A Level Music Tech Dictionary

Table of Contents

General Music Tech Terms	
Amplify	Increases the amplitude or level of an electrical audio signal. An amplifier is a piece of hardware that boosts the signal
Amplitude	Another term for loudness or level, measured in decibels (dBs)
Analogue	An analogue signal resembles a sound by replicating the amplitude of the sound's wave. This can be recorded electrically or magnetically.
Attack	The time it takes for sound to reach its full velocity after being triggered.
Attenuate	Posh word for 'cut'.
Compressor	Reduces the dynamic range. (makes loud sounds quieter and quiet sounds louder)
DAW	Digital Audio Workstation
dB (decibels)	A unit of level or amplitude
Decay	The time it takes for sound to fall from its peak velocity to its sustain level.
Delay	It records the input signal and then plays the sound back at a later time (delay)
Digital	A process of representing a waveform in a binary code (0's & 1's).
Dynamic Processing	An effect that deals with level, amplitude or volume
Envelope (ADSR)	A way of automating the shape of a sound, most commonly through the amplitude and 'ADSR' controls,
Equalisation (EQ)	An insert effect that can alter the timbre (tone) of the input sound using filters.
Filter	An effect that boosts or cuts frequencies within a spectrum in order to shape the timbre (tone) of a sound.
Insert	A effect that is added in 'series' within a signal chain. i.e. one effect after another (NOT Parrallel!)
Frequency	Scientific measurement for different pitches. Measured in hertz. The limits of human hearing, known as the '*frequency spectrum*' is 20Hz to 20Khz.
Hardware	A piece of physical equipment such as a microphone or 'outboard' compressor
Interface	Converts analogue signals to digital code that can be recorded and stored on a computer.
MIDI	Music Instrument Digital Interface. The standard protocol that allow synths and drum machines to communicate with a sequencer (DAW)
Mixer	A device that is used to blend the levels of incoming sounds together and output them as a combined audio signal
Modulation	Change in a specific parameter over time. For example, pitch modulation is called vibrato
Monitor	A process of listening to a sound through speakers or headphones

Mono	A single audio tracks that has no left or right stereo information
Microphone	A device for recording sound that converts variations in air pressure level into an electrical signal.
Multi-track Recording	The ability to record more than one sound source at a time onto discrete tracks for mixing at a later date.
Noise Gate	A dynamic effect that will block out unwanted sounds that occur under a specified level (threshold). I.e. to get rid of guitar hum when not playing.
Production Technique	A technique that is used to improve a recording of performance.
Release	Time it takes for the sound to reach zero after the note is released.
Reverb	An effect that replicates that ambience of a given space, such as a church or bathroom.
Sample	An audio clip that can be manipulated in a similar way to a synth
Samplers	A type of digital recording device that allows audio to be controlled via MIDI, such as the Akai S1000
Software	A digital version of a piece of hardware (software compression)
Signal processing	What is done to audio in order to change its characteristics
Stereo	An audio track that has both left and right channels played at the same time.
Sustain	The volume/amplitude of the note when it is held.
Synthesis	A process of creating sounds using an combination of oscillators which create waveforms (such as sine & square) and tone shaping processes e.g. filter & ADSR envelope
Pan	Short for 'panorama'. Sending sound to left and right channels (speakers) independently.
Recording Studio	A place where you record sound, usually comprised of a live room (for your musicians) and control room (for your recording engineer and producers)
Quantise	A process that snaps MIDI note to the designated rhythmic subdivision. E.g. 1/8 Quantise will snap the notes to the nearest 8^{th} note or quaver.
Timbre/tone	The type or characteristic of a sound. For example the difference in character of a Saxophone and violin.

1.1 Recording and Production Techniques

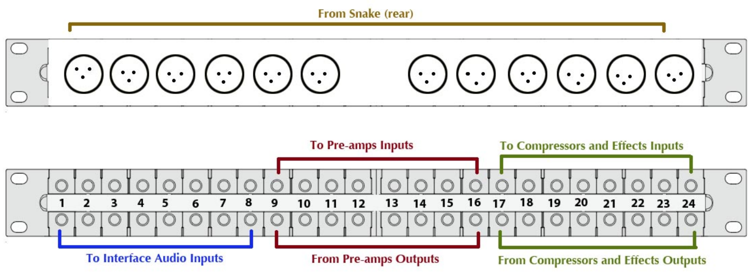
DAW functions

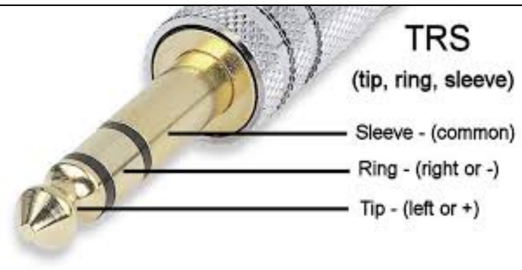
Audio Punch	When you set a DAW to record for a short window in order to correct mistakes recording audio.
Bounce	When multiple audio 'stems' are combined into one.
Overdub	Recording audio 'over the top' of pre-existing audio. Also known as multi-track recording.
Patch	A jargon-y word for a MIDI synth pre-set. Comes from when physical cables had to be 'patched'
Playhead	The line where the music is playing in Logic.
Region	The green boxes that contain the notes.
Transport Bar	Incorporates play, record and stop.
Velocity	How hard the notes are played. Affects timbres as well as volume.
Bars, beats, divisions and ticks.	Bar = 4 beats. Beat = 1/4 note. Division = 1/16th note. Tick = 1/240th of a division (1/3840th of a beat!!).

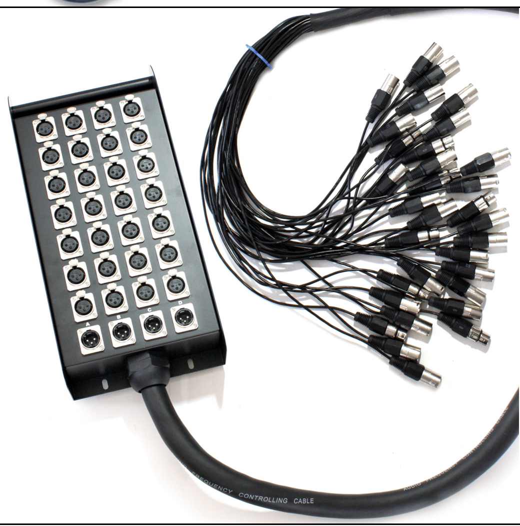
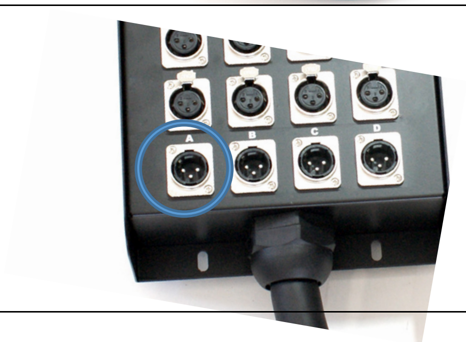
Automation	
Read/touch/ latch/write	**Read**: Logic plays back automation. **Latch**: Records values that you change using playback and leaves them at that value (e.g. Pan -63) **Touch**: Logic will record automation but as soon as you let go of the controls will jump back to the original position. **Write**: overrides **everything**. NEVER USE.
MIDI Automation	Records automation to MIDI file not just Logic. This will save the MIDI automation if the file is exported as .mid. These can then be opened in other DAWs

Studio Hardware		
Microphone		Turns variations in air pressure into an electrical wave form. A form of transducer. A transducer converts one form of energy into another form of energy.
audio interface		
microphone pre- amps		Gives an extra gain boost or reduction BEFORE the interface. Improves the signal to noise ratio and generally makes the recording sound great.
DI Box (active/Passive)		Direct injection – converts unbalanced jack signal to balanced XLR. Better **signal to noise ratio.**

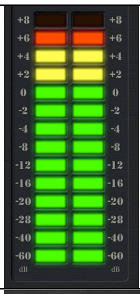
mixing desk		Turns multiple inputs into one stereo output. Enables you to add effects to separate channels such as EQ, pan, gain etc**.**
guitar pedals		Foot-controlled effects units for guitars/bass. Can be daisy-chained with mini jack leads. Connected in SERIES.
outboard effects (external effects unit)	Physical effects units. Signal affected BEFORE interface/DAW	
Digital Multi-tracker	Precurser to DAW – piece of hardware that looks like a mixer with mixing capabilities and a hard-disk recorder attached.	
Drum machine	A type of sampler/synth sequencer or hybrid which has presets specifically designed to sound like drums.	
Sampler	Stores/triggers/edits pre-recorded sounds	
Reverberation Unit	External reverb unit	
Compressor	External compressor	
EQ Unit	Boost/cut frequencies.	

Other things found in the Studio		
Monitors 'Monitoring'		
Patch bay		A panel in the studio used for connecting different effects units/instruments. Stops you having to plug things in and out from the back.
Control room		Where the sound engineer can monitor recording from and mix.

Acoustic shields (or screen)/ baffles		Absorb sounds so there is less unwanted room noise/reverb.
Sound/Isolation Booth		Soundproof and very 'dry' room for recording soloists. Acoustically treated and insulated. 'Room within a room' which removes spillage. Most useful for drums (which spill a lot) and vocals (which need to be isolated)
TRS Jack See 2.3 leads and signals for more info		
XLR (cannon)		Balanced 3 pin lead. Used for mics and reduces hum. See 2.3.
Mic Stand		A stand. For mics.

Clip		The important bit everyone forgets about.
Pop Shield		Filters out plosives and sibilance.
Jack Lead		Used to connect guitars to amps. It is unbalanced.
Looms/stage box or 'multi-core'		Found in the live room. Extension lead for audio. Enables you to plug mics into the interface from a distance.
'Return'		Enables the performer to hear guide tracks and sounds.

‘Programming environments’	
MIDI	Musical Instrument Digital Interface. A protocol (computer language) which expresses musical performance in numerical form (binary). Allows all DAWs and synths/samplers made by different manufacturers to talk to communicate other through a globally understood language.
Open Sound Control (OSC)	A **new** protocol invented for synths and computers to communicate. Messages are transported across the internet using UDP (User Datagram Protocol) instead of MIDI allowing for more accuracy and flexibility.
New and emerging software on music production	
Ableton live	
MAWs	Mobile audio workstation. E.g. Garage band on your phone.
FL Studio	
Steinberg Cubase	
Studio One	

Types of Mixer	
Live sound mixer (analogue)	Hardware that is used to mix audio signals in a live environment
Digital/software mixer	A mixer that exists solely on a computer
Recording Mixer	A mixer that has many inputs for recording
Mixer Controls	
Pot	
Bus	Sends the output signal of a channel to an auxiliary (or 'aux' channel before the output.
Meter (LED)	
VU (Volume Unit) Meter (see 2.4 levels for more detail)	
Channel Fader	
Mute (Cut)	
Phase Reverse Ø	Switches the polarity of the signal to avoid cancellation of frequencies
Solo/PFL	**Solo** = only the track(s) soloed are outputed (post fader). Found in

	DAWs. **PFL** = 'Pre-fade listen'. Found on analogue mixers. Solos the track, *but* **bypasses the fader** and *only* solos the track for the **monitoring/headphones**, *not* the master output. Useful for line checking the input gain as it ignores the fader.
Group Faders	
'Line'/'Mic/'instrument input'	(see 2.3 Leads and signals (impedance) for more detail)
Insert	
Tape	
Trim/gain	
Hi mid/lo mid etc	 
Q	
Aux Sends	
Aux Masters	
Unity Gain	

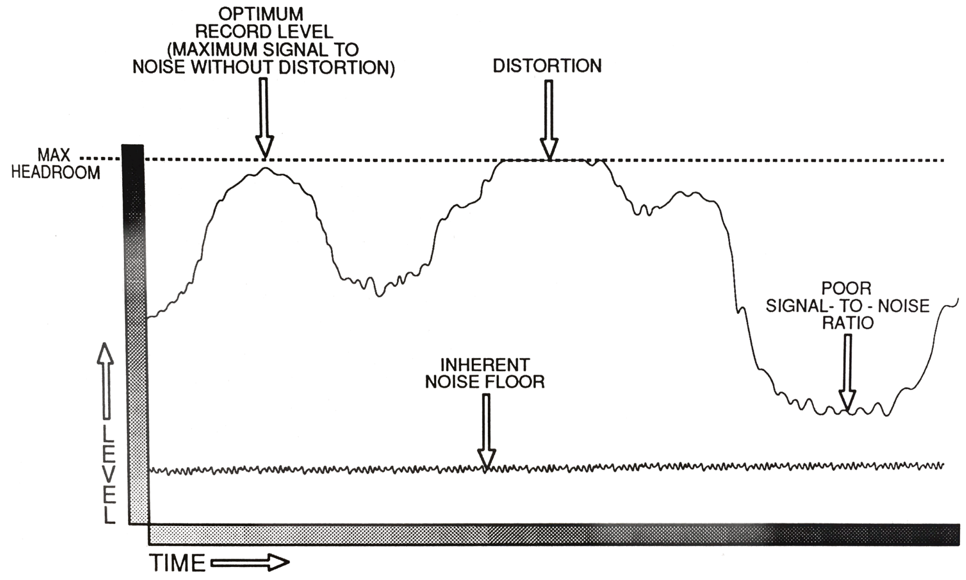
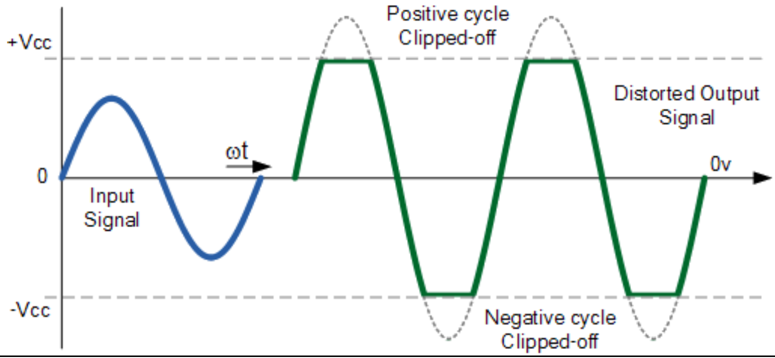
Pan	
Monitor section	
Flying faders	
Master out	
Tape out	
Channel Strip	
Talk Back Mic	
Controls of an interface	
Gain/Trim	
Headroom	
Clock sync	

