Music

Sound Engineering and Production: Revised Concepts Glossary

[MULTI-LEVEL]

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Introductory note

Headwords in these five sets of glossary listings from Access 3 through to Advanced Higher are given in **bold type**. The explanation or definition of each headword is given in ordinary type. Within the explanation cross-references to other headwords are shown in *italic type*.

A cumulative index at the end of the pack lists all the headwords and shows their level (pp 53–6).
acoustic – The ‘sound’ of a room or space. The acoustic of any space is defined primarily by its size and the types of surfaces therein. These two characteristics in turn determine how a sound wave is dispersed within the space. A church, for example, is generally a large space with hard surfaces on the walls, ceilings and floors. A sound wave therefore takes a long time to disperse in such a space as the hard surfaces absorb very little of the wave’s energy and reflect it back into the room. But a domestic living room is a much smaller space and will have soft furnishings, curtains, etc., that will absorb more quickly the energy of the wave. See also reverb and ambience.

acoustic guitar – A six-string or twelve-string guitar that produces sound acoustically without the aid of electronics; although some may have pick-ups attached or built in. Acoustic guitars may have either nylon or steel strings. In the case of classical, or ‘Spanish’ guitars, the strings are nylon and give a much softer sound than steel-strung guitars which tend to be used more for rock, pop, jazz and folk music.

acoustic screen – A large panel of absorbent material that can be positioned to provide separation between musicians recording with microphones within the same space. This minimises leakage between instruments and microphones.

amplifier – Electronic device designed to take a very small-level signal and increase it to an audible level. See also pre-amplifier.

arrangement – The instruments used, the parts they play and the structure of a song or piece of music. A skilful arranger can take any piece of music and totally change its feel or tone by adjusting these variables and the piece’s tonality.

audio – Electronically produced or reproduced sound.

backing vocals – Vocal lines in an arrangement that are secondary to, but support and enhance, the lead vocal.

bass – The lower range of audible frequencies: nominally between about 20 Hz and 320 Hz.
**bass guitar** – Originally devised as an electric double bass from which it takes its open-string tunings (EADG), the bass guitar has become an instrument in its own right providing the bass parts in rock, pop and occasionally jazz music. While the standard instrument is four stringed, it is not unusual to see five-string basses with an additional lower (C or B) string, or even six-string instruments with an additional lower and upper string. Most bass guitars are *active* in that they require battery power for a circuit that controls the *tone* of the instrument.

**cardioid/uni-directional** – The predominant *pick-up pattern* of microphones. Cardioid or uni-directional microphones typically have a heart-shaped pick-up pattern (from Greek ‘cardia’ = heart) and are sensitive in only one direction. This means they have less sensitivity to instruments to the side or behind them. In studios this means that they pick up best the instrument they are pointing directly at and are less sensitive to other instruments in the room. In a live situation this means the same, with the additional benefit that being less sensitive to the outputs from loudspeakers means they cause fewer problems than other types of microphone with regard to *feedback*. See also hyper-cardioid and super-cardioid.

**CD** – Compact Disc. The standard optical, read-only format of consumer digital audio devised in the 1980s by Sony and Philips. A CD has a continuous track of digital data represented by pits burned into the surface of the disk read by a laser. A CD can hold up to 750 MB of information which equates roughly to 74 minutes of stereo audio at a *sampling frequency* of 44,100 Hz and a *bit depth* of 16. These standards for audio reproduction via CD are called the *red book standard*.

**channel** – On a mixing desk the channel is the series of electronic circuits designated to an input source. This is then duplicated a number of times to accommodate more inputs. A 16-channel desk therefore has 16 sets of the same circuitry to accommodate 16 different input sources.

**chorus (song structure)** – The part of a song that is normally repeated a number of times within the song. The chorus of a song may also contain the hook.
circuit breaker – Electrical safety device that automatically switches off the power supply as soon as it detects a problem or fault. Modern circuit breakers feature a trip switch which can be simply reset once the fault or problem has been rectified.

cconnector – The plug or termination at either end of a lead or cable. A number of different connectors exist in sound engineering, the most often encountered being the jack plug for general connections, the XLR for microphone and professional connections, and the phono or RCA plug for Hi-Fi, S/PDIF and non-professional connections.

ccontrol room – In a recording studio, the room where all the equipment and the engineer is situated.

count-in – The beats before the song or piece of music starts, to give the performers the start point and tempo. Like the click track (which may incorporate the count-in) this should be eliminated from the final mix.

distortion – The rasping, grating sound generated when an incorrect (too high) setting is used. While generally it is an undesirable effect, on some instruments, the electric guitar and the organ, for example, it has become a standard creative effect. See also overload.

drum kit – The group of drums and cymbals that have been pieced together and standardised over the years to create a drum kit includes a bass drum, snare drum, usually 2–4 tom-toms, a pair of hi-hats and at least one crash and one ride cymbal. Rock and fusion drummers have managed to take this to extremes, however, and it is not unusual to find kits that incorporate two bass drums, two snares, countless toms and cymbals, a gong and various other bits of kitchen hardware.

dry – A signal that has had no effect added to it.

echo – The physical reflection of a sound wave from a reflective surface which diminishes gradually in energy, thus getting quieter. Sound travels at roughly 340 meters per second (mps), so for a naturally occurring echo to be 1 second, the surface off which the original sound wave was reflected would have to be 170 metres away (170 m there and 170 m back).
**electric guitar** – A version of the *acoustic guitar* which derives its signal entirely electronically from a series of *pick-ups* positioned close to the steel strings (nylon strings won’t work due to the use of the electromagnetic principle – see *dynamic microphones* and *pick-up*). While electric guitars have jack sockets, their output is more like that of a microphone; therefore, when recording there are three preferred techniques:

1. positioning a microphone in front of the amplifier speaker
2. plugging the guitar into a *DI box*
3. using a guitar pod/processor which fulfils the roles of both the amplifier and the DI box.

**fade in** – When a track or piece of music increases in volume gradually from silence.

**fade out** – The opposite of a *fade in* – when a track or piece of music decreases in volume gradually to silence. This has become a widespread practice in *mixdown* technique as a tidy way of ending a song.

**gain** – Amplification. Gain is determined by the amount an electronic circuit amplifies the input signal. The gain control on any device is therefore, very, very important. Setting a gain too low will mean the engineer has to compensate for low-level signals by increasing output volumes. This results in increased noise levels. Too much gain, and the signal will *overload* the input circuitry and result in *distortion*. All recording devices have a gain control as part of the *pre-amplifier*. It makes sure the signals from all the different sources are at a suitable level for the following electronics as *mic-level* sources generally have a much lower output signal than *line-level* sources. The gain control evens them out.

**headphones** – A small *stereo* loudspeaker system that can be worn on the head or in the ears to allow isolated *monitoring* of signals. May be referred to as ‘cans’.

**hiss/white noise** – Electronically generated high-frequency noise. All audio devices will generate a small amount of hiss. It is the sound engineer’s job, through correct operation, to minimise this at all costs.
**hum** – Electronically generated low-frequency noise. Hum is usually the result of interference from mains cables or poorly earthed or grounded equipment. It is worth noting that only faulty or incorrectly wired equipment will generate hum.

**input** – The connection to send a signal into a device.

**introduction** – The first or opening part of a song. Often abbreviated to intro.

**jack plug** – A basic form of connector found on guitars, keyboards and mixers. The jack plug is normally a mono signal carrier, but can also carry a stereo or balanced signal if the plug has a Tip/Ring/Sleeve (TRS) configuration like that found on headphone jacks. While the jack plug is probably the widest utilised of all the connectors in sound engineering it has its pitfalls; its is, for example, a non-latching design so it can be easily and accidentally unplugged.

On mixing desks, guitars and keyboards, jack plugs are normally the larger 1/4 inch version although more and more the compact 3.5mm version is being utilised by manufacturers primarily to save space on increasingly smaller devices.

**lead/cable** – The wire that joins two connectors. In sound engineering it is standard to refer to the leads and cables by their termination connectors, for example an XLR to XLR cable would be referred to as either an XLR cable or a microphone (mic) lead. There are essentially three types of cable in sound engineering:

1. ‘screened’, unbalanced cable usually used to connect between guitars and keyboards, hi-fi and non-professional devices. Normally these terminate in either jack or phono plugs. The ‘screening’ is usually a braided cable that surrounds an inner core cable. The screen is connected to earth and the core carries the signal. This screening eliminates interference and hum.

