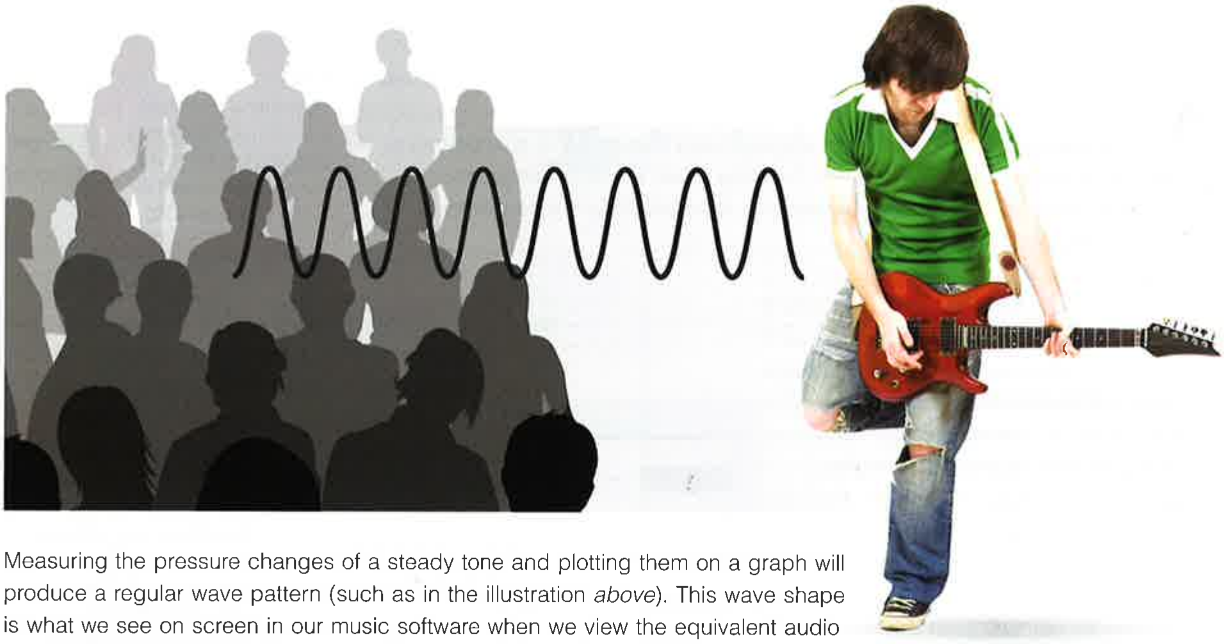
01

CHAPTER 1
THE BASICS

HEARING SOUND

▶HOW DO WE HEAR SOUND?
Our ears hear sound by reacting to changes in the pressure of the air around us. These air pressure changes are turned into chemical and electrical signals which the brain can understand. Ears are very sensitive and can cope with a wide range of sound levels – although they can be damaged by too much sound pressure.

▶HOW DOES SOUND TRAVEL?
A simple way of understanding air pressure changes is to think of them as **waves** in the air between whatever is making the sound and whoever is hearing it.



Measuring the pressure changes of a steady tone and plotting them on a graph will produce a regular wave pattern (such as in the illustration *above*). This wave shape is what we see on screen in our music software when we view the equivalent audio signal or digital waveform.

▶PITCH
We understand sounds to be high or low and call this pitch. With musical notes, though, we are more specific than just high or low – we label the notes with the letters A to G (as on a keyboard) or with the **frequency** of the wave.

▶VOLUME
We also hear changes in the volume of sounds, which we call 'loud' or 'soft'. How our brains make this judgement is affected by a variety of properties of the sound but, to keep things simple, the volume of a sound wave is usually known as its **amplitude**.

▶HOW IS SOUND USED IN MUSIC TECHNOLOGY?
To make use of sound for performing and recording, sound waves are turned into electrical signals. These signals travel through wires (such as a microphone lead), and eventually, using a **loudspeaker** or a **monitor**, are turned back into sound waves in the air again. However amazing our technology becomes, these are the basic steps we need to be able to make.

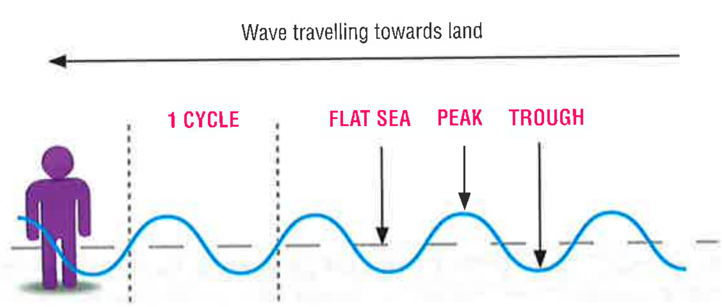
FREQUENCY

Sound waves in the air are invisible, but we can compare them with other waves such as those in the sea.

If you stand in the sea, waves continually pass you by and you will notice that they have high points (**peaks**) and low points (**troughs**).

Frequency simply means how often something happens. So the frequency of a wave in the sea is the number of peaks to pass you in a certain amount of time – for example, 30 per minute.

A WAVE IN THE SEA



▶ CYCLES

Each time a wave reaches you, the sea water will have been through some changes including a peak, a trough and moments when the water is flat. The whole sequence between any two similar points is called a **cycle**, so a wave could have a frequency of 30 cycles per minute.

> **FACT**
> Frequency is measured in **Hertz**, named after the scientist Heinrich Hertz. One cycle per second is called 1 Hertz (or 1 Hz).

