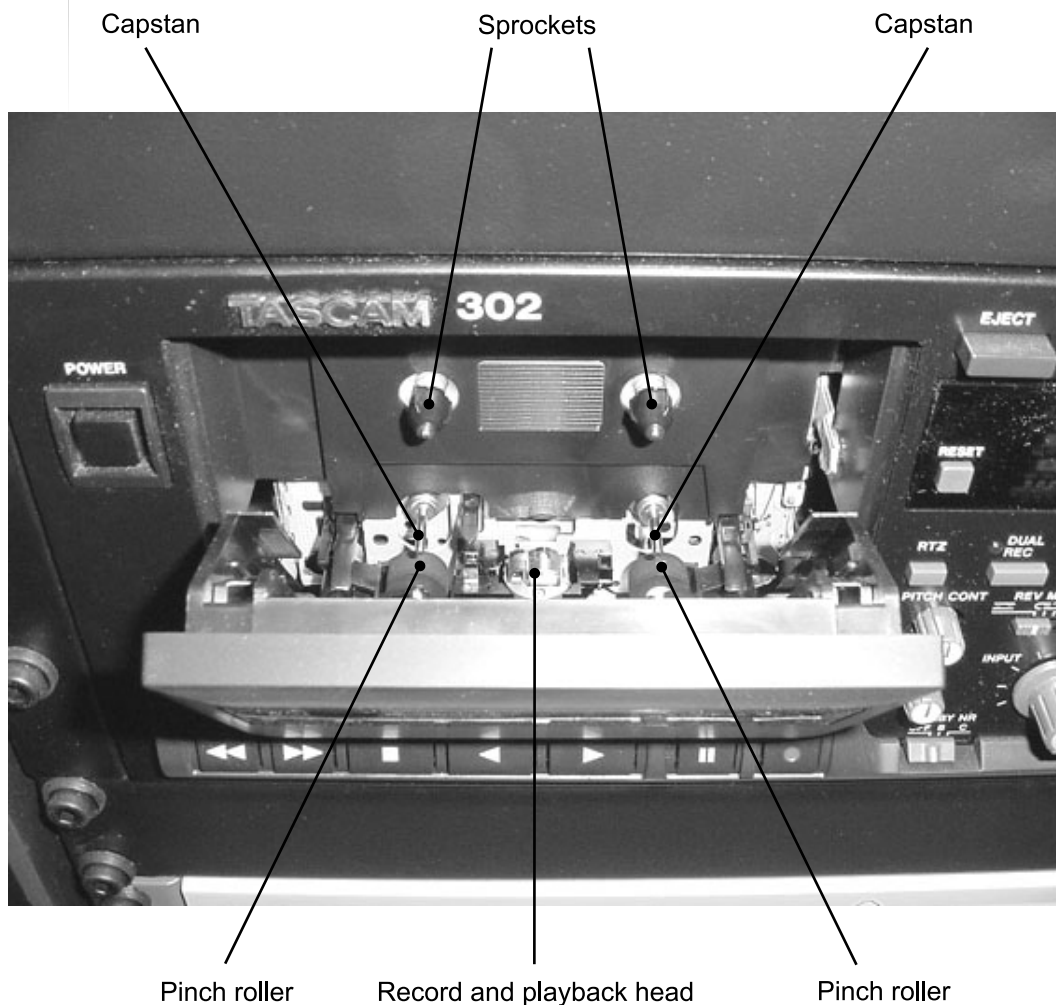
View inside a typical tapedeck

There are two sprockets that engage the spools in the cassette. These sprockets spin one of the spools to pull the tape during recording, playback, fast-forward and reverse.

Below the two sprockets is the record and playback head, containing the two tiny electromagnets.

On the right and left are the capstan and the pinch roller. The capstan revolves at a very precise rate to pull the tape across the head at exactly the right speed so that the sound is accurate. The standard speed is 1.875 inches per second (4.76 cm per second). The roller applies pressure so that the tape is tight against the capstan.

How to handle and maintain your cassette tape recorder

As with all the devices in the studio, you must read the user manual that is supplied for specific information.

The tape deck needs very little maintenance. The most important thing to remember is to clean tape heads and the compartment.

Cleaning the heads

The tape heads should be cleaned after every five hours of use. To clean the heads, take a cotton wool earbud and cleaning alcohol such as surgical spirits. Both earbuds and surgical spirits are available from most pharmacies.

Dip the earbud into the spirits so that is moist but not dripping wet. Then gently rub the earbud over the heads in the tape deck. Avoid touching the plastic and rubber parts of the device with the spirits.

Cleaning the compartment

The main dirt that collects in the compartment will be dust and fluff. It is difficult to avoid getting dust in the compartment, but you can minimise the problem by keeping the compartment closed when not in use and by regularly vacuuming the studio.

If you see any bits of fluff or other dirt, gently pull them out using a pair of tweezers.

Tape Types

Broadcast tape decks often have controls for different tape formulations. This relates to the type of oxide used to coat the tape. This is always labelled on good quality tapes. There are three types of tape in common use:

• Standard ferric oxide tape referred to as "Normal Bias". (Basic tape sound quality.)

• "Chrome" or CrO_2 tape. The ferric oxide particles are mixed with chromium dioxide. (Better than Normal Bias sound.)

• "Metal" tape. Metallic particles rather than metal oxide particles are used in the tape. (The best sound.) Metal tapes differ from Normal Bias and "Chrome" tapes as follows. You can play a metal tape on any tape player, but if you want to record, there must be a setting on the device for metal tape.

Look on your recorder to find the control for tape type, and always make sure you have set it to correspond to the kind of tape you are using. This will ensure the best possible sound.

Reducing unwanted noise

Most tape decks come with a noise reduction system to minimise unwanted noise during recording and playback. Dolby Noise Reduction is the most common system. Using the Dolby system properly will improve tape sound quality.

For the best results, a recording made using a noise reduction system must be played back using the same system. If your station chooses to use noise reduction, then all the tape decks in the station need to be set to use the same noise reduction system.

A problem can arise when tapes from outside the station have been made using use a different system, or none at all. When this happens, applying a noise reduction system to the tape will also reduce the quality of the recording, and in this case it is best not to use the noise reduction system.

Tape deck connections

Most tape decks have an unbalanced stereo output, although some more expensive broadcast tape decks have a balanced stereo output.

Under B, Balanced and Unbalanced Wiring, for a discussion of balanced and unbalanced signals.

RCA connectors are usually used for the unbalanced output. The left and right channels are usually colour coded, with red indicating right, and white left.

These outputs are connected to the tape channel inputs on the mixing desk. In some instances these can be passed through a balancing amplifier before being connected to the mixer. In some studios they may also be connected on the studio patch panel.

Under P in the A-Z for more about patch panels.

Tape decks also have a stereo input for recording. Again this is most often through unbalanced RCA connectors.

The tape deck is usually fed with audio directly from the recording bus of the mixing desk, or via a distribution amplifier connected to the recording bus. Some broadcast tape decks have a control input that allows the tape-deck to be controlled from the mixer.

Compact Disc (CD)

The compact disc (CD) was introduced in 1982. Philips, Polygram records and Sony jointly developed it. CD players were readily obtainable by about 1985 and this new digital audio playback system rapidly began to replace vinyl records, as the developers had hoped. Today CDs have nearly completely replaced records as the standard medium for listening to music.

The compact disc is a digital audio medium. The disc stores stereo 16-bit audio sampled at 44.1kHz.

Under A in the A-Z for more on Digital and Analog Audio.

The audio items such as songs on the CD are organised into tracks, and each track has a track number. Sony, Philips, and Polygram developed the specifications for CDs and CD players. These specifications were contained in the standards document referred to as the *Red Book*, published in 1982.

The *Red Book* specified the maximum playing time of a CD as 74minutes 33seconds, although many newer CDs are as long as 78 minutes. From this information, you can work out the total amount of data that can be saved on a CD:

44,100 samples/channel/second x 16 bits (2 bytes)/sample x 2 channels (left and right) x 74 minutes x 60 seconds/minute = 783,216,000 bytes per CD.

This is a lot of information.

A CD is only 120mm (millimetres) in diameter and about 1.2 mm thick. Most of a CD consists of a piece of clear polycarbonate plastic. During manufacturing, this plastic is impressed with microscopic bumps, arranged as a single, continuous, extremely long spiral track of data.

These bumps mark the binary numbers, that is, the 1's and 0's (bits), which make up the digital audio on the disc. In the case of a data CD that is used in a computer, the bumps mark the bits that make up the data files stored on the CD, otherwise it works in exactly the same way.

Once the clear piece of polycarbonate is formed, it is covered with a thin, reflective aluminium layer. A thin acrylic layer is then sprayed over the aluminium to protect it. The label and artwork is then printed on the acrylic.

The drawing below shows a cross section of a CD.

Cross-section of a CD (not to scale)
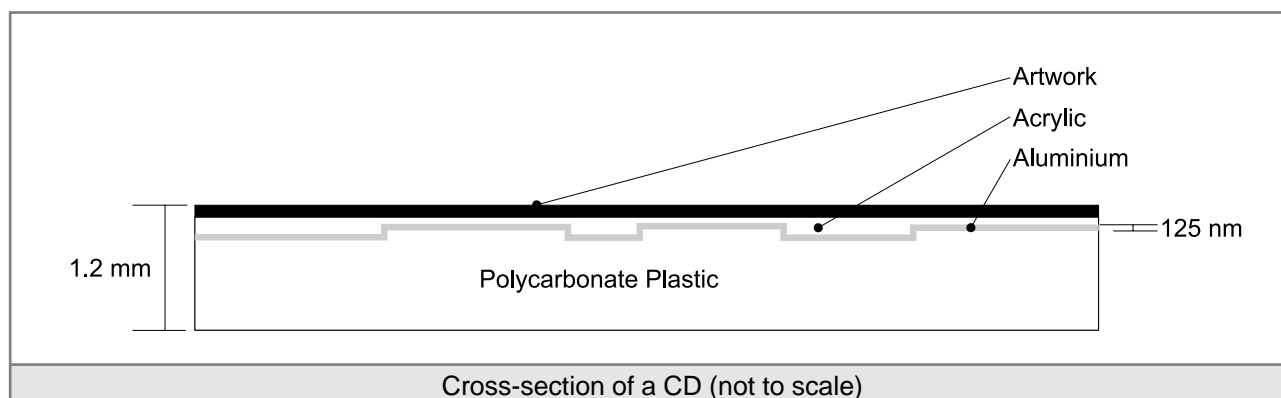
The data track on the CD spirals outwards from the centre to the edge, as shown in the diagram below. This is why it is possible to use smaller diameter discs. The spiral of data on a smaller disc still starts in the centre of the disc, but it ends earlier. So disc holds less data.

The data tracks are extremely small and are measured in microns. A micron is a millionth of a metre. The data track is about 0.5 microns wide, and the gap separating one track from the next is only 1.6 microns 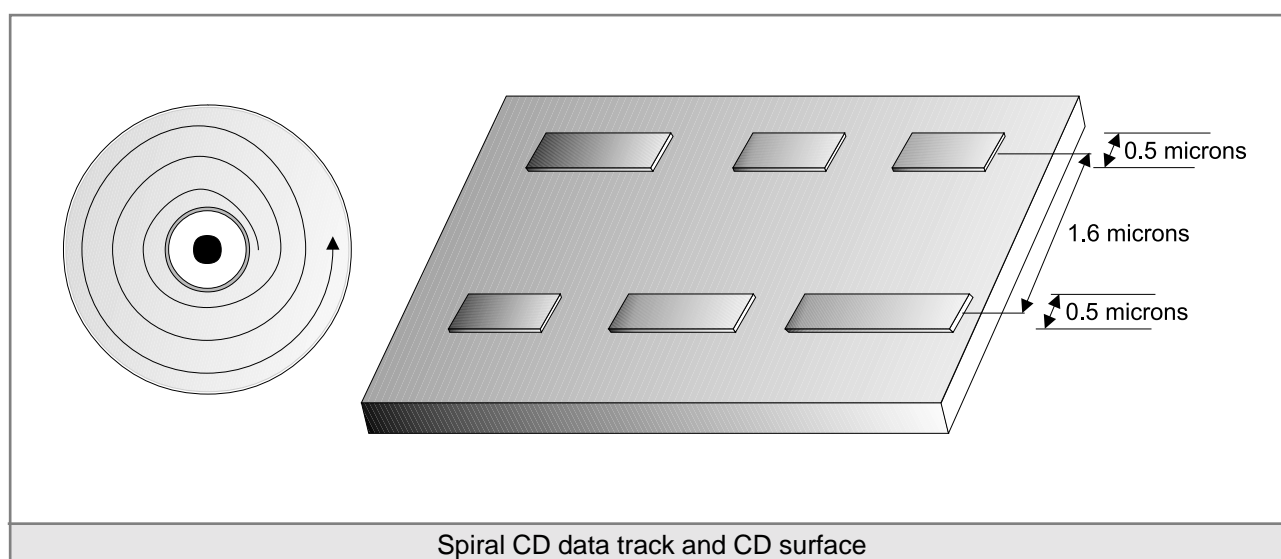wide. The bumps that make up the track are each only 0.5 microns wide, a minimum of 0.83 microns long and 125 nanometers high. A nanometer is a billionth of a meter. If you magnify the surface of the disc, it will look something like this:

0.5 microns

1.6 microns

0.5 microns

Spiral CD data track and CD surface

The microscopic sizes of the bumps make the spiral track on a CD extremely long. If you could lift the data track off a CD and stretch it out into a straight line, it would be 0.5 microns wide and over 5.5km long!

How the CD Player Reads the Disc

To read something this small requires an extremely accurate disc-reading mechanism. A CD player uses a low power laser to read the data track from the disc.

The disc spins in the player and a sensitive tracking system keeps the laser beam focussed on the spiral data track. The laser shines through the polycarbonate layer onto the aluminium backed data track. The shiny aluminium layer reflects the laser light back off the surface of the disc.

The light is either reflected straight off the aluminium, or is deflected off a bump in the data track. The CD player detects these differences. It registers the light deflected off a bump as a 1, and light that is not deflected (the reflected light) as a 0. In this way, the CD player creates the stream of digital information that can be converted into an audio signal.

The digital information on the CD is coded to ensure that as long as the CD is well looked after, it will work perfectly.

The CD also contains other digital data, called "sub-code". The sub-code contains a Table of Contents that organises the data on the CD. The sub-code also contains information about the tracks on the CD, such as the number of tracks and their duration. The sub-code makes it possible to find different tracks on the CD easily, and allows the CD player to display things such as the elapsed and remaining playback time of a track.

Handling Tips, and Cleaning CD's

The CD is a very delicate system, using the deflections of light off microscopic bumps to read audio. The CD is coded with a variety of error-correcting techniques to make it more reliable, but unless you look after your CDs, and handle them properly, they will be damaged and start "skipping" or not play at all.

- To take a disc from its case, press down on the centre of the case and lift the disc out, holding it carefully by the edges.

- Fingermarks and dust should be carefully wiped off the disc's recorded surface (the silvered side) with a soft cloth. CDs do not have any grooves that can collect dirt and dust, so gently wiping the disc with a soft cloth should remove most dirt. To clean the disc, wipe in a straight line from the centre to the edge. Wiping in a circular motion will scratch a large part of the spiral data track and will cause the disc to "skip".

- Never use chemicals such as record sprays, anti-static sprays, benzene, methylated spirits or thinners to clean CD's. Such chemicals can do irreparable damage to the disc's plastic surface.

- Return discs to their cases after playing to avoid scratches that could cause the disc to "skip".

- Do not bend the disc.

- Do not leave the discs in the sun, or in areas of high temperature or humidity. Long periods in high temperatures can cause the disc to warp.

- Do not stick labels on the label side of the CD. If you need to write on the label side of the CD, write very gently with a permanent felt-tipped or other soft-tipped marker. Do not use a hard-tipped pencil or ballpoint pen. Never mark or write on the recorded side of the disc.

Compact Disc Player (CD Player)

A Compact Disc player is used to play CDs. The machine consists of a very sensitive reading mechanism that reads digital information from a CD.

Under C (CDs) in the A-Z for more about how a CD player reads digital information from a CD.

The CD player also contains digital-to-analog conversion circuits.

Under A in the A-Z for more about digital and analog signals.

These digital-to-analog conversion circuits convert the digital data stored on the CD into an analog audio signal that can be fed into a mixer or amplified for a loudspeaker.

For many community radio stations, CDs are the main source of audio other than the microphone. As a result, there is an enormous demand on the broadcast CD player. These machines are often in continuous use for up to 18 hours a day, and are opened and closed thousands of times each month.

As we've pointed out in previous sections, a radio station environment demands ruggedness, reliability, and instant cueing from audio source equipment. This is why studio CD players are built differently, and have controls and other features that are different to those found on a domestic CD player. However, if you are familiar with domestic CD players, then a few minutes with the manual of a broadcast CD player will teach you all of the extra features.

The basic operation of a CD player is straightforward. It involves putting a CD in the machine, choosing the track you want to play, and then pressing the play button to start playback.

Consult the user manual for your particular CD player for specific instructions on its features and operation. The instructions given below are generally applicable but may vary for your CD player.

Inserting the CD

An open/close button opens the CD player's disc tray. Press it to open the tray and insert the CD.

Always place the CD in the disc tray with the label facing upwards. CDs can only be played on one side, and putting a CD in the wrong way round will cause the CD player to "freeze up" as it tries to read the CD. It will eventually give up trying, and you will be able to open the tray and take out the CD, but this wastes precious time.

Some exotic hi-fi players require the CD be put in the player label side down, but typically such a player would not be used in a radio studio.

When the CD is in the tray correctly, press the open/close button again to close the tray.

Do not push the tray closed, as this will damage the motor that opens the tray.
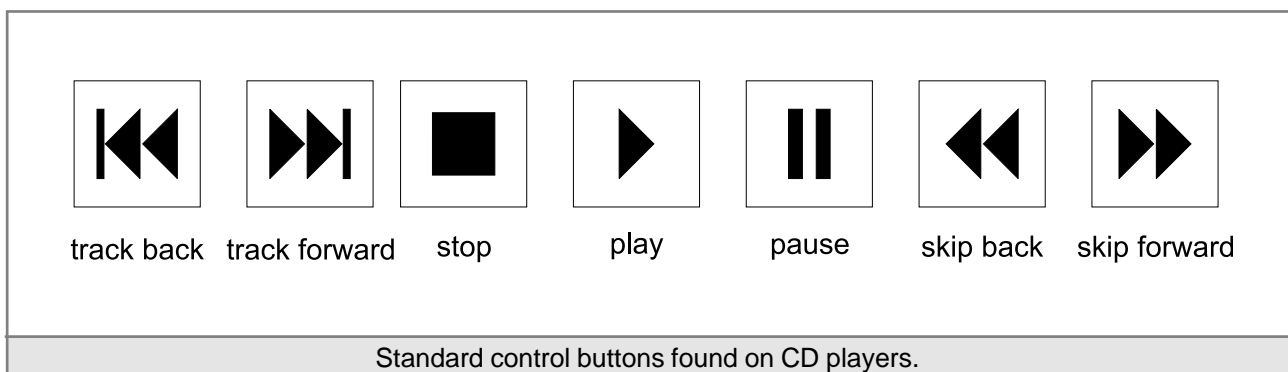
Damaging the motor may cause the player to refuse to open or close at all.

Avoid using special shaped CDs (heart shaped CDs, octagonal CDs, etc.). Trying to play them may damage your CD player.

Many broadcast CD players disable the open/close button if a CD is playing in the machine. This stops presenters from accidentally ejecting the wrong CD.

Selecting a Track

Choosing tracks on the CD involves pressing the "track" or "search" buttons. The diagram below will show you the symbols on the track and search buttons. These are standard symbols that are used on most recording and playback equipment.

| track back | track forward | stop | play | pause | skip back | skip forward |

Standard control buttons found on CD players.

Some CD players use a jog wheel instead of a button to select tracks. A jog wheel is a control that is turned instead of pushed like a button. Turning it clockwise (to the right) will increase the track number. Turning it anti-clockwise (to the left) will decrease the track number.

Playing the CD

Once you've selected a track, press the "play" button to play that track.

One of the marks of a good quality broadcast CD player is that when you press play, the machine starts playback instantly. There is no delay. These CD players "cue" the track when it is selected. This means the tracking mechanism in the machine immediately reads the sub-code on the disc and positions the read laser at the start of the track that is to be played ready for playback. These players often have a CUE button instead of a pause button, or an AUTO CUE control. This control flashes for the short time it takes the player to find the track. It then stays lit once the track is cued.

Cueing Tracks

Some CD players may take a few seconds to start playback and do not start instantly once you've pressed PLAY. This short amount of time is taken by the CD player to look for the track you've chosen. To minimise the "dead air" created while the player locates the track, you can cue the player as follows:

• Use the track controls to select the track you want, and then press PAUSE.

- Make sure the pause indicator is showing, or that the elapsed time is shown as zero.

- When you are ready, press PLAY – and the track will play instantly.

The SKIP Buttons

The SKIP buttons allow you to skip or scan through a track. Pressing these buttons will advance or reverse the track at high speed. This allows you to find particular parts of a track.

Broadcast CD players can skip through the track to within parts of a second, called frames. A frame is one seventy-fifth of a second. By searching frame-by-frame through a track, very precise cueing is possible.

Some players will also automatically skip past any silence at the beginning of a track, as part of "cueing".

The PAUSE Button

If the disc is playing and the PAUSE button is pressed, playback will pause at the point where the button was pressed. Pressing PAUSE or PLAY again will typically cause playback to continue from that point.

Many players use the same button for PLAY and PAUSE.

The STOP Button

Pressing the STOP button stops playback and clears any track selections. It is important to use the STOP button with care if you have pre-selected tracks.

Controlling the CD Player from the Mixer

Broadcast CD Players often use "fader start", allowing them to be controlled from the studio mixing desk.

In this case, the start button or the on-off buttons on the mixing desk can be used to play and pause the CD player.

The mixing desk and the CD player have to be capable of using this feature if you want to use "fader start".

Under M in the A-Z for more about mixers.

Counting the Seconds

In this section, the term "elapsed time" means the amount of time that has passed since the beginning of the track that is playing. The term "remaining time" means the amount of time left before the track comes to an end.

Normally, the CD player's display shows elapsed time. That is, it counts up the seconds and minutes that have passed from the time the track started playing.

A CD player used in a broadcast studio should also be able to display the remaining time, or the time left before the end of the track. This is an essential feature as it allows presenters to see how much time is left on a track, so that they can prepare the next item.

Presenters don't want to be surprised when the track comes to an end. They must be prepared for the end of the track so that there is no dead air time while the set up the next item.

You can choose whether elapsed time or remaining time is shown on the CD player's display by pressing a button marked TIME or DISPLAY.

Playing Tracks One At A Time

Another important feature of broadcast CD players is that they should be able to play tracks one at time. Domestic players play tracks continuously from the beginning of the CD to the end. It is very rare that a station will play an entire CD from beginning to end. Normally stations play only one track from a CD and then an item from a different source – another CD player, or the MD, or a microphone, etc.

It makes your job much easier if you don't have to worry about stopping the CD player while starting the next item. This is why it is important that the CD player is capable of playing one track and then pausing itself.

A button labelled SINGLE/CONT, or the AUTO CUE control, normally controls this. Pressing this button will put the player in single play mode - that is, it will play one track and then pause. Pressing the same button again make the machine continue with the next track. The play mode will be shown on the display.

Pitch Control

Some broadcast CD players feature pitch control. Pitch control allows the playback speed to be changed in the same way as on a turntable.

The Loop Function

A "loop" function allows a part of a track to be looped – that is to be played over and over.

Repeat

The "repeat" function allows a track, or the whole CD, to be played repeatedly.

Programming the CD

A programme play function allows the running order of tracks on a CD to be programmed.

The player manual will contain details of these features and instructions on how to use them.

Choosing a CD Player

After the mixer and the microphones, the CD player is possibly the most important item in a community radio studio. So it is important to choose the right CD players for your station.

As we've said, your CD players need to be rugged and reliable. They must be chosen for their ability to play even dirty or scratched CDs.

There are dramatic differences between CD players. There are many factors that are not easy to judge without experience. The experience of fellow broadcasters is valuable here. Ask around before buying CD players and see what other stations have to say.

CD-Rs

CD-Rs (compact disc-recordable) and CD-RWs (compact disc-rewritable) are increasing finding their way into many radio studios as the technology is becoming more affordable and easier to use.

Some community stations have CD-Writers, and many are using them to distribute programming.

The disadvantage of CD-RW is that most common CD payers cannot read them. CD-R can be read by almost any conventional CD player. A "finalised" CD-R looks to any CD player like an ordinary CD, provided that the disc has been "finalised" to record a final Table of Contents track on it.

Often CD-Rs are not properly finalised, and it may be worth investigating if your choice of CD player can play "unfinalised" CD-Rs, as this feature may become more useful in the future.

Under C in the A-Z for more about CD writers.

Maintaining and Cleaning Your CD Player

Maintaining the CD player involves keeping the unit clean both externally and internally.

The lens used to focus the laser beam that reads the CD and the optical sensor that picks up the reflections off the CD can gather dirt and dust. Using a "Cleaning CD" approved by the player's manufacturer once a month should prevent this from becoming a problem.

Also, keeping the CD tray closed and your CDs clean (no sticky fingerprints and dust) will help prevent dirt getting into the machine.

Outputs

Typically, broadcast CD players have two stereo outputs, one balanced and one unbalanced.

Under B, Balanced and Unbalanced Wiring, in the A-Z, for more about balanced and unbalanced signals.

These usually use male XLR connectors for the balanced output, and RCA connectors for the unbalanced output. The left and right channels are usually colour coded, with red indicating right and white indicating left.

These outputs are connected to the CD channel inputs on the mixing desk. In some studios they may also be connected on the studio patch panel.

Under M in the A-Z for more about mixers.

Under P in the A-Z for more about patch panels.

Most professional CD players also have digital outputs. Using these outputs, the CD player feeds the digital information from the CD directly to a digital mixer or the digital input of a digital recording device, such as a computer, minidisc or DAT recorder. This avoids any loss of quality that may result from D/A conversion.

Some broadcast CD players have a control input that is fed from the mixing desk. Standard CD players have no audio inputs.

Compact Disc-Writer

Today, writable CD drives (CD-Writers or CD-Burners) are standard equipment in most new computers. A CD-Writer allows you to take audio or data files from your computer and place them on CD. CDs containing audio can be played in a CD player, or if the CDs contain data, they can be used in a computer.

CD-Writers are relatively inexpensive, as are the blank CDs they use. CDs have rapidly begun to replace cassette tapes and floppy discs as the medium of choice for storing recordings and data.

Before reading further, turn back to the section on Compact Disc (CD) in the A-Z and read about CDs.

Under C in the A-Z for more on CD-ROM drives and computer hardware.

CD-ROM

CD-ROM stands for Compact Disc Read Only Memory. All standard audio and data CDs, such as pre-recorded audio or software CDs, are CD-ROMs. They are a digital storage medium that uses an optical laser to sense microscopic bumps on the disk. The bumps represent the bits of information stored on the disc. A CD-ROM, as the name says, is a Read Only medium. This means you can only read data from the disc, but cannot write any data onto it.

How CD-ROMs Work

CD-ROMs are mass-produced through a complicated manufacturing process. The pattern of bumps and flat areas that makes up the digital information on the CD is etched onto a glass disc. This is used to make a mould of the CD. This mould is used to press the pattern onto acrylic discs. The discs are then coated with aluminium to create the readable reflective surface. Finally, the disc is coated with a transparent plastic layer to protect the reflective aluminium layer. This elaborate process is not practical to use unless producing hundreds, thousands or millions of CD copies.

However as CD became the standard medium for audio, the demand for a simple CD recording technique grew. At the same time, computer users needed a medium that could store more data than floppy discs. In response to this demand, electronics manufacturers introduced the CD-R and the CD-Writer.

CD-R

CD-R stands for CD-ROM Recordable. A new CD-R does not have microscopic bumps on its surface. That is, it contains no digital data. The CD-R is coated with a special chemical film into which bumps can be burned using a CD-Writer.

How CD-Rs Work

As we've said, a CD-R does not have any bumps or flat areas at all. The disc has a smooth reflective aluminium layer that rests on top of a layer of light sensitive dye. When the disc is blank, the dye is translucent and light can shine

through it and reflect off the metal surface. But when the dye layer is heated, it darkens so that light can't pass through it.

By darkening particular points along the CD track, and leaving other areas of dye translucent, you can create a digital pattern that a standard CD player can read. The light from the player's laser beam will only bounce back to the sensor when the dye is left translucent, in the same way that it will only bounce back from the flat areas of a conventional CD. So, even though the CD-R disc doesn't have any bumps pressed into it, it behaves just like a standard disc.

A CD-Writer is used to burn this digital pattern onto a blank CD-R. A CD-Writer has two lasers: a standard "read laser" (like a normal CD player), and a "write laser." The write laser is more powerful than the read laser and is intense enough to darken the dye material on the CD-R. The weaker read laser does not affect the dye. This means that information on the disc will not be affected by the disc.

Speed of the CD-Writer

CD-Writers can create CDs at different speeds. At 1x (1 times) speed, the CD spins at about the same rate as it does when the player is reading it. This means it would take you about 60 minutes to record 60 minutes of music. At 2x speed, it would take you about half an hour to record 60 minutes, and so on. Current CD-Writers operate at 8x speed or faster. Faster writing speeds need a faster connection between the computer and the writer and a blank disc that is designed to record information at higher speed.

Advantages and Disadvantages of CD-R Discs

The main advantage of CD-R discs is that they work in almost all CD players and computer CD-ROM drives. CD-Rs are also the cheapest media available. At about R10 per blank disc, they are even cheaper than most high-quality cassette tapes.

The only drawback of CD-R is that once you've burned in the digital pattern, it can't be erased or re-written. It is possible to leave out some areas on the disc for later writing, but this creates a "multi-session" CD. A multi-session CD can not be read properly in a standard CD Player and some older CD-ROM drives.

To overcome the problem that the CD-R cannot be erased or rewritten, a new format was introduced in the mid-90s, called CD-RW

CD-RW

CD-RW stands for CD-ROM Rewritable. A CD-RW can have bumps burnt into the disc, but can also have them removed. The laser of a CD-RW writer, called a CD-Rewriter, can both burn bumps into the media and also melt the media back into its original state.

How CD-RWs Work

In place of the dye layer in the CD-R, a CD-RW disc contains a chemical compound that can change its form when heated to certain temperatures. When the compound is heated above its melting temperature, to around 600 degrees Celsius, it becomes a liquid. At around 200 degrees Celsius, it turns into a solid.

The solid form of the compound is translucent (light can show through it), while the liquid is dark. On a new, blank CD-RW disc, all of the material in the writable area is in the solid form, so light will shine through this layer to the reflective metal above and bounce back to the light sensor.

To write information on the disc, the CD-Burner uses its write laser, which is powerful enough to heat the compound to its melting temperature. These "melted" spots serve the same purpose as the bumps on a conventional CD and the opaque spots on a CD-R: they block the "read" laser so it won't reflect off the metal layer. Each non-reflective area indicates a 0 in the digital code. Every spot that remains solid is still reflective, indicating a 1.

As with CD-Rs, the weaker read laser does not change the state of the material in the recording layer. The erase laser falls somewhere in between: while it is not strong enough to melt the material, it is strong enough to heat the material to the solid point. By holding the material at this temperature, the erase laser restores the compound to its solid state, erasing the dark spot and the encoded 0. This clears the disc so new data can be encoded.

Advantages and Disadvantages of CD-RW

CD-RW discs do not reflect as much light as the other CD formats and cannot be read by older CD players and drives. Many new drives and players, including all CD-Rewriters can work with all the different CD formats. But since CD-RWs will not work on most existing CD players, these are not a good choice for music CDs. For the most part, they are used to back-up computer files.

CD-Rewriters can write to both CD-R and CD-RW discs, whereas CD-R drives can't write to CD-RW discs. CD-RW discs are, however, about the twice the price of CD-R's.

CD-Rs and CD-RWs are written using light and for this reason should be kept away from strong direct light, as this can corrupt the information stored on the disc.

Speed of the CD-Rewriter

The CD-Rewriter works at different speeds. In fact, the device is usually specified by its write speed, its rewrite speed and its read speed. For example a CD-Rewriter would be specified as 16x8x32. This means it can write CDs at 16x speed; it can re-write CDs at 8x speed, and can read data from a CD at 32x speed. The speed for re-writing is often slower than the write speed, as rewriting needs the extra step of first erasing the existing data on the CD.

Writing CD-Rs and CD-RWs

In the radio station, CD-Writers are most often used to create audio CD's. Typically, you may write a programme onto CD to archive it, or to distribute it to other stations. In these cases, the CD is intended for playback in a CD player and so you would use CD-R media, as most CD players can't read CD-RWs properly.

However, you may also want to make regular back-up copies of important computer files, such as the sound files used by your computer playout system. For this it is a good idea to use CD-RWs, as they can be re-written each time you make a new back-up.

Under D in the A-Z for more on digital playout systems.

Internal and External CD-Writers

Most often the CD-Writer or Rewriter is part of a computer, and the information written to the CD comes from a file stored on the computer. The CD-Writer can be fitted into the case of the computer (an internal writer) or can be an external unit that connects to the computer via the USB or PC Card connections.

Internal CD-Writers are relatively easy to install, and are about twice as fast and half the price of external writers. However external writers do have some advantages.

• Installing an external writer is much easier and you don't have to open up your computer's case.

• External writers are portable, this means that the writer is not tied to a particular computer and can be connected to any of the computers at your station. It is also a good solution if you want to use the CD-Writer with a laptop computer.

• External drives are usually compatible with different types of computers. If your station has an Apple Mac in the production studio and a Windows Computer in the office, an external CD-Writer would probably work with both.

CD-Writing Software

Using a CD-Writer attached to your computer requires CD writing software. Basic CD writing software will be supplied with the CD-Writer. The software allows you to decide what type of CD to make (audio or data) and choose the files you want to write to the CD. Programmes such as *Nero*, *Adaptec Easy CD Creator* and *Toast* are very popular.

If you do a lot of CD writing it is worth buying CD-writing software, as the basic software supplied with the CD-Writer can be limited.

'Making a Coaster'

Writing a CD places large demands on your computer and things can go wrong. Do not be surprised if some of the CDs you are burning don't come out right.

CD-Rs cannot be overwritten. This means that when an error occurs while writing the disc, you have to throw away the whole disc. This is often called "making a coaster" – as the only thing you can do with a damaged CD is use it as a coaster (the round mat you put under your glass or mug to protect the surface of the table).

If you continually have problems burning CDs, try to reduce the write speed. The most common problem when writing CDs is called a "buffer underrun". This happens when the computer is unable to send data to the CD-Writer fast enough. Reduction of the write speed can overcome this problem.

Separate CD-Recorders

Not all CD-Writers are part of a computer. There are a number of stand-alone CD recorders available for both domestic and professional use. Often, these have two drives that allow you to record music tracks directly from one CD to another.

Remember that copying pre-recorded CDs raises very serious copyright issues. Breaking copyright laws can lead to legal action that could cost you a lot of money.

These stand-alone recorders can also have audio inputs, allowing them to record incoming signals, much like a cassette or minidisc recorder. These writers are usually fast and accurate, but typically can only be used to create music CDs. Professional models are also very expensive when compared to computer-based writers.
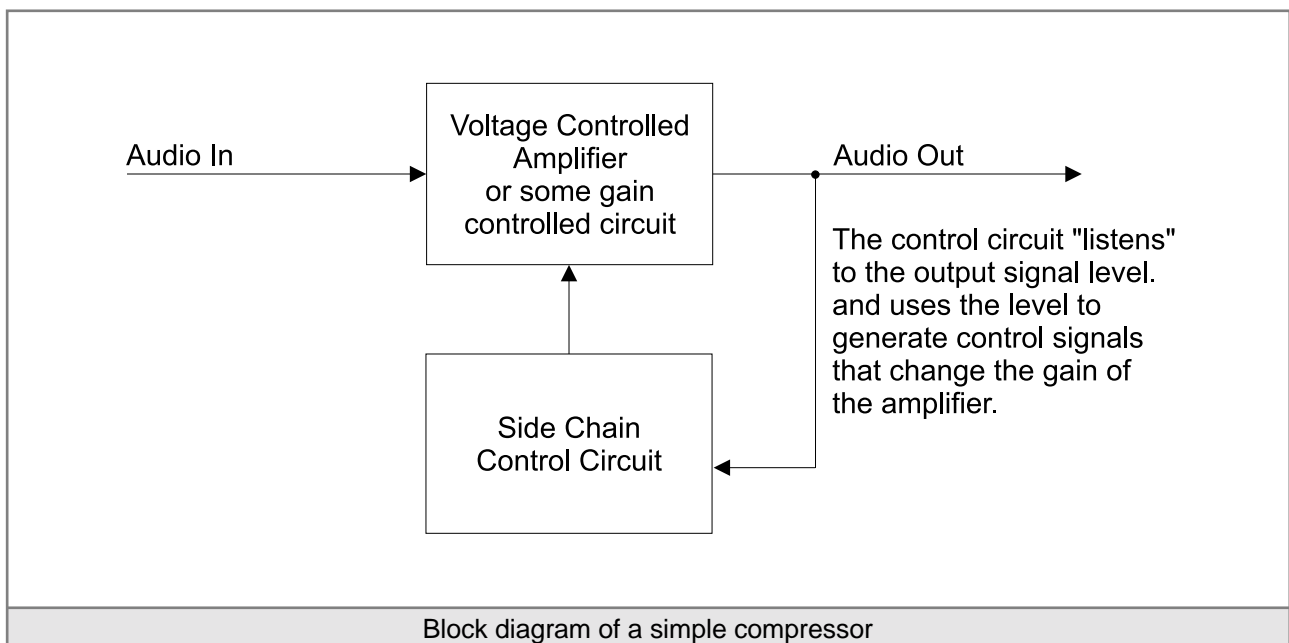
Compressor Limiter

A compressor limiter (compressor) is a device used in most community radio stations. The compressor limiter is used to process the signal that comes from the broadcast studio before it is transmitted.

What the Compressor Does

The compressor limiter keeps the level of your station's signal constant and at the best possible volume, both for your listeners and to compete with other stations.

In broadcast terms, we say the compressor is used to control the dynamic range of the station's sound. Dynamic range is the range between the loudest and quietest sounds that are broadcast. Reducing the dynamic range of the signal by using a compressor is called compression. The effect of compression is to make your station sound louder and clearer on air.

The diagram below shows how a compressor limiter works.

Audio In → **Voltage Controlled Amplifier or some gain controlled circuit** → Audio Out

Side Chain Control Circuit

The control circuit "listens" to the output signal level. and uses the level to generate control signals that change the gain of the amplifier.

Block diagram of a simple compressor

Not all compressors work exactly as shown above. For example, some compressors monitor both the input and the output audio. Also, there is computer software to carry out the role of the compressor. However the underlying principle remains the same.

The control panel of a typical compressor is shown below.

Gain Reduction		Output Level

Threshold   Ratio   Attack   Auto   Release   Gain   Stereo Link   Bypass

Typical compressor control panel
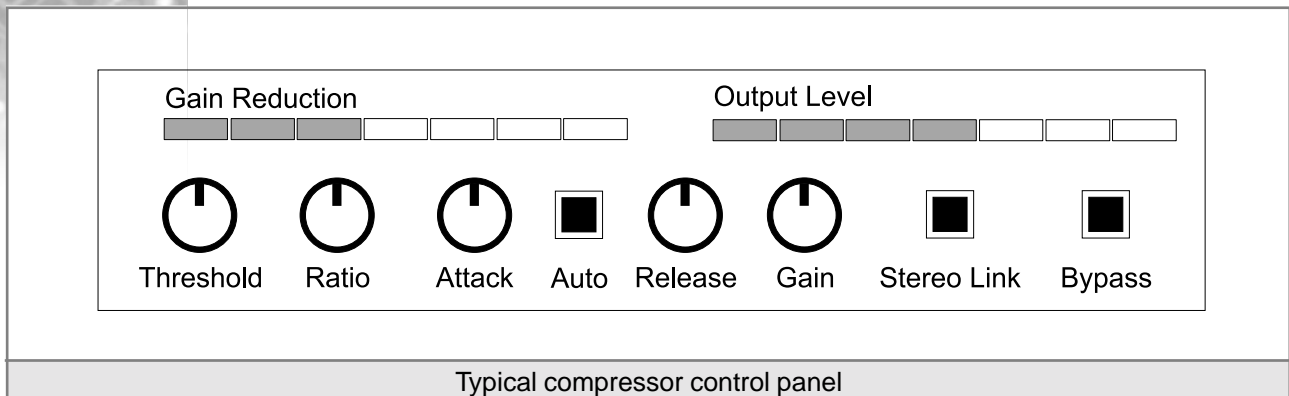
As with all equipment, we recommended that you consult the user manual for your compressor for specific instructions on its features and operations. The instructions provided here are generally applicable, but may be different for your compressor.

Controls and Meters

From the diagram, you can see that the compressor control panel has two meters:

- one meter (on the left) indicates the amount of gain reduction or compression

- the other meter (on the right) shows the signal level of the compressor's audio output.

Threshold Control

The threshold control is used to set the signal level at which you want the compressor to begin reducing it.

Ratio Control

The ratio control controls the change of output level for a given change in input. For example a compression ratio of 2:1 means that for a 2dB (dB = decibel) increase in input signal level, the output signal will only increase by 1dB.

The diagram below shows how different compression ratios affect the output signal.