2. Screened, balanced cable normally used for microphone leads and to interconnect professional and semi-professional equipment. These normally terminate in either XLR or TRS jack plugs.
3. Speaker cable used for connecting the output of amplifiers to loudspeakers or PA systems. Speaker cable must be unscreened. Using speaker cables for audio connections other than speakers will result in hum and possibly radio frequency interference.

lead vocal – The main vocal part or track in a song.

level – The degree of intensity of an audio signal.

loudspeaker – A transducer that converts the electronic audio signal back into a sound wave. Due to physical constraints, it is difficult for one loudspeaker to convert accurately the entire audio spectrum, so manufacturers use a number of drivers in a single cabinet to properly reproduce the full frequency range. Each frequency-range speaker is given its own name. High-frequency units are called tweeters, mid-frequency units squawkers and low-frequency units woofers. It is not uncommon to also get very low-frequency units called sub-woofers. It has become a common practice for manufacturers to incorporate amplifiers within the speaker cabinets. Such loudspeakers are known as active loudspeakers.

mains multiblock – A mains adapter that splits a single outlet socket into four or more outlets. Many today come with built-in circuit breakers for added protection and even anti-surge protectors which protect against sudden surges in voltage which could potentially damage equipment or crash computing devices.

mic stand – The heavy-based stand which holds a microphone. See also boom stand.

microphone – A transducer designed to convert a sound wave into an electrical current. Microphones are one of the most important elements of any signal path as they are the initial conversion point for any signal to be recorded. It is therefore important that high-quality microphones are used in recording studio situations. All microphones have their own ‘sound’ due to tiny fluctuations in their frequency response, and many microphones are manufactured for a particular purpose or even instrument.

minidisc (MD) – A digital, optical, record-and-read storage medium. Similar to the CD in operation, minidisc uses compression software to limit the material recorded in order to get all the information onto a very compact format. While minidisc recorders/players are generally 2 track devices, some multitrack versions exist.
mix(down) – The act and art of creating a balance of all the recorded tracks, processing where appropriate and necessary, and creating a *two-track*, stereo-mixed version of the music.

mono(phonic) – A single channel of *audio*.

noise – Any unwanted signal. See also *hiss* and *hum*.

output – The connection in an audio device from which its signal comes. Outputs should always be connected to inputs.

PA – Public Address. A PA system is a large-scale *louderspeaker* system designed to help musicians be heard clearly in a large space. A PA system is similar to a recording studio, but without the necessity for multitrack recorders as the mixed sound goes straight to amplifiers and loudspeakers. A sound engineer using a PA system will employ largely the same microphone, mixing and processing techniques that one would use in a studio.

phono plug – An *unbalanced* two-pin connector predominantly found in domestic hi-fi systems and non-professional equipment. Due to their normally being used in stereo systems, phono plugs tend to come in pairs coloured red and white. Generally the red connector will be used for the right-hand signal and white for the left.

record – To store a performance onto a medium so it can be played back or edited.

riff – A repeated rhythmic chord sequence within a song or around which a song may be based.

session – The time spent in a studio creating a recording.

session log – A note, usually formalised, of the activities carried out and completed within a *session*. Session logs are a great way of keeping on top of a recording project. Keeping a note of settings, microphone placements, even problems you have encountered in a session means you can always come back to the log in the future to reproduce the settings or overcome a similar problem.

signal – An electrical representation of a sound.
**stereo** – A two-channel audio system with the channels designated as left and right. Devised primarily because we have two ears, stereo reproduction of recorded sound has been the norm for many decades as it offers an excellent representation of what we hear acoustically. Any *multitrack* recording has to be *mixed* to stereo in order for it to be played on a standard domestic hi-fi system.

**studio** – The room in which the performers play in a recording *session*. Ideally, a studio will be soundproofed so noise cannot penetrate inwards or outwards, and also have an element of adjustment to its *acoustic*, perhaps a live sound at one end and a dead sound at the other (known as lede design). Only instruments that require microphones need be played in the studio. Instruments that can be *DI’d* can perform just as well in the *control room*.

**synthesiser** – Electronic instrument, usually keyboard based, that uses electronically generated waveforms through filters and processors to emulate (or synthesise) acoustic sounds. While most of these emulations of real instruments are at best approximate, synthesisers are capable of generating a wide range of sounds that no acoustic instrument ever could. Thus they have become an important element of modern sound production as an instrument in its own right.

**take** – The recorded performance of a part or track of a song. Standard studio practice has the performer do a series of takes and the best take, or a combination to make up the best take, will be used in the *final mix*.

**tape** – Linear magnetic storage medium. As tapes are a magnetic storage format, they are susceptible to strong magnetic fields. Care should be taken, therefore, to ensure tapes with information recorded on them are not placed in the vicinity of strong magnetic fields, such as *loudspeakers* or power supplies in case the information is wiped from them.

**tone control** – A basic form of *equalisation* on basic devices. The tone control will not have the sophistication of studio equalisers and will in general have only three controls – bass, mid and treble – to *boost* or *attenuate* a range of pre-assigned frequencies.

**track** – A single channel of recorded audio. Can also be the final finished product.
**track sheets** – Useful forms that keep a note of what instruments have been recorded on what *tracks*. Can be an essential part of a *session log*.

**trim** – Also known as the *gain* control, the trim control adjusts the level of a signal coming in to the *pre-amplifier* of a *mixing desk*.

**two-track recorder (2-track)** – A recording device that records on only two tracks, such as a cassette recorder, a minidisc recorder or CD recording device. Two-track devices are designated as *stereo* devices as the tracks are assigned to the left and right *master output* of the *mixing desk*.

**wet** – A signal that has had an effects process applied to it. See also *dry*. 
ambience – Similar to a room’s acoustic although generally the term only refers to small spaces. Ambience is the acoustic sound generated by a room. With the judicious use of microphone techniques, the ambience of a room in which an instrument is being recorded can be picked up and added to the direct sound of the instrument. This gives a greater sense of space within the recorded sound and can lead to a very natural-sounding stereo image.

attenuate – To reduce the level of a signal or series of frequencies (see equaliser).

balance – The relative levels of all the parts of a recording. A well balanced mix will have all the elements of the recording audible but no one part either dominating or masking the others.

boom (stand) – A microphone stand which has an additional boom on top of the standard upright, enabling better positioning of the microphone in relation to the instrument.

buss – A common connection of a number of different signals along which they all may simultaneously flow. A mix buss, for example, is a mixed output of a range of different signals. Mixing desks can have many different busses for different uses. An auxiliary buss can send a series of signals to a single-effects processor or amplifier and a mixing desk may have up to ten or more of these.

clean – A signal that is unaffected by hiss, hum, distortion or any type of effect.

click track – A metronome track recorded onto one track of the multitrack recorder to provide a guide tempo and count-in for the performers. Click tracks are usually generated electronically and so ensure that drummers don’t slow down or speed up. Care must be taken not to include the click track in the final mix of the music.

close miked – When a microphone is positioned between 2 cm and about 30 cm from an instrument, it is said to be close miked. Close miking helps to reduce problems with leakage from other instruments in the proximity, but can lead to other problems related to sound level and the proximity effect. It can also mean that performers may hit the microphone or that the microphone will also
pick up the sounds of the instrument being played (keys on a flute moving, for example). As with all microphone techniques, the potential problems have to be weighed up against the benefits.

cue – A mix of previously recorded tracks sent to the performer(s) to enable them to play along in time and in synchronisation. See also foldback and monitoring.

DI (box) – Direct Injection. Technique whereby the output of a device is plugged directly into the mixer without the intervention of a microphone. This is particularly useful for devices that have an electronic output – keyboards, for example – but can also be used with guitars and bass guitars with the intervention of a DI box. A DI box serves two purposes.

1. It can turn an unbalanced signal into a balanced one, making it less susceptible to interference.
2. It splits the signal into two or more outputs with an equal signal level at each output. A bass guitar could therefore be simultaneously plugged into a mixing desk and an amplifier via a DI box with no signal degradation.

direct sound – Sound that travels directly from the sound source (instrument/singer) to the microphone without reflecting off any surfaces. Where no reflections are desired, the engineer can increase the amount of direct sound utilising close miking techniques.

earth/ground – The common connection for all electrical and electronic devices to, literally, the earth or 0 volts. This is used as a safety feature, but in audio can also be the source of unwanted hum if a device is not properly grounded or earthed. This is also what the screen of a properly balanced cable is connected to, again helping with the elimination of interference of any sort.

effects unit/processor – A device that adds effects processes to any signal. Many of these are multi-effects processors and have programmes for a wide range of processes and even programmes for a number of simultaneous processes. For example, a single unit may be able to add equalisation, compression, chorus and reverb to the input signal.

equaliser – The tone control. Equalisers split the full range of audible frequencies into up to four manageable ranges: low frequency (LF), low-mid frequency (LMF), hi-mid frequency (HMF) and high frequency (HF). This gives a greater diversity of control over the
entire frequency range for both corrective and creative purposes. There is a range of different types of equalisers for different roles in audio; all are useful, all are potentially damaging to the signal, so equalisers should be used sparingly.