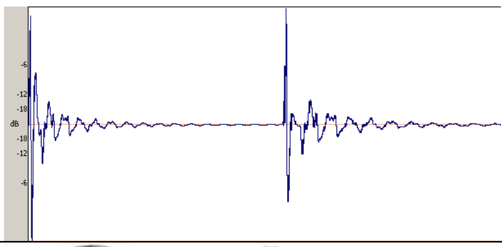
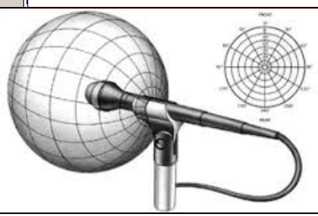
<table>
<tr><td></td><td></td></tr>
<tr><td>Kettle Socket</td><td></td></tr>
<tr><td>ADAT
(optical in)</td><td></td></tr>
<tr><td>FIREWIRE</td><td></td></tr>
<tr><td>Main Out</td><td></td></tr>
<tr><td>XLR input</td><td>
</td></tr>
<tr><td>Combi- input</td><td>

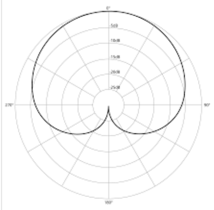
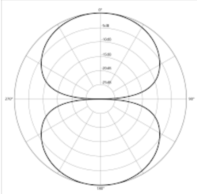
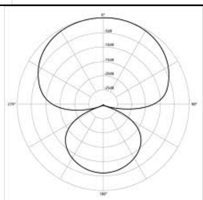
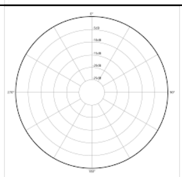
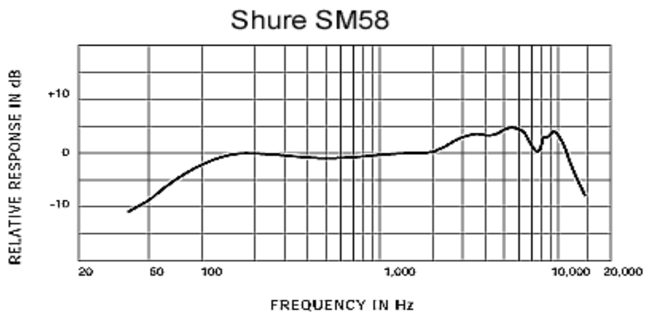
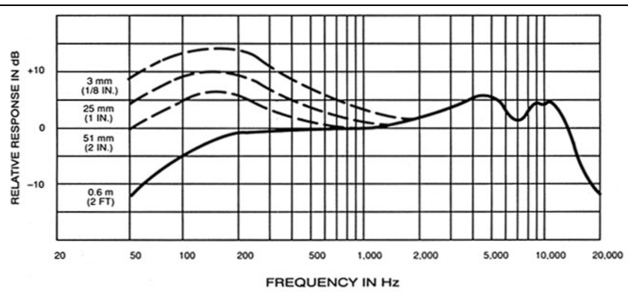
don’t forget about the locking tabs….</td></tr>
</table>

1.2 Microphones

Gain

Signal – to – noise ratio	
clipping	
interference and hiss	
Pre-amp controls:	
Phantom Power	

pad	
Mics and their characteristics	
Capacitance	
Condenser	
Dynamic Mic	
Cradle	
Rumble (Low pass) Filter	
Transient sounds	
Polar Pick-up pattern	

Cardioid	
Figure of 8 pattern	
Hyper-cardiod	
Omni-directional	
Frequency Response	
Proximity Effect	
Micing techniques	
Background noise	
Spill	**When other ounds are picked up beyond the sound source**
Plosives/ sibilance	
Ambient Recording	
Close Micing	

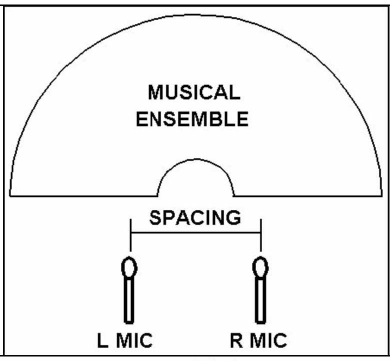
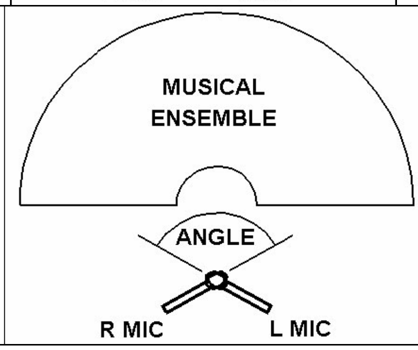
On/Off Axis	
Stereo Pair	

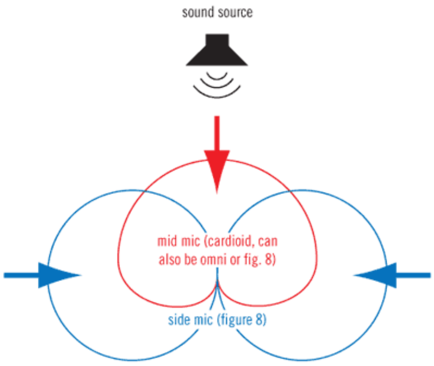
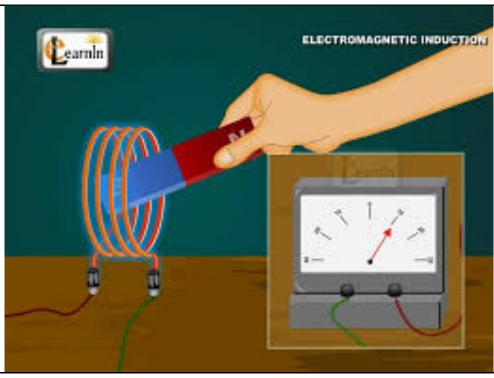
How to mic up instruments

Instrument	Mic + Polar Pattern + Placement (distance)	Technique/notes
Electric Guitar		
Bass Guitar		
Acoustic Guitar		
Drum overheads		
Snare		
Toms		
Kick		
Cello		
Violin		
Double Bass		

Brass (Trumpet/trombone)		
Flute/Clarinet		
Saxophone		
Vocals		
BVs		
Congas		
Xylophones		

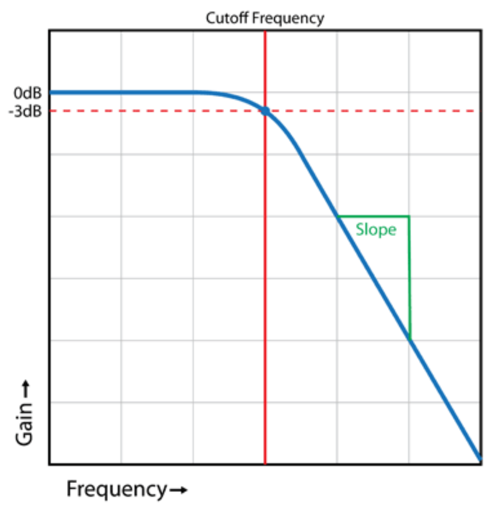
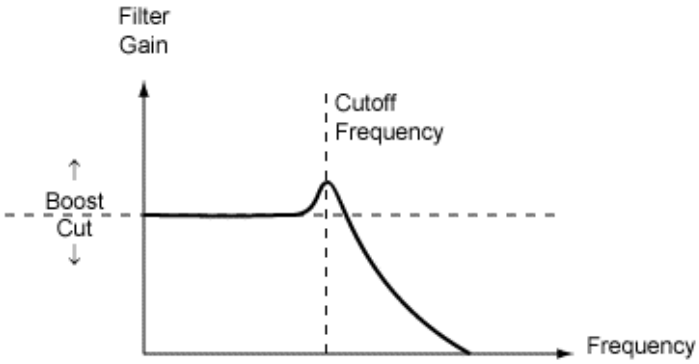
Advanced Micing Techniques

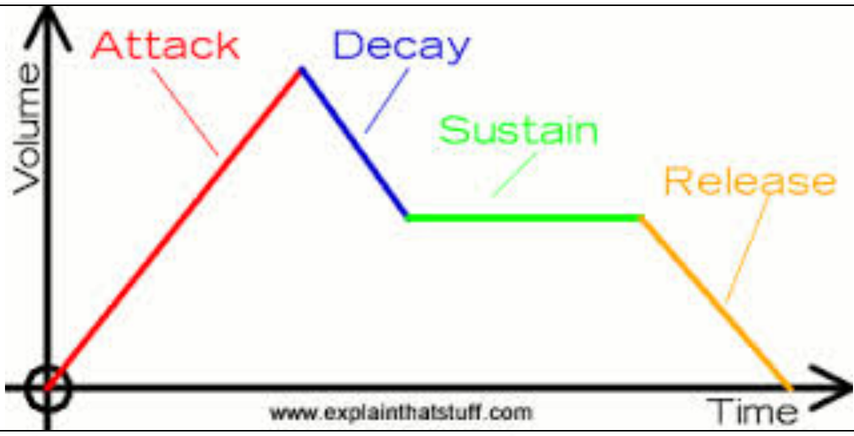
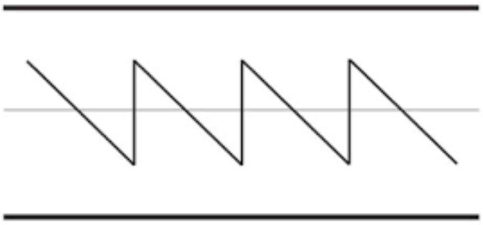
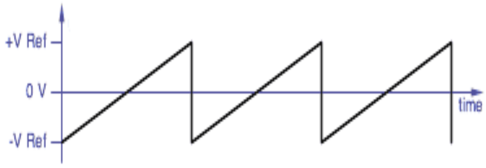
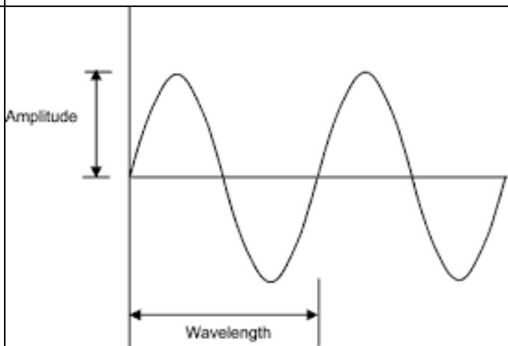
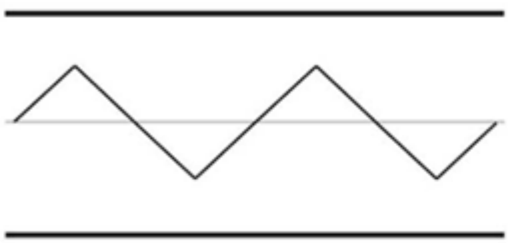
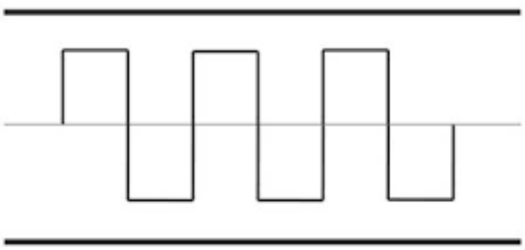
Spaced/AB Pair	
X/Y or coincident pair	

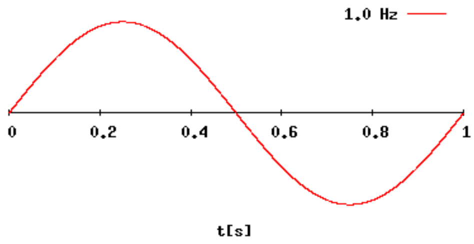
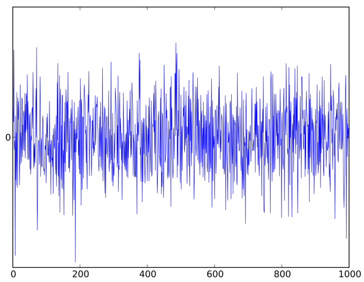
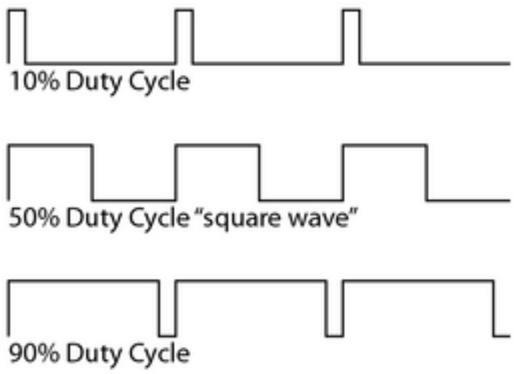
M/S (mid side)	
How mics actually work	
Electro-magnetic induction	
Front/ back plate	
Transducer	
Diaphragm/ membrane	

