MICROPHONES

The basic principle of the dynamic microphone
The condenser microphone
Electrical impedance
Frequency response
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Omnidirectional Microphones
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Microphone Cabling

Microphones are transducers. Transducers are devices that convert energy from one form to another. In normal musical activities, such transformations between different types of energy are very common; for example, as a pianist presses a key, electrical changes in his brain and nervous system are converted into movements in his muscles, which are in turn converted into acoustic energy by the action of the hammer setting the string into motion. When the sound-waves created by the piano strike the listener's eardrums, a reverse process takes place which changes the wave's kinetic energy back into electrical energy, the medium of thought and perception. Here we will be considering the change from sound energy to electrical energy in the microphone.

The microphone should be regarded as a device that is, or at least in skilled hands has the potential to be, as subtle as any musical instrument. Each microphone has its own characteristic properties which make it more or less useful in any recording or live situation, and like a musical instrument, its overall quality is often (though not exclusively) related to its cost, with more refined ones requiring more demanding manufacturing tolerances and better components.

The microphone was an invention of the late nineteenth century, and its development was closely tied to that of telegraphy and telephony. In 1876, Alexander Graham Bell patented his 'speaking telegraph' a device that was the forerunner both of the telephone, and the modern 'dynamic' microphone. Bell's transducer (which was both microphone and earpiece) depended on electromagnetic induction - the principle of the inter-convertibility of magnetism and electricity, which had been demonstrated by Michael Faraday in the 1830s: if a magnet is moved through the centre of a coil of wire, an electric current is 'induced' in the wire which is proportional to the magnet's motion. This simple principle lies at the heart of the technological revolution, for the ability to generate and make use of electricity has more thoroughly transformed mankind, whether for good or ill, than almost any other discovery in its history.

The basic principle of the dynamic microphone

Many of the different types of microphone invented over the last century or so have a feature in common with the human ear - a membrane that is the receptor for sound waves. In the microphone this is a thin circular plastic diaphragm that vibrates in sympathy with the sound waves striking it. It is not usually visible, being hidden beneath a protective wire-mesh cover and foam wind-shield. There are several methods of converting the diaphragm's motion to electricity, but perhaps the simplest is found in the dynamic (or moving-coil) microphone. Attached to base of the diaphragm of the dynamic microphone is a coil of thin wire which moves up and down in a narrow gap between the magnetic poles of a specially manufactured circular magnet whose poles form concentric circles. See figure 1.

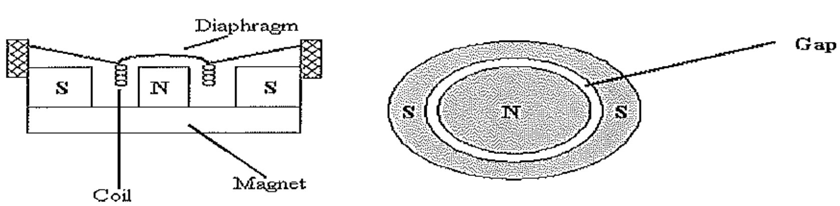


Figure 1. Cross section through dynamic microphone components (left), and magnet viewed from above (right).

As the coil moves through the magnetic field between the north and south poles of the magnet, a current flows between the two ends of the coil. The current which is thus induced is an alternating current (AC), in other words it cycles between positive and negative values as the diaphragm moves in and out, in the same way that the sound pressure fluctuates in the sound wave being captured by the microphone.

The condenser microphone

The other commonly found microphone type, the condenser, depends on a different electrical property, that of capacitance. A capacitor is an electronic component formed from two oppositely charged plates (electrodes) with a gap between them. Although a detailed discussion of the principle of capacitance is beyond the scope of this hand-out, it should be noted that the output of a capacitor is related to the size of the air gap between its plates. In the condenser microphone, a 'polarizing' voltage is applied to the plates in the capacitor from a phantom power supply (often 48 V from a mixing desk or separate supply). One of the plates is in the form of a thin metal-coated (often aluminium or gold) diaphragm which, like the equivalent in the dynamic mic, moves in response to soundwaves (Figure 2). As the diaphragm moves in and out, the air gap between the plates changes, which has the effect of altering the output voltage from a constant 48 V DC to 48 V plus the AC signal described above for the dynamic microphone. This new signal is an example of 'modulated' signal. A further capacitor removes the DC element leaving just the AC signal.

A somewhat cheaper and less high-quality condenser microphone is the electret or back-electret. This does not require phantom powering because it has a charge permanently fixed between its electrodes during the manufacturing process (usually around 100 V). Users will notice that such microphones do require a power supply, however, usually in the form of a 1.5 V pen-torch battery. This is not required to charge the capacitor element, but to power other circuitry built into the microphone, namely a pre-amplifier. The purpose of the pre-amplifier is to reduce the very high impedance of the condenser element, and allow efficient transmission of the signal along the microphone cable to a mixing desk or other device.