**fader** – The linear sliding control that adjusts the channels output. A fader is not a volume control, it is a variable *attenuator*. When the fader is fully down, it is at maximum attenuation, and when it is fully up, it is at minimum attenuation. The signal, therefore, is always present; the fader just determines how much of the signal is allowed to pass through. This can be depicted as similar to a sluice gate in a lock. While the gate is shut or down, no water is allowed to flow. When the gate is raised, the water may flow. Opening the gate further lets more water flow.

**final mix** – The version of the *mixdown* that will actually be submitted as a stereo *master*. The final mix features a balance of instruments that all involved are happy with, additional effects that enhance the overall *production*, and perhaps the application of some *dynamic processors*, usually *equalisers* and *compressors*, to the overall mix.

**flat** – A signal that is unaffected by *equalisers* or *filters* and therefore has no peaks or troughs in its frequency range and could be represented graphically as a flat line.

**frequency** – The number of times an event happens in a pre-determined time scale. For example, if a sound wave is said to have a frequency of 440 *hertz*, this means that the wave repeats itself 440 times in one second.

**guide vocal** – A vocal track that is recorded in the early stages of the project to give the performers an indication of the progression of the song. This will generally be replaced later in the project by a more carefully performed and recorded *lead vocal* track.

**hard disk recorder** – A recording device, either *two-track* or *multitrack*, that stores the digital recorded information on a computer-based hard disk drive.
**headroom** – A safety margin between the signal level and the amount of signal the device can handle before *distortion* occurs. While it is always important to have the maximum possible level going into a device to ensure it is functioning properly, it is good practice to reduce this input level slightly to give some headroom. Performers are notorious for setting a level and as soon as the sound engineer presses record, they invariably play louder.

**hertz (Hz)** – The unit of measurement of the *frequency* of sound waves: 20 Hz, for example, means 20 repetitions or cycles of the wave in 1 second.

**indirect sound** – Sounds picked up as reflections of an original *direct sound*. Indirect sound and reflections are very slightly delayed from the original direct sound and are the primary features of *ambience*.

**master fader** – The linear fader on a mixing desk that determines the overall output level. If performing a *fade in* or *fade out*, it is the master fader that should be used.

**mixing desk/mixer** – The device at the heart of a studio set-up through which all the signals can be *routed* for recording, processing or monitoring. Mixers are defined as a set of input channels and a set of output channels – *subgroups* and *master* – so a desk that has 24 inputs, 8 subgroups and a stereo master output is defined as a 24–8–2 desk. The number of input channels will always be the most numerous. In a studio, the mixer serves four main functions: matching the input signals from *mic level* and *line level* sources; processing, by means of built-in *equalisation* circuits; *routing* signals to outputs; and collating signals into a mixed output.

**monitor** – To listen to the signals either through the main monitoring or *loudspeaker* system, or through a secondary system, such as headphones.

**multitrack** – Multitrack recording devices have two or more tracks with the ability to *monitor* or *cue* one track while recording on the other. This allows the process known as *overdubbing* whereby a single musician can build up a song by performing each of the parts one after the other. Recording each instrument onto its own track also allows the sound engineer a great deal of control over each track. An *equalisation* setting, for example, can be added to one track and another setting to another track and so on. Multitrack recorders come in many formats these days from 4-track devices to 24-track
devices and computer-based hardware and software systems that feature almost infinite multitrack recording capabilities.

**mute** – See cut.

**omni-directional** – Microphone *pick-up pattern* describing equal sensitivity in all directions (omni = all). Omni-directional microphones have limited practical use except for picking up the *acoustic* of a hall in live recordings. Due to the fact that they have all-round sensitivity, omni-directional mics should never be used in live PA systems otherwise *feedback* will occur.

**overdrive** – See overload.

**overdub** – In *multitrack* recording, the act of playing a new track of material in *synchronisation* with one previously recorded.

**overload** – The *distortion* that happens when a signal exceeds the stated input level. Meters are generally provided to ensure that sound engineers do not exceed these levels. It is always a good idea, however, when running at maximum level, to turn down the *gain* slightly to get some *headroom*. Overload, like distortion, is often used as an effect on guitars whereby the output from the pre- amplifier is turned up to maximum in order to overload the input stages of the amplifier. Guitarists tend to refer to this as *overdrive*.

**pan(ning)** – The pan control serves two functions:
1. In a mix it places a *mono* signal in the stereo sound field from left to right.
2. In **tracking** it works in conjunction with the **routing** switches to determine which tape output the signal will be sent to. Panning to the **left** will send the signal to the **odd** numbered outputs and panning to the **right** will send it to the **even** numbered tracks.

![Diagram](image)

**peak** – Maximum level of any signal.

**pick-up** – A transducer found on **electric** and **bass guitars** that translates the vibrations of the strings into a varying current in a similar fashion to the **dynamic microphone**.

**pick-up pattern** – Term applied to the shape of the sensitivity pattern of a microphone. The most predominant are **cardioid**, or heart shaped, **omni**, sensitive in all directions, and **figure-of-eight** or bi-directional, sensitive to the front and back, but not the sides. Each pattern has its applications in recording and live sound, but most microphones are **cardioid**. High-quality studio-condenser microphones may have a switch which makes them multi-pattern as they can switch between all the main pick-up patterns, increasing their versatility. The pick-up pattern may also be referred to as the polar pattern.

**popping/blasting** – The explosive sounds in singing and speech that cause audible pops and thumps in a recorded vocal. These can be effectively reduced using a **pop-shield**.

**pop-shield** – A thin mesh positioned in front of a microphone that disperses the focused blast of air from explosive consonants (P and B) in speech and singing that can cause problems in a recording.

**remix** – The art of taking an original mix(es) and material and re-organising it or changing settings to change the feel, sound or structure.
reverb – The natural series of very short and dense echoes of a sound that occur in a confined space such as a room or a hall. While echoes with a longer delay would be discernible, in reverb the echoes happen so fast and are so dense, it is impossible for the listener to hear individual repeats. Reverb is the essence of natural sound. Listening to a close miked instrument is like having the instrument play in your ear in a very small room. The addition of reverb to a sound makes it appear as if the instrument is being played in a real acoustic. Nowadays reverb can be emulated digitally very easily and nearly all effects processors have a wide range of reverb types for different applications. See also gated reverb.

solo – A mixing desk control that mutes all other channels in order to isolate the soloed channel for monitoring. The solo function is also usually the control that aids the initial set-up and gain setting of a channel. On live mixing desks in particular, the solo function may also be referred to as Pre-Fade Listen (see PFL).

stage monitor – Normally a wedge-shaped loudspeaker cabinet pointed at the performers that delivers a foldback or monitor mix so the performers can hear themselves.

stereo master – The final mixed recording of any project. As most replay systems are stereo, the multitrack recording has to be mixed down to a two-track master in order for it to be replayed.

sweet spot – When positioning microphones, the position at which the sound is best. Also the ideal position for the listener between a set of loudspeakers to get the best possible sound.

talkback – A system, like an intercom, to enable the sound engineer or those in the control room to talk to the performers in the studio. Talkback systems usually consist of a small microphone built or plugged in to the mixing desk with a switch that sends the signal from the mic through the foldback or monitoring system.

time-domain effects – Those types of effects processes that change the time characteristics of an input signal by adding to it delay, reverb, chorus, phasing or any of the delay or reverb-related effect variations.

tracking – The act of recording the individual parts of a project onto a multitrack recorder. See also production.
**verse** – The part of a song after the *introduction* and before the *chorus* and then repeated between choruses. The lyrical content of the verse tends not to be repeated, but holds the story or narrative of the song.

**wah-wah** – Effect that uses a narrow band *filter* swept across the audio spectrum giving the descriptive wah-wah sound. While generally this is a guitar-pedal-based effect, it can be utilised on just about anything. Much used in funk music.

**windshield** – Foam shield placed over the top of a microphone to protect it from interference from wind. Generally only used outdoors and not to be confused with a *pop-shield*.

**XLR** – Three-pin, latching professional audio connector for *balanced lines*. The type of connector used for *microphones*. 
**AB comparison** – Technique whereby an engineer can set up two (or more) different devices or settings and switch between the two to hear any differences. For example, two different microphones can be positioned in front of a musician and the engineer can, using AB comparison, choose the one he/she prefers.

**AFL** – After Fade Listen. A switch system on mixing desks to let the engineer monitor a signal being sent after its level has been determined by the fader. This is mainly used for checking the output-to-effects processors from a post-fade auxiliary send.

**analogue** – A device that utilises a changing voltage or current to represent an acoustic signal.

**autolocate** – A function on recording devices that can store positions throughout the duration of a project. This enables the engineer to jump to these points automatically once they have been stored.