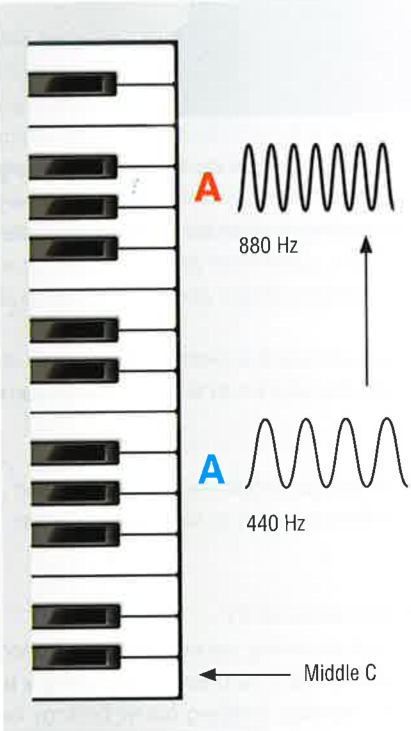
▶ PITCH AND FREQUENCY

Sound waves in the air are much faster than waves in the sea, and have a much higher frequency.

The frequency of a sound wave determines its pitch. The higher the frequency, the higher the pitch.

The note A in the middle of a piano has a frequency of 440 Hz, meaning that the air is vibrating 440 times per second.

As the frequency goes up, so does the pitch of a note. There is a neat relationship here, as every time a frequency doubles we hear a note one octave higher – so the next A up on the keyboard has a frequency of 880 Hz.



> **FACT**
> The human ear can hear sounds from about 20 Hz to about 20,000 Hz (20 kilohertz, or kHz), although the higher frequencies become less clear with age. Some animals can hear much higher frequencies than us.

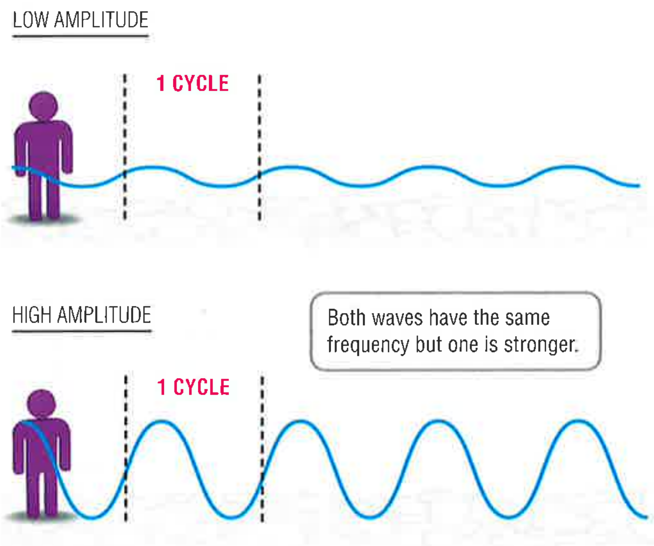
AMPLITUDE

Sometimes the sea can be really calm and almost flat – the waves then have a **low amplitude**. When they swell up and the peaks become taller, they have a **high amplitude**.

The amplitude of a sound wave affects its volume.

A sound wave with a low amplitude has a low volume (i.e. a soft sound).

A sound wave with a higher amplitude has a higher volume (i.e. a louder sound). This is where the familiar word 'amplifier' comes from – a piece of electronic equipment that boosts the signal of a microphone, sound system or instrument.



dB	Sound
140	Jet plane
130	Fireworks
120	Rock concert
110	Chainsaw
100	Personal stereo
90	Timpani
80	Motorcycle
70	Shouting
60	Traffic
50	Phone ringing
40	Piano practice
30	Conversation
20	Rainfall
10	Refrigerator
0	Birdsong
	Library
	Whisper
	Breathing
	Silence

▶ VOLUME MEASUREMENT

The measurement of volume is usually shown on a scale of **decibels**. Each decibel (dB) is one tenth of a Bel, named after Alexander Graham Bell, an early pioneer of the telephone.

Decibels compare sound levels rather than actually measuring them. As the scale goes up, the changes become much more noticeable. For example the difference between 70 dB and 80 dB is much greater than than the difference between 10 dB and 20 dB.

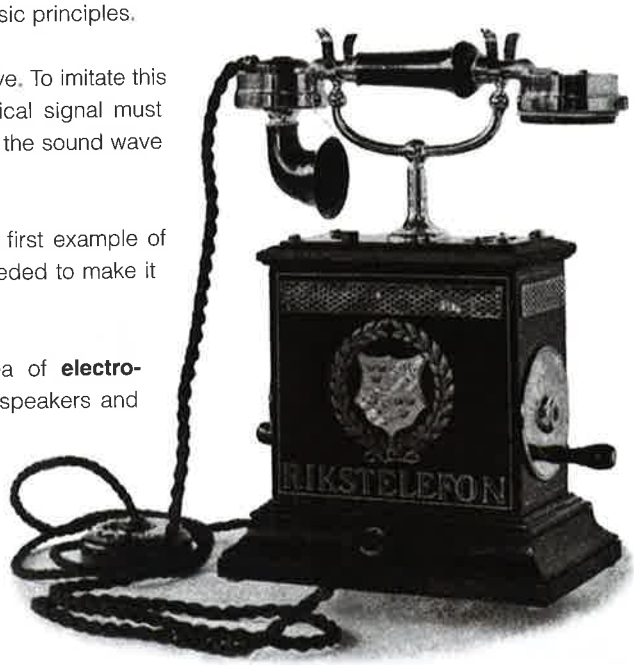
If you listen to sounds above about 85 dB for any length of time, damage to your ears will result. At about 130 dB sound levels start to be painful and damage to your hearing is likely without protection. This is worth keeping in mind when working in studios with drums and amplified music.

SOUND AND ELECTRICITY

▶ FROM SOUND TO ELECTRICITY

We are all familiar with the idea of singing into a microphone and listening to loudspeakers, and in this section we are going to look at how these work. You don't need to understand all the science and maths involved to grasp music technology, but it helps to have some idea of the basic principles.

When a sound is made, the air vibrates and creates a wave. To imitate this effect with electricity, the strength (voltage) of an electrical signal must change (go up and down in strength) in the same way as the sound wave is changing.



