# MICROPHONES

# EXAMPLES OF EXAM QUESTIONS

## Short Questions

(b) Whilst recording, the vocalist was very close to the microphone.

(i) What is the main advantage of placing the vocalist close to the microphone?

(1)

..................................................................................................................................................

(ii) Identify two problems close mic'ing the vocal could introduce to the recording. How could these problems be reduced during the mix?

(4)

Problem 1

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Solution 1

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Problem 2

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Solution 2

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(b) The bass guitar was recorded using DI. There is hiss on the recording. Excluding signal processing, identify **three** precautions that could be taken to reduce hiss whilst recording DI bass guitar.

(3)

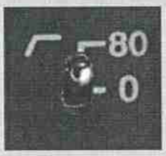
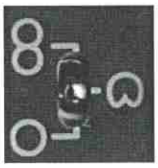
1 ..................................................................................................................................................

2 ..................................................................................................................................................

3 ..................................................................................................................................................

# Table Questions

(a) The vocal was recorded with a condenser microphone using the switch settings shown in the table below. Identify the switches, describe what they do and explain why the settings have been selected for this recording. An example has been given.

| Switch | Identify the switch | Describe what this switch does and explain why this setting has been selected |
|---|---|---|
| **Example:**   | High Pass Filter | • Engages filter to cut frequencies below 80 Hz<br>• Reduces proximity effect<br>• Removes rumble |
| (i)  | (1) | (3) |
| (ii)   | (1) | (3) |

Complete the table below to describe how you would mic a singer to achieve a similar recording to that heard in "vocal.wav". Give reasons for your choices. An example is provided for you.

| | What you would choose | Reasons for choice |
|---|---|---|
| Microphone distance | 10-30 cm | · Less reverb<br>· Less background noise |
| Microphone type | (1) | (2) |
| Microphone polar pattern | (1) | (2) |

(a) These tracks were recorded using DI.

(i) What does DI stand for?

(1)

(ii) Briefly describe how this would be achieved.

(1)

(b) An amplifier simulator has been applied to the DI guitar recording. Complete the table below to describe how you would mic an amplifier to achieve a similar distorted tone to that heard in tracks 4 and 6, giving reasons for your choices.

| | **What you would choose** | **Reason for choice** |
|---|---|---|
| Microphone type | Dynamic | • can withstand high SPL<br>• coloured frequency response gives a punchy sound |
| Microphone polar pattern | (1) | (2) |
| Microphone placement | (1) | (2) |

# Picture

(c) The vocal was recorded with a condenser microphone using the switch settings shown in Figure 1. Identify the three switches. Explain why the settings have been selected for this recording.

| | Identify the switch | Explain why this setting has been selected for this recording |
|---|---|---|
| (i) | ................................ (1) | ................................<br>................................<br>................................<br>................................ (2) |
| (ii) | ................................ (1) | ................................<br>................................<br>................................<br>................................ (2) |
| (iii) | ................................ (1) | ................................<br>................................<br>................................<br>................................ (2) |

(i)

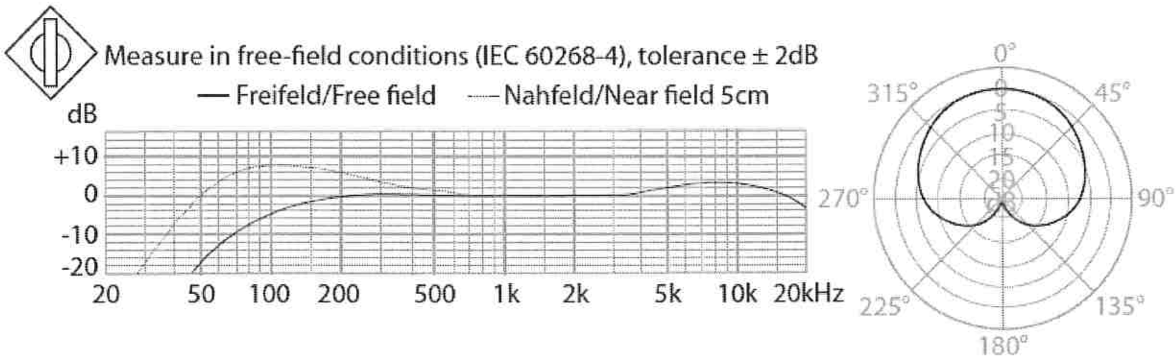
(ii)

(iii)

**Figure 1**

4 (e) The diagram shows technical specifications for a microphone. Evaluate the suitability of the microphone for recording vocals.

(8)



# Essay Questions

Microphones have been used since the late 1800s to record sound. Describe what a microphone does. Explain how a dynamic microphone and a condenser microphone function and identify the benefits of each type.

Figure 1 (refer to Figure 1) is a picture of a microphone placement for a female vocalist. Evaluate the recording techniques used.

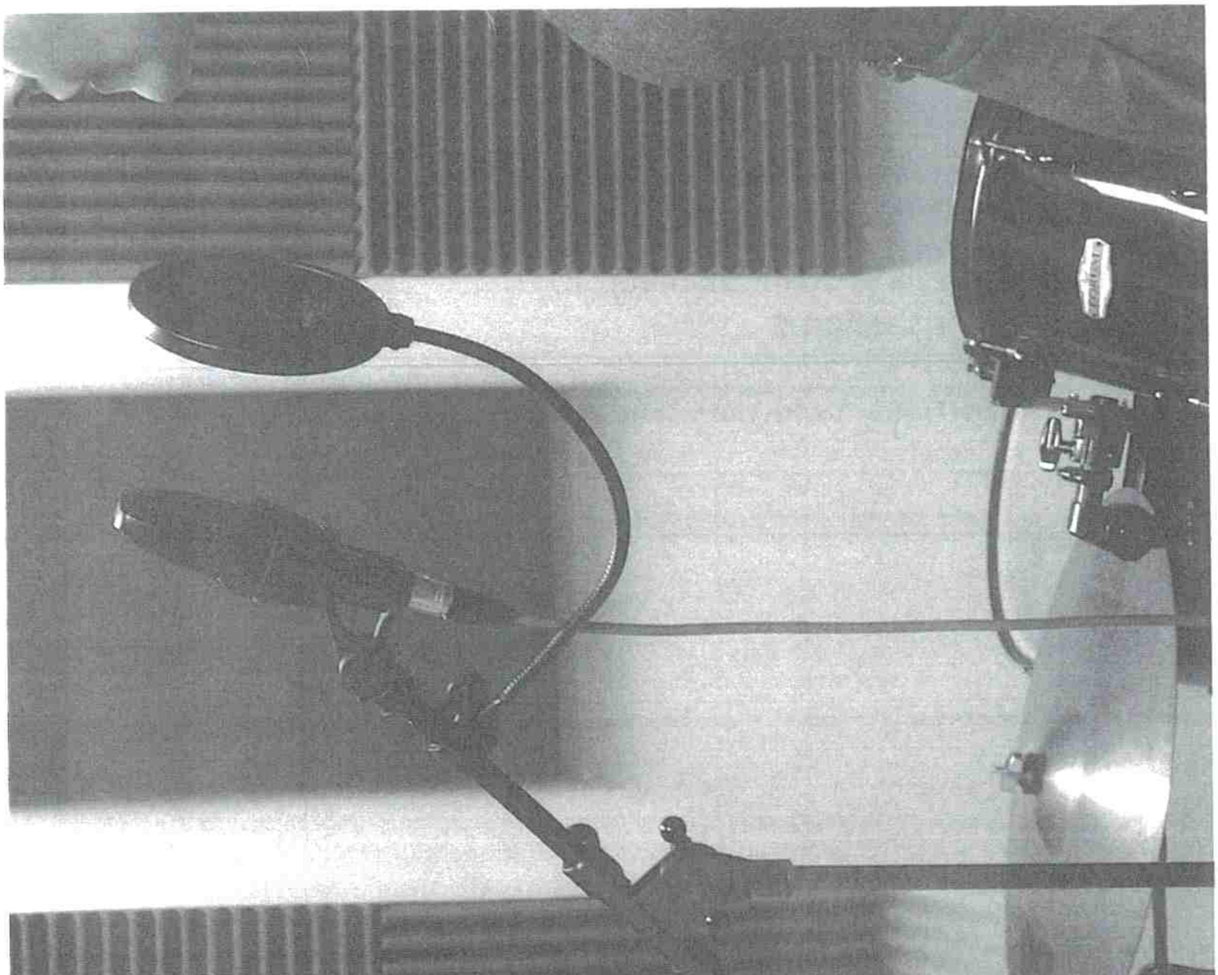


Figure 1

Figure 2 (refer to Figure 2) is a picture of a microphone placement on an acoustic guitar. Evaluate the studio technique used.


**Figure 2**

Describe how you would mic up a standard drum kit for a rock band in a contemporary recording. Explain any decisions you make. How would this have been done differently in the mid-1960s?

## Sound and Audio Fundamentals

### Microphones

There are several types of microphones which each use different technology to convert the acoustic energy into electrical energy. The most common types are:

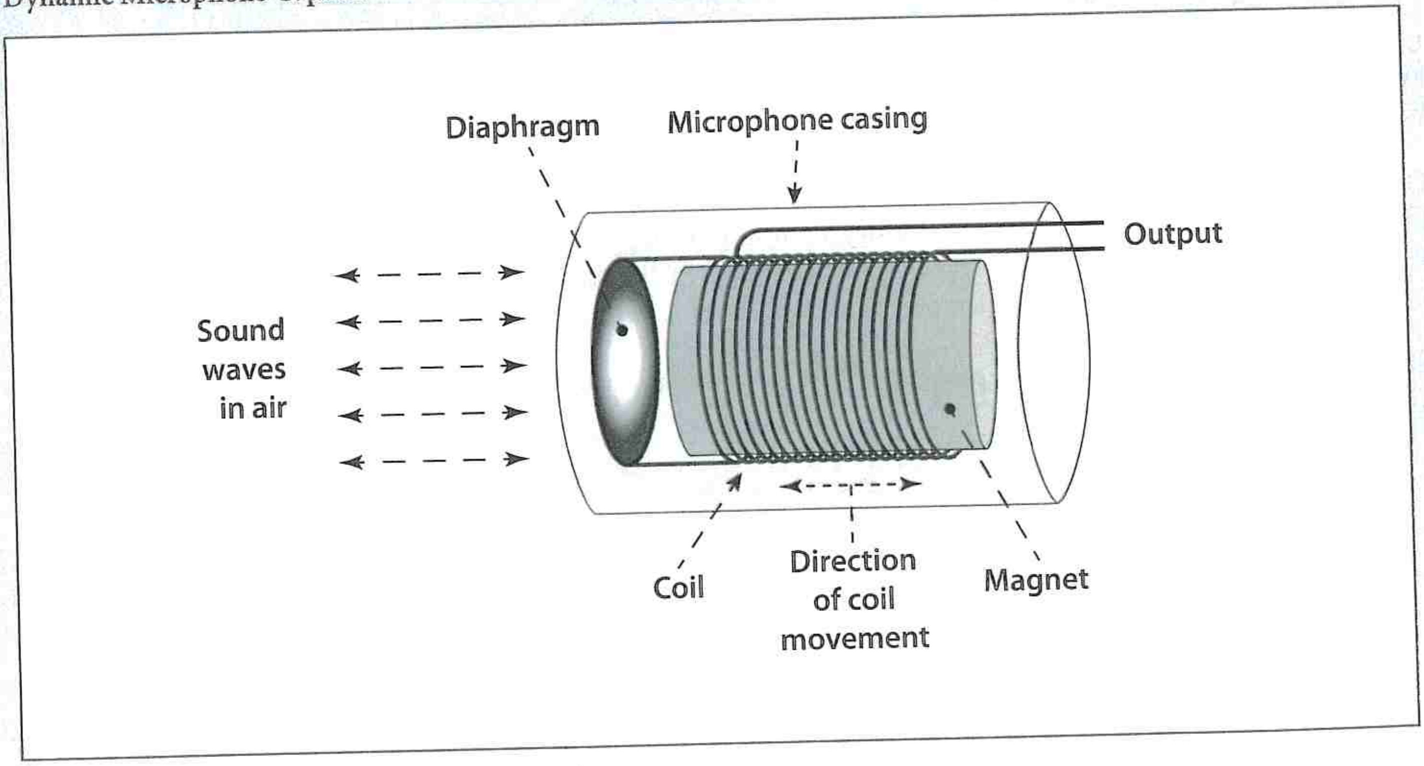
- Dynamic microphones
- Condenser microphones
- Valve microphones
- Ribbon microphones

Dynamics and condensers are by far the most used in both live and studio situations.