**auxiliary send/return** – A mixing desk function that allows a signal or group of signals to be sent to a separate output – an auxiliary output – for either monitoring or processing. In the case of monitoring a pre-fade send will be used. For effects processing a post-fade send will be used and the signal with the process added to it will then be returned to the mixing desk.
boost – To increase the level of a signal or series of frequencies (see equaliser).

bridge – Passage connecting two other passages of a song, for example between the chorus and verse.

CD-R – Recordable Compact Discs. A write-once-only format of the CD. CD-R’s have become one of the widest used storage formats in the world for both audio and other data. For a few years now it has been both cheap and simple to incorporate CD burning devices on home computers. Like the CD, a CD-R can hold up to 750 MB of information.

CD-RW – Re-Writable Compact Disc. A write-many-times version of the CD-R which gives the user the opportunity to re-write over old information previously recorded onto a CD. It is important to note that CD-RW’s cannot be used for audio recording as they are a data only storage medium and cannot comply with the red book standard.

centre frequency – The frequency around which the attenuation or boost of an equalisation filter is centred. While in any filter a range, or bandwidth, of frequencies is affected by boosting or attenuating the filter, the bandwidth curve always has a centre. For example on a graphic equaliser, the frequencies assigned to each control are actually the centre frequencies of the filter.

chorus (effect) – An effect whereby short delays and slight modulations are added to a signal to make it sound as if there is more than one player. It therefore applies a detuning effect which can be detrimental to some instruments (for example, the acoustic piano) but can be very effective on others (for example, the electric guitar).

compressor – A dynamic processor that can automatically control the gain of a signal. Once the incoming signal has reached a predetermined threshold, the compressor reduces the output of the signal by an amount determined by the ratio control. Effectively this is like a fraction; so if a ratio of 2:1 is set the amount of signal above the threshold will be halved; a ratio of 4:1 means it is quartered and so on. Compressors also have an attack control which determines how quickly the compressor reacts and a release control which determines how quickly the compressor stops compressing once the signal has gone below the threshold again.
condenser microphone – Also sometimes referred to as a capacitor microphone, these microphones measure the capacitance between two charged plates to gain their signal. This signal is typically very small, so condenser microphones require a pre-amplifier to give this signal a boost prior to its going to any inputs. Condenser microphones therefore require a power source. This may be from a built-in battery, but more often it is from phantom power, a voltage from the mixing desk that flows down the microphone cable. Condenser microphones are of a consistently high quality with a very wide frequency response and are excellent for studio use; however, they can be fragile and are therefore not generally particularly suited to live sound applications.

cut/mute – To turn a channel or a track off totally. Cutting and muting are mainly used in mixdown to either eliminate unwanted parts of a track or reduce the noise from an unused channel or track.

delay – The interval between an original signal and its repetition. While this is achieved electronically, it is similar to, and is used practically, as echo. Modern digital-delay processors can repeat the original sound forever and with almost an infinite initial delay time.

digital – An electronic representation of analogue sounds that utilises 1’s and 0’s. See also A/D converter and D/A converter.

dynamic microphone – A microphone that generates a signal using the electromagnetic principle of moving a coil in a magnetic field. The microphone’s diaphragm is connected to the coil which in turn is suspended in a magnetic field. When the diaphragm vibrates, the vibration is transferred to the coil and the current is induced in the coil giving the signal. Dynamic microphones can cope with very high volumes of signal and can also handle rough treatment. They do, however, tend to have a slightly limited frequency response. While they can be very useful in the recording studio, they are invaluable for live use.

dynamic range – The range of volume levels in a piece of music or in the output of an instrument. The dynamic range can be reduced for control purposes by using a compressor.
effects loop (FX) – A loop of information whereby the signal, or group of signals from the mixing desk are sent to an external device which adds its own processes and then returns the signal back to the mixer.

feedback
1. Acoustic feedback occurs when the output of a loudspeaker can be picked up by a microphone that is being amplified by the same speaker system. It is characteristically a high-pitched squeal.
2. Positive feedback occurs, similarly, when an output is fed back to its own input.

filter – An electronic circuit designed to boost or attenuate a designated range of frequencies. See also equaliser.

foldback – The speaker or headphone system that lets musicians hear themselves and others either on stage or in a recording studio. Foldback mixes can be separately controlled by the sound engineer from the mix that is being heard in the control room or in the venue. These mixes are controlled by the pre-fade auxiliary sends on the mixing desk and can be tailored for each performer. In live situations bands may have either loudspeaker-based foldback systems or, nowadays, in-ear-based systems. In-ear systems have many advantages in that they don’t have to be extremely loud and they allow the performer to move around the stage without having to position themselves next to a foldback speaker.

frequency response – The range of audio frequencies that a device is sensitive to. The frequency response of a condenser microphone may be between 50 and 20,000 Hz. This means it is not sensitive to any frequencies below 50 Hz or above 20,000 Hz.

impedance – The resistance of an electronic circuit or component to an AC current. In audio engineering the AC current is an audio signal, and, unfortunately, impedance varies with the frequency of the signal fed through the circuit or component. Basically, audio devices are either high impedance or low impedance and a high-impedance device or output shouldn’t be plugged into a low-impedance device or input and vice versa. With the correct use of connectors and pre-amplifiers such connections and errors should be virtually impossible.

I/O – Abbreviation for Input/Output. Sometimes also I/P and O/P.
key change/modulation – The part of a song where the key changes, usually upwards, to lift the song and maintain interest.

leakage – The overspill from one instrument into another instrument’s microphone. This will only occur where more than one instrument is being simultaneously miked up in the same room. Leakage can be minimised by using directional microphones and acoustic screens; but it is difficult to totally eradicate it. There may also be leakage from a pair of headphones if the monitoring or foldback volume is turned up particularly high. It is not unusual, for instance, for a click track monitored through headphones to leak into one or more of the drum microphones. Also known as bleeding or spillage.

line level – The output from a purely electronic source, a keyboard for example or any processing device. The actual output level is set by the manufacturer to industry standards depending on the standing of the equipment as ‘semi-professional’ (–10 dBV) or ‘professional’ (+4 dBu).

masking – Problem in a mix where the level of a track or tracks is such that other tracks and instruments cannot be heard.

mic level – The level or voltage of a signal produced by a microphone. Typically mic-level signals are considerably lower than line-level signals, so a pre-amplifier must be used to boost their output. In some condenser microphones, the output of their built-in pre-amplifier is high enough not to require any more boosting.

middle 8 – The part of a song where a new or altered piece of music is introduced, usually after the second chorus. While it is normally eight bars long – hence its title – it can be much more or even much less.

MIDI – Musical Instrument Digital Interface. A digital language that enables devices to talk to one another in a standardised format. While MIDI was originally devised for keyboards and musical instruments, more and more effects processors and devices are responding to it and may be programmed using MIDI.

MP3 – Moving Pictures Executive Group Level-1 Layer 3. A digital data compression format that reduces an audio file to around one-tenth of its original size. Devised primarily as a fast and efficient method of downloading large audio files via the internet, MP3 players have become popular due to their compact size, high quality and huge
storage capacity. It is now possible to have an entire CD collection stored on an MP3 player the size of a mobile phone.

**mS** – Abbreviation for millisecond or one-thousandth of a second; thus 500 mS is half a second. Most effects processors use the mS as the time constant for programming.

**noise gate** – A signal-activated switch. If a signal reaches a preset threshold, the noise gate opens and allows the signal to pass through. If the threshold is not met, the gate stays shut eliminating any lower-level noise or hiss. Gates are very effective and useful devices in the studio, operating as automatic mutes or cuts to reduce low-level background noise while recording using microphones.

**patchbay** – A device that localises all the input and output sockets of a range of studio devices so that they may easily be interconnected using patchleads.

**patchlead** – The short leads that allow two sockets on a patchbay to be connected.

**PFL** – Pre-Fade Listen. See *solo*.

**phantom power** – A voltage (up to 48 v) sent down the microphone cable from the mixing desk in order to power a condenser microphone. While this power source normally comes from the desk, stand-alone phantom power units are available for situations where a desk is not being used. Phantom power can only be sent down balanced microphone cables.

**pitch bend** – A control message on keyboards designed to change the notes pitch in relation to a performance wheel or lever. The term may also be applied to the guitar technique that bends the strings in order to change the pitch of the note played.

**PPM** – Peak Programme Meter. A segmented bar-type meter designed to register peaks in a signal rather than just an average level.