Threshold and Compression Ratio
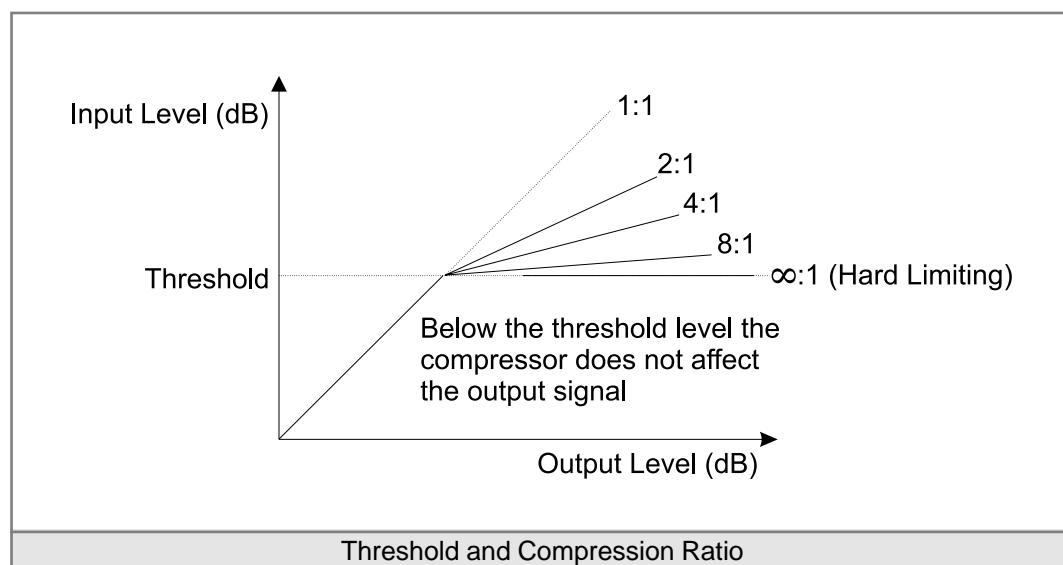
As you can see, the higher the ratio, the less the output will increase for a given input. Once the compression ratio exceeds 8:1, the output signal will increase very little, even if the input signal increases dramatically.

At around the 8:1 ratio, the compressor begins to act more like a limiter. This means that the compressor limits the signal at the threshold level, so that no matter how large the input signal becomes, the compressor's output stays the same (at the threshold level). Some compressors have a separate peak limiting control that can be set independently of the compression ratio.

Attack and Release Controls - Setting the Compressor Limiter

The speed at which the gain is reduced in response to an increase in input signal level is called the "attack time". This is usually specified in milliseconds, and is set using the attack control.

The speed at which the gain is restored to its original level after the input is removed is called the "release time". This is set using the release control.

You need to listen to your signal very carefully in order to set these parameters. If the attack time is set too fast, the compressor will respond to even the shortest peaks (loudest sounds), causing the level to change very quickly.

For example, the beats in a dance track often cause short peaks in the signal. If the attack time of the compressor is so short that it responds to the individual beats, the compressor will reduce the signal level on each beat, and produce a very unnatural sound. On the other hand if the attack time is too slow, the compressor's output may exceed the desired maximum before the compressor acts on the signal.

Too fast a release time causes "pumping" or "breathing" as the gain changes rapidly. If the release time is too short, quiet sections of music will be lost as the compressor will still be reducing the gain, even though the loud input signal is no longer there.

The Auto Switch

Many compressors have an "auto" (automatic) switch for the attack and release time. When the auto switch is on, the attack and release times are set dynamically, based on the input.

Attack and release settings used for speech vary from those used for music. You will also need different attack and release settings for different types of music.

Community radio stations broadcast a mixture of talk and music programmes, and broadcast different kinds of music (gospel, pop, kwaito, house music). A compressor that automatically adjusts attack and release is therefore ideal for a community radio stations.

The Gain Control

Extra "make up" gain is normally provided so that the output level can be matched to any subsequent pieces of equipment. This is controlled by the "gain" control. The output level resulting from using the gain control can be monitored on the output level meter.

Under A (Amplifier) in the A-Z for more about gain.

The Stereo Link Button

If you are using a compressor to feed stereo audio from your studio to your transmitter, then using the stereo link button can be useful. The stereo link button links the two channels of the compressor together so that both channels process their signals in the same way.

Not all compressors have this facility. If your compressor doesn't have a stereo link button, you will have to set controls for each channel to exactly the same level.

The Bypass Button

Pressing the "bypass" button bypasses the compression circuitry, and allows you to quickly compare the compressed and uncompressed output.

To Sum Up….

The compressor has a variety of possible settings. Small adjustments to any of them can make a considerable difference to your station's on-air sound.

It is a good idea to document the compressor settings that work best for your station and compressor. Keep this record available for reference.

Also, once the compressor has been set for your station, put a guard or protective covering on the front panel to prevent anyone from accidentally changing the settings.

The compressor settings you choose will depend very much on your preferred sound. However, many stations have been happy using

- a threshold level of 0dB

- compression ratio of between 1.5:1 and 4:1

and setting the output gain so that the output level meter reads the ideal input signal level for subsequent equipment (often 0dB).
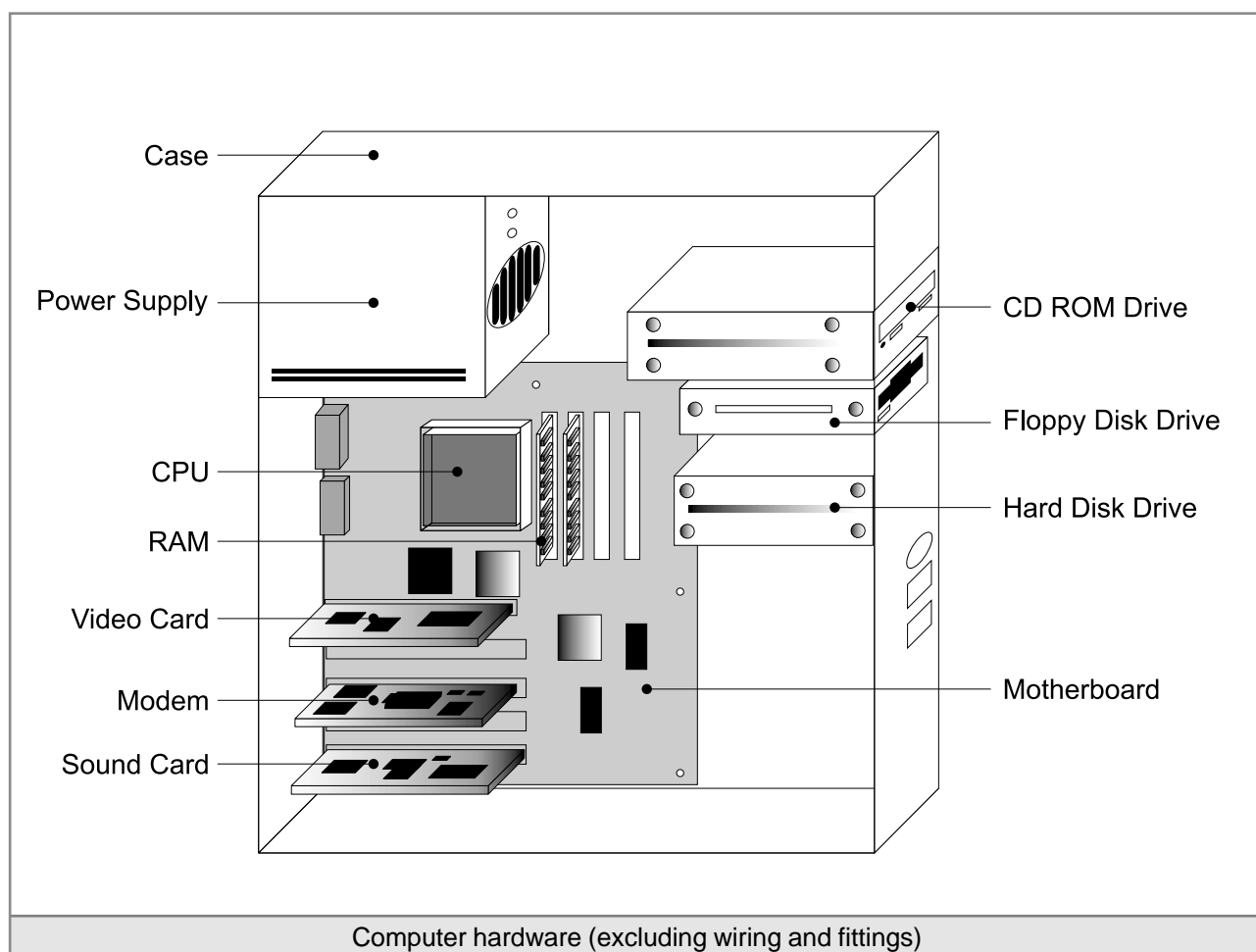
Computer Hardware

There are two vital components to any computing system:

- **The hardware.** This is the wiring, electronic circuits, disc drives, monitors, keyboards and so on that make up the machine we call a computer.

- **The software.** This is the term used to describe the programmes that we use on the computer.

Hardware and software must work together if we are to get anything useful done with a computer.

Your choice of computer hardware will differ according to your needs and according to the kind of software you are planning to use on your computer. For example if you use an Apple Mac computer then the hardware and software that make up the computer will be different to those of a Windows PC.

In general, however, all computers share certain basic hardware. The diagram below shows the hardware found in a typical computer. For clarity in the diagram, we have excluded other hardware, like the monitor (screen) keyboard and wiring.

Case
Power Supply
CPU
RAM
Video Card
Modem
Sound Card

CD ROM Drive
Floppy Disk Drive
Hard Disk Drive
Motherboard

Computer hardware (excluding wiring and fittings)

The Case

All computers need a case to house all the electronic components. Cases can come in a variety of colours and shapes, but all of them have the same purpose – that is, to hold and protect the electronic components.

The computer's power supply is normally supplied with the case.

The Motherboard

The motherboard is a large circuit board. All the other components are plugged into the motherboard. The motherboard supplies power to all the components and connects them together.

Motherboards vary in terms of speed and features. More expensive boards tend to be faster, more reliable and support more features and advanced components.

The motherboard is mounted in the computer's case.

The Microprocessor or Central Processing Unit (CPU)

The CPU can be thought of as the brain of the computer. It is the part of the computer that does the actual computing, and co-ordinates the actions of the whole system. Software programs are written to give the CPU a set of instructions. The CPU will follow the instructions to accomplish a specific task.

Like all other components, the CPU is connected to the motherboard. It is easy to spot on the motherboard, as it usually the largest microchip on the board. This is why it is often called the "chip" for short. The CPU is covered by a cooling fan.

Computers are differentiated by the type and speed of their CPU. For example, in terms of type, people may refer to a computer as a 486, a Pentium, Pentium II or Pentium III. This describes the different type of CPU used.

In terms of speed, people may refer to a computer as a Pentium III 600. The 600 refers to the speed of the CPU. In this case, the CPU has the capacity to operate at a frequency of 600 MHz. The CPU is capable of 600 million operations per second.

Memory (RAM and ROM)

RAM

The CPU has to process digital data to carry out instructions. This data has to be stored so that the CPU can quickly retrieve the data, process it and save it again for further processing. This storage space used to hold the data that the computer is working with at any time is referred to as the computer's memory.

Memory is like the CPU's scrap paper - somewhere to write down notes and calculations as it works. It is referred to as Random Access Memory (RAM). RAM is storage space that is available for short-term storage of data.

RAM relies on a constant presence of electrical charges, and operates only when the computer is turned on.

Computers are also differentiated by the amount of RAM they contain. The more RAM in the computer, the faster it works.

ROM

The computer has another kind of storage called Read Only Memory (ROM). ROM contains data that is permanently etched onto a chip, and typically stores the commands necessary for a computer to boot up, or start.

Storage Devices

In addition to RAM and ROM, computers have secondary storage devices such as floppy or stiffy discs, hard drives, and CD-ROMs.

These devices are responsible for long-term storage of data and software programmes. They can hold much more data than RAM and ROM, and are much less expensive.

They are also much slower than the primary storage devices. Data stored in memory can be accessed by the CPU in nanoseconds (a nanosecond is a billionth of a second), while data on a hard drive is accessible in microseconds (a thousandth of a second). This means your hard drive is about a thousand times slower than RAM!

Most computers currently come equipped with at least 32 megabytes (32 million bytes of data) of RAM, and have hard discs capable of storing several gigabytes. A gigabyte is a thousand megabytes.

Video Card and Screen

The video card and screen allow us to communicate with the computer. We need to give the computer instructions and data before it can do anything. We then need to see the results of the instructions displayed in a way that we can understand.

The monitor, or screen, works much like a television screen. Like a television, the computer monitor needs a video signal to display a picture. The video card creates this signal. The video card connects to the motherboard and converts the computer's digital output into text or pictures for display on the computer monitor.

Keyboard and Mouse

The keyboard and the mouse provide us with a way to communicate with the computer. By typing on the keyboard and pointing and clicking with a mouse, we are able to enter data and give instructions to the computer. The instructions are converted to digital data.

Sound Card

Adding a sound card to a computer makes it possible for the computer to play and record sound.

The card converts audio signals into a digital format that can be processed by the computer. The card also converts digital audio stored on a computer into a format that can be sent to other equipment such as loudspeakers and the mixing desk.

Under A in the A-Z for more about analog and digital sound.

Sound cards vary enormously in quality and features. In the radio studio the primary purpose of a computer is to play and manipulate sound. Therefore, there are more demands on the studio sound card than on a standard office or home computer.

The right choice of sound card used in a studio computer is vital. Because the studio sound card tends to be more sophisticated than others, it is also often more expensive.

Communications Hardware

Computer communications are becoming a part of everyday life. More and more people are using the Internet and e-mail as a resource and as a means of communication. By connecting our computer to a phone line or a network connection, we are able to communicate with other computer users.

Modems

A modem makes it possible for a computer to communicate through phone lines. Modem is an acronym for **Mo**dulator/**Dem**odulator. Modems translate digital computer information into analog signals used over phone lines. They can also work in the reverse and translate the analog signal from a phone line into a digital signal used by a computer.

Modems are distinguished by the number of bits per second of information that they can transmit. Nowadays, commonly used modems operate at 33 600 and 56 000 bits per second. However, older modems that work at lower speeds are still used.

Networking Hardware

If you have more than one computer in your station, you will probably want to share files and resources such as printers amongst them. To achieve this your computer needs networking hardware. This hardware is a **n**etwork **i**nterface **c**ard (NIC), often just called a network card.

Most new computers are supplied with a network card. The network card allows the computer to talk with other computers on the network (that is transmit and receive data).

If you have more than two computers that you want to connect together, you need a network hub to which you connect all of your computers. The hub switches information between the computers.

Under D (Digital Playout) in the A-Z for more about networking computers.

Peripherals

In addition to the hardware already listed, there is a wide variety of external hardware, often referred to as peripheral devices or "peripherals" that can be connected to a computer.

The more common devices are CD-Writers used to make audio and data CDs; printers for printing information onto paper, and scanners to convert text or pictures on paper into a digital format for the computer.

Many stations today use CD-Writers and store and distribute programmes they have produced. It is also useful to have a printer for printing out text reports, letters and other information.

Under C in the A-Z for more about CD-Writers.

Computer Software

There are two parts to any computer system: the hardware, and the software.

Software is the term used to describe the programmes that we use on the computer. Both hardware and software are very important, and we need the right hardware and software if we want to get anything useful done with a computer.

Under C (Computer Hardware) for more about computer hardware, and the interaction between hardware and software.

Types of Software

There are different software programmes or packages that perform different functions. For example to type and edit a script, or a letter, you need a word processing programme such as Microsoft Word or Word Perfect. For accountants there is a wide variety of software you can use to make accounting easier. Graphic designers use software packages for design and layout. Using computers for radio work also needs specialised software.

Operating System

Software programmes such as the many different versions of Windows (95 / 98 / NT / 2000 / XP), Mac OS, Linux and others are called operating systems. All computers use an operating system. The operating system controls all of the computer's hardware and provides a base upon which other software can be used.

The operating system manages the hardware for the user, and provides on-screen images that you can understand when you are working on the computer.

Audio Editing Software

In the radio production studio, software programmes such as Pro-Tools, Soundscapes, Cool Edit Pro, Netia and many others are used to record and change audio on the computer. These programmes let us mix and edit sounds, add sound effects, and generate digital audio files that can be sent to another computer for playout in the studio, or for writing onto a CD-R.

Under C in the A-Z for more about CD-Writers.

Advanced audio editing programmes often demand more advanced sound hardware, and the soundcards used in audio production computers are usually more sophisticated than those on a standard computer.

As audio editing is usually graphical – the software displays pictures of the sound waves you are working with – more sophisticated video cards and larger screens are usually used with this software.

Playout Software

Stations that use computers in the broadcast studio will use playout software.

Playout software provides a way for presenters to quickly access audio files stored on the computer and then to play them on the air. Programmes such as Netia, On the Air, Wavecart, Radiohost and many others have been especially designed for use in the radio studio. Again, using these programmes often requires more sophisticated sound hardware.

Under D in the A-Z for more about digital playout systems.

Software for Anything and Everything...

There are millions of other types of useful software available.

Outside of the studio a community radio station may have another computer with different software, for example, word processing software for typing; Internet browsing software such as Internet Explorer or Netscape Communicator for accessing the Internet and e-mail; software such as Outlook Express or Pegasus Mail for sending and receiving e-mail.

Stations might also use accounting software to manage finances or desktop publishing software to design flyers, notices and posters for the station.

In fact, there is very little you can't find software for nowadays. Many software packages are available for download directly from the Internet.

Screenshot of the software package, CoolEdit Pro, which is used for audio editing

Connectors and Connections

All the audio equipment in the in the community radio studio is connected in some way to other pieces of equipment in the studio through audio and/or power cables.

At the end of each of these cables is a connector that is used to join the pieces of equipment. Knowing the names of these connectors and how they are wired is essential. This section introduces the power and audio connectors most often used in community radio stations.
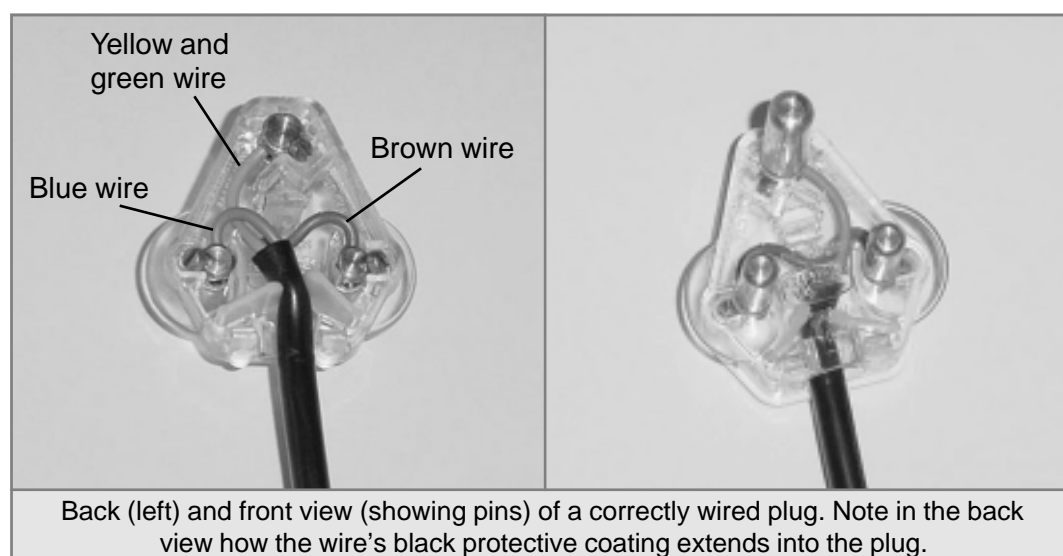
Power (Electricity) Connectors

The Plug

The standard plug is the most common power connector. It connects equipment to the mains power supply. The mains supply is deadly. Incorrectly wired plugs will damage your equipment and can cause electric shocks that can kill or injure you.

It is essential to wire all power connectors correctly. The photograph below shows how mains plugs are wired.

Back (left) and front view (showing pins) of a correctly wired plug. Note in the back view how the wire's black protective coating extends into the plug.

To wire a plug, disconnect the plug from the mains supply. Then follow the colour coding of the wires.

• The blue, neutral wire connects to the pin on the left as seen from the back of the plug. This pin is most often labelled "Neutral", "Blue" or just "N".

• The brown, live wire connects to the pin on the right as seen from the back of the plug. This pin is often labelled "Live", "Brown" or "L".

• The yellow and green, earth wire connects to the top pin of the plug. The earth pin is often labelled "Yel/Grn", "Earth" or marked with the earth symbol: ⏚ Not all pieces of equipment use the earth wire, and you will have to read the wiring instructions to see whether or not 'earthing' is needed.

The wiring must also be properly cut and secured inside the plug. The earth, live and neutral wires of a power cable are surrounded by a rubber or plastic coating (mostly white, grey or black). To wire the plug, you will have to remove a section of this coating. When you do this, make sure that this protective coating extends right into the plug.

The earth, live and neutral wires are often stuck together. You will have to separate them by pulling them apart. When you do this, make sure the insulation (the blue, yellow / green and brown coating) surrounding the copper wire remains intact. Also, make sure that the three separated sections are just long enough to fit comfortably into the pins.

Then remove a very short section of the yellow / green, brown and blue coating to expose the copper wire. You must remove just enough so that the copper can fit into the hole at the base of each pin – but not touch any other part of the plugs casing. Screw the screws that will hold the copper wire in the pins down firmly, making sure that they are holding the copper wire down.

Plugs that have loose pins, do not have back covers, or are broken, cracked or damaged in anyway must be replaced immediately. They can endanger your life, and will definitely cause breakdowns and harm your equipment.

IEC Connector

The IEC connector is often called a "kettle plug", as they are often used on kettles. They are used to supply power to many pieces of studio equipment, and to computers.

There are two kinds of IEC connector the male and the female connector, as shown below. You may find the formal names for these connectors in some manuals, as follows: the male connector is the IEC320/C-14 and the female connector is the IEC320/C-13.

| The male connector, or IEC320/C-14 | The female connector, or IEC320/C-13 |

The female connector usually conducts power from the power source, and the male plug normally receives power. Typically, this is the power connector found on most broadcast equipment.
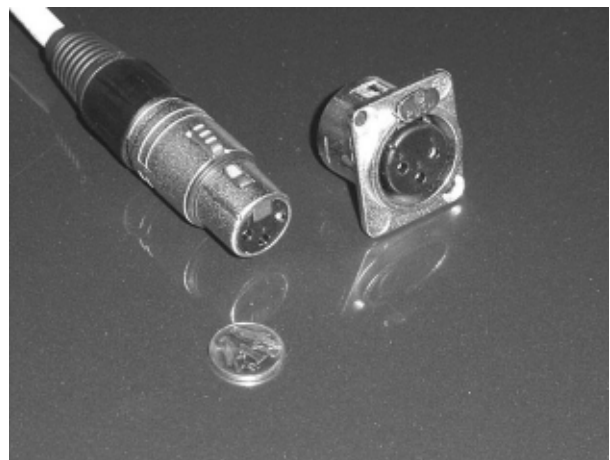
A standard three-core power cable is used with these connectors. The connectors have three pins. The middle pin is Earth, the left pin is Neutral and right is Live, as seen from the front.

Audio Connectors

XLR Connectors (Male and Female)
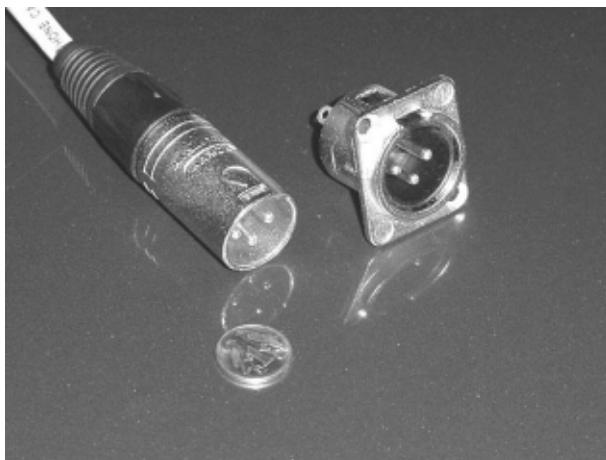
| Male XLR connectors | Female XLR connectors |

The XLR connector is the most common connector in professional audio. Convention has it that signals move in the same direction as the pins of the connector, that is, they go into a female XLR connector and come out of a male XLR connector.

The XLR is most commonly a 3-pin connector, although 4 or 5 pin types are used for some microphones or other specialised connections.

The XLR connector has a locking mechanism and will not simply pull out. There is a tab on the connector and only once you've pressed it will you be able to pull the connector out.

The pins on the male and female connector line up only one way, making it impossible to connect them incorrectly.

The three pins on the connector make a balanced connection possible. The predominant standard is that Pin 1 of the connector is used for the shield; Pin 2 for the in-phase or "hot" signal, and Pin 3 for the out of phase or "cold" signal. The XLR an also be used for unbalanced connections. In this case, the "cold" pin (Pin 3) is not used.

Under B in the A-Z for more on balanced and unbalanced connections.

¼" Jack and Plug

| ¼" TRS plug | ¼" Mono plug |

The words jack and plug are often used interchangeably, causing some confusion. The right way to use the terms is as follows: plugs are the connectors that plug into jacks. We will stick to this rule in this section.

The ¼" plug can have two or three contacts. The three contacts are the tip, the ring and the sleeve. The three contact connectors are often called TRS plugs. The three contacts are used for stereo (tip = left, ring = right, sleeve = ground) and balanced (tip = hot, ring = hot, sleeve = shield) connections. The two-contact jacks only have a tip and a sleeve. Two-contact jacks are used for unbalanced connections.

GPO Jack and Plug

| GPO plug |

The GPO, or just PO, plug takes its name from the connectors used in the **P**ost **O**ffice for connecting lines together in a telephone switchboard. For this reason, many American books will refer to these connectors as phone plugs.

The GPO connector is most commonly used for jackfields or patchbays. The connector has three contacts: the tip, the ring and the sleeve. These provide the connections for a balanced audio signal with a screen.

These connectors are used on short patch cords that are used to make the connections between the jacks on a patchbay.

Under P in the A-Z for more on patchbays.

RCA / Phono Plugs (Jacks)
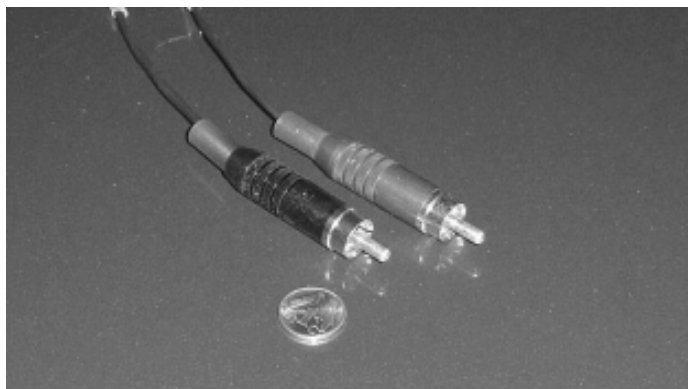
RCA or Phono connector

These devices have two names: RCA jack is the American term; phono plug is the British term. In South Africa, it seems most people use the term "RCA" to describe the connector. The RCA is an unbalanced connector designed for domestic use, but it is used on some studio equipment.

Mini-Jacks
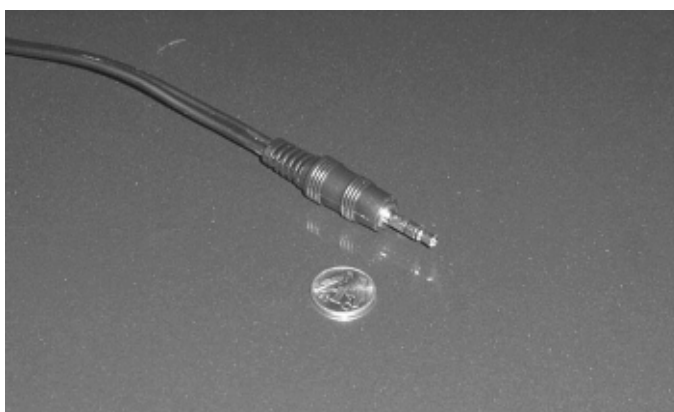
Stereo Mini-Jack

These are miniature versions of the ¼" stereo plug. They are used on portable equipment. They are also often the input and output connectors on many computer sound cards, and they are used as connectors on stereo headphones.

The three contacts on the connector are used for unbalanced stereo connections (tip = left, ring = right, sleeve = grand). Their small size and delicate construction makes them particularly flimsy and prone to loose connections and shorts.

D-Type Connectors

Female D-Type connectors

These are multi-pin connectors. The D-type is available as a male or female connector in sizes with between 15 and 50 pins. They are common on digital equipment and are also often used on studio mixing desks.

RF Connectors

BNC Connectors

Male (left) and female (right) BNC connectors

The BNC connector uses a similar connection to a bayonet light bulb, and locks into place by turning the connector. The 75-ohm BNC connector is used with video equipment. Most often the BNC connector in the radio station is a 50-ohm BNC used with transmission and other radio frequency (RF) equipment.

The TNC connector is also quite common. It is basically a screw version of the BNC connector

N-Type Connectors

N-Type connector

The N-Type connector is a larger version of the TNC connector. It is able to handle considerably higher RF powers. It is commonly found on transmission and RF test equipment.

Telephone and network connectors

RJ-11 Connector
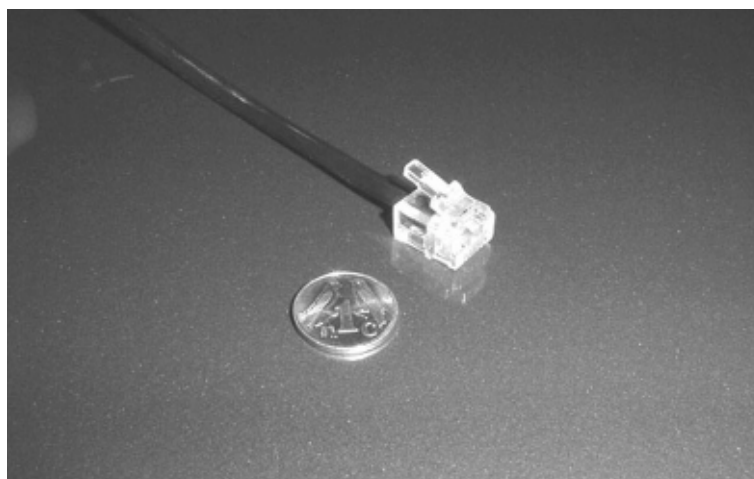
RJ-11 connector

The RJ-11 connector is most often used for telephone connections. It is a small four-pin plastic connector. It requires a special crimping tool to be attached to the cable.

RJ-45 Connector
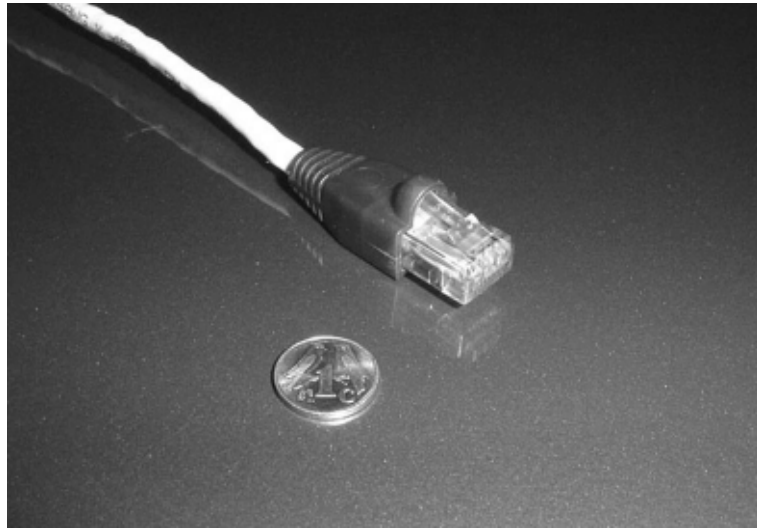
RJ-45 connector

The RJ-45 connector is a larger version of the RJ-11. It has eight pins and is used for data network cables such as ISDN cables and computer network cables. Standard computer network cards and network hubs have sockets for this connector.

The Decibel

Equipment specifications are full of symbols like dB, dBu, dBV and dBv. These symbols can confuse even the most experienced people. dB is the abbreviation for "decibel".

The decibel, or dB, always describes a ratio of two quantities. This can be the ratio of two powers, voltages, currents or sound intensities. Most often it is a ratio of power quantities.

The dB is a logarithmic quantity. A logarithm is a mathematical function that is used to describe large numbers with many digits with much smaller numbers. When working with audio, using logarithms makes it much easier to describe the quantities we are working with. The sensitivity of our ears is more or less logarithmic, so dB values relate more closely to how we hear. The idea behind using the dB is to make things easier.

The Bel is a unit named after Alexander Graham Bell, the inventor of the telephone. There are 10 decibels in a Bel. The decibel is more commonly used, as the Bel is inconveniently large. The decibel is described as 10 times the common logarithm of the power ratio. Thus to express the relationship of two power levels, P_1 and P_0, in decibels, we'd write:

$$dB = 10 \log \left(\frac{P_1}{P_0} \right)$$

It is not really important if you're not familiar with logarithms. What is important is to grasp that dB always represents a ratio of two quantities, and not a quantity itself.

To demonstrate this let's put some real numbers into the dB equation. Let's assume an amplifier produces 2 Watts of power at its output when 1 Watt of power is fed to its input. This means that the ratio of the two powers and the gain of the amplifier can be described as follows in dB:

$$dB = 10 \times \log \left(\frac{P_1}{P_0} \right)$$
$$dB = 10 \times \log \left(\frac{2}{1} \right)$$
$$dB = 10 \times \log (2)$$
$$dB = 10 \times 0.301$$
$$dB = 3.01$$
$$dB \approx 3$$

So the ratio of 2 Watts to 1 Watt is 3dB. Thus a gain of 2 is described as 3dB of gain.

Here is another example of a decibel calculation. The output power of a CD player is typically 2.5 milliwatt or 0.0025 Watts. To power a loudspeaker an amplifier would need to increase the power of this signal to 20 Watts. Lets put these values into the decibel equation.

$$dB = 10 \times \log \left(\frac{P_1}{P_0} \right)$$

$$dB = 10 \times \log \left(\frac{20}{0.0025} \right)$$

$$dB = 10 \times \log (8000)$$

$$dB = 10 \times 3.90$$

$$dB = 39\ dB$$

The ratio of 20W to 2.5mW can be described as 39dB. This is the gain of the amplifier. This example points out one of the useful qualities of the decibel. The power of the CD player is actually amplified to 8000 times its original value in this case. This is much easier to describe using a smaller number such as 39dB.

The table below shows a variety of power ratios described in dB relative to 1 Watt

Power Value P_1	Level in dB (Relative to 1 Watt P_0)	Power Value P_1	Level in dB (Relative to 1 Watt P_0)
1	0	100	20
1.25	1	200	23
1.6	2	400	26
2	3	800	29
2.5	4	1000	30
3.15	5	4000	36
4.0	6	8000	39
5.0	7	10000	40
6.3	8	20000	43
8	9	40000	46
10	10	100000	50

Decibels can also be used to express voltage ratios. Where this happens, the formula is slightly different. Without going into detail, power is proportional to the square of the voltage, so to express a voltage ratio in decibel the dB relationship is doubled relative to power, hence:

$$dB_{voltage} = 20 \log \left(\frac{V_1}{V_2} \right)$$

(V_1 and V_2 in the above equation are the voltage values.)

The important concept is that "dB" in itself has no absolute value. But if 0dB is specified as some reference value then any number of dB above or below that zero reference can be used to describe a given quantity. To say that a CD player's nominal output level is +4dB means very little. It's a lot like saying the CD player can do 4. The CD player can do 4 of what? To say that the CD player's output level is 4dB above 1 miliwatt however actually tells us that the CD player delivers 2.5 miliwatts into some load. There are a number of reference levels that are commonly used.

Most modern audio equipment is sensitive to voltage levels. The standard dB term used for expressing input and output voltage is dBu. 0dBu is corresponds to a voltage of 0.775 volts. So for example our CD player with a nominal +4dBu output can deliver 0.775 x 2.5 or 1.9375 volts. Another dB unit that is often used is the dBv, which is equivalent to the dBu. An older dB unit that is used is the dBm. 0dBm is 1 miliwatt. dBm has no direct relationship to voltage. However the dBm and dBu units are equivalent to one another if the dBm value is derived with a 600-ohm load.

Digital Audio Tape (DAT)

Digital Audio Tape (DAT) was developed in the mid-1980s by Sony and Phillips, before Minidiscs and CD-R. Digital music was made popular by compact discs, and DAT was intended as a new digital recording format. It was designed for domestic use, but never became popular with home users. However, it has become a well-established standard format for professional use.

DAT cassettes contain magnetic tape that is used to store digital information. Storing digital audio requires storing large amounts of data.

Under C, Compact Discs, in the A-Z for more about storing digital audio.

Storing Data on DAT

To store large amounts of data, the DAT player/recorder uses a rotary-head format, where the read/write head spins diagonally across the tape at 2000 revolutions per minute. This is the same way that VCRs (video cassette recorders) are able to store the large amounts of data that generate pictures and sound.

The proper name for the DAT is "R-DAT", with the "R" standing for rotary. This term is rarely used, because the "S-DAT" (S for stationary) from which it was to be distinguished, never went into production.

Studio reel-to-reel and tape decks are able to use stationary heads because they have wider tape and faster tape speeds. As the DAT was intended to replace analog compact cassettes, it was kept small, and the rotary-head compromise was made. This means more moving parts and potential for problems, but the results are very acceptable.

Most DAT player/recorders seem to work like a cross between a typical analog cassette deck and a compact disc player. In addition to audio, you can record subcode information such as the number of the track (so you can jump between songs in a certain order), or absolute time (counted from the beginning of the tape) onto a DAT.

The tape speed of DAT is much faster than a regular cassette deck (you can rewind 30 minutes of music in 10 to 25 seconds).

Inputs and Outputs

DAT decks have analog inputs and outputs, just like any other tape system. They also have digital inputs and outputs to transfer audio digitally to and from other digital devices, like CD Players, MD Players and computers.

The Tape

As the DAT deck is similar in technology to a video deck, the tape is similar in design to a video cassette. The tape is about half the size of an analog cassette and even smaller than a MD.

The tape is protected by a hinged door, like a video cassette. When you play the tape, the door opens and the tape is pulled out and wrapped around the spinning head.

A tape can be as long as 90 or 120 minutes. Because there is only one "side" to the tape, that is uninterrupted time. This is more than the maximum playing length of CDs and MDs.

A Digital Audio Tape (DAT)

DAT can record in several modes.

These are:

- 2 channel (stereo), 48KHz Sample rate, 16-bit linear encoding, 120 min max

- 2 channel (stereo), 44.1Khz Sample rate, 16-bit linear encoding, 120 min max

- 2 channel (stereo), 32KHz Sample Rate, 12-bit non-linear encoding, 240 min max

The first mode uses a higher sampling rate than CD, and should deliver slightly higher quality than CD.

The second mode is the same as specified for CD and MD.

The third mode reduces the sample rate making it possible to record for a longer time on the cassette, but at lower quality. The effective "sample rate" of an FM stereo system is about 32kHz, so the loss in quality in the third mode is marginal for broadcast proposes. The third mode makes it possible to pre-record programmes of over three hours in length.

Under A, Analog and Digital, in the A-Z for more on sampling.

Operating a DAT Machine

We recommend that you consult the user manual for your DAT deck, as machines tend to vary significantly in operation. Generally, playing a DAT involves placing the tape in the player and pressing play. The operation is very similar to both a MD and a CD player.

• Press OPEN/CLOSE to open the tape tray, gently load the tape with the printed side facing up, and press OPEN/CLOSE to close the tray.

• Press PLAY to begin playback.

• To play through sections quickly, press REWIND or FAST FORWARD during playback.

• If the tape has indexed segments (tracks) recorded onto it, you can use the SKIP buttons to move between segments. Press a SKIP button once to jump one segment, twice to jump two segments, and so on.

Recording on DAT

Recording on DAT is similar to using a standard cassette tape or MD.

• Press OPEN/CLOSE to open the tape tray, gently load the tape with the printed side facing up, and press OPEN/CLOSE to close the tray.

• In order for absolute time to be recorded with the tape, you must do one of the following:

– If you wish to overwrite what is on the tape, press REWIND to rewind the tape to the beginning.

– If you do not wish to overwrite what is already on the tape, press END SEARCH. The tape will fast forward to the end of the previously recorded material.

• To begin recording, press RECORD and then PLAY.

Indexing DAT

DAT can also be indexed using Start, Skip, and End Ids (identifications). These are written onto the tape as inaudible subcode, much like the subcode on CD and MD.

The IDs are used to identify "tracks" on the DAT, marking where different items start and finish. Using the IDs makes it easier to find items on a DAT.

Most decks have a "forward to the next Start ID" button (or the SKIP button we previously mentioned). The deck will also have a skip back button.

These buttons are used to locate an item on the DAT that was marked with a Start ID. Many decks can be set to automatically move to the next Start ID when a Skip ID is encountered while playing.

The End ID marks the logical end of the tape; all decks will stop playing when it is reached, and some will automatically rewind the tape.

To index your DAT tapes consult your equipment manual on how to add and remove the Start, Skip and End Ids.

Common DAT Problems

Most people who have worked with DAT have a DAT horror story about how the delicate mechanism of the player chewed up a vital recording. Or you'll hear the sad story of how the tape deteriorated after only a few days, ruining the sound.

The only advice we can give is – never trust a single DAT for mastering or long-term storage. Always make a backup copy of work in progress or work you want to keep.

Store on CD-R Rather Than DAT

The cost of the hardware and software used to make CD-Rs has dropped enormously, as has the cost of blank CD-Rs. So if you have the technology, or the money to buy it, it may be worth considering CD-R instead of DAT for storing recordings or for long-term archiving of material.

DAT Do's and Don't's

The following article, TOP 20 DAT RECORDING DO'S and DON'TS, comes from a TASCAM user's guide. It provides some very practical hints for using your DAT player/Recorder. No author was listed, so credit is given to Teac Corporation.