### Dynamic Microphones

Dynamic microphones use a capsule which contains a magnet and an electrical coil. As the vibrations in the air move the diaphragm of the microphone, the coil moves around the magnet causing an electrical current to flow through the cable.

Dynamic Microphone Capsule

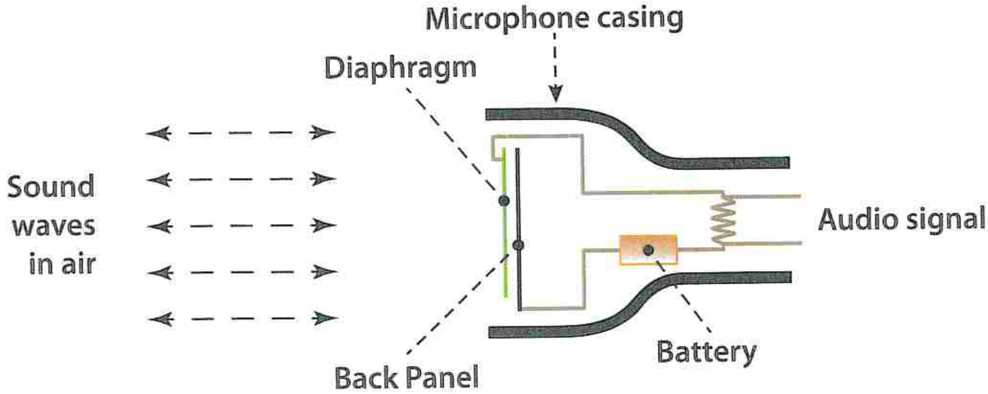



Most dynamic microphones are constructed to be quite tough and are therefore good in loud situations, such as close to guitar amplifiers or inside kick drums. The construction of the casing is also very hardy which is why they are often the microphone of choice for live performances.

### Condenser Microphones

Condenser microphones create electrical current by using an electronic concept called capacitance. The diaphragm of the microphone forms one plate of the capacitor with another inside the capsule. One plate has a positive charge and the other a negative charge. As the vibrations in the air move the diaphragm, the distance between the two plates varies with more current flowing when the plates are closer together.

### Condenser Microphone




Condenser microphones are a little more delicate than dynamics, although there are some fairly solid designs out there suc as the AKG C1000. They tend to sound clearer and brighter than dynamics and are subsequently a little more expensive t buy. This extra clarity makes them very good for vocals and acoustic instruments such as pianos and acoustic guitars.

Condenser microphones require phantom power to operate which provides the charge difference between the two plates o the capacitor. You will more than likely find a switch on your mixing desk or audio interface which is labeled as +48v. Thi is the phantom power switch, sending 48v down the XLR cable to the microphone. It's important to only use this switch fo condenser microphones as it may damage other designs.

Some condenser microphones can be powered with a battery which will be helpful if your mixing desk or audio interfac doesn't have phantom power available.

Condenser microphones can also be referred to as 'capacitor microphones' or 'electrostatic microphones'.

### Loops

Loops are small sections of audio or MIDI which flow naturally when repeated. For example, a one bar drum loop whic starts on the kick drum on beat 1 and ends on the last moment of beat 4. If you were to immediately play the loop straig afterwards it would sound as if the drummer had continued playing with the kick drum on the 1 again.

Some DAWs offer a selection of loops as a way of getting started. Apple loops (as used in Logic and Garageband) are eve more useful because they will help you ensure you're always in the right key and at the right tempo.

As well as using the provided loops, it is possible to create your own loops from your own audio recordings and MI performances. This makes it possible for you to build up a song quite quickly once you have got the overall groove an chord progression together. It may be that you decide to re-record the audio you've arranged using loops at a later date t get a more natural performance from the musicians but it's fairly common in electronic music to build an entire track usin nothing more than loops.

Loops are created by selecting the required length of audio or MIDI and separating it from the rest of the performance s that you end up with one region of the exact length required, ready to duplicate.

The talent in creating loops lies in ensuring that you are selecting exactly the right amount of time, i.e. the very start of be 1 to the very end of beat 4. This is certainly a skill that is worth developing as it takes a keen ear and accurate editing.

## Sound and Audio Fundamentals

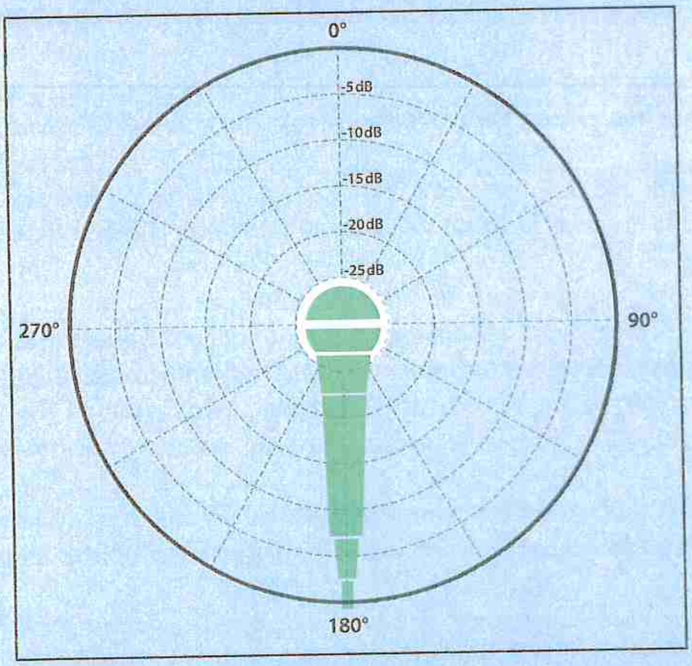
### Microphones

**Polar Patterns**
Microphones are designed to be able to pick up sound in different ways whether that be from every direction or focused on some directions and not others. The area that the microphone is sensitive to is known as its polar pattern.

The ability to choose which polar pattern is needed makes them far more practical in a recording situation.

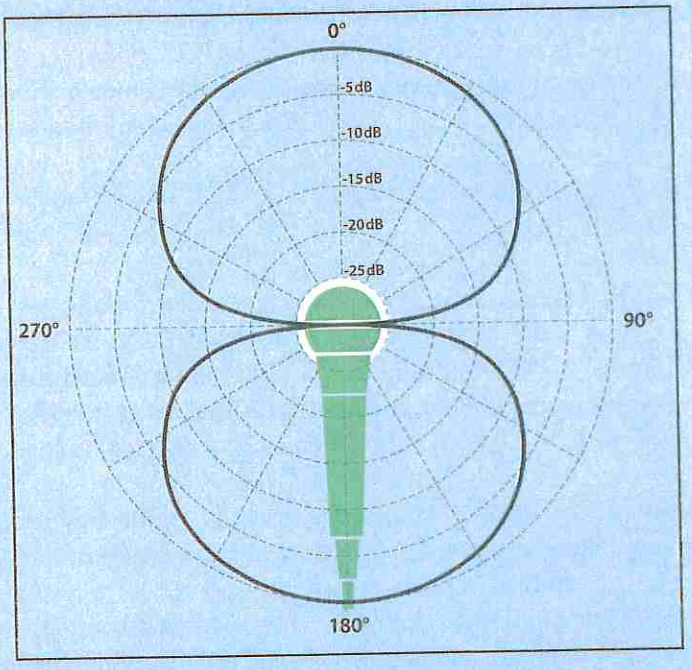
The most common polar patterns are:

**Omnidirectional**
A microphone with this polar pattern is sensitive to sound from every direction equally. Remember that this is through three dimensions: Up, down, left, right, forwards and backwards.




**Example Omnidirectional Microphone:**
The Shure KSM44A is a large multi-pattern microphone that can be configured for omnidirectional work.


**Figure of 8**
A figure of 8 microphone (sometimes abbreviated to fig-8) is most sensitive to sound from both infront and behind, creating a number 8 style pattern around the capsule. It rejects sound from the sides.

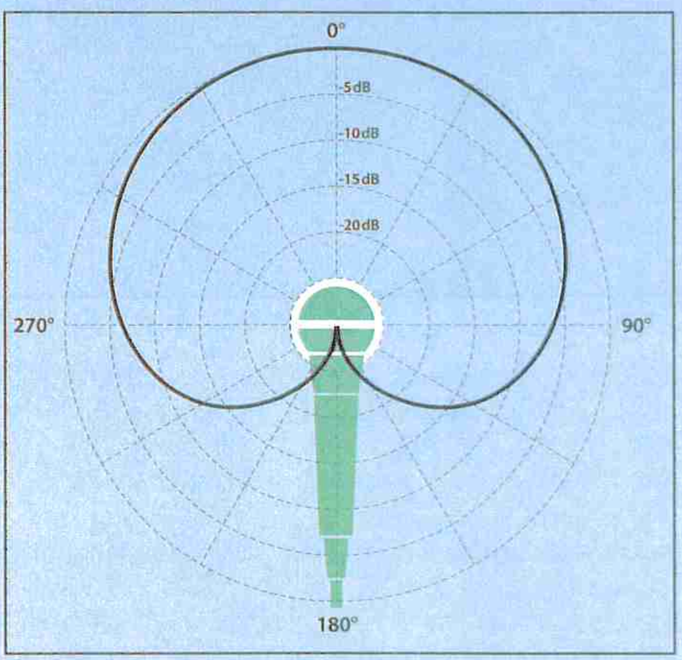



**Example Figure of 8 Microphone:**
The AKG C414 is another multi-pattern microphone capable of working in a figure of 8 configuration.

## Cardioid

A microphone with a cardioid polar pattern is more sensitive to sound from in front of its capsule, i.e. whatever the microphone is pointed at is loudest. This is probably the kind of microphone you are most familiar with.

A side effect of the cardioid design is proximity effect. When a cardioid microphone is placed close to the sound source, the bass frequencies are emphasised. This can be a good or bad thing, depending on what you're trying to achieve. Remember that it is there and exploit the effect when you want something to sound a little warmer or fuller. This can be good for 'breathy' vocals'

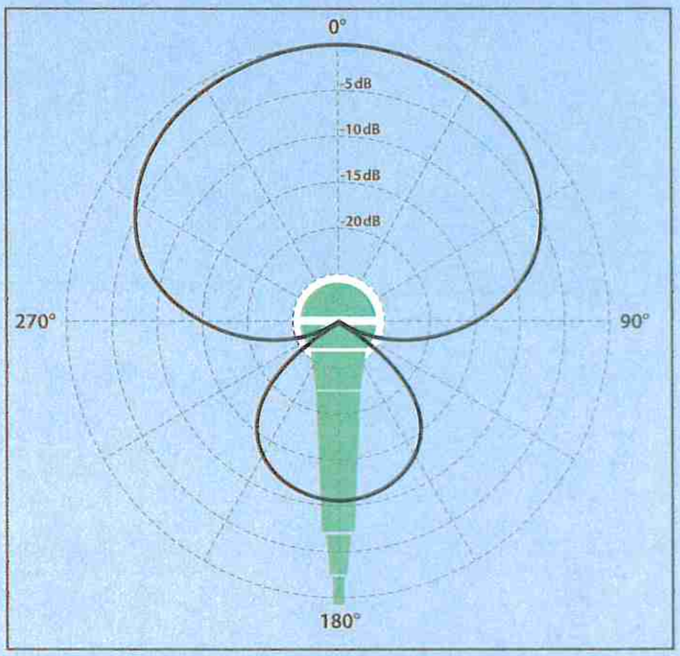



**Example Cardioid Microphone:**
Sennheiser MD421

## Hypercardioid

A hypercardioid microphone is very similar to a cardioid, except that the front focus is exaggerated, causing it to reject even more sound from the sides. This makes it very useful in situations where you need to record a very specific sound source where there is lots of unwanted sound around it, e.g. a lead violin in an orchestra or when recording dialogue for a film in a noisy street location.

There is a trade off for having a microphone that's this focused and that's that it is also sensitive to some sound from behind.