**pre-amplifier** – The first stage of amplification of a device, normally a microphone. Condenser microphones have pre-amplifiers built into them, but the input stages of a mixing desk are also referred to as the pre-amp and it serves to increase a mic level signal to that required by the electronics of the mixing-desk channel. A pre-amplifier is also the input stages of a guitar or bass guitar amplifier, housing the
equalisation section prior to the signal being passed on to the main power amplifier.

proximity effect/bass tip up – A low-frequency boost that occurs in cardioid microphones when they are placed particularly close to the sound source. This unnaturally colours the sound and can be detrimental to the overall signal. However, in some live situations it can help lift a vocal out of the mix slightly.

punch in/out – drop in/out – A technique in multitrack recording that lets a performer record over mistakes or change parts previously recorded by punching or dropping in and out of record mode while the machine is in playback. Punching or dropping in can be performed by an engineer pushing the right buttons at the right time, the performer hitting a foot switch at the required point, or by advanced use of the machines autolocate functions whereby the multitrack recorder can be programmed to drop in and out of record mode automatically.

reflection – The parts of a waveform which reach the listener or microphone after bouncing off a surface. See also reverb and direct sound.

routing – Sending a signal to an output. Mixing desks have many types of output for many different purposes: auxiliary sends, tape and subgroups, outputs, etc. Routing is the process by which the engineer can take any input signal, and using either controls or a patchbay can send that signal to any or all of those outputs for processing, recording or monitoring. See also panning and buss.

shock mount – A moving suspension mount for a microphone. Many microphones, especially condenser microphones can be susceptible to movements or vibrations that travel through the floor and up the mic stand. A microphone placed in a shock mount is isolated from these vibrations.

sibilance – High-frequency (normally between about 5 kHz and 10 kHz) lisping or spitting noise on vocal recordings that occurs on ‘s’ or ‘sh’ sounds. Sibilance is usually caused by bad microphone technique or over use of equalisation. While it is predominantly an issue on vocal tracks, it can also be heard on cymbal tracks. Eliminating sibilance should be attempted at source; however a device called a de-esser may be employed to remove the problem.
**signal chain/path** – The route that a *signal* takes through an audio system from input to outputs. The route may be simple, such as a microphone plugged into an amplifier and loudspeakers plugged into the amplifier to create a basic PA system. But in the case of a professional recording studio, it can be very complex, involving large numbers of processors and *monitoring* systems. It is important for the sound engineer to understand each of the different routes any signal may take in order to correctly connect and operate the equipment.

**signal-to-noise ratio (S/N ratio)** – The ratio of maximum *signal* level to any residual noise present. The S/N ratio is expressed as a number of decibels, so the higher the number of dB indicated in the specification of a piece of equipment the less noisy it is, and the lower the number, the noisier it is.

**spillage** – See *leakage*.

**splitter** – A device that splits a signal into two or more separate signals without signal degradation. Such devices are useful in large-scale PA systems where more than one *mixing desk* may be used, but get their signal from a single microphone. See also DI box and Y lead.

**squawker** – One of the speakers in a *loudspeaker* cabinet that handles only mid frequencies.

**subgroup** – An output channel on a *mixing desk* that controls overall a small number of *input* channels. Subgroups in modern mixing desks have two main functions:

1. In a recording situation, they can be assigned as the outputs to tape tracks, so a number of instruments can be routed to a single tape track;
2. In a live or *mixdown* situation a number of instruments assigned to a subgroup can have their overall volume increased or decreased with a single *fader* movement rather than having to adjust a number of different levels; for example, a drum kit may have as many as eight microphones. Assigning these eight
channels to a single subgroup means only the subgroup fader need be moved to adjust the overall drum-kit volume. See also routing.

**sub-woofer** – The speaker in a *loudspeaker* system that handles the very low frequencies. In actuality these speakers operate at frequencies that can be felt rather than heard, i.e. those below 20 Hz.

**sweep** – The control on a *parametric* or *semi-parametric equaliser* that determines the *centre frequency* of the filter.

**synchronisation (sync)** – When two or more tracks or devices play at the same time, in time.

**texture** – The manner in which the different parts of a recording are woven together. If a piece is essentially a melody supported by a chord accompaniment, the texture is said to be harmonic; if the melody is accompanied by other melodies, then the texture is contrapuntal or polyphonic. The instrumentation used in an arrangement may also change its texture.

**transient** – A short, loud signal with a very fast *attack* and *decay* time.

**tweeter** – One of the speakers in a *loudspeaker* cabinet that handles only high frequencies.

**VU meter** – Volume Unit meter. These are analogue voltage measurement meters with a flickering needle display calibrated to show the relative volume of the input signal. VU meters are found on old equipment and new equipment that has retro styling; however, many engineers prefer their metering because it is less harsh than the *PPM*. 
**woofer** – One of the speakers in a *loudspeaker* cabinet that handles only low frequencies.

**Y lead** – A lead that has one connector at one end and two at the other in order to split the signal into two. While this technique has its uses, the split signal will be degraded somewhat. It is far better to use either a *splitter* or a *DI box*. 
**accent (or ‘spot’) microphone** – A microphone positioned at a single instrument or group of instruments in a multiple or stereo mic array in order to pick up the instrument separately from the signals received at the other microphones. A technique largely used in classical recording whereby a general *stereo pair* of microphones picks up the orchestra while the accent microphone is positioned at the solo instrument.

**active** – A circuit which requires power to operate. A *bass guitar*, for example, may have an *equaliser* and switching circuit which requires battery power. Circuits that do not require power to operate are known as *passive*.

**ADSR** – *Attack, decay, sustain, release*. The four primary elements of the *envelope* of a sound.

Here we can see the *envelope* of a short percussive sound (a snare drum). It is graphically represented using the elements of time and gain. As a short loud sound, the *attack* time is sharp and it reaches maximum gain quickly. The following *decay*, *sustain* and *release* times are short too. The sound dies to nothing very quickly.
Compare this with the envelope of a violin. Here the *attack* time is much longer and, as the bow can move continuously over the string, the sustain is also much longer. The *release* time is long too, as it takes a while for the string to stop vibrating once the player has stopped playing.

**amplitude** – The ‘height’ of a waveform. The amplitude determines the volume of the wave.

**attack**
1. Part of the *envelope* of a sound (*ADSR*). The attack time is the time it takes for an acoustic sound to reach its maximum initial *amplitude* or volume. Percussive sounds will, by nature, have a very fast attack time, whereas bowed sounds, a violin for example, take a longer time to reach maximum volume, and so have a slower attack time.
2. The control on a *dynamic processor* that determines how quickly it will react once the *threshold* has been reached.

**audio frequency** – Signals and waveforms which fall within the spectrum of human hearing. Normally this is in the range between 20 Hz and 20,000 Hz; however, research has proven that humans can perceive frequencies both below and above this range.

**balanced wiring** – A system of wiring that minimises interference. A balanced cable has three conductors – two as part of an inner core and one overall *screen* which effectively wraps around the inner two conductors. The inner two conductors carry the signal as positive and negative phase and the outer screen is connected to the *earth* or ground. This means that interference such as that from radio frequencies (RF) cannot penetrate the screen so the cable doesn’t
end up operating like an aerial. It is important to connect balanced outputs to balanced inputs. Microphones that require phantom power must always use balanced connectors.

**bandpass filter** – A filter which operates by excluding a range of lower and higher frequencies and letting through a band of frequencies between these upper and lower frequencies. See *equaliser*.

**bouncing** – A multitrack recording technique whereby previously recorded tracks are mixed onto adjacent spare or empty tracks to free up more tracks for recording; i.e. once the tracks have been bounced, they can be recorded over with new material. This is a technique routinely employed on multitrack devices with a limited number of available tracks (4- and 8-track recorders for example) to give the engineer more scope for adding additional instruments and parts.

**boundary microphone/PZM** – Devised by Tandy in the 1980s, boundary microphones feature a condenser capsule mounted on a plate. The plate can then be positioned on or attached to a flat surface, effectively increasing the microphone’s *pick-up pattern*. Tandy originally intended these microphones for discreet use on lecterns, but many sound engineers realised their effectiveness in many applications including overheads on drum kits, classical recordings and stage performances. The PZM (Pressure Zone Microphone) is a version of the boundary mic.

**clipping** – Severe and potentially damaging form of distortion that happens when a signal is too high for the piece of equipment it is being fed into. This can be particularly damaging to loudspeakers. Manufacturers include many safeguards to avoid clipping in their equipment. It is very important to monitor meters and input lights. Flashing red is never a good sign.

**coda** – The end section of a piece of music that serves to sum up and finish off the ideas.

**crosstalk** – The undesirable transfer of a signal (typically a very high-level one) from one track or channel to another due to the proximity of the tracks or the channel busses. Thankfully with digital recording devices, crosstalk between tracks is virtually impossible; however, some analogue mixing desks may still be susceptible.
**DAT** – Digital Audio Tape. A format of recordable digital tape devised in the 1980s by Sony. This was the first commercially available digital recording format using small one-sided magnetic tapes. Still found in many recording studios, DAT has been largely superseded by the CD-R and minidisc.