1. Do fast forward new DAT tapes (or analog tapes, for that matter) all the way to the end, then rewind to the beginning before using. This "unpacks" the tape and disperses any lubricant more evenly over the tape surface.

2. Listen to a tape after recording to verify that there are no glitches. However, note that some top-of-the-line DAT decks, such as TASCAM's DA-60, let you monitor off the tape as you're recording. These machines are more costly because they use 4 heads, but being able to verify that signals are being properly recorded on tape can be invaluable, especially for live recordings.

3. Don't overuse "cue" and "review" modes. These modes are similar to rewind and fast forward, but maintain contact between the tape and head so you can hear what's on the tape as it whizzes by. You usually enter these modes by pressing rewind or fast forward while the tape is playing instead of pressing stop first. Cue and review speeds are slower than rewind and fast forward but faster than normal playback; although the monitored sound won't be high fidelity, it will be good enough to let you know where you are on the tape. The down side of these modes is that the continued tape contact causes head and tape wear, so unless you absolutely need to monitor the sound as it goes by, press stop before selecting rewind or fast forward to disengage the tape from the heads.

4. Do clean the DAT heads periodically (the manual should mention how regularly this needs to be done), or if there seems to be an increase in errors. DATs can be cleaned with commonly available dry-cleaning cassettes; never use wetting agents of any kind, or do analog tape recorder cleaning techniques. If you experience any problem with a rotating head-based machine, first run a cleaning cassette and see if the problem fixes itself. This will provide a solution in most cases; if not, take the DAT deck to your local authorized service centre for professional cleaning and inspection.

5. Don't write on DAT labels with pencil or ball point pens. You don't want graphite or ink particles flaking off of the DAT machine transport; use a felt-tip marker that doesn't leave any solid residue. Do back up your tapes. It's as important to make DAT copies as it is to make safety copies of analog tapes. It's best to do digital transfers, but if it isn't possible, analog copies still sound fine. If at all possible, keep an additional set of tapes offsite in case of fire, flood, earthquake, volcanoes, or UFO visitations.

6. There is a major caution concerning digital copies: "consumer" grade DAT recorders often include SCMS (Serial Copy Management System), which is designed to thwart unauthorized digital dubbing and piracy by inserting "flags" in the digital data stream. You can generally make one copy of a tape, but you can't make a digital copy of that copy. (SCMS does not affect the analog inputs and outputs.) Why is this a problem? Suppose you mix down several tunes to a DAT, then "assemble" these tunes in the proper order by copying them to a second DAT. If you try to make a backup copy of the master, SCMS will prevent this. Pro decks either dispense with SCMS entirely, or provide some means to defeat it. Which brings us to...

7. Do defeat SCMS on the DA-30. There's no reason for professionals to have to put up with SCMS, since pros often need to make copies of their own music. The basic procedure is to unplug the DA-30, remove the top cover, and locate the digital printed circuit board (looking from the front of the unit, this is the large green board to the right of the transport). Locate a short jumper wire labelled W-402, then use a small diagonal cutters to clip this wire (or, desolder the wire and install a switch if you want to be able to choose between enabling and disabling SCMS). Put the cover back on and you're done. For more details on how to do this simple modification, see Issue #8 (Spring/Summer 1992) of the TASCAM Users Group guide.

8. Don't overlook the fact that DATs are compatible with many hard disc recording systems. If you don't have two DATs to make backups, you can copy the data from the DAT to the hard disc recorder, then load in a second DAT tape and bounce the hard disc data to it.

9. Do use the write protect tab to differentiate between a "current" and "old" backup. Many pros keep two backups. Only one is updated at a time; that way if a problem happens during the backup process, the older backup is still available. On the next backup, the older backup is updated. protect the current backup and unprotect the older backup so that the next time you back up, there won't be any mistaking which tape should be backed up to.

10. Do keep DAT tapes in proper environmental conditions: dust-free, low humidity, and cool. Since DATs aren't very big, you may be able to store back-up copies in a climate-controlled office environment, which gives the additional advantage of off-site redundancy. Humidity and heat are particularly bad; moisture can rust or corrode metal tapes.

11. Do record at 44.1 khz if you expect your material to end up on CD. Although DAT's 48 khz sampling frequency theoretically gives better high frequency response than 44.1 khz, the real world difference is negligible. Material recorded at 48 will have to undergo a sample rate conversion to 44.1 khz prior to appearing on CD, which costs between $5O and $100 and can degrade the sound to a minor degree.

12. Don't be too afraid to go over 0 VU. No, that's not a typo! Granted, 99.9 percent of the time you'll want to avoid overload like the plague. But on tunes where extremely short percussive transients are the reason for the overload, they can be slightly clipped. This allows for a higher average signal level without compression, assuming of course that the duration of any distorted signals are so short that you don't hear any adverse effects to the overall sound. (Tapes recorded on the DA-3O can actually sound good with a little clipping, since the DA-30's clipping characteristics are somewhat more like the "soft" clipping associated with analog gear.

13. Do date your DAT tapes. After a tape has been around more than five years or so, it's prudent to make a digital copy of it to a new tape, and keep the older tape as a backup.

14. Don't leave tapes in a DAT machine after the power is turned off. Should there be any type of problem (e.g., blown fuse) when you turn the machine on again, it may be difficult to get the tape out.

15. Do keep abreast of software changes in your machine. DATs, like most devices these days, are software-driven and occasionally their software is updated or improved. Check with the company periodically to see if any updates have been made available that are pertinent to the gear you own.

16. Don't go over and over the same piece of tape when doing final mixes. The more you use a piece of tape, the more it deteriorates and the greater the likelihood of errors. Do consecutive mixes, and note which one is the final mix. You can then do a digital transfer to another DAT deck to build up a tape with nothing but final mixes (but remember the earlier caution about SCMS-equipped gear.)

17. On a related topic, don't leave the DAT in pause mode; pause is intended for a very specific function. When stopped, DAT tape is disengaged from the transport, and is not in contact with the heads. When you enter play or record modes, it takes a few seconds to engage the tape and have it contact the head. However, in pause mode the tape remains engaged and the head remains in contact; if you then initiate play or record, the tape kicks into action much faster. Since pause mode causes head and tape wear, use this only when you need to go from a paused to playing or recording status in the minimum possible time.

18. Do leave at least a minute of blank, but recorded, tape at the beginning of a tape before laying down any of your tunes (two minutes is probably safer). Also make sure you don't record all the way to the end. With virtually all tapes, the ends are the most prone to having quality problems.

19. Don't let DAT tapes get near magnetic fields. As with analog tapes, this can erase signals on the tape.

20. Do give each tape a unique ID. DAT tapes have a way of proliferating, and it's important to be able to locate what you want quickly. Similarly, do keep a notebook of what's on each tape. DATs can hold two or even three hours of material, and that's a lot of tunes – you don't want to record over something accidentally!

Digital Playout System

The term digital playout system usually refers to using a computer or computers to play sound for broadcast. Most computers are capable of playing back audio, through use of a piece of hardware called a sound card, which is installed in the computer.

A sound card contains electronics that make it possible for the computer to process digital audio data. The sound card also converts digital audio to analog audio that can be fed to a mixing desk.

Under A in the A-Z for more about analog and digital audio.

More advanced sound cards can offer many other audio processing capabilities.

Playout Software

In addition to the sound card hardware, you need playout software to make a good playout system. It is the combination of software and hardware that make it possible to play sound from a computer.

Basic audio playout software is usually part of most computer systems. However the special requirements of radio mean that most stations would want to use specialist playout software designed to make the job of the presenter easier. The software should make it easy to find the sounds that you want to play. It should also make it possible to mix together sounds and play them simply and quickly with the mouse or keyboard.

Using the System

A digital playout system is similar to a CD or MD player, in that you still have to find your items and then play them out. But instead of having to find the right CD and choose a track to play with the CD player, now the task is to find a sound file on the computer and play it by using a mouse or keyboard. A sound file or audio file is a file saved on the computer that contains digital audio data that can be played by the playout software.

A large part of using a playout system is creating and managing the sound files the system uses.

For example, imagine you have a CD with several tracks that you want to be able to play on the computer. Firstly, the audio stored on the CDs needs to be converted into sound files that the playout system can use. This means you must record the CD onto the computer.

There are a few ways of doing this: sometimes recording software is used and the sound from a CD player is fed to the computer and the CD is actually recorded. The recording is then saved as a sound file on the computer.
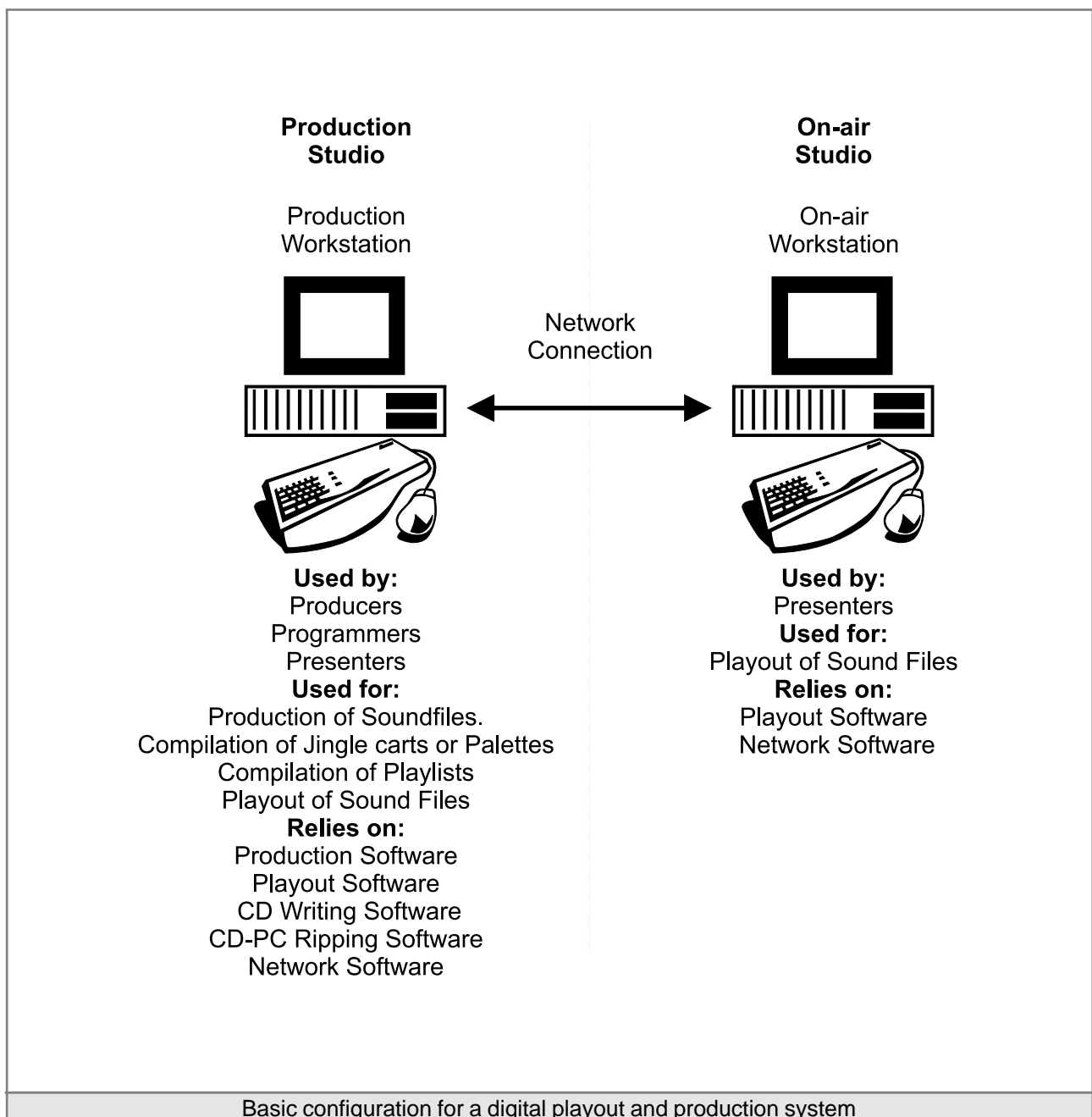
More often, the digital audio that is already on the CD is extracted from the CD using the computer's CD-ROM drive and software designed for this purpose. This process is called digital audio extraction (DAE) or "ripping". Ripping software makes it possible to copy the digital audio from the CD to the computer at very high speed.

No matter what method you use to get the sound from the CD onto the computer, this process has to happen before tracks from the CD can be played by the playout system.

All other items, such as inserts, jingles, pre-recorded interviews and so on, that you want to play from the computer playout system also have to be transferred to the computer and saved as sound files.

A Typical Basic Playout System

The system architecture illustrated below is typical to a basic playout system. It consists of two workstations (computers) in separate studios, one capable of production work and both capable of playout. The two computers are linked by a network connection that makes it possible to send files from one computer to the other.

Production Studio

Production Workstation

On-air Studio

On-air Workstation

Network Connection

Used by:
Producers
Programmers
Presenters
Used for:
Production of Soundfiles.
Compilation of Jingle carts or Palettes
Compilation of Playlists
Playout of Sound Files
Relies on:
Production Software
Playout Software
CD Writing Software
CD-PC Ripping Software
Network Software

Used by:
Presenters
Used for:
Playout of Sound Files
Relies on:
Playout Software
Network Software

Basic configuration for a digital playout and production system

Operating the system requires producing sound files on the production workstation that are compatible with the playout software. These files are then copied using the network connection to the on-air workstation. Once they have been stored on the on-air workstation, the sound files can be played out using the playout software.

You need two workstations because the on-air workstation cannot be used for production work while broadcasting.

Once the sound file has been created, it is vitally important that it can be found on the computer for playout. This means ensuring all of the sound files on the computer's hard drive are organised so that they can be easily retrieved.

Organising Your Files

Depending on the playout system you use, and your station's preference, this can be done in many ways. Organising the sound files in the playout system is the most demanding part of using the system.

It is essential that everyone who uses the system knows how the sound files are organised. If sound files are not descriptively named and properly organised the playout system becomes completely chaotic and it becomes impossible to find sound files that you may have spent hours producing and that you need to broadcast.

Imagine the thousand sound files on a playout system are a thousand sheets of paper. We all know that if we don't organise a thousand sheets of paper by filing them and clearly labelling the files, we can waste hours finding the one we want.

A radio show host does not have hours to find the right file. Mostly s/he will want the file immediately.

Many people use the playout system in a community radio station, so it is vital organise your sound files. Set up the system when you install your digital production or playout system, and discuss it with your equipment supplier.

Different Ways of Organising Files

Often, sound files are grouped by type. This means that in the case of a Windows or an Apple computer, sound files of the same type are placed in the same folders so that they can be easily retrieved.

Usually, the computer is set up with a folder where music sound files are saved, a folder for adverts, jingles, finished productions such as interviews, sound effects and so on.

Some systems already have folders for storing different items but will still rely on the user to label the item or save it in the right place.

Different playout systems will require a different approach – so consult your manuals.

Using a Playout System

A good playout system is very easy to use. Volunteer presenters or other staff who already know the studio essentials – that is, how to use the mixer, the mics and other source equipment — will really only have to learn how to use a mouse and a keyboard to get sound from the computer onto air.

Advantages and Disadvantages of a Playout System

With a playout system all music is stored on computer, so music cannot be stolen. Computers sound files do not deteriorate and cannot get dirty or scratched like tape and CDS.

At the same time – computers sometimes "crash" (break down as a result of software or hardware problems). While a library full of CDs may be unreliable because people may take CDs home and not return them, or lose them, it will never crash like a computer can.

Playout systems make the presenter's job much easier. At the same time, the station will need extra computer skills to maintain the playout system, and this can make the technical department's work more difficult. These skills can be difficult to learn and so it is important to have a good relationship with your supplier to talk through problems as they arise.

Automation Systems

A digital automation system is a more sophisticated playout system. An automation system "automates" the entire broadcasting process. These systems are used extensively by private and public broadcasters.

With an automation system, virtually all of the station's programming is played according to a pre-determined schedule from the computer. The presenter is only there to link the items together, as they are already pre-set on the computer. If there is no presenter, the automation system can take over and play out a pre-set schedule of programming without any human assistance.

The trick of the automation system is generating the schedule of programming that is to be played out and making this programming available on the studio computer at the right time. This demands a lot of work from the station's programming, music, technical and production departments.

It also requires maintenance of a much more complicated computer network - needed to link production, on-air and administrative facilities, the time-consuming loading of sound data onto the computer and the difficult and ongoing task of managing the data once it is on the computer.

Generating the programming, music and advertising schedules well in advance of the actual broadcast demands a full-time programming team. This team needs to enforce an absolute vision of the station's format. The advertising department staff will also need to be able to generate scheduling and billing information using a computer-based financial system.

The demands that a complete automation system place on the staff of the station are severe. It is difficult to imagine a station maintaining an automation system without continuous access to a qualified computer specialist to keep the system running, in addition to the staff who will generate the daily schedules. Complete automation also requires a much higher level of computer literacy from many of the staff members at the station.
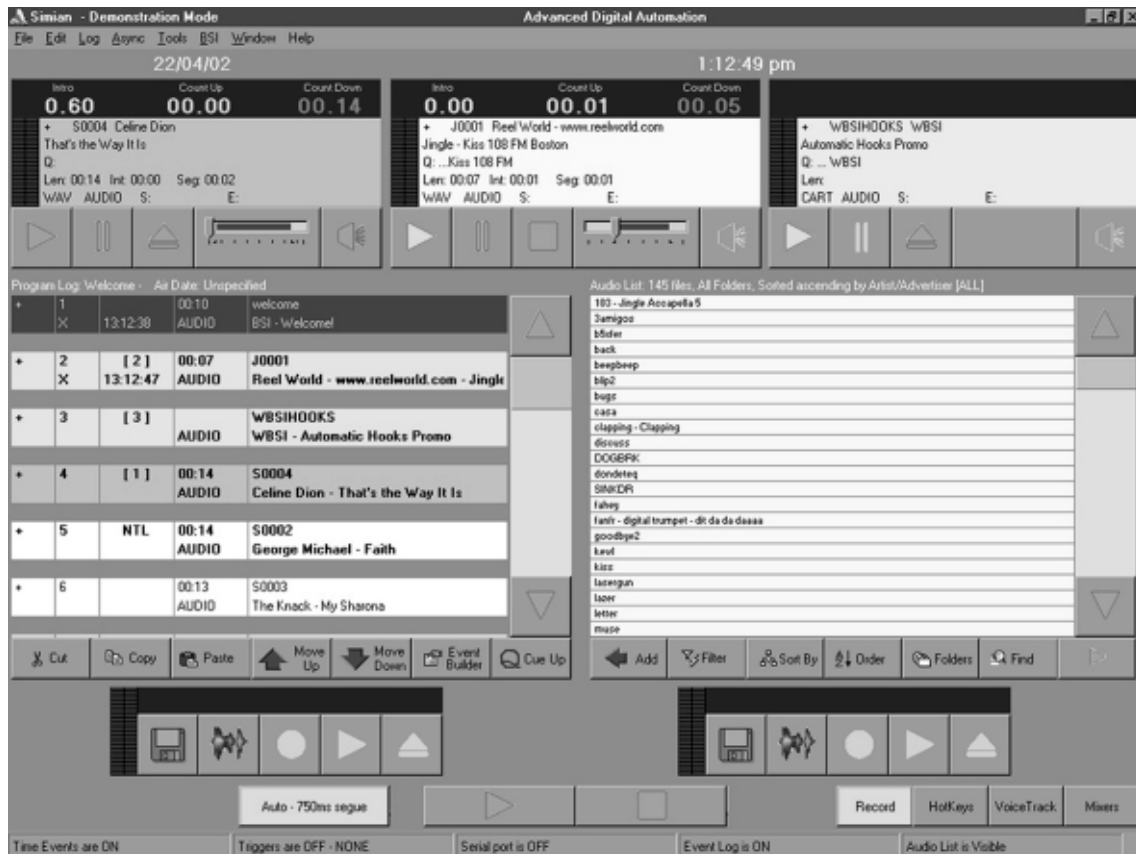
Screenshot of BSI Simian. This is a playout and automation system. The programme has three playback decks across the top of the screen.

Effects

Effects are designed to enhance good sound recordings, not to hide bad recording technique. Properly used effects can add a whole new dimension to recorded sound. Effects can make adverts more appealing; dramas more dramatic, and jingles literally out of this world.

Most community radio stations encounter effects as part of the station's digital editing system. Nearly all computer-based digital-editing systems now come equipped with a toolbox of effects.

Some stations also use dedicated effects processors. These are extra "boxes" that are connected to the rest of the studio equipment. The connection of the effects processor to the studio system is often specific to the studio configuration.

Unfortunately, explaining the many combinations of effects using different configurations of auxiliary (aux) sends, aux returns, insert points and patchbays is beyond the scope of this manual. Let's just say that if you are going to be using a dedicated effects processor in your studio, you should consult thoroughly with the equipment supplier to ensure it can be used in a way that suits your station's needs.

However there are many different types of effects, we discuss some that you are likely to encounter either on your computer or your effects unit.

Equalisation (EQ) is discussed in a separate section, even though it is considered to be an "effect". This is because EQ is part of so many pieces of studio equipment.

The same applies to compression. A compressor limiter is a standard device in many stations and it is discussed separately.

Under C in the A-Z for more about compressor limiters, and under E in the A-Z for more about equalisation.

Nevertheless there remain a number of effects worth mentioning:

Reverberation (Reverb)

When recording in a controlled studio, voices and instruments often sound small or "dead". This is because the studio space does not have the acoustic ambience of a large hall or club. Reverb is an electronic effect that can be applied to audio to re-create this ambience.

Reverberation occurs naturally when sound is reflected and re-reflected off the walls, ceilings and floors within a large room. When a sound is produced in a room, that sound is echoed many times off the various surfaces in the room. These echoes happen so fast that we don't hear them as distinct from one another.

Electronic reverb re-creates this effect by generating thousands of "reflections" electronically.

Using reverb can create the impression of sound in a real room. It can also be used to create new effects that have no obvious real counterpart.

Delay / Echo

An echo is a repetition or a partial repetition of a sound resulting from the sound being reflected off some surface. We hear the original sound and a short while later the reflected sound reaches our ears. This delayed reflected sound is called an echo.

Delay is a way of electronically producing echoes by delaying an audio signal. There are often several settings to a delay effect, including the length, number and the volume of the echoes.

Many effects units and software programmes use a multi-tapped delay. These generate multiple echoes at different delay times. It also possible to pan the individual delays from left to right to create interesting spatial effects.

Modulation Effects

Modulating the delay time creates modulation effects. Modulation in this context means using a low frequency oscillator to generate a signal that varies the delay time in accordance with the frequency of the oscillator signal.

For example, applying a short delay to a sound and modulating the delay time creates the chorus effect. This produces a slight wavering in the pitch of the delayed signal. An equal mix of the original sound and the modulated delayed sound creates an effect that is like two voices singing the same part, but with slight differences in timing and tuning. Hence the name chorus

Chorus is one of several modulation effects. Using different length delays and different modulating frequencies and other slight variation creates the flanging, phasing and vibrato effects. Phasing uses a very short delay time and adds the slightly delayed signal to the original, creating a filtered sound.

Flanging uses a longer delay and also adds feedback to create a dramatic, swirling effect.

Vibrato is created by using only the delayed sound. The delay is kept very short and is modulated to create a change in pitch.

Pitch Shifting

Pitch shifting is another popular effect. A pitch shifter can change the pitch of a sound without changing the speed.

Time Stretching

Time stretching is similar to pitch shifting, but in this case the speed of the sound is changing and the pitch is retained.

And Many More…

There are many other effects out there. and it is worth experimenting with your software or hardware to find effects that will work with the material at your station.

The Electromagnetic Spectrum

Throughout this manual, we discuss signals: audio signals, digital signals and radio signals. These are not the only signals that are out there. It makes sense to understand where these signals fit into the bigger picture.

All signals are a form of electromagnetic radiation. Electromagnetic radiation is the transmission of energy in the form of waves that have electrical and magnetic components.

It is not possible for a wave to have an electrical component and not a magnetic component, and vice versa. The most familiar forms of electromagnetic radiation are radio waves and visible light waves. Although it doesn't seem that way, radio waves and light are essentially the same thing. But they differ in terms of the frequency and the wavelength of the radiation. Mathematically, waves are represented by sine waves. The sine wave is a graphical representation of wave phenomena. The diagram below illustrates the important quantities of a sine wave.

Representation of an Electromagnetic Wave: Two Complete Cycles of a Sine Wave
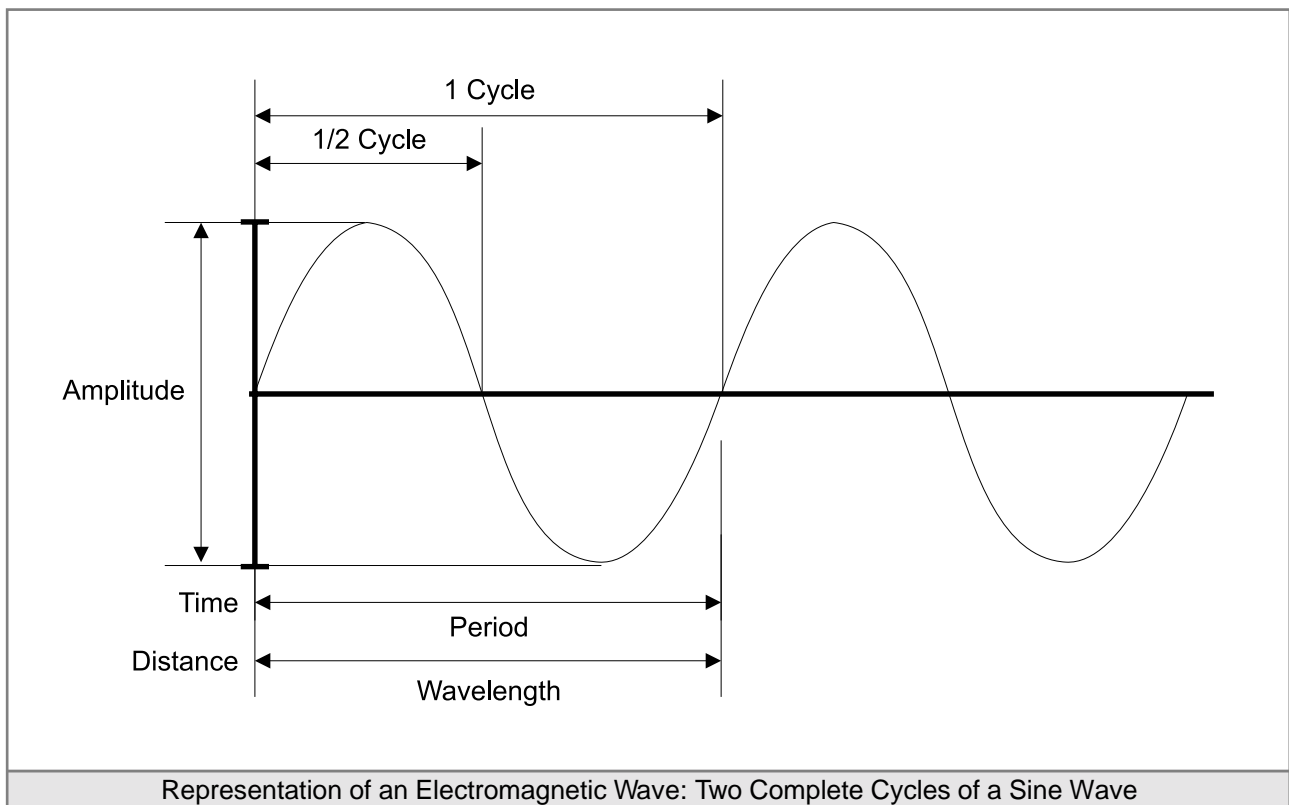
A sine wave follows its pattern repeatedly. Each time it returns to its starting point, this is called a cycle. The time it takes to complete one cycle is called the period. The number of cycles a wave completes in a second is called the frequency of the wave and is measured in Hertz (Hz). In the diagram the amplitude is the height of the sine wave. This is a measure of the signal level.

All electromagnetic waves travel through empty space at the same velocity (speed) namely the speed of light. The speed of light is given the symbol c and is 299 792 458 metres per second. For most calculations the approximate value of 300 000km/s is used. The speed, wavelength and frequency of any electromagnetic wave are related to one another by the following equation:

$$Wavelength = \frac{Speed\ of\ Light}{Frequency}$$

Different forms of electromagnetic radiation are characterised by wavelength and frequency. When electromagnetic waves are ordered by frequency or wavelength, this ordered array is called the electromagnetic spectrum. The electromagnetic spectrum represents all electromagnetic signals, from very low frequency and long wavelength to very high frequency and very short wavelength. The diagram below illustrates part of the spectrum.

A portion of the electromagnetic spectrum showing the frequency bands and wavelengths allocated to various purposes
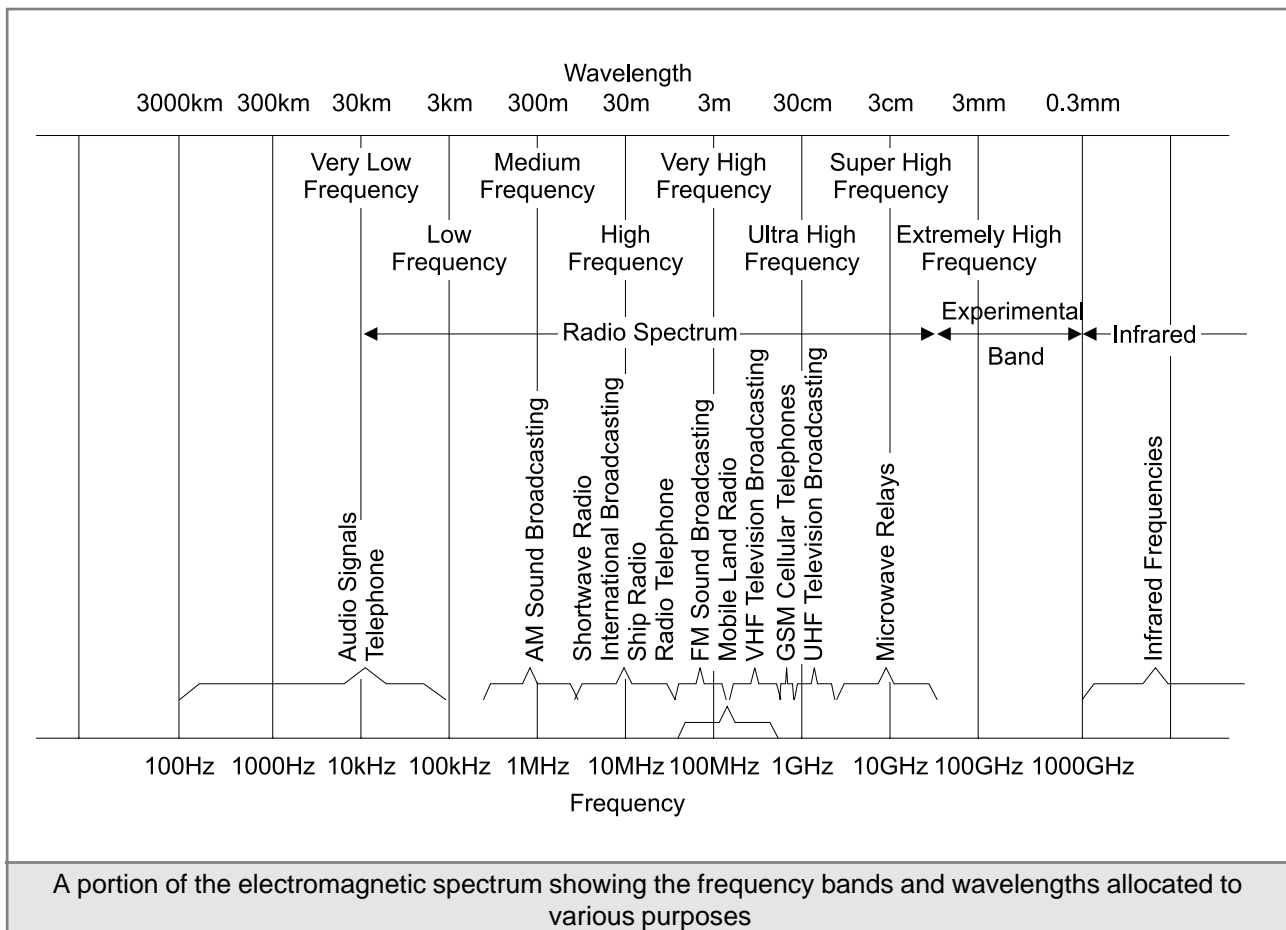
The allocation of the spectrum to different purposes is governed by international treaties and regulated by the International Telecommunications Union. In South Africa, this responsibility is delegated to the Independent Communications Authority of South Africa, who manage the use of the frequency spectrum in line with international laws.

Equalisers (EQ)

Equalisers (EQ) are generally grouped under the heading of "effects," as equalising is a form of signal processing that alters the sound you are producing.

Under E in the A-Z for more about effects.

However EQ and filtering is a common feature found on production mixers, microphones and even field recorders, so that a more complete explanation is necessary.

You may know equalisers as the tone controls found on a hifi set. The tone controls that let you change the bass or the treble are really the simplest form of equaliser.

They are called shelving equalisers and they cut or boost frequencies above or below a set cut-off frequency. The EQ works a lot like a volume control, except it only turns up (or turns down) a part of the sound. Equalisers are based on electronic filters. These filters make it possible to change a certain range of frequencies.

Filters that only affect signals above a certain frequency are referred to as low-pass filters. The name comes from the fact that while these filters change signals above the cut off frequency, signals lower than the cut off frequency pass through unchanged.

A treble EQ on a mixing desk is based on a low-pass shelving filter. Altering the treble EQ has no effect on the low frequencies they pass through the filter without being affected, while the higher frequencies are cut or boosted by the control. Filters that affect signals below a certain frequency are referred to as high-pass filters. A bass EQ is a high-pass shelving filter. It boosts or cuts the lower frequencies, but lets the higher frequencies pass through. Typical frequency responses of these equalisers are shown below.

Response of a high-pass and a low-pass shelving filter
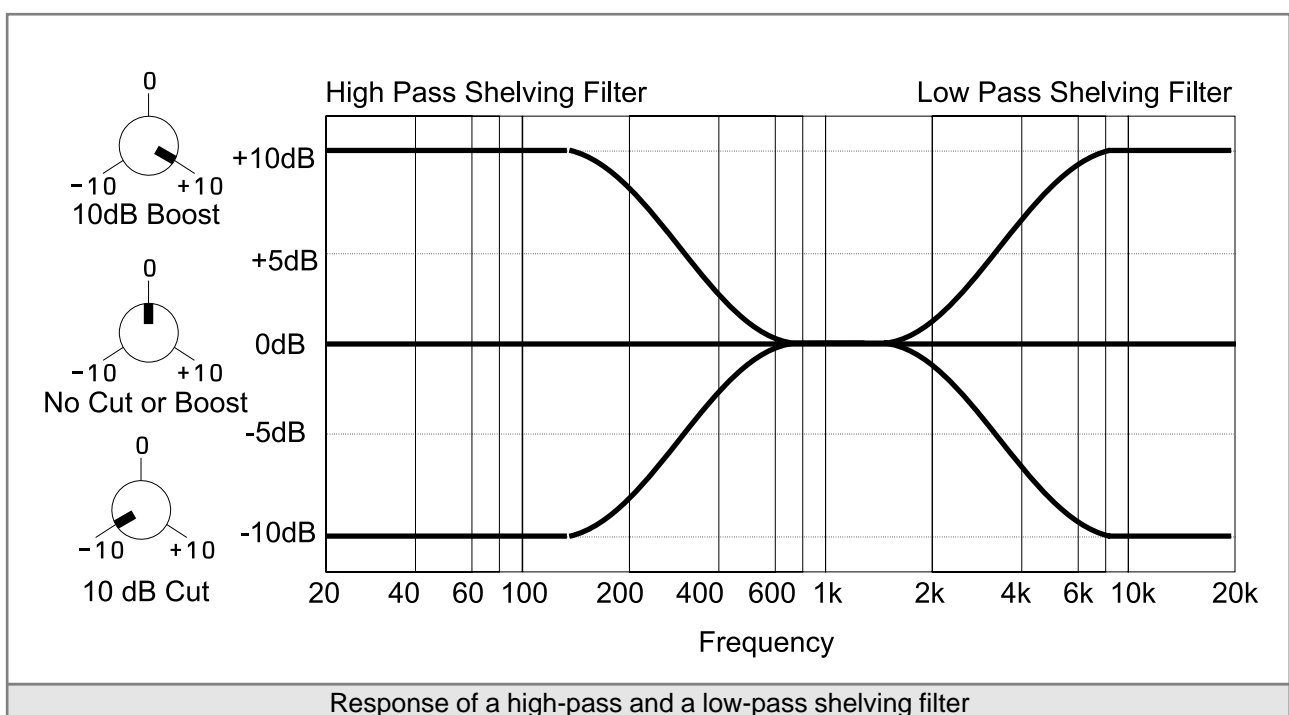
A filter that affects frequencies between two limits, such as the midrange EQ on a mixer, is called a bandpass filter. On many mixers the bandpass filter not only provides cut and boost, but it will also be tuneable so that its centre frequency can be changed. This form of EQ is known as a sweep equaliser. The frequency response of a bandpass filter is shown below.

Response of a bandpass filter
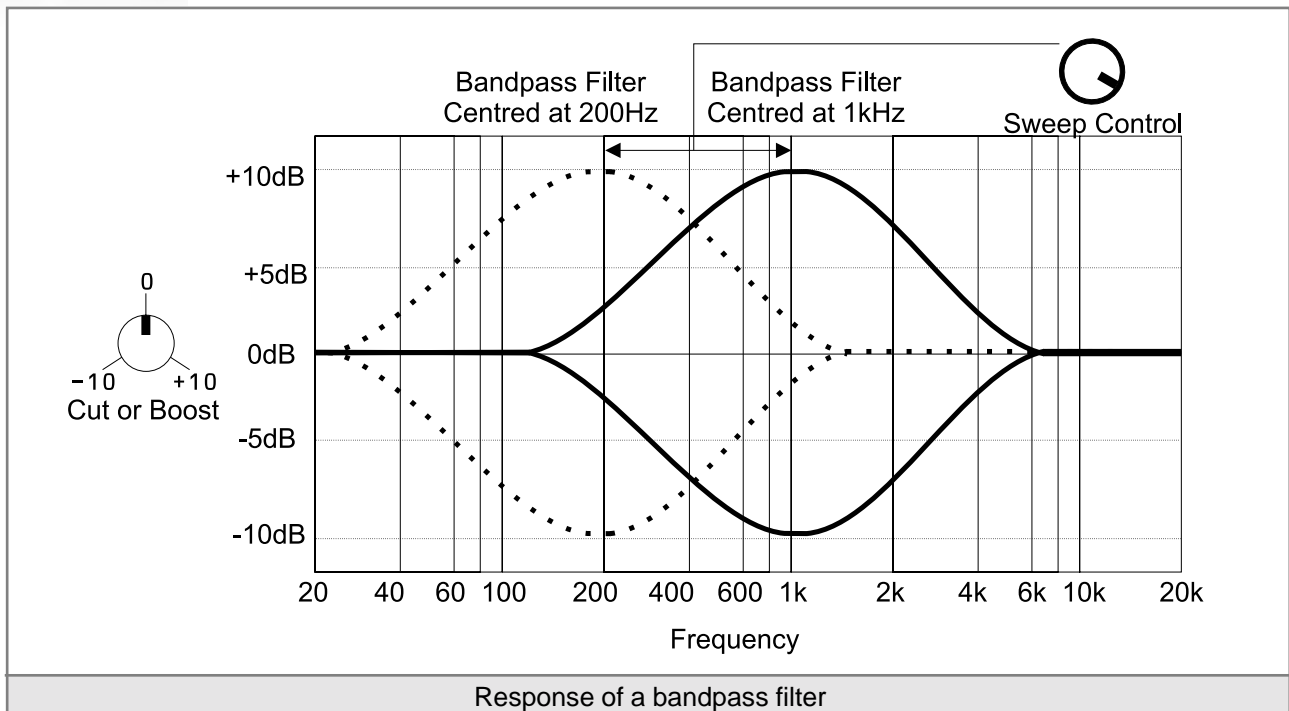
Similar to the sweep EQ is the parametric EQ. The parametric EQ has a third control that allows the width of the filter to be adjusted. The width of a filter is often described as its Q value. Q is defined as the ratio of the centre frequency of the filter to the bandwidth that the filter affects. A low Q value means that the filter only affects a small band of frequencies, a larger Q value and the filter affects a wider range of frequencies.

Another type of EQ you may encounter is the graphic EQ. A graphic equaliser whether as a stand-alone unit or as an effect on your computer is recognised by a row of faders across the front panel. Each of these faders is used to control a very narrow band of frequencies. By altering the faders the frequencies in each of these bands can be cut or boosted.