**Example Hypercardioid Microphone:**
Audio-Technica AT4053b

### Microphone Technique

Microphone techniques can be separated into three areas:

- 'Close Miking'
- 'Multi-Miking'
- 'Stereo Miking'

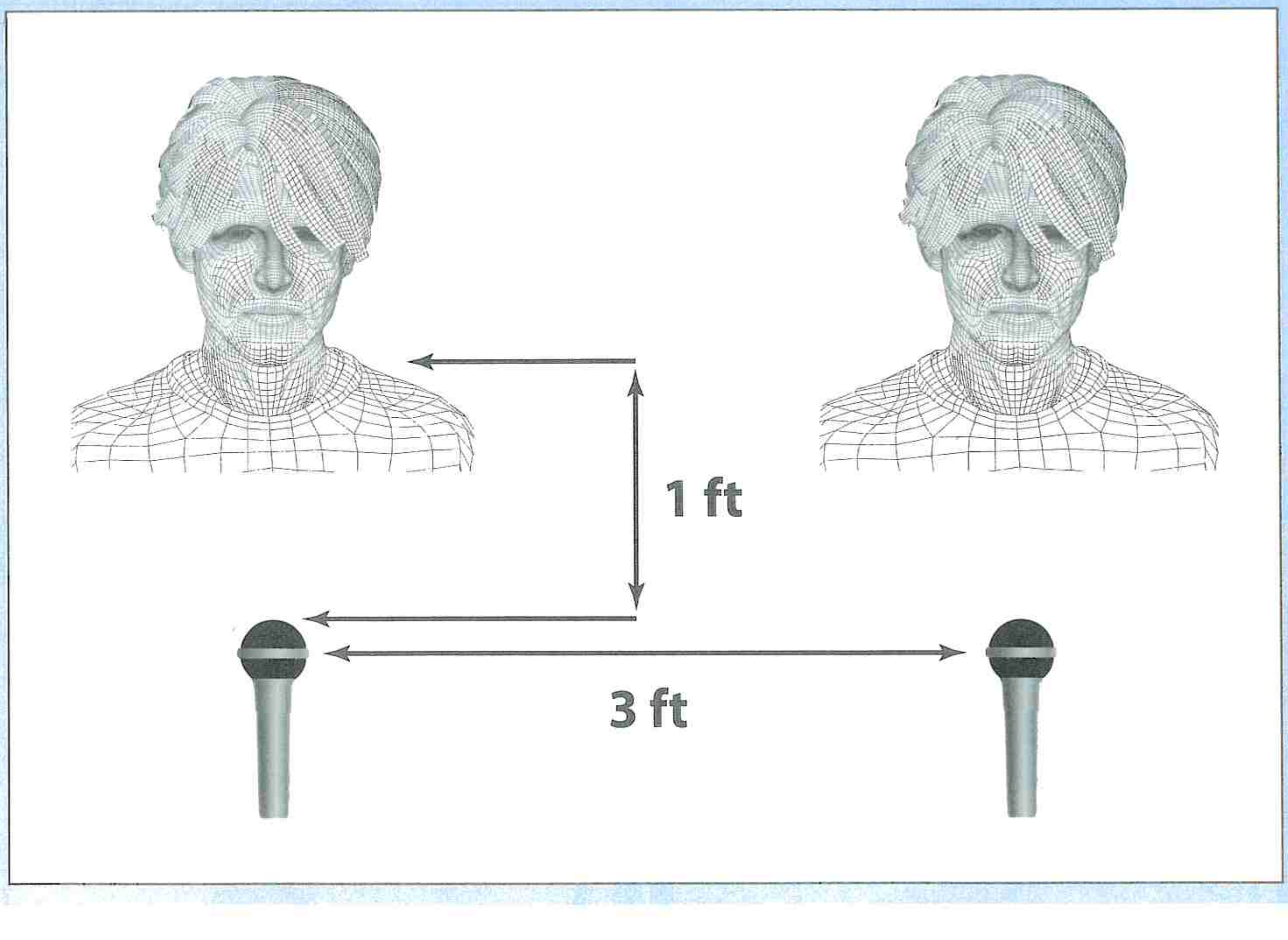
**Close Miking**

This is where you place a microphone close to the source with the intention being that you record only the desired source and as little as possible of anything else in the room. This technique is very common in modern recording, particularly in rock and pop music where a tight sound is required.

**Multi-Miking**

This is where more than one microphone is used to record a source with the intention that the microphones are mixed together later. This could be a combination of close miking and stereo miking techniques, although the result is not necessarily mixed in stereo.

When using more than one microphone, a good rule of thumb for placement is to use the 3:1 rule. When using more than one microphone, you should ensure that the distance between two microphones is at least three times the distance to the source.



## Stereo Miking

This is where two microphones are used to record a sound source with the intention being to create a stereo representation of what that instrument sounds like in the environment that it is recorded in. This is achieved using two microphones, one of which will normally be panned hard left and the other panned hard right.

For example, you may choose to record an acoustic guitar or string section in a nice sounding chamber with wooden panels to create a live reverberant sound.

By using a stereo microphone technique you can better capture more of what the instruments sounded like in that nice sounding space, providing the listener with a more realistic auditory experience.

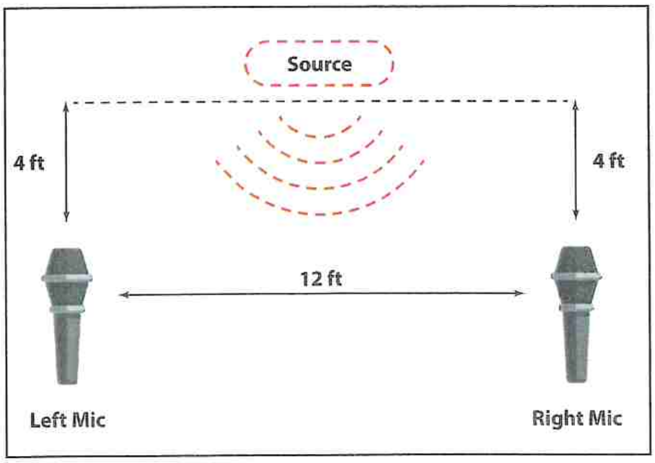
There are numerous tried and tested stereo microphone techniques. Here are some examples:

### Space Pair / Spaced Omnis (also known as AB)

This is a technique which requires two omnidirectional microphones although cardioids can also be used if omnis aren't available.

The two microphones are placed at a similar distance to the source but with a distance between them. For best effect, that distance should adhere to the 3:1 rule (see p.18).

This microphone technique results in a spacious sound in which the room is very apparent and the stereo image is quite wide. As a result, this can cause problems with phase cancellation between the two microphones, so adjust the positioning of the microphones until you get the best sounding result and check that it still sounds good when in mono by panning the two microphones to the centre or using a mono switch on the mixing desk if you have one.

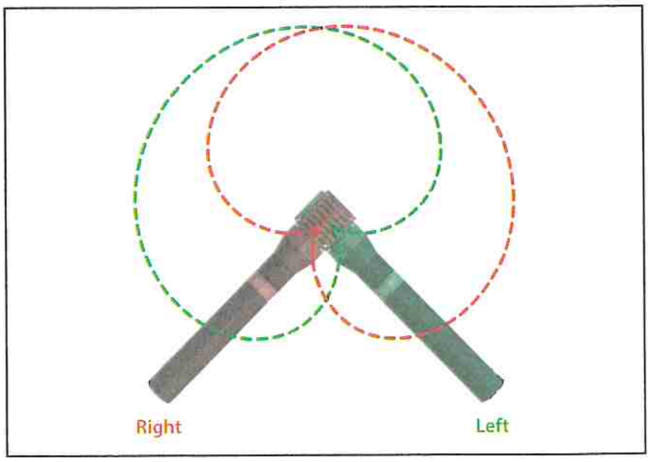



### Coincident Pair (also known as XY)

This technique requires two cardioid microphones which should be placed with the capsules directly above each other. They should then be directed in different directions to each other at an angle of between 90 and 135 degrees.

This technique results in a controlled stereo image with the sound source still in focus.

You might choose to use this technique for recording acoustic guitar or for drum overheads.



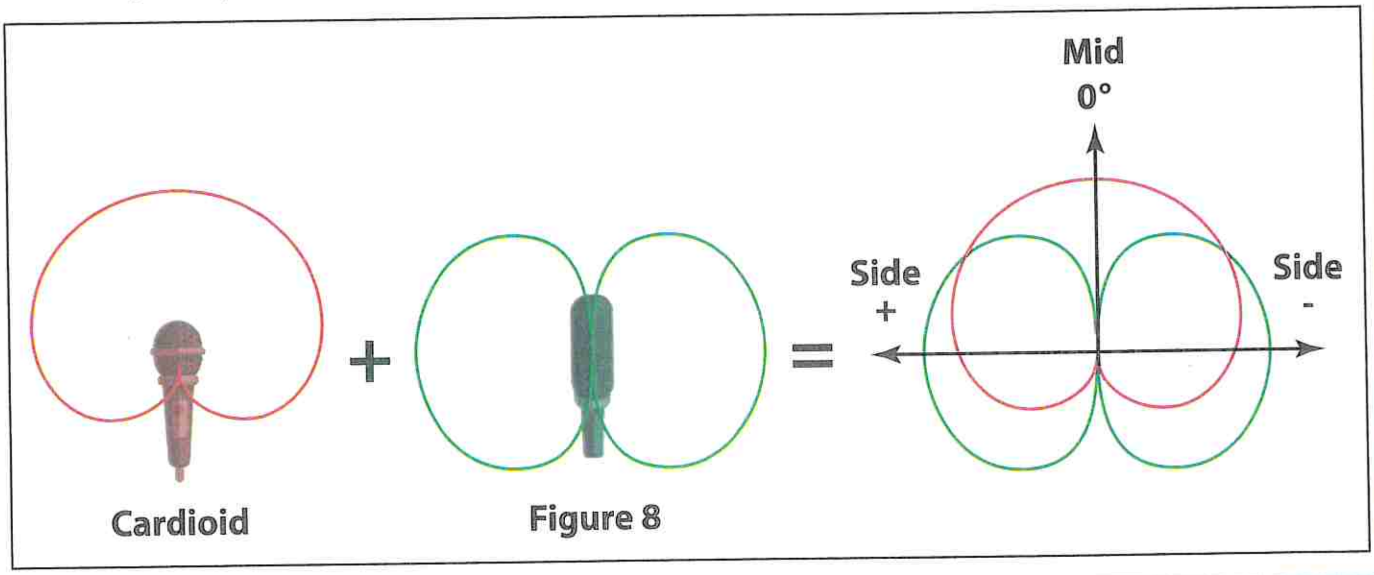
**Mid-Side (also known as MS)**
This technique requires one cardioid microphone and one figure of 8 microphone.

The two microphones are placed with the cardioid facing the desired sound source and the figure of 8 placed at a 90 degree perpendicular so that the front is pointing to the left and the rear is pointing to the right.

The combination of the two microphones is now picking up sound from both the sound source and the room around it.

The trick with this technique is in how the signals are handled at the mixing desk. The cardioid microphone is panned to the centre, while the figure of 8 microphone's signal is split, panned left and right with the polarity inverted on one side.

As the left and right will then be cancelled out when summed to mono, this makes the technique very good for mono compatibility.



## Sound and Audio Fundamentals

### Recording Technique

Remember that the marginal gains we're hoping for start at the very beginning of the chain with what is actually being recorded. There is no sense in spending lots of time and money recording something if the music or instruments themselves don't sound very good.

Once you are happy that the source sounds good, you can concentrate on the recording stage.

In order to maximise the quality of your recording, think about:

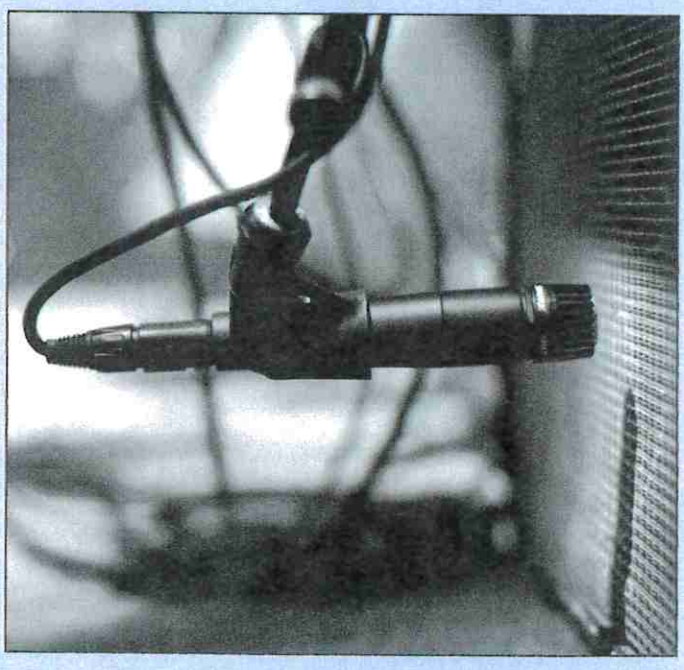
- **The room:** does the sound of the room compliment the recording?
- **The microphone type:** dynamic, condenser, valve or ribbon?
- **The microphone make and model:** cheap or expensive?
- **The microphone position:** close, ambient, or a stereo technique?