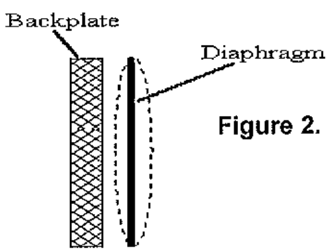


Figure 2. The backplate and diaphragm of a condenser microphone.

Electrical impedance

Impedance is measured in ohms, can be regarded as a measure of resistance to the flow of an alternating current (e.g. an audio signal) in a circuit. The term is used in two contexts: output (or source) impedance; and input (or load) impedance. The output impedance is thus effectively a gauge of the resistance in a device which is producing an audio signal (for instance a microphone or a synthesizer), and the input impedance is the resistance at the input of the equipment it is connected to (a mixer, amplifier or loudspeaker). Two important facts should be remembered about impedance: the input impedance should always be higher than the output impedance; and in situations involving long cable runs, a low impedance source should be used. Professional-quality microphones are usually low-impedance devices, which means they will have output impedances of 600 ohms or less, and often between 100-200 ohms; the microphone inputs on a mixing desk will usually be between 1000-2000 ohms. Mismatching output and input impedances will at best produce an inadequate or distorted signal, and at worst physically damage the equipment. With cable runs of more than about 20 feet, high-impedance microphones (usually greater than 25 k) will suffer signal loss, whereas low-impedance mikes may pass along hundreds of feet of cable without signal degradation

Frequency response

A further basic characteristic of microphones is frequency response. Musical sounds are made up of many different frequencies of the audio spectrum. If certain frequencies are made louder (by raising a slider on a 'graphic equalizer', for instance), the tonal qualities of the sound will be changed. If a microphone has a flat frequency response, it will not 'colour' the sound by boosting or cutting frequencies within it - it should sound 'natural'. Because they are not burdened with an attached coil of wire, condenser microphones tend to have smoother, more linear response patterns than dynamic ones. Some dynamic microphones that are intended for vocal work or speech actually make a virtue of their emphasis of certain frequencies, called a presence effect. The region affected usually lies between 2 kHz-5 kHz, and can result in an improvement in the articulation of speech, at the expense of an increase in background noise. Associated with the linearity (or smoothness) of the response of the microphone is the usable frequency range. This will often be described in specifications as, for example, 50-15000 Hz. Ideally, a microphone with a flat response from 20-20000 Hz is required to accurately reproduce musical performances, but in many cases, compromises must be made on grounds of economy.

Sensitivity

The actual signal output of a microphone is determined by its sensitivity. This is often measured relative to a sound with a 94 dB sound pressure level (SPL), which equates to somewhere between a symphony orchestra playing fff and a loud disco. Unfortunately, as with many audio measurements, there is no single method of indicating a microphone's sensitivity, but an output of 2 - 20 mV (thousandths of a volt) at the sound pressure level described above is typically found. As well as the signal, every microphone will add a little 'self noise' to its output. Generally this is of a low enough level to be disregarded

So far, the discussion has considered features that are common to different types of microphone. They are further categorised according to their polar response pattern;- a description of how sensitive they are to sound coming from different directions. As a generalization, there are three classes of polar response pattern: omnidirectional, undirectional and bidirectional.

Omnidirectional Microphones

An omnidirectional microphone picks up sounds coming from all directions (omni means all); it's polar response pattern is illustrated by Figure 3 below.

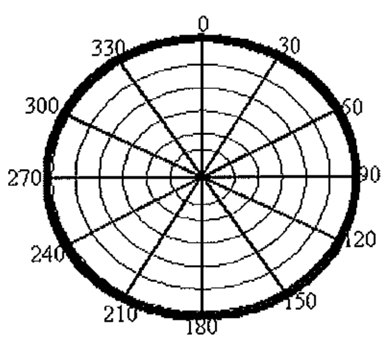


Figure 3. Polar response pattern for an omnidirectional microphone.

The microphone in figure 3 is at the centre of the innermost concentric circle facing forwards toward 0 (imagine that you are looking down from above the microphone). The response pattern is represented by the outer thicker line, the magnitude of its response at any angle being represented by the line's distance from the centre (the closer to the centre, the higher the microphone output). Thus for an ideal omnidirectional microphone, its response will be the same all the way around. The diagram is misleading in two ways, however: firstly it appears that the response pattern forms a flat plane (like a disk) - it should really be seen as a sphere; secondly, any real omnidirectional microphone will become unidirectional (it will tend to pick up more from in front of it) at high frequencies (usually between 10-15 kHz). Omnidirectional microphones are relatively easy to recognise, for they have a grille only at the front end (This is because they are *pressure* microphones - they rely only upon the pressure on the front of the diaphragm, in the same way that the ear responds to pressure arriving at the eardrum from the ear canal. While the all-round response of the omnidirectional microphone is clearly useful, there are many circumstances where a more directional response would be beneficial. Omnidirectional microphones can be problematic in PA (public address) sound reinforcement, where they can increase the possibility of acoustic feedback, the high-pitched screeching sound caused by the continual re-amplification of certain frequency components, often lying around a microphone's response peak (see presence effect above). This is an example of positive feedback - a vicious circle in which the sound from a loudspeaker is picked up by a microphone, amplified, and then reproduced again by the loudspeaker, with the cycle being continually repeated. A microphone with a unidirectional response can provide some shielding from such effects of feedback by suppressing sounds coming from its rear.