**dB** – Decibel. The unit of measurement for audio.

**decay** – Part of the envelope of a sound. After the initial attack of any sound, there is a small amount of time where the maximum volume decreases and begins to level off. This is the decay time.

**diaphragm** – The part of a microphone that ‘collects’ the vibrations of the sound wave. In a *dynamic microphone* these vibrations are transferred to the coil to create the current required for a signal. In a *condenser* microphone the diaphragm is one of the charged plates through which the capacitance is measured to create the signal.

**DSP** – Digital Signal Processing. Technique whereby a signal is digitised and modified by having other digital signals or processes applied to it.

**dynamic processor** – Processors which control either the gain, envelope or frequency of an input source for both corrective and creative purposes. The most utilised are *equalisation, compression* and *noise gates*; but there are others such as *limiters* and *enhancers*. As the fundamental nature of the original sound source is being changed, it is important that dynamic processors have the signal flowing through them prior to its routing elsewhere, otherwise the effect is negated. This is why many compressors have microphone *pre-amplifiers* built into them. From a mixing desk, engineers should always use the *insert-point* to properly route the signal to be processed.

**enhancer/exciter** – A *dynamic processor* which can replace the lost *harmonics* of an original signal making it sound brighter, rounder and revitalised. While at the outset its effect appears similar to that of *equalisation*, it is in fact generating new frequencies that have been lost in the recording process.

**envelope** – The ‘shape’ of a sound in relation to time and volume. As a note is played it has variations in volume and these can be generally summed up as the note’s *attack, decay, sustain* and *release* (or *ADSR*). Different instruments exhibit different general envelopes. Percussive sounds, by their nature are short and loud, but wind and
string instruments take a while getting to their maximum volume and then can stay there for a prolonged period of time due to either blowing or the movement of a bow. Synthesisers can emulate these characteristics of acoustic instruments using an envelope generator circuit which controls each of these parameters.

**figure-of-eight mic** – A microphone *pick-up pattern*. Figure-of-eight or bi-directional microphones are sensitive to signals at both their front and rear, but have a drop in sensitivity at the sides. These two ‘lobes’ of sensitivity give the shape of a figure-of-eight – hence the title. Such microphones have some studio uses – recording *backing vocals* with two singers positioned either side of the microphone, for example – but are generally used for more advanced applications.

**flanging** – An effect whereby the original signal is *delayed* and then fed back on itself possibly a number of times with gradually increasing delays. This gives a swishing sound that can be rather dramatic.

**fundamental** – The lowest frequency of a musical note that has the most energy and is therefore the loudest part of the waveform. See also *harmonics*.

**gated reverb** – An effect whereby a *noise gate* is applied to the output of a *reverb* processor. The natural decay of the reverb is therefore cut off sharply resulting in a rather startling unfinished sound. The effect is most often used on drums and gives a powerful, if slightly obvious, sound. Nowadays, effects processors tend to have gated reverb settings preset within them with varying reverb characteristics and gate times.

**graphic equaliser** – An *equaliser* that has a series of preset filters spanning the audio spectrum. Each filter is designed to either *attenuate* or *boost* a narrow band of frequencies around a *centre frequency*. The more bands a graphic equaliser has, the narrower the bands need to be. For professional live applications, it is fairly standard to have a 31-band graphic equaliser. This means that there are around three separate filters for each octave of the audio spectrum. This enables sound engineers to adjust the output of the system to the venue to improve sound quality and reduce the possibilities of *feedback*.
high pass filter – A filter which operates by attenuating completely a range of lower frequencies, letting the high frequencies pass through unaffected.

insert point – A mixing desk connection that takes the signal from an input and sends it to an external processor in such a way that the signal flows through the external processor before being returned to the rest of the mixer channel. Insert points are predominantly used for patching in dynamic processors due to the fact that such processors need to change the basic nature of the signal in some way, shape or form. Insert points tend to be in the form of a single TRS jack socket where the tip of the socket sends the signal out, the ring of the socket returns the signal to the desk and the sleeve is earth. More complex mixers may have a separate socket for each function, send and return.

limiter – A dynamic processor that stops a signal from going over a predetermined limit. Essentially a limiter is a compressor with fairly extreme settings – a high ratio and a very fast attack time. Limiters are used in live sound as protection devices. If there is a sudden spike in a signal, the limiter can react quickly and prevent loudspeakers getting damaged.

low pass filter (LPF) – A filter which operates by attenuating completely a range of high frequencies, letting the lower frequencies pass through unaffected.

mastering – The art of taking a final mix and preparing it for mass production. This may entail the removal of clicks or glitches, editing the track so that it has neat start and end points, and also the overall addition of equalisation and/or compression. Mastering is part of the post-production of a project.

MTC – MIDI Time Code. A form of time code transmitted via MIDI cables to synchronise MIDI devices to one another.

outro – The end section of a song that finishes it off. The opposite of intro (or introduction).

pad – An attenuation switch normally found on condenser microphones, but may also be found on mixing desk pre-amplifiers. The pad switch reduces the output of the pre-amplifier by a predetermined amount, usually 10 or 20 dB. These tend to be needed on condenser microphones as they have a built-in pre-amplifier.
circuit. If a condenser mic is placed at a source and when the source is playing loudly, the mixing desk is being overloaded even though the gain control is fully down, then a pad switch would be employed to reduce the mic’s output.

**parameter** – A variable value that affects an aspect of a device’s performance or programming.

**parametric EQ** – An equaliser that has such control that it can pick out any range of frequencies from the audio spectrum and either boost or attenuate them. To achieve this, a parametric EQ has three controls per filter:

1. A gain control to either boost or attenuate the selected frequencies.
2. A sweep control to select the centre frequency of the range of frequencies.
3. A bandwidth or Q control to increase or decrease the range of frequencies to be adjusted. Parametric EQs can make very focused adjustments due to this range of control and are mainly found in professional recording environments.

**passive** – An electronic circuit that requires no power to operate, such as the basic tone controls in electric guitars. Some DI boxes are also passive as they only provide a basic splitting function.

**phase cancellation** – Where two versions of the same signal exist and are combined, and if the delay between the two signals means that the peak of one waveform combines with the trough of the other, then the resultant is nothing. The two identical waveforms have cancelled each other out. While total phase cancellation is rare and difficult to achieve, phasing problems can occur when two microphones spaced unequally apart are picking up the same instrument. The combined signal at the mixing desk then has the two waveforms interfering with each other and this creates a swirling sound or a high-pitched whine. Phase problems like this may be avoided if the second microphone is placed more than three times the distance from the source as the closer mic.

**phasing** – An effect whereby the original signal is delayed and then played back on top of itself. This gives a swirling filter effect as the frequencies in turn cancel each other out.
pitch shift – A process to change the pitch of an input signal without changing its duration. This is the basic form of a harmoniser.

post fade – A signal that is monitored or routed after it has passed through the channel fader and is therefore determined by the level of the fader. Post-fade auxiliary sends are used to send signals to effects processors. The fact that the amount of signal going through a post-fade send is determined by the fader position gives the engineer greater control over the positioning of the sound. More fader level and less aux send will make the sound closer, while less fader level and more aux (auxiliary) send makes it sound further away. This gives the mix engineer control over the front-to-back dimension of the music.
**post-production** – The elements of a project that are achieved after the main recording sessions. This includes *mixdown, mastering* and *manufacture*.

**pre-fade** – A signal that is monitored or routed before it has passed through the channel *fader* and is therefore independent of the fader position. *Pre-fade* *auxiliary sends* are used primarily for *monitoring* or *foldback* mixes, enabling musicians to hear a mix that is separate to that being monitored in the *control room* or heard through the PA system.

**pre-production** – The parts of producing a song prior to the studio sessions where the music will be recorded. This includes writing and *arrangement* of the material, rehearsing it and scheduling studio time and personnel.

**presence peak/colouration** – A mid-frequency emphasis in a sound, generally unwanted, that can occur due to a microphone’s frequency response, the microphone technique used, the proximity of the microphone to the instrument, the room’s acoustic or even a fault in a piece of equipment. These peaks are said to ‘colour’ the sound.

**production** – The act of committing a song to, usually multitrack, tape in the recording studio. See also *pre-production* and *post-production*.

**Q/bandwidth** – The width of an *equalisation* filter. This is the range of frequencies affected by the filter overall. Q stands for ‘quality factor’ and is an indication of being either a narrow bandwidth filter or High Q(uality) which affects a restricted number of frequencies, or a wide bandwidth of Low Q(uality) which affects a large range of frequencies. Q is determined as a number between around 0.5 and 10: the higher the number, the higher the quality factor, so the narrower the filter. Some *digital* filters have the capability of very high Q factors up to around 100. A filter this narrow would be affecting an extremely narrow bandwidth of frequencies, perhaps even a single frequency.