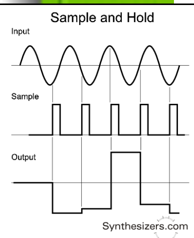
1.3 Synthesis

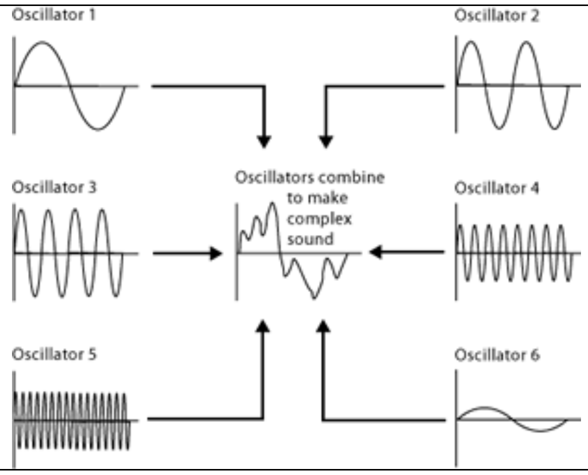
Parts of a Synthesiser

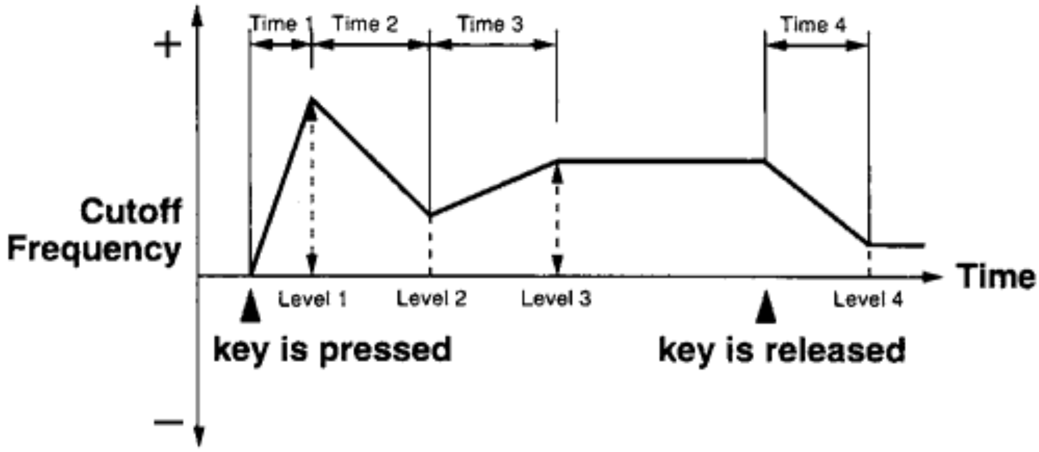
Oscillator	 Generates an electrical wave form Used for a synth. Several oscillators can be blended/ pitched to create different timbres.
Cut off/cut/frequency/ Centre Frequency/LPF/ HPF/BPF	 The point (in hertz) at which the filter starts working. (E.g. 1KHz)
Resonance	 A boost at the cut off frequency of a filter. Creates a whistling sound.

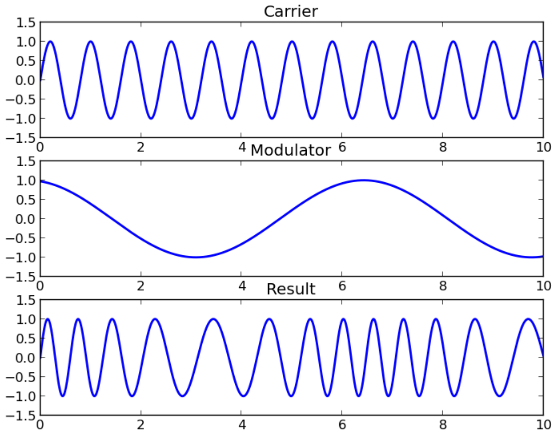
Envelope (Amplitude)	
Sawtooth Wave	
Ramp Wave	
Sine Wave	
Triangle wave	
Square Wave	

LFO	 Low frequency oscillator (between 0-25Hz). So low you can't hear it. It can then be assigned to different parametres (e.g. pitch/volume). The oscillator/synth the LFO is assigned to will be modulated according to the wave shape, rate, depth(intensity) and parameter of the LFO. e.g. if LFO is at 7Hz and is a sine wave, and assigned to pitch, it will produce a vibrato effect.
White noise	 Hissing sound that contains all the frequencies
Pink Noise	Rumbling sound with all the frequencies with emphasis on lower frequencies.
PWM (Pulse width modulation	 A square wave with an adjustable amount of time in between each cycle before the the voltage drops from maximum to minimum. The percentage of time that the signal is high is known as a **duty cycle**. The duty cycle 20% can then be modulated using an LFO. It's particularly good for creating synth string sounds.
Pitch Bend	A wheel on a MIDI controller which varies the pitch of a sound.
Modulation Wheel	Wheel on the side of a MIDI controller that can be routed or assigned to any parameter in a DAW for real-time automation.
Assign/Route	When you connect different parts of a synthesiser or effects unit to each other. Modular synths use these to connect together different parts.
Arpeggiator	A piece of software that turns a chord into a broken chord. You can change things like octave and direction.
Amplifier	The part of the synthesiser which amplifies the signal. The last stage in the synthesis process.

Other parameters of a synthesizer	
Voices	The number of voices that can be played simultaneously. Monophonic means only one voice at once.
Glide (secs)	The time it takes in seconds to slide from one note to another note. Also known as **portamento.**
Drive	An overdrive on a filter which boosts the signal.
Legato	Setting which means that the attack and delay envelope(s) are not retriggered after the first not is played. It mimics the sound of slurring notes to create a smooth sound. Particularly noticeable with filter envelopes.
Blend	The amount of signal coming from each oscillator (e.g. square and sine)
series/parallel	A routing option for filters. Series = one after the other. Parallel = both at the same time.
Detuning	Makes all the Oscillators a bit out of tune.
Coarse/fine tuning	Coarse = tuning by semitones Fine = tuning by cents (see below)
Cents	The increments between semitones. 1/100th of a semitone. 50 cents = quartertone.
2'/4'/8'/16'/32' (feet)	 8ves on a synth. Measured in 'feet' (the length), as originally organ pipes were measured by their length in feet.
S/H	 Samples a wave (e.g. sine) and holds a signal at the sampled value until the next sample is taken.
Polyphonic	Synthesisers that can play more than one note at once.
Bender Range	Measured in semitones. How many notes the pitch bend will bend by. See Sequencing (1.5) for info about how MSB and LSB are used to make pitch bend smooth.
VCO/VCF	Voltage controlled oscillator/filter. The name for filters on analogue synths.
DCO/DCF	digitally controlled oscillator/filter. The name for filters on digital synths.
Subtractive synthesis	The use of oscillators that produce harmonically complex waveforms (e.g. saw wave) then putting them through filters to shape the sound further.

<table>
<tr><td>Additive Synthesis</td><td>

Creating complex timbres by using multiple oscillators that generate simple wave forms (i.e. sine waves)</td></tr>
<tr><td>Filter Envelope</td><td>

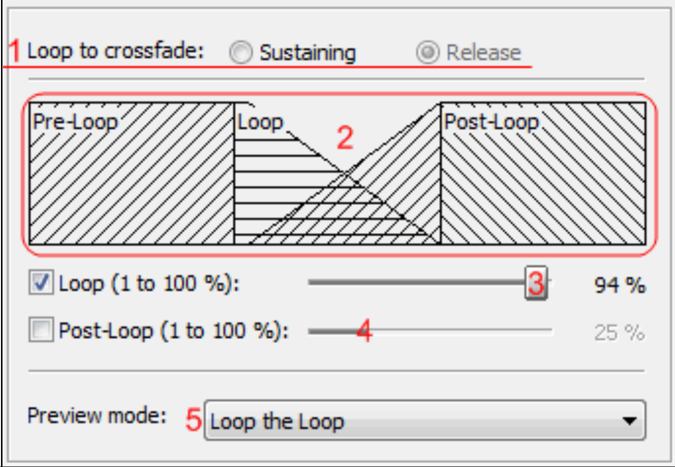
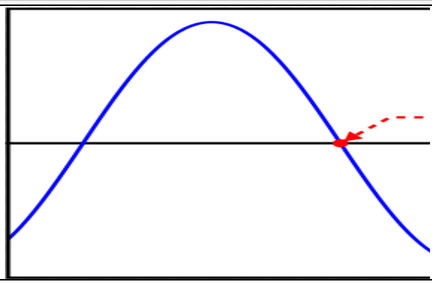
When the an envelope is assigned to filter cut off.</td></tr>
<tr><td>FM synthesis</td><td>

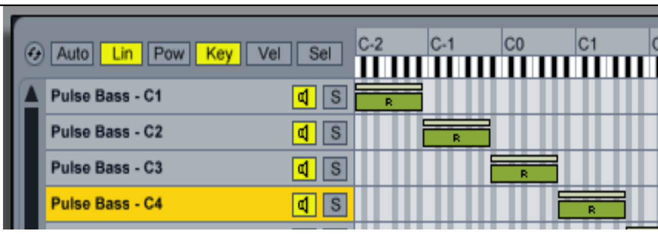
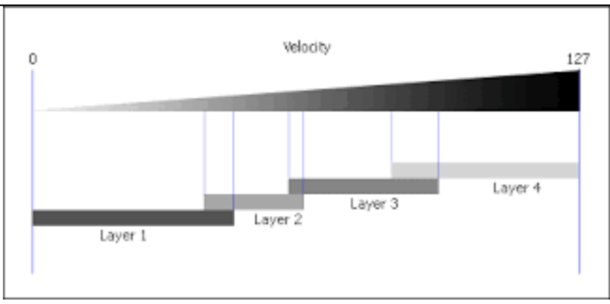
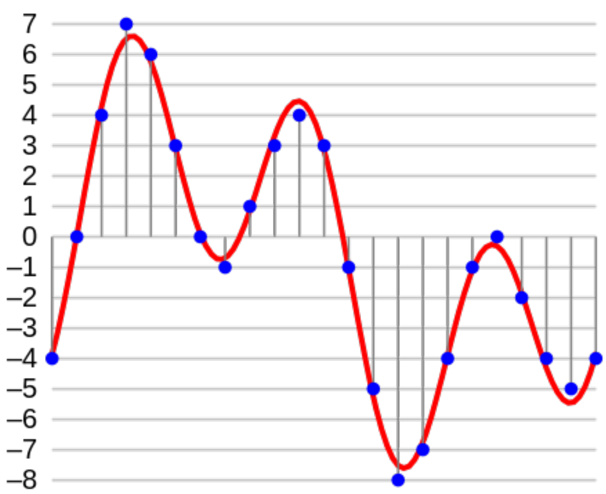
Control the output frequency of one oscillator (the carrier) with another oscillator (the modulator). This can produce lots of different results.
More excitingly, it can have several modulators at audible frequencies producing elaborate timbral changes and extra harmonics.</td></tr>
</table>

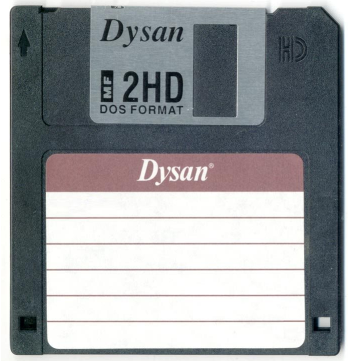
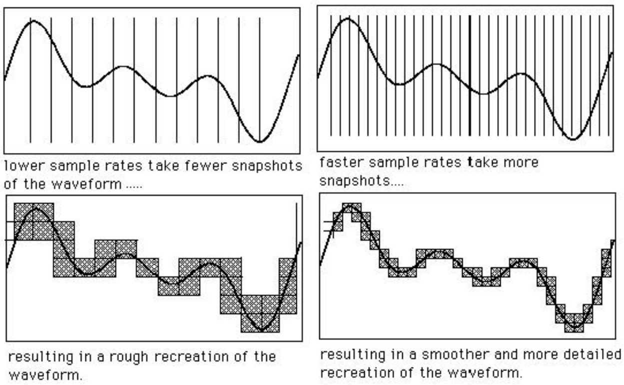
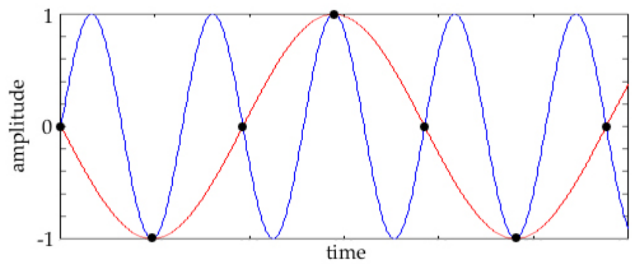
1.4 Sampling

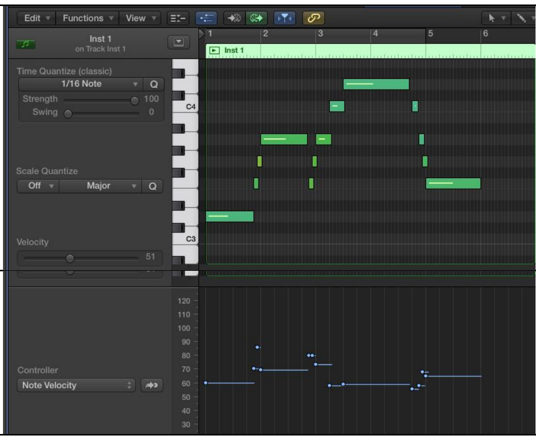
Editing Samples

Cutting/ trimming/ truncating	Editing the length of a sample.
Tuning samples	Pitch is affected by speeding up or slowing down samples.

Looping	 Repeating to create an endless sound without a click.
Cross/x-fade	 Fade out the end of a loop as you fade in a the beginning of a loop so there is no click.
Zero-crossing point	 Slicing the sample when the amplitude is 0, where the wave meets the X-axis, to avoid a click.
Pitch Mapping/ Transposing	Using the pitch shift insert to change the pitch without speeding up or slowing down.