Directional microphones

Directional microphones work on a pressure-gradient principle; instead of simply responding to the pressure on the front of the microphone capsule, they also depend upon the difference in pressure on the front and rear of the diaphragm. If a microphone is designed to react to the pressure difference alone (in other words if the sound can be picked up by the front and back of the diaphragm equally), it will, if properly designed, have a bidirectional or figure-of-eight polar response (Figure 4). This type of response pattern has two *lobes*, lying at the front and rear of the microphone; sound from the two sides is suppressed. If an omnidirectional capsule is combined with a bidirectional capsule, a new type of polar response pattern can be produced - a unidirectional response (Figure 5).

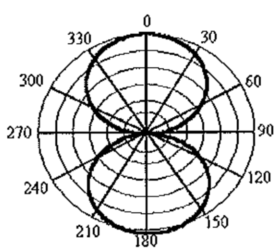


Figure 4. Bidirectional (figure-of-eight) response pattern.

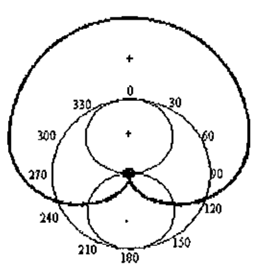


Figure 5. The unidirectional (cardioid) polar response pattern (bold line) produced from the combination of an omnidirectional and a figure-of-eight microphone.

As can be seen from Figure 5, the pattern is heart shaped, and is thus described as *cardioid* - a term derived from the Greek root for the word heart. Its polar response favours sounds from in front, and suppresses sounds from the rear; one should note, however, that not all sound from the rear of the microphone is suppressed. It is thus suited to work in which isolation of individual instruments or instrumental groups is required, and in conditions where one is trying to avoid feedback - cardioid and related patterns can be brought relatively close to loudspeakers without creating squeals and howls. It is expensive to build a unidirectional microphone by combining two other microphone capsules, so most cardioids use a single capsule which has an 'acoustic delay network' built into the back of it. With careful design and engineering it is possible to cheaply construct capsules which have cardioid polar response patterns like that depicted above.

Intermediate between cardioid and figure-of-eight polar responses are those of supercardioid and hypercardioid microphones. If the 'acceptance' angles in degrees (in other words the angle within which sound will be picked up whose centre point is formed by the microphone) of these different types is compared, it can be seen that they get narrower as one moves from cardioid to figure of eight: cardioid 131; supercardioid 115; hypercardioid 105; figure of eight 90. However, the size of a central rear lobe increases from nothing for the cardioid, to the same size as the front lobe for figure of eight.

Figure 6. Cardioid, supercardioid and hypercardioid polar response patterns.

A useful analogy when considering different microphone types is that of light sources. An omnidirectional polar response is the equivalent of a conventional household light bulb or fluorescent tube which lights up an entire room. Unidirectional polar responses can be likened to torch beams, with a hypercardioid polar response being like a narrow and very sharply focused beam, and a cardioid being somewhat wider and more diffuse .

As well as the various types of unidirectional transducers described above, there are a number of special-purpose microphones which have a very directional (lobar) polar response, and can be likened to telephoto lenses for cameras: the 'rifle' microphones. These are up to about two feet in length, usually surrounded by a windshield, and often used in film and TV sound work to isolate individuals being interviewed or acting on camera.

Pressure-zone microphones

Pressure-zone or boundary microphones (PZM) have a very different appearance. They are hemispherically omnidirectional devices (though directional versions have also been manufactured) usually formed from electret elements mounted very close to a substantial flat metal plate, with a small gap of around 1mm between the diaphragm and the plate (Figure 8). The effect of this set-up is to produce a microphone with a very flat, smooth polar response.

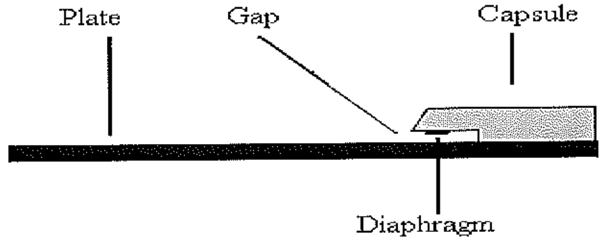


Figure 8. The PZM or boundary-effect mike.

Microphone Cabling

Professional microphones are generally connected to a mixer or other equipment using three-pin XLR connectors. These come in two types: male, which have three pins and which connect to the mixing desk; and female that have three holes, and connect to the base of the microphone. While the basic requirement for equipment interconnection is for just two wires in the cable (one for the signal, one for the ground), a third wire provides for a *balanced line*, a means of reducing noise which is induced in cables over long runs. In the balanced line, one wire carries the signal, and one the ground, as described above, but a third carries an inverted version of the signal (its polarity is reversed). Illustrated in Figure 10 below is a sine wave (bold line) superimposed on its inversion (dotted line). If these two signals, which are exact and opposite, were added together, they would cancel each other out, producing silence.

