Monitoring

- Monitoring is how we listen back to the audio being recorded.
- In a recording session, both the musicians and the engineer will need to monitor the recording.
- Performers will need to hear a backing track or other instruments to play in time with.
- On a mixing desk, monitor mixes can be set up using the Aux sends on each channel strip.
- On an Audio Interface, turning on the Direct Monitoring function will allow performers to hear their performance before it is processed by the DAW reducing latency.

Headphones

- Closed-back headphones are best for monitoring by performers.
- They allow less sound to escape, or spill, as the back of the speakers are fully enclosed.
- An Engineer could use open-back headphones.
- These give a more acoustically balanced sound as the speakers are not enclosed.

Speakers

- An Engineer might also use monitor speakers in the control room where there are no issues with spill.

The Recording Studio

Recording Studios often consist of 2 rooms - a Control Room and a Live Room.

Control Room

This is where:
- the producer, recording or mix engineer will be
- the audio is recorded to a DAW
- the audio will be edited
- the mix will be created
- instruments like electric guitar or bass are recorded using D.I

Equipment:
- Computer
- Audio Interface/mixing desk
- DAW
- Talkback Microphone
- Headphones
- Speakers
- Desk
- Acoustic Treatment
- Chairs/Stools
- Instruments

Live Room

This is where:
- the artist or musician(s) will be
- the microphones will be set up
- musicians will perform

Equipment:
- XLR Cables
- XLR Wall Box
- ¼ Jack Cables
- Microphones
- Microphone Stands
- Pop Filters
- Acoustic Treatment
- Chairs/Stools

Audio Interface

- Many Recording Studios will have a mixing desk as the Audio Interface.
- These often have 8 or 12 inputs allowing for a larger number of microphones to be used.
- More on Audio Interfaces & Mixing Desks can be found in CA2.

Pop Filter

- Provides a shield for the microphone to reduce plosives.
- It consists of a piece of semi-transparent material stretched across a frame

XLR Cables

- XLR cables have 3 pins in the male end of the cable and 3 sockets in the female end.
- They also have a locking mechanism to stop them from disconnecting accidentally.
- XLR cables are a balanced connection type

Mic Stands

- short stands can be used for recording kick drums and guitar amps.
- Taller stands are required for overhead microphones.
- Most stands will have a boom arm that will allow the microphone to be positioned close to the sound source.
- With a vocalist it can be advisable to use a fixed stand.
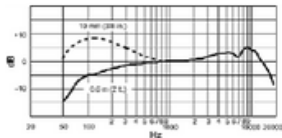
Microphones & DI

Dynamic Microphones

- Have a moving coil.
- When sound waves hit a dynamic microphone:
  - A magnetic coil moves back and forth.
  - This movement generates a small electrical voltage.

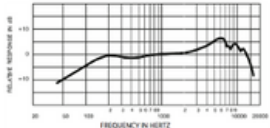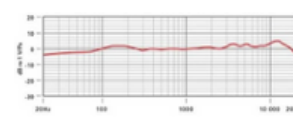Sound Waves   Dynamic Microphone   Electrical Energy
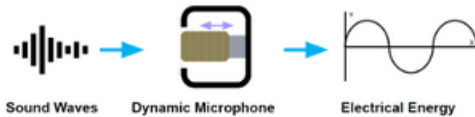
Frequency Response

- All microphones have different frequency responses.
- This means that they will cut or boost different frequencies.
- Even if they are the same brand of microphone, they will be slightly different.

Proximity Effect

- As you get closer to a microphone the sensitivity to the low end will increase.
- This is the proximity effect.
- This effect only occurs on cardioid and figure of 8 patterns.

Condenser Microphones

- Have a capacitor consisting of two charged magnetic plates.
- When sound waves hit a condenser microphone:
  - The distance between these two plates changes
  - This movement creates a change in voltage
- They require a power supply of +48 Volts to work, known as phantom power.
- Usually supplied by the audio interface through the XLR cable.