More sampling Parameters	
Reversing	Playing the sample backwards.
Stuttering	Re-triggering the beginning of the sample repeated.
ADSR!	You can assign these synth parametres to audio samples.
Pitch key zones	 multi-sampling where one sample is stretched over as many notes as possible without affecting the timbres.
Velocity Layering	 Different samples are assigned to different velocity ranges BECAUSE there is a difference in timbre when velocity changes. X-fade between each sample to smooth transition.
Time Stretching	**Change the length of the sample** but the pitch is the same. Measured in 100%.
Multi-sampling	Sample only covers a limited range of notes. See pitch key zones.
Bit Depth /resolution	 How much information per sample. Howe accurately the amplitude of the sample is recorded. E.g. on a scale fro m -8 to 7.
Nyquist Theory	The sample rate should be double the desired frequency response.

Floppy disc	 An old storage system.
Sample Rate.	 The number of samples per second.
Pitch Shift	Speeding up or slowing down the sample the change the pitch.
Binary	0s and 1s.
Aliasing	 The sample rate is too low so frequencies (within the audible range) are incorrectly recreated. The result in unwanted **artefacts**.
Dithering	Introduction of small amounts of unobtrusive randomly generated noise in the the conversion process. Why? It randomises the effect of quantiseation error which occurs when analogue is converted to digital.

1.5 Sequencing	
Real-time input	
Step-time input	
Step Grid	
Pencil Tool	
Quantise	
¼,1/8, 1/16	
1/6, 1/12	
% quantise	
Groove templates	
humanise	
Note length quantise	
Editing skills	
Velocity	
List editor	
Piano Roll	
Velocity Editing	
Note length edit	
Global Editor	

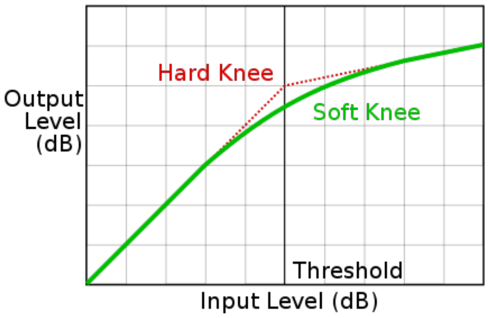
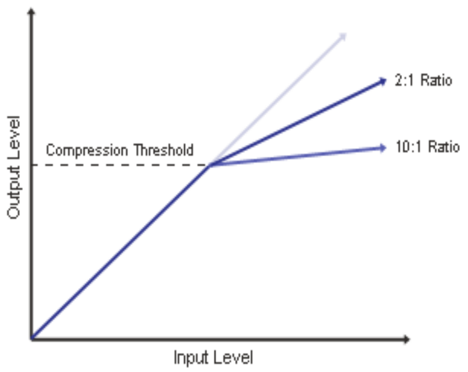
How MIDI works	
MIDI controller	
Sound Module	
Patch	
Protocol	
8-Bit	
Channel/Event message	
Event message (and event editor)	
Controller Change (CC)	
Continuous controller	
Switched controller	
System Messages	
General MIDI	
GM2	
Inspector Window	
Matrix Editor	
Hyper Draw	

Hyper Editor	
MIDI IN/ OUT/ THRU	In: Out: Thru:
Jitter	
Latency	
Format 0, 1 & 2	
Soundboard	
Mbps	
Audio Buffer	
Wurlitzer Sideman	
Moog 960 Step Sequencer	
Status byte	
Databyte 1	
Databyte 2	
MSB/LSB (in pitch bend)	
ATARI ST	
Cubase	

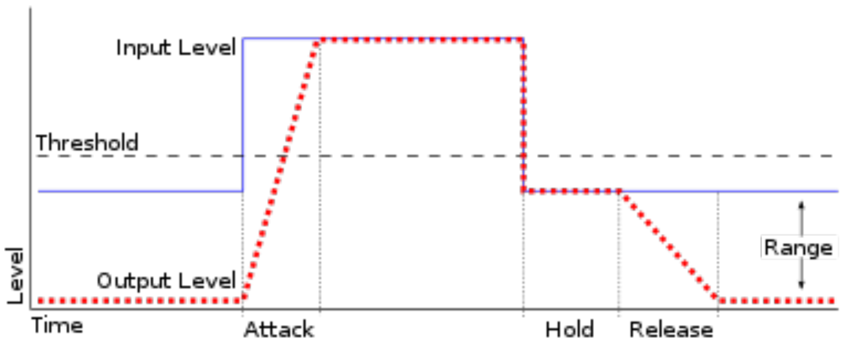
Logic	
Pro Tools	

1.6 Audio Editing	
Truncating	
Scissor Tool/split/splicing	
Overlap/no overlap	
Removing clicks and noise	

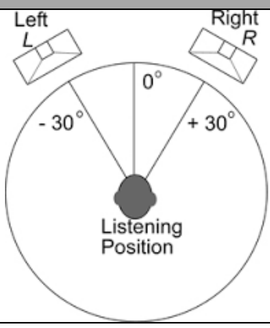
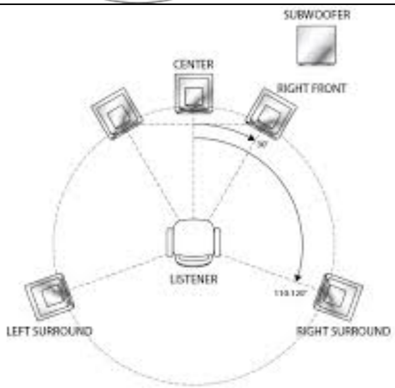
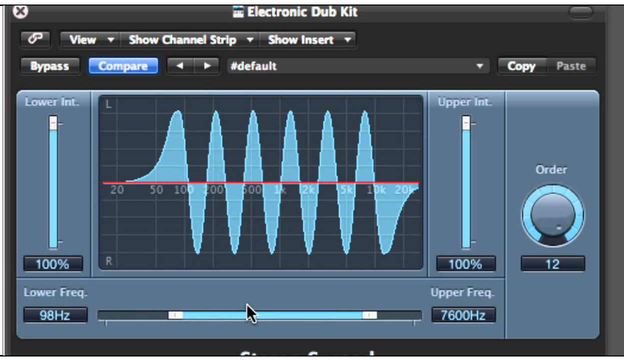
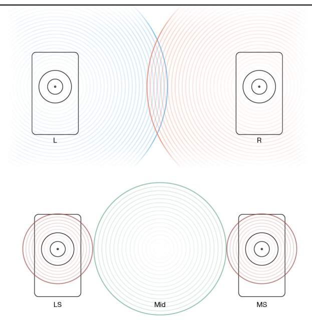
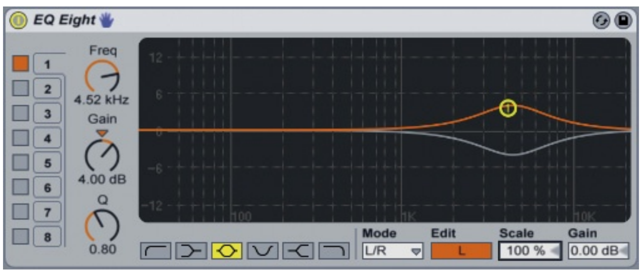
1.7 Pitch and rhythm correction	
Pitch correction	
Response time	
Mix	
Formant shift	
Pitch shift	
How to correct inaccuracies in rhythm	
Audio Quantise	Quantises transient peaks in audio so they are in time with a click.
Flexi-time or elastic audio	

1.9 Dynamic processing	
Advanced Dynamic processers	
Limiter	
Expander	
De-esser	
Pumping	
Compressor Parametres	
Knee (hard/soft)	
Threshold	 The level at which the compressor starts working
Attack (ms)	
Release (ms)	
Circuit Type	
Stereo Link	
Multiband compression	
Side-chaining	

Noise Gate Parametres	
Threshold	
Range/ Reduction	
Attack (ms)	
Hold (ms)	
Release (ms)	
Monitor	
Side-chain	

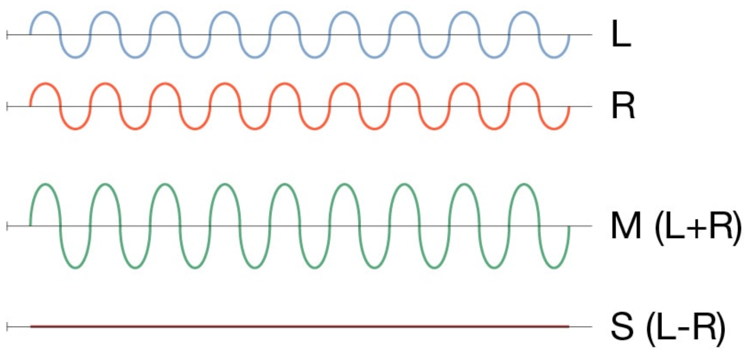


ratio	The amount of compression

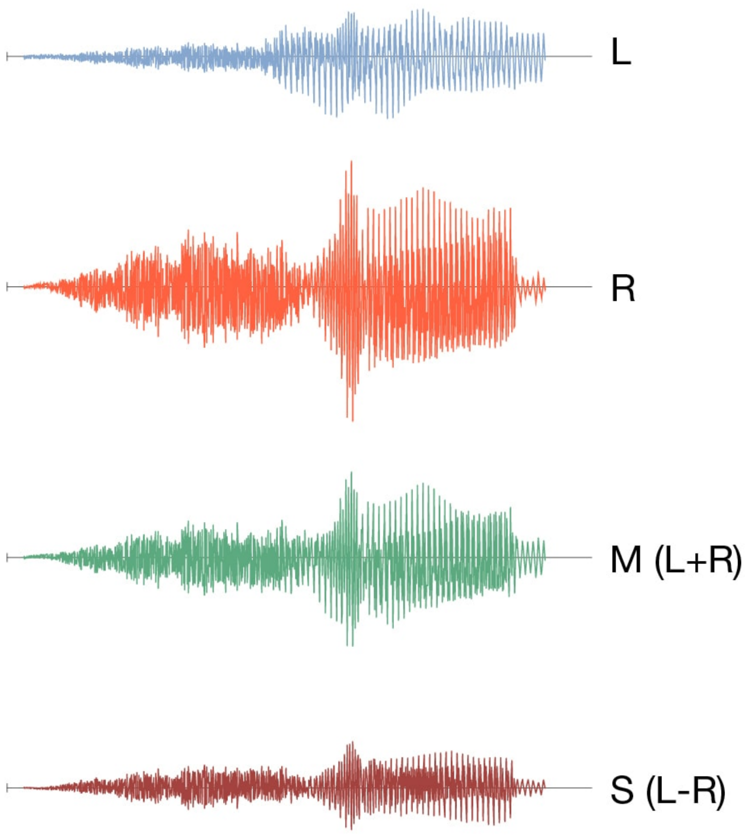
1.10 Stereo	
Stereo Set up	
Surround Set Up	
Pan	
Stereo Widening (Stereo spreader)	
Mono-compatibility	
Panning law	
mono-summing	
mid-side (M/S) processing (EQ)	 

Based on the micing technique.

The mid signal contains all the signal components that are distributed equally left and right (sum). These create a mono signal. Conversely, the side signal is composed of everything that differs between left and right (difference). Together the mid and the side signal combine to create to a complete stereo signal.



simplified waveform

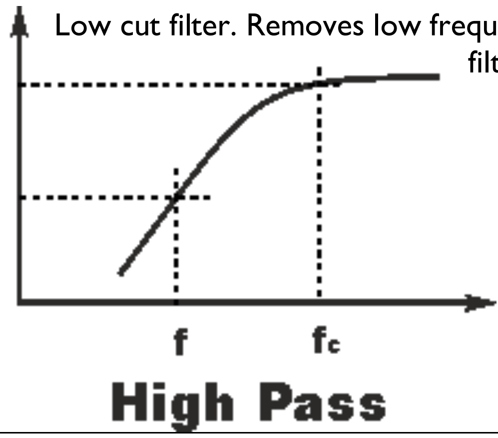
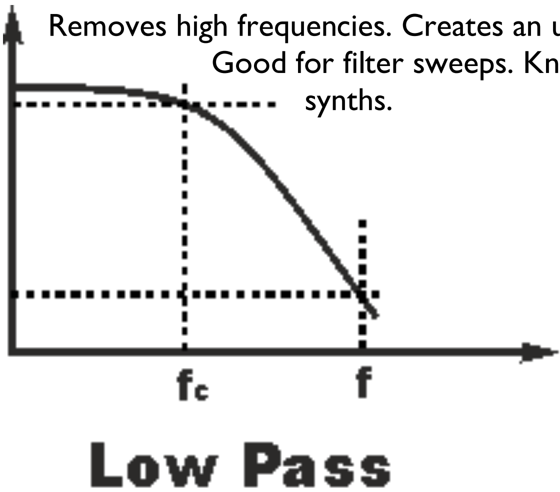
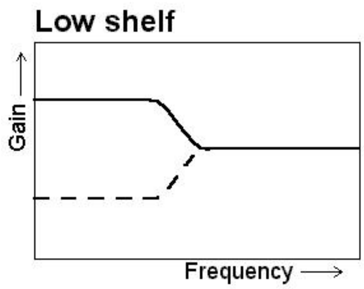