When the two signals reach the receiving apparatus, assuming it is equipped to deal with balanced signals, the inverted signal will be turned the right way round, and added to the original un-inverted signal. The effect of this is to remove, or at least heavily attenuate, any noise induced in the cable, and to increase the original signal amplitude. It is assumed that the same noise will be induced into both the signal-carrying wires, so that when one is re-inverted and added to the other, the noise in one wire will have the opposite polarity to that in the other, and they will cancel each other out, whereas the signals will have the same polarity, and their addition will create a signal with double the amplitude.

MICROPHONE TECHNIQUES

Using microphones
Solo instruments
Stereo Microphone Technique
Multi-microphone set-ups

The selection of a microphone for a particular recording or PA situation clearly requires careful consideration. A number of basic questions must be asked before selecting equipment, including:

- Is accuracy and 'transparency' (the absence of unnatural 'colouration' of sound by a microphone) important? – This is often a function of the genre. EG in classical and acoustic music accuracy and transparency are generally considered essential, in alternative rock a highly coloured or unnatural sound may be desireable.
- How wide are the dynamic and frequency ranges of the instruments we wish to record?
- Do we wish to improve the articulation of speech, or help make the text clearer in vocal music?
- Do we need to 'insulate' the sound of a performer from other sounds, such as audience noise?
- Do we want to produce a recording in which the reverberant space (the 'ambience' of a hall, say) is a feature?

In real recording situations, these, and other considerations, will probably not be mutually exclusive. Whatever the case, it is unlikely that a single microphone type will fulfil all the requirements of an engineer or producer.

Solo instruments The simplest case is the recording of a solo instrument, perhaps to produce a demonstration tape or a digital sample. It should be obvious that the further the microphone is placed from the instrument being recorded, the lower will be the energy of the sound wave reaching it, and the higher will be its pickup of ambient noise. If the microphone is brought too close to an instrument, however, there will be an increase in mechanical noise such as fret-noise, string-squeaks or bow scrape. There may also be a substantial low-frequency boost caused by the proximity effect - a feature regularly used by pop vocalists, who often hold the microphone several inches from their mouth, though this can be reduced if the microphone or channel- strip has a bass roll-off switch.

Clearly, no hard and fast rules can be given, though in general terms the placement of a cardioid microphone a short distance from the sound source is a useful opening gambit. It is important to understand where the bulk of a musical instrument's sound is generated: this is not always obvious (see acoustic guitars, below) and considerable experimentation may be required to determine a satisfactory position. In some cases, the optimum position may not even be in front of the player - for instance, the flute may benefit from a microphone placement behind or above the player. A good starting point the apex of an equilateral triangle immediately in front of the source (Figure 9).

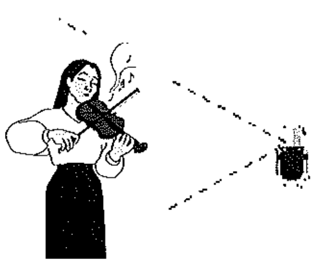


Figure 9. Microphone placed at the apex of an equilateral triangle.

In pop and rock engineering, the microphone is often positioned close to the bell of the brass and woodwind, though slightly angled away, so that it is 'off axis'. It should be remembered that the tonal characteristics produced by this method may be rather different from the 'natural' live sound as the "room sound" will be diminished. Brass instruments can produce very substantial sound pressures from their bells, and care must be taken to avoid overloading the microphone and causing distortion, or even damage; dynamic microphones tend to be more robust, and better suited to close use for loud instuments, than condensers. It is probably sensible to start with a position which is 'off axis', in other words not pointing directly at the bell, but above it pointing down, or below it pointing up.

The piano is one of the most difficult instruments to balance. There are many possible positions, from inside the instrument to some distance from it. A position on an arc from the foot of the instrument to the treble strings (right-hand side facing the keyboard) in which the microphone is able 'to 'see' the strings (or rather the greater part of the soundboard' is a good compromise.) multiple mic set-ups have traditionally produced the best results (a couple of bi-directionals is a good, though expensive method. A combination of ambient "spot" mikes can also work. A cheap & cheerful sure-fire winner is to tape a couple of Tandy, or similar PZMs to the underside of the open lid of a grand piano. Great results for less the than cost of a half-decent dynamic mike.

Drums & percussion, like brass instruments can produce tremendous volumes of sound, and similar care should be taken in the selection and positioning of microphones. A kit may well require a large number of separate microphones, with one per component not being unusual. As well as these close-placed 'spot' microphones, a crossed pair and spaced pair (see the next section for more details of these terms) may be placed above the kit to pick up ambient sound.