The telephone, invented in the 1870s, was probably the first example of sound technology. Many years of experimenting were needed to make it work.

Some of the first telephones were based on the idea of **electro-magnetism**, and this is still the way in which most loudspeakers and some microphones work.

▶ ELECTROMAGNETISM

An electromagnet is created whenever electricity passes through a wire. The wire becomes magnetic and will attract metal objects towards it. This effect becomes more obvious when the wire is coiled up and even stronger if it is wound around some metal.

The opposite also happens – magnetism can generate electricity. If a magnet is moved across a wire then electricity starts to flow in the wire. Again, the effect is much stronger if the wire is coiled and wound around a metal shape of some kind.

On the opposite page, you can follow how, using electromagnetism, a microphone converts a sound wave into an electrical signal and how a loudspeaker then converts that signal back into a sound wave.

SOUND TO ELECTRICITY AND BACK AGAIN

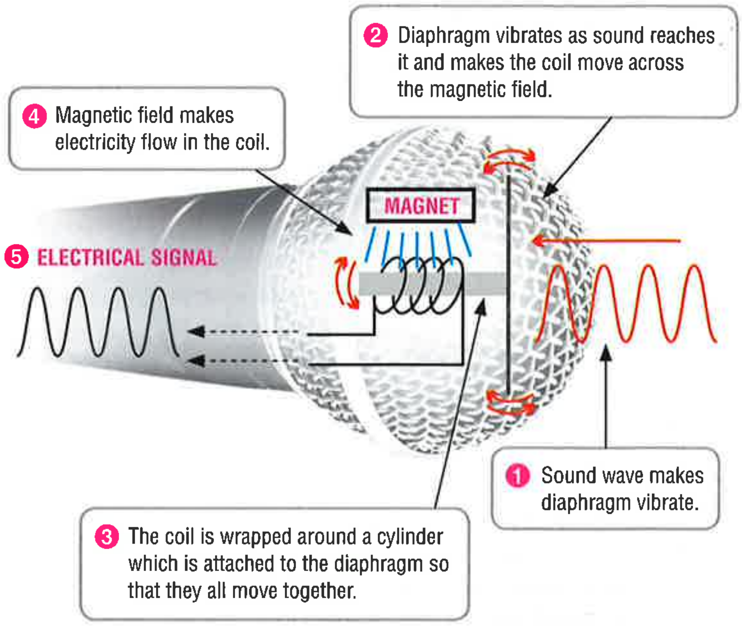
▶INSIDE A MICROPHONE – HOW SOUND IS CONVERTED INTO ELECTRICITY

Sound waves in the air ❶ hit a thin metal plate called a **diaphragm** ❷ inside the microphone.

This diaphragm is attached to a cylinder with a coil of wire wrapped around it ❸, and when it is moved by sound waves it makes the coil move rapidly across the field of a magnet.

The magnetic field creates a small electrical voltage in the coil ❹ which is continually changing as the sound wave changes. We usually call this flow of electricity a signal, or audio (sound) signal ❺.

In **Chapter 4** we will look at microphones in more detail.

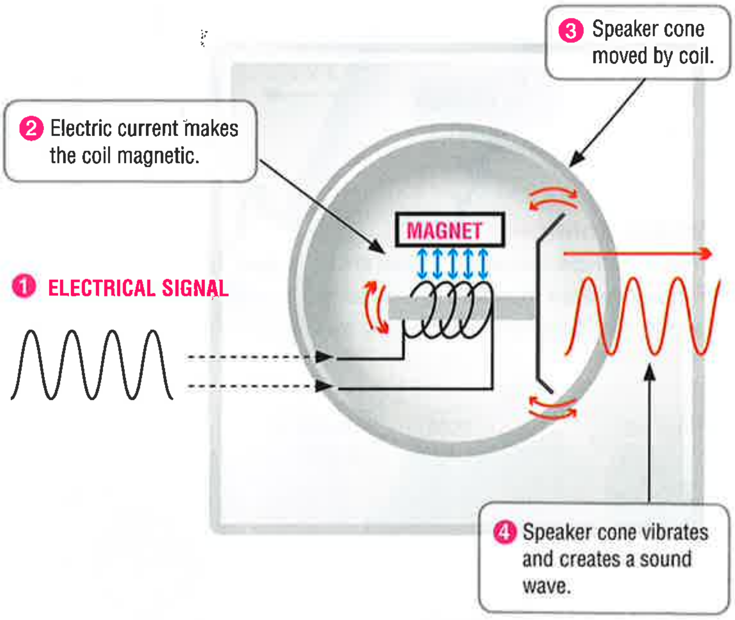


▶INSIDE A LOUDSPEAKER – HOW ELECTRICITY IS CONVERTED INTO SOUND

In a loudspeaker, an electrical audio signal ❶ is fed to the wires of a coil of wire surrounding a cylinder ❷. This makes the coil magnetic.

The amount of magnetism created varies rapidly as the electrical signal changes, and this makes the coil vibrate as it is attracted to a fixed magnet nearby.

The cylinder is attached to a cone made of card (or some other flexible material) ❸. As the coil vibrates it makes the cone move too, causing the air around it to vibrate and creating a sound wave ❹.



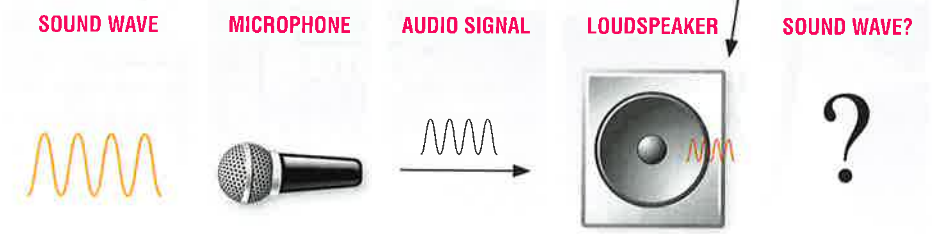
AMPLIFYING SOUND

These three diagrams show how a sound wave can be converted into an electrical signal and back to a sound wave.