Using EQ

In using equalisers, a good rule is to first get the best sound possible without using them. Then if you have to use the EQ, preferably cut before boosting. For example if you want to accentuate the mid-range in a recording, rather cut the treble and bass a little before boosting the midrange. The less EQ boost you use in most cases the more natural your recording will sound.
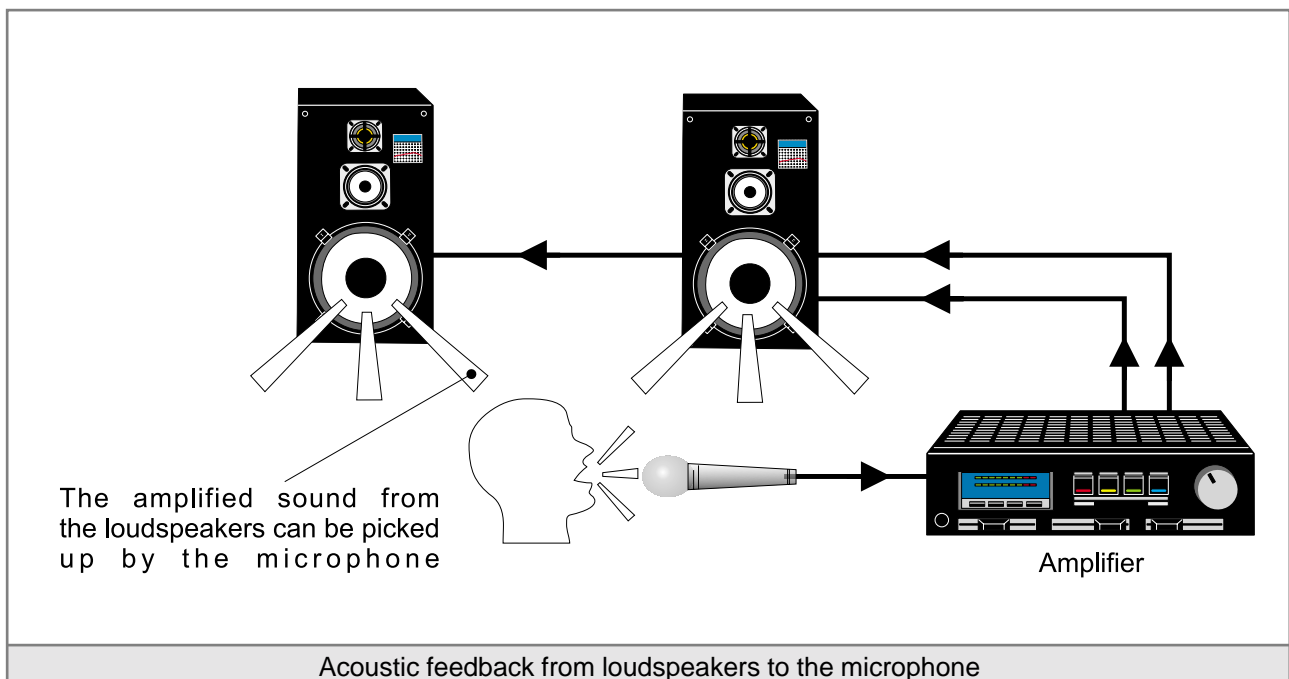
Feedback

Feedback is most often experienced as the nasty screeching you sometimes hear coming out of speakers at a concert, school play or a church service when someone is not using the microphone properly. It is a horrible sound – but even worse, if you don't quickly stop it, it can damage your equipment.

What Causes Feedback?

Feedback is produced when an unwanted sound output (or audio signal) from a loudspeaker or audio device is fed back to an earlier part of the sound reproduction system. The unwanted audio signal is picked up and amplified (made louder) by the electronic circuits in the system. When happens, the system becomes overloaded, and this is heard as a nasty screeching sound through the loudspeakers.

The diagram below shows how feedback happens.

The amplified sound from the loudspeakers can be picked up by the microphone

Amplifier

Acoustic feedback from loudspeakers to the microphone

A person is speaking into a microphone (mic). The electrical signal from the mic is fed to an amplifier. The amplifier increases the size of the signal to a level that can be fed to a pair of loudspeakers. The loudspeakers convert the signal into a much louder version of the original voice, as they are meant to do. The problem is that the mic is placed in front of the two loudspeakers. This means that the mic will pick up the sound of the person's voice as s/he speaks, in addition to the amplified sound from the loudspeakers. This creates a feedback loop of sound being fed from the loudspeakers to the mic, back to the speakers and so on. This feedback loop overloads the system, which causes the screeching noise.

How To Stop Feedback

To stop feedback, you have to break the feedback loop.

In the example described above, you will have to move the mic to a place where sound coming from the mic is not picked up by the speakers. Or you can move the loudspeakers so that they don't pick up sound from the mic. Moving either the speakers or the mic will break the feedback loop, and the system will work properly.

The Problem With Feedback

You don't want acoustic feedback, for two important reasons:

Firstly, it always affects your sound output. The noise is horrible and it will make your sound unlistenable. Even a small amount of feedback can cause echoes or boosting or cutting of certain frequencies.

Secondly the very high signal levels associated with feedback can destroy or damage loudspeakers and other equipment that are part of the feedback loop.

Controlling Feedback in the Broadcast Studio

Acoustic feedback is not acceptable in the broadcast studio. To make sure feedback from the mic to the loudspeakers doesn't happen, broadcast studios have special switching units. These units, often incorporated into the mixer, turn off the studio loudspeakers whenever a microphone is turned on.

The same switching unit also normally triggers the RAG lights or On-air lights. These are indicator lights placed inside and outside the studio to show that a microphone is switched on, and that someone is ON AIR. It is good practice never to enter or leave the studio when the on-air light is on.

Feedback Related To Recording

Another kind of in-studio feedback relates to recording equipment. The diagram on the next page shows how this happens.

Input 1   Input 2   Input 3   Output

Recording Input   Play Output

Minidisc Recorder

The minidisc recorder is being used to record the audio from two microphones connected to a mixer. The recorder's output is also connected to one of the mixer's channels. If that channel's fader is lifted, a feedback loop is created, as the audio sent to the recorder's input is fed back from the recorder's output into the mixer.

Mixing Bus

Mixer

Recording feedback

In the diagram above, a minidisk (MD) recorder is being used to record a signal from the recording output of a mixing desk. If the MD recorder's output is added to the mix on the mixer's recording bus, and you lift the MD recorder's fader, feedback will occur.

Under M in the A-Z for more on the mixer, and the mixer's recording bus.

We know that a feedback loop is created by sound being fed from one part of the sound system to an earlier part of the system. In this case the MD recorder is recording a signal fed to it from the mixing desk. By pushing up the MD fader, on the mixing desk, the output signal that the MD produces is being fed to the mixing desk. The mixing desk then feeds it back to the recording input of the MD, and a feedback loop is set up.

Sometimes this form of feedback can be used to good effect. If you are using reel-to-reel recorders, you can create tape echo by producing a small amount of feedback. This alters the sound by adding echo.

But generally, when recording to a MD, tape, reel, computer, DAT or any other recording device, avoid unwanted feedback by ensuring that the output of the device is not sent back into its recording input through a mixer or other equipment.

Field Recorders

The term field recorder refers to any portable recording device that allows reporters to go into the field with the means to capture sound. The most popular type of field recorder used by community radio broadcasters in South Africa is a portable cassette tape-based recorder.

Under C in the A-Z for more cassettes and cassette tape.

However, MD field recorders are becoming more commonly used, and will probably become standard equipment within the next few years, provided they aren't overtaken by another format.

Some stations also use portable DAT recorders as field recorders.

Under M in the A-Z for more about minidiscs.

Under D in the A-Z for more about DAT.

Because there are so many different kinds of field recorders you can buy, it is impossible to provide operating instructions that can be generally used. But there are a number of issues that apply to all recorders:

Field Recording Kit

The typical field recording kit consists of the following pieces:

The recorder

This could be a cassette, MD or DAT recorder. Field recorders should be built for inexperienced users and rough treatment. The recorder should have a carry strap, large buttons, easy to read displays and very rugged connectors and case. The recorder is not used in the protected environment of the studio and it is not securely mounted in an equipment cabinet. It is carried by someone, often over long distances and using public transport. One day, someone will drop it. Or someone may have to run carrying the recorder and it will get bumped around. Make sure your recorder is chosen for outdoor use and is ready for any eventuality.

A microphone

Several recorders have built-in microphones. However, the quality of these microphones is very poor, so external microphones are generally used. Like the recorder, these microphones need to be especially reliable and robust, as they will encounter rough treatment in the field. Omnidirectional microphones are most often used for fieldwork, as they are the most versatile. However directional "shot-gun" microphones can be useful in some cases, for example, a noisy press conference when you only want to record the speaker's voice.

Under M in the A-Z for more about microphones.

Microphone Cables

To get an audio signal from the microphone to the recorder, you need a microphone cable. Typically these cables are short - maybe only a metre long.

It is a good idea also to buy a second, much longer cable. This allows you to set up the microphone far away from the recorder. At a press conference you will often not be able to sit right next to the speaker. Placing your microphone near the speaker and having enough cable to retire to a quiet spot where you can monitor your recording is useful.

Headphones

Before starting any recording, you must be sure that the recording levels are properly set and that your machine is recording properly. All field recorders will have a headphone socket to allow you to listen to the recording as it is being made. It is essential to have headphones to listen to the sound as it is being recorded.

Media

All of the field recorders record onto some medium: tape, MD or DAT. Your field recording kit is not complete without one of these. A recorder without a tape, MD or DAT is really only good as a paperweight.

Batteries

All field recorders need a power supply when in the field. Batteries provide this power supply. Different recorders use different batteries. Most new recorders are equipped with rechargeable batteries, much like those in a cell-phone. These recorders have a dedicated charger that you will use when you need to re-charge the batteries. All recorders will also run off mains power through an adapter supplied with the recorder. Always check the state of the batteries before taking a recorder into the field: a recorder without power, like a recorder without a mic cable or recording media, is useless to you in the field.

Carry case

It is recommended that you keep all of the items that make up the field recording kit in a durable carry case. This prevents items being left lying around and forgotten, protects the equipment and provides a place to put spare recording media. Something like a medium size photographer's bag is ideal.

Microphone stand

A microphone stand is a useful optional item for a field recording kit. It frees up the reporters hands from holding the microphone during an interview. It can also be used in situations where it is impossible to hold the microphone, for example, at a speech or at a press conference.

Using Your Field Recorder

Before going into the field check that your field recording kit has all of the items listed above. There is something especially depressing about travelling hundreds of kilometres to cover a story – only to find that when you get there you don't have a microphone cable … or any one of the other essential items!

Using the field recording normally involves the following steps:

1. Connecting the microphone to the recorder. Before starting a recording, you must connect the microphone cable at one end to the microphone, and at the other to the recorder. The microphone cable usually has a female XLR connector at the microphone end. The recorder end of the cable could use a variety of input connectors, including male XLRs, ¼" plugs and mini-jack plugs. Insert these connectors gently and correctly. Remember to press down the tab on XLR connectors.

Under C in the A-Z for more about connectors.

Do not force these connectors. It is easy to damage your field recorder by forcing a microphone connector. Excessive use of force may push it right through the case and into the delicate machinery inside.

2. Connecting headphones to the recorder. Gently push the ¼" plug or mini-jack plug into the jack for the headphones.

3. Insert the media. Follow the operating instructions of your recorder and gently insert a recordable tape, MD or DAT, as is appropriate, into the recorder.

4. Switch on and check your levels. Before you start recording, conduct a short test recording to check your recording level. Tape based recorders have a play and record button that must be pressed together to start recording. MD and DAT recorders may only have a record button. Check your user manual for the exact instructions for your machine. The recorder should have a level meter. Once you have pressed the right buttons to start recording, say a few words and have your guest say something. While this happens, check the level on the meter. If you are not getting a good level, adjust the microphone position or the recorder's input level control to get an acceptable level. Check the quality at the same time by using the headphones to listen to this test recording.

Under M in the A-Z for more about meters.

5. Make the recording. Once you've checked that everything is connected, and the levels are right, you are ready to record.

6. Once you have finished your recording, take your tape, DAT or MD from the recorder and label it. Stations can have many tapes lying around. The aim of the label is to help you find your recording again. The best way of marking the tape is to list the contents, the date of the recording and your name.

F

7. Replace the recorder in its case. Make sure you have all the components of the field recording kit, and put them back in their carry case. Gently coil the microphone cable. Do not twist the cable tightly, as repeatedly doing this will break the wires inside the cable. If you disconnect the microphone cable from the microphone and the recorder, disconnect them gently by pulling on the connectors and not the cable. Remember XLR connectors have a locking tab that must be pressed in before they will disconnect.
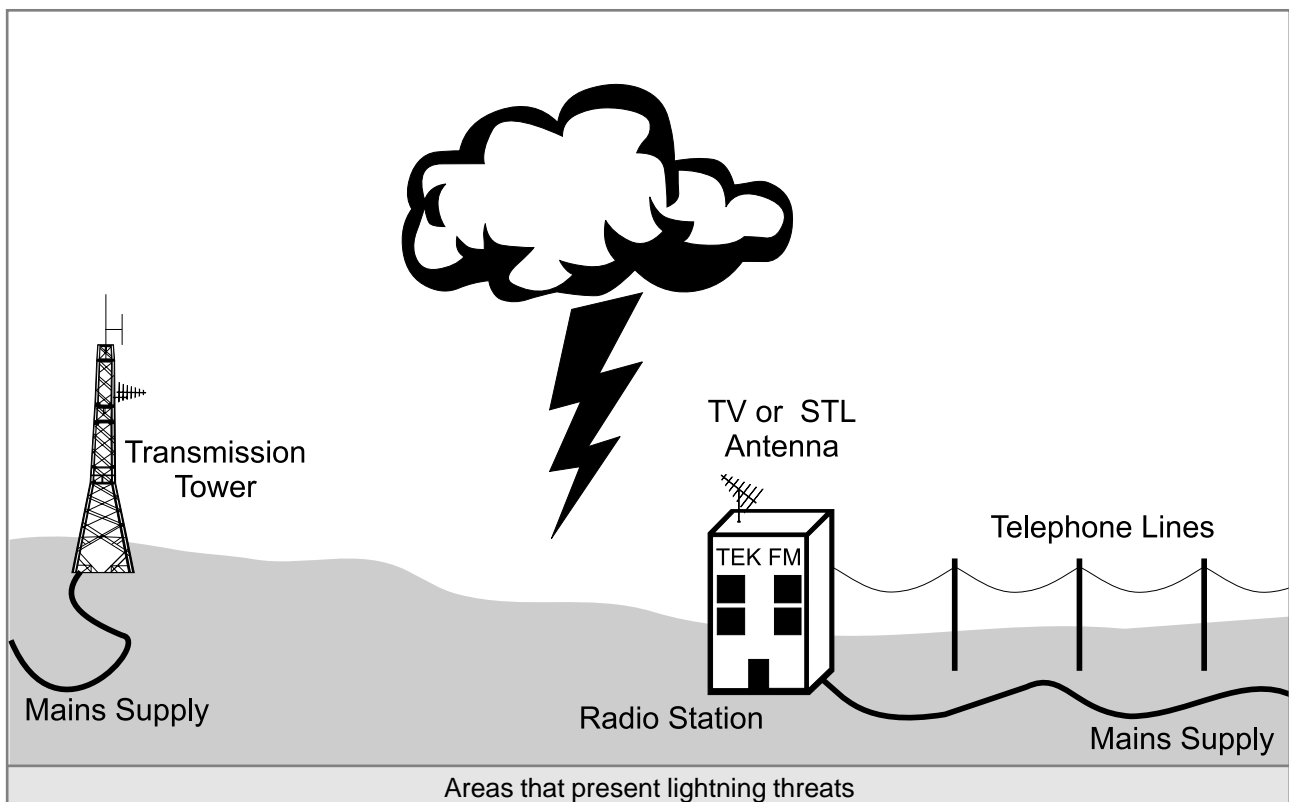
Lightning Protection

Many parts of South Africa have frequent electrical storms, and lightning is a real threat to studio equipment.

Lightning is a massive release of electrical energy from charged particles in the atmosphere. This enormous surge of electrical energy can sometimes find its way into the studio through various electrical connections. When this happens, the equipment that encounters this power surge will be severely damaged.

To avoid lightning damage, many stations take measures to protect themselves from electrical surges resulting from lightning or other irregularities in the mains supply.

Direct lightning strikes only account for a very small proportion of damage to electronic systems. Most damage results from nearby lightning strikes, when power enters the electrical system. We are not going to go into the scientific details of how power enters into the electrical system of your radio station. However, it is very important to know how we can block it and prevent or limit damage. The next diagram shows the main areas of concern for lightning protection.

Transmission Tower

Mains Supply

TV or STL Antenna

TEK FM

Telephone Lines

Radio Station

Mains Supply

Areas that present lightning threats

Lightning can enter the radio station through a direct strike, the telephone lines and the mains supply.

The Transmission Tower

The transmission tower is usually a target for direct strikes, as it is often mounted in high areas and is often the highest point. For this reason, transmission towers should be properly earthed and be installed with a down conductor.

A down conductor starts with a metal spike at the top of the transmission tower. This spike should stand much higher than the transmitting antenna. The aim is to try to ensure that if lightning strikes your tower, it will strike the spike before the antennas.

The spike is connected to a thick copper cable that runs down the tower and is earthed at the base of the tower. This leads the lightning to earth through a path that doesn't pass through your very expensive transmission equipment.

Correct earthing procedures for the down conductor are essential. If your station uses its own transmitter, you should discuss earthing the transmitter tower with your supplier and other transmission experts to ensure that it is properly earthed.

The Mains Supply

The mains supply at both the transmission tower and the station is another potential source of lightning damage. Direct lightning strikes to power lines kilometres away and inductive coupling to power lines can result in electrical surges on the mains supply. These power surges can seriously damage or even destroy studio audio and computer equipment.

To protect against this, some form of mains protection is advised. This can take many forms. The most common method used in community radio stations is to connect all studio equipment to the mains supply via a protected power outlet.

Many suppliers install power conditioning and surge protection units. These units plug directly into the mains supply and studio equipment is then plugged into the unit, which provides them with a "clean" protected power supply.

Telephone Lines

The telephone lines coming into the station are another potential source of lightning damage. Nearby lightning strikes can cause enormous surge voltages on the phone lines.

This large amount of electrical power can damage equipment connected to the phone lines, such as telephone hybrids, modems and fax machines.

The equipment connected to these devices is also at risk. A lightning strike to the phone lines at Barberton Community Radio severely damaged not only the telephone hybrid, but also several channels on the studio mixing desk, a CD player and a microphone.

Phone lines can be protected using a variety of surge protection devices. Generally, these connect in line with the phone line. Request that your supplier

installs these devices as part of the telephone hybrid system. It may also be worth contacting Telkom or your local town council's electrical department, as they will probably have experience with appropriate lightning protection measures in your area.

Equipment Connected to the Studio Building

A final concern is antennas connected directly to the studio, such as the STL, or TV antennas or satellite dishes. The position of these antennas can often make them attract direct lighting strikes. As they are connected directly to studio and transmission equipment, any surge along the antenna cable can cause severe damage to that equipment.

Reputable technicians should install the antennas and you should be happy that they have addressed any concerns you have about lightning protection. Surge protection devices are also available for the antenna cables.

The northern parts of South Africa experience many electrical storms each year, and for stations in these areas lightning protection is essential. When you consider how much it costs to repair or relace damaged equipment, the cost of lightning protection seems very little. A single lightning strike can put you off air, or compromise your sound. This will cost you thousands and you will lose listeners. Rather make sure your station is properly earthed and protected.

Loudspeakers

Our perception of the sound we are producing in the studio is based on what we hear through the studio loudspeakers. This means that we must use high quality speakers, if we want an accurate idea of what we are doing. As if to emphasise the importance of the speakers in a studio they are often called "monitors". We don't just listen to sound in the studio, we monitor (check) it for quality.

A speaker is an electro-acoustic transducer that produces sound waves in response to an electrical audio signal. Effectively, a loudspeaker works in the opposite way to a microphone, where a microphone produces an electrical audio signal in response to sound waves.

Under S in the A-Z for more about sound.

Under M in the A-Z for more about microphones.

Loudspeakers are used in combination with an amplifier. The amplifier increases the power of the audio signal from a piece of source equipment, such as a CD player.

Under A in the A-Z for more about amplifiers.

The signal from the amplifier is sent to the speaker, which turns the audio signal into sound waves that we can hear. When everything is working as it should, the speaker produces nearly the same sound waves that a microphone originally recorded onto the tape, CD, etc.

Different Parts of the Speaker, and What They Do

The speaker consists of three main sections: drivers, a cross over network and an enclosure (the speaker box).

Traditionally, speakers have one or more drivers. A driver produces sound waves in a way that is similar to the way a microphone records sound, but in reverse. The driver, like a microphone has a flexible diaphragm, but instead of responding to a sound wave and generating an audio signal, which is what a microphone does, the driver responds to an audio signal by rapidly vibrating the diaphragm or cone, to produce sound waves.

The diagram on the next page shows how a driver works. It consists of a cone, usually made of paper, plastic or metal. The cone is attached at the wide end to the surround.

This rim of flexible material (the surround) allows the cone to move, and is attached to the driver's metal frame, called the basket.

The narrow end of the cone is connected to the voice coil. The coil is attached to the basket by the spider, a ring of flexible material. The spider holds the coil in position, but allows it to move freely back and forth. Some drivers have a dome instead of a cone. A dome is just a diaphragm that extends out instead of tapering in.

Cross-section of a moving coil cone-type driver
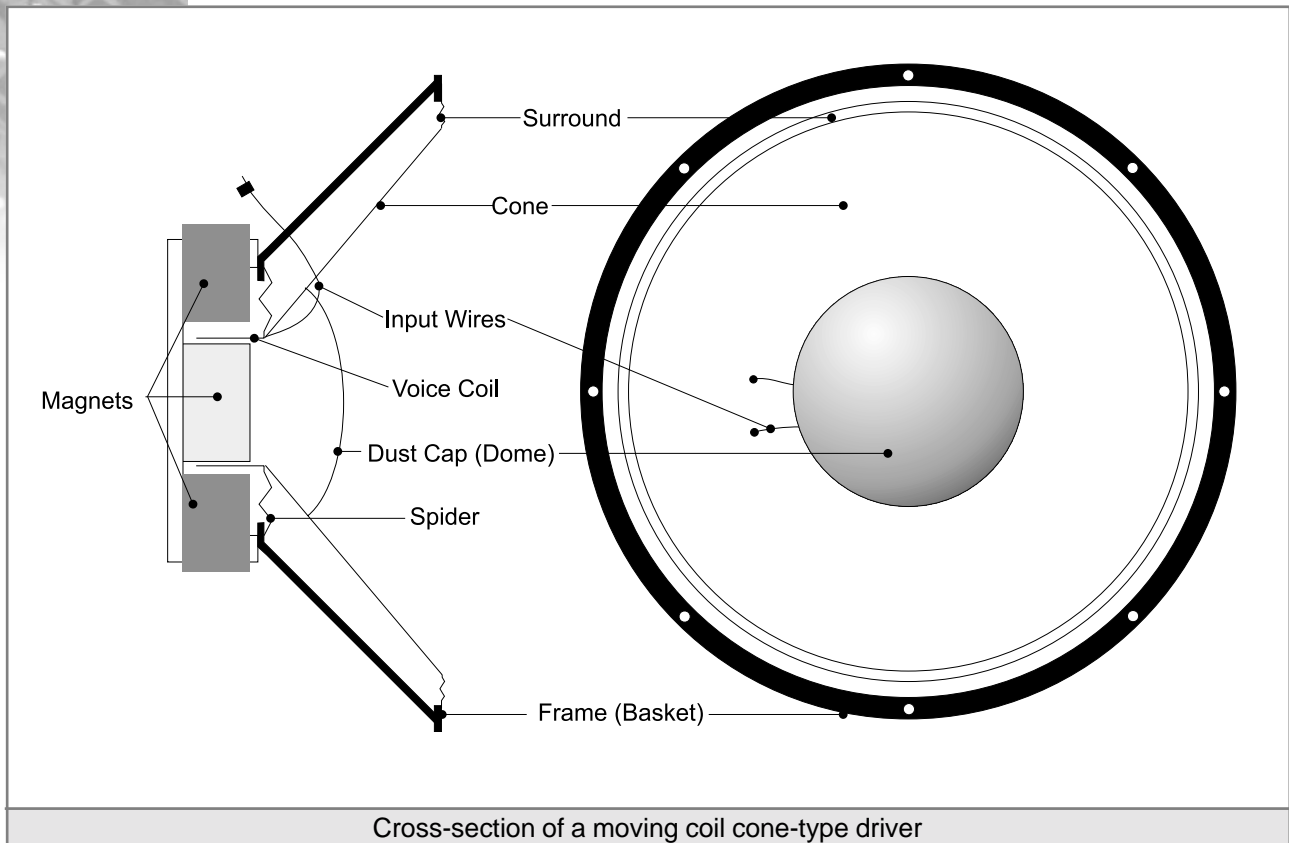
The voice coil is an electromagnet, a coil of wire, wrapped around a piece of magnetic metal, such as iron. Running electrical current through the wire creates a magnetic field around the coil, magnetising the metal it is wrapped around. As the electrical current changes, so the magnetic field changes. The voice coil is mounted in a constant magnetic field created by a permanent magnet.

These two magnets - the electromagnet and the permanent magnet - interact with each other as any two magnets do. The changing electrical current in the voice coil causes its magnetic field to keep changing. Depending on this magnetic field the voice coil is either attracted or repelled by the permanent magnet. The changing audio signal constantly reverses the magnet forces between the voice coil and the permanent magnet. This pushes the coil back and forth rapidly. When the coil moves, it pushes and pulls on the speaker cone. This vibrates the air in front of the speaker, creating sound waves.

The Woofer and Tweeter

The larger the cone, the more difficult it is to move it backwards and forwards at higher frequencies. For this reason, speaker units typically have a large driver, called a "woofer", for low frequencies up to about 2 kHz, and a smaller driver, called a "tweeter", for frequencies above 2kHz. If a speaker has more than one driver it needs a crossover network. The crossover network is a filter that feeds the correct band of frequencies to each of the drivers.

The Enclosure

The drivers and the crossover are housed in a speaker enclosure. The enclosure serves a number of functions. Firstly they make it much easier to set up the speakers. Everything is in one unit, and the drivers are kept in the right position, so they work together to produce the best sound.

They are usually built with heavy wood or another solid material that will effectively absorb the driver's vibration. If you simply placed a driver on a table, the driver would vibrate so much it would probably drown out the speaker's sound.

The speaker enclosure affects how sound is produced. The speaker cone moves backwards and forwards, so it is actually producing sound waves behind the cone as well as in front. We only want the sound waves generated from the front of the speaker, so the enclosure has to deal with these "backwards" waves.

One way of dealing with these backward waves is to put the driver in a sealed enclosure. This enclosure is completely sealed, so no air can escape. This means the forward wave travels outward into the room, while the backward wave travels only into the box. This works well with small, low power speakers. For higher power speakers with larger cone movements, like the ones usually found in the studio, other techniques are used, such as the vented enclosure.

The vented enclosure has an opening, usually in the front panel of the speaker. The backward motion of the cone pushes sound waves out of this opening, boosting the overall sound level. The design of the interior of the cabinet, and the size and position of the opening, must be carefully calculated to make this design work properly. The main advantage of these enclosures is efficiency. The power moving the driver is used to emit two sound waves rather than one.

These are some of many different speaker designs. Typically the monitors used in community radio stations are moving coil speakers, with two drivers and vented enclosures.

Driving the Loudspeakers

The monitor speaker has to driven by an audio power amplifier. The power amplifier in most studios is fed an audio signal from the mixing desk, and the level control for the studio monitors is usually found on the mixing desk.

Active monitors are speakers that have amplifiers built into the speaker enclosure. These monitors are often more expensive, but have several advantages. Because the amplifier and the speaker are designed to work together, the speaker is often better sounding, and there is no risk of damaging the speaker by overdriving it.

Maintenance of Speakers

Speakers should not require any maintenance. The greatest risk to speakers is that they can be "blown". This happens when too much power is used to drive the speaker and the voice coil tears itself from the cone. Stations are advised to make sure the late night and weekend presenters are discouraged from boosting the sound to levels that will blow the speaker!

The speaker cones and domes are often made of rigid, yet delicate materials. It is advised that you keep the covers on the monitors. This protects the cones from direct sunlight that can cause them to perish over long periods of time, and also stops people from touching them. Studio visitors love pressing the exposed cones and the domes on monitors. But this will damage the speaker.

Meters and Metering

Most professional audio equipment used for recording will have a meter that can be used to monitor the sound level.

In the studio, mixers will have one or more sets of meters, for monitoring the level of broadcast audio. Tape, MD and DAT recorders and field recorders should also have a meter of some kind to allow monitoring of the audio level.

Using meters to monitor the sound level is essential to ensure that your radio programme or recording signal level is not so high that it will cause distortion, or so low that noise and hiss will degrade the sound.

Very few people can accurately measure a signal level by just listening to it. Natural hearing is excellent for monitoring the quality of an audio signal, but is not appropriate for a quantitative measurement. This is why meters are so essential.

Different pieces of equipment will use different types of meters. Being able to identify the type of meter on your equipment, and knowing how to read that meter, will ensure that your broadcasts and recordings are always of high audio quality.

The VU Meter

One of the most common measuring devices is the VU meter. VU is an abbreviation for *volume unit.* On a VU meter, the zero (0VU) mark indicates the maximum distortion-free level that can be handled by the device. Other values greater or less than the zero level are indicated as positive and negative decibels.
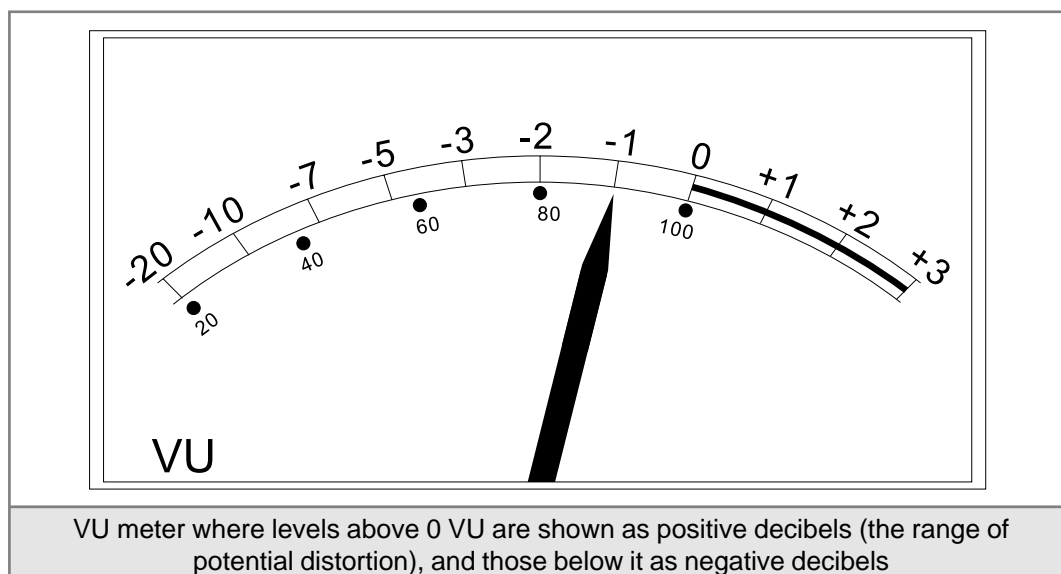
Under D in the A-Z for more about decibels or dB.

0VU (zero VU) corresponds to a signal level of +4dBu and represents a voltage level of 1.228 volts. In practice, analog devices have considerable *headroom* above 0VU to allow for peak levels. This means that during programme playback the signal level should be controlled so that the VU meter reads around 0VU, peaking at not much more than +2VU

The VU meter is most useful as a line-up aid. When the mixer is set up, the channels are lined up so that each piece of equipment produces an equal signal when the fader is at a reference position. Usually, the reference position for the fader is –10 (minus 10).

Under M in the A-Z for more about mixers.

The VU meter is used to measure the level from a piece of equipment when playing a 1000Hz test tone at a level of 0dBu. 0dBu is used as a reference level which corresponds to –4VU. The gain of each channel on the mixer is typically set so that when the fader is at the –10 position, the VU meter reads –4VU in response to the test tone. The diagram below shows a VU meter with a decibel scale from -20 to +3 dB.

VU meter where levels above 0 VU are shown as positive decibels (the range of potential distortion), and those below it as negative decibels

The VU meter responds to the energy in a signal and not the peak voltage of the signal. This means it is not the best meter for measuring the peaks in a signal. The very rapid fall time of the needle of the VU meter also makes it difficult to read accurately, as the needle is in constant motion when measuring a programme signal. It is for these reasons that many people prefer a meter that measures the peaks in the audio signal under measurement.

The PPM

An alternative to the VU meter is the PPM. PPM stands for Peak Programme Meter. This meter accurately measures the peak voltage of a programme signal. The meter illustrated below is scaled 1 – 7 with 4dB per division. On the PPM a 0dBu reference level will read as 4. When used to monitor programme level, peaks should be controlled so as not to exceed 6PPM.

The PPM Scale
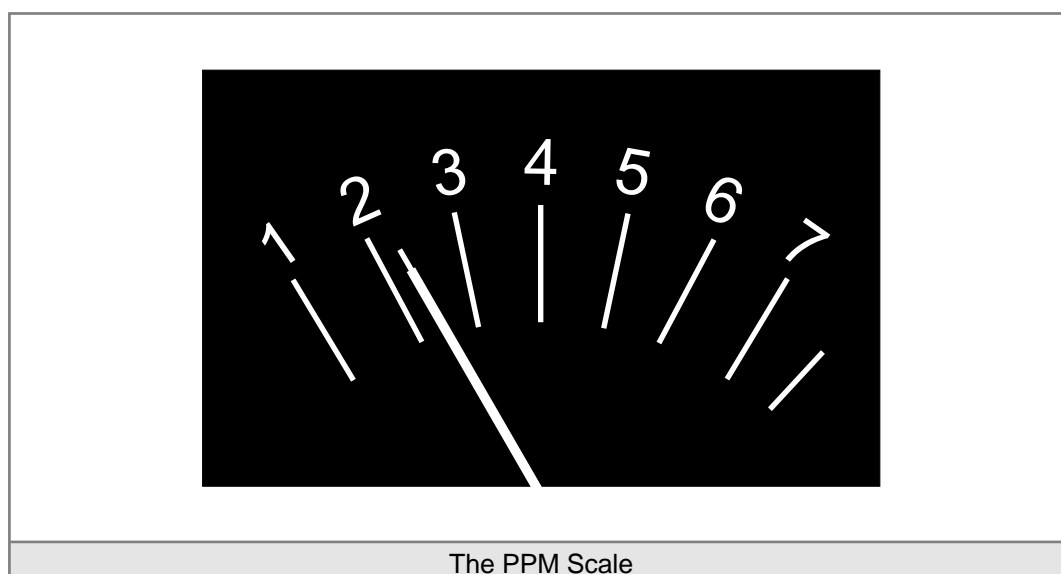
Digital Meter (DATs, MDs and Computer)

A third type of meter is the digital meter found on MD and DAT recorders, and on computer recording systems. There is no clearly defined standard for the meter, and often the scales used on different meters of this kind will be different. A typical digital meter is shown below.

L										
-dB	50	40	20	12	8	•	2	•	0	OVER
R										

A meter commonly found on digital recording equipment
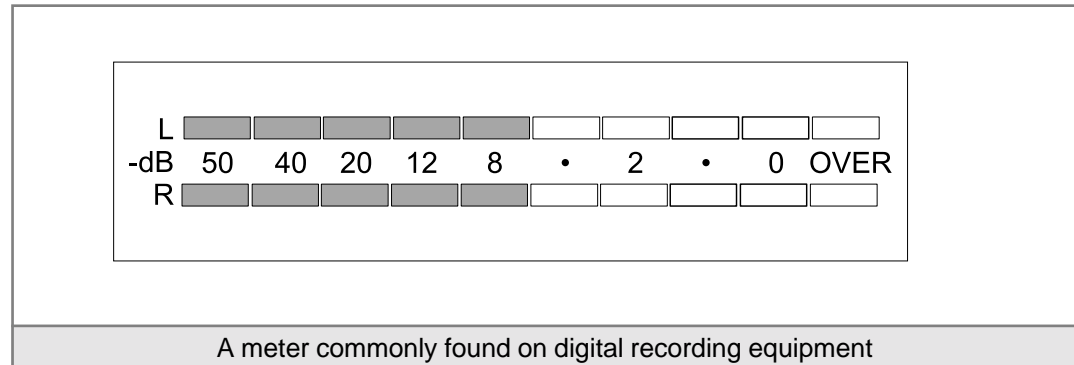
The meter is usually a LCD or LED meter, and is part of the rest of the display on the particular machine. The meter needs to be used very differently from the VU meter or PPM. The other meters are analog meters and provide a large amount of headroom, which is why in practice, the VU meter can move above 0VU, and the PPM above 4, without signal distortion.

Digital recording systems, on the other hand, have little or no headroom. This means that if the level moves above the 0 level on the digital meter, very real distortion will be heard as the digital to analog converter circuits are overloaded.

Under A in the A-Z for more about analog and digital audio.

Equipment manufacturers often do not give specific instructions on how to read these meters. Typically a DAT or MD manual will say something like "at the optimal recording level the overload light should not light up". Most manufacturers also tend to leave some headroom before the DA converters are overloaded. This means that if the overload light should flicker on briefly, it does not always cause distortion, but this varies from manufacturer to manufacturer.

The digital meter is primarily used to monitor recording levels and it is always desirable to maintain as high a level as possible without distortion when recording.

Considering the above explanation, and because no standard yet exists, let's assume that when the overload light comes on it means that the recording is distorting. A general guideline for using a digital meter to measure the signal would be as follows:

Control the signal so that the signal level is kept around the middle of the meter scale. Typically this is around –8dB or –6dB. This leaves enough headroom, so that if the signal should suddenly peak, the signal level will probably still not exceed the 0 level. This is a very general guideline and experimenting with your particular MD or DAT deck will give you a better idea of how high the input level can be pushed before distortion is heard.
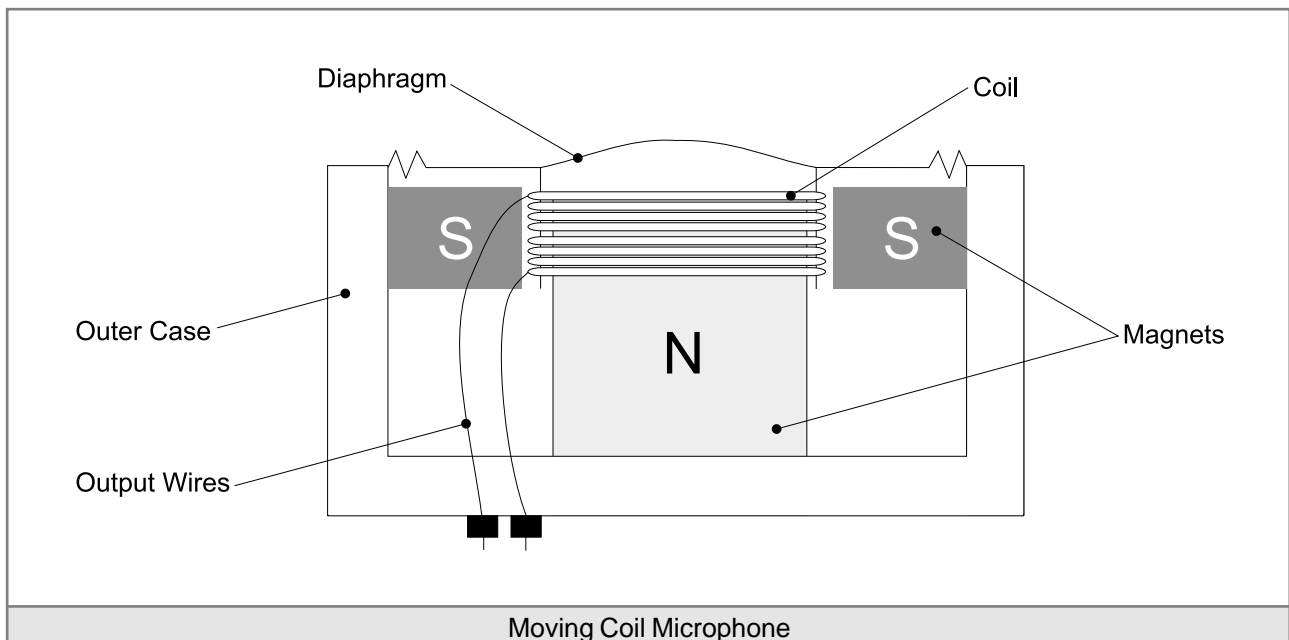
Microphone

A microphone is an electro-acoustic transducer. A transducer is a device that converts energy from one form to another. The term electro-acoustic tells us that a microphone is a transducer that converts sound waves or acoustic "energy" into an equivalent electrical signal. This electrical signal can then be fed to a tape recorder or other audio equipment. The microphone's ability to turn sound waves into electrical signals is what makes recording possible. Microphones are usually called mics (pronounced as "mikes").

How a Mic Works

To understand how a microphone works, we'll look at the operation of the moving coil dynamic microphone. We show this microphone in the diagram below.

Moving Coil Microphone

The microphone consists of a very light circular diaphragm, usually made from a thin plastic film, that is attached to a coil of very fine wire. The coil fits into a gap between the North and South poles of a permanent magnet.

When sound waves travelling through the air hit the diaphragm, the diaphragm moves back and forth (vibrates). The movement of the diaphragm causes the coil of wire to move. The coil sits in the magnetic field of the two permanent magnets. Because of electromagnetic induction, the movement of the coil in this magnetic field causes a small electrical current to flow in the wires of the coil.

This current will vary in exactly the same way that the movement of the diaphragm varies. For example, when someone shouts into the microphone, the high sound pressure will cause the diaphragm to move much more than it will when they whisper. This larger movement will produce a larger current in

the coil. The current produced in the coil is very small, and must be amplified to a useful level by a microphone pre-amplifier, such as the input of the "mic" channel of your mixer.

Under S in the A-Z for more about sound.

Some Different Kinds of Microphones

The moving coil dynamic microphone is only one of many different kinds of microphones that are available. The main types of microphones are as follows:

Dynamic Microphones

Dynamic microphones can be of the moving coil (described above), or ribbon variety. The ribbon microphone operates on the same principle of electromagnetic induction. But instead of a coil and a diaphragm, the motion of a thin, usually aluminium, ribbon, within a magnetic field creates the electrical signal. Moving coil dynamic microphones are used extensively in the broadcast and production studios and for field recording.

Condenser or Capacitor Microphones

A capacitor is an electrical component that is made up of a pair of parallel metal plates separated by insulating material. If an electrical voltage is applied to the two plates, then the capacitor can store electrical charge. If the distance between the parallel plates changes once the capacitor is charged, the voltage across the two plates will change. This principle is used in the condenser or capacitor microphone.