Here is a summary of the microphone techniques you might like to use:

**Technique:**
Mono close mic

**Description:**
One microphone placed close to the source.

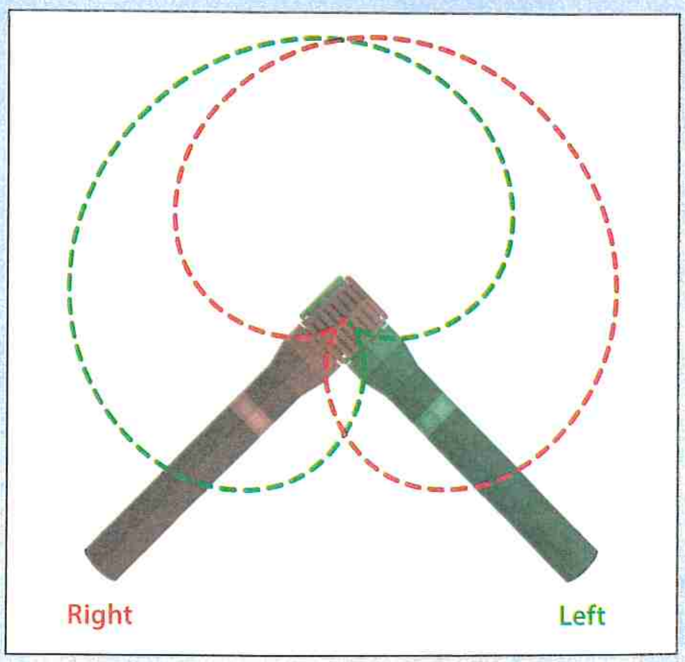
**Sound:**
Close, detailed, intimate, with very little room sound.



**Technique:**
Coincident (XY) Pair

**Description:**
Two cardioid microphones, placed with the capsules adjacent to each other at an angle of 90–130 degrees.

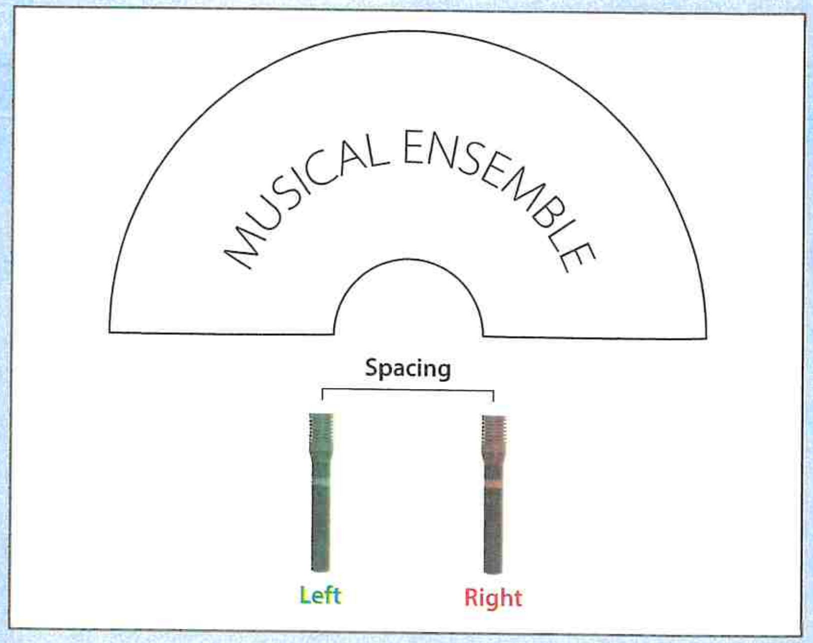
**Sound:**
Stereo, with good detail and focus on source. The sound arrives at both capsules at the same time, so good mono compatibility due to phase coherence.



**Technique:**
Spaced Pair (AB)

**Description:**
Two omnidirectional microphones (or cardioid if omnis aren't available) spaced apart by several feet.

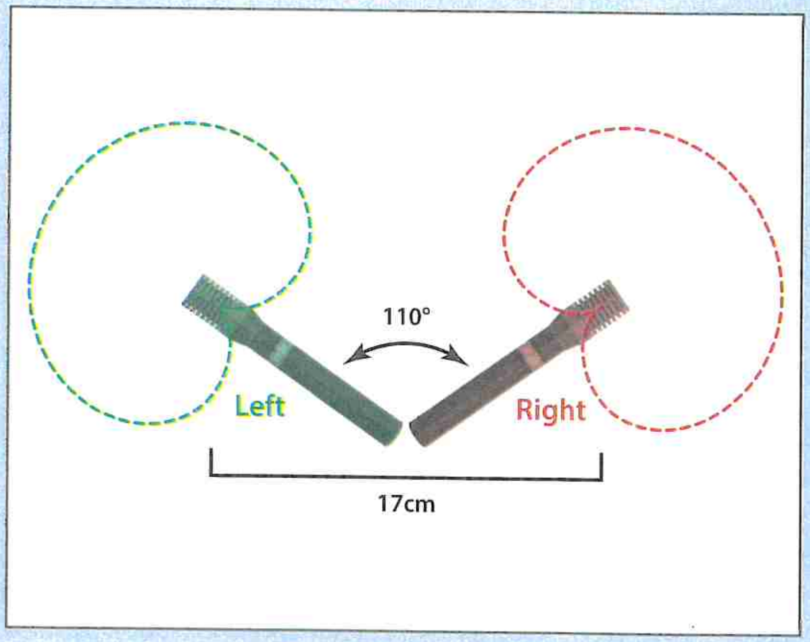
**Sound:**
Stereo, with a wide stereo image, good balance of room sound and source which can be adjusted by moving the microphones closer or further from the source.




**Technique:**
Near coincident pair (ORTF)

**Description:**
Two cardioid microphones, placed with each capsule pointing away from the other about 18cm apart.

**Sound:**
Similar to coincident pair, but with a slightly wider stereo image. Not quite as mono compatible due to the time distance between the capsules.




**Technique:**
Mid-Side

**Description:**
One cardioid pointing directly at the source, and a figure of 8 microphone placed perpendicular to capture the left and right stereo image. Requires coding and decoding by the mixing desk (see Glossary).

**Sound:**
When used in stereo the image is very good, with good focus on the source due to the cardioid. Not as wide an image as spaced pair. Good mono compatibility because left and right will phase cancel.

## Sound and Audio Fundamentals

### Microphone Technique: Recording a Drum Kit

One of the most complex recordings you might undertake is to record a drum kit. This is a challenging task which takes practice and experimentation to get right.

The challenge comes from the number of microphones needed. There are numerous theories on how this should be approached, none of which are necessarily right or wrong.

Here are a couple of ideas to get you started.

**The Glyn Johns technique.**
Glyn Johns is a legendary sound engineer who worked with numerous big names in music including The Beatles, The Who, The Rolling Stones and Led Zeppelin. The drum sound he achieved with these acts was very distinctive, particularly the sound he achieved when recording John Bonham's performances for Led Zeppelin.

What makes this technique special is that only four microphones are required.

The arrangement of the microphones is as follows:

**Microphone 1:**
Should be placed close to the kick drum. Use a dynamic microphone that can cope with the high levels, such as an AKG D112.

**Microphone 2:**
Should be placed close to the snare drum, pointing at the place where the drummer is striking the skin. Be careful not to get the microphone so close that the drummer hits it. Use a dynamic microphone such as a Shure SM57.

**Microphone 3:**
Should be placed on a high microphone stand above the drum kit, more or less central, pointing down towards the snare drum. A condenser microphone would be appropriate here as it will provide the clarity required for the cymbals.

**Microphone 4:**
Should be placed to the side of the drum kit, beyond the floor tom also facing the snare drum. This should also be a condenser microphone.

The important thing to remember to make this technique effective, is to ensure that both microphone 3 and 4 are the exact same distance from the snare drum. This will ensure that the snare drum is the focus of the sound and will help manage the phase relationship.

Record each of these microphones to different tracks in your DAW, then when mixing, pan the kick and snare microphones to the centre, pan microphone 3 to half left and microphone 4 to full right. This will create a nice stereo image with a powerful sound.

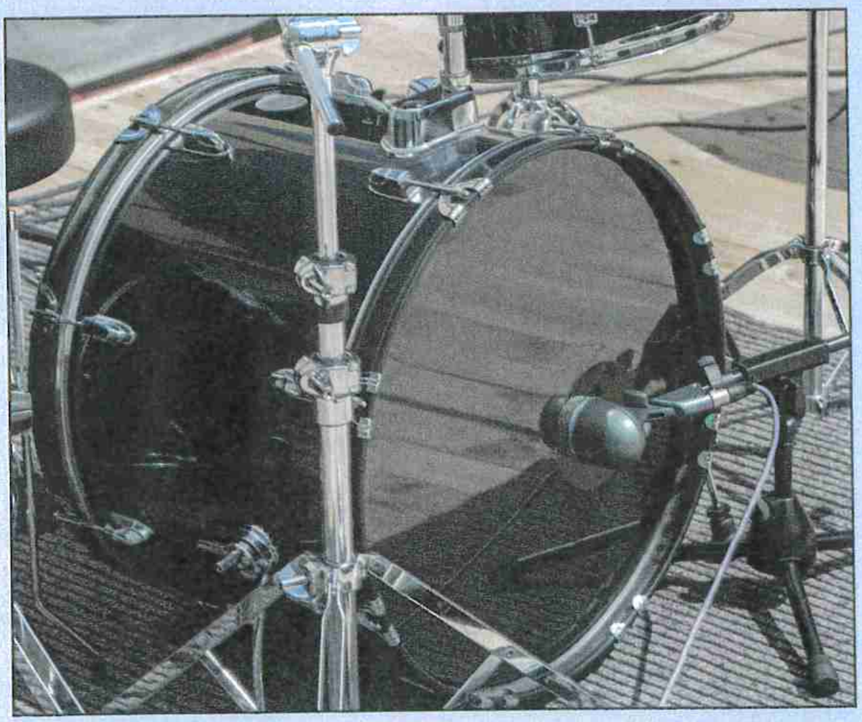
This is of course just a starting point and you should tweak and adjust your microphone positions until you hear it working well. Remember to always use your ears to make that judgement, not what it looks like.

### More Microphones

Another common drum recording technique is to use a further combination of close microphones and ambient microphones to allow more flexibility in the sound achieved.