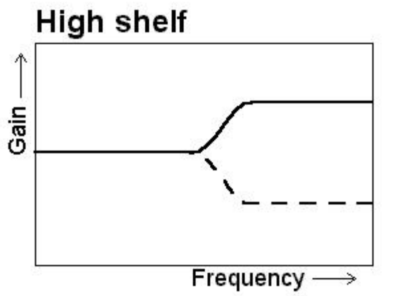
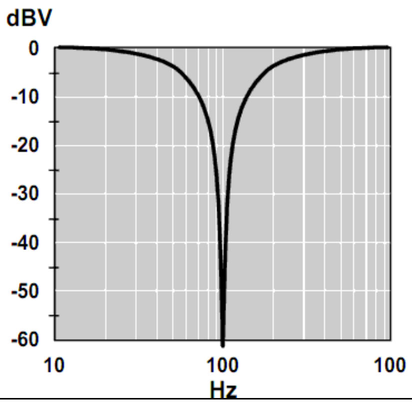
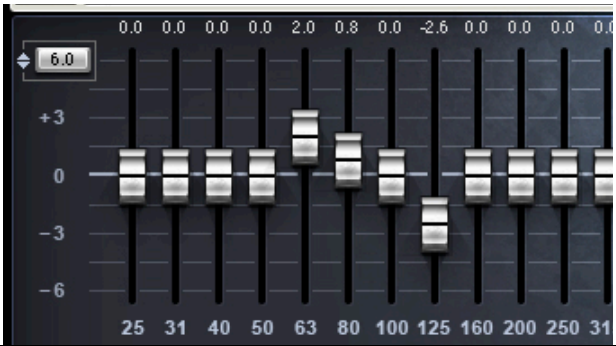
actual waveform

sonible.com

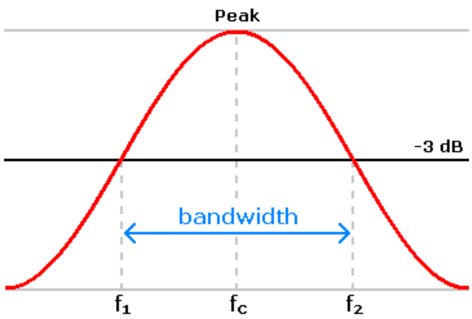
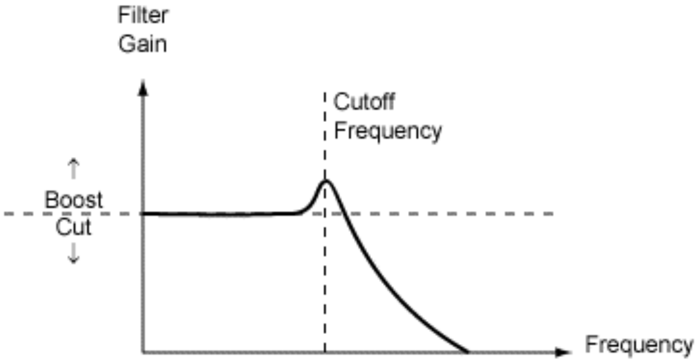
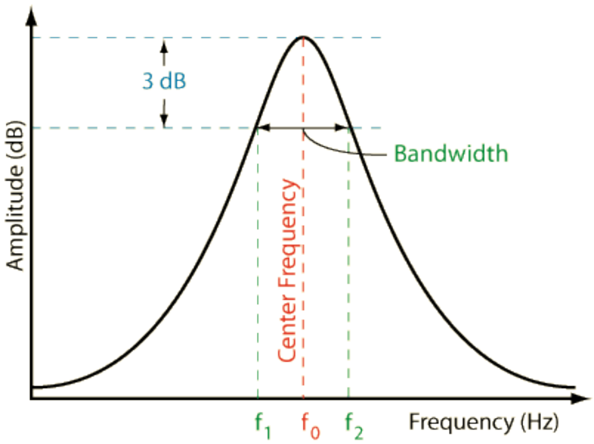
1.11 EQ

Different types

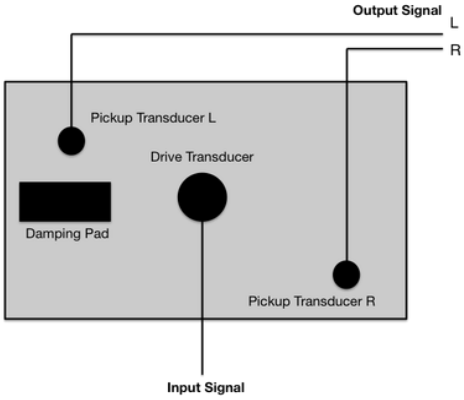
Frequency Spectrum	20 – 20K hertz. The limit of human hearing.
dB	Measurement of loudness of a logarithmic scale.
kHz/Hz	**Hertz (measurement of frequency).** Cycles per second (cycle = 1 wavelength)
HPF	Low cut filter. Removes low frequenceis/rumble. Around 20-120hz. For filter sweep 
LPF	Removes high frequencies. Creates an underwater effect. Reduces hiss/spill. Good for filter sweeps. Knowns as the cut off filter in simple synths. 
Low shelf	

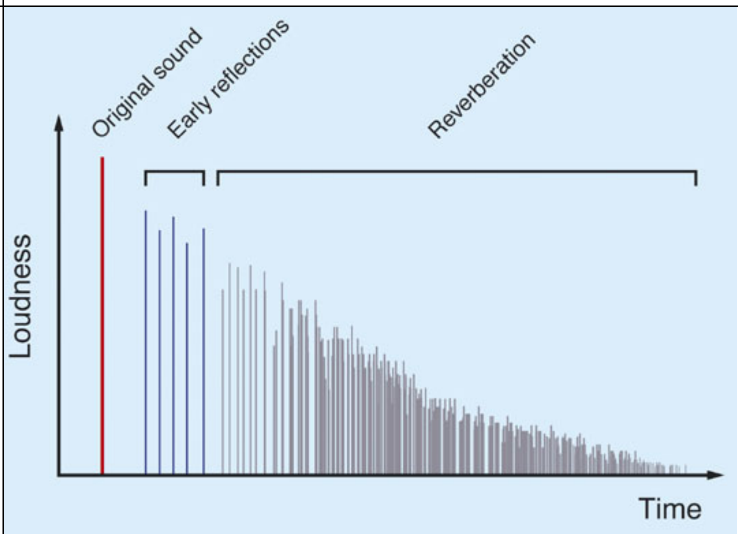
High Shelf	
Band EQ (notch filter)	
Filter Sweep	
Graphic EQ	 **Fixed band EQ. Normally 25-31**
Parametric EQ	
Bypass	
Frequency ‘Bands’	
Rumble Filter	A high pass filter designed for cutting low bass frequencies during recording.

Analyser	

Advanced EQ parameters	
Bandwidth/Q	
Resonance	
Centre Frequency	
F stop/ F turnover (slope)	
Match EQ	
'Ringing out'	

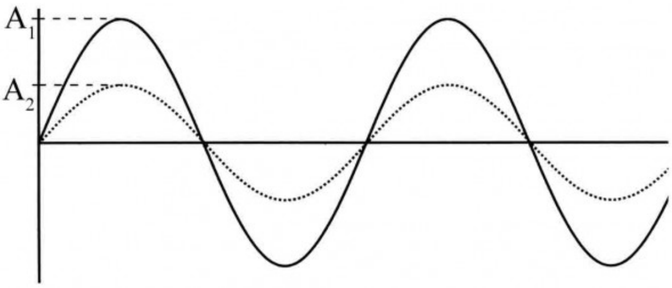
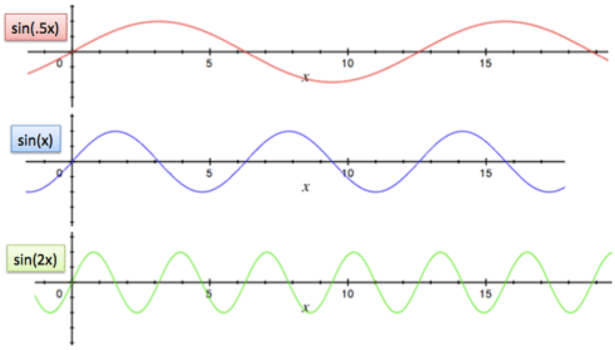
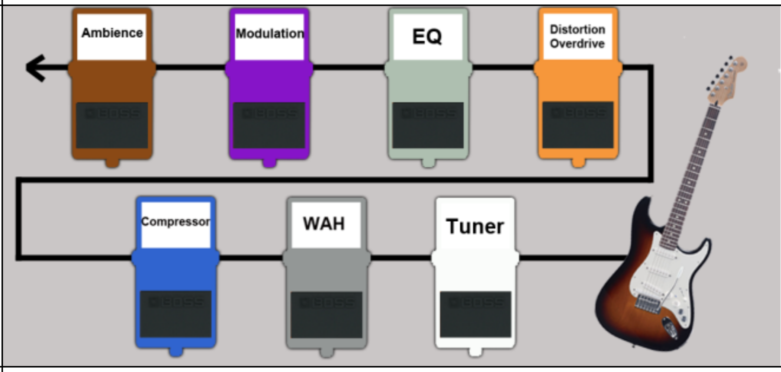
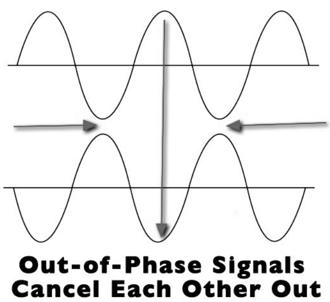
1.12 Effects	
Core Parametres	
Wet/Dry	
Mix	
Bypass	
‘Additive’ effects	
‘In the mix’ effects	

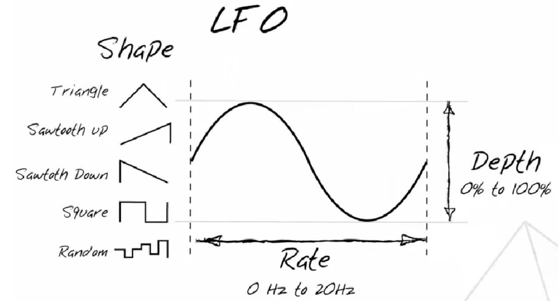
<table>
<tr><th colspan="2">Reverb</th></tr>
<tr><td>Reverb</td><td>After a sound has been made, it dies away or dissapates and reflects of differetn serfaces.</td></tr>
<tr><td>Spring Reverb</td><td>
</td></tr>
<tr><td>Plate reverb</td><td>
</td></tr>
<tr><td>Echo Chamber</td><td></td></tr>
<tr><td>Digital reverb</td><td></td></tr>
<tr><td>Room Size</td><td></td></tr>
<tr><td>Reverb time (seconds sec)</td><td></td></tr>
</table>

Reverb advanced parameters	
Density	
High/low frequency control	
Convolution Reverb	
Impulse Response	
Excitation signal	
Natural Reverb	
Dampening Pad	
Reverb time graph	
RT60	
Pre-delay	
Early reflections	

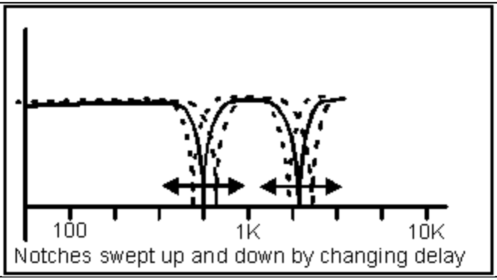
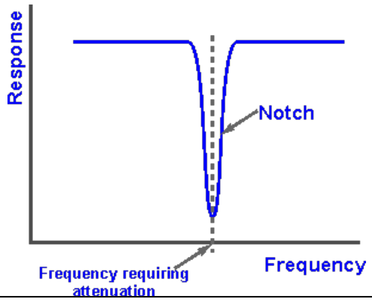
<table>
<tr><th colspan="2">Delay</th></tr>
<tr><td>Delay Time (repeat rate)</td><td>Time between delays</td></tr>
<tr><td>Feedback (in %)</td><td></td></tr>
<tr><td>Tempo Delay/timed delay/sync</td><td></td></tr>
<tr><td>Low cut/high cut</td><td></td></tr>
<tr><td>Bass/Treble</td><td></td></tr>
<tr><td>Mono Delay</td><td></td></tr>
<tr><td>Stereo Delay</td><td>Left and right delay can be treated independently</td></tr>
<tr><td>Pingpong Delay</td><td></td></tr>
<tr><td>Multitap delay</td><td></td></tr>
<tr><td>ADT</td><td></td></tr>
<tr><td>Slapback echo</td><td></td></tr>
<tr><th colspan="2">Delay advanced parameters and Analogue tape delay</th></tr>
<tr><td>Copycat</td><td>
</td></tr>
<tr><td>Delay Designer</td><td></td></tr>
</table>

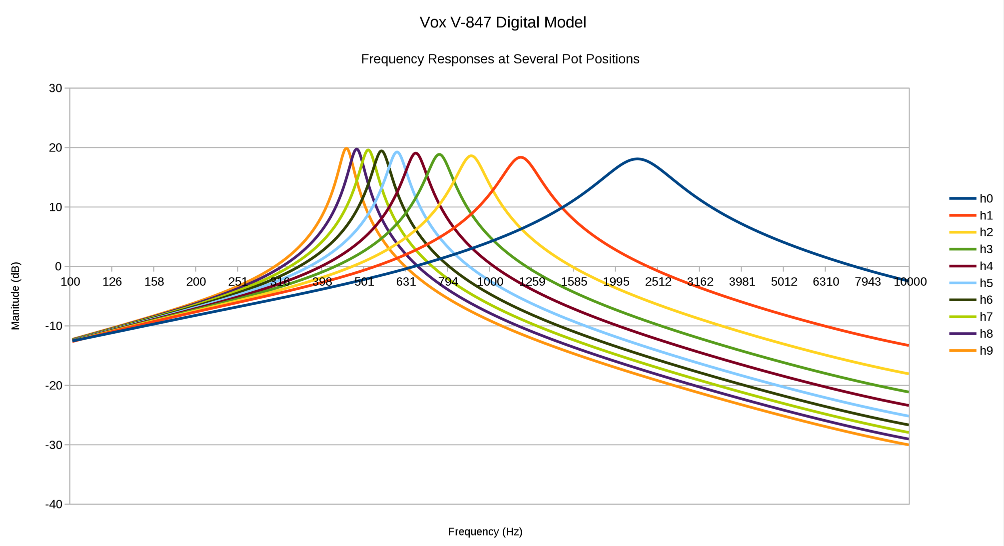
(delay pan and EQ)	
Tape Loop	
LFO (in delay)	
Freeze	