1 DOWN THE WIRE

If a microphone turns sound into electricity and a loudspeaker turns electricity into sound, can we just join them together and send a sound along a wire?

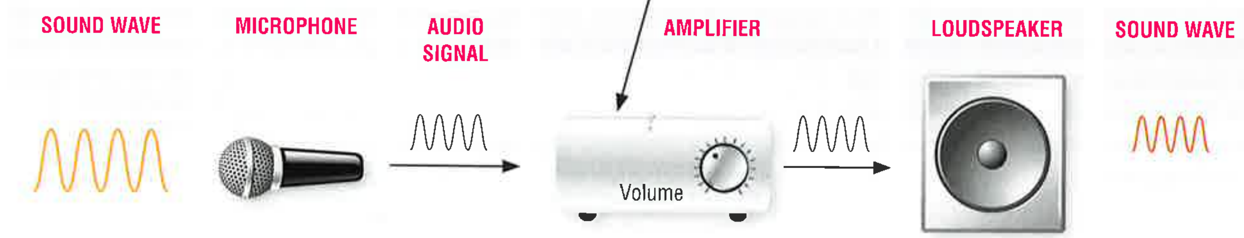
SIGNAL STRENGTH
The voltage from a microphone is extremely small – measured in thousandths of a volt. It isn't strong enough to produce a sound we could hear easily from a loudspeaker.



2 MAKING YOURSELF HEARD

To be of any use, the tiny electrical signal from a microphone needs to be made larger – amplified. So you need to plug a microphone into the input of an **amplifier** and then connect a loudspeaker to the output of the amplifier.

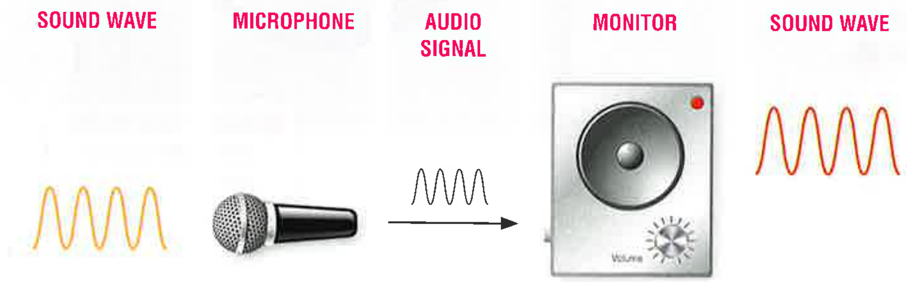
ELECTRONICS
In most modern amplifiers **transistors** are used. These are able to take a small voltage and change it into a larger one which is strong enough to make a healthy sound from a loudspeaker.



3 POWERED MONITORS

Sometimes a loudspeaker and amplifier are combined into one piece of equipment. These are known as **powered monitors** (or sometimes just **monitors**). They range in size from tiny desktop computer speakers to huge units that can fill a hall or stadium with sound.

POWER
When you use an amplifier it will usually say on the case how much power it can produce. This is measured in **watts** (named after James Watt who developed the steam engine). A small amplifier will produce less than one watt and the largest over a thousand.



MONO AND STEREO

So far we've looked at the idea of sound going into a microphone, becoming an electrical signal, being amplified and then coming back out of a loudspeaker.

If sound is being played through only one loudspeaker then it is described as 'monaural' – usually shortened to **mono**. A guitar amplifier is a good example of mono sound – one guitar plugs in and is played through one speaker.

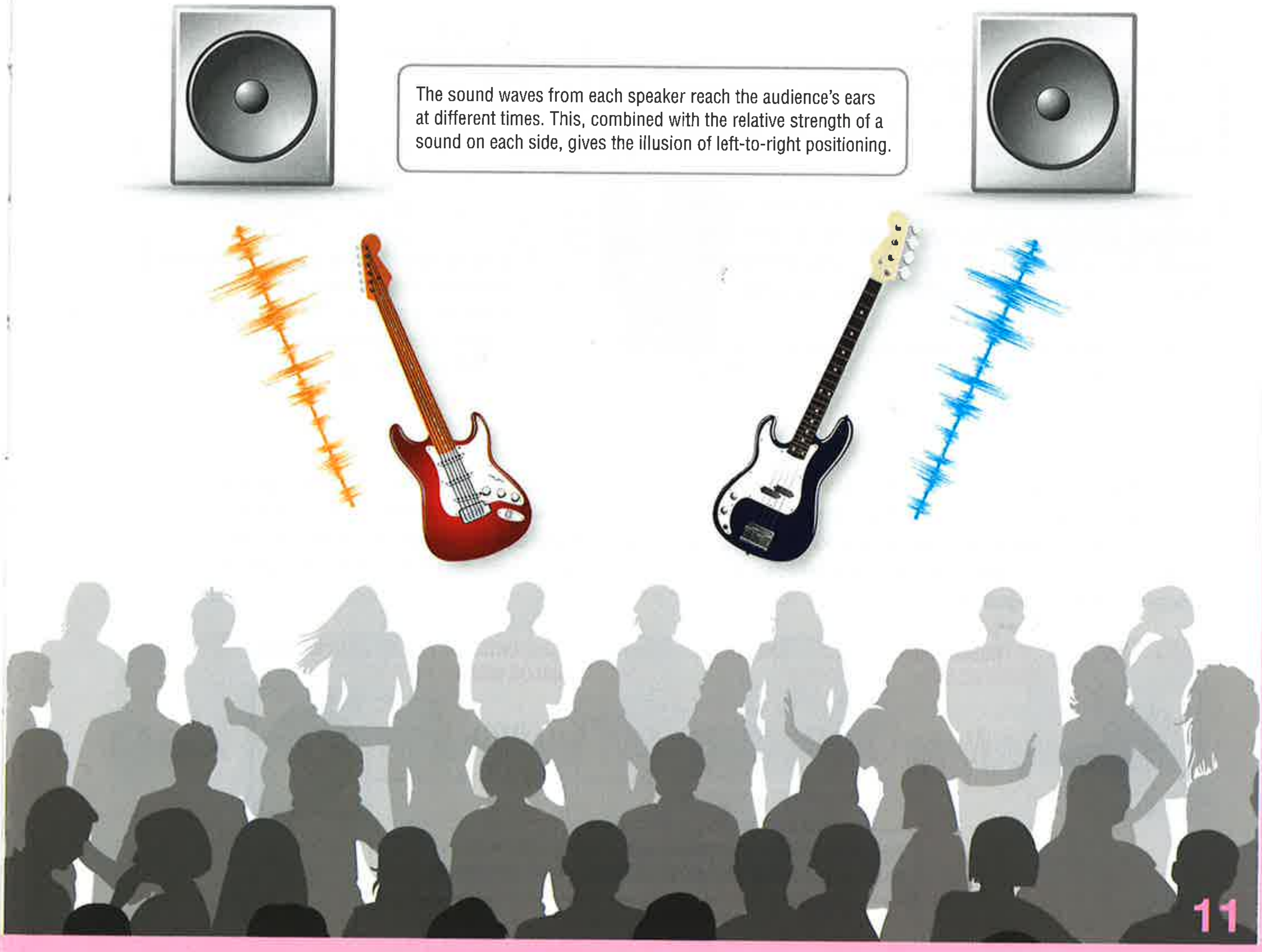
Almost all equipment we use these days to listen to music or speech is 'stereophonic' – abbreviated to **stereo**. This has two loudspeakers which give us a sense of left and right, an effect which is missing with mono sound. Most televisions, CD players, car sound systems and personal music players have two speakers or earphones.