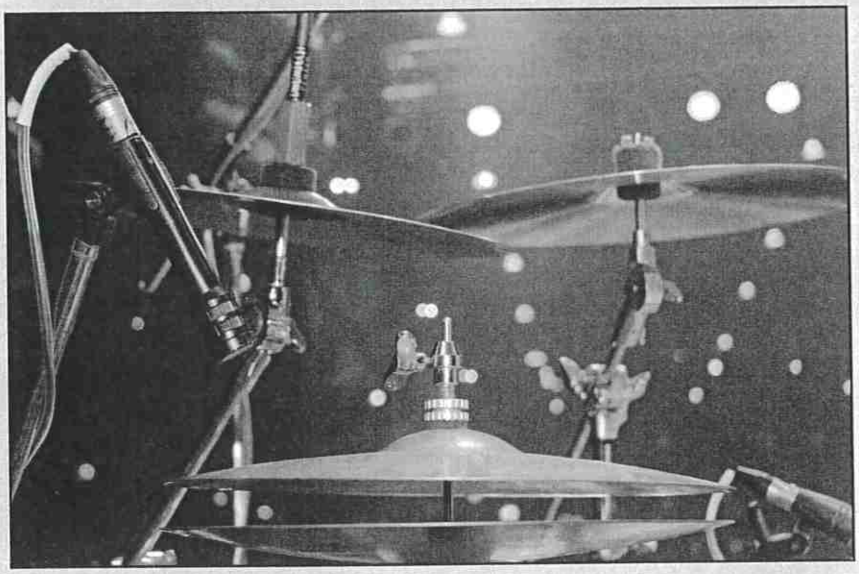
Here is an example of how you might approach recording drums with more microphones.

| Channel | Position | Type | Example | Tip |
|---|---|---|---|---|
| Kick Drum | Inside drum, close to where the beater hits the skin. | Dynamic | AKG D112 | For a clicky sound, use a hard beater or attach a penny to the skin where the beater makes contact. |

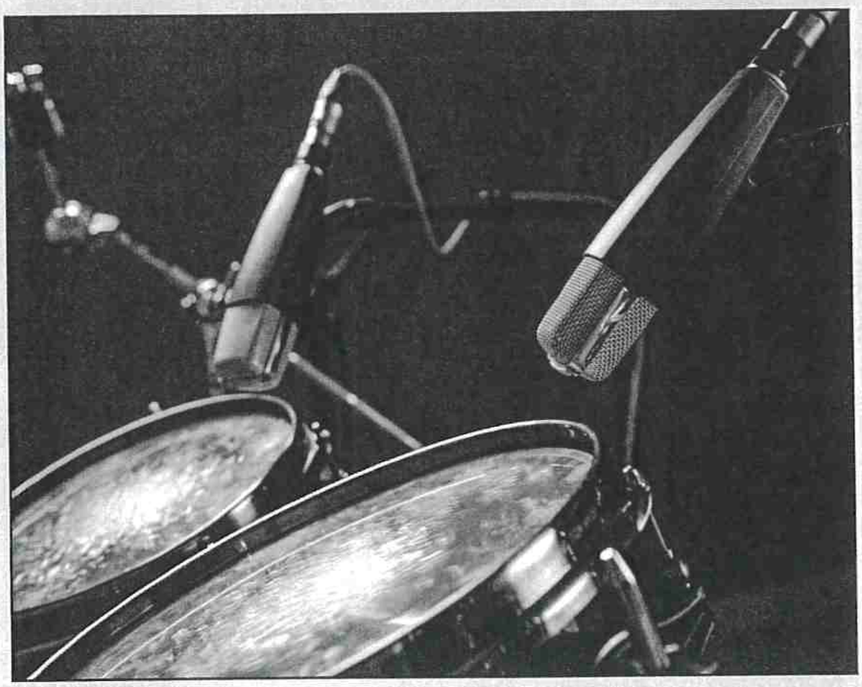


| | | | | |
|---|---|---|---|---|
| Snare | As close as possible. Pointing towards the place where the stick strikes the skin. | Dynamic | Shure SM57 | Point away from the hi-hat to reduce spill. |

| | | | | |
|---|---|---|---|---|
| **Hi-Hat** | Pointing towards the place where the stick hits the top cymbal. | Condenser | AKG C451 | Point away from the rest of the kit, straight down at the cymbal to reduce spill. |

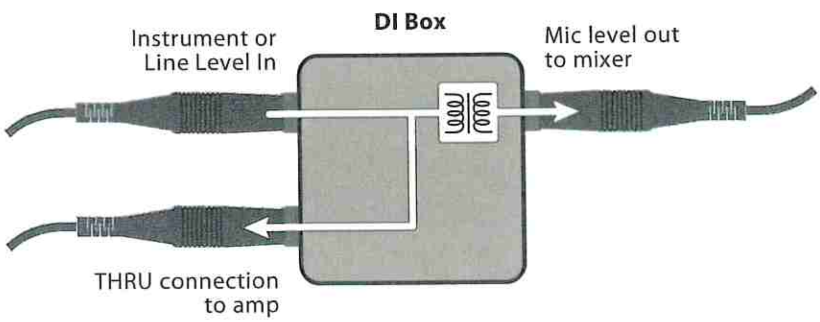


| | | | | |
|---|---|---|---|---|
| **Tom 1** | As close as possible. Pointing towards the place where the stick strikes the skin. | Dynamic | Shure SM57 | If there are cymbals nearby, then point the microphone away from them to reduce spill. |
| **Tom 2** | As close as possible. Pointing towards the place where the stick strikes the skin. | Dynamic | Shure SM57 | If there are cymbals nearby, then point the microphone away from them to reduce spill. |



| | | | | |
|---|---|---|---|---|
| **Floor Tom** | As close as possible. Pointing towards the place where the stick strikes the skin. | Dynamic | Sennheiser MD421 | If there are cymbals nearby, then point the microphone away from them to reduce spill. |

DI Box

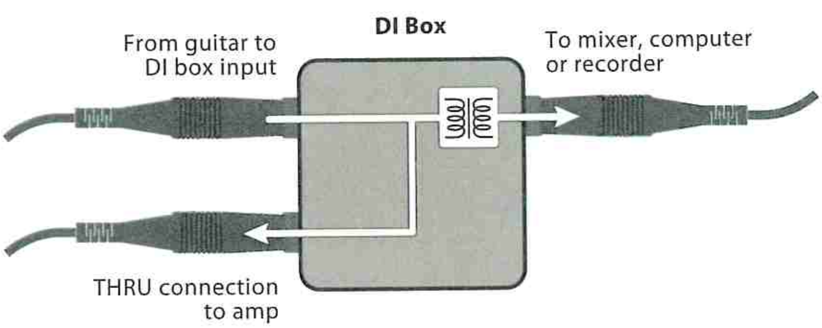




If a DI box is unavailable, you can use a dedicated balanced to unbalanced cable. In this cable the cold signal is connected to the ground at the other end. This won't eliminate any hum from a ground loop as this would require the transformer in the balancing circuit.

DI boxes can be very useful devices, which can also work in the opposite direction. For example, re-amping guitars.

Re-amping is the process of taking a recorded guitar signal and sending it to an amplifier to shape the required guitar sound. The best results are achieved by recording the guitar clean into the recording device by using the instrument input of a DI box, then when you want to add the amplifier you send the signal back out via a DI box to convert the signal back to instrument level for the amplifier input.

It's quite common for studio engineers to use a DI box to split the signal coming from the guitar when recording, so that the guitar is connected to the amplifier at the same time as being recorded clean directly into the DAW. This enables the engineer to record the sound that the guitarist likes through the amplifier, while leaving flexibility for the tone to be changed at a later date.



### Nominal Line Level

The nominal line level is the invisible line that runs through all the equipment in the studio, defining how we measure the signal level. If this didn't exist we could assume the signal level on one device is '5', while we define the same level on another device as '8'.

Having a nominal line level means that everything is being measured on the same scale, which helps us control the signal level, maximise our signal to noise ratio and prevent distortion.

There are two common reference levels used in audio equipment:

- +4dBu
- -10dBv

**+4dBu**
This is considered the *professional level*, and is most common in commercial studio equipment. Most balanced equipment will operate at this level.

**-10dBv**
This is considered *consumer level*, although many professional pieces of equipment also operate at this level, either because they are of a certain age, or they may have a switch to enable you to flick between +4 and -10 depending on the kind of equipment you're connecting it to.

There are two different ways of measuring level, one of which is measured in dBu and the other in dBv. The difference in level between the two is approximately 12dB (not to be confused as 14dB – remember that they are measured using different units).

Sometimes you may need to connect a piece of balanced equipment to a piece of unbalanced equipment and neither device has a +4/-10 switch. For example, connecting the audio outputs of a keyboard to the microphone input of a mixing desk.

In this case a DI box (direct injection) can be used to match the levels of the two devices.

**DI Box**

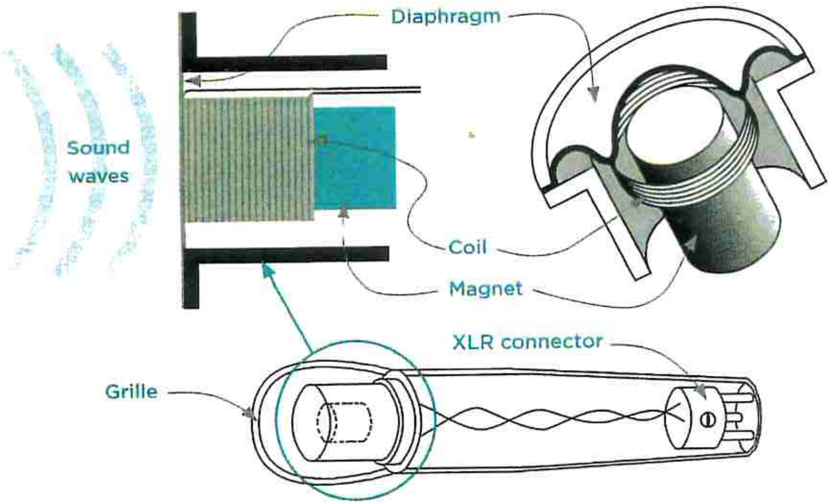
# Principles and practice

## Capture of sound

- A microphone is a **transducer**; a device that converts between different types of energy
- In this case, it is converting sound, and thus variations in air pressure into electrical energy
- The sensitive transducer element of a microphone is called its **capsule**
- Different types of microphone work in different ways; you need to know about dynamic, condenser and ribbon microphones.

### Dynamic microphones

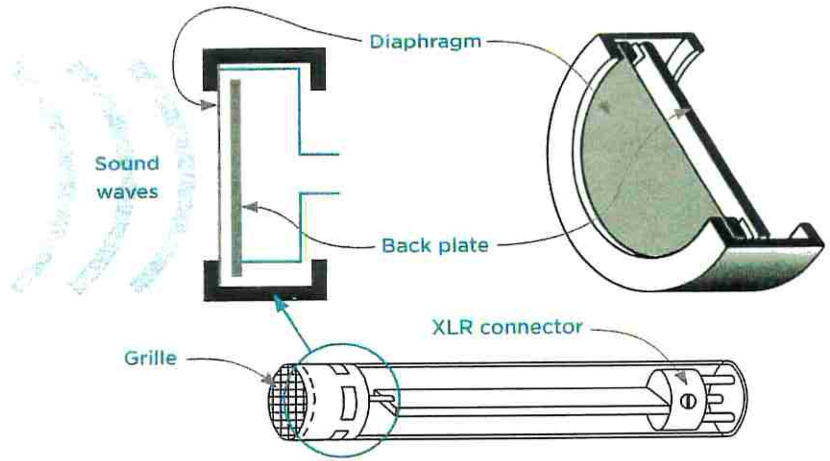
- **Dynamic microphones** are sometimes known as **moving coil microphones**
- A movable coil is positioned in a magnetic field, attached to a **diaphragm**
- The diaphragm is a thin piece of material that moves in response to the changes in air pressure
- When sound enters the microphone, the diaphragm vibrates
- The coil moves in the magnetic field and a varying electrical current is induced that is proportional to the changes in air pressure
- This is called **electromagnetic induction**.






### Condenser microphones

- **Condenser microphones** contain a capacitor, which in turn consists of two plates
- Sound causes one of the plates to vibrate, which functions as a **diaphragm**
- When the diaphragm vibrates, the gap between the two plates changes
- The capacitor plates are powered; therefore, moving the diaphragm causes a change in **capacitance** and thus a current flows.

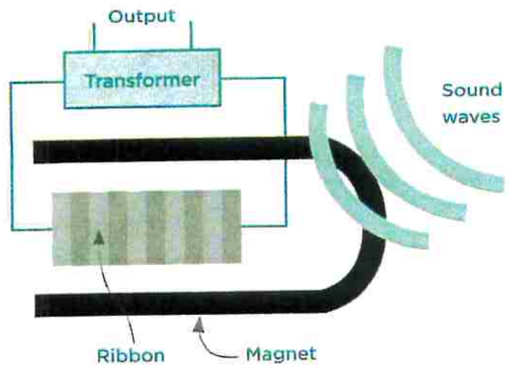



### Comparing dynamic and condenser microphones

| Dynamic microphones | Condenser microphones |
|---|---|
| ■ Generally inexpensive<br>■ Robust<br>■ Can withstand high SPL/ volume<br>■ Resistant to moisture<br>■ Good for live use<br>■ Does not require **phantom power**<br>■ Limited HF response; suitable for bass instruments. | ■ Sensitive, giving effective **capture** of quiet sounds<br>■ Flat and accurate **frequency responses**<br>■ Able to capture a wide frequency range<br>■ Generally able to capture a brighter signal than **dynamic microphones**<br>■ Good **signal-to-noise ratio**; high output volume and thus low **noise**<br>■ Wide dynamic range<br>■ Suitable for most studio work. |

## Ribbon microphones

- Sound vibrations disturb a metallic ribbon suspended in a magnetic field
- This generates a voltage that is proportional to the movement of the ribbon
- 'Warm' sound when used as a close mic with emphasised low frequencies
- Fragile and often very expensive
- Damaged or broken by phantom power.