Sound Waves   Condenser Microphone   Electrical Energy
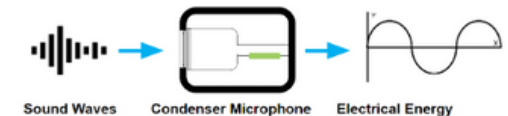
Direct Input (DI)

- A method of recording that does not use microphones to capture audio.
- Commonly used to capture electric bass, guitars and keyboards, which are plugged in using a Jack cable.
- The DI box converts the unbalanced signal level from the instrument to match the balanced output of a microphone, so it can be plugged into an XLR connection on a mixing desk or interface.
- A DI box will need phantom power to work.
- The link output can send a copy of the direct guitar signal to an amplifier.

Dynamic	Condenser
Can withstand high sound pressure levels (SPL)	Can generally handle lower sound pressure levels (SPL)
Very robust	More fragile
Does not require phantom power	Requires phantom power
Requires more gain: less sensitive to picking up faint and/or distant sounds	Requires less gain: more sensitive to picking up faint and/or distant sounds
Generally, narrower frequency range	Flatter frequency response
Curved frequency response	Generally, wider frequency range

Polar Patterns & Stereo Techniques

Polar Patterns

- All microphones have what are called polar patterns.
- Polar Patterns are represented by a diagram that shows where the microphone is more or less sensitive

CARDIOID   FIGURE-OF-EIGHT   OMNI
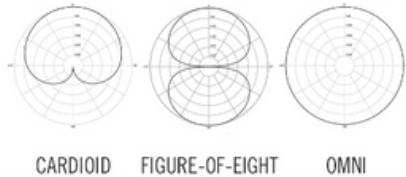
Cardioid

- Cardioid polar patterns are the most common polar pattern used in microphones and are most suitable to record a singular instrument or part of an instrument:
  - Voice (including voiceover and dialogue)
  - Acoustic guitar
  - Electric guitar amplifier
  - Drum Shells (Kick, Snare, Toms)
 - Brass/Woodwind

Figure of 8

- Figure of 8 polar patterns are useful to record two sound sources whilst only using one microphone.
- E.g. it would be suitable to record a duet such as voice and acoustic guitar.

Omnidirectional

- Omnidirectional patterns are useful to record an entire space equally.
- Omnidirectional would be suitable to record an entire ensemble in a room with only one microphone.

XY Stereo Technique

- Consists of two cardioid microphones set at a right angle.
- One microphone captures the left and the other the right side.
- Cardioid polar patterns combine to create a stereo image.
- Best used with condenser microphones because they have a flatter frequency response and a wider frequency range.
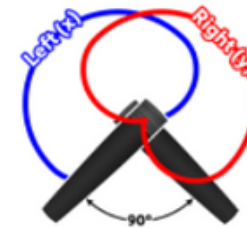
Spaced Pair (AB)

- Two cardioid or omnidirectional microphones spaced at least 20cm apart.
- Commonly used for drum overheads or capturing room and ambient sounds when recording an ensemble.

A/B Stereo Miking
7in / 17cm
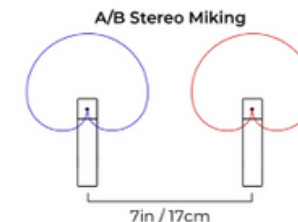
Planning, Health & Safety

Recording Session - Key Roles

- Musical artist: creating, developing and performing musical material.
- Engineer: in charge of setting up and managing the technical aspects of the recording.
- Producer: someone who is in charge of leading and directing a musical project.

Planning a Recording Session

When planning a recording session, you must consider:
- time constraints and timeline for completion
- the length of each session
- What you want to achieve in the session (agreed goals)
- What the hardware requirements are for the session (equipment required)
- What the software requirements are for the session (DAW)
- What the musical artist's requirements are

Multitrack Recorder

- Most modern recordings are recorded using a DAW as the Multitrack Recorder.
- Digital hardware recorders are still available and present a portable option but are limited in track number.
- Some recording studios may still offer the option to record to tape for artists who want to use this method.

Health & Safety

- Identifying hazards and risks when using a DAW was covered in CA2.
- The same hazards apply to the recording studio.
- However, the risk of some of these hazards causing harm is much higher, e.g. slip, trip and fall hazards are far more likely to cause harm in the recording studio.
- Control measures will minimise the risk posed by the hazard.