Here we can see that the notch between *a* and *b* is much wider than the notch between *c* and *d*. The notch between *a* and *b* affects a far greater range of frequencies that surround the *centre frequency* – here 1 KHz – than the notch between *c* and *d*. It is therefore said to have a low-quality factor or low Q, while the notch between *c* and *d*
has a high-quality factor or high Q. \( a-b \) could also be described as wide bandwidth and \( c-d \) as narrow bandwidth.

**ratio** – The control on a *compressor* that determines how much compression is applied to the signal once it has exceeded the *threshold*. The ratio can be treated much like a fraction; for example a ratio of 2:1 applied to a signal that exceeds the threshold by +2 dB will result in an output of +1 dB – half of 2 dB; similarly a ratio of 4:1 on the same signal will reduce the signal to +0.5 dB – a quarter of 2 dB. It is important to note that the threshold is still exceeded regardless of the ratio applied. Extreme ratios result in an effect known as *limiting*.

**red book standard** – The standard format applied to audio CDs as defined in 1980 by Sony and Philips. Originally CDs were purely an audio format, but they have been since utilised as mass storage media.
for all sorts of applications. There are therefore a number of other ‘book colours’ for different formats. The red book standard maps the basic requirements for CD audio reproduction including the *sample rate* of 44,100 Hz, the *bit depth* of 16, the maximum playing time of 74 minutes 33 seconds and even the physical dimensions of the disc itself.

**release**

1. The last element of a sound’s *envelope*. The release time is the time it takes for the note to die away to nothing after its *sustain* time has been met.
2. The control on a *compressor* or *noise gate* that determines at what speed the gate will close or compressor will stop compressing once the incoming signal has returned below the *threshold*.

**roll-off filter** – A filter, found either in an *equaliser* or on a *microphone*, that attenuates a series of pre-defined low frequencies. This can be helpful for eliminating stage or handling noise from the microphone.

**sample** – A digital snapshot of an acoustic sound. An *A/D converter* takes a constant stream of samples in order to convert acoustic sounds into *digital* information. A sampler can take a short series of these snapshots, alter their pitch and duration and play them back as tuned notes.

**shelving equalisation** – An *equaliser* filter that either boosts or attenuates constantly below or above a set frequency. On a graph, therefore, the equalisation curve resembles a shelf rather than a bell-shaped curve.

**slave** – A device that is controlled by another or master device. The master device generates the information for the slave device to react to.

**SPL** – *Sound Pressure Level*. The acoustic pressure of any sound wave. Expressed in dB as a value above the threshold of hearing, essentially the higher the SPL, the louder a sound is.

**stereo pair** – A *matched pair* of microphones used for accurate stereo recording. As the signals are to be played over a stereo system, the microphones are designated left and right and their signals will in turn feed those *loudspeakers*. 
submix – A mix within a mix, such as an overall mix of a drum kit routed to a subgroup which then feeds the master mix (see mastering).

sustain – The third element of the envelope of a sound at which the level of the sound stays constant. String instruments can have an infinite sustain via the movement of a bow. The sustain in wind and brass instruments relies on the breath of the player. Percussion instruments have only short sustain times.

tempo – The speed of a song measured in beats per minute (BPM).

threshold – Control on various dynamic processors that determines the point at which the process is applied to the signal. For example, on a noise gate, the threshold is the point at which the gate opens and lets the signal pass. Signals that do not reach the threshold remain unaffected.

transducer – Any electronic device that converts one form of energy into another; for example a microphone, a loudspeaker or a pick-up.

tremolo – An effect whereby the signal is varied up and down in volume. It can be particularly effective on guitar and electric piano sounds.

TRS jack – Tip, Ring, Sleeve. A jack plug and socket with three parts to the connection like a headphone jack. The tip element can be used to send a signal from an insert point to an external device, while the ring element is the return. The sleeve is connected to earth or ground and is common to both signals.

unbalanced wiring/connectors – Audio cables that have a single inner core to carry the signal surrounded by an overall screen to prevent interference. Such cables are used primarily for jack-to-jack leads from guitars or keyboards to an amplifier. They cannot be used for long cable runs as they are more susceptible to interference than balanced wiring.

waveform – A graphic representation of a signal or a sound wave's variation over time.

wavelength – The length between corresponding points of successive waves. Low-frequency waves have long wavelengths and high-frequency waves have short wavelengths. A simple rule of thumb is, ‘Double the frequency, half the wavelength’ and vice versa.
A/D converter – Analogue-to-digital converter. An electronic circuit that takes an analogue waveform and converts it into digital binary information that can be stored, and processed by digital devices.

ASIO – Audio Stream Input/Output. Devised by Steinberg, an ASIO driver is a software interface between an audio or sequencing package and a hardware audio device. Without this software, the software and the hardware cannot ‘talk’ to one another.

automated mixing – A system whereby elements of a mix, or complete mixes can be stored and recalled in either snapshot or real-time modes. This means that a mix engineer may adjust parameters during a mix and a computer will store and recall those adjustments in synchronisation with the music playback, thus allowing potentially very complex and dynamic mixes.

bit depth/rate – When an analogue signal is digitised, one of the values applied is the bit depth. This is the length of the ‘digital word’ and relates to the range of amplitudes that can be measured by the A/D converter. When the converter measures the sound, the amplitude may be either increased or decreased to fit into a set bit depth. This is called quantising. Therefore the greater the number of bits the converter is capable of processing, the smaller the quantisation that needs to be applied. This means that higher bit depths result in better-quality A/D conversion. Typically, standard CDs have a bit depth of 16, but it is not unusual to find digital devices capable of 20-, 24- or even 32-bit resolution.

byte – Digital data that is comprised of 8 bits.

coincident pair – A matched pair of microphones that are placed symmetrically close to each other in order to pick out the instrument(s) they are recording with a natural stereo sound that will include the acoustic and ambience of the recording venue. May also be referred to as an XY pair. Often used in classical and live recordings.

control surface – A control surface looks like a mixing desk, but it is an extension of computer-based audio software in a physical form.
Rather than clicking with a mouse on a screen to change a value within the software, the engineer uses the control surface; so a fader on the screen has its physical counterpart on the control surface. Due to their complexity and the number of functions available, control surfaces need to be connected to the host computer via a fast interface, normally Ethernet or FireWire.

**cut-off frequency** – The frequency above or below which attenuation begins in any filter.

**D/A converter** – Digital-to-analogue converter. Electronic circuit that takes a previously digitised signal and converts it back to analogue information. See also A/D converter.

**DAW** – Digital Audio Workstation. Any digital recording workstation that features recording, editing and mastering is referred to as DAW.

**Dolby** – Originally the manufacturers and developers of noise reduction circuitry for analogue recording devices, Dolby laboratories now specialise in surround-sound formats for movie theatres. Dolby’s system differs from that of DTS in that it only features one rear speaker.

**DTS** – Digital Theatre System. An enhancement of surround sound theatre systems which typically has 5 speakers and a sub-bass unit (the 0.1) to deliver true surround sound. The current configuration is 5.1 – front left, front right, front centre, rear left and rear right, however the EX or 6.1 system has been widely installed which includes a rear centre speaker. Some theatres even have a 7.1 system with 3 speakers at the front and 4 additional speakers providing the rear and surround effects. More and more the 5.1 and 6.1 formats are being used to mix music due to the upsurge in home theatre systems.