## Pre-amps

- A **pre-amp** converts a signal into a workable **line level**
- There are a number of controls on a pre-amp that help us to work with microphones and recording equipment
- The table below shows some of the controls you often find on pre-amps.

| | |
|---|---|
| Phantom power | 48V from a **mixing desk** or an **audio interface** to provide power for **condenser microphones** and DI boxes. |
| Gain | The amount of boost applied to the pre-amp stage of an audio channel. Used either to boost signals to an operable level or to boost beyond that point to drive a signal into **distortion** for musical purposes. |
| Pad | A switch on equipment that **attenuates** the gain by a set amount to prevent **clipping**. |
| High pass filter | A type of filter that removes only lower frequencies below a set cutoff frequency and allows high frequencies through unaffected. Sometimes known as a '**rumble filter**'. |
| Polarity | Inverts the polarity of the signal. |
| Clip/Activity LEDs | These will illuminate when a signal is clipping/distorting, and when a signal is going through the channel. |

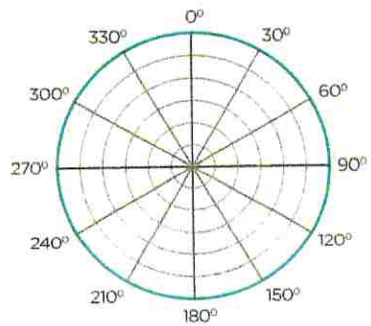
## Phantom power

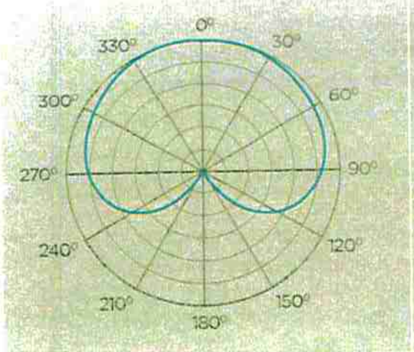
- **Phantom power** is required to charge the capacitor of a **condenser microphone**, and for its internal pre-amp
- Some condenser microphones can also be powered by a battery
- Many mixing desks and **audio interfaces** allow you to switch phantom power (48V) on and off; this is important as it can damage some equipment (e.g. ribbon mics).

## Polar patterns

- A microphone's **polar pattern** describes how it picks up sound from around the **capsule**.

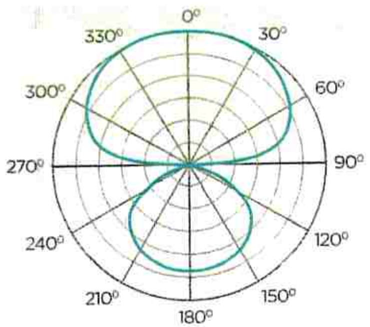
Although 2D on the page, the diagrams below represent how a microphone picks up sound in three dimensions:

| Polar pattern | Characteristics |
|---|---|
|  | **Omnidirectional**<br>- Picks up sound from all around the **capsule**<br>- **Captures** room **ambience**<br>- Provides little isolation so can lead to spill being captured<br>- Useful if the space sounds nice or the instrumentalists are all around the microphone. |




**Cardioid**

- Rejects sound from behind the microphone
- Minimises the amount of **reverb** and/or **noise** from behind the microphone that is captured.
- Used for most close-mic work.




**Hypercardioid**

- Picks up sound from in front of the capsule and gives some capture of those behind
- Effective at providing partial isolation from other instruments but capturing some of the ambience of the room.

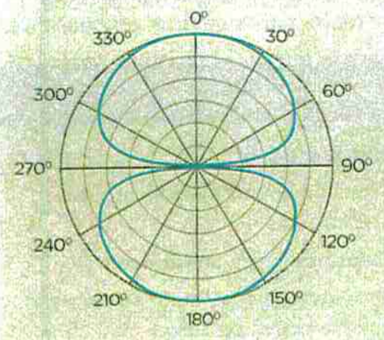



**Figure-of-8**

- Picks up sound from in front of and behind the capsule
- They are used in M/S (mid-side) recording techniques to give a stereo image of the sound
- Figure-of-8 microphones strongly reject sound from the sides of the capsule.

- Directional **polar patterns** all exhibit the **proximity effect**.

Turn to page 19 to learn more about the proximity effect.




- Engineers can avoid capturing unwanted background **noise** and spill by:
  - Ensuring performer(s) wear (closed-back) headphones
  - Keeping the monitor mix relatively quiet in headphones
  - Using acoustic screens/isolation booths
  - Making use of **overdubbing** and directional microphones.

## Frequency response

- A microphone's **frequency response** tells us the frequencies it picks up
- It is common to see a graph of a microphone's frequency response
- The flatter the graph, the more 'true' the sound is
- If a graph has any particular **peaks**, this means that those frequencies are captured louder than they are in 'real-life'.

Turn to page 89 to find out more about graphs of frequency response.

## Transient response

- This is how quickly the **diaphragm** can move when disturbed by a vibration
- Small diaphragm **condenser microphones** tend to have the fastest **transient response** as they have the lightest and easiest to move diaphragm
- This means that their high **frequency response** is often better
- Because the diaphragm in a **dynamic microphone** is connected to a heavy coil, it does not move as quickly. This is particularly true of large diaphragm dynamic microphones used to record instruments such as kick drums
- This can introduce a form of acoustic **compression**.

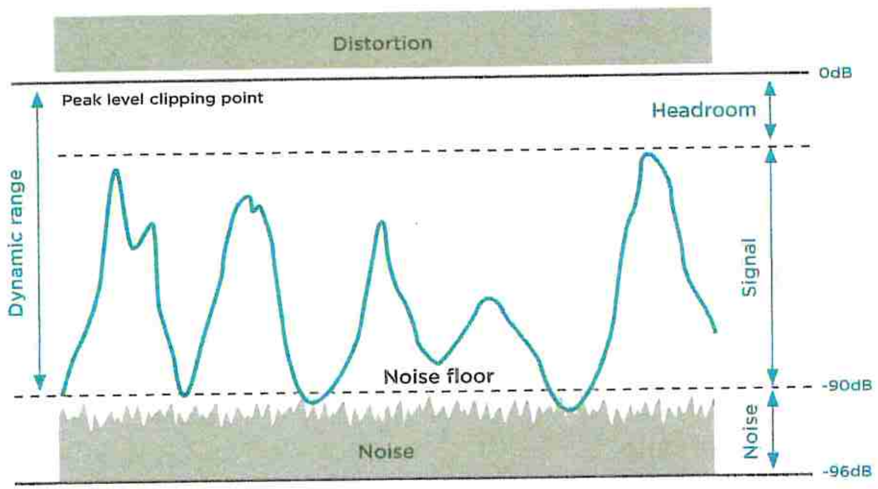
## Setting the gain

- When recording, you should carefully consider your gain structure
- This means that you will use the dynamic range of audio equipment to its best advantage in order to minimise **noise** and unwanted **distortion**
- To do this, it is necessary to set appropriate **gain levels** on each piece of equipment
- If the volume on an electric piano is set at a low level, and the **audio interface** gain is turned up to the maximum, you will **capture** lots of hiss
- If the volume on the electric piano is at its highest level, you risk capturing a distorted signal
- It is important to record at a good level in order to maximise the **signal-to-noise ratio**, but not sufficiently loud that the signal's **peaks** are clipped

- Normally, the first gain stage to adjust is that of the instrument itself, the next is the gain on the **pre-amp** or interface, and so on
- The gain at each stage of recording needs to be well above the noise floor but with enough headroom to be comfortably below the point of **distortion/clipping**.

## Signal-to-noise ratio

- The **signal-to-noise ratio** of a recording is how we describe the difference in volume between the signal you want to **capture** and the **noise**
- A poor signal-to-noise ratio will mean that noise is more prevalent in a recording
- Because the signal is quieter and is thus closer to the volume of the noise, the engineer will have to boost the volume of the signal, which also boosts the volume of the noise.




The diagram opposite represents a 16-bit system. In digital audio, 1 bit corresponds roughly to 6dB of signal-to-noise ratio. Thus, for a 16-bit system like CD, this gives a maximum of 96dB. **Dithering** adds noise at a lower volume than the least significant **bit** to prevent the quantisation distortion of quieter parts of the signal. Thus, in a 16-bit system, the usable dynamic range is around 90dB.

For more information on dithering, turn to page 88.



## Headroom

- Headroom is the gap between the loudest **peaks** of your mix or audio and the point at which digital **clipping** begins
- Sometimes, **analogue soft clipping** is used to add warmth to a recording, but digital clipping can sound harsh and unmusical.

## Microphone, line and instrument level

- Signals have a difference in impedance (sometimes labelled on equipment as 'Z'). You might see 'Lo-Z' and 'Hi-Z' inputs on audio interfaces and amplifiers. Lo-Z inputs are normally used for microphones, and Hi-Z for instruments such as electric and bass guitar (and are often labelled as such)
- DI boxes can convert a signal at instrument level to microphone level and from Hi-Z to Lo-Z
- Microphone and instrument levels are much lower than line level. Microphone level is the lowest, and instrument level is a little louder. Both need boost from a pre-amp to raise them up to a workable line level
- This is most important when connecting different pieces of equipment together; connecting a line level source to a microphone level input will cause distortion because the signal is louder than the input is designed to accept
- Equally, the signal from an instrument such as an electric guitar plugged into a line input will be much quieter than the input is designed to accept; the signal will require lots of boost and the resulting output will have a poor signal-to-noise ratio
- The impedance also needs to match because as well as a level/noise difference, frequency response will be adversely affected if this is not the case
- Therefore, it is important to connect instruments to the correct pre-amp input – labelled 'instrument' rather than 'line'
- Microphone level signals tend to travel through XLR connectors, and instrument level signals through TS jacks
- Line level signals often use balanced TRS jack connectors, or unbalanced phono (often for consumer audio).

## Proximity effect

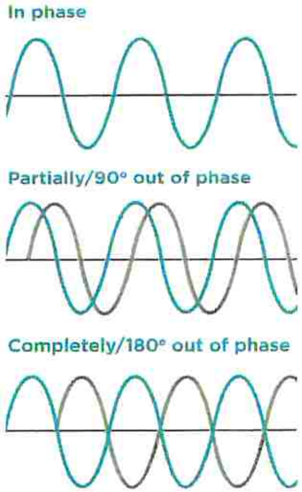
- The increase of low frequencies depending on how close the microphone is to the sound source. When closer, there will be an increase in the captured signal's lower frequencies
- Particularly apparent when recording male voices and acoustic guitars
- Can be used positively for kick drums and bass guitars
- Directional microphones exhibit the **proximity effect**
- EQ or an HPF/rumble filter can be used to reduce its effect, or you can alter the microphone position, moving it further away when capturing the sound.

**Microphones are often mounted in an elastic suspension mount/cradle to isolate against stand vibrations and avoid capture of low frequency rumble.**

## Hiss and hum

- Filters can be used to remove unwanted low or high **frequency** noise
- **Low pass filters** could be used to remove hiss
- **High pass filters** could be used to remove hum
- A parametric EQ with a narrow Q could be used to remove a specific frequency that is causing a problem.

## Phase




- If sound waves are in phase, then their peaks and troughs line up
- If sound waves are out of phase then the peaks and troughs do not line up and cause destructive interference, which sometimes can completely cancel out the sound of the wave
- Phase must be considered in any situation where you are using lots of microphones to record the same sound source, for example during drum recording
- It is particularly problematic if you are recording using two microphones on opposite sides of a drum (e.g. the snare)
-  These switches are often seen on analogue mixing desks. They are most commonly referred to as phase switches, but actually invert the polarity of the signal.

The diagram above shows various **phase differences** compared to the original wave.