Acoustic guitars & other stringed instruments are not as easy to record as it would appear. A common mistake is to point the mike at the instrument's soundhole. The soundhole is merely a tuning-port, and produces no more volume than other parts of the instruments sound-board. (Try covering the soundhole & see what happens) a mike placed too close to the sound-hole produces excessive bass, especially when combined with the mike's proximity effect. Experiment with mike position while monitoring on headphones until a pleasing sound is produced. (try a position beside the players ear. Instruments always sound good to the player. Watch out for the player's breathing or sniffing though!) a coincident pair of small-diaphragm condensers placed 10" to 18" in front of the instrument is a good starting point. The condenser's higher output & sensitivity allows the mikes to be placed further away from the instrument, thus avoiding the proximity effect.

Vocals present many problems, and classical and pop techniques can be usefully contrasted. In order to naturally reproduce the vocal sound of a classical singer, the microphone should not be brought too close to the mouth - generally a distance of greater than 1 metre is recommended. Pop musicians, however, regularly use very close miking, and as explained above, this gives rise to the proximity effect that can usefully contribute a warmth and solidity to the sound. However, this manner of performance also produces the danger of explosive consonants (especially "P" & "B" sounds) and sibilance ('s' sounds). Placing a 'pop shield' between the singer and the microphone can counteract the former. (a pair of tights attached to a wire coat hanger makes an acceptable pop shield!) For the latter a frequency-conscious compressor, known as a "De-esser" must be used.

Stereo Microphone Technique

Early recordings were monaural (mono), being made with a single microphone and reproduced on a single speaker. Whilst such recordings conserved the relative levels of instruments in an ensemble, they failed to provide any left-right clues to instrumental locations in an ensemble, all sounds appearing to come from the same central source. This was clearly unsatisfactory and a means was required to reproduce the position information of a performance. Alan Blumlein patented a method of stereophonic recording In 1933, which made use of intensity (or loudness) differences between two microphones (and thus two loudspeakers when the sound was reproduced) to create the illusion of space.This 'classic' method uses two figure-of-eight microphones set at right angles to each other their grilles almost touching.

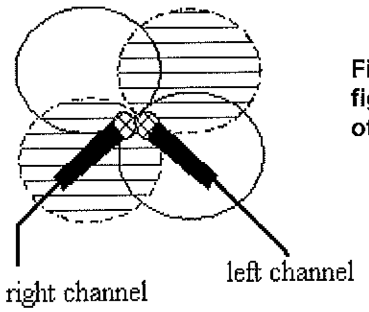


Figure 10. The polar response pattern resulting from two figure-of-eight microphones set at an angle of 90 to each other in the coincident (Blumlein) technique.

Whilst this setup produces an accurate stereo image when placed in the appropriate position relative to the performers (often close to the conductor's head in an orchestral or choral recording), its acceptance angle is restricted to 90, and it will pick up sound from behind. This latter problem is compounded by the fact that the microphone picking up the front right will also be picking up the sound from the back left! Using cardioid or hypercardioid microphones, the angle between the capsules can be adjusted, and unwanted noise from the rear can be rejected.

Whereas the coincident-pair technique described above makes use of the level differences between the two channels for its stereo effect, a second method, the spaced-pair uses the time differences between channels (Figure 11).

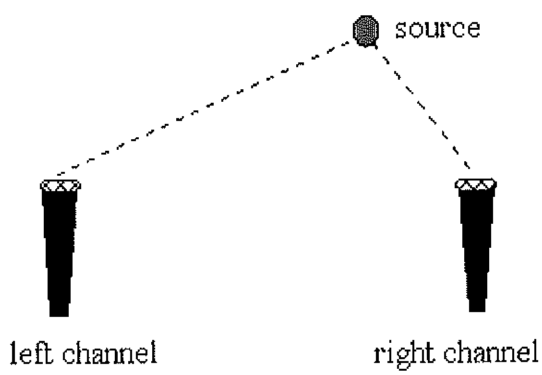


Figure 11. The spaced-pair technique.

As can be seen in Figure 11, the sound from the source will have to travel further to reach the left microphone than the right one, and will thus take longer. Sound travels at roughly 343 metres per second, or around one foot per millisecond, and so each extra foot will take around 1/1000th second. When the sound is reproduced by the loudspeakers, it will appear at the right speaker slightly before the left, which when coupled with the level difference will help localize the sound: a sound coming from the right-hand side will reach our right ear fractionally before it reaches the left and will thus be perceived as stimulus from the right side. Some suggest that the omnidirectional microphones used with this technique should be spaced by no more than 1/3 of the width of the ensemble being recorded. Others feel a spacing of 10-12 feet is reasonable, but recommends the use of a third microphone placed halfway between the other two, and fed equally to the two channels. Care must be taken with these techniques, as the phase errors which are the essence of the technique when reproduced in stereo, cause serious problems with mono-compatibility. (Hence the “mono check” switch on studio desks)

A third common stereo-microphone technique is the so-called near-coincident pair. In this case the unidirectional microphones cross over each other so that the grilles are separated by about seven inches, the handles forming an angle of around 110 (Figure 12).