One plate of a capacitor is used as a diaphragm that can move in response to sound waves. The motion of the diaphragm changes the voltage across the plates in direct response to the sound waves that caused the motion. This changing voltage can then be fed into a mixer, or recording equipment.

Condenser microphones require a power supply to charge the two plates. This is usually supplied by a 48V "phantom" power supply from the mixing desk or microphone pre-amplifier.

Note – if you ever need to swap a condenser microphone for a dynamic microphone, turn the power supply off to prevent damaging your dynamic microphone.

Condenser microphones are often used in the broadcast and production studios as they offer the highest sensitivity and best noise performance of any studio microphone.

Electret Microphones

These are similar to condenser microphones, in that they have a permanent electrostatic charge on the plates, but they do not use an external power supply. This makes them cheaper, but they often deliver much poorer quality audio than a condenser microphone. Electret microphones are often small in size and can frequently be found on domestic audio equipment, such as portable tape recorders.

Carbon Microphones

These are used in telephone handsets. These are inexpensive and only have a frequency response of about 300 to 3000 Hz, making them unsuitable for regular use in the studio.

Directional Response

Microphones do not only differ in the way that they are physically constructed, but also in their sensitivity to sound from different directions. The pattern of sensitivity of a microphone can be called the *directivity*, *directional characteristic, directional response, field pattern* or *polar pattern*. A polar diagram is normally used to illustrate the directional response of a microphone.

The polar diagram shows the microphone's output sensitivity with respect to direction over 360°. Usually the microphone's directional response is measured for various frequencies, the results of which may be combined in a single diagram. In many cases, a microphone that has a uniform response over a large frequency range is desirable.

Omnidirectional Mics

Microphones are broadly classified as having either an omnidirectional polar pattern or a directional polar pattern. Omnidirectional microphones are sensitive to sound from all directions, while directional microphones are sensitive to sounds only from certain directions. The polar pattern of an omnidirectional microphone is illustrated below: The 0° direction on the polar diagram is often referred to as "On Axis".

Omnidirectional Microphone Polar Pattern
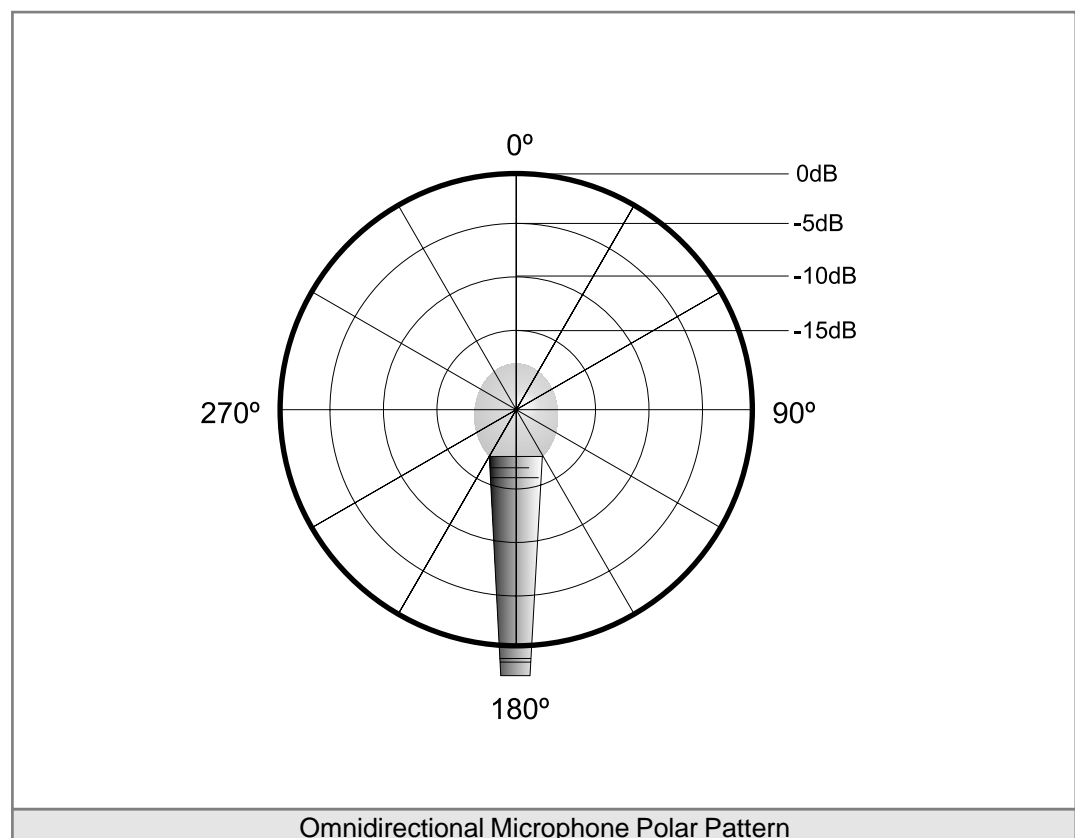
Cardioid Mics

Directional microphones can have many different polar patterns. The pattern commonly used for presenter's microphones in the radio studio is the cardioid pattern illustrated below.

0°

0dB
-5dB
-10dB
-15dB

270°

90°

180°

Cardioid Microphone Polar Pattern
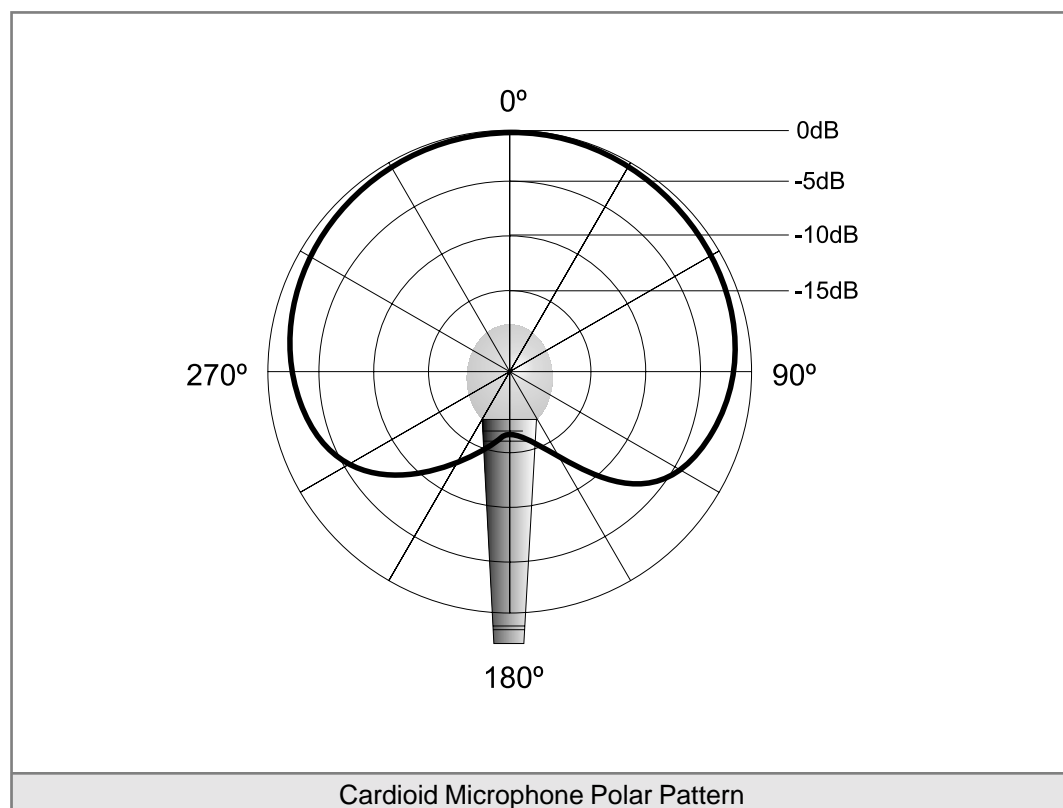
The microphone is described as "cardioid" because the polar pattern is shaped a lot like a heart. The microphone is most sensitive to sounds that come from the front and least sensitive to sounds from the back. This design means that the microphone will be very sensitive to a presenter's voice - that is, on axis, but will not pick up much of the unwanted background noise in the studio.

There are variations on the cardioid microphone's response, such as the hypercardioid and supercardioid microphones. These are even more directional designs. The polar pattern of a hypercardioid microphone is shown on the next page.
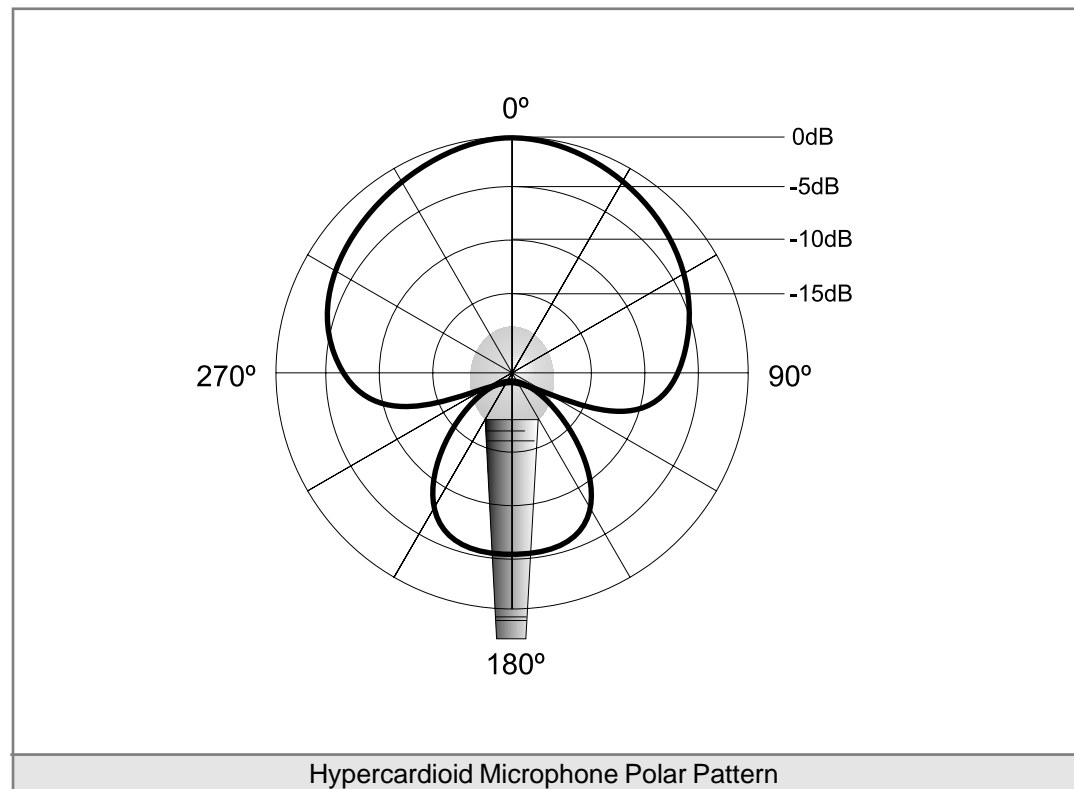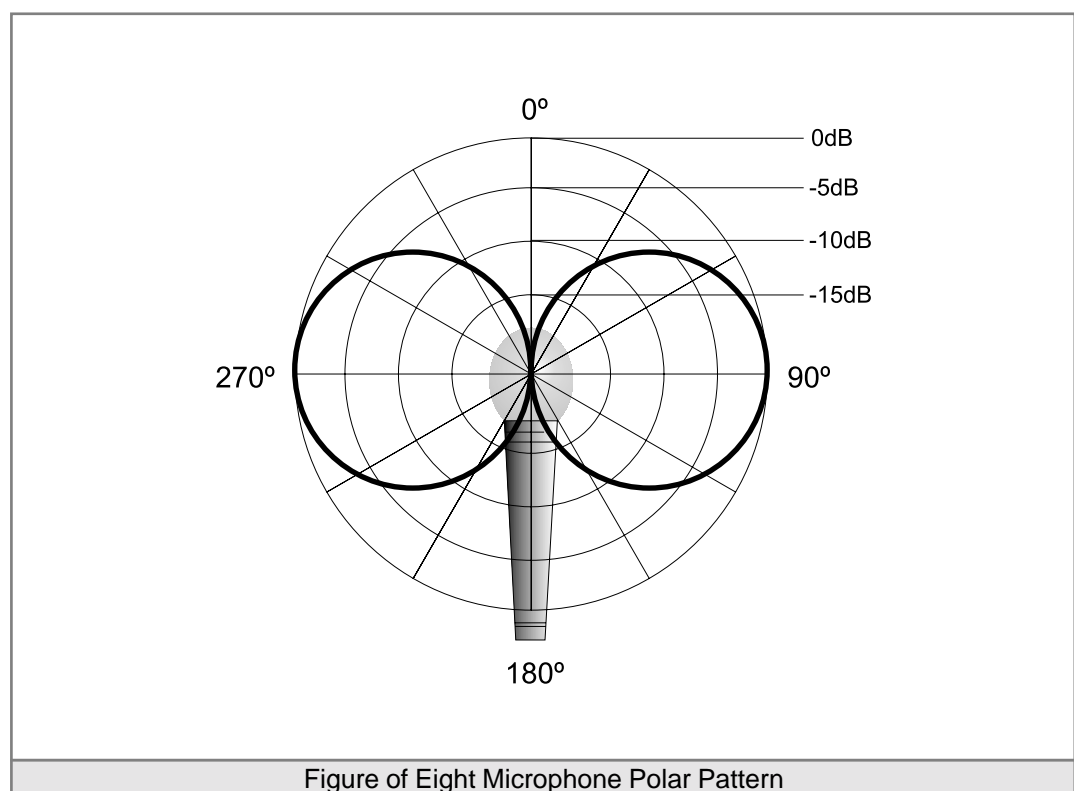
Hypercardioid Microphone Polar Pattern

Figure of Eight Mics

The figure of eight microphone is sensitive to sound coming from the sides of the microphone. The polar pattern of a figure of eight microphone is shown below.

Figure of Eight Microphone Polar Pattern

The figure of eight microphone is usually only used for specialist applications. It has often been used for live back-up vocals allowing two singers to share one microphone, with one on either side.

Tips for Using Mics

It is worth noting that the physical construction of cardioid and figure of eight microphones causes them to exhibit the proximity effect. The proximity effect boosts low frequencies when the sound source is very close to the microphone. This effect benefits a presenter or vocalist by giving a larger than life sound. On the other hand, if it is not taken into account, the proximity effect can cause voices to sound distorted and muddy.

The proximity effect is one of many things that can cause microphones to produce unexpected sounds. Handling the microphone can cause large amounts of noise. This is why whenever possible a microphone should be mounted on a solid boom stand. Certain voice sounds, particularly the plosive P and B sounds, cause the microphone to pick up a popping noise. This is caused by blasts of air from the presenter's mouth slamming into the microphone's diaphragm.

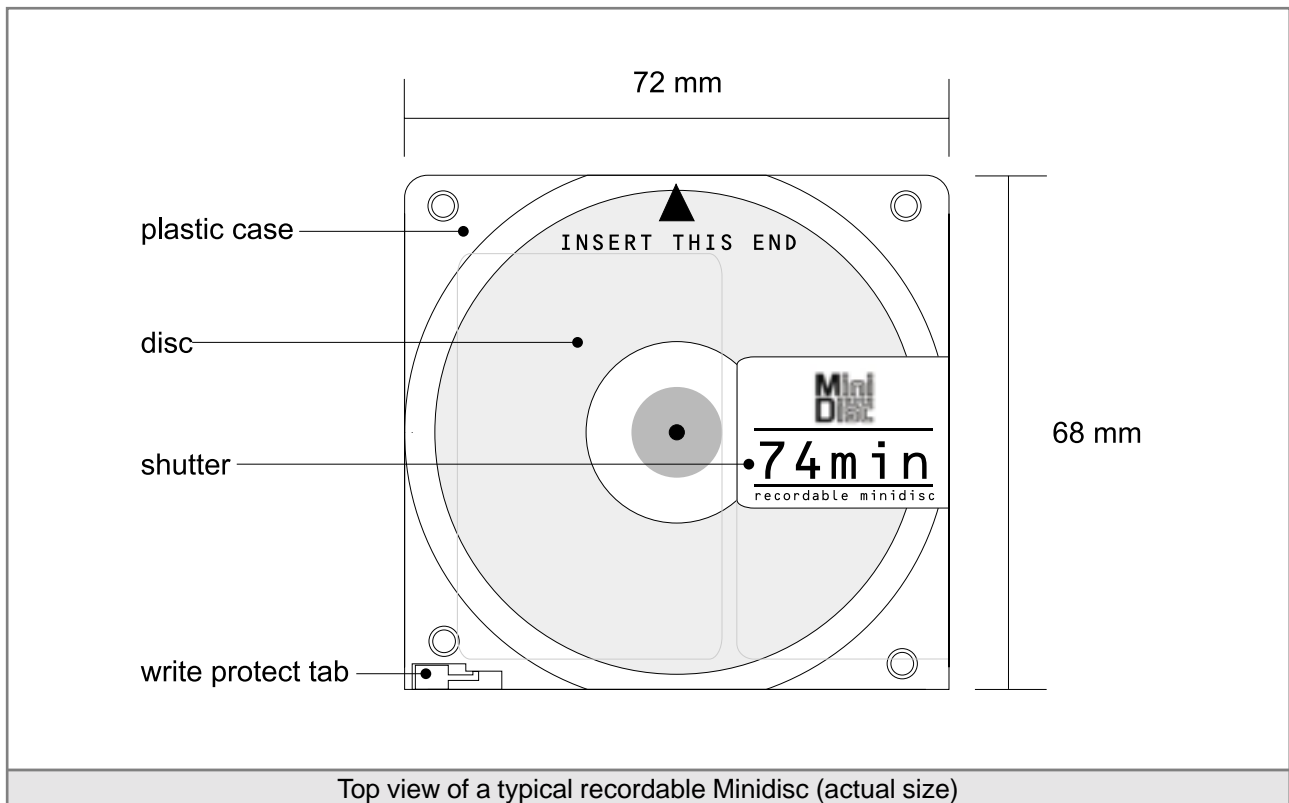To reduce this effect, the presenter should point the microphone just above or below the mouth, or if necessary, use a pop shield. A pop shield is a fine mesh of plastic or metal that is placed between the mouth and the microphone. In general, a good working distance for a cardioid presenter's microphone is between 10 and 25 cm from the microphone. This is, of course, highly dependent on the presenter's voice.

Minidisc

The Minidisc (MD) format was introduced in 1992, by Sony. MDs were intended to replace cassette tape as a new digital audio playback and recording system. The MD, like the cassette, was intended to be portable. This meant it had to be small and able to withstand vibrations and rough treatment outside the safety of the home or studio.

The diagram below shows the make-up of a typical recordable MD.

72 mm

plastic case

INSERT THIS END

disc

MiniDisc

shutter

74min

recordable minidisc

68 mm

write protect tab

Top view of a typical recordable Minidisc (actual size)

The diagram shows just how compact the MD is. The plastic case for the MD is only 5mm thick, making it much smaller than a computer "stiffy" disc.

The silvered disc that stores the audio is enclosed by the plastic case. To read data from the disc, a MD player opens the shutter to reveal the disc surface underneath. The disc stores compressed 16-bit, 44.1-kHz sampled digital audio. This means that the MD delivers audio quality approaching that of a CD.

Under A in the A-Z for more about analog and digital audio.

Much like a cassette, the MD has a "write protect" tab. A cassette has tabs on the top of the cassette that can be broken off to prevent accidentally copying over the tape. The MD has a sliding tab. When you slide the tab to the left (as seen in the diagram), you can use the MD for recording and editing. When the tab is slid to the right the MD is "write protected". In this case no new audio can be written to the disc and none of the existing audio on the MD can be edited.

Stereo and Mono

MDs are capable of recording in both stereo and mono. You can also store stereo tracks and mono tracks on the same disc.

Stereo recording is obviously preferable when recording from a stereo source. However the advantage of using mono is that it allows you to store twice the amount of data on the MD. This means that a 74-minute MD can store 148 minutes of mono audio. For a radio journalist who mostly uses a MD field recorder with a mono microphone, this is very useful.

Types of MD

There are three different kinds of MD:

• The recordable MD shown on the previous page is the type most commonly used in radio.

• Pre-recorded MDs are also available. The shutter on these MDs only covers the bottom of the disc and they cannot be used for recording.

• A third type of disc has both a pre-recorded section and a recordable section.

Why MDs Are So Useful

There are several things that make the MD such a useful tool for radio.

A MD can be recorded with the ease of a cassette, but with much higher audio quality. When recording in mono, the MD can record for much longer periods of time than any other medium.

Recordings on a MD are stored as individual tracks, similar to tracks on a CD. This means you can easily access any track, without the fuss of having to rewind and fast-forward a tape.

But what really makes the MD special, is that the audio stored on the disc can be edited using any MD player / recorder. You do not need to copy audio to a computer or splice tape. This means that reporters can edit their recordings in the field.

Editing on MD

The MD uses a table of contents ("TOC'") data structure to link sections of audio scattered about the disc into a continuous stream. This is what makes it possible to edit tracks. Tracks can be segmented, combined, moved, or deleted, with an edit point accuracy of 60 milliseconds (12ms on modern units).

Because of the TOC structure, space freed by deleting data becomes available for further recording. MD allows you to delete tracks you don't want and with a few button pushes replace them with new ones, placing them in any order you like on the disc.

Under M, Minidisc Player / Recorder, in the A-Z for more about editing on MD.

Audio Compression on MD

CDs are 12cm across; the MD is only 64mm across, yet they both store 74 minutes of digital audio. To make this possible the MD compresses digital audio using a system called ATRAC (Adaptive TRansform Acoustic Coding).

By compressing the audio, the amount of data that is stored on the MD is reduced, making it possible to use a smaller disc. ATRAC is an audio compression system based on psychoacoustic principles.

Psychoacoustics studies how people hear, and models what parts of a sound a person with normal hearing is actually capable of hearing. The ATRAC system uses this to reduce the amount of digital data that is stored on a MD. ATRAC leaves out the parts of a sound that most of us can't hear, in such a way that most people cannot tell the compressed audio from the original signal. Nevertheless, there is a difference, and uncompressed digital formats such as CD and DAT do have greater fidelity than MD.

Handling Tips for Minidiscs

- Do not touch the disc by opening the shutter. The shutter and disc will be damaged if the shutter is forced open.

- Do not place MDs in direct sunlight, areas of high temperature, or high humidity, for example, in your pocket.

- If dust gets into the MD cartridge, wipe it with a soft DRY cloth. Do not use any liquids to clean MDs.

- When putting a label on a MD, make sure it is fixed to the correct position for labels on the disc. If the label is not properly fixed it may roll up or come loose and could cause the cartridge to get stuck in the MD player.

Minidisc Player/Recorder

The MD Player/Recorder is the machine that is used to play and record MDs. The controls on most MD players are very similar to those on a CD player. This is because playing a MD is very much like playing a CD – you insert the disc, select a track and press play.

You might like to revisit the section on compact disc players, under C in the A-Z, to have another look at how a CD player's controls work before reading more about MDs.

All MDs are marked to show which end of the disc goes into the player.
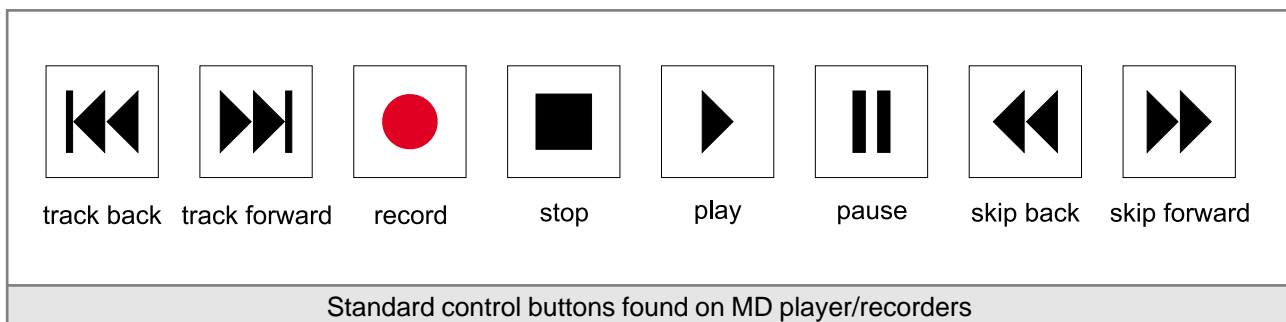
Under Minidisc in the A-Z for a diagram of a MD.

Always make sure that the disc is inserted correctly. Forcing a disc into the player in the wrong way will damage the player, and discs can get stuck inside the machine.

Standard Control Buttons on the MD Player/Recorder

The buttons on a MD player are usually marked with the standard symbols used on most recording and playback equipment. These are shown in the diagram below.

| track back | track forward | record | stop | play | pause | skip back | skip forward |

Standard control buttons found on MD player/recorders

The TRACK Buttons

Choosing a track on the disc involves pressing the TRACK or "search" buttons. Some MD players use a jog wheel instead of a button to select tracks.

Playing and Cueing the MD

Once you've selected a track, press the PLAY button to play the track. Many MD players do not instantly start playback once PLAY is pressed. This is because it takes a short time for the player to locate the playback data on the disc.

To avoid the "dead air" that this short delay creates, MD players can be "cued" before playback. Cueing means getting the disc ready to play by selecting the track in advance. You do this by selecting the track, and then pressing the PAUSE button before the PLAY button.

The SKIP Buttons

The SKIP buttons allow you to skip or scan through a track. By pressing these buttons, you advance or reverse the track at high speed, allowing you to find particular parts while editing or playing back.

The PAUSE Button

If the disc is playing, and the PAUSE button is pressed, playback will pause at the point where the button was pressed. Pressing PAUSE again, or PLAY, will cause playback to continue from that point.

This is a useful function for editing a MD. You play the disc until the desired edit point is found, and at that point, you press PAUSE. Edits can then be made easily at that point.

The STOP Button

The stop button returns the player to the same state as if the disc had just been put into the player. Any track selection is lost.

Programme Play

MD players are often used to play adverts and jingles. MD players that are used for this purpose must be able perform programme play. Programme play allows you play several tracks at once in any order. To do this usually involves entering a "programme" mode and then selecting the tracks in the order you want them to play out. This is vital for any station using its MD for jingles and adverts, because it is often necessary to play a jingle, and immediately follow it with an advert.

Recording on MD

Recording a MD is very similar in operation to recording a standard cassette. Even the recording controls on the MD recorder are nearly identical to those on a cassette machine.
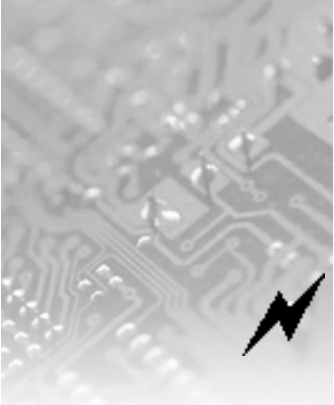
Firstly, you insert a recordable MD into the recorder. When you press the RECORD button, the recorder will enter "record and pause" mode. This means that the player is paused but ready to record.

The display on the front of the machine will then also monitor the recorder's input signal. Check this level to ensure that your recording will not distort or be too quiet.

Under M in the A-Z for more about meters and metering.

The signal level should first be adjusted at the mixing desk to make sure the recorder is receiving a good level. If the level from the mixing desk or source equipment is acceptable, but the level received by the MD recorder is too high or too low, then the input sensitivity of the recorder must be adjusted. Some recorders do this automatically, while others have an input level control. This is typically a rotary control that is turned to increase or decrease the input level, as necessary.

Once the input level measured on the MD recorder's display is acceptable, press play to start recording.

Most professional players automatically add a new track to the MD when record is pressed. However some players begin recording from the beginning of the MD. These recorders have to be skipped past the last track before you pressing record, or you will lose tracks already recorded and stored on the disc. If you select a track on the disc and press RECORD, the recorder will overwrite the selected track and all subsequent tracks on the MD for as long as the recording lasts.

Editing on MD

The MD format allows the digital audio on the disc to be edited using a MD player. Most players have several editing functions. Different manufacturers also use different names for the functions, which can make it a little confusing.

Fortunately however, all machines have the same standard editing functions. This means that even if the names of the functions are different on a new machine, you can quickly figure out how to edit with a little thought and patience.

Typically, selecting a track and then pressing a button labelled EDIT, accesses the edit functions. Each time the EDIT button is pushed, the machine will offer another editing function. Once the function you want is displayed, you press the ENTER or YES button to choose the function.

Some MD players have a dedicated button for each function. This is more convenient if you plan to do a lot of editing on MD.

The standard MD editing functions are as follows:

ERASE/DELETE

Erase can be used to remove a track from the disc or to remove all tracks from the disc.

To use ERASE, you must first select a track. Then you press the EDIT button until "erase" or "delete" appears on the display. Press ENTER or YES to select the erase function. The player will confirm the track you want to erase. If the correct track is shown, press YES/ENTER, and the track is removed from the MD. If the wrong track has been selected, press NO/EDIT, and the track will not be affected. Erasing a track decreases the number of tracks on the MD. The space on the MD that is freed up by erasing tracks can then be used to record more material.

JOIN/COMBINE/T MARK OFF

The JOIN function allows you to combine two adjacent tracks to make one track, as shown in the diagram below.

| 1 | 2 | 3 | 4 | |

↓ Join

| 1 | 2 | 3 | |

Track Numbers Decrease

Joining tracks on a MD
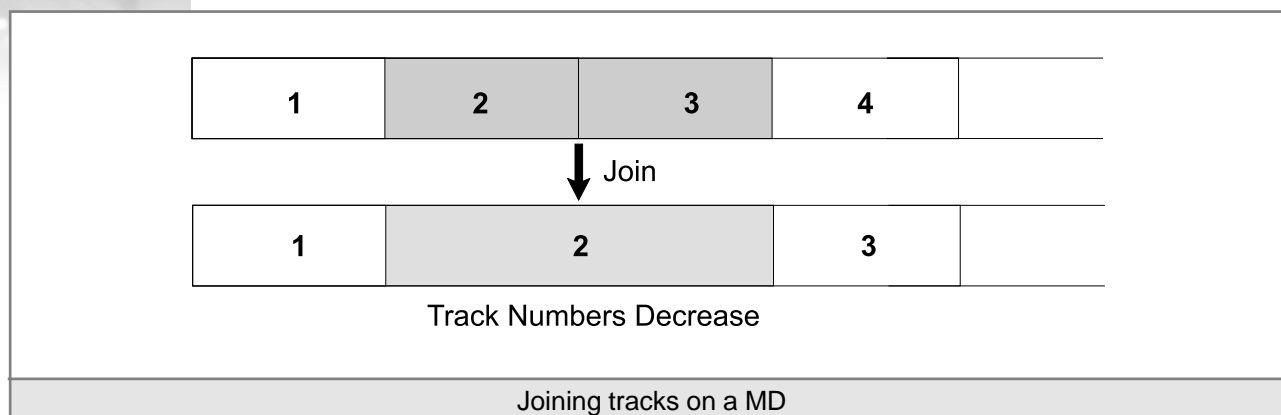
Joining the tracks as shown in the diagram involves selecting track 3. You then select JOIN/COMBINE, and then you select ENTER/YES. Tracks are always joined to the end of the track before them. Once the tracks have been joined, the total number of tracks on the MD decreases by one.

DIVIDE/CUT/T MARK

DIVIDE allows you to split up existing tracks on the MD. This allows you to give different parts of a track their own track numbers. The operation is shown in the diagram below.

| 1 | 2 | 3 | |

↓ Divide

| 1 | 2 | 3 | 4 | |

Track Numbers Increase

Dividing tracks on a MD
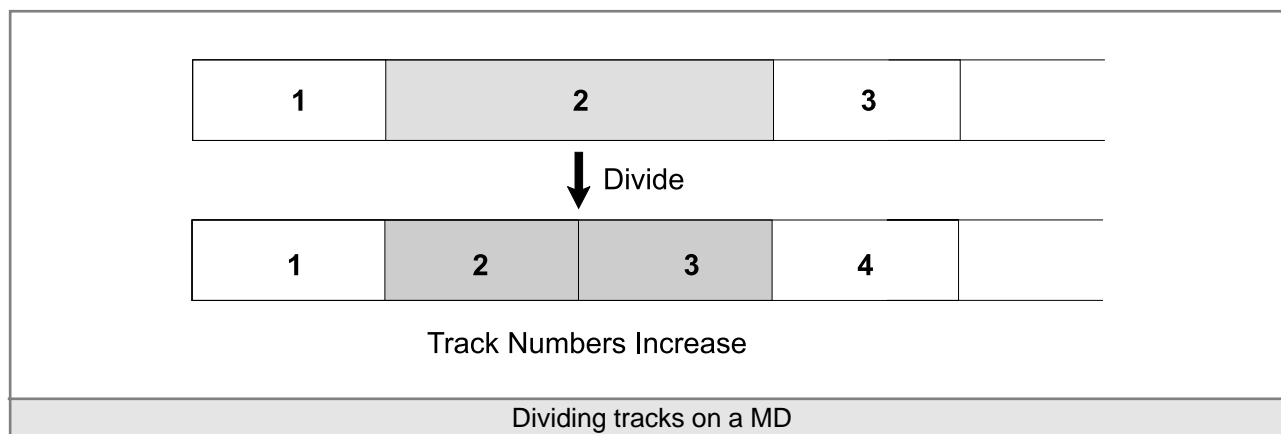
Firstly, play the track until the point that you want to divide or split it, and then press pause. Use the EDIT button to select DIVIDE. When you press ENTER/YES, the track is cut into two tracks at the selected point. When a track is divided, the number of tracks on the MD increases by one.

MOVE

The MOVE function allows you to change the order of the tracks on the MD. The function is illustrated in the diagram below.

Before Moving

	1	2	3	4	
	A	B	C	D	

Move

	1	2	3	4	
	A	C	B	D	

After Moving

Moving or changing the order of tracks on a MD
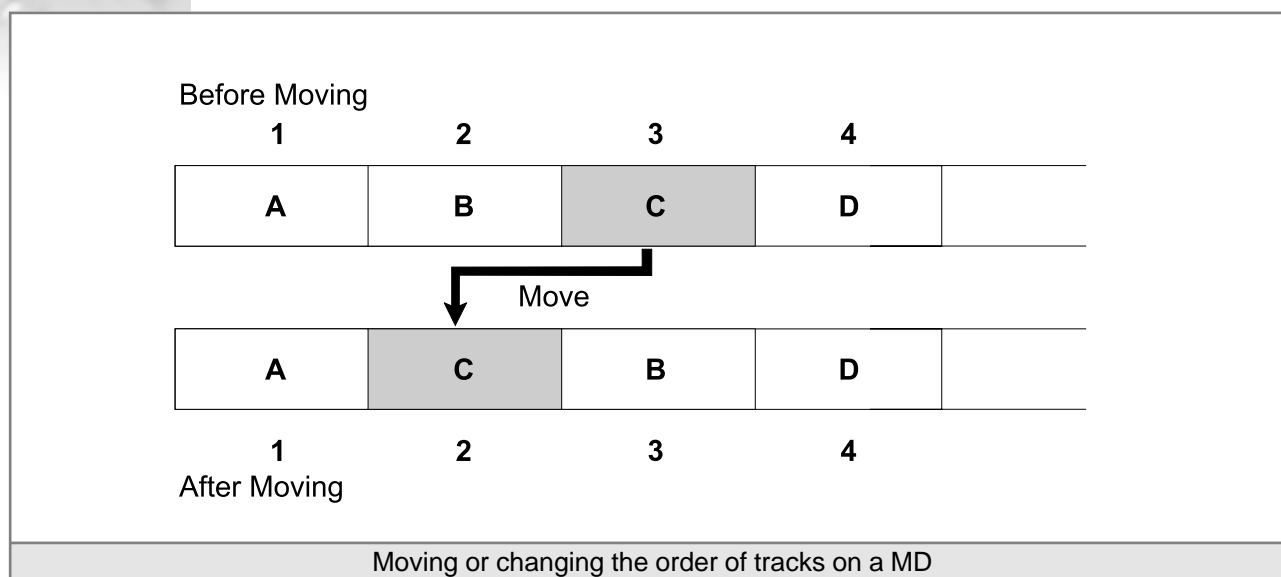
Select the track that you want to move. Then press EDIT and select MOVE. The MD player will then allow you to choose the track number that you want to move the track to. Press ENTER/YES when you have chosen the new track number.

Moving tracks will not erase any tracks, and the total number of tracks on the MD is unchanged after moving a track.

TITLE/NAME

Tracks on a MD can also be given titles. The player will display the title when a track is selected. TITLE can be accessed by using the EDIT button. Letters and numbers making up the title are often selected using the "track" button to scroll through a list of characters.

Using descriptive titles for MD tracks makes it much easier to find items on the MD. For example, don't just call your track *Thabo* – rather label it *Prim Reddy interviews Thabo Mbeki on 15th June 2002.*

Connecting to a Computer Keyboard

Many professional MD players can connect to a standard computer keyboard. The keyboards can be used to control the various MD functions. This can make editing, and particularly titling tracks, much easier and faster.

Check Your Manual

The sequence of operations to use the editing functions on your MD player will be explained in detail in the MD player's operating manual. Go through the manual for your specific MD player carefully before you start to edit.

Mixing Desk (Mixer)

There are many uses for a mixing desk, or mixer – it can be used in the broadcast studio, or the production studio, or on stage for a music show or drama, or for an outside broadcast, and in many other applications. Wherever it is used, the mixer is the centre of the audio system.

Every piece of sound equipment that you use in your studios – the microphones, the CD players, the minidisk, the effects units, turntables etc. – will somehow be connected to the mixing desk.

Before reading further, you might want to review the sections on the broadcast and production studios, on pages 5 to 19.

The best way to become familiar with a mixer is to get to know the controls of a typical analog mixer. There are many different kinds of mixers – different models of analog mixers, digital mixers, virtual mixers on a computer screen – but they all follow the same basic logic and principles. So once you've learnt about how the controls of a typical analog mixer work, it won't be too difficult to apply your learning to other kinds of mixers.

The Purpose of the Mixer

Most people's first response to a broadcast mixer is: "How do you remember what all those buttons are for?" This reaction often gets worse when you see a production mixer, which has a lot more buttons and knobs. Take comfort. The mixer is not as hard as it looks. If you take your time, work through the mixer slowly, and don't try to master everything at once, everything will fall into place.

Firstly, it is important to understand what the mixer does.

The obvious answer is that it mixes sound, and that will impress many visitors to the studio. In broadcast terms, we say the basic purpose of a mixer is to allow two or more different audio signals from different sound sources to be combined or mixed together, allowing independent control of the level of each sound in the mix.

How the mixer works

The diagram on the next page shows how a mixer works.

Input 1   Input 2   Input 3   Output

0   0   0   0

10   10   10   10

Input Level Faders   Master Level Fader

∞   ∞   ∞   ∞

Mixing Bus
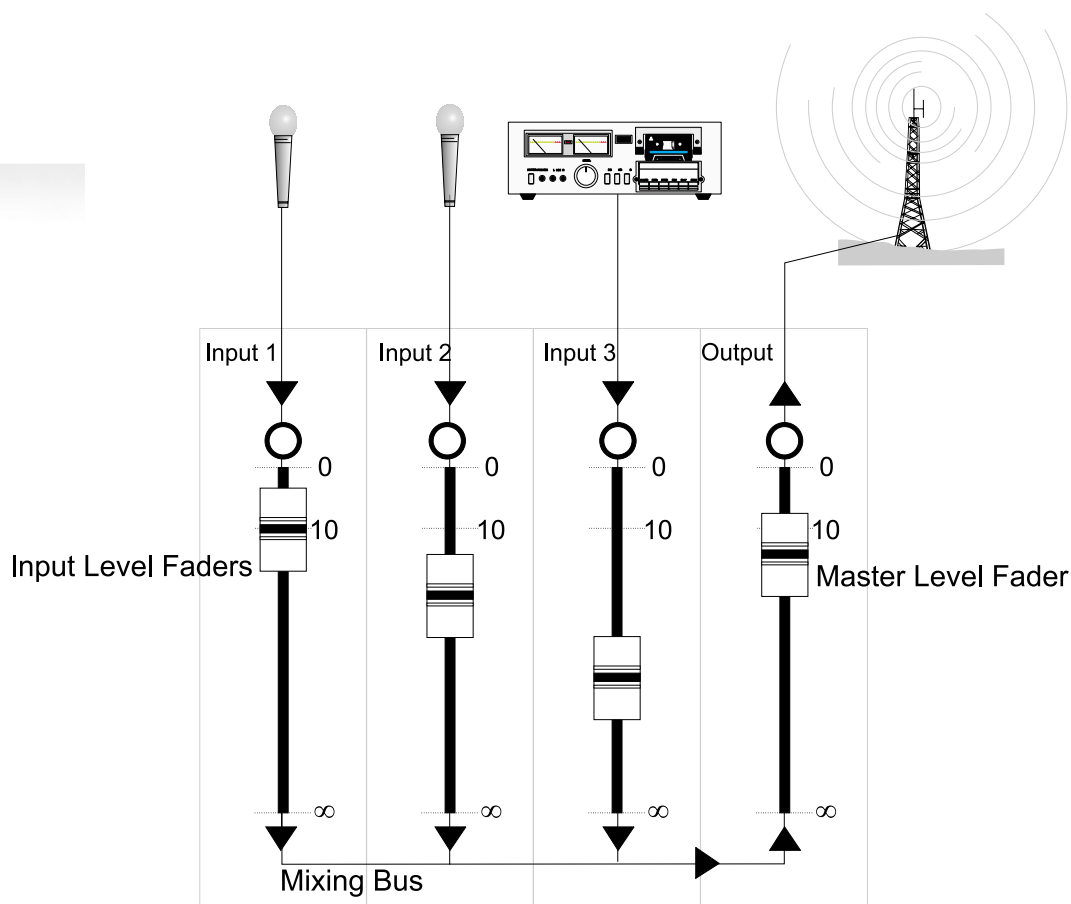
Basic operations of a mixer

In this diagram, there are two microphones and tape deck. Audio from these devices is being mixed and then sent to transmitter.