### POLARITY AND PHASE

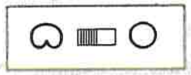
The terms polarity and phase actually refer to two different things. Phase implies a shift in time relative to an initial wave, whereas polarity refers to the reversal of two connections on a cable, for example the hot and cold on an XLR, the tip and ring on a jack or the positive and negative connectors on a speaker.

This is what happens with a polarity switch; the polarity of the signal is reversed. However, you will commonly hear this process referred to as 'inverting' or 'flipping' the phase, as in the Logic 'Gain' plug-in.



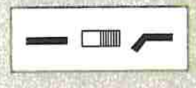
## Microphone switches

**Polar pattern**



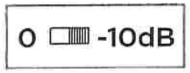
- The polar pattern describes how sound is picked up around the capsule
- On the switch shown, the microphone is switched to a cardioid polar pattern; you could also choose omnidirectional.

**High pass filter**



- Removes all frequencies below the cutoff. The cutoff frequency tends to be fixed between 80-150Hz
- This is sometimes known as a **rumble filter**.

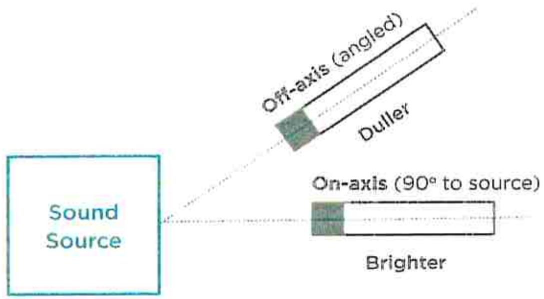
**Pad**




- Changes the sensitivity of the microphone
- On this microphone, you can reduce the sensitivity by 10dB.

## On and off axis microphone placement

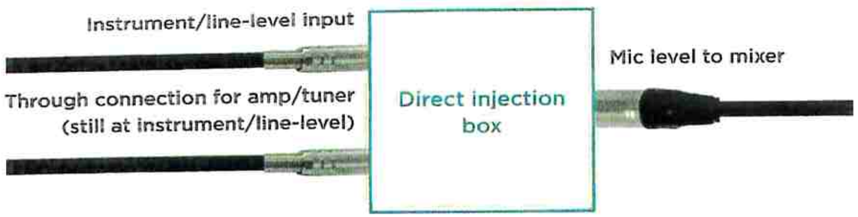
- Placing the microphone at different angles changes the range of frequencies that are captured
- Placing the microphone **on-axis** will **capture** a brighter sound and **off-axis** will make it duller.




## Direct injection

- A **DI** box converts a signal at instrument or **line level** to microphone level, and **unbalanced** signals to **balanced** signals
- DI boxes are used to eliminate the need to mic up electronic instruments, giving a direct connection to an **audio interface** or mixer

- Active DI boxes require **phantom power**, or a 9V battery, whereas passive DI boxes do not require external power.




## Plosive sounds

- **Plosive** sounds have a strong initial transient ('p','d') which can create a large disturbance in air pressure on the **diaphragm** and a 'pop' sound
- A pop filter is used to disperse the air more evenly to avoid such a quick and large diaphragm movement
- It is possible to reduce the impact of plosive sounds using EQ and **compression**, but by far the best solution is to re-record, or comp in the word or phrase from another part of the song.

# Recording techniques

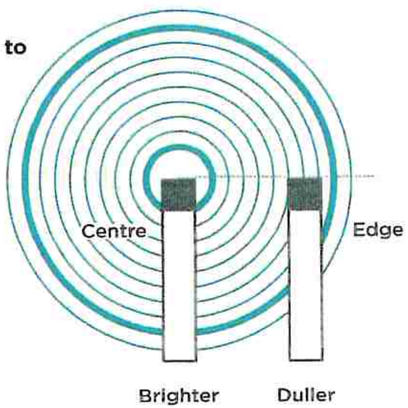
**You will already have used many of these recording techniques when completing your coursework. However, you may also be asked questions that relate to different microphone techniques and different instruments in your Component 3 and Component 4 exams.**

**Below is a brief summary of the key points to remember when close and stereo miking different instruments and groups.**

## Close mic techniques

### Electric guitar/bass guitar

- **Dynamic microphone** (for high **SPL**) with a cardioid **polar pattern** placed in front of the **amplifier**, around 15cm away (close to speaker grille)








- The placement on speaker gives brightness – centre of cone is bright, edge of cone is duller
- If the microphone is **on-axis**, the sound will be brighter than when it is **off-axis**.

See page 21 to find out about on and off axis mic positioning.

- You could also use a **condenser microphone** with **pad** switched on
- A DI box could be used, and the DI signal processed using amp plug-ins in a **DAW**
- You could also use both a microphone and a DI box and decide on the best signal later/use both; it is important to look at the potentially differing phase characteristics of each signal
- To add presence, a second microphone can be added; this is usually a condenser set about 5m away, depending on the room acoustics and sound required.

### Acoustic guitar

- Acoustic instruments are generally recorded with condenser microphones because of their wide **frequency response** and accurate reproduction of high frequency content
- Condenser microphones also capture the wide dynamic range of the guitar with good sensitivity
- A small diaphragm condenser microphone will give a fast transient response, and thus will likely best capture the full frequency range and brightness of the instrument
- A large diaphragm condenser microphone could be used for greater sensitivity
- A cardioid condenser microphone should be placed approximately 15-60cm in front of the 12th fret of the guitar (where the neck meets the body)
- You could also use two small diaphragm condenser microphones pointing at the sound hole (around 30cm away) and at the fretboard (around the 11th fret).

### Brass and woodwind

| Instrument | Microphone | Technique |
|---|---|---|
| **Trumpet/ trombone** | Dynamic or condenser | Place in front of the bell, or slightly to the side for a more mellow sound. Use pad on condenser. |
| **Saxophone** | Dynamic or condenser | Condenser for slight distance (pad may be necessary). Dynamic for spot miking – be careful to avoid the sound of the keys. |
| **Flute/ clarinet** | Condenser | Sound emanates from the key holes and mouthpiece so a condenser placed above is best for recording. |

**Clip on microphones for trumpet, trombone and saxophone are commonly used for convenience in live sound, but are seen less often in studios.**

## Grand piano

- Open the lid of the piano and use **condenser microphones** to **capture** the brightness of the top strings
- The microphones should be far enough away so as not to emphasise a few single notes. About 30cm is normally a good distance, with a microphone used for the treble and one for the bass
- The condensers should use shock-mounts and be pointed towards the hammers. An **XY pair** just outside the piano give as more realistic and less close sound
- A single microphone could also be positioned over where the strings cross. This method is also commonly used in live sound.

## Upright piano

- Remove the front panel of the piano and mic the strings with a spaced or XY pair of condensers
- You could also mic the soundboard of the piano if it is away from nearby walls; if doing so, the microphones pick up less **noise** from the pedal thumps and piano action noises, but the captured sound will be duller.

## Vocals

- (Large **diaphragm**) **condenser microphone** (to **capture** full **frequency** range/ wide dynamic range)



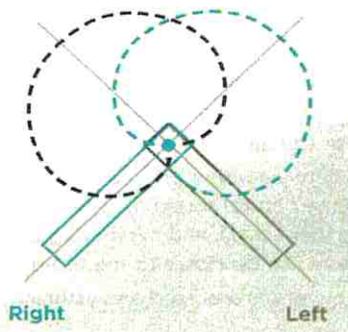
- Performer could be up to 30cm from **pop shield**, which itself should be 8-12cm from the microphone itself
- Use shock mount to isolate the microphone from vibrations through stand
- Use pop shield to avoid capture of **plosive** sounds.

## Drum kit

- Spot microphones are widely used for the different elements of a drum kit
- However, it is not easy to avoid spill between microphones, and drum miking is a compromise between capturing the natural sound of the drum (often achieved when the microphone is further away), and minimising spill and **ambience** (meaning the microphone has to be closer to the drum)
- Prepare and tune the kit, ensuring any rattles are fixed and listening to the sound of each drum
- Acoustic screens can be used to reduce spill if other musicians are in the live room
- For the kick drum, use a large **diaphragm** dynamic microphone to pick up the low frequencies. Sometimes, another microphone is used on the other side of the kick drum. These two mics can be used together to pick up the sound from the port and the sound of the beater. Polarity inversion may be required to avoid cancellation (often this is referred to as 'phase inversion' or 'flipping the phase')
- Use **dynamic microphones** such as Shure SM57s or Sennheiser MD421s for the toms; position the microphones up to around 15cm away from the skins at approximately 30°. It is also common to use clip on dynamic microphones for live sound
- A dynamic microphone should be used for the snare; alternatively, a condenser with **pad** switched on to reduce sensitivity. Snares can be miked from above and/or below; below emphasises the rattle of the wires (but issues can arise with spill from the kick drum) and above produces a rounder sound. You might need to use the polarity (often referred to as 'phase') switch to avoid cancellation if you're using a microphone above and below
- A small diaphragm condenser is suitable for the hi hat; it needs to be placed above the top cymbal between 10-15cm away, pointing down to avoid air blasting. If the microphone is placed too close, issues may arise with proximity effect
- The main cymbal sound and overall kit sound is best captured with a pair of overhead condensers. You can also add spot mics for other cymbals such as the ride and the crash
- A **spaced pair** gives a wider stereo picture but could leave a hole in the middle; an XY or **mid-side pair** can be used to avoid this, and also have better **mono** compatibility
- The recordings of each part of the whole kit should be checked against each other for phase cancellation issues.

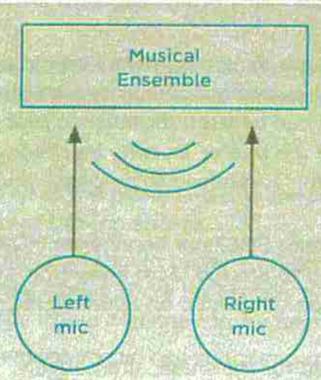
## Stereo mic techniques

### Coincident (XY) pair



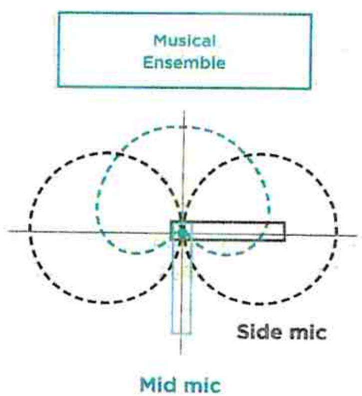

- Two cardioid microphones placed so that the **capsules** are right next to each other
- This combines the **polar pattern** of the two capsules to create a stereo image
- Because the mics are basically touching, a good **mono** compatibility is maintained.

### Spaced (AB) pair




- Two **omnidirectional** microphones placed around 30-50cm apart (depending on ensemble size)
- Useful for the recording of large ensembles and performances that require a sense of **ambience**.

### Mid-side (MS) pair

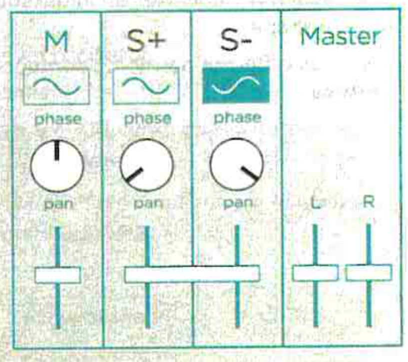



- A cardioid and a figure 8 microphone
- Set up at 90° to one another
- The cardioid microphone should point straight at the sound source, and the figure 8 microphone picks up the sound from the sides.



## Processing mid-side recordings

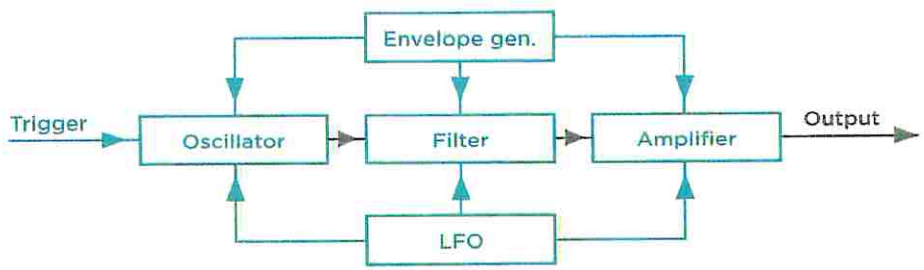
- At the mixer or on the **DAW**, the signal captured from the sides on the figure-of-8 microphone should be duplicated, and one version phase inverted
- They should then be hard panned in opposite directions at the same level
- If the side tracks are panned centrally, they will completely cancel out, and therefore mid-side recordings exhibit excellent **mono** compatibility
- Changing the volume of the side tracks will adjust the stereo width.