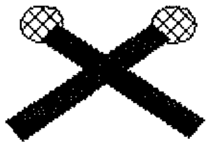


Figure 12. The near-coincident pair of unidirectional microphones.

This method relies on a combination of time and level difference (and is thus a compromise between the two methods described above). It produces a sharply focused three-dimensional image. A further technique is the coincident pair, where the mike capsules are almost touching. The image is not as defined as the above methods, but this technique is absolutely mono-compatible.

A number of other methods of producing a stereo are in common use including the *Middle and Side* (M&S) technique, which is widely used in film and television work, and uses the combination of a figure of eight (side) and a unidirectional or omnidirectional (middle) microphone. Using a special decoding box called a *matrix box* or a mixing desk equipped with phase-reversal switches, it is possible to extract the left- and right-hand channels by adding together the output of the middle and side microphones (producing the left hand channel), and subtracting the side output from the middle output to produce the right-hand channel.

Multi-microphone setups

The techniques outlined above are utilised throughout the production of recordings, regardless of stlye or genre. However classical, orchestral or ensemble recordings tend not be made using the "overdub" procedure in contemporary rock & pop production. Many microphone placements are feasible for orchestral and ensemble recordings. Illustrated in Figure 13 is a setup which uses a main stereo pair, with spot microphones for individual instruments or sections, and a spaced pair of omnidirectional microphones in the centre of the audience for ambience (in the case of a "live" as opposed to a studio recording)

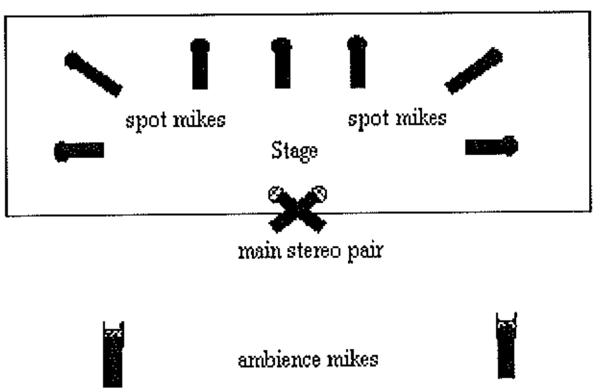
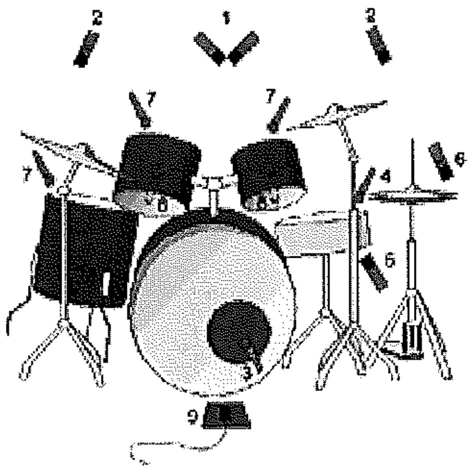


Figure 13. A multi-microphone setup.

The heart of the set-up is the stereo pair (coincident or near-coincident pair of cardioid microphones) which provides the bulk of the recorded signal, though the balance can be subtly varied by adjusting the level of individual spot microphones (again cardioids or hyper-cardioids) at the mixing desk. In many cases the mixer channels for spot mikes may be completely closed (fader at the bottom of its track), and the ambience mikes kept at a low level.

Miking drums

Modern Drum-kit miking techniques, demand that individual instruments are given their own microphone and their own track on a multitrack recorder. This is then supported by a stereo pair for the "room-sound" See figure 14



1. Coincident pair overheads.
or
2. Spaced pair overheads.
3. Kick drum.
4. Snare (batter head).
And/or
5. Snare (snare head).
6. Hi-hat.
7. Toms.
8. Alternate toms (inside).
9. PZM for kick drum.

Figure 14. Suggested multi-mike set-up for a standard drum-kit

It should be noted that the use of microphones in different positions does present a possible hazard: the sound will take longer to reach the main pair than it does to reach the individual spot microphones (roughly 1/1000 second for each extra foot) which could cause odd 'ping-pong' effects or colouration to the sound because of phase changes (the waveform being superimposed on itself with a slight time delay). There will be time delays at some frequencies that could potentially cause the cancellation of waves, and seriously compromise mono-compatibility.

Critical evaluation of recordings

Record engineering and production are intensely musical occupations, even though many producers & engineers are not musicians. It is in the hands of these individuals to make a performance into something that can be listened to and enjoyed repeatedly, and yet when successful, attention should not be drawn to their participation. (though some producers do manage to achieve "star" status) Instruments and voices must be carefully balanced, dodgy room acoustics must be accounted for, and an appropriate ambience must be achieved. In modern pop production the final product is literally created, by the piecing together of lots of 'takes', to assemble a performance that was never actually performed.