▶ DOUBLING UP

Having two loudspeakers means that, to make a sound louder, you will need two amplifiers. In fact, any part of the circuitry in stereo equipment will need to be doubled up. Of course, we don't usually have a separate box of tricks for each side – the electronic circuits are duplicated on their boards, and two of everything is fitted into the same casing. The two circuits which run through a stereo amplifier are called the left and right channels.

▶ PANNING

When a sound is sent to a stereo amplifier, the **pan** control divides its signal between the left and right channels. If an equal amount of signal goes to both channels, the sound seems to be in the centre – it can then be shifted either left or right by sending more signal to one of the loudspeakers than the other. For more on **controls**, see page 134.



The sound waves from each speaker reach the audience's ears at different times. This, combined with the relative strength of a sound on each side, gives the illusion of left-to-right positioning.

ANALOG SOUND

▶WHAT IS ANALOG SOUND?

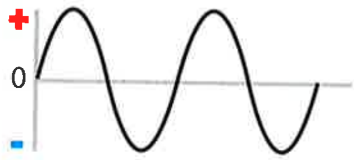
A microphone turns sound into an electrical signal. The changes in strength and pitch of the sound are reflected in the strength and frequency of the electrical signal, thus making a comparison or 'analogy' between them.

Common formats for analog recording are **vinyl records** and **magnetic tape**.

▶FEATURES OF ANALOG SOUND

- The changes in both sound and electrical signal are continuous
- If the sound becomes louder, the electrical signal becomes stronger
- If the pitch of the sound goes up, the frequency of change from positive to negative electricity becomes faster.

During playback, the continuous variation in electrical strength and frequency is changed back into sound waves by a loudspeaker.

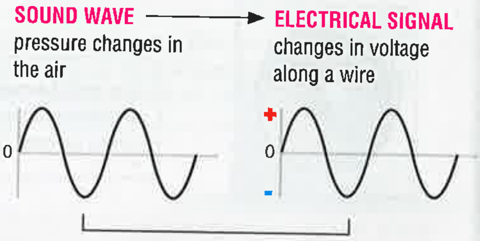


Changes in the signal are continuous

▶OTHER EXAMPLES OF ANALOG

In an analog thermometer, the temperature is shown by the mercury. If the mercury is high, the temperature is high, and changes are continuous. The position of the mercury line is analagous to the temperature.

Analog sound represents sound waves as electrical signals. Changes in pressure travelling through the air (the sound wave) are represented as changes in the strength and frequency of an electrical signal travelling along wires.



At each point in the cycle, the electrical signal is comparable with the sound wave.

Many devices in daily use represent the world in an analog form, such as a clock face, thermometer or a speedometer that has a needle moving across a dial. They form a 'picture' of the information we need to read.

▶BACK AND FORTH

Once sound has been turned into an electrical signal, it can be amplified or changed. If the electrical signal is increased or distorted, those changes will be heard when it is turned back to a sound wave. Guitar amplifiers sometimes add distortion to sounds, and occasionally have reverb effects built in. Effects are generally quite difficult to achieve with analog signals alone, needing either complex electronic circuits or sometimes a mechanical process to alter the signal as well.

The electrical signal is amplified and distorted as it passes through the amplifier.

DIGITAL SOUND

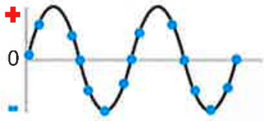
▶WHAT IS DIGITAL SOUND?

Instead of comparing the characteristics of sound to something else, digital sound measures it and represents it as numbers. By taking a measurement at regular intervals of pitch, volume, timbre and any other useful information, a mathematical description is built up of the sound. Each measurement is called a **sample**. This is a two stage process – the sound becomes an electrical (analog) signal first and is then 'sampled'.

Common formats for digital recording are **CD**, **DVD**, **DAT**, **MiniDisc** and **MP3**.

▶FEATURES OF DIGITAL SOUND

- The changes in the samples are not continuous but are in discrete steps
- As volume or pitch changes, the numerical information changes
- If a sound was only sampled a few times per second, the playback quality would be poor with lots of gaps, rather like the sound when you 'fast forward' through a CD or DVD.