## Synthesisers

**A synthesiser is an electronic sound generator capable of creating and manipulating synthetic sounds. It has become common to use synthesisers as DAW plug-ins, but the sounds, warmth and authenticity of vintage analogue equipment are highly regarded by many.**

### How does a synthesiser work?



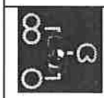
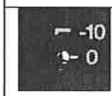
| Question Number | Question | Mark |
|---|---|---|
| **3(a)** | Reverb has been added to the vocal. Identify the reverb time. | **1** |
| | Acceptable Answers | |
| | **C** 2.0 seconds | |

| Question Number | Question | Mark |
|---|---|---|
| **3(b)(i)** | What is the main advantage of placing the vocalist close to the microphone? | **1** |
| | Acceptable Answers | |
| | Less spill / less reverb / less noise / better signal to noise ratio / wider dynamic range / more control over effects/ambience in mixing (1)<br>Proximity effect (1) | |

| Question Number | Question | Mark |
|---|---|---|
| **3(b)(ii)** | Identify two problems close mic'ing the vocal could introduce to the recording. How could these problems be reduced during the mix? | **4** |
| | Acceptable Answers | |
| | Uneven dynamics (1): compression (1).<br><br>Too dry (1): add reverb / delay (1)<br><br>Loud breaths / lip smacks (1): expander / gate / cut breaths out / volume automation (1)<br><br>Plosives / pops /p (& b) (1): HPF/100Hz filter / low shelf cut / volume automation / dynamic EQ reducing LF / multiband compression reducing LF / cut plosives out (1)<br><br>Sibilance / s (& t) (1): de-esser / EQ with high mid cut with narrow band / high cut EQ on reverb / volume automation / multiband compression reducing high mids / dynamic EQ reducing high mids (1)<br>NOT LPF / just 'EQ'<br><br>Proximity effect (1): HPF/100Hz filter / low shelf cut / dynamic EQ reducing LF / multiband compression reducing LF (1)<br><br><u>Headphone</u> spill (1): gate / expander / cut spill out manually between vocal phrases (1)<br><br>Max 2 for problems.<br>Max 2 for solutions. | |

| Question Number | Question | Mark |
| --- | --- | --- |
| 3a | The vocal was recorded with a condenser microphone using the switch settings shown in the table below.<br>Identify the switches, describe what they do and explain why the settings have been selected for this recording.<br>An example has been given. | 6 |

Acceptable Answers

| Switch | Identify the Switch | Describe what this switch does and explain why this setting has been selected |
| --- | --- | --- |
|   | Polar response / pick-up pattern / polar pattern (1) | Directional characteristics (1)<br>Figure of 8 / Cardioid / Omni (1)<br>Proximity effect (1)<br>Changes level of ambience / reverb / noise / spill (1)<br>Selects between capsules (1) |
|   | Pad / attenuator / sensitivity switch (1) | Reduces / alters sensitivity of microphone / volume / gain (1)<br>Reduces distortion (1)<br>Recording loud sounds (1)<br>dB (1)<br>Vocals not loud / not high SPL / mic needs to be at max sensitivity (1) |

| Question Number | Question | Mark |
|---|---|---|
| 3 | Complete the table below to describe how you would mic a singer to achieve a similar recording to that heard in "vocal.wav". Give reasons for your choices. An example is provided for you. | 6 |
| | Acceptable Answers | |

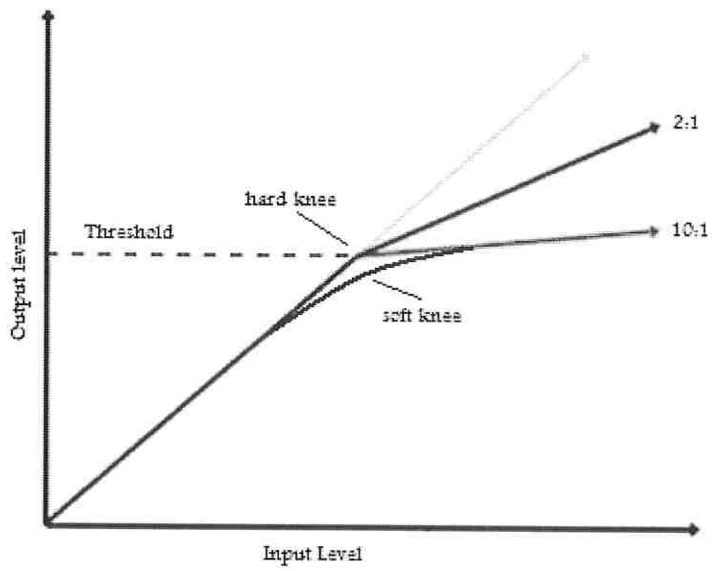
| | What you would choose | Reasons for choice |
|---|---|---|
| Microphone type | Condenser<br>Capacitor | Vocals have a <u>low</u> SPL (1) so would not distort or damage (1) a condenser microphone.<br>Good at picking up high frequency detail/wider frequency response/flat frequency response (1).<br>Sensitive (1)<br>Wide dynamic range/low noise (1) |
| | LDC | Fast transient response (1)<br>(LDC) is good for capturing warm tones of a male vocalist (1) softens transients (1). |
| | Dynamic<br>(1) | <u>High</u> SPL (1)<br>Max 1 because recorded using condensor<br>**(2)** |
| Microphone polar pattern | Cardioid<br>Hyper cardioid<br>Super cardioid<br>Uni-directional<br>**(1)** | Rejects background noise / spill (1)<br>Directional/pick ups in front (and sides) (1)<br>Less reverb / less ambience / dry (1)<br>NOT "only the sound of the vocal is picked up"<br><br>Proximity effect (1) adds warmth (1) by emphasising low frequencies (1)<br>Presence peaks / high mid range boost (1) helps the vocals cut through (1)<br>**(2)** |

The "What" box must be correct to award a mark in the "reason" box.
Award accurate diagrams for "What" box

**(Total for Question 3 = 6 marks)**

| Question Number | Question | Mark |
|---|---|---|
| 4a | Describe how you would mic up a standard drum kit for a rock band in a contemporary recording. Explain any decisions you make. How does this compare with mid-1960s drum recording technique? | 16 |
| | **Acceptable Answers** | |
| | *In this mark scheme, italics mean that the mark should not be credited multiple times.*<br><br>Underlined technical terms must be spelt correctly<br><br>Tune drum kit (1).<br>Dampen skins with gaffer tape / gels / cushions (1)<br>Remove unused parts of drum kit / rattles (1).<br>Screens to reduce spill (1).<br>Credit any discussion of room acoustic (1).<br>Place mics where they won't be hit (1).<br>Multiple mics used for better control of mixing / EQ / compression / processing / balance etc (1)<br><br>Directional microphones e.g. cardioid / figure of eight (1) to prevent spill (1).<br>Closer = less spill (1) and proximity effect (1) correct reference to low frequencies (1)<br>Further / higher = more natural / picks up sound of whole drum / more reverb (1)<br><br>Any correct discussion of phase/polarity (1)<br><br>**Kick/bass drum**<br>*Large diaphragm (1) to pick up low frequencies (1).*<br>*Dynamic / D112 / D6 / PG52 / beta 52 etc (1) high SPL / loud (1) to pick up low frequencies (1).*<br>Inside (1) less spill (1).<br>Up to 10 inches / 25cm / close from (beater) skin (1).<br>Additional mic placed in front / outside of kick drum (1) e.g. woofer speaker cone (1) for sub-bass (1) *Condenser / capacitor (1)*<br>Modify the beater / glue coin/credit card to the skin / clicky beater / upper mids to cut through the mix (1)<br><br>**Snare & toms**<br>*Clip-on mics (1)*<br>*Dynamic / SM57 / D40 / i5 (1) high SPL / loud (1) robust (in case it's hit) (1).*<br>*Condenser / capacitor / C1000 / C414 / C451 (1) for brightness / HF (1) pad (1).*<br>*Up to 6 inches / 15cm (1).*<br>Snare drum mic should be as far away as possible from hi-hat to prevent spill (1).<br>Snare bottom mic (1) to pick up the rattle / snare / wires (1).<br>Beware of spill from kick drum (1). | |

| | | |
|---|---|---|
| | **(Floor) tom**<br>*Large <u>diaphragm</u> (1) to pick up low frequencies (1).*<br>*Dynamic / D112 / PG52 / MD421 etc (1) high SPL / loud (1) to pick up low frequencies (1).*<br><br>**Hi-hat**<br>Not always needed in mix because of overheads / snare spill / importance of hi-hat in the song (1).<br>Small diaphragm (not 'pencil') (1) to pick up high frequencies (1).<br>*<u>Condenser / capacitor /</u> C1000 / NT5 / 414 (1) to pick up high frequencies (1).*<br>*Pad (1) to prevent distortion (1).*<br>*2-6 inches / 5-15cm (1)*<br>Above hi-hat to prevent air blasting (1).<br><br>**<u>Overheads</u> (1)**<br>Mostly to pick up cymbals / overall picture of kit / more reverb than the spot mics (1). *Stereo / left and right (1).*<br>Any ref to additional spot mic cymbals (1)<br>Stereo track in DAW / 1-2 / 3-4 pairs (1).<br>*Condenser / ribbon / C1000 / NT5 / 414 (1) to pick up high frequencies (1).*<br>*Pad (1) to prevent distortion (1).*<br>1-4ft / 30cm-120cm from cymbals / snare (1).<br>Equal distance from snare (1) so snare centre (of stereo picture) (1).<br>*Spaced pair (1)* gives wider picture (1).<br>X-Y / co-incident pair / Blumlein / ORTF (1), middle and side / M-S (1)<br>reduces hole in the middle / natural stereo [not just "stereo"] (1).<br>*Omni / figure of 8 (1).*<br>Glyn Johns (1) mic overhead and mic to side of kit (1)<br><br>Large diaphragm condenser/LDC popular for heavier music (1) - presence peak in high mid range accentuates snare (1)<br>Small diaphragm condenser/SDC (1) for fast transient response (1)<br>Balance between drums and cymbals improves with height / moving mics around side or behind drummer (1)<br><br>**Room**<br>*2 mics / stereo (1)*<br>*Spaced pair (1)*<br>*Condenser / ribbon / PZM / boundary (1) sensitive (1) low SPL / quiet (1).*<br>*Omni / figure of 8 (1).*<br>More than 6 ft / pointing away from drum kit (1) to reduce dry signal / give lots of reverb (1).<br><br>**Additional mics**<br>Extra mic for special effects / more compression / distortion (1) | |

| Question Number | Question | Mark |
|---|---|---|
| 4b | Figure 1 shows a valve compressor. Many of the controls are similar to those of a software plug-in. Explain the function of the controls and specifications that can be seen in the picture. | 16 |
| | Acceptable Answers | |
| | Underlined technical terms must be spelt correctly<br><br>All comments must relate to the correct knob/socket.<br><br>**GENERAL DESCRIPTION OF COMPRESSOR**<br>Reduces dynamic range/automatic volume control/evens out volumes (1)<br><br><br>Output AND Input correctly labeled (1)<br>Threshold correctly labeled (1)<br>Ratio correctly labeled (1) (*allow examples, e.g. 3:1*)<br>Hard knee / Soft knee correctly labeled (1)<br><br>**INSTRUMENT**<br>Jack / TS / tip-sleeve (1) usually used for instruments such as electric guitar/synths / DI (1).<br><br>**INPUT**<br>Pre-amp (1).<br><br>MIC 48V:<br>Phantom power (1) used to supply power to condenser microphones (1) or DI boxes (1).<br><br>MIC:<br>Used for dynamic / ribbon (1).<br><br>Credit any reference to impedance / resistance / sensitivity (1) | |

| | | |
|---|---|---|
| | **1960s drum recording**<br>Mono (1)<br>Fewer mics / not spot mic'ed/close mic'ed / any valid description of 60s mic'ing (1)<br>Further away / bigger room / brighter room / more ambience / reverb / more spill (1).<br>Indistinct kick drum / poor balance (1)<br>Less LF / HF (1).<br>Fewer tracks (1)<br>Often bounced/combined with other tracks (1)<br>Ribbons more common (1).<br>Valve mics more common (1) | |