Musicians tend to find the process of critically listening to a *recording*, rather than a *performance*, very difficult; it is, however, an extremely revealing exercise to try to deduce how a performance was recorded. A number of questions about the recording should be considered, here's just a few:-

- Do the instruments sound 'natural'?, if they do not, is this deliberate?
- Is there a good stereo image with depth as well as width?
- Is there a natural ambience, or have room simulations been used effectively & convincingly?
- Are the instruments well balanced, are any instruments "fighting" for space?
- Is the mix appropriate for the musical style or genre?
- Is the recording free from distortion & or noise?

DYNAMIC PROCESSING

Compression

Though far more subtle than the use of EQ, reverb, delay or modulation effects, dynamic control, using compression, expansion and limiting is a vital component in achieving master-quality recordings.

DYNAMIC RANGE

Compressors (as well as limiters and expanders) act upon the dynamic range of an instrument, voice or signal. Dynamic range is the difference between the quietest and loudest parts of a performance, the minimum and maximum volume an instrument or other sound source can produce, or the highest and lowest signal levels in a recording. Dynamic range is expressed in decibels (dB). Sound sources that have a wide dynamic range present problems for the recording engineer in achieving consistent signal levels. Dynamic processors such as compressors are designed to solve these problems.

LEVEL CONTROL

We must have effective control of levels going into the mixing console & recorder. This control needs to be faster than an engineer can monitor meters, make a decision about correction and begin to adjust faders. The compressor therefore, becomes an extra pair of eyes, ears and hands. Compression therefore can be seen as *automated* level control, constantly monitoring the signal and making adjustments according to parameters defined by the engineer. Loud sections or peaks, are attenuated to prevent them overloading (distorting) the desk or recorder.

What the knobs do......

THRESHOLD

The THRESHOLD control determines the point at which compression begins. If the signal level is below the threshold nothing will happen. When a louder signal is received the Compressor will reduce (attenuate) the output by an amount determined by the ratio control.

RATIO

The RATIO control governs how aggressively the compressor will lower output in response to strong signals. The lowest setting (1:1) means no reduction. 2:1 means that for every 2dB over the threshold the level will only rise by 1 dB. 4:1 means that for every 4 dB over the threshold the level will still only increase by 1 dB.

An infinite ratio (∞:1) is in effect "limiting" which tells the compressor not to let the signal ever be louder than the threshold

ATTACK

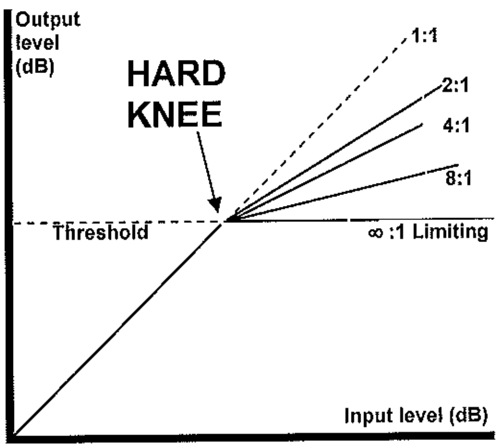
When a string is plucked or a drum is hit, the very beginning of the sound is usually the loudest. The brain uses these "transients" to identify and categorise sounds. Sometimes we want these initial transients to pass uncompressed. In other cases, for example a loud snare-hit or a percussive bass guitar part, which could send the meters into the red, we would want compression to begin immediately. The ATTACK control determines the length of time it takes compression to start and is expressed in milliseconds (usually between 0.1ms & 200+ ms)

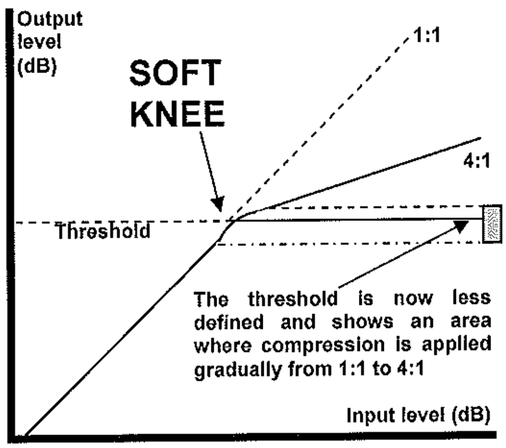
RELEASE
This control determines how quickly the signal returns to normal after compression.
A long release time has a cause a sound to sustain. (Often used by guitarists to achieve sustain while using a clean tone) Shorter times can sound more natural but need to be used with care in order to avoid audible "Gain-pumping". Expressed as time, from about 50ms to several seconds.

OUTPUT
Also called gain make-up, used to compensate for overall loss in volume due to processing.

COMPRESSION MODES
PEAK; In peak mode the compressor looks at instantaneous input levels. This is good for making sure that fast transients e.g. drums, do not overload the recorder or desk.
RMS; also called *auto mode*. Here the compressor looks at the average overall signal level and makes gradual adjustments. In this mode the attack and release times are not user-definable but are program dependent. This mode is good for vocals or for complete mixes.