Recording Process

Microphone Placement	Ensure microphones are placed to achieve the best result from each instrument.
Selecting Input Source	In the DAW settings, ensure that the correct input and output devices are selected. When adding an Audio Track, ensure that the correct Input Channel is selected. Alternatively, this can be changed in the channel strip.
Optimising Gain	The gain level should be set on each channel being recorded. This is done by adjusting the gain dial on the interface or mixing desk. If the desk or interface has a visual guide, the signal level should be in the green or just going orange at the loudest points. This will ensure that the signal is captured at a good level without it clipping (distorting).
Overdubbing	This is the process by which new parts are added to an existing recording.
Multiple Takes	It is standard practice to record more than one take for each performer. In the DAW, each take will be put into a take folder. These takes can then be compiled (or comped) into a complete take of the best parts of each performance.

Mic Placement

Electric Guitar & Bass

- Electric Guitars and Basses can either be recorded by Direct Input (DI) or by placing a dynamic microphone in front of the amplifier.
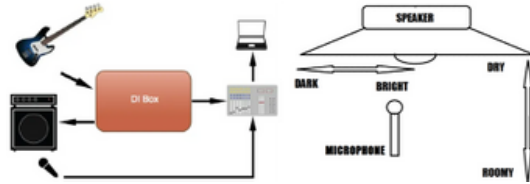
Acoustic Guitar

- Can be captured in mono or stereo depending on how much room sound is wanted.
- If recorded in stereo, XY and AB patterns can pick up different parts of the instrument's sound.
- Condenser microphones are best suited to capture the detail and the brightness of an acoustic guitar.

Vocals

- Captured in mono using a cardioid condenser microphone, although dynamic microphones can also be used.
- Microphones are usually placed just above the vocalist's mouth and angled slightly off the vertical to reduce the impact of plosives and sibilance.
- A pop shield is also recommended to reduce the impact of plosives.

Drum Mic Choices

Part	Mic Type	Polar Pattern
Kick	Dynamic	Cardioid
Snare Top	Dynamic	Cardioid
Snare Bottom	Dynamic	Cardioid
OH Left	Condenser	Cardioid
OH Right	Condernser	Cardioid
Room	Condenser or Dynamic	Omnidirectional

Drum Mic Placement

Kick Drum Placement:
- Using the port hole to get inside the drum
- Pointing at the beater
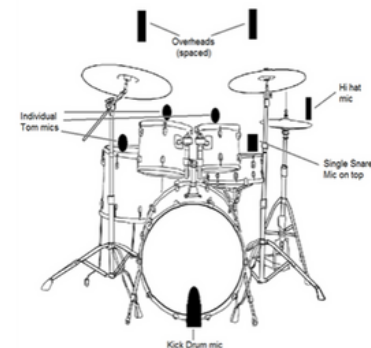- Roughly half way inside

Snare top placement:
- Above the edge of the drum
- Pointed towards the centre

Snare bottom placement:
- Below the snares
- Pointed towards the centre

Overhead microphones:
- Placed in either XY or AB positions

Editing & Routing

Corrective Editing

Removing unwanted audio	- This may be a gap before a performer starts playing or a gap in between sections where they play. - These issues is can be fixed by using audio editing to remove the sections of unwanted noise. Short fades can then be added to the audio to avoid any pops or clicks. - Noise gates could also be used to remove spill from a vocal or drum mic. - It is common practice to trim regions at the beginning and end so that there are no unnecessary empty or silent regions at the start and end of the recording. This is called topping and tailing.
Correction of Timing Errors	- Slip Editing can be used to correct timing errors. This involves cutting a recording into small sections and moving them in time with the metronome. - Crossfades are then applied to remove any unwanted clicks. This can also be done through functions such as Flextime editing in Logic. - Timing issues should be corrected before recording new parts, otherwise it will be harder for performers to play along to parts that are out of time
Correction of Pitch Errors	- For isolated instances, you can cut out the individual note and transpose that region to the correct pitch. Alternatively, it could be replaced by a copy of the same note from elsewhere in the recording. - For vocals, it might be more appropriate to use pitch correction to automatically correct the notes to the key and scale of the recording. This is more commonly

Polarity

- When recording anything with multiple microphones the polarity must be checked.
- E.g. a snare drum might be recorded with microphones that are 180° opposite from each other, so the waveforms go in the opposite direction.
- The snare top waveform is positive and the snare bottom waveform is negative.
- This means that frequencies will be cancelled out.
- The polarity or phase can be inverted using a plugin such as the gain plugin in Logic.