**DVD** – Digital Versatile Disk. A mass-storage format similar in size and shape to CD, but which can hold up to eight times as much information without compression. DVDs achieve this by: (a) being double sided, (b) having a much tighter spiral of burned pits, and (c) having multiple layers, the information being accessed by a re-focussing of the laser. This enables manufacturers to place on disk entire feature films with additional features including surround-sound formats. Nowadays we also have DVD-RW – re-writable DVDs, DVD video recorders and an audio-only DVD format called DVD-A. Most home computers come with both DVD players and writers installed as standard.
editing – The art of taking a series of takes in audio and creating one good one by use of cut-and-paste techniques or by rearranging recorded material into a different sequence.

expander – The opposite of a compressor, an expander increases the dynamic range of a signal by making low-level signals lower and high-level signals higher.

glitch – A short and nasty ‘click’ in digital audio. This may be caused by a corruption of the digital information or a poor edit of the soundfile.

harmonics – The overtone frequencies created by a fundamental which are an exact multiple of the fundamentals frequency. Whenever a note is sounded, it generates a range of frequencies above it at varying levels. If the fundamental note has the frequency of 150 Hz, then the first harmonic will be 300 Hz, the next will be 450 Hz, the next 600 Hz and so on, adding 150 each time. These harmonics and the balance of them give instruments their individual sound or timbre.

harmoniser – An effects process that takes an input musical note and adds harmonies to it by applying a short delay to the original signal and pitch, shifting it either up or down, or both, to create a series of intervals. While early harmonisers could only achieve a single fixed interval, harmonisers nowadays can be programmed to follow correctly the chord structure of a song and apply harmonies that fit accurately with the performance.

hyper-cardioid – A microphone pick-up pattern similar to cardioid, but with a much tighter and narrower heart-shaped field of sensitivity. Such microphones are ideal in live situations as the tight pick-up pattern is excellent at helping to eliminate feedback problems. These microphones are sensitive only to signals that are directed right at them.

instrumental break – The part of a song where the lead vocal stops and the instruments take over, possibly featuring a solo instrument.

latency – The delay between a signal going into a processor and coming back out again. While latency may occur to a small degree in most audio devices where what is being input is being simultaneously monitored, it predominates in A/D converters and D/A converters in computer-based recording set-ups. This is due to the time it takes for the computer to digitise and then un-digitise the audio information.
and is directly related to the processing speed of the computer. Faster processors significantly reduce any latency.

**LFE** – Low Frequency Effects. The sub-bass unit in *surround sound* audio systems. Its presence in the specification of surround systems is indicated by the .1, hence a 5.1 surround system means 5 normal speakers and 1 LFE, a 7.1 system is 7 normal speakers and 1 LFE, etc.

**LFO** – Low Frequency Oscillator. An oscillator used as a low-frequency modulation source; for example in the *chorus* effect, whereby the delayed signal is detuned by LFO modulation.

**link passage** – See *bridge*.

**matched pair** – A pair of microphones of the same make and model that have been manufactured in such a way as to be electronically and physically identical and therefore sound the same. Such pairs of microphones are used in *stereo* recording techniques such as *coincident pairs*.

**normalising** – System of connection in studios where a device that will normally be connected to another input or output is plugged into it permanently via a *patchbay*. If the device needs to be connected elsewhere, then inserting a *patchlead* into the socket will break the normalised connection.

**optical link** – A type of digital interface that changes the information into optical information generated by a laser rather than an electrical current. This is particularly useful where the information is to be passed over a long distance where fibre-optic cables are used rather than normal wires. The type of information coming out of (or going into) an optical interface is similar to that of *S/PDIF* information in that it is two channels of information; although optical interfaces exist that can carry up to eight audio channels.

**playlist** – A list of previously *edited* files, usually in a *DAW*, upon which the engineer can draw to create a final version of a piece of music.

**plug-in** – A software programme that can apply effects processes to an audio file. Plug-ins can only be used as part of an audio recording and editing package and not as stand-alone software. Generally speaking, the plug-in software draws on the internal processing power and capabilities of the computer rather than using its own hardware except in some professional situations where in order to...
operate fully, the plug-in requires additional hardware which has to be connected to the computer. There are currently two main types of software technology applied to plug-ins – RTAS and VST. While these are standards for different manufacturers of software, conversion software is available to enable the use of VST plug-ins on RTAS programmes and vice versa.

**pre-delay** – Part of the sound of reverb either in an acoustic or as part of a processor programme. The pre-delay of reverb is the time that it takes for the reverb to build up to an audible signal after the initial direct sound has reached the listener. By its nature, pre-delay helps the listener to determine the size of the space, since the bigger the space the longer it will take for the reverb to build, with a longer pre-delay time.

**quantising (audio)** – Rounding up or down. In digital audio, waveforms are measured within preset ranges. If the waveform being recorded doesn’t fit exactly into one of these ranges, then it is either increased or decreased to the closest value to fit neatly. This is quantisation. While in general A/D converters do this extremely well and without any noticeable degradation of the original waveform, occasionally there can be a problem in the rounding up or down and digital noise may occur. This is known as quantisation noise or error. See also A/D converter and bit depth.

**ribbon microphone** – A type of microphone, similar to a dynamic microphone, that operates using a thin film of metallic material, or ribbon, suspended in a magnetic field. As the ribbon vibrates, tiny fluctuations of current are induced in the ribbon giving the signal. While ribbon mics have an exceptional frequency response and little, if no, colouration of the sound, they tend to be fragile and therefore easily damaged. It is worth noting that sending phantom power to a ribbon microphone may also damage it beyond repair.

**RTAS** – Real Time Audio Suite. A software-based effect that uses the excess processing power of a DAW to add effects or processes in real time to a signal. Such software is included as plug-ins on DAW software and hardware.

**SACD** – Super Audio CD. An alternative to DVD audio. Super audio CD has been developed by Sony and Philips to deliver higher sound quality than standard CDs and also 5.1 surround sound. Because of this, an SACD does not conform to the red book standard and cannot
be played in a normal CD player, but may be played in an appropriate DVD player.

**sample rate/frequency** – The speed at which an *A/D converter* takes snapshots of the incoming signal in a second. The more samples it can take in a second then the greater the increase in the frequency response and therefore the better the quality of the A/D converter. CDs typically feature a sample rate of 44,100 Hz, or 44,100 individual snapshots in any 1 second, but it is not unusual to find digital recording systems and hardware with sample rates up to 192,000 Hz.

**scrub** – On a DAW, the function of playing backwards and forwards over a small area of the soundfile in order to select an edit point. The term originates from the same function using analogue reel-to-reel tape machines whereby the engineer would move the tape back and forth over the tape head to find the required edit point.

**semi-parametric** – An *equaliser* that has a *parametric* level of control over a restricted series of frequencies. Such equalisers tend to have preset high-frequency and low-frequency controls, but low-mid and hi-mid controls with *gain*, *sweep* and *Q* or *bandwidth* controls. This is the type of equaliser found on most high-quality *mixing desks*.

**SMPTE** – Society of Motion Pictures and Television Engineers. This is a time code standard defined by that society in America. Basically SMPTE (pronounced ‘Simpty’) is a digital clock code that can be recorded as a continuous stream of information onto a tape track, enabling two or more devices that can read the SMPTE code to remain in *synchronisation* with each other as they play.

**spaced pair** – A *matched pair* of microphones in a *stereo* recording system that are spaced apart (normally several feet), but are positioned at the same height. Such techniques are ideal for larger ensembles in larger spaces, symphony orchestras or choirs, for example.

**S/PDIF** – Sony/Philips Digital Interface. Digital devices can ‘talk’ to each other in the digital domain. Signals that have been digitised therefore don’t have to be *D/A converted* to send them along a digital signal path; in fact, to do so would be detrimental to the signal. A number of purely digital information interfaces exist, therefore, and S/PDIF is one of the most basic. As two channels of information can be sent or received by an S/PDIF socket, only one socket is required for a stereo input or output. S/PDIF uses *phono* sockets as standard.
S/PDIF outputs may be found on many CD players, whereas S/PDIF inputs and outputs are becoming standard on most semi-professional equipment.

**super-cardioid** – A cardioid microphone with an extremely narrow angle of response. Such microphones can be used where feedback or ambient noises are a problem. They are generally found being used by sound engineers recording interviews in the open air, where it is beneficial to eliminate as much of the surrounding noise as possible. In live applications, super-cardioid microphones have excellent rejection to feedback.

**surround sound** – A multi-channel sound recording and reproduction system that utilises an array of speakers around the listener to create special effects and a realistic sound field. See also DTS, Dolby and LFE.

**TDIF** – Tascam Digital Interface. A multi-channel stream of digital audio information that can transmit up to eight channels simultaneously.

**total recall** – An automated mixing system on mixers that stores the settings of a mix in a central computer for recall at a later date. While total recall systems store the settings, it is still down to the engineer to physically alter the settings.

**total reset** – An automated mixing system on mixers that stores the settings of a mix in a central computer for recall at a later date. Total reset systems differ from total recall systems in that the desk will recall all the settings automatically. No physical work or adjustment by the engineer is necessary.

**vocoder** – An electronic process that can apply synthesiser and instrumental control over an input signal; for example, that from a microphone. Vocoder can create interesting ‘talking’ synthesiser effects.

**VST** – Virtual Studio Technology. A software standard that emulates a physical studio, with all its processing, etc., in a virtual environment such as a computer’s memory. Devices tend to be represented with icons and certain functions can be achieved by patching devices together on screen. VST has also become a software programming standard for plug-ins, while VSTi’s are Virtual Studio Technology instruments – software-based instruments that use the computer’s excess processing power and memory to create virtual synthesisers and instruments.
There follows a cumulative A–Z listing of all the glossary headwords in this publication. Reference to this index facilitates cross-referencing between the five separate lists in the main glossaries.

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