This is a typical situation in the radio studio, where you will have a microphone for the presenter, another for a studio guest and a tape deck to provide background music.

The microphones and tape outputs are all connected to the input of a channel on the mixer. The fader on each channel is used to control the signal level from each piece of equipment.

Levels

You can see from the diagram that the levels for each device are set differently. Look at the levels of the faders, and you will see that first microphone channel is set at the highest level; the second microphone is set at a lower level, and the tape deck is at the lowest level.

Imagine that the presenter is using the first microphone and an inexperienced guest is using the second microphone. The guest is shouting into the microphone. To ensure that the sound of the two voices is even, and there is no distortion, you have to drop the level of the second mic. The tape deck provides a music background (commonly called a "music bed") for both voices. Its

level is the lowest. The aim of using a music bed is to cover any silence, but the music must not be so loud that it is difficult to hear the voices.

The Mixing Bus

The three signals described in the example, at their different levels, are combined in the mixing bus. The term mixing bus is used to describe the pathway that the audio signal travels down.

Most mixers have more than one mixing bus. This makes it possible to feed different combinations of signals down each pathway (bus) at the same time.

For example, you might want to broadcast the interview we've described, as well as record it for your archives. The mixing desk may have two buses: a programme bus, and a recording bus. The programme bus is the pathway for signals that we want to be part of the radio programme. In the example, we want the presenter and guest microphones to be part of our programme, as well as the music. So we would send these signals to our programme bus.

For the recording of the interview, we may only want the presenter and guest microphone, but not the music. So we would connect these two mics to the recording bus, but not the tape deck.

The Master Fader

The master fader controls the level of the mixed signal from the mixing bus (which is hopefully the perfect combination of the two voices and background music). This mixed signal is then fed out of the output of the mixer to the transmission equipment for broadcast.

While it isn't necessary to go into too much depth about the details of mixer design, it is worth noting that there is more to the electronics of the mixing bus than just joining together each of the inputs to a piece of wire, as shown in the simple diagram above. To mix sound, specialised mixing circuits are needed, which is why we need a mixer and why it is often so expensive!

Features of a Broadcast Studio Mixer

Channels

A good broadcast studio mixer should provide many more features than those explained above. First of all, the mixer should have many more than just three channels. The mixer should have enough channels for all of the equipment you are planning to connect to it.

A/B Switches

Most mixers also have "A/B" switches. An A/B switch allows two pieces of equipment to be connected to each channel.

By selecting "A" the first piece of equipment is available on the channel; "B" then selects the second piece of equipment.

This is an essential feature so that your studio doesn't "outgrow" the mixer because you run out of inputs.

Mixing Buses

The mixer should offer more than one mixing bus allowing different mixes of signals to be sent to different equipment.

Stereo

A broadcast mixer would also have stereo channels. These allow you to control the left and right signals from a piece of stereo source equipment with one fader.

Under S in the A-Z for more about stereo.

Controlling Source Equipment

The mixer should also be capable of controlling the operation of source equipment. Most broadcast mixers make it possible, for example, to play and stop source equipment. This is often referred to as fader start, and using this feature also requires that your source equipment is capable of using it.

Meters

The mixing desk will also give you a way of measuring the levels of signals. The mixer will have a set or several sets of meters that show the level of an audio signal.

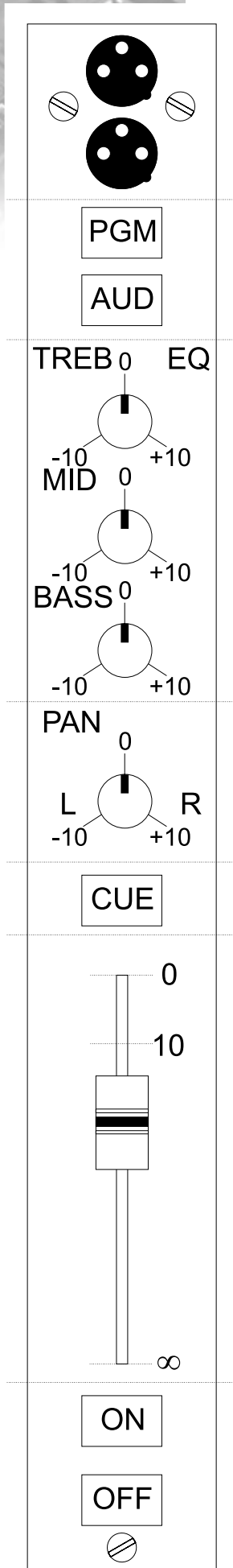Under M in the A-Z for more about meters.

Modular Design

Most broadcast mixers are modular in design. This means that the mixer is built up from different modules, depending on what you want to do with it.

To explain some of the controls on a mixer better we'll work through a typical stereo channel module, as shown in the diagram on the next page.

A typical broadcast mixer

The Connectors: Each module provides connectors for connecting equipment to the channel. Our channel is a stereo channel, with XLR connectors, one for the left signal and one for the right signal.

Routing Buttons: PGM and AUD are the names given to the buses on many broadcast mixers. PGM is an abbreviation for ProGraMme and AUD stands for audition. The buttons are used to route the sound from the channel to one or both of the buses. For example, if you press the PGM button, the sound from this channel is mixed with the PGM bus and will be part of the mixed PGM output. Most desks are set up with PGM feeding the transmission equipment and AUD feeding the recording equipment. Production mixers may be different: instead of buttons the mixer will have a rotary knob that is used to adjust the level of the sound sent to one of the buses. The buses on a production mixer may also have different names.

Equalisation: Many broadcast mixers do not have Equalisers or EQ controls on the desk. However, EQ is a standard feature on a production mixer. The equalisers are similar to the tone controls on a hi-fi system. They allow you to boost or cut certain bands of frequencies. On our channel there is a bass EQ that can control the lower frequency components, a Mid-range EQ that controls the frequencies in the middle of the audio range (the voice frequencies), and a treble control that controls the high frequency sounds. Before using the EQ it is best to try and get the best possible source sound, unless you need EQ for a particular effect. It is good practice to use EQ sparingly, especially when using it to boost a signal.

Pan Controls: Our channel is a stereo channel, and many stereo channels will have a pan control. The pan control allows the signal to be steered between the left and right channels. When the control is in the centre position, the signal is equally split between the left and right channels. When the control is turned fully to the right, the entire signal goes to the right channel output; turning it to the left sends the signal to the left channel. Any position in between varies the balance between the left and right.

Cue: The Cue or PFL (standing for **P**re-**F**ade **L**isten) button allows you to preview or cue the channel signal on headphones or monitor speakers, and often check the level on one of the desk's meters. This allows you to both check the level of a signal and find out if that really is the sound that you want to play before you play it.

The Fader: The fader controls the level of the signal on the channel. Moving the fader down will cause the signal level to drop; moving it up will cause it to increase. Faders are usually marked as illustrated. The position marked as 10 would typically be the normal level for the fader when the channel is in use. This leaves room for the fader to be pushed up when extra gain is needed for an especially low level signal. The fader is marked at the bottom with an infinity symbol (∞). This means that when the fader is at this position, ideally the signal on the channel is reduced by an infinite amount. In other words no signal comes through the channel. The 0 at the top indicates that the signal is reduced by zero, in other words not at all.

ON/OFF Remote Start Buttons: These buttons are used to switch the channel on and off, and are often used to control equipment connected to the channel. Even if the fader is up and a signal is playing through the channel, pressing the OFF button will turn the channel off, and remove the signal on that channel from the mix. Pressing ON turns the channel back on. Let's imagine the ON/OFF buttons are also used to control the signal from a CD player. In this case, the channel will be connected to a CD player that supports remote control or "fader start", as it is often called. Once a CD is put in the player and a track is selected, playback could be started by pressing the ON button. Pressing the button not only switches the channel on, but also sends a signal to the CD player telling it to start playing. Typically, pressing the OFF button will not only turn the channel off, but also cause the CD player to pause or stop playback of the CD. This remote control function is available on most professional audio equipment, so your broadcast CD, MD, DAT, reel and tape players could all potentially be controlled directly from the mixing desk.

Modulation

Modulation is a technique that plays a vital role in transmitting a radio station's signal. Let's say an audio signal has a frequency of 3kHz. This is a typical frequency component of a recorded voice. At this frequency the wavelength of the voice signal is about 100 000 metres.

A Before reading further, turn to the E in the A-Z and read about the electromagnetic spectrum.

A It might also be useful to read the section under S in the A-Z on sound and audio.

Antennas with dimensions that are less than a quarter of a signal's wavelength are very inefficient. This means an antenna that is at least 25 kilometres long is needed to efficiently broadcast a 3kHz voice signal. This is highly impractical.

For this reason, it is necessary to raise the frequency of the signal before transmitting it. Modulation is a technique that makes this possible. It allows broadcasters to use high frequency radio waves to transmit their programmes. Using higher frequency signals means that practical, efficient antennas can be used.

A Under T in the A-Z for more about transmitters and antennas.

Modulation is a technique that "adds" the information of the audio signal produced in the studio onto the radio wave that carries the audio signal to the listener. The radio wave or *carrier wave* is modulated or modified in accordance with the characteristics of the audio signal, which is the *modulating wave*. The resulting signal is called a *modulated wave.*

In radio broadcasting, the carrier wave is modulated in one of two ways.

• The amplitude of the carrier wave can be modulated or changed to carry the audio signal. This technique is called amplitude modulation (AM).

• The frequency of the carrier wave can be modulated to carry the audio signal. This technique is called frequency modulation (FM).

A Under E in the A-Z for more about the electromagnetic spectrum.

Amplitude Modulation, or AM

In amplitude modulation (AM), audio information is impressed on a carrier wave by varying the amplitude of the carrier wave above and below its unmodulated value, to match the fluctuations in the audio signal being transmitted. This is illustrated in the diagram below.

Amplitude Modulation Transmission
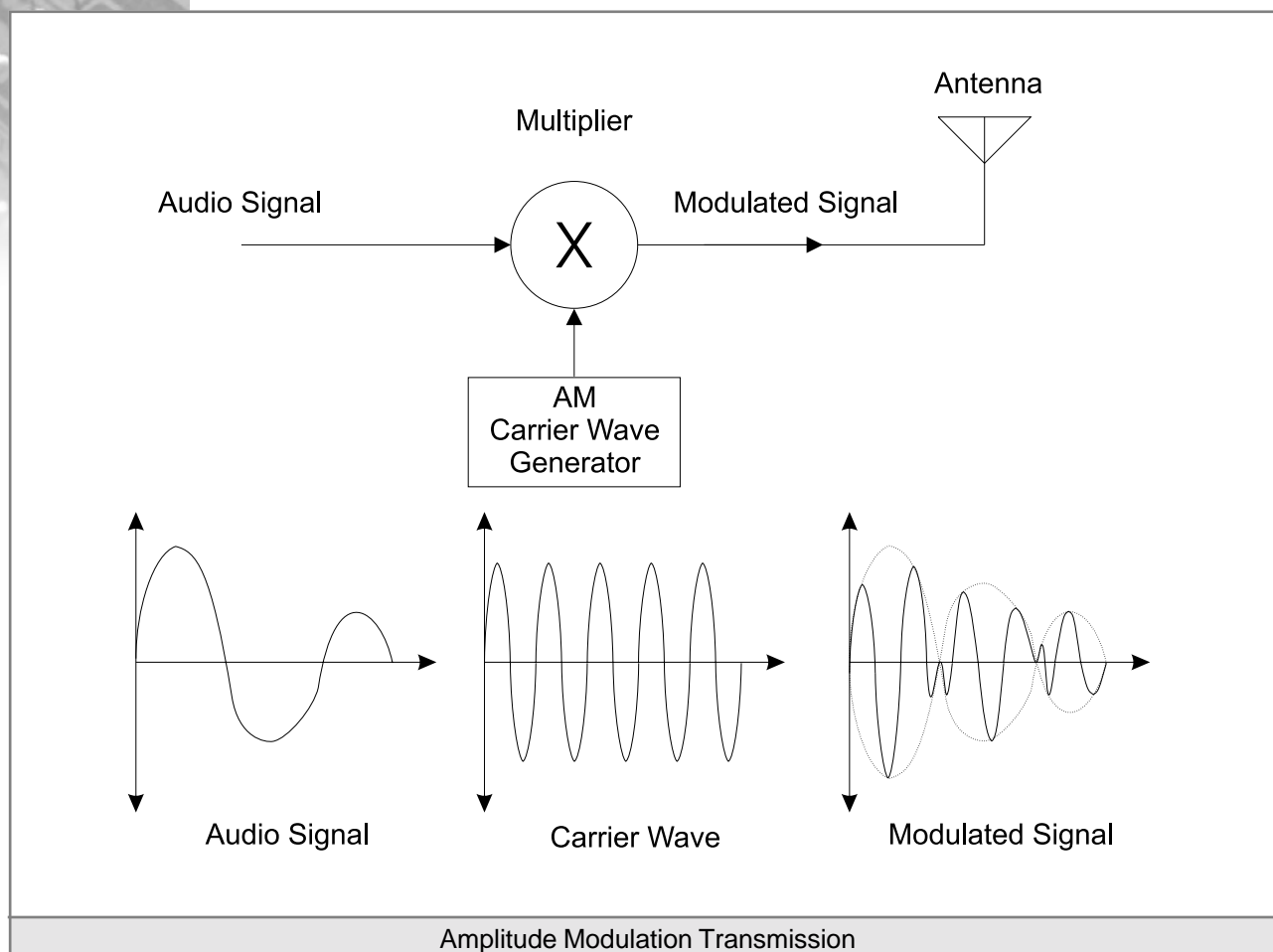
AM is the oldest method of broadcasting radio programmes. Commercial AM stations operate at frequencies spaced 10 kHz apart between 535kHz and 1605kHz. The wavelength of radio signals at these frequencies ranges from 560 to 187 metres. This means that the quarter-wavelength antennas needed to broadcast AM are relatively large.

The advantage of AM is that radio waves in this frequency range can be detected by receivers hundreds of kilometres away. In addition to its use in commercial radio broadcasting, AM is used for long-distance short-wave radio broadcasts and for transmitting the video portion of television programmes.

Frequency Modulation, or FM

In FM, the amplitude of the carrier is kept constant, but its frequency is altered in accordance with variations in the audio signal being sent. This form of modulation was developed by the American electrical engineer Edwin H. Armstrong during the early 1930s in an effort to overcome interference and noise that affect AM radio reception. The diagram below illustrates FM modulation.

The frequency of the modulated wave increases or decreases in response to variations in the audio signal.

Audio Signal   Carrier Wave   Modulated Signal

Frequency Modulation Transmission
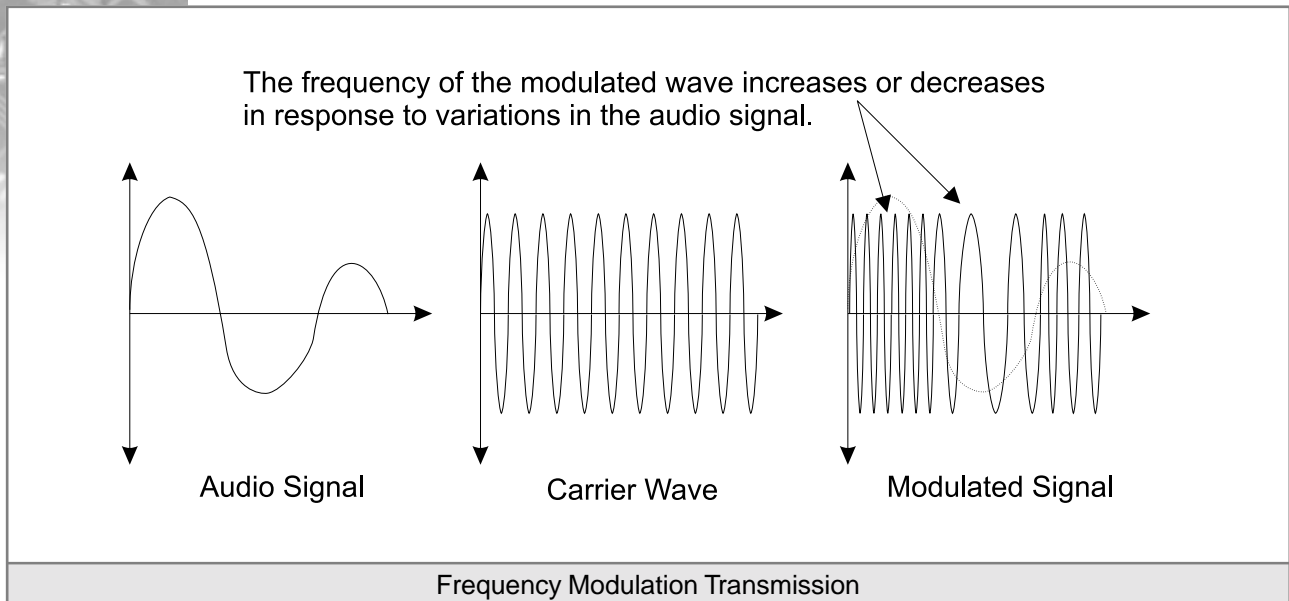
FM is less susceptible than AM to certain kinds of interference, such as that caused by thunderstorms, electrical currents from power lines, machinery and other sources. These noise-producing signals affect the amplitude of a radio wave, but not its frequency, and so an FM signal remains virtually unchanged.

Differences between AM and FM

FM broadcasting stations are assigned higher frequencies than AM stations. These frequencies, spaced 200 kHz apart, range from 88 to 108 MHz. This means that FM antennas are much smaller than AM antennas, typically a metre or so in length. This makes FM antennas much easier to install than AM.

The much larger 200kHz bandwidth assigned to FM means that FM can transmit much more information than AM. This results in FM radio having higher audio quality than AM. FM is capable of accurately transmitting audio signals ranging from 20 Hz to about 16kHz. AM is capable of transmitting audio in the frequency range from a few hundred Hertz to about 5kHz. This is fine for transmitting voice, which is why AM is often used in South African talk radio. However, the limited bandwidth does compromise the quality of most modern music. FM is also better suited to the transmission of stereo sound than AM.

A Under S in the A-Z for more about stereo.

The disadvantage of FM is that the higher frequencies rely on "line of sight" between the transmitting and the receiving antennas.

"Line of sight" means that the receiving antenna has to be able to "see" the transmitting antenna, and the signal path cannot be obstructed by mountains or other physical features, like buildings. This severely limits the reach of FM signals when compared with AM. Even in an area without obstructions, the curvature of the earth limits the reception area of FM to a radius of less than a hundred kilometres around the antenna.

Placing FM antennas on high sites, such as towers on top of buildings, or mountains can increase this distance. The limited reach of FM is why public and commercial broadcasters use more than one FM frequencies for transmission, as it is impossible to cover a large geographical area with a single FM transmitter.

AM signals are able to cover much greater distances. The lower frequencies that are used for AM are effectively reflected back to earth by the earth's atmosphere. This is called the skef wave effect. In addition, AM signals are conducted through the ground, called the ground wave effect. These two effects combine to give AM a much greater reach than FM.

The diagram below shows the differences in coverage area for an antenna broadcasting AM and FM.

Ionosphere

AM signals reflected off the earth's atmosphere.

Line of Sight

Antenna

Earth

Grey shading shows the Line of Sight Reception Area.
It can be seen how a mountain or other physical obstruction such as a building can block FM line of sight transmission.

The propagation of the sky wave and the limitations of line of sight from an antenna
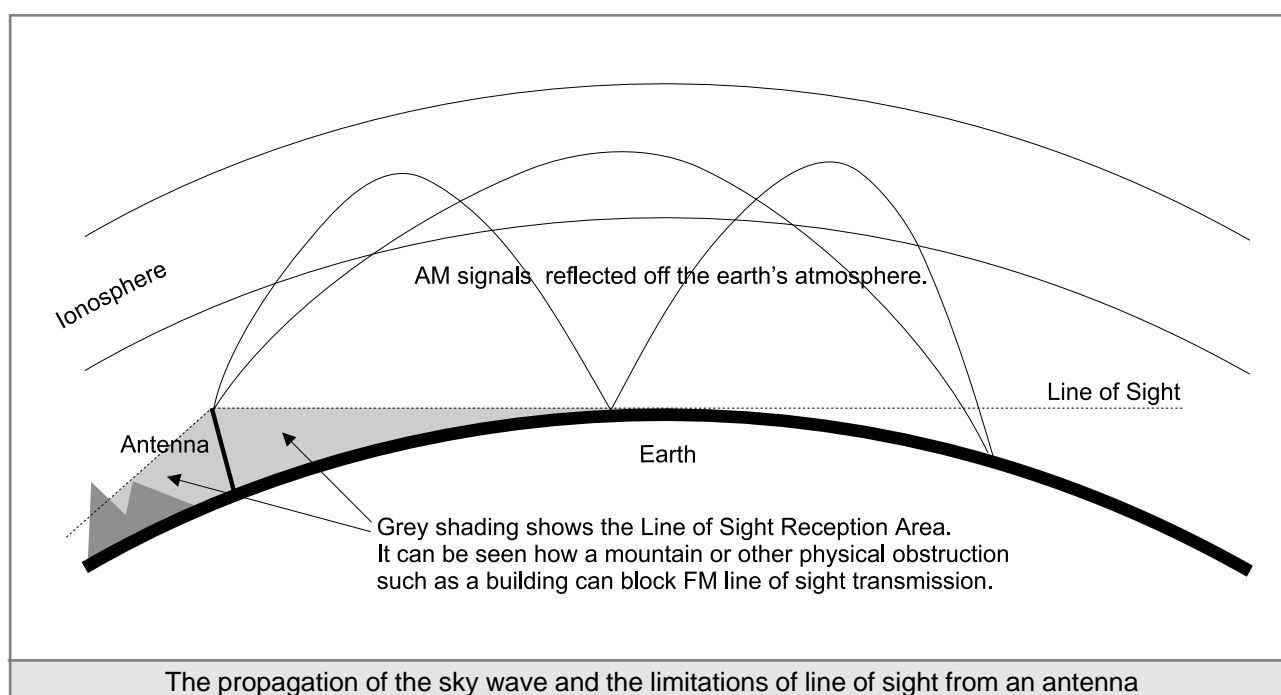
Demodulation

The radio receiver is used to reverse the process of modulation. The reverse process is called demodulation. This allows the original modulating wave (the audio signal) to be retrieved from the modulated radio wave.

Patchbay / Jackfield

A patchbay, also called a jackfield, is one of the most useful items in a studio.

Inputs and outputs that you regularly use are wired to a patchbay so that they can be conveniently patched together with short signal leads. There are a variety of patchbays available in different sizes that use different types of connectors.

The patchbay most often found in South African community radio stations uses GPO jacks and plugs. This patchbay normally consists of a 1U (one unit) panel with 2 rows of 26 sockets, with each socket on the top row paired with the socket below it. The top row of sockets is for outputs from source equipment; the bottom row is for inputs to the mixer or other recording or transmission equipment.

Patchbays are most commonly used to gain access to mixer inputs, auxiliary sends and returns, equipment inputs and outputs and insert points. However not all patchbays are wired in the same way.

We recommend that you read the section on Mixers in the A-Z before reading more about patchbays or jackfields.

The diagram below illustrates a patchbay that is typical of the kind found in a community radio studio. However, for convenience, we've only drawn 8 inputs and outputs. As mentioned, a real patchbay could have many more (a typical patchbay would have 26 sockets). The appearance and abbreviations used for the labelling of the diagram are typical of labelling you will find on a real patchbay, so take care to study the labelling style.

CD1L	CD1R	MD L	MD R	PGM L	PGM R	COMP L	COMP R
CH 1 L	CH 1 R	CH 2 L	CH 2 R	COMP L	COMP R	TX L	TX R
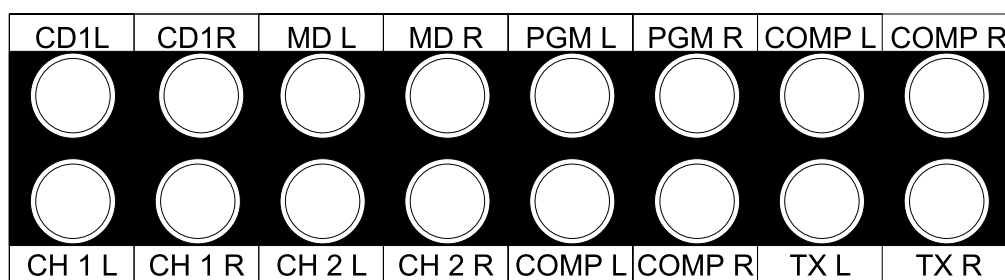
Patchbay layout with 8 Inputs and Outputs

Outputs

The top row of jacks are outputs, and are labelled as follows:

CD1 L and CD1 R: These are the Left and Right outputs from CD Player Number 1.

MD L and MD R: These are the Left and Right outputs from a MD Player.

PGM L and PGM R: These are the mixer's Left and Right programme outputs.

COMP L and COMP R: These are the Left and Right outputs from the compressor.

Inputs

The bottom row of jacks are inputs, and are labelled as follows:

CH 1 L and CH 1 R: These connect to the Left and Right inputs of Channel 1 on the mixer.

CH 2 L and CH 2 R: These connect to the Left and Right inputs of Channel 2 on the mixer.

COMP L and COMP R: These connect to the Left and Right inputs of the compressor.

TX L and TX R: These connect to the Left and Right inputs of the transmitter.

How the Patchbay Works in a Broadcast Studio

Imagine that this patchbay is in use in the station's broadcast studio. Why is it useful to have these jacks? Let's work by example:
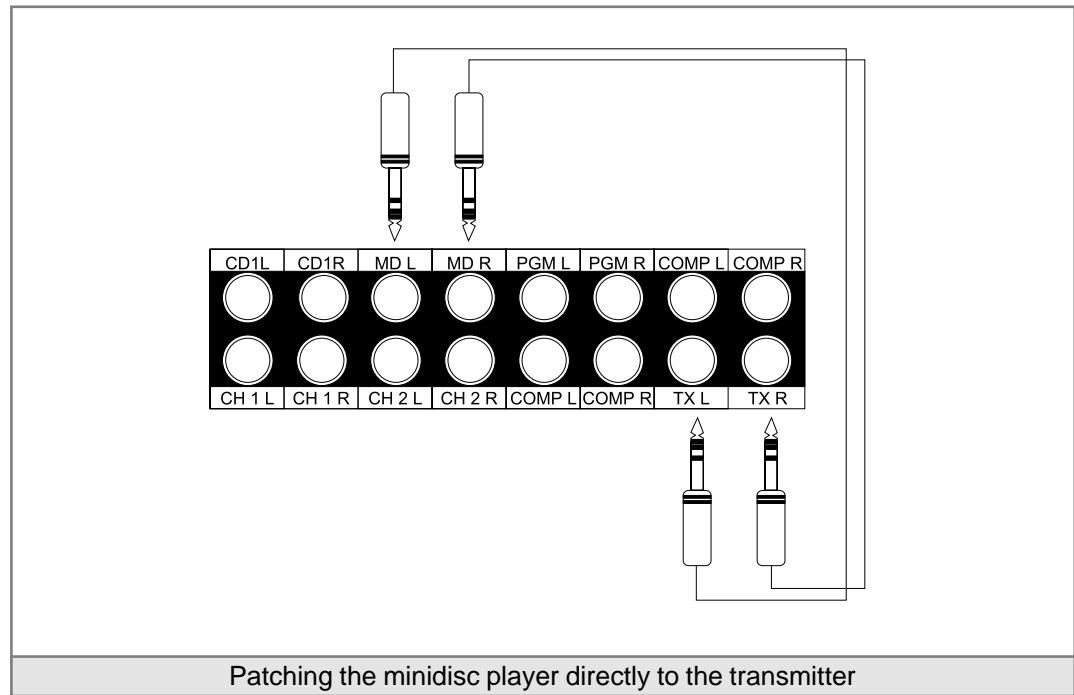
Example 1:

Assume that you need to switch off the broadcast mixing desk for cleaning and maintenance. This could mean taking your station off air while the mixer is switched off. However with a patchbay you can avoid this.

There are several options for staying on air while the mixer is switched off. The outputs of CD1 and the MD player are available on the patch panel, as are the inputs of the compressor and the transmitter.

Look at the diagram below. By using patch cords, you connect MD L and MD R to TX L and TX R. By doing this, you are bypassing the mixer. So while you switch off the mixer to complete your repairs, your station is still on air, because the output of the MD player is connected directly to the input of the transmitter.

If the compressor limiter normally processes the signal before transmission, you could have patched the MD to the compressor input. This would be a better solution, as the compressor will control the level from the MD player.

Patching the minidisc player directly to the transmitter

Example 2:

Assume that the station is doing an outside broadcast (OB). We have a stereo input coming into the studio from Telkom, but have no spare channels on the mixing desk to plug it into. It will be difficult and time-consuming to change the connections on the mixer, but the patch panel provides a simple solution.

The diagram below shows how you can patch the OB lines into the mixer without having to change any of the studio connections.

Patching OB lines to Channel 1 of the mixing desk
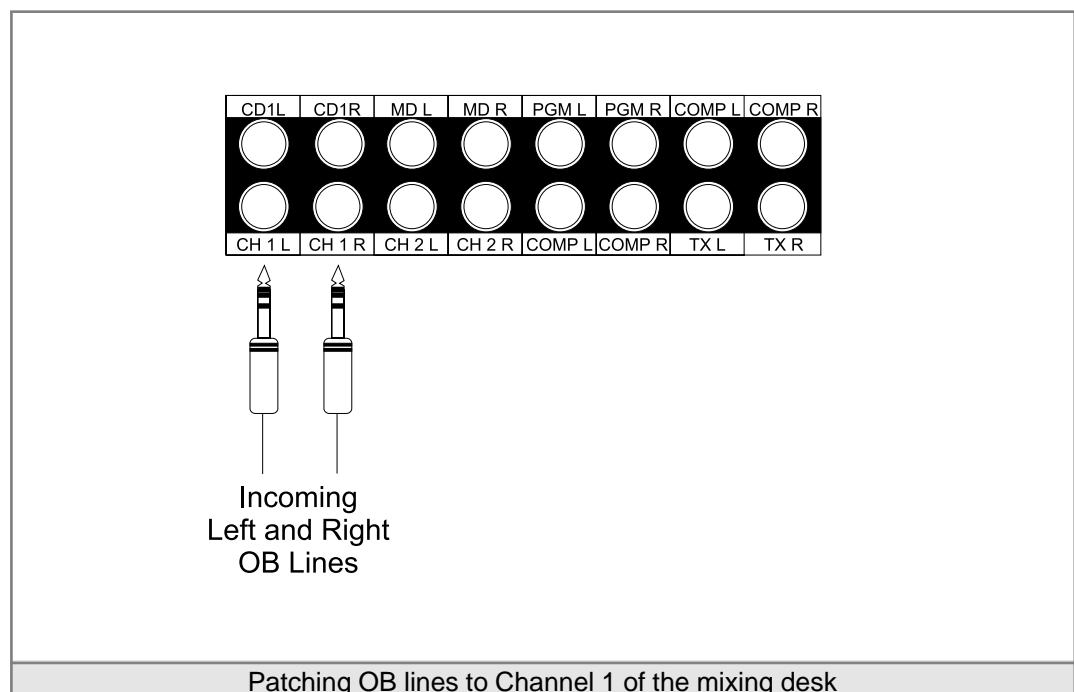
Connecting the OB Lines to the Channel 1 input on the patchbay brings the OB signal to Channel 1 on the mixing desk. The OB can then be played through the mixing desk, and the level can be controlled with the Channel 1 fader.

You might be wondering what happened to the equipment that was connected to Channel 1 before we plugged in the OB line. Patchbays are normally built in such a way that if you put a plug into the jack, the existing connection is broken, and our new connection "overplugs" the existing connection. When the plug is taken out the old connection is restored.

In our example, CD1 is normally connected to Channel 1 of the mixer. Plugging in the OB line disconnects CD1 from the mixer in favour of the OB line. Taking the plugs from the OB out will restore the connection from CD1 to the mixer.

These two examples demonstrate the versatility that a patchbay brings to the studio. Any variety of signals can be re-routed by using the patch cords.

However, note that a patch cord must always run from an output to an input. This means that the opposite ends of a patch cord must be plugged into different rows on the patchbay. This is very important – connecting two outputs together can damage the output circuits of both pieces of equipment involved.

Using the Patchbay to Find Faults

You can use patchbays to find faults by working through a problem systematically. For example, if there is no sound coming from the CD channel on your mixing desk, you could use the patchbay to find out whether the problem lies with the CD player or with the mixing desk before taking equipment out of its racks or altering cabling.

If both the mixer and the CD player are connected to the patchbay, as shown in the example above, you can try to find the fault as follows:

Check the output of the CD player by plugging a pair of headphones into the outputs labelled CD1 L and CD1 R. Then,

• If there is no sound coming through the headphones, the fault lies between the CD player and the patchbay. So check the CD player and its cabling.

• If there is sound, then the fault lies between the patchbay and the mixer. So you will check the CD channel on the mixer and the cables from the patchbay to the mixer.

This method of using the patchbay to find faults can save a lot of time and trouble.

Reel-to-Reel Recorder

The reel-to-reel recorder has been standard equipment for recording and editing in radio studios for many decades. However, reel-to-reels are being fast replaced by computer-based recording and editing systems. The lower cost and greater capabilities of computer-based systems have rendered reel-to-reel obsolete in many areas of radio production. Since the mid-90s, most new stations have no longer invested in reel-to-reel machines and their use is becoming limited to stations that have an archive of reel-to-reel tape that they need to continue playing. Reel-to-reel is similar to cassette tape in that it uses magnetic tape to store audio information.

Under C in the A-Z for more on cassettes and cassette players and how they store audio on magnetic tape.

Tascam BR-20 Reel-to-Reel Recorder often used in SA community radio stations

As the picture of the Tascam BR-20 Reel-to-Reel Recorder shows, the tape is not housed inside a cassette or cartridge. Reel-to- reel is a medium that depends on two separate reels of tape.

Another difference is that the tape is much wider. Reel-to-reel tape is available in a number of widths used for different purposes. The tape used in radio production is a quarter of an inch (written as ¼") wide, and records two tracks of audio. This is why reel-to-reel recorders are often referred to as 2-track analog recorders. The two tracks are used for the left and right channels of a stereo recording.

The ¼" (6.35mm) tape is much wider than the 4mm wide tape used in cassettes. The use of wider tape by the reel-to-reel recorder provides better audio performance than cassette tape.

Under S in the A-Z for more on stereo.

Length and speed of reel to reel tape

Reel-to-reel tape is sold by length. Common lengths are 600, 1200 and 2500 feet. Depending on the length of the tape it is supplied on different sized reels. There are 5", 7" and 10" reels. The recorder in the picture is fitted with 10" reels.

Reel-to-reel recorders can record at a number of different speeds. The higher the speed of the recording, the better the recording quality. However the higher the speed of the recording, the more tape is needed.

Recording speed is specified in inches per second, that is, how many inches of tape are needed for each second of the recording. The most common recording speeds are 15 inches per second (ips) or 38cm per second, and 7.5ips or 19cm per second. If you are using 1200 feet of tape, and recording at 15ips you can work out how long your recording can be as follows:

There are 12 inches in a foot. This means a 1200-foot tape is 14400 inches long. (1200x12 = 14400)

The recorder uses 15 inches of tape for every second of recording time. So we can work out the number of seconds of recording time by dividing the length of the tape by 15.

$14400 \div 15 = 960$

So when recording at 15ips, 1200 feet of tape will record 960 seconds, that is 16 minutes of audio. The slower the recording speed, the more recording time is available on a length of tape. The longest recording that is possible using a single reel of tape on a recorder like the Tascam BR-20 shown in the picture, is just over 60 minutes.

Operation of the Recorder

Much of the content of the following sections on reel-to-reel is drawn from an excellent tutorial on analog recorders, by Peter Elsea of the Electronic Music Studio at the University of California, Santa Cruz. *The Analog Recorder* and several other essays are available at http://arts.ucsc.edu/ems.

The diagram below shows the parts that make up a typical reel-to-reel recorder.

Parts of a typical Reel-to-reel recorder
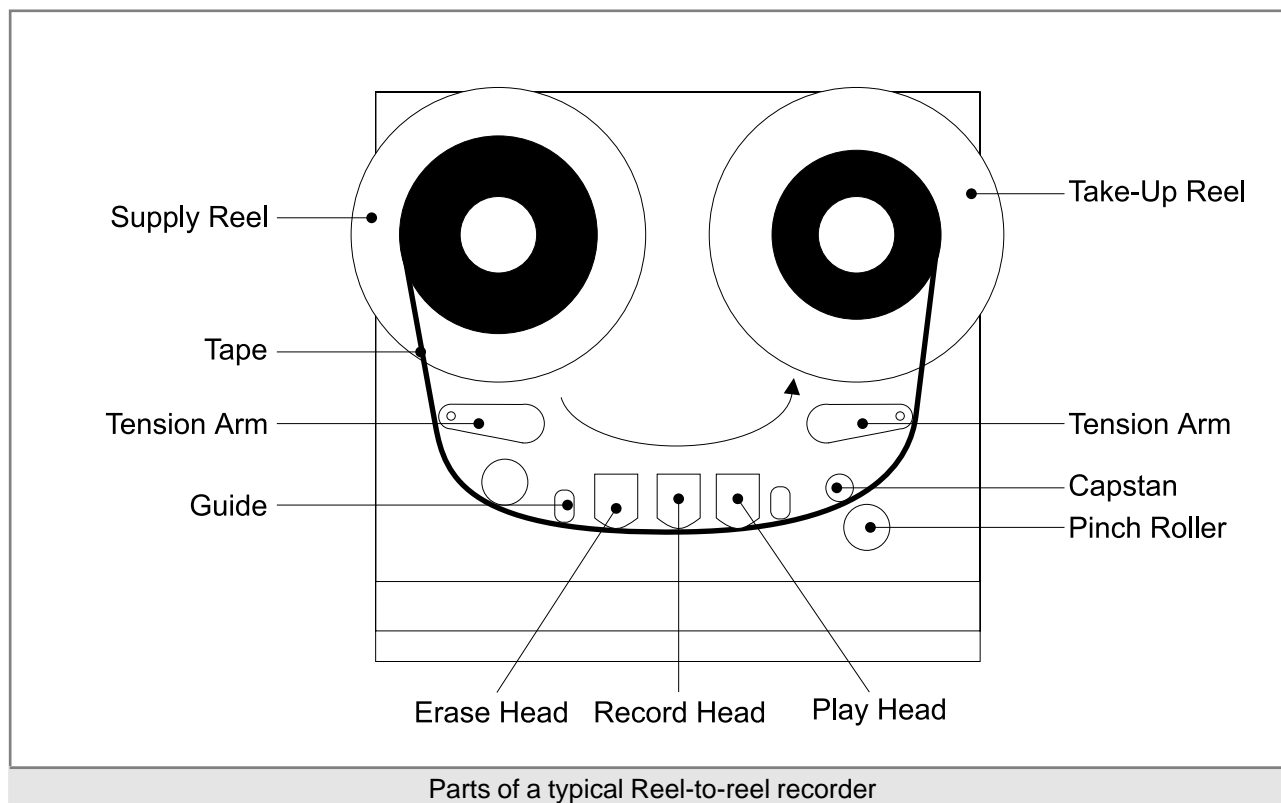
The tape is threaded from the left to the right, as shown by the arrow in the diagram. The tape starts from the supply reel. It runs past one or more guides and a tension arm, past the erase, record and play heads, between the capstan and the pinch roller, past more guides and arms, and finally it winds onto the take-up reel.

The capstan controls the movement of the tape past the heads. The capstan is a steel shaft that is always turning at a constant speed. When PLAY is pushed, the pinch roller squeezes the tape against the capstan, and the tape is pulled past the heads. The take-up reel is turned to take up the tape as this goes on.

The take-up reel and the supply reel each have motors that turn during playback. During rewinding or fast-forwarding they turn at high speed to move the tape from one reel to another. There are brakes on both reel motors to stop the tape quickly and gently. When rewinding or fast-forwarding the recorder lifts the tape from the heads to prevent wear and tear of the tape.

Controls

The controls of the reel-to-reel recorder are very similar to the controls on a cassette deck. The PLAY, REWIND, FAST FORWARD and STOP controls are used to control the movement of the tape. The recorder will have a RECORD button. Usually RECORD and PLAY must be pressed together to start a recording.

Typically, the recorder will have a monitor selector. This allows the choice of listening to the input signal, or the signal on the tape. The input signal is the signal being fed to the recorder, that is, the signal that you want to record. The recording will have a speed control to select the recording speed and also a pitch control allowing for fine speed adjustments. Like any other professional recording equipment, the reel-to-reel recorder will have a set of level meters. These are usually VU meters that can be used to monitor the recording and playback levels.

Under M in the A-Z for more on meters.

The recorder will have a tape counter to keep track of your location on the tape. Professional recorders also have "locate" features linked to the counter that can remember positions on the tape for you.

Cleaning Heads

You cannot expect to make good recordings if the recorder's heads are dirty. It is good practice to clean the heads of the reel-to-reel before any recording session.

Heads can be cleaned with alcohol or surgical spirits on a cotton wool swab or ear bud. This is easily done: just dip the ear bud in the alcohol and scrub each head. If there is anything on the head that will not come off with this method, find a technician. If you notice that the capstan or tape guides are dirty, you may clean these as well. Do not clean anything else.

Recording Tips

Always label your tapes carefully and thoroughly. Proper labelling reduces the number of times a tape must be played, because you won't be playing it over and over to find out what's on it. The less a tape is played the longer it will last. Labelling should include:

• your name

• name of tape

• date recorded

• speed and format

• heads or tails

Heads or tails refers to whether your tape is in the rewound and ready-to-play state, which is heads (a supply reel) or if it is in the just-played state, which is tails (a take-up reel).