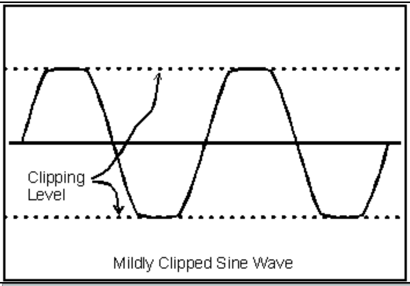
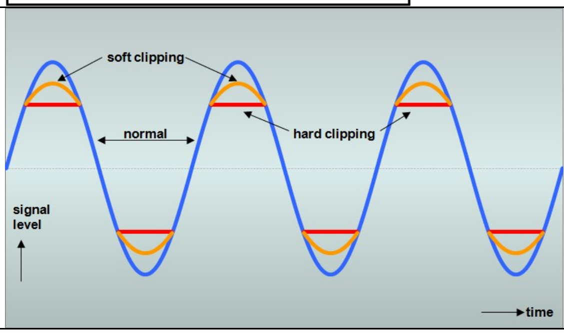
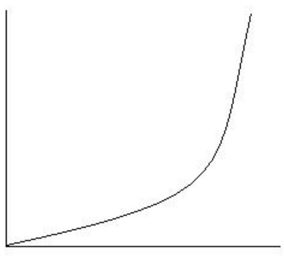
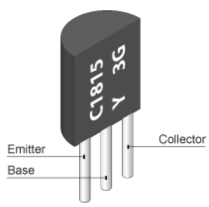
Basics of modulated delay		
Depth (Or intensity)		A = amplitude
Rate (Hz)		
¼ inch jack		
9V battery		
Daisy Chaining		
Phase Cancellation		

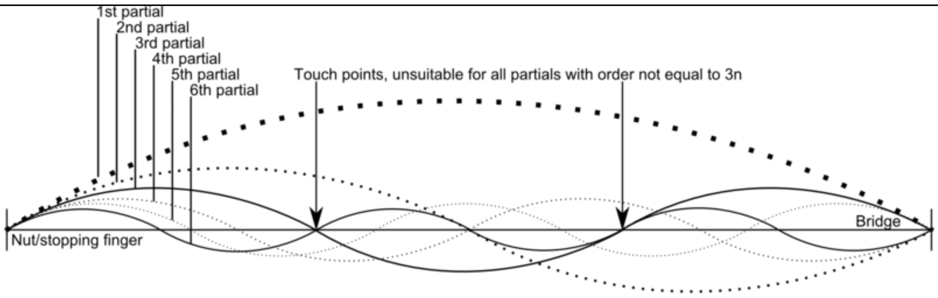
LFO Wave	
Flange	
Flanging	
Rate (Hz)	
Depth (Or intensity) Also ‘Width’	
Feedback or Resonance (Res)	
Manual	
Comb filter	

Chorus	
Depth (Or intensity)	
Rate (Hz)	
Rate (Hz)	
Mix (E. Level)	
EQ/Tone	
Sync	
LFO	

Phaser	
Phasing	
Rate (Hz)	
Depth (Or intensity) Also 'Width' Or Floor/ceiling	
Feedback or Resonance (Res)	
8/10/12 Or 90/180/360	
Notch (fliter)	
Wah wah (pedal)	
Wah Wah	
Foot switch	
Intensity	
Drive	
Sync	

Wah-Wah spectrum	

Distortion	
Distortion	
Clipping	
Fuzz (hard clipping)	
Overdrive (soft clipping)	
Valve Distortion	 
Transistor Distortion	

Harmonics/ Partials	
Compression	
Distortion insert Logic	
Distortion Pedal/Stomp Box	
Re-amping	
Digital distortion	

Tremolo	
Tremolo	
LFO Depth	
LFO Rate	
Phase	
Smoothing	
Vibrato	
Vibrato	
LFO Depth	
LFO Rate	
Vocal effects	
Autotune	
Pitch Corrector (Shifter)	
Vocoder	

Talk Box	
Ring Modulator	
Ring Modulator	A circuit that combines two incoming signals and outputs only the sum and differences of the frequencies.

1.14 Mastering	
General	
Perceived volume	See levels 2.4
Limiting	Brick wall compressor. Used to stop tracks from clipping.
Mastering for different formats	EQ/Compression/panning will be different for vinyl, tape and digital. E.g. Bass will have to be cut in vinyl masters.
Full-range speakers e.g. NS10/Aura tone	Speakers that replicate the full frequency spectrum as accurately as possible with a flat frequency response so as not to add any colour to the sound.
Reference Tracks	Pre-existing audio used as exemplar material for how the final master should sound.
Mastering plugins	Use algorithms to add EQ/dynamics processing/reverb holistically based on analysis of the waveform to create the loudest/more clean output for you.
Re-mastered tracks	Old tracks re-released after being mastered on new technology.
Domestic Play-back Mastering	Mastering using HiFi systems to make the track sound as good as possible on home speakers
Processes	
Noise-Reduction (Dolby)	Removing unwanted noise from a signal either white noise or hum to achieve a better signal to noise ratio.
Exciters	Saturation/boost added to frequencies C. 3.5Hz and up which adds brightness and crispness to the audio.
Modern 'Loudness Wars'	Recent trend in production to make mixes as loud and powerful as possible whilst not distorting.

Dithering	**See Sampling 1.4**
Redbook Standard	Technical specification for CD formats. 16bit 44,100Hz.
CD track Markers	Don't necessarily interrupt the playback on an concept album but signify where each track begins.
Silence Gaps	Added during the mastering process to make sure the gaps between tracks on a CD are the correct length and truncating if neccessary.
Radio Edits	Cutting out sections/creating a shorter version or deleting swear words for radio play.
Stereo width	Changing the overall panning e.g. narrow in verse and wide in chorus. Stereo spreaders can do this too.
Master Reverb	Adding reverb to the output of the entire track to 'glue' the mixes together**.**

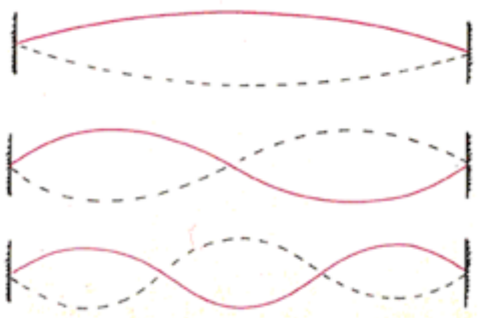
PRINCIPLES OF AUDIO AND SOUND TECHNOLOGY

2.1 Acoustics

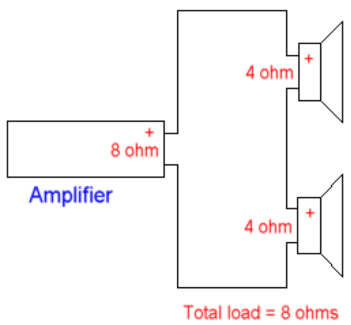
(he way the surfaces in a room absorb, reflect and diffuse sound)

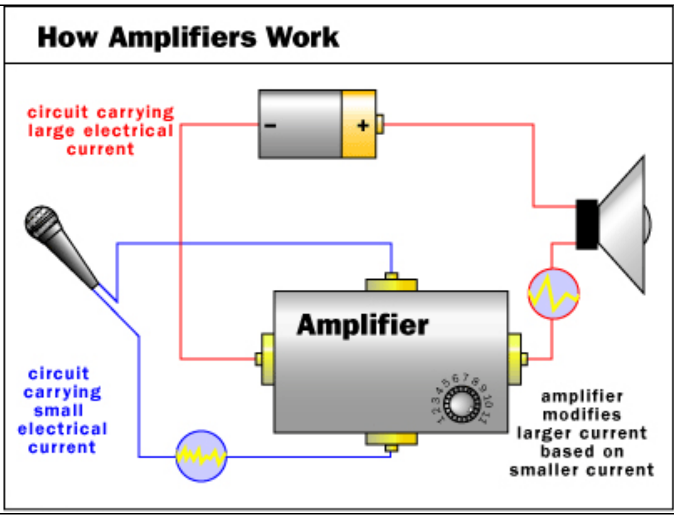
Live room acoustics

For more detail in reverb, see 1.12

Room size	The larger a room, the bigger the pre-delay before the early reflections.
Absorption	When the surface doesn't reflect all of the sound waves back – it takes in some of the sound energy.
Monitor Speakers	Studio monitors with a flat frequency response designed for mixing, meaning no frequencies are artificially emphasised.
Colourisation	Changes in tone/timbre of a sound as a result of unwanted (early) reflections within a room.
Reflections	Sound waves bounce off any reflective surfaces in a space. Reflections can cause cancellation of, or an increase in amplitude. The more reflective a room, the longer the RT_{60}
Diffusion	Diffusion scatters sound waves from angled surfaces over a wide area.
Natural reverb	Reverb captured in the intended sound source (e.g. a church)
Chamber reverb	Reverb created by use of an acoustically treated echo chamber.
Standing waves	 Can cause phase cancellation or increase in the amplitude of a frequency depending on the acoustics of a room and the frequency of a sound. It's a bad thing.
Bass Traps	Absorbs bass frequencies in order to attain a flatter low frequency response in a room. The essentially turn sound energy into heat through friction.

2.2 Speakers, amps and monitors

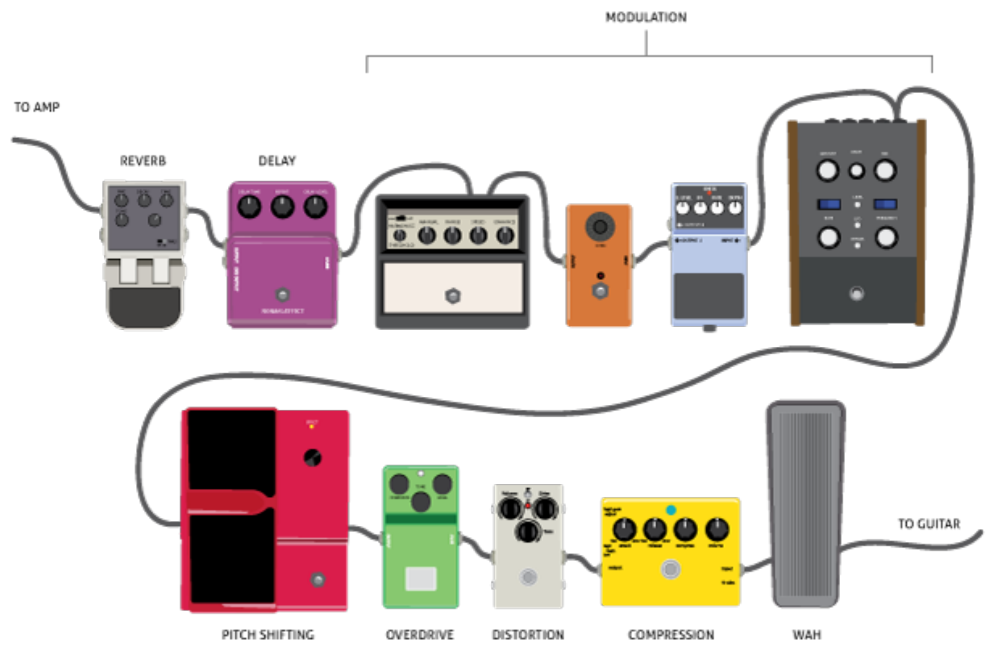
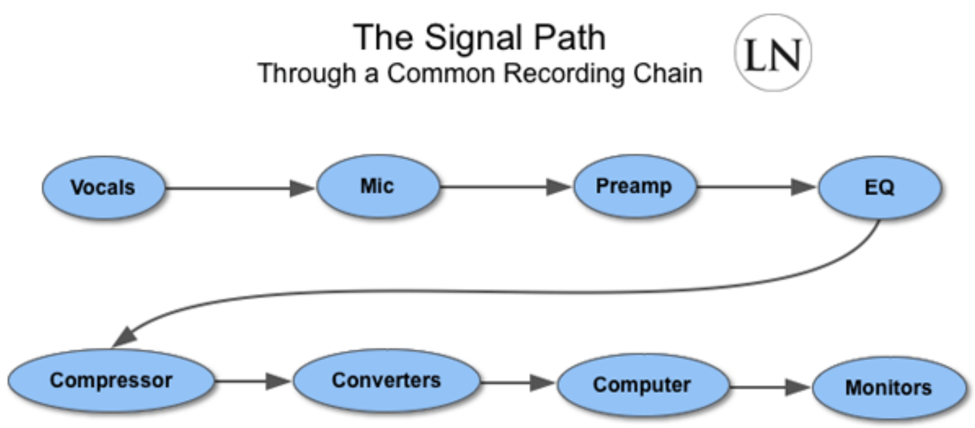
Types and characteristics	
Tweeter	Small speaker cone that is designed to play high frequencies.
woofer	Large speaker cone designed to play low frequencies
Subwoofer	Very large speaker cone designed to play frequencies below about 30Hz
Driver unit	Cone and magnet and coil device only. Everything else is the speaker.
crossover	Device for filtering out high/low frequencies that go to the woofer/tweeter.
Active speaker	Powered (by kettle lead). Does not require an amplifier.
Passive speaker	Require an amplifier.
How speakers work	
Speaker cone	Moves forwards and backwards based on the electrical charge from the copper coil, which in turn moves the magnet, which is attached to the speaker cone. Creates variations in air pressure.
Electro-magnetic induction	An electrical current is created when a magnet is passed through/moved through a copper coil.
Speaker impedance	Measured in ohms. It measures the 'resistance' of a speaker (speakers a big electricity resistors). If you think of electricity as water flowing through a pipe, the lower the ohms, the lower the resistance so the 'pipe' is bigger and more than flow through.
ohms	 *Diagram 3* The lower the ohms the lower the impedance (resistance) and the better quality the speaker. Professional level speakers are 4ohms. *Impedance is frequency dependent.*
Types of monitoring	
Headphones	