Measurements (samples) are taken at intervals along the waveform

The number of samples per second is known as the **sample rate** and as it increases, we cease to notice the tiny chunks of sound. The amount of information measured in each sample is known as the **bit depth** or **resolution**. Digital sound for CD quality has 44,100 samples per second with a 16 bit resolution.

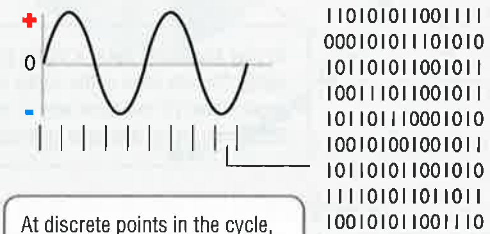
▶OTHER EXAMPLES OF DIGITAL

A digital thermometer 'samples' the temperature around it at regular intervals, so changes are not continuous. If the sampling was only once per hour, it would not be very helpful. Once per second would give a useful result. Digital devices do not form a direct 'picture' of information, but instead store a mathematical representation of it which can be displayed on a screen.

Each time the waveform of a sound is sampled, information about it is measured as a number, which can be stored. For playback, the numbers can be used to calculate the waveform again.

ELECTRICAL SIGNAL → NUMERICAL SAMPLES

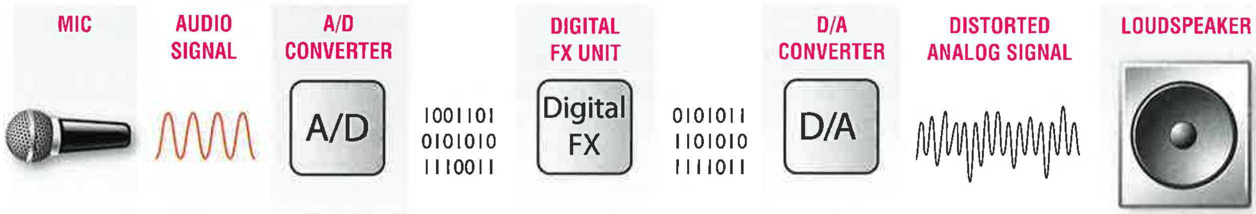
changes in voltage along a wire



At discrete points in the cycle, information about the electrical signal is analysed as a number.

▶BACK AND FORTH

An analog-to-digital (shortened to A/D) converter analyses the signal and produces the numerical information about the sound. This is done using a microprocessor (similar to a computer 'chip'). While the sound is in digital format, it can be changed by altering the numerical information. Distortion, reverb, delay and all popular effects can be added inside the processing circuit. A digital-to-analog converter (D/A) reprocesses the numbers back into an electrical signal, which can be played back through a loudspeaker.



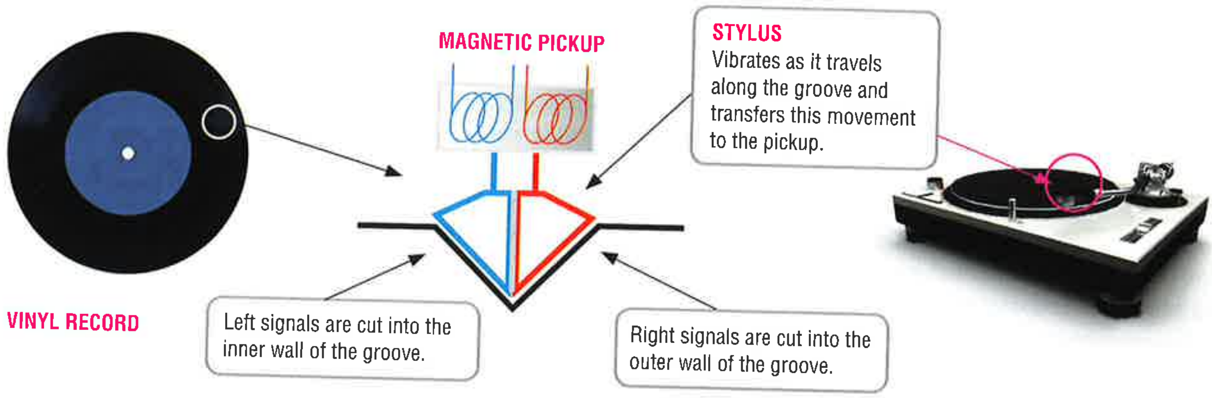
The digital information is altered as it passes through the effects unit – the resulting audio signal is therefore different too.

ANALOG RECORDING

▶VINYL RECORDS

To make a vinyl record, a groove is cut into the vinyl disc by an electromagnetic cutter that vibrates as the audio signal passes through it. The groove varies as the music changes in pitch and volume. For stereo sound, a v-shaped groove has the left channel cut into one side and the right channel cut into the other. The groove follows a spiral pattern starting on the edge and working inwards. The groove is 'analogous' to the audio signal, meaning that it changes in a similar way.

CROSS-SECTION VIEW OF A GROOVE IN THE VINYL



To play the sound back, a stylus (sharp point) travels along the groove as the record spins. The vibration of the stylus in the groove makes a coil move and generates a small signal (in the same way as a microphone does), which can then be amplified. For stereo sound a double stylus/pickup is needed.

Until the 1980s, vinyl records were the main medium for distributing 'Hi Fidelity' (high quality) music. Long playing records (LPs) had several songs on each side and later became known as 'albums'. Smaller records (singles) could have one song on each side. Early gramophones played back the sound mechanically, by picking up the vibrations of the stylus in the groove and transferring them to a 'horn' which in its turn made the air vibrate to create sound. These were replaced by record players with electrical pickups whose signals could then be amplified. The stereo vinyl record arrived in the late 1950s, adding greatly to the enjoyment of listening, and this became the common format from the 1960s onwards.