Routing - Buses and Sends

- When the output of a group of channels is changed to send them to a different destination than the stereo output.
- This allows the same processing to be applied to a group of tracks using only one plugin.
- It can also help with navigating around a project.
- Can be set up manually or by using track stacks.

Mixing: Balance & Plugins

Balance

- Balancing a mix involves adjusting volume levels of each part relative to the others.
- It also involves placing parts in the stereo field to achieve width in the mix but also so that one side is not stronger than the other.
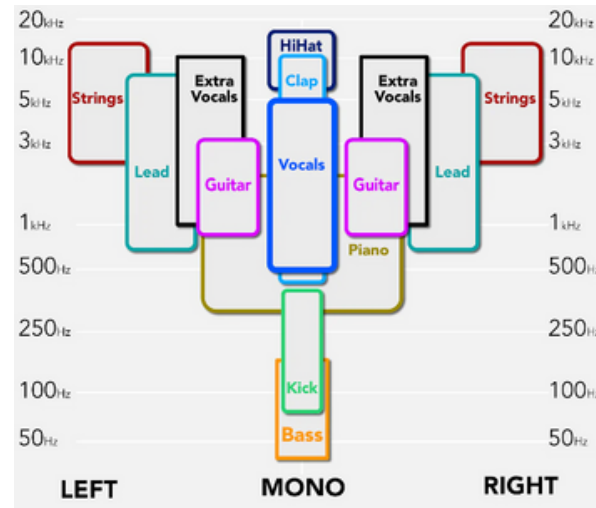
Automation

- Automation can be used to make changes to volume, panning and effects sends over time.
- This could be used to automate the volume of a guitar track so that it is louder in the chorus than in the verse.

EQ

- The process of turning up or down the level (dB) of specific frequencies (Hz).
- EQ can be used to improve the overall balance of a recording through filtering, cutting and boosting.
- The most common type of EQ in a DAW is a parametric EQ.
- HPF and LPF are used to remove frequencies that are not needed in recordings.
- Notch or band pass filters are used to remove frequencies that do not sound good on certain instruments and to provide space for other instruments.
- Notch filters and shelf filters can be used to boost desired frequencies.


Noise Gate

- Turns down a signal once it falls below a set volume/allows a signal to be heard once it goes above a set volume.
- It can be used to remove/turn down unwanted audio such as spill in a recording.
- The parameters - threshold & reduction - will need to be adjusted to suit the source and how much the spill needs to be reduced.

Compression

- Dynamic processing is essential for creating a balanced mix so that every part can be heard at an appropriate level.
- Compression is used to control the dynamic range of tracks, that is the difference between the loudest and quietest parts of the tracks.
- Parallel compression is when a compressed signal is mixed with the natural signal. This can either be done by using an auxiliary channel or by adjusting the mix balance in the compression plugin.
- Each sound source will need carefully adjusted parameters.

Uncompressed   Compressed
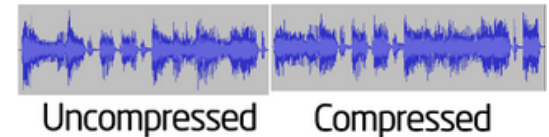
Limiter

- Limiters are used in modern music to increase the average volume of a mix.
- This makes the mix louder and ready to be played on the radio, in clubs and streamed online.
- Increased volume comes at the cost of reduced dynamic range, so limiting should be applied with caution as too much can ruin a great sounding mix.