Flat speakers	
5.1	
Beats headphones…	
Bad speakers…	
Monitoring mixes	
Input/recording Mix	
Monitoring Mix	
Cue Mix	
Types of Amps	
How an amp works	
External Pre-amp	**See 1.1**
internal Pre-amp	
Power Amp	

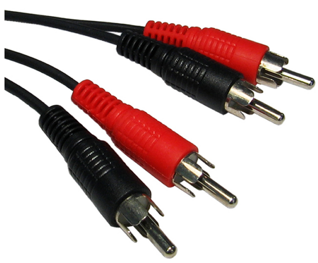
HIFI Amp	
Active Speaker	
Instrument Amp	
Bass Amp	
Guitar Amp	
Keyboard Amp	

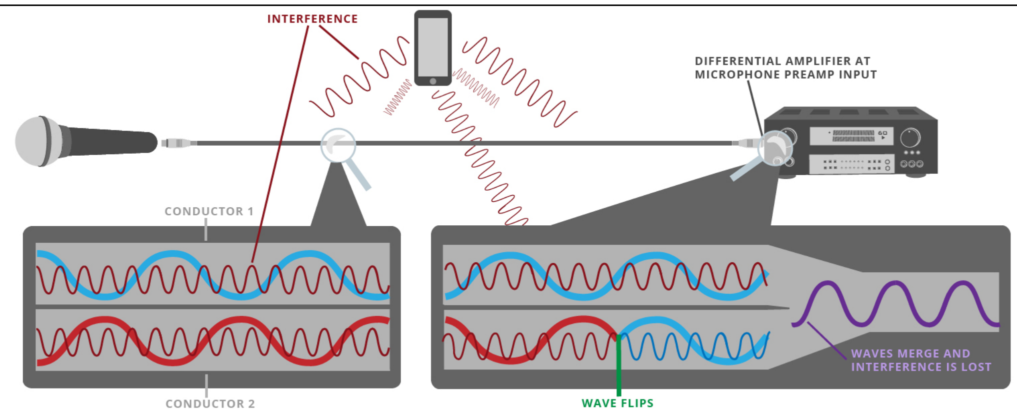
2.3 Leads and signals

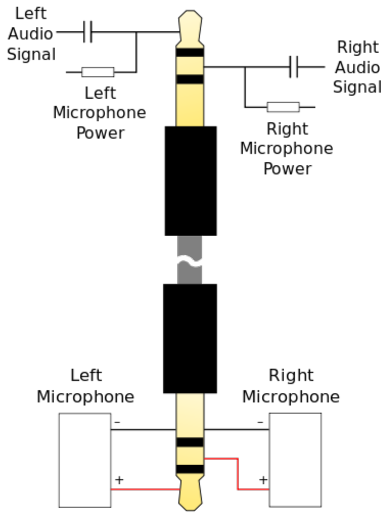
Connectivity including signal path and signal types

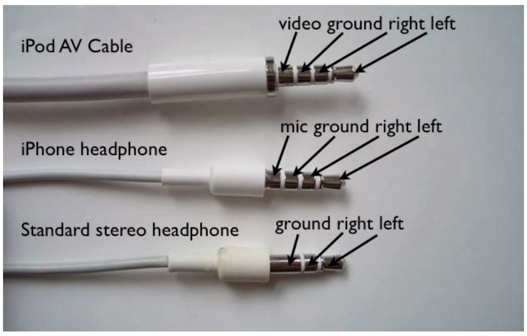
Effects signal path	 The generally accepted order of placement for effect types begins with the wah and proceeds to compressor, distortion or overdrive (if you're using both, place the higher-gain unit before the lower-gain pedal), modulation units (if you use multiple modulation units, as shown here, experiment with the order), delay, then reverb.
Recording signal path	 It is in this order because…

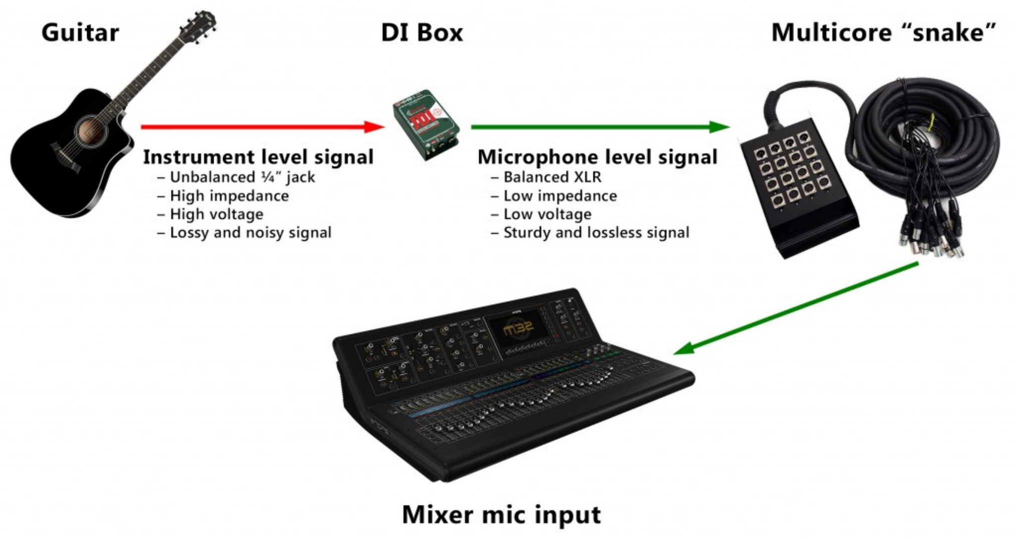
The different types and uses of leads	
What connections look like:	
Hiss/hum/rumble	
TRS Jack	Balanced stereo Jack. T= tip. R = ring S = sleeve.
XLR	Three pin. Used to connect microphones, DI boxes and other audio signals in a studio. It is Balanced. Ground, positive and negative.
TOSLINK	Fibreoptic-cable ‘light pipe’. Used on consumer Audio Visual (AV) products. Used for ADAT.
ADAT	Has toslink a connection (see above). Found on audio interfaces. Can send multi audio channels in one digital cable. Up to 8 channels simultaneously. Used to expand the input/output count of an audio interface by connecting (using ADAT) with an expansion unit with more inputs so you can record more channels all at the same time.
FIREWIRE	Like USB but faster. Less common, but still used on some interfaces.
USB	Universal serial bus. USB A/B/C available.
S/PDIF	Looks like a phono cable but carries digital audio. You only need one cable as digital can carry both.
TS	Mono jack. Just tip and sleeve.

RCA	See Phono. (Stands for Radio Corporation of America)
MIDI	5 pin din connector. Largley obsolete. Computers not transmit MIDI by USB. Used to be used for other audio stuff in the studio. Some synths only have MIDI out (you can get MIDI USB converts).
Phono	 Normally used for Hifis and AV equipment. Comes from the word ‘phonograph’. Popular with DJs. It is unbalanced. Can only carry a single audio source so two are required for stereo. Phono connectors can also be known as RCA. Used for digital audio in a different form. See SPDIF
Thunderbolt	Form of USB.
Impedance	
Impedance	
Instrument input signal level	e.g…
Mic input signal level	e.g…
Line input signal level	e.g…

<table>
<tr><th colspan="2">How leads work</th></tr>
<tr><td>Balanced Connection (XLR)</td><td>

Step 1: Signal inverted by 180 degree and sent over two wires
Step 2: signal reinverted at the end by amp.
Therefore:
interference which enters both cables is…..
Original signal is actually…</td></tr>
<tr><td>Unbalanced connection</td><td>Inside the recording device


The microphone wiring</td></tr>
<tr><td>Phone cables!</td><td>
</td></tr>
</table>

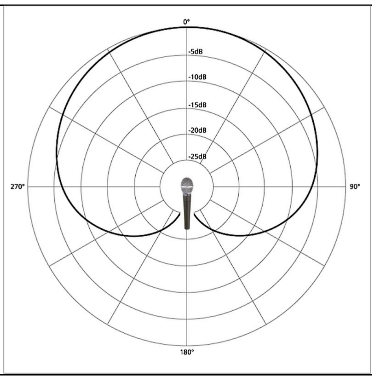
DI box	
Ground Lift	
The advantages/disadvantages of different leads and connectivity	
DI Box	
Jack Lead (Unbalanced)	
XLR (Balanced)	
Digital connections (Firewire; USB; S/PDF)	

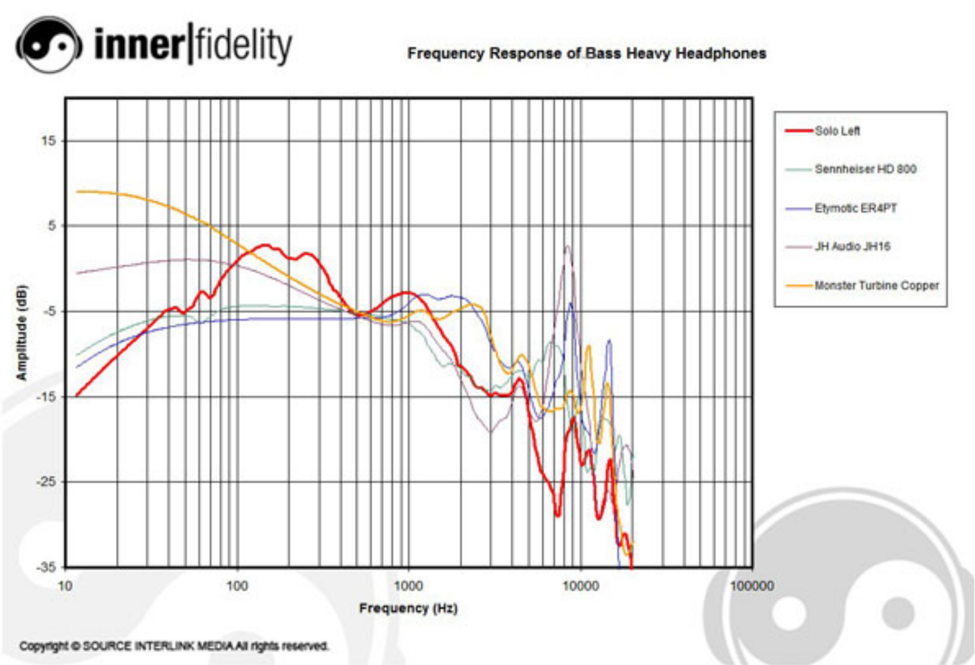
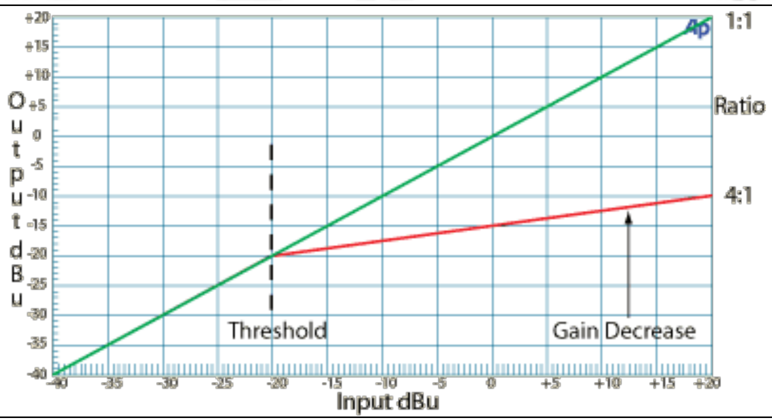
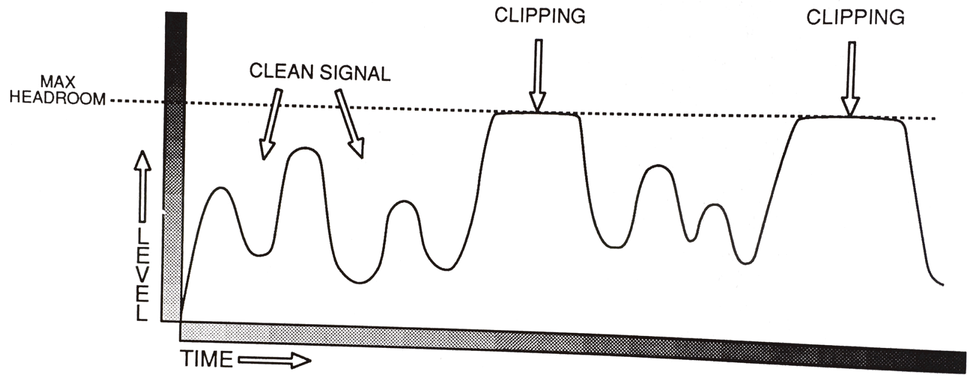
2.3 Numeracy