To better preserve your tape it is recommended that you store your reels tail out. (That means, store them in the just played rather than the just rewound state.)

Editing on Reel

The reason the reel-to-reel became so popular for radio production, is because it is possible to edit recordings very precisely on reel tape.

Reel-to-reel tape can be edited by simply cutting the tape. For example, if you recorded an interview on reel, and your guest repeated herself during an answer, you could locate the repetition on the tape, mark where it starts and where it stops on the tape and use a blade to cut out that section of tape. The two ends of your tape could then be joined back together using some splicing tape to make the interview clearer and shorter for better listening.

In the same way, if you wanted to add something to a recording, you could record it on another piece of tape and splice it into the recording. Splicing means cutting the original recording, and inserting the new section, sticking it to the original using splicing tape.

Editing Modes

The recorder will have one or more edit modes that you use to find the spot on the tape where you want to edit. There are different names for these, but the most common are:

- STOP EDIT: This allows you to hear the tape as you move it by hand so you can find specific sounds.

- DUMP EDIT: This is a play mode in which the take-up reel doesn't turn. This allows large sections of tape to be run off into a wastebasket.

- CUE: Allows you to hear the tape in the fast modes.

Making the Edit

To find the edit point, enter edit mode, put a hand on each reel, and move the tape back and forth over the heads to find the beginning or end of the sound you are looking for. The faster you turn the reels, the more natural the tape sounds, so the best way to move the tape is in quick, short jerks. You will soon find that it is easier to find the beginnings of sounds than the ends.

Once you have found the sound, mark the tape with a dot over the play head, using a grease pencil. Now that you have marked the edit point, you are ready to cut or splice the tape.

Splicing the Tape

Before splicing the tape, make sure your hands are clean and free of oil. Your recorder will have a splicing block. This block has a groove for resting the tape while you cut it. To complete the edit, place the tape in the splicing block black side up (the black side of the tape, with the manufacturer's name should be pointing up).

Line up the mark you made on the tape with the diagonal notch in the splicing black. The tape will stay put because of the way the block is grooved. Using the notch as a guide, take a razor blade and cut the tape with a quick motion towards yourself. Pull the separate pieces of tape out of the ends of the block and repeat the process at the other splice point.

Be sure to use non-magnetic industrial razor blades, otherwise you will pick up clicks and thumps on your tape.

Place the two pieces of tape you wish to splice in the block so that the ends touch but do not overlap. You now need to apply the splicing tape.

It is best to cut a number of 2cm long "tabs" of splicing tape before you start editing, rather than to make them one at a time as you are going along.

Use the point of the razor blade to pick up a tab of splicing tape and put it in the block across the splice. Rub the back of the splicing tape until it sticks. Remove the spliced tape from the block by pulling upwards smartly at both ends. The tape should snap out without wrinkling.

Set the splice (join) on the raised portion of the block and rub on the back of the splicing tape until all of the bubbles are gone. The diagram below shows how the completed splice should look:

Bottom

Top

Side

Correct Splice
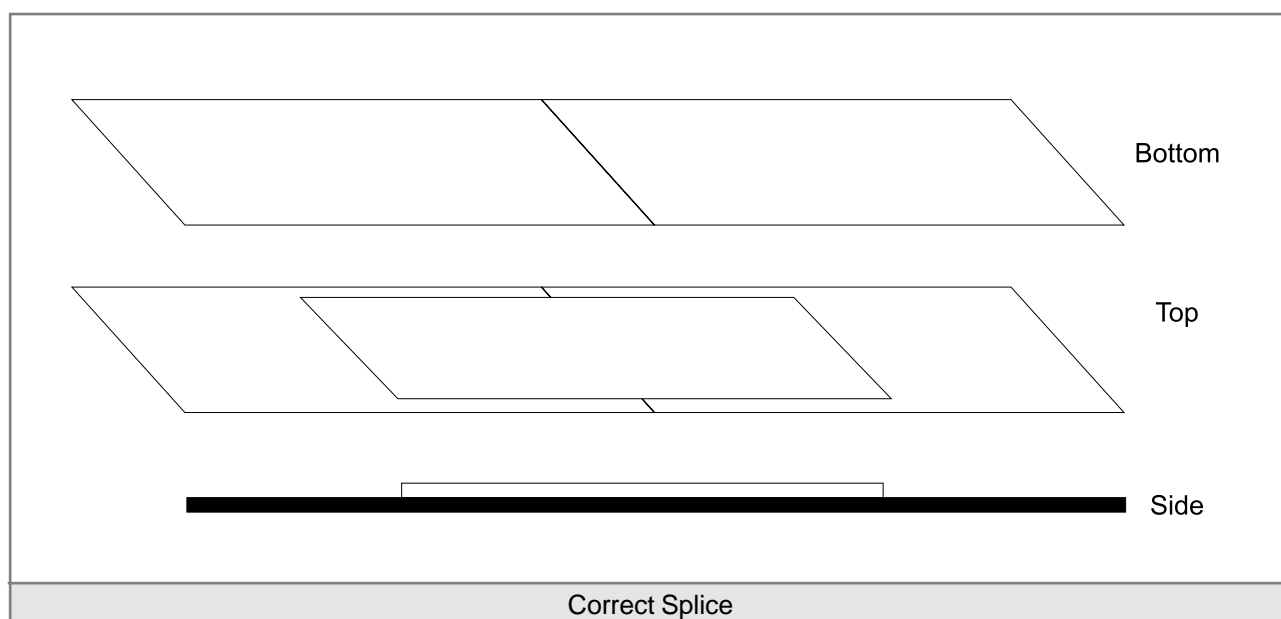
Once this is complete, examine the splice carefully for any of the following mistakes.

Bottom

Top

Side

Splice Gap
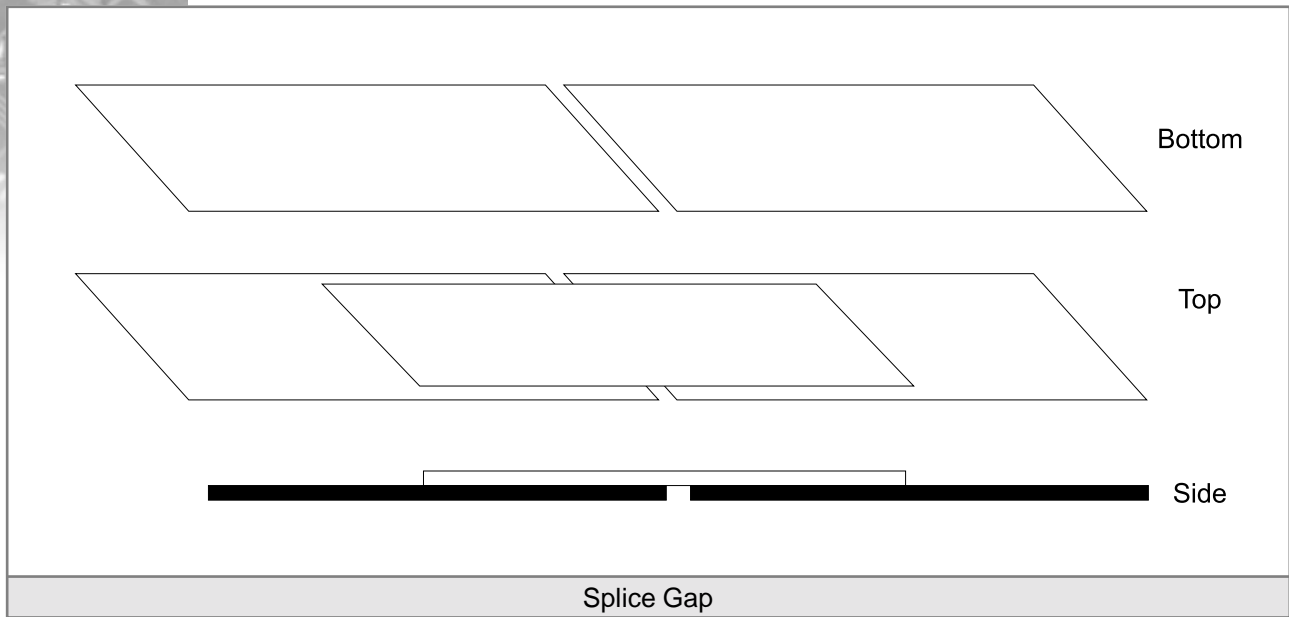
The diagram above shows a splice with a gap between the tape ends. The exposed splicing tape can cause the tape to get stuck in the reel. This is usually caused by a difference in the way the two pieces of tape are cut. To avoid this problem, use the same diagonal of the splicing block for both sides of your splice.

Bottom

Top

Side
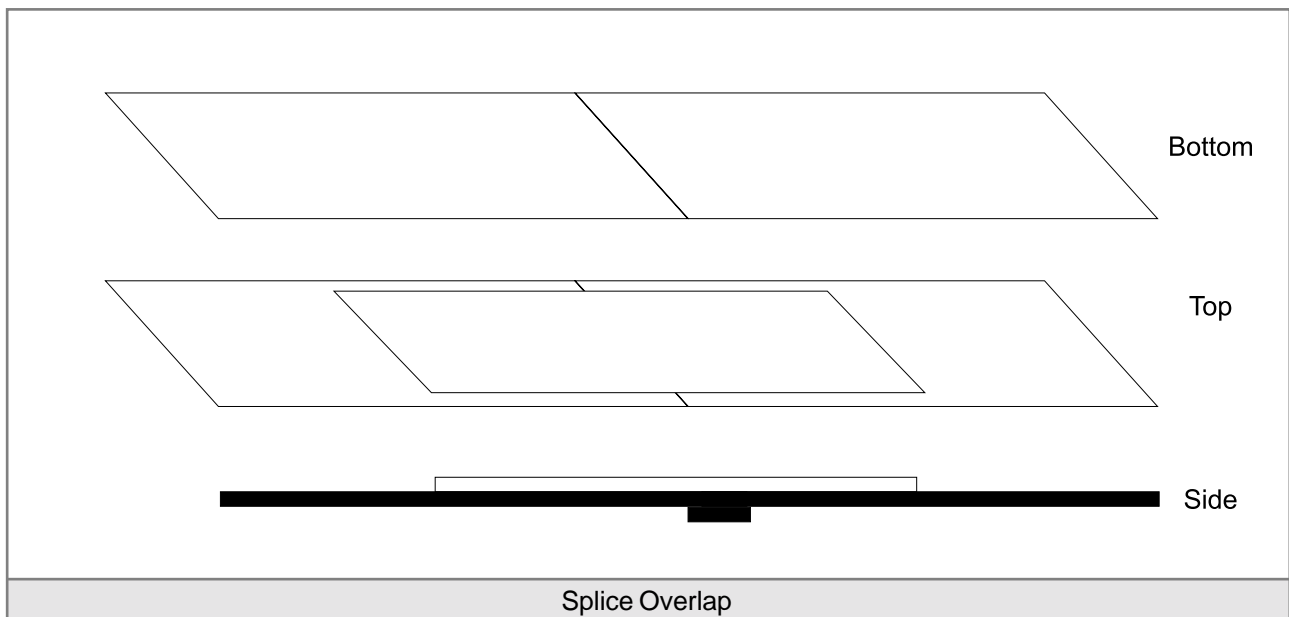
Splice Overlap

The diagram above shows a splice where the pieces of tape are overlapped. This will cause a thump as the tape is played.

Bottom

Top

Side

Splice Overhang
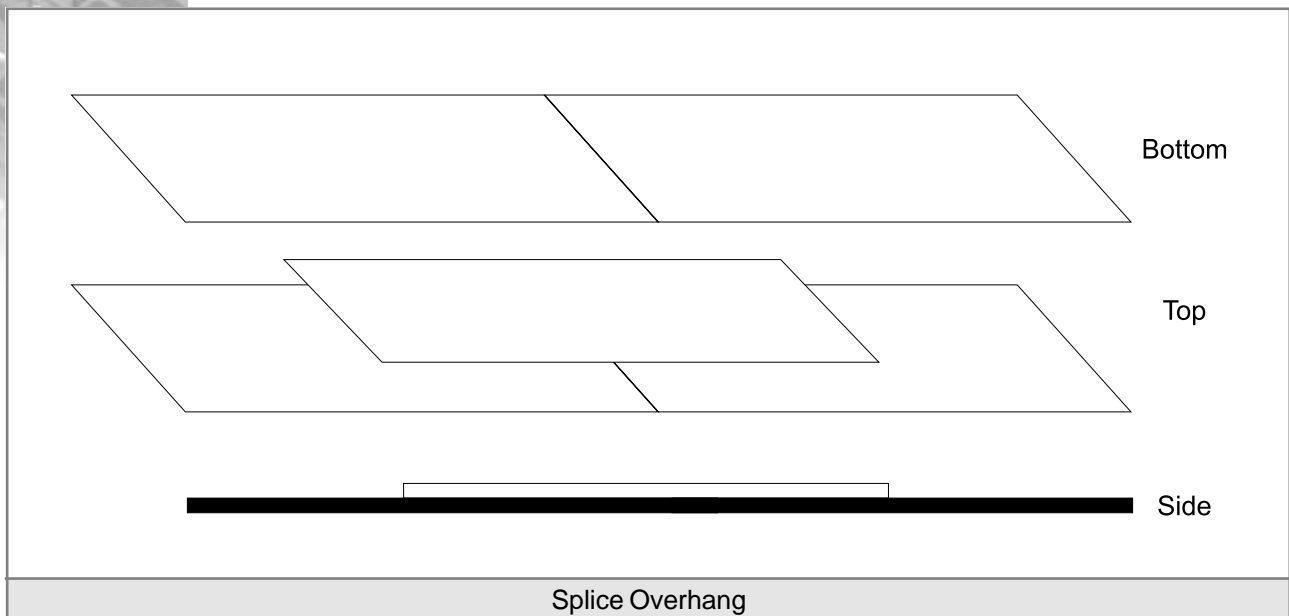
The diagram above shows a splice where the splicing tape is hanging over the edge of the tape. This will cause the tape to get stuck in the reel or to stick to the tape guides.

Bottom

Top

Side
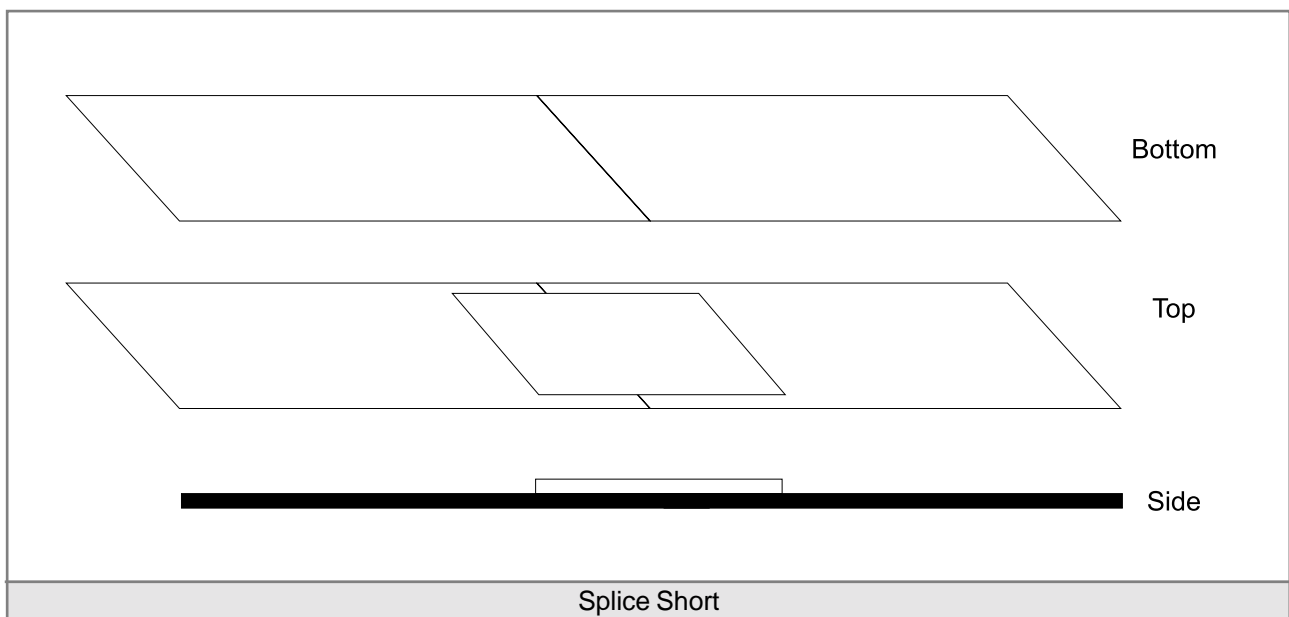
Splice Short

The diagram above shows a splice that is too short. This type of splice will not last very long, causing the tape to break.

	Bottom
	Top
	Side
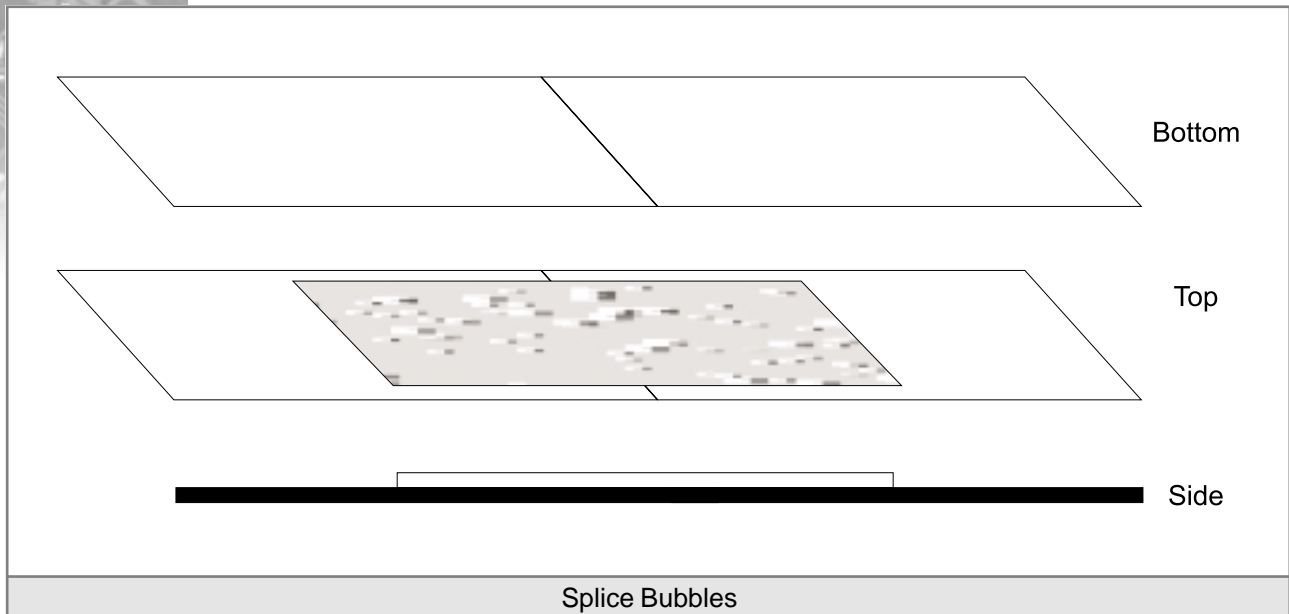
Splice Bubbles

The diagram above shows a splice where the splicing tape is still full of air bubbles. This happens when the spicing tape is not rubbed down after being applied. This type of splice will not last very long.

The splice will also not last long if you handle the tape with greasy fingers. If you make the piece of splicing tape too long, the tape may "thump" as it passes the heads.

To take a splice apart, bend it backward right at the join and peel the recording tape off the splicing tape.

Connections to the Reel-to-Reel Recorder

The reel-to-reel recorder will have at least one balanced pair of outputs that can be connected to your studio mixing desk. The recorder may have one or more pairs of inputs. Often recorders have microphone inputs in addition to a pair of balanced inputs that can be fed from the studio mixing desk or distribution amplifier. A professional recorder should also have connections that allow for fader start.

Sound and Audio

In this manual, we have used the term sound to refer to sound waves that travel through air, and the term audio to refer to electrical signals that travel through cables.

Sound

Sound is the effect perceived by the brain when vibrations travelling through air stimulate the ears. These vibrations are acoustic energy. Acoustic energy consists of varying waves of pressure that travel through a physical medium – usually air.

You cannot see these acoustic or sound pressure waves, but you can visualise them. Imagine the waves that are created when a stone is dropped into a pool of water. The movement of the waves away from where the stone hit the water is similar to the movement of sound pressure waves from a sound source.

Anything that vibrates that is in contact with the air can create sound pressure waves. This could be hands clapping, a loudspeaker cone, or a person's vocal chords. These vibrations disturb air molecules causing waves of alternating high and low pressure. The ear is a sensitive transducer that responds to the changes in pressure.

The shape of the outer ear directs sound pressure waves into the aural channel. The eardrum is a stretched drum-like membrane at the end of the aural channel. The eardrum vibrates in response to the sound waves. These vibrations are passed on to three tiny bones connected to the eardrum. These three bones act as an amplifier by amplifying the eardrum's vibration many times over, and as a limiting protection device.

The bones transfer the vibration to the inner ear. The inner ear has two fluid-filled chambers that are lined with very fine hairs. Vibrations transmitted to these hairs are perceived as hearing. Hearing loss occurs when these hairs are damaged, often by too much exposure to very loud sounds, and they can deteriorate with age.

Take care of your ears – sound engineers who damage their ears will have very short careers!

Sounds that we hear as being louder (higher amplitude) are caused by greater changes in air pressure. Quieter sounds (lower amplitude) are caused by lesser changes in air pressure.

In order to be classified, as sound, air pressure waves must change at a rate of between 20 and 20 000 complete cycles per second. The number of complete cycles that are completed in a second is called the frequency of the sound wave. Frequency corresponds closely to the musical attribute of pitch. The higher the frequency, the higher the perceived pitch of the sound.

The unit of frequency is the Hertz (Hz). The normal human ear responds to frequencies between 20Hz and 20 000Hz. However, the ability to hear higher

frequencies decreases with age, and most people begin to loose about 1Hz of range each day from the start of their teens. The amount of time needed for one complete cycle of a sound wave is called the period of the wave.

Sound waves travel through air at a speed of 344 metres per second (1238.4km/h). This is measured at sea level at a temperature of 15 ºC. The speed of sound is independent of frequency. The physical distance that is travelled by one complete cycle of a given sound frequency as it passes through air is called the wavelength. The wavelength is expressed by the equation:

$$Wavelength = \frac{Speed\ of\ sound}{Frequency}$$

All of these quantities are illustrated in the diagram below.

Representation of a sound wave: A complete cycle of a sine wave
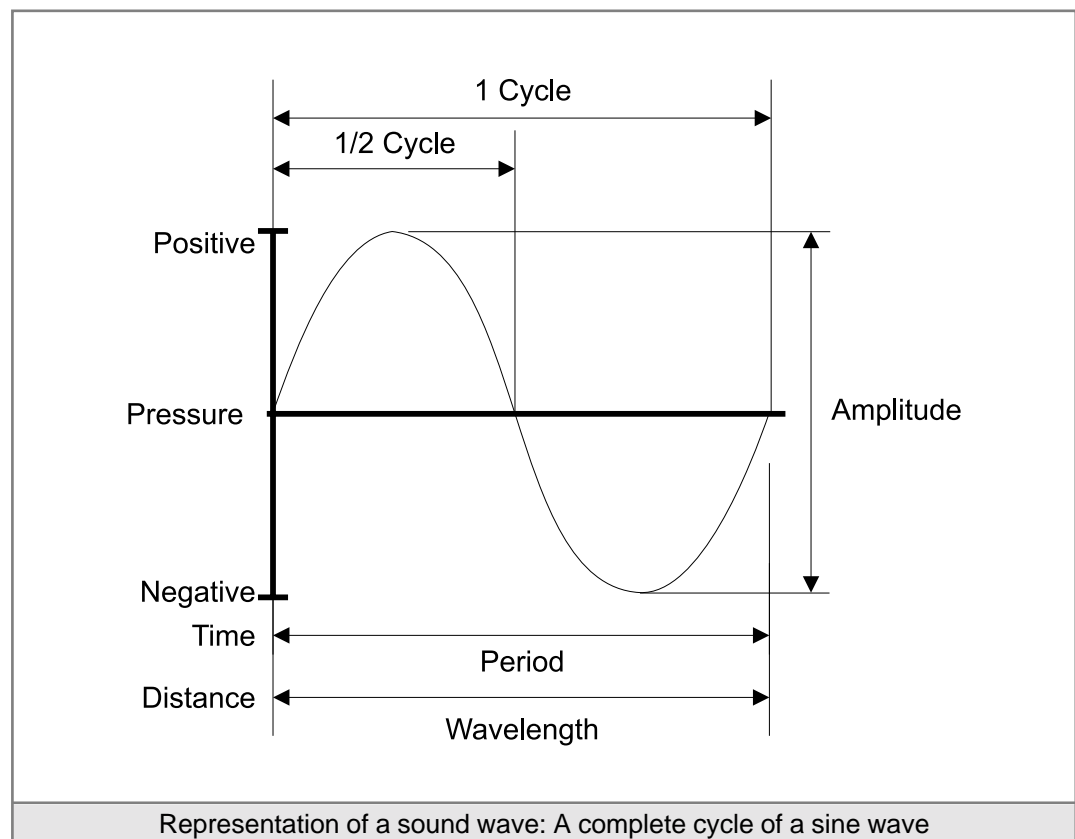
Audio

An audio signal is an electrical representation of a sound wave, in the form of a varying voltage or current. Within the limits of the audio equipment, the audio signal voltage or current will vary in frequency, and proportionally in amplitude, in exactly the same way as the sound wave it represents. The amplitude of an audio signal is called the signal level. Level is normally specified in decibels.

Under D in the A-Z for more about decibels.

Converting Sound Into Audio

To convert sound to audio we use a transducer. A transducer is a device that converts energy from one form to another. In radio, we deal with electro-acoustic transducers. These are devices that convert acoustic energy (sound) into an electrical signal (audio), or vice versa. The most common input transducers encountered in the radio studio are as follows:

Air Pressure Microphones – convert sound waves in air to an audio signal in the microphone cable.

Under M in the A-Z for more about microphones.

Tape Heads in cassette recorders, which convert the varying magnetic fields imprinted on magnetic recording tape into an audio signal.

Under C in the A-Z for more about cassette recorders and tape heads.

Phonograph (turntable) Pick-Ups – convert the physical movement of the stylus ("needle") within the grooves of a record into an audio signal.

Laser Pick-Ups – convert the imprinted pattern of pits on a CD into a digital data stream. This stream is then translated by a digital-to-analog converter (DAC) into an analog audio signal.

Under A in the A-Z for more about analog and digital audio.

Magneto-Optical Pick-Up – the minidisc uses a combination of magnetism and laser light to write digital information onto a MD. The pick-up converts changes in magnetic fields to a digital stream. This is translated by a DAC into an analog audio signal.

Converting Audio Back Into Sound

Output transducers convert audio signals back into sound. There really is only one type of output transducer used in the studio, and that is the loudspeaker. The loudspeaker converts an amplified audio signal into vibrations of a speaker cone that generate sound waves.

Under L in the A-Z for more about loudspeakers.

Any electronic device that processes audio signals is called an audio frequency device or audio device. Most of the equipment in the radio studio falls into this category.

STL (Studio - to - Transmitter Link)

All stations need to feed the audio signal from the broadcast studio to the transmission equipment. However studios and transmission sites are most often not located in the same place, and this is why you need an STL.

It is desirable to locate a community radio studio in a highly visible and accessible place, somewhere like a shopping mall or community centre, or where people live, so that members of the community can see what is happening and take part in the station's activities.

However, transmission towers tend to be placed on the highest point in the broadcast area, somewhere like the top of a mountain. The higher up the FM transmitting antenna is situated, the larger the station's reception area, or "footprint", will be. AM transmission towers are also generally not located near the station. Because the transmission tower and the studios are not in the same place, an STL is used to link the studios to the transmitter.

Different Types of STL

There are a variety of STLs, all with their own advantages and disadvantages.

Cable

The simplest STL is a cable between the studio and the transmission equipment. If the station's transmission site is located within a few hundred metres of the studio, very high quality audio cable can be used to connect the two. But often the two sites are too far apart to connect them with a cable.

Advantages of this system are that it is simple and cheap. But placing the transmitter close the studio means you may not get the best coverage.

Wireless STLs

A wireless STL uses a low power transmitter at the station, and a receiver at the transmission site to create a radio link between the two sites. The signal from the studio is fed to the STL transmitter, which beams the signal to the transmission site. Here the STL receiver picks up the signal and feeds it to the broadcast transmitter.

Wireless STLs use much higher frequencies than broadcast transmitters. These frequencies require "line of sight" between the STL's transmitting antenna at the station and the receiving antenna at the transmission site. "Line of sight" means that mountains or other physical features, like buildings, cannot obstruct the signal between the two antennas. This often means that the STL's transmitting antenna must have its own tower, or be mounted on the station's roof, to prevent surrounding buildings from obscuring the signal.

Wired STLs

If the distance between the studio and the transmitter is more than a few hundred metres, and a wireless STL is not suitable, you will have to use a wired STL.

In South Africa, community radio stations that use wired STLs lease the wire to the transmission site from the telephone company, Telkom. The extent of the telephone network makes it possible for Telkom to carry signals across great distances, something also used for outside broadcasts. Stations typically lease a permanent "high quality" analog or a digital connection from the station to the transmission site.

An STL is used when the studios and the transmission tower are not co-located. The STLfeeds the sudio audio to the tranmission point.

Transmission Tower

Wireless STL

TEK FM

Radio Station

Wired STL

Illustration of different types of Studio Transmitter Link (STL)
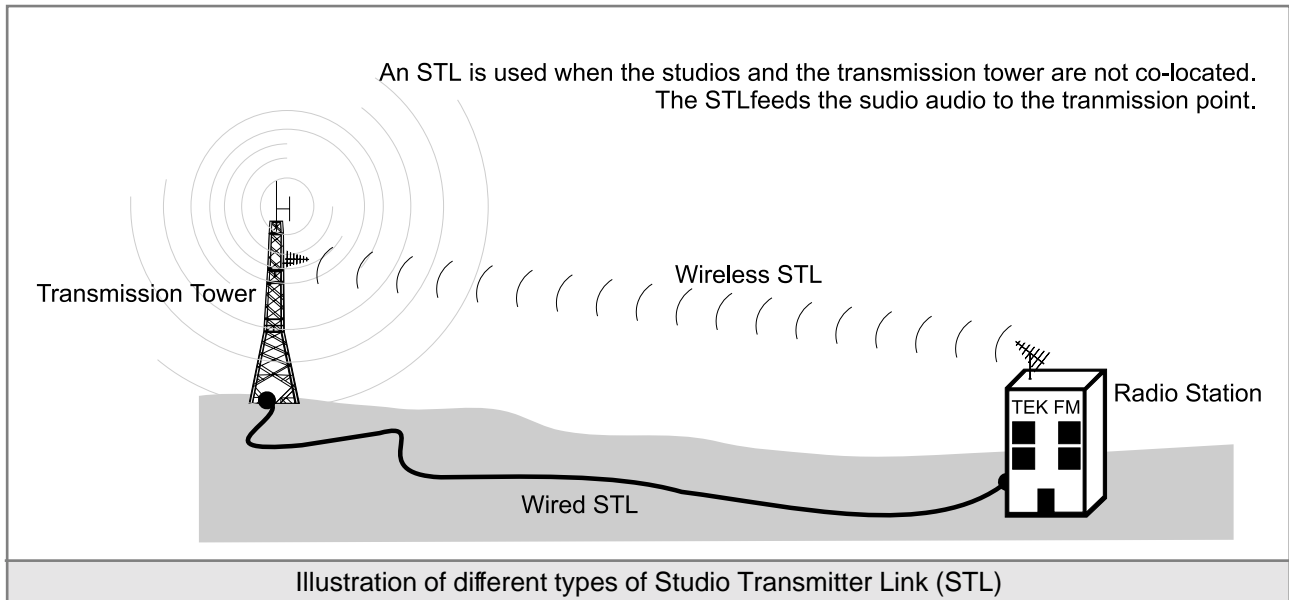
Your choice of system will depend on the station. Buying a wireless STL is often much cheaper. The initial outlay of buying the STL equipment is large, but once this has been paid, there are no monthly fees. When leasing an STL from Telkom or Sentech, you will be charged a monthly fee.
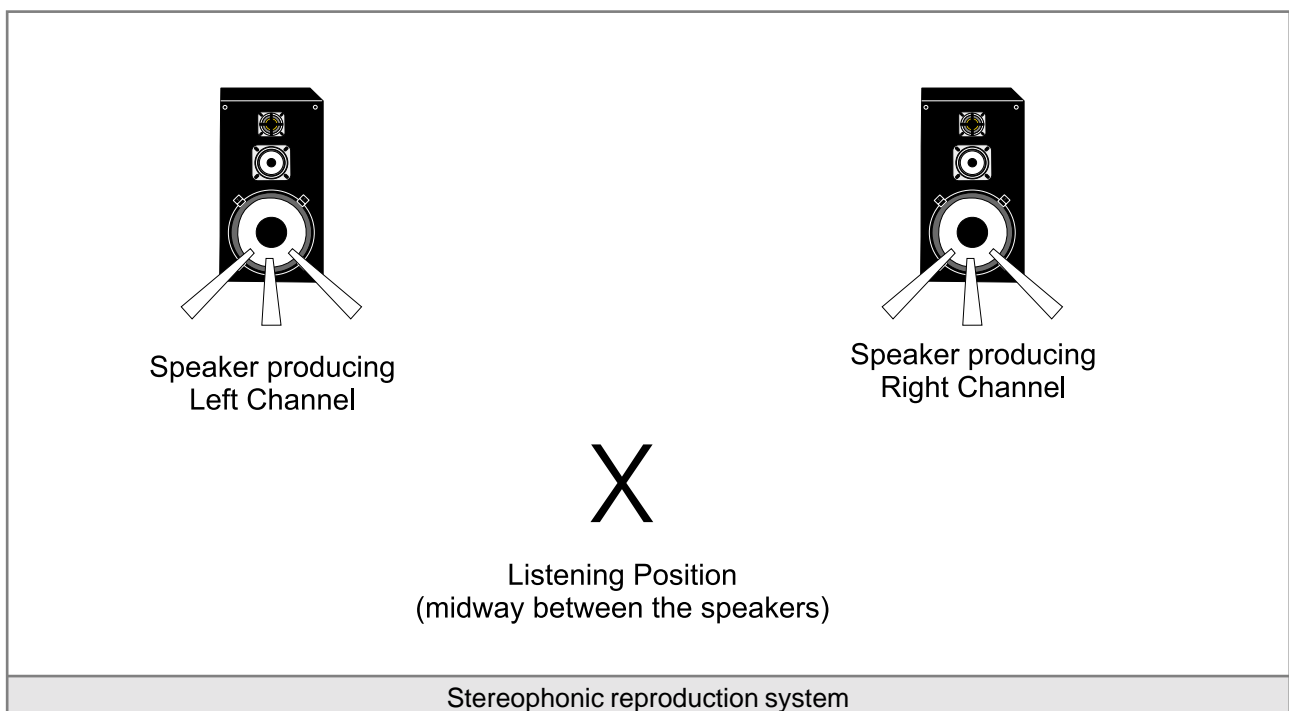
The wired STL has the advantage that it does not require line of sight and can often be used across greater distances. The wired STL is also less affected by weather conditions. The advantage of leasing an STL is that the station is not responsible for maintaining the link.

Stereo

Stereo is a term that today is taken for granted. It is widely assumed that FM stations broadcast in stereo and that stereo is "better". But what is stereo? Stereo is actually an abbreviation for stereophonic. Stereophonic is a word based on the Greek words "stereos" meaning solid, and "fonos" meaning sound.

Stereo is not so much solid sound as it is sound that can be moved about in space.

Stereo reproduction records and transmits sound by using two channels. The two stereo channels are referred to as the left and right channels. The two channels feed two loudspeakers that should placed to the left and right of the listening position, as shown in the diagram below.

Speaker producing
Left Channel

Speaker producing
Right Channel

X

Listening Position
(midway between the speakers)

Stereophonic reproduction system

The aim of using two speakers to produce sound is to create a stereo sound image.

The stereo sound image adds a two-dimensional perspective of laterality and depth to the sound that is heard. In a broadcast or production studio, you can use the pan controls to create the effect of lateral movement, that is, movement from side to side. The sound that is heard through the speakers, as we move the pan control, moves from one side to the other.

Stereophonic reproduction was developed in the early 1930s by Bell Telephone Labs in North America and by the British sound engineer, A.D. Blumlein in Europe. Before this, all recordings we made monaurally (mono). Monaural reproduction uses only one channel for recording and reproduction.

Stereo works because of the way we hear. Our hearing allows us to locate the direction from which a sound originates. In a mono system with a single speaker, our hearing instantly places the speaker's position, and we always hear the sound as coming from that one point.

The combination of two sound sources working together in a stereo system creates an illusion of space. By varying the signal in each channel of a stereo system, the sound engineer can position sounds so that they sound as if they are coming from somewhere other than the speakers.

Stereo Connections

All stereo equipment uses paired outputs and inputs. You need paired connections because you have to connect both the right and left channels. Stereo connectors are labelled R, or Right, and L, or Left. They are normally also colour-coded, with a RED mark on the connector for the right channel, and a WHITE or BLACK mark on the connector for the left channel.

To maintain the quality of the stereo sound image, it is very important to keep the stereo connections consistent throughout the studio. This means that left channels are connected to left channels and right channels to right channels. You may not at first notice if you accidentally swap the channels – but with careful listening, you will pick up the changes in the stereo image.

Recording in Stereo

When recording music from CD to MD or cassette tape, you will most often be making a stereo recording. This means that the left and right channels from the CD or other source equipment must be connected to the left and right inputs of your MD recorder or other recording equipment.

In day-to-day radio work, recordings from a microphone are in mono. A voice is a monaural sound source, so it is not necessary to record it using stereo. The microphone is usually set up in such a way that its mono signal is fed equally to the left and right stereo channels. This is ideal for the presenter, as it places the presenter's voice exactly in the middle of the stereo sound image. However, you can create interesting effects in the production studio by using panning to "move" voices recorded with mono microphones.

Recording live sounds in stereo requires using a special stereo microphone or involves special arrangements of two or more microphones. For example the "kunstkopf" method uses two microphones that are inserted into the ears of the recording engineer to capture the depth and perspective of normal hearing. The "Blumlein microphone technique" uses two figure-of-eight microphones placed at 90° to each other, nearly touching. Alternatively, two cardioid mikes are placed at 120° to one another.

Under M in the A-Z for more about different kinds of microphones.

Stereo microphones are expensive and require good microphone technique to be used effectively.

Telephone Hybrid

A telephone hybrid makes it possible to talk to callers on-air. It allows you to connect audio from a telephone line to your studio equipment, and also to send audio from the studio to a caller.

It is an essential piece of equipment in a community radio broadcast studio as it enables the station to interact with listeners. The hybrid also makes it possible to interview people who are far away. Using a hybrid you can have on-air guests from all over the country.

Telephone Basics

Telephone circuits are designed for voice communication and only deliver audio in a frequency range from 300Hz to 3.4kHz. This is far from the full audio spectrum that extends from 20Hz to 20kHz. This is why sound from a telephone has a very distinctive "tinny" quality that is instantly identifiable to the listener.

Under S in the A-Z for more on sound and audio.

The telephone company supplies your phone handset with power via the phone line. So there is always a voltage present on your phone line. This voltage is something you want to keep away from your audio equipment.

The term "telco" is often used to describe the channel on a mixing desk that connects with the phone. Telco stands for "telephone company" – the telco channel is the channel connected to the telephone company.

What Does a Hybrid Do?

The telephone system carries both sides of a telephone conversation on the same wires. The job of a hybrid is to separate these two signals from one another. The two signals are:

• The received audio, that is the voice of the caller, which must be fed to the studio mixing desk. This signal is then mixed with other programme audio so that listeners can hear the caller's voice.

• The send audio, that is the presenter or interviewer's voice and any other signals from the studio. This must be sent to the caller so that s/he can hear the presenter or interviewer or other voices from the studio.

If these signals are not separated they will interfere with one another, creating feedback and degrading the sound of the programme.

Look at the diagram below to see how the hybrid separates these signals.