Mixing: Effects & Sends

Amp Simulators

- Guitars and Basses recorded using DI will often need an amp simulator applied to make the recording sound as though a real amplifier has been used.
- This plugin simulates the sound of a real amplifier using software processing.
- This will allow a mix engineer to create a guitar sound that suits the rest of the recording.

Distortion

- Distortion plugins such as overdrive or other pedals can be used to add further individuality to guitar sounds.
- These can be automated to imitate a performer turning the effects on an off for different sections of the song.

Chorus

- Replicates the sound of multiple musicians playing the same thing together by modulating the frequency of the incoming signal.
- Creates changes in pitch and time giving the sense of multiple musicians playing.
- Chorus can be used to thicken certain parts.
- This is usually done via an effects send.
- It is most commonly used on vocals and guitar but can also be used on bass

Effects Sends

- When a copy of a signal or group of signals is sent to a bus.
- This allows for parallel processing to be applied to a channel whilst keeping the original signal.
- Most commonly used for adding delay and reverb effects.
- The amount of signal sent to the bus can also be controlled.
- It can be set up either as pre-fader or post-fader.
- Pre-fader will mean the level sent to the bus will not be affected by the level of the fader on the channel. If you still want to hear the effects when the dry signal is turned down choose this option.
- Post-fader will mean the level sent to the bus will be affected by the level of the fader on the channel. If you want the level of the effects to turn down relative to the dry signal choose this option.

Tremolo

- A tremolo plugin modulates the volume of the signal using an LFO.
- Most commonly used on guitars and can be found on some amps or as an effects pedal.

Flanger

- Flanging can be used to create a subtle double of an incoming signal similar to chorus.
- If used with extreme settings it creates a metallic swooshing effect.
- Commonly used on Guitars.

Phaser

- A phaser recreates the effect that the cancelation of frequencies has on an audio signal.
- It has a characteristic swooshing effect and is most commonly used on guitar and occasionally on drums.

Wah-wah

- Wah-wah is a filter effect.
- It uses a HPF or LPF to create a change in frequency content of a sound.
- It was originally a guitar pedal in the 1960s but is now emulated in a DAW.

Reverb & Delay

- Reverb and delay are used, sometimes in combination, to create a sense of space.
- Where parts are recorded in a dry studio, reverb and delay give the impression of a larger space that would be similar to a live performance venue.

Monitoring and Mixdown

Reference Material

- When mixing, it is useful to have what is known as a reference track.
- This can be any piece of recorded music that is similar to how you want your recording to sound like.
- During a mix, you can refer to these recordings to compare the quality of your mix.

Monitoring

- When mixing, it is good to use a range of different monitoring equipment to test how well the mix will translate to different environments.
- This can include Headphones, main and near-field Studio monitors (speakers).

Stereo Mixdown

- Projects can be mixed down either as a whole mix or as stems (stereo masters).
- Stems refers to exporting parts of the project as a mix e.g. a rhythm section (drums, bass, rhythm guitar) or a string section in a film score.
- Commonly projects are exported to uncompressed stereo files, either WAV or AIFF, with a sample rate of 44.1kHz and a bit depth of 16 bit, known as CD quality.
- Increasingly, projects are being exported at either 48kHZ or even 96kHZ with a bit depth of 32 or 64.
- However, if audio has only been recorded at 16/44.1 then there is no benefit to exporting at higher rates.
- Sometimes projects are exported to compressed stereo file types such as MP3.
- MP3s can be encoded at different bit rates.
- 192kbps (kilobytes per second) was the early standard but as devices increased their capacity, 320kbps has become more common.

Monitoring Equipment

	Advantages	Disadvantages
Headphones	- Isolated sound - Easier to hear detail in audio - Portability - Easier to hear the stereo field	- Uneven frequency response - Enhanced low end - Do not take in to account physical listening environment - Can have a more limited frequency response.
Main Studio Monitors	- Flat frequency response - Wider frequency response - More dynamic range	- Requires a well acoustically treated room - Listener must sit in an equilateral triangle to speakers - Not what most consumers will listen to music on
Near-field Monitors	- Closer to a consumer listening experience. - If a mix sounds good on these it will translate well.	- Limited frequency response - Less dynamic range - Listener must sit in an equilateral triangle to speakers