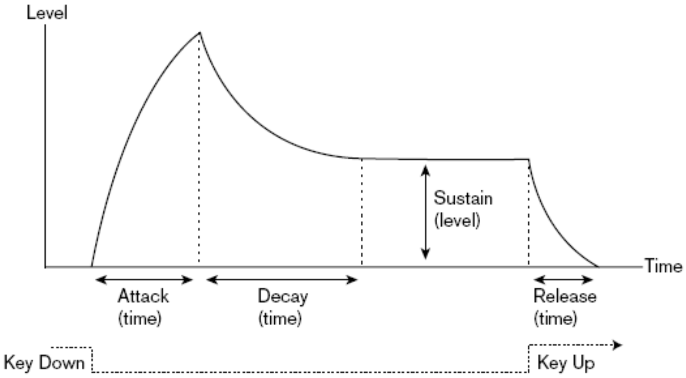
Parameters you need to know

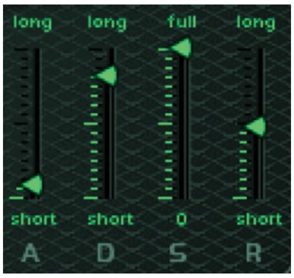
Loudness dB	
Frequency (Hz)	
LFO Frequency (Hz)	
Delay time (m/s)	
Tempo (BPM)	
8ve ‘feet’ settings	Middle C (261 Hz) = 2’
Semitones	
Cents	
Feedback (%)	
Mix (%)	

Graphs

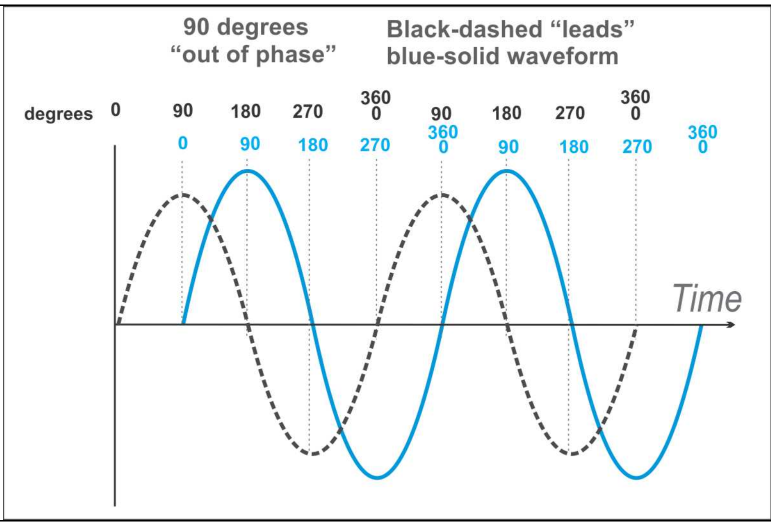
Polar Response	

Frequency Response	
Compressor response	
Clipped wave form	
Amplitude envelope	

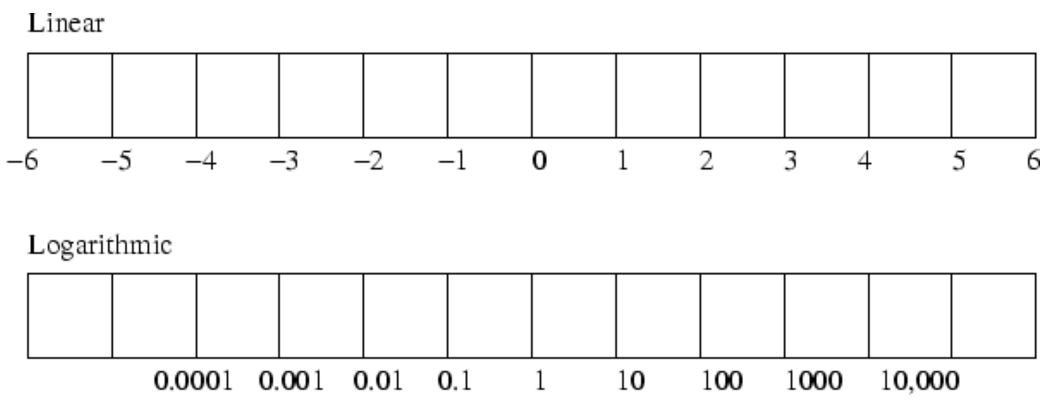
<table>
<tr><td></td><td>

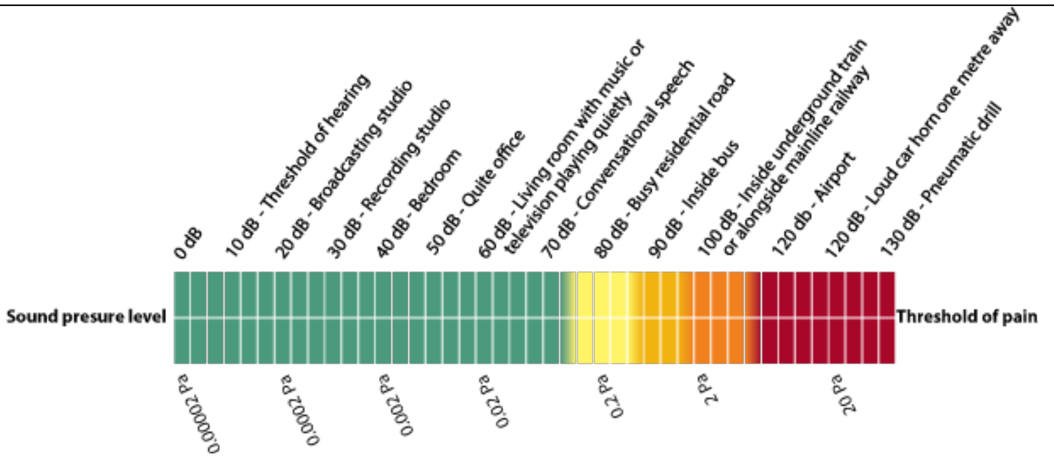


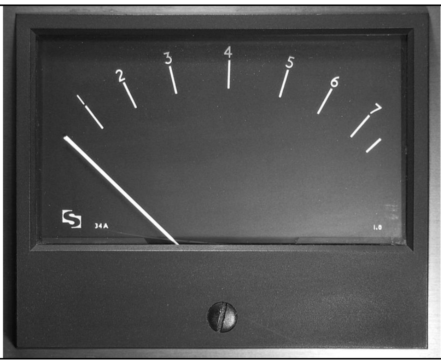
Do these match? Why/ why not?</td></tr>
<tr><td colspan="2">Calculations based on sound waves</td></tr>
<tr><td>Amplitude</td><td>The amplitude is…

</td></tr>
<tr><td>Rate</td><td></td></tr>
<tr><td>‘Out of phase’</td><td>
</td></tr>
</table>

2.4 Levels

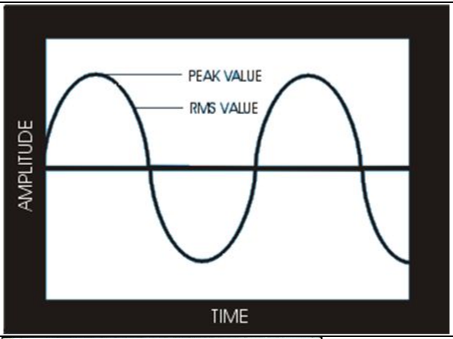
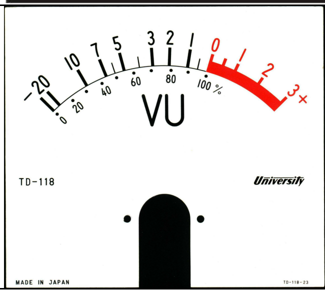
<table>
<tr><th colspan="2">Principles of levels and metering</th></tr>
<tr><td>Perceived loudness</td><td>How loud humans hear things. It is a combination of: sound pressure level (dBs), frequency (Hz) – some frequencies are louder than others and length of sound (RMS).</td></tr>
<tr><td>Psycho-acoustics</td><td>A scientific discipline which studies how humans perceive sound.</td></tr>
<tr><td>Logarhythmic Scale</td><td>


</td></tr>
<tr><td>(dB)SPL</td><td>


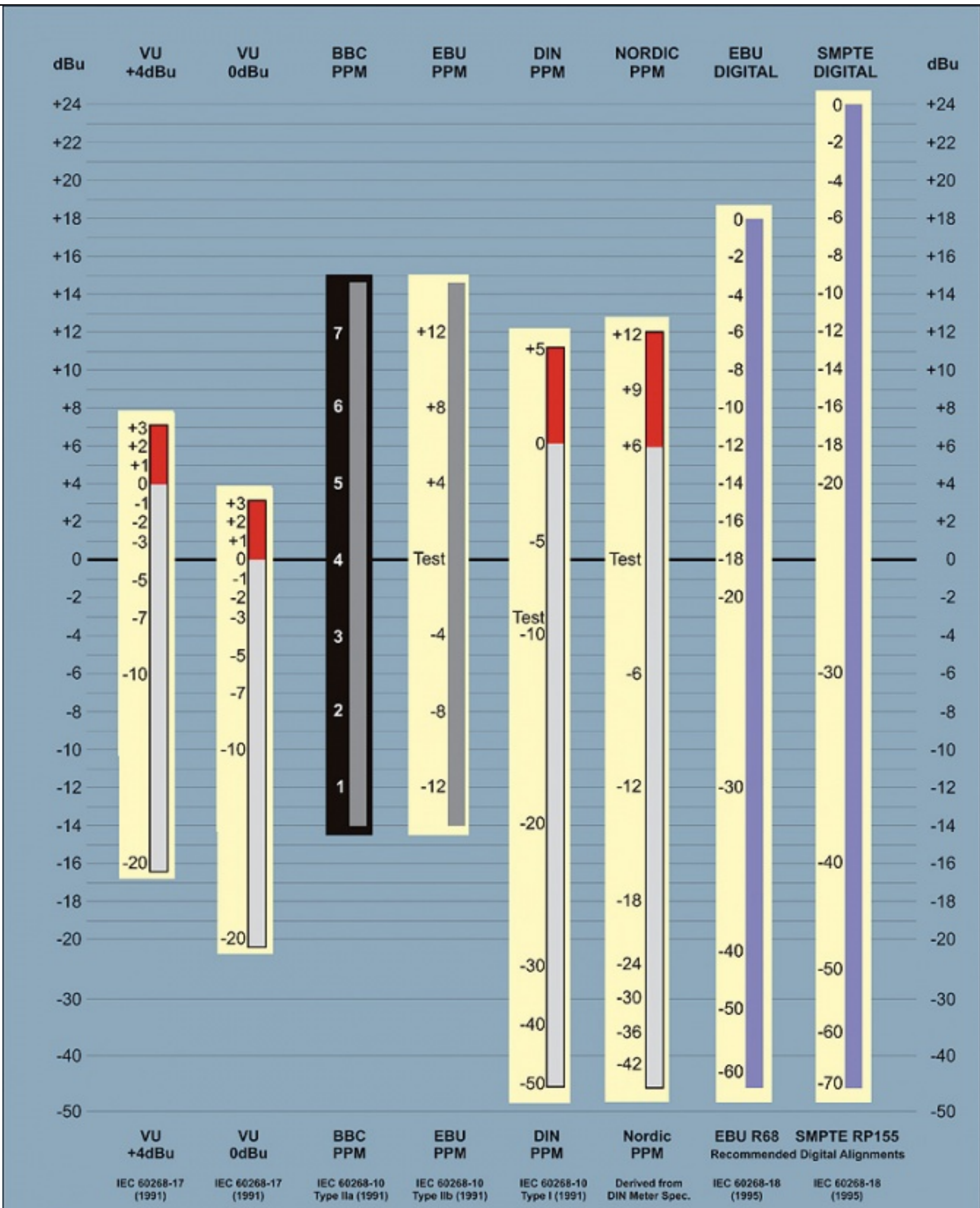
</td></tr>
<tr><th colspan="2">Levels and metering scales</th></tr>
<tr><td>Amplitude (dB)</td><td>The height of a sound wave measured in decibels. Measurement of loudness.</td></tr>
<tr><td>Peak Programme Meter (PPM)</td><td>

Best used for:

Measuring all peaks in the sound.
Making sure there is no clipping that might damage the equipment.

</td></tr>
</table>

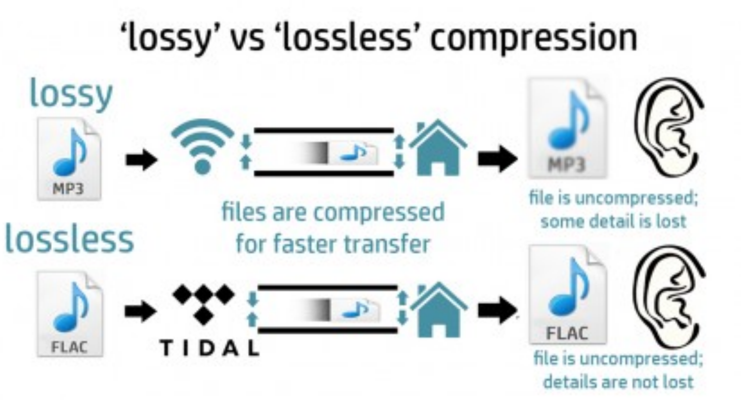
RMS Meter (Root mean squared)	 Best used for: • Measuring average loudness, just like our ears do. • It takes into account both the amplitude *and* the duration of the peaks. • It averages out peaks and troughs to give you the perceived loudness as our ears work in a similar way.
VU Meter (or SVI meter)	
0dB/Unity gain	The *input* is the same as the *output* (i.e. the sound level is equivalent to the alignment level of 0dBSPL 20 µPa). IT DOES NOT MEAN SILENCE
-30dB	
+3/+10dB	
Normalize	
dBu	Lots of **professional equipment** uses dBu with an *alignment level* of '0' on their meters to **+4dBu** (which is aligned to a voltage of 1.228V (rms)). U = unterminated, if you're interested.
dBv	Used mainly for semi-pro and amateur equipment. Their alignment level is -10dBV to 0.316mV (millivolts). V = volts
dBFS	Starts at 0 and work downwards. Used for all digital equipment. They have an alignment level **of -18dBfs with 0dBu**. **LOGIC USES THIS.**

Comparison of different meter types.

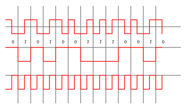