HARD KNEE/ SOFT KNEE
The difference between these 2 modes becomes more apparent when a ratio of 4:1 or above is selected. In hard knee mode the full compression ratio is applied immediately. In soft knee mode the compression sneaks in over a range of several dB, making the effect less abrupt.





COMPRESSION PROBLEMS & SIDE EFFECTS

For every dB of compression applied background noise increases by 1dB. A compressor only affects sound above its pre-set threshold, which it then attenuates. However if we reduce the highest peaks by 10dB, we would then have to make up the gain reduction by turning up the output. If we increase output by 10dB to compensate, any unwanted noise will also increase by 10dB. This will be particularly nasty during quiet passages with nothing to mask the noise.

EXPANDERS & GATES
Many compressors now have in-built expanders and/or noise gates to go some way to solving the above problem. An expander is the opposite of a compressor, & will increase the dynamic range of a signal particularly when the lower limits have been reached by the presence of noise or hiss. An expansion ratio of 2:1 will produce on output of 2dB for an input of 1dB. Expansion is used as a corrective measure , a much better technique is to ensure the original signal is a noise-free as possible by correct level-settings achieving optimum signal to noise ratios.

HIGH FREQUENCY LOSS

Most of the energy in typical rock and pop music is in the lower frequency ranges and is generated by the kick drum, bass guitar and snare drum. This shows up a major weakness in using compression. High frequency sounds which contain less energy, (i.e. are quieter) occuring at the same time as low frequency sounds can be reduced to the point of being almost inaudible. E.G. a hi-hat strike at the same time as a kick-drum strike will be reduced in level even though it is below the compressors threshold.

To avoid this problem a longer attack time must be set to allow the transients through uncompressed. (see *attack*) though if heavy compression is applied to the complete mix the sound can lose "presence" & "sparkle"

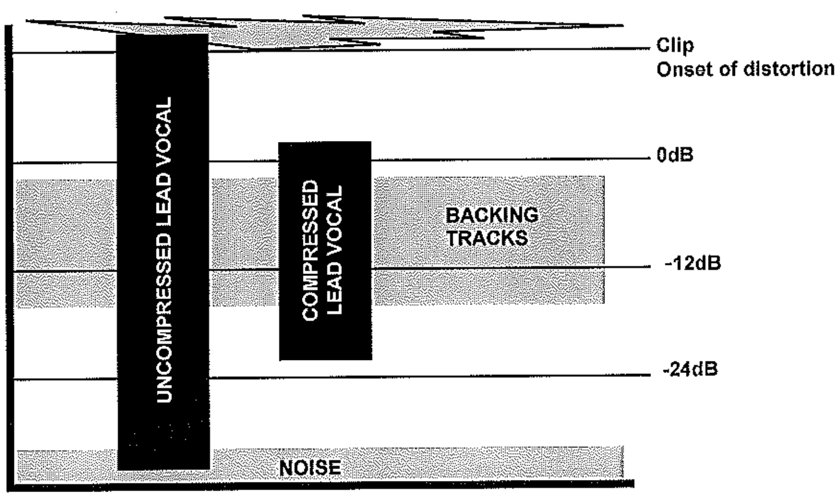
Some newer compressors include an "exciter" to re-introduce HF after heavy compression, Others have a distortion feature which introduces a small amount of clipped signal. This goes some way to mimic early valve compressors, which had their own characteristic distortion (in a similar way to a tube guitar amp) & made HF loss less apparent. Great care must be taken when using "exciters" particularly during recording. A little restores sparkle, too much can produce a over-bright and harsh mix.

WHERE DO WE CONNECT DYNAMIC PROCESSORS

When we use time delay or modulation effects, we take a small part of the signal, usually from a mixing desk's Aux send, or an amplifier's FX loop, add the effect and mix the "wet" signal back in with the original "dry" signal. Dynamic control must be applied to the whole signal, therefore the Aux sends are not suitable. Any instrument producing a line-level signal (keyboard, guitar pre-amp, acoustic-electric guitar) could be plugged directly into the compressor. Microphones do not have a strong enough output for this & need to pass through a mic pre-amp, found at the top of the mixer's channel-strip, or a stand-alone unit. Each mixer channel will have an insert point, (often terminating at a patch-bay) where compressors can be added.

WHEN DO WE USE COMPRESSION

Most engineers compress while recording & again during mixing. Compressing while recording ensures consistent levels, avoids overloading the desk or recorder & makes optimum use of the recording medium's dynamic range. However, as any effect added during recording is irreversible, it is best to be cautious. Compression used during recording will be in peak mode, during mixdown the compressor will be switched to RMS or auto mode.



This diagram shows compression used to improve a vocal track. On the right is a level meter.
The grey band at the bottom shows the background noise level of a recording system.
The central grey band represents the average level of the backing tracks. If the lead vocal is recorded without compression it can peak into distortion, or be low enough to be lost in the mix, obscured by noise or allow background noise level to become obtrusive. After compression the vocal stays in an area where it stands out from the backing track, yet is in no danger of clipping.