▶LIMITATIONS

- Noise can be a problem with vinyl records – the groove and stylus can become worn and any dust or dirt affects the playback of the sound
- The disc itself can warp – if it is not truly flat then the edges move up and down as it spins and the sound varies, producing an effect known as 'wow and flutter'.

Vinyl enthusiasts look after their records carefully. A lot of cleaning is needed, and very expensive and stable turntables have been developed for the best possible playback.

FACT
If you listen to tracks from the beginning of the 1950s or earlier, you will hear the noise and frequency range limitations of the recording methods of the times. Louis Armstrong and Elvis Presley are two artists with long recording careers. A comparison of their earlier and later work clearly shows the improvements in sound technology.

DIGITAL RECORDING

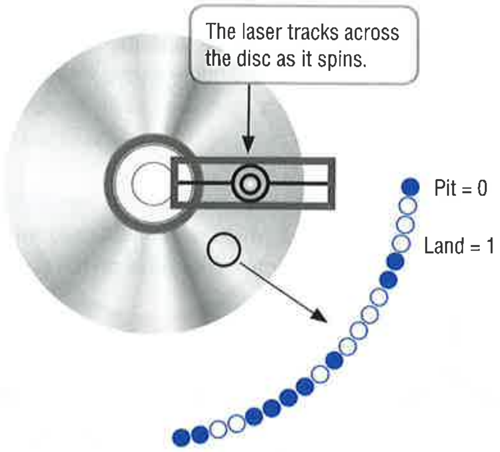
Instead of directly representing sound as a groove or changing magnetic field, digital recording simply stores the numerical information about the sound. The method of storage can vary – it can be a disc, a computer file or a personal media player. Any differences in sound quality that occur when the file is played on different devices will come from the digital-to-analog conversion circuits and the amplifier circuitry in the device you are using.

▶CD RECORDING

It's easy to put a blank CD into a computer and to 'burn' a disc – but what actually happens when we do this?

The numbers in a digital sound file are all either 0 or 1. This is called 'binary code'. The 0s and 1s are grouped together to make much larger numbers which a computer processor can decode. To store the numbers on a disc, a laser light burns a small mark (called a 'pit') into the surface every time there is a 0 and leaves a blank ('land') wherever there is a 1. These markings follow a spiral pattern starting from the centre and working outwards.

FACT
If the spiral series of markings on a CD was unwound to a straight line, it would be 5km long.



▶CD PLAYBACK

To play the sound back, a less powerful laser shines onto the disc and is reflected back to a detector. The reflection is different when the light bounces off a pit and so the detector recognises a '0'. A '1' is detected when it bounces off the flat, or lighter land. This means that the numbers can be read from the disc and then converted back to an audio signal by a processor similar to that in a computer.

▶COMPUTERS, PLAYERS AND DIGITAL SOUND

Most sound recording and editing now involves computers. Digital sound is stored as files on a hard drive or other memory system. Digital sound files are stored in miniature players, phones, answering machines and a host of other devices.

ANALOG AND DIGITAL

In the 1980s, the development of digital recording was so fast that the CD replaced vinyl records and tape as the medium of choice in a matter of a few years. Digital recording is virtually noise-free, and every disc is an exact copy of the master recording. With tape and vinyl, the copying process gradually introduces imperfections which build up as it is repeated.

For some people, though, digitally processed sound is too clean and clinical. Analog recordings have a perceived warmth and richness which some producers seek out, rebuilding old equipment and even running digital recordings onto tape and back again to capture the 'sound'.

All recordings have an analog stage. With the availability and affordability of modern equipment it is possible to use a mixture of analog and digital equipment in accordance with your own preferences.

MAGNETIC TAPE

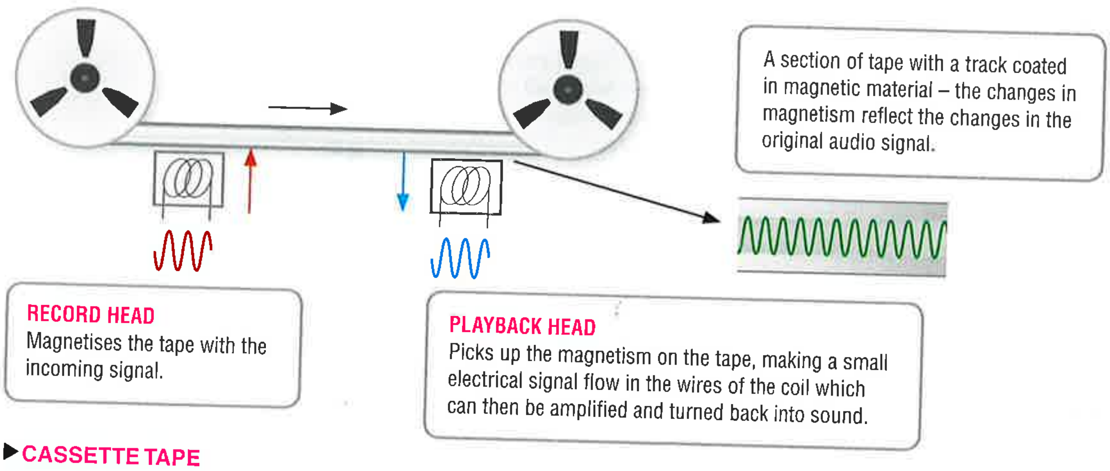
From the late 1940s, tape recorders developed rapidly and became central to the recording of music and sound for broadcast. Large tape machines such as the one pictured are known as 'reel-to-reel'. As the tape is moved between reels by the 'transport' mechanism, it is pressed against tape heads which record, erase or play back the sound.

A tape recorder was used to make the master recordings and for distribution these were transferred to vinyl records. Tape could be edited and re-used introducing a new level of flexibility to the recording process. Earlier in the 20th century, recordings were made directly to vinyl records, wax or tin foil cylinders in one take. Sometimes the musicians were asked to play faster so that the piece would fit onto the recording medium!