Operation of a telephone hybrid
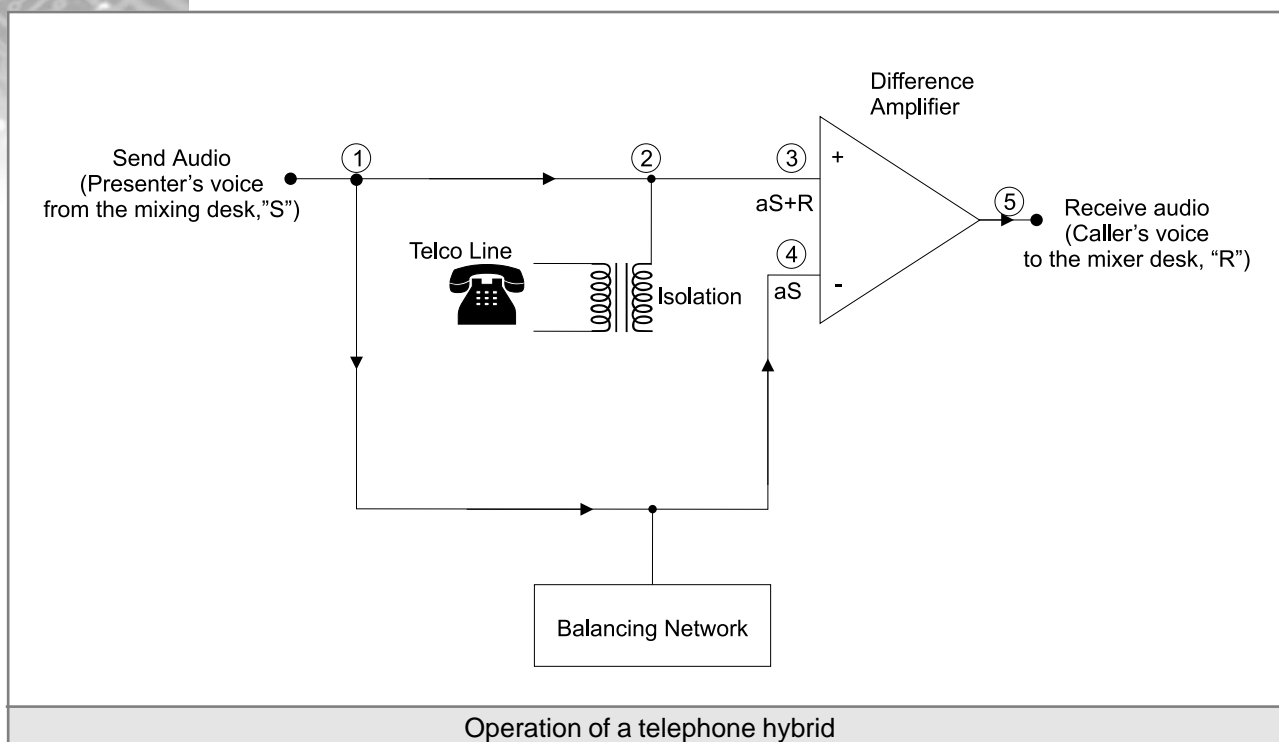
In the diagram, the mixing desk sends a presenter's voice to the hybrid. This is the send audio at Point 1. We'll call the presenter's voice signal "S". The hybrid has to send the presenter's voice to the phone line at Point 2, so that the caller can hear the presenter. But the electrical characteristics of the phone line will change the presenter's voice by an amount we'll call "a", giving you an unwanted signal we will call "aS".

The voice of the caller is also coming from the phone line. We'll call the caller's voice (the receive audio) "R". So the signal at Point 2 is made up of both the unwanted presenter's voice plus the required caller's voice or "aS+R".

The signal we need to send to the mixing desk must only consist of the caller's voice (R). To get the caller's voice (R) from the signal that is available at Point 2, the hybrid must remove the unwanted part of the signal (aS). Extracting the caller audio, R, from this combined presenter and caller signal is the most important aspect of the hybrid's operation.

To do this, the hybrid subtracts the unwanted signal from the caller's voice. However it first needs to recreate the unwanted signal. This is done by feeding the presenter's voice to a balancing network. The balancing network is designed to balance out the characteristics of the telephone line by changing the presenter's voice in the same way the telephone line does. So it creates the signal "aS".

The signal from the phone line at Point 3 is "aS+R". The balancing network creates the signal "aS" at Point 4. If the signal at Point 4 is subtracted from the signal at Point 3, you get R, as follows:

aS + R - aS = R

This is the caller audio that is needed by the mixing desk. The hybrid uses a difference amplifier that performs the subtraction shown in the diagram. The output from the difference amplifier is only the caller audio and this can be sent to the mixing desk.

The isolation element shown in the diagram is a transformer that prevents the telephone line power supply from being transferred into your audio equipment.

The accuracy of the balancing network in matching the characteristics of the phone line defines how well a hybrid will work and also how much you will pay for it. If the balancing network is not very accurate, then the subtraction operation will not completely remove the send audio. In this case the signal sent to the mixer will still contain components of the send audio. These components can cause feedback and distortion of your radio programme.

The amount that these components are reduced by is called separation. Separation will be listed in the hybrid specifications in negative decibels. The greater the negative number, the better the hybrid. An analog hybrid does not cancel all of the send audio, but usually enough is cancelled so that and feedback and distortion are not a problem. More expensive digital hybrids effectively cancel out all of the send audio.

Mix Minus

Mix minus is a term that you will often find with reference to hybrids. The signal that is sent from the mixing desk to the hybrid send input is called a "mix minus" or "cleanfeed" signal. It is called mix-minus because it is the mix of all the sources you want to feed to the telephone, minus the phone hybrid output. The hybrid output is the caller audio. This signal is needed for two reasons:

- First, you want to feed the caller an audio signal so that he or she can hear everything that over-the-air listeners hear.

- Second, if caller audio is fed into the hybrid send input, the hybrid is unable to properly perform the "subtraction" described previously. Sending the caller audio into the hybrid input also creates a "feedback loop" similar to the when you send the output of a recording device back to its input.

Under F in the A-Z for more on feedback.

Most broadcast mixers are supplied with dedicated "telco" modules. These modules provide at least one mix-minus.

If you don't have a dedicated mix-minus, then you need to make sure that the signal feeding your hybrid does not contain caller audio. Feeding the hybrid input from one of the mixer's auxiliary busses can do this. Just make sure that all your programme inputs are connected to this bus and that your hybrid output is not.

Under M in the A-Z for more about mixers.

Using the Hybrid

While the inner workings of the hybrid are quite sophisticated, operating the hybrid is very easy. Operating a hybrid is just like using a hand phone, only you use buttons on the hybrid rather than an actual phone. Normally, hybrids don't "ring" out loud. A hybrid should have a light that flashes to alert you to the fact that someone is calling. This prevents the sound of ringing telephones unexpectedly interrupting your programme.

When you see that there is a caller, you answer the call by pressing an answer or ON button. If you have a hybrid that can handle multiple phone lines, you press the button for the line that is ringing. This connects the call to the mixing desk and you can then usually speak to the caller through one of your studio microphones. When the call is over you would press an OFF or DROP button to disconnect the call.

A hybrid may have a keypad that allows you to make calls, but many hybrids can be connected to a standard telephone handset that is used for dialling.

A multi-line hybrid can handle several incoming lines. This allows you to "stack up" callers who can then be put on the air as needed. A multi-line hybrid must allow you to easily move between each caller so that the programme can progress smoothly.

Settings

Analog hybrids usually require that you tune the balancing network described previously to get the best separation. In this case, the hybrid usually has a "null" adjustment. Setting this involves making a call and adjusting the "null" until the best quality audio is heard through the mixer.

Transmitter

Transmitter is a general term used to describe the combination of equipment that turns an audio signal from a broadcast studio into an FM or AM radio signal that can be picked up by a radio receiver.

It may be useful to take another look at the section on transmission on Page 29 before reading further.

The diagram below shows the components and signals that make up a typical FM stereo transmission system.

Typical components of a stereo FM transmission system
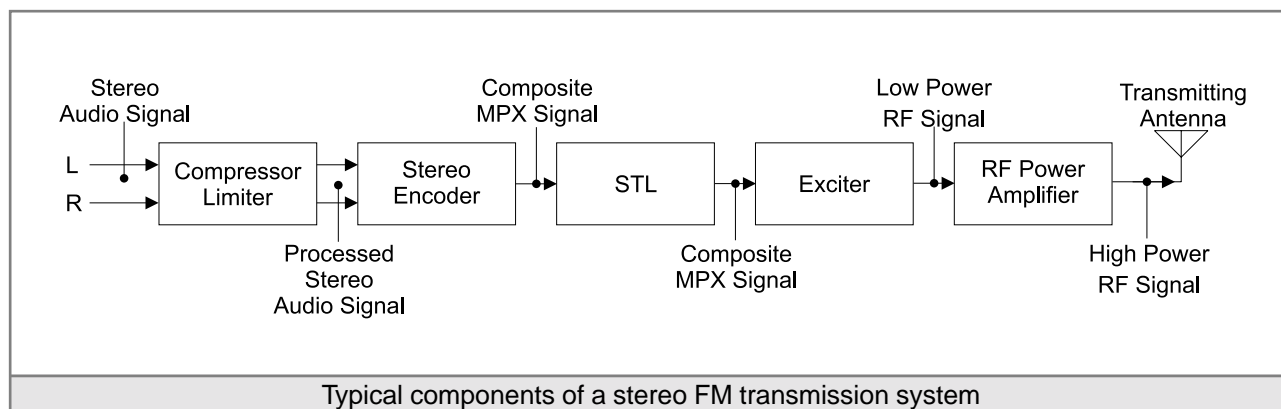
Often, all of the components shown in the diagram are provided as one piece of equipment – a transmitter. In this case, all the different devices and functions would be located in one box.

But, just as often, the components are built into separate pieces of equipment.

Components of a Stereo FM Transmission System

Compressor Limiter

The diagram starts on the left, with the stereo audio output of the broadcast studio. This output is fed into a compressor limiter, or other final processor.

Under C in the A-Z for more about compressor limiters.

The compressor limiter is often a stand-alone piece of equipment. The compressor limits the level of the signal to prevent overdriving the other equipment, and applies compression and processing to improve the broadcast sound.

Stereo Encoder

The processed audio signal is then fed to a stereo encoder. The encoder combines the audio of the left and right channels, and generates a 19kHz pilot tone that is used by receivers to detect a stereo broadcast. The encoder can also add other components to the signal.

Under S in the A-Z for more about stereo sound and audio.

Because the output of the encoder consists of several components, it is called a composite or multiplexed output. Multiplexed is abbreviated as "MPX" This abbreviation is often used to label the outputs of encoders and exciters.

STL

The arrangement in the diagram is typical of a station that uses a wireless studio-transmitter link (STL). In this case the STL equipment generates the MPX signal.

Stations that use wired Telkom STLs often send processed audio to the transmission site. In these cases, the STL precedes the stereo encoder in the transmission system.

Under S in the A-Z for more about STLs.

Exciter

The functions of the encoder and the exciter are most often performed by one piece of equipment – an exciter. The exciter is the device that generates the radio frequency carrier wave. The exciter also modulates the carrier with the encoder's MPX signal. This creates a radio frequency (RF) signal that can be transmitted by an FM antenna.

Under M in the A-Z for more about modulation.

The exciter is normally specified in terms of how much output power it can deliver. A dedicated exciter normally delivers up to 50 watts of power. Many community stations are licensed to broadcast using less power than this. In these cases, the stations feed the transmitting antenna directly from the exciter.

RF Power Amplifier

Stations that are licensed to broadcast with more power may follow the exciter with one or more RF power amplifiers. These amplifiers boost the power of the RF signal. The power boost can be enormous and commercial broadcasters will amplify the RF to several kilowatts.

Antenna

The final stage of the transmission system is the transmitting antenna. A transmitting antenna radiates the RF energy into space. The listener's radio set, or receiver, has a receiving antenna (aerial) that receives some of this energy from space.

• AM Antenna

For AM, a quarter wavelength transmitting antenna is used. The low frequencies used for AM transmission mean that the dimensions of a quarter wavelength are considerable between 50 and 140 metres. So the AM antenna is a large structure. It has to be designed to match the broadcast frequency of the station. Great care must be taken in locating the antenna, and a lot of excavation work will have to be done to secure the tower and install ground mats.

- **FM Antenna**

The FM transmitting antenna is much smaller and easier to install than the AM antenna. There are several different kinds of FM antennas. Many community radio stations use a dipole antenna. This is an omnidirectional antenna, which radiates equally in all directions. The antenna's dimensions are the same as half a wavelength of a station's broadcast signal. Because FM signals are at frequencies so much higher than AM frequencies, half a wavelength is much smaller, and FM antennas are less than two metres in size.

How the Signal is Broadcast and Received

You are probably still wondering how the signal sent to the antenna gets into the air. The RF signal in an antenna is an electric current, and all electric currents have an electromagnetic field associated with them. As current flows in the antenna, an electromagnetic field is created around the antenna. An antenna is designed to radiate this electromagnetic field. The electromagnetic field travels from the antenna in all directions at the speed of light.

This electromagnetic field is what is detected by the listener's radio. The electromagnetic field produces a current in a receiving antenna (aerial) on a listener's radio. This received RF current is amplified in the radio and the transmission process explained above is basically applied in reverse to retrieve the audio signal that can be played through the radio's speaker as sound.

Under E in the A-Z for more about the electromagnetic spectrum.

Uninterrupted Power Supply (UPS)

All of the studio and transmission equipment at your radio station requires a source of electrical power to work. Eskom or your local council supplies this electrical power, or mains supply. Using mains power normally requires connecting a plug to a power point in the wall.

Many stations are located in areas where mains power supply is unreliable. Unless you have back-up, a mains power failure will interrupt your broadcast. Some stations have been off air for over a day as a result of mains power supply failure. This is a serious problem – remember, your listeners will also be affected by the electricity failure, and will want to rely on their radio station to keep them informed about it!

This is the purpose of an uninterrupted power supply (UPS). As the name suggests a UPS provides uninterrupted power to the station even when the mains supply is disturbed.

There are many different UPS systems that supply varying degrees of back-up protection. The choice of system for your station should be made after considering the quality of the power supply in your area. For example, stations located in areas that suffer from constant power disturbances will need a more complete power back-up system than stations located in areas where the power supply is more stable.

Different Kinds of Power Disturbances

The mains supply is meant to be a stable AC voltage of 230 volts at 50Hz, but there are several kinds of power line disturbances that affect the mains supply.

"Spikes" or Power Surges

Lightning, power network switching and the operation of other high power equipment in your building, such as elevators, spot welders and so on will cause "spikes" in the mains voltage. A power spike or a power surge is when the mains voltage jumps to well over 230 volts for a short time.

Undervoltage

The mains voltage can also can dip below 230V, providing an undervoltage supply. This can be caused by faults in the power network and sharp load changes. This kind of condition is often seen in light bulbs that dim in intensity.

Noise

The mains supply can also be "noisy" when signals at frequencies other than 50Hz find their way onto the power lines.

Power "Blackout" or Failure

Lastly, the supply can black out. In this case there is a total voltage loss and all electrical equipment is left without power. Blackouts can last for a few moments, or sometimes they can last for a number of days.

What the UPS Does

A stable power supply is best for the smooth and safe operation of your station's broadcast equipment, and this is what a UPS should provide. A UPS counters power disturbances by fulfilling two vital functions:

- Firstly, a UPS acts as a buffer between the power sensitive studio equipment and the fluctuating mains power supply, providing protection from power surges, dips and line noise.

- Secondly, a UPS provides back-up power in the event of a complete power failure. The duration of the availability of back-up power is dependent on the batteries used as part of the UPS. A generator can also be included to provide back-up power for extended periods.

A UPS usually looks a lot like a computer. UPS's are supplied in an off-white case that is more or less the same size a computer's case. This is because the most popular use of a UPS is to protect computer systems, and the cases are meant to look at home in rooms full of computers.

Types of UPS

There are three main types of UPS: the off-line, line interactive and online UPS. Each one has different features:

The Off-line UPS
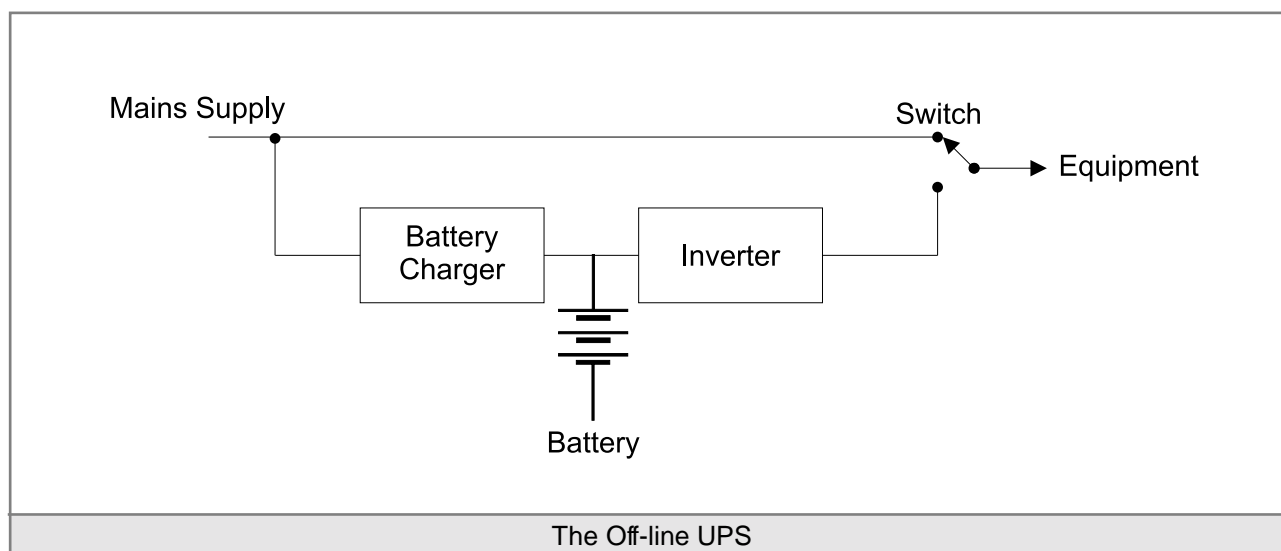
The Off-line UPS

Off-line or standby UPS's are those power systems where equipment is normally powered directly from the mains supply. The offline UPS only supplies the equipment when the mains supply is not available. These systems are intended for equipment that can tolerate momentary loss of power while the system switches to back-up power. They provide no protection from surges or dips in mains voltage.

The diagram above shows a battery system used to provide back-up power. The inverter shown in the diagram is a device that converts the direct current

(DC) supplied by the battery to the alternating current (AC) that is needed by equipment.

If you do not wish to use a battery, you could use a generator to provide back-up power in the same way.

This kind of system is appropriate for areas with minimal power problems.

The Line-interactive UPS

The Line Interactive UPS
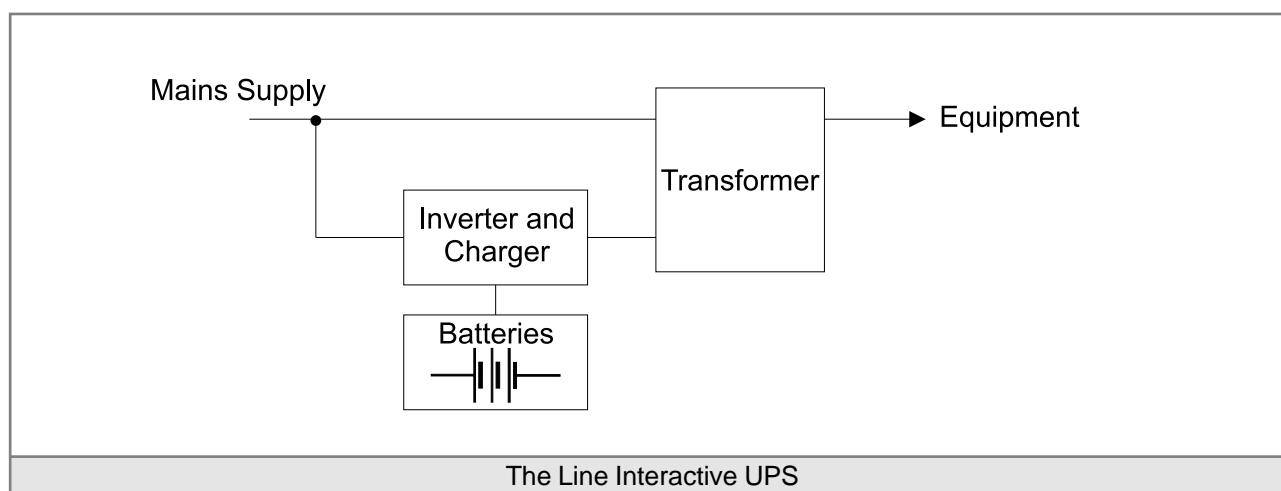
Line-interactive or single conversion UPS's are used when mains voltage fluctuations (dips and spikes) are a problem. In the line-interactive system, the mains supply is fed directly to the equipment through an inductor or transformer. Switching elements, in combination with the inverter, monitor the mains supply to control the supply to the equipment.

The name "line-interactive UPS" comes from the fact that the inverter interacts with the mains line to drop, boost, or replace AC power as needed to maintain voltage control.

These UPS's are cost effective but still expose equipment to short power fluctuations while switching happens.

The On-line UPS
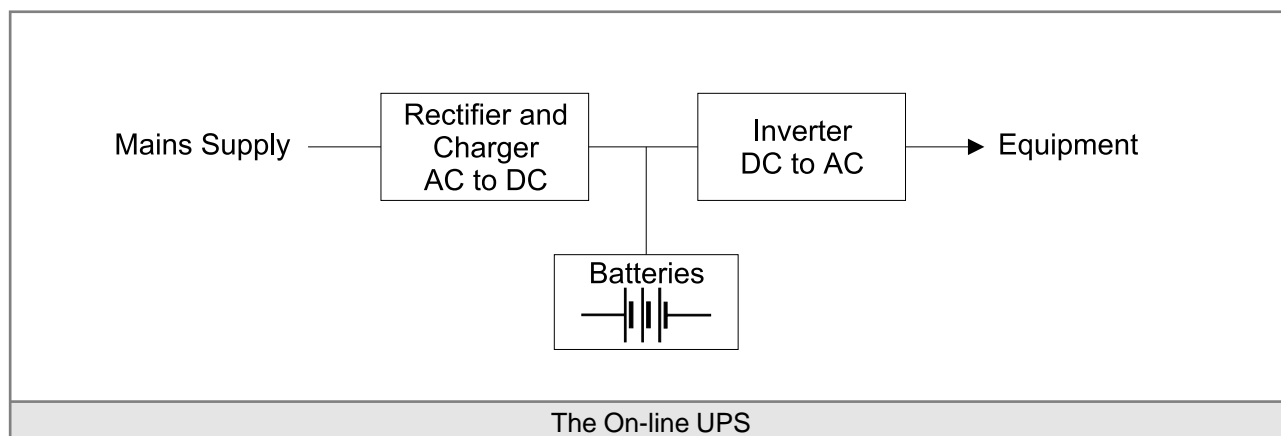
The On-line UPS

The on-line UPS provides the highest level of power protection, as the equipment is always powered by the UPS and not the mains supply.

The on-line UPS is also called a double conversion on-line UPS, as input power is converted twice, as follows:

• Firstly, input power is converted to DC, through use of a rectifier. This eliminates any surges or dips and line noise.

• Secondly, an inverter converts the DC into a continuous, completely re-conditioned AC supply.

In the event of a complete mains blackout, the inverter draws DC power from the UPS batteries. This is also the case with the line-interactive UPS.

On-line UPS's are the most expensive back-up system. However, if your budget allows, these are the most appropriate for the radio studio equipment.

UPS Specifications

Once your station has settled on what type of UPS to use, there are two more specifications to consider.

Power Rating

Firstly, a power rating, usually specified in kilovolt-amperes (kVA). This is determined by the power requirements of your equipment. Each piece of equipment you intend to protect by using a UPS draws a certain amount of power. By measuring the power consumption of all your equipment you can find out how much power a UPS must be able to supply to keep all the equipment running. This is usually a task best left to your UPS supplier or electrician, although a good estimate would be that about 1kVA of UPS power is needed for each studio. This figure excludes back-up power for lights and transmission equipment.

Battery Power

The second specification is the duration of back-up power that the UPS batteries can supply. This specification greatly influences the size and cost of the UPS. Depending on their size, batteries can supply from a few minutes up to a few hours of back-up power.

Stations in areas where the power supply is often lost for long periods might choose a battery – generator combination. In this case you would have a UPS battery with power limited to a short period, and connect a generator to the system to provide back-up for long periods. The batteries then only need to provide enough back-up time to start or refuel the generator.

Stations in areas where blackouts happen often, but do not last for very long, might not be able to justify the expense of a generator, but would still like a reasonable amount of back-up. These stations should choose a system with a few hours of battery power.

Consult With Others In Your Area

It is useful to consult with your local council's electrical department on this issue. They should be able to give you detailed information on the nature of power disturbances in your area. This information will help you to make the best decision on a UPS. Hospitals, police stations and other emergency services in your area are also likely to make use of UPS systems. Contacting them and hearing their experiences will put you in a position to make a better UPS decision.

Generator

A generator is a petrol- or diesel-driven engine that produces electricity. Generators are used to supplement the battery power of a UPS, or can be used as the sole source of back-up power in an off-line system.

If a generator is used on its own in an off-line system, the station will be without power for the time it takes to get the generator running.

Modern back-up generators that are used to supply several kVA of power to a studio are generally small and relatively portable. However, a generator presents an additional maintenance burden to a station. A generator is an engine much like a car engine, and like a car engine it needs regular use and servicing to stay in good working order.

An electrical generator also presents a safety hazard. A generator operates with deadly voltages and currents. Improper use or installation of a generator can cause property damage, serious injury and even death.

If your station uses a generator read and follow the operating and safety instructions contained in the user manual to protect yourself, your station's equipment and, most important, staff and volunteers.

Generator Safety Tips

1. The installation of the generator and the disconnect and transfer switches MUST be performed by a qualified electrical contractor. This is not a job for the station's technical department. Do NOT attempt to install these devices to your electrical panel, it is EXTREMELY DANGEROUS!

2. ALWAYS check and follow local and national fire and electrical codes. You may find out, for example, that your building lease or equipment insurance attaches very strict conditions to using a generator.

3. NEVER run an electric generator inside a building.

4. ALWAYS store the generator fuel (petrol or diesel) in proper containers, in a safe and secure place.

5. NEVER fuel an electric generator while the generator is running!

6. DO NOT smoke or bring naked flames near the generator, or when handling fuel.

7. Ensure proper ventilation around the generator.

8. ALWAYS have a fully charged, approved fire extinguisher located near the generator.

9. ALWAYS disconnect from the mains supply BEFORE starting your backup generator. The generator should normally be installed so that this happens automatically.

10. NEVER remove or tamper with safety devices – they are there to protect you.

11. NEVER attempt to repair an electric generator yourself. ALWAYS refer repairs to your supplier or other qualified serviceman.

12. Many engine parts are very HOT during operation. Don't touch them as you could get severely burnt.
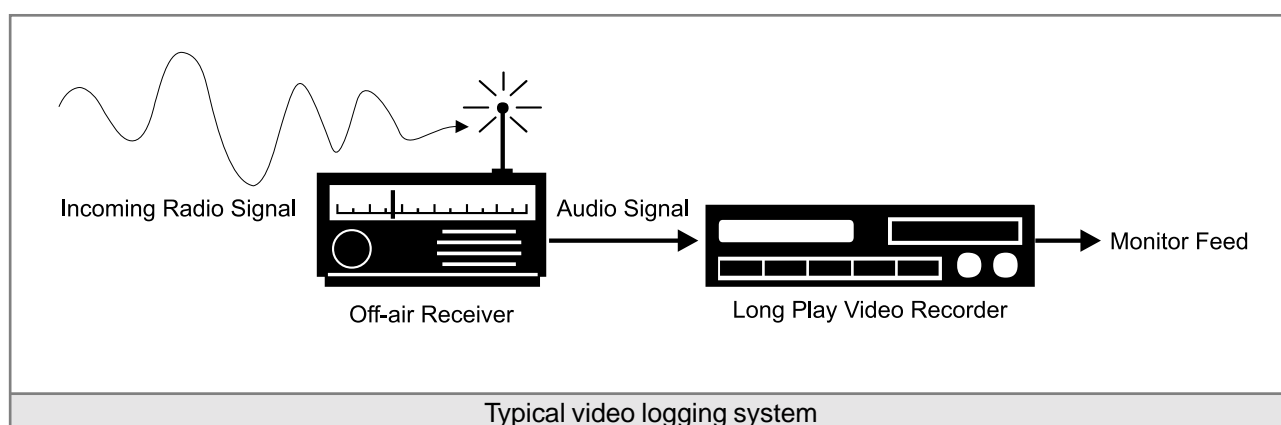
Video Logging System

It is a legal requirement for all South African radio stations to record their broadcasts. It is also required that these recordings are kept for at least 40 days after the broadcast has taken place.

It is important to read this section carefully – if you fail to log, or record your broadcasts as the law requires, you run the risk of having your licence taken away.

To make these recordings, most stations make use of a video logging system. The system consists of a tuner (radio receiver) and a video recorder (VCR). The tuner is tuned into the station, and the audio from the tuner is fed to the video recorder. A tuner is used rather than a feed from within the studio so that what is logged (recorded) is a record of exactly what was broadcast.

The diagram below shows a typical logging system.

Incoming Radio Signal · Off-air Receiver · Audio Signal · Long Play Video Recorder · Monitor Feed

Typical video logging system

Why You Use a Video Recorder

The reason for using a video recorder is that videotapes can store more audio than many other media can.

Video recorders can also record at half speed in "long play" mode. This means that double the amount of audio can be stored on one videocassette. If you use long play, a three-hour tape can store six hours of reasonable quality audio. This means that only four tapes are needed to record an entire day's broadcast. To store and archive 40 days worth of broadcasts, as the law requires, your station will need 160 three-hour tapes.

A video recorder and videotapes are much cheaper than other bulk storage devices. As the recordings are not used for broadcast, the relatively poor audio quality of video recordings is not a concern.

How a Video Logging System Works

Several community radio stations have experienced problems using their logging system. But the system is simple to operate and easy to understand. To comply with regulations, it is important to ensure that the logging system works.

Firstly, the tuner must be tuned to your station. This is obvious – if the tuner feeding the logging system is tuned to another station, the recorder will record that station and not yours.

We suggest that once your tuner has been tuned in, you cover the tuner's control panel or put the tuner in a cabinet where no-one will tamper with it. Visitors to studios often twiddle knobs… sometimes with disastrous results.

The tuner is usually connected to the video recorder's AV input. The video recorder must be set to record from that input. This not as simple as it sounds – a video recorder can often have more than one AV input and can receive signals on many other channels.

It is vital that the video recorder is at all times set to record audio from the input connected to the tuner. As mentioned, this is usually the AV input. Select the right channel by using the video recorder's remote control or front panel.

Correct setting of the VCR is very important. If the recorder is not set to the AV input, your logging tapes will contain hours of silence or static, which won't help you if someone complains about your broadcast or decides to take you to court for something you said on air.

Managing a Video Logging System

The recorder is the only part of the system that needs to be handled regularly. This is why most stations put the video recorder in the broadcast studio. Every few hours the videotapes used in the video recorder need to be changed. To do this, the presenter takes out the old tape, inserts a new one, and presses the record button on the recorder.

Choose the times that the tapes are changed very carefully to ensure someone is available to do it. Also, try to make sure that when you change the tapes you don't interfere with the broadcast.

Sometimes the video recorder can also accidentally be switched off "long play". When this happens, your tapes will only record for half as long as you expected. It is very important to keep the recorder set on "long play".

If, as we've mentioned above with the tuner, you put a guard or some kind of protection over the VCR's panel once you've set it up, no-one will be able to change the settings.

Monitoring the System

The diagram shows a monitor feed from the video recorder. This is a very important feature. Most stations feed the audio from the video recorder to the studio mixing desk or some other equipment that allows you to listen to the sound from the recorder. This allows you to listen to old recordings and also to check every now and then that the recorder is recording properly and that the tapes are still in reasonable condition.

To Sum Up….

The logging system is an ICASA requirement, and many stations find it tedious and difficult to manage. Given the number of tapes you need, it is also quite expensive. However, there are other advantages.

The logging system can provide a valuable resource and record of the station's programming. You can also use the logging tapes for evaluating presenters and programmes.

But most important, the logging tape is the only legal record of a broadcast that the station can use if there is a legal dispute or complaint. The only way to review your broadcast to check the validity of a complaint, and formulate a response, is to listen to the logging tapes.

Appendix A:

Abbreviations

ADC	Analog-to-Digital Converter (often written as AD Converter)
AM	Amplitude Modulation
BPM	Beats per minute
CD	Compact Disc
CD-R	Compact Disc Recordable
CD-ROM	Compact Disc – Read Only Memory
CD-RW	Compact Disc Rewritable
CPU	Central Processing Unit
CR	Community Radio
DA	Distribution Amplifier
DAC	Digital-to-Analog Converter
DAE	Digital Audio Extraction
DAT	Digital Audio Tape
dB	Decibel
EQ	Equaliser
FM	Frequency Modulation
FX	Effects
Hz	Hertz
IBA	Independent Broadcasting Authority
ICASA	Independent Communications Authority of South Africa
ICT	Information and Communications Technologies
I/O	Short for Input and Output
IPS	Inches per second
ISDN	Integrated Services Digital Network
ISP	Internet Service Provider
LCD	Liquid Crystal Display
LED	Light Emitting Diode
MD	Mini Disc
NCRF	National Community Radio Forum
OS	Operating System
OSF-SA	Open Society Foundation for South Africa
OB	Outside Broadcast
PC	Personal Computer
PFL	Pre Fade Listen
PPM	Peak Programme Meter
RAM	Random Access Memory
RF	Radio Frequency
ROM	Read Only Memory
SPL	Sound Pressure Level
STL	Studio to Transmitter Link
TRS	Tip, Ring, Sleeve. Used to describe a jack with 3 connections
TX	Transmitter
UPS	Uninterrupted Power Supply
VU	Volume Unit
XLR	Type of connector used to carry balanced signals

Appendix B:

Prefixes

Throughout this manual, and in many other technical documents, units are prefixed by a letter that denotes magnitude. For example the unit of frequency is Hertz (Hz), but in FM transmission we deal with Megahertz, written as MHz. The "Mega" and the "M" used as a prefix denote that we are working a unit of a million Hertz. It's clumsy and a little ridiculous telling people that they can tune into your station on eighty nine million five hundred thousand Hertz (89500000 Hz). It is much easier to talk about 89.5MHz.

There are several prefixes. Some of them should be quite familiar to you. For example, kilo as in kilometre, meaning a thousand metres. Another is milli, in millimeter, meaning a thousandth of a metre.

Some of the more common prefixes are tabulated below:

Power of Ten	Number	Prefix	Abbreviation
10^{-9}	0.000 000 001 (a billionth)	nano	n
10^{-6}	0. 000 001 (a millionth)	micro	m
10^{-3}	0. 001 (a thousandth)	milli	m
10^{-2}	0.01 (a hundredth)	centi	c
10^{-1}	0.1 (a tenth)	deci	d
10^{3}	1000 (a thousand)	kilo	k
10^{6}	1 000 000 (a million)	mega	M
10^{9}	1 000 000 000 (a billion)	giga	G

Appendix C:
South African Broadcast Equipment Suppliers

Broadcast & Installation Engineering cc (B&I)

B&I supply broadcast equipment and studio installation, with a focus on transmission.

PO Box 1318, Highlands North 2037, Johannesburg

Tel:	(011) 452 4962
Fax:	(011) 452 4964
Contact:	Frans Jooste

Broadcast Solutions Electronics (BSE)

BSE manufacture, supply and install FM transmission equipment primarily for radio, but also for television.

PO Box 446, Bergvliet 7864

Tel:	(021) 762 4521
Fax:	(011) 761 1737
Contact:	Thys de Beer

D Audio Services

D Audio services supply broadcast equipment and studio installation.

Box 628, Little Brak River 6503

Tel:	(044) 690 4100
Fax:	082 320 5923 or (044) 690 4100
e-mail:	d.h@mweb.co.za
Web-site:	www.daudio.co.za
Contact:	Dean Holdstock

Eminently More Suitable (EMS)

EMS supply broadcast equipment.

PO Box 1026, Melville 2109

Tel:	(011) 482 4470
Fax:	(011) 762 2552
e-mail:	mail@emsafrica.co.za
Web-site:	www.emsafrica.co.za
Contact:	Mike Shepstone

Globecom

Globecom supply and install broadcast equipment and provide service and maintenance.

PO Box 50559, Waterfront 8002

Tel:	(021) 555 4701
Fax:	(021) 555 4711
e-mail:	matt@globecom.co.za
Web-site:	www.globecom.co.za
Contact:	Matthew Buck

Sound Fusion Media

Sound Fusion supply and install broadcast equipment and provide service and maintenance.

PO Box 3073, Pinegowrie 2123, Gauteng

Tel:	(011) 880 5233
Fax:	(011) 880 5234
e-mail:	soundf@global.co.za
Contact:	Mike Collison

New Installation Company (N.I.C.)

NIC supply and install broadcast equipment and provide service and maintenance.

PO Box 1202, Florida Hills 1716

Tel:	(011) 794 5984
Fax:	(011) 794 5204
e-mail:	karl@nic.co.za
Contact:	Karl Britz

Prosound

Prosound supply, install and rent broadcast equipment.

PO Box 261458, Excom 2023

Tel:	(011) 334 6550
Fax:	(011) 334 6826
e-mail:	jhb@prosound.co.za
Web-site:	www.prosound.co.za
Contact:	Lee Thomson

Sentech

Sentech provide signal distribution, as well as transmission service, advice and information.

Private Bag X06, Honeydew 2040

Tel:	(011) 471 4400
Fax:	(011) 471 4758
Contact:	James Odendaal

Telemedia (Pty) Ltd.

Telemedia supply and install broadcast equipment.

PO Box 1853, Rivonia 2128, Gauteng

Tel:	(011) 803 3353/4
Fax:	(011) 803 2534
e-mail:	telemedia@pixie.co.za
Contact:	Richard Fulton

Tru-Fi Electronics

Tru-fi supply and install broadcast equipment.

PO Box 111, Kyasand 2163

Tel:	(011) 462 4256/7/8/9
Fax:	(011) 462 3303
e-mail:	trufi@trufi.co.za
Web:	www.trufi.co.za
Contact:	Flemming Ravn

Wild and Marr

Wild and Marr supply and install broadcast equipment.

PO Box 318, Plumstead 7801

Tel:	(021) 762 8615
Fax:	(021) 762 8619
e-mail:	Mario@wmc.co.za
Web-site:	www.wildandmarr.com
Contact:	Mario Cupido

Appendix D:

Useful Websites

The Internet is a good place to start when considering equipment. Most equipment manufacturer's websites provide detailed information on their products and often list recommended prices. This allows you to make more informed decisions when looking at equipment quotations. Visits to these sites and subscribing to some of the manufacturer's mailing lists is one of the easiest ways to keep up to date with new products and technical developments.

Equipment Manufacturers:

The list of equipment manufacturers presented below is by no means complete, but does cover many of the manufacturers whose equipment is currently used in South African community radio stations. (All web page addresses were correct at the time of printing.)

AEQ: www.aeq.es

AEV: www.aev.net

AKG: www.akg-acoustics.com

AKG manufacture microphones, headphones, and accessories.

Behringer: www.behringer.com

Behringer produce a wide range of signal processors including mixers and pre-amplifiers and effects units

Beyerdynamic: www.beyerdynamic.com

Beyerdynamic produce microphones, headphones and conference systems.

Denon: www.del.denon.com

Denon produce a broad range of equipment, both professional and domestic. Denon CD players are a standard piece of equipment in many community radio stations.

Electrovoice (EV): www.electrovoice.com

EV produce a broad range of equipment. EV microphones are popular for radio use.

Dateq: www.dateq.com

Dateq produce mixing desks.

Fostex: www.fostex.com

Fostex manufacture recording equipment, particularly multi-track recorders. Their headphones are used in many community stations.

Gentner: www.gentner.com

Lexicon: www.lexicon.com

Lexicon produce effects processors.

Marantz: www.marantz.com

Neumann: www.neumann.com

Neumann produce high quality microphones.

Neutrik: www.neutrik.com

Neutrik manufacture a wide range of audio connectors.

Orban: www.orban.com

Orban produce broadcast audio processors and digital editing systems.

Otari: www.otari.com

Otari produce a range of recording equipment from MD player/recorders to digital editing systems.

Rane: www.rane.com

Rane produce a wide range of signal processors including mixers, pre-amplifiers and effects units.

RØDE Microphones: www.rode.com.au

RØDE manufacture microphones. The RØDE "broadcaster" is used in many community stations.

Roland: www.roland.co.uk

Shure: www.shure.com

Shure manufacture a wide range of microphones and accessories.

Soundcraft: www.soundcraft.com

Soundcraft produce mixing desks. Their RM 100 and RM 105 mixers are some of the most widely used in South African community radio.

Sony: www.pro.sony-europe.com

Sony manufacture a complete range of audio products from microphones to digital editing systems.

Tannoy: www.tannoy.com

Tannoy manufacture studio monitors.

Tascam: www.tascam.com

Tascam produce a wide range of recording equipment. Tascam MD, DAT, tape and reel recorders are used in many community radio stations.

Technics: www.panasonic.com/consumer_electronics/technics_audio/index.htm

Although Technics are not strictly a broadcast equipment manufacturer, their products often seem to find there way into community radio studios.

Telos: www.telos-systems.com

Telos manufacture telephone hybrids and digital codecs.

Software Manufacturers

The list of software manufacturers is also nowhere near complete, but again covers much of the software currently used in South African community radio. If your radio station is using broadcast playout software, the manufacturer's website is an invaluable resource. The website should provide user updates and software downloads, allowing you to get the most out of your playout system. (The web page addresses were all correct at the time of printing)

Audiovault: www.audiovault.com

Broadcast Software International: www.bsiusa.com

Makers of Wavecart and Simian Windows based digital playout software.

Digidesign: www.digidesign.com

Makers of Pro-Tools.

Netia: www.netia-broadcast.com

Netia provide playout/editing and radio management software.

On-The-Air: www.softronmedia.com/ibs

Broadcast playout software for the Apple Mac.

Radiohost: www.radiohost.com

Radiohost is playout software that also includes scheduling modules.

Radio Computing Software: www.rcs-works.com

Steinberg: www.steinberg.net

Steinberg produce a range of music production and editing software including Cubase VST, Nuendo and Wavelab.

Syntrillium Corporation: www.syntrillium.com

Syntrillium make Cool-edit Pro an editing and production programme for Windows.

Virtual Radio: www.aev.net

Other Sites

Minidisc Community page: www.minidisc.org.

Everything about minidiscs- History, scientific papers, product reviews and more.

Handbook for Acoustic Ecology: www.sfu.ca/sonic-studio/

This site provides definitions and explanations of hundreds of sound and audio terms.

Appendix E: References

We used the following books and papers to write this manual. They are recommended reading for anyone wanting to know more about broadcasting and sound.

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19. White, P. 1999. <u>Basic Effects & Processors.</u> Sanctuary Publishing

20. White, P. 1999. <u>Basic Microphones.</u> Sanctuary Publishing

21. White, P. 1999. <u>Basic Mixers.</u> Sanctuary Publishing

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