By the end of the 1950s, stereo tape machines were available and costs had come down to the point where home taping was practical, although the machines were still very large.

▶HOW DOES IT WORK?

Tape is coated in stripes of material that 'remembers' magnetic changes applied to it by the tape head. The head has a coil inside it which is connected to the electrical signal from the recorder – this creates a changing magnetic field which affects the material on the tape.



▶CASSETTE TAPE

In the 1970s, small cassette tape became a popular means of distribution and a serious competitor to vinyl. One great advantage of cassette tape was that you could copy your favourite songs to make a 'mix tape'. Very small players became possible, and the Sony Walkman started the personal media player revolution which continues with the digital MP3 players of today.

▶LIMITATIONS

- All tape recordings include some 'hiss' – the noise caused by background electrical and magnetic activity on the tape itself
- Tapes are also liable to stretch and break.

Tape recording machines became very sophisticated during the 1970s and 1980s, enabling synchronised multi-track recording at a very high quality. Although tape is still used by some, during the 1990s computer-based digital recording took over the recording, mixing and mastering functions of tape in most studios.

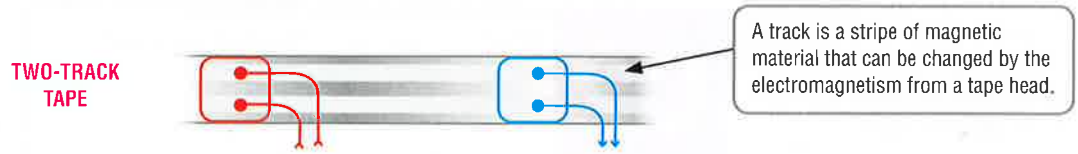
MULTI-TRACK RECORDING

▶ BEYOND STEREO

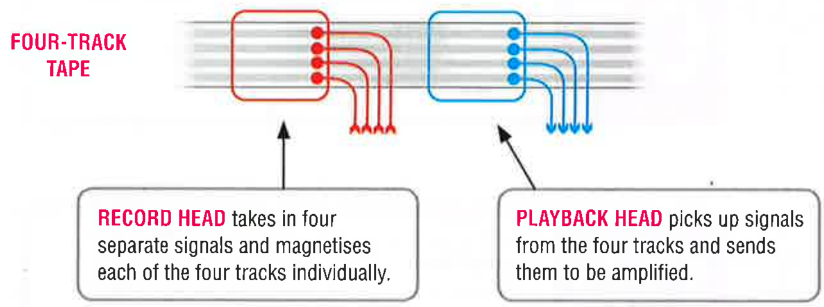
As we saw on page 11, two channels of sound will give us a stereo effect – left and right. For a long time this was the limit of recording technology. Performances had to be recorded 'live', with only one chance to balance and pan the sounds. Microphones would be set some distance away, to capture the whole performance.

The earliest multi-track recording equipment was developed in the late 1940s by Les Paul, also famous for his guitars. The technology progressed slowly, though, and it was not until the early 1960s that it was widely used.

Stereo tape has two 'tracks' of magnetic material which move across the tape heads:



The challenge early on was to fit more tracks onto one reel of tape. Two became three, and later standardised at four tracks for some time. As one track plays back it is possible to record onto another – a process known as overdubbing.



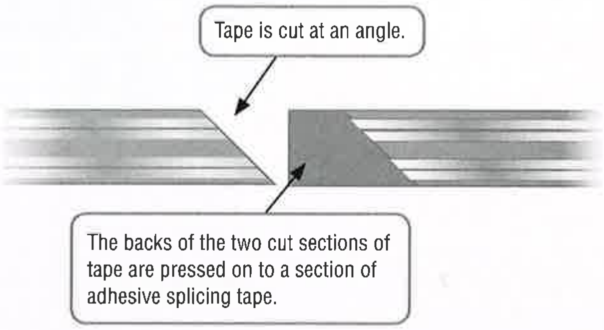
This step-by-step recording technique also allows for close positioning of microphones. The idea of a track of sound is now used to represent recordings visually on a computer screen.

During the 1960s, eight-track recording become possible, rising to 16 and even 24 as time went on. This involved very expensive and complex machines that needed careful handling, and could only be used in recording studios run on large budgets.

Among the first commercial artists to record using four- or eight-track machines were the Beatles and the Beach Boys during the 1960s. The Album 'Pet Sounds' (1966) by the Beach Boys cleverly uses eight-track technology to individually record and mix the voices.

▶ EDITING

Early recording technology was primitive and inflexible. It was necessary to record a performance in one take, and there was no possibility of correcting anything afterwards. Tape offered the possibility of editing. A performance could be recorded more than once and the best sections could be literally cut and spliced together. The angled cut creates a crossfade – as the join passes the tape head, the proportion of each section playing changes, avoiding a sudden change.



PROJECTS AND SUGGESTIONS

▶ TAPE AND BEYOND

You may be wondering what the point is of looking at tape, vinyl and other recording formats that you may never have used or even seen.

The main reason is that all audio technology that has ever been invented has left some kind of legacy. The language and ideas used in modern digital work often come from older technology. We still cut, splice and crossfade. We still use tracks and channels and even the labelling of sockets and controls on equipment still sometimes refers to 'tape'.

Apart from this, some understanding of basic science, electronics and technology will enable you to produce better recordings in the same way that understanding your car will make you a better driver.

Understanding of ideas such as sound waves and electronics can happen on many levels. In this chapter, the science has been presented in a basic way – for a greater understanding you may wish to find out more about some of the topics such as:

- Sound and waves
- Binary numbers and digital sound
- AC and DC electricity
- Basic electronics.

▶ TEST YOURSELF

01 What are the units used to measure volume of sound?

02 Which units are used to measure frequency?

03 What is the difference between a loudspeaker and a powered monitor?

04 Which units are used to measure the power of an amplifier?

05 How many channels of sound can a stereo amplifier process?

06 Digital sound sampling analyses a waveform a certain number of times per second – what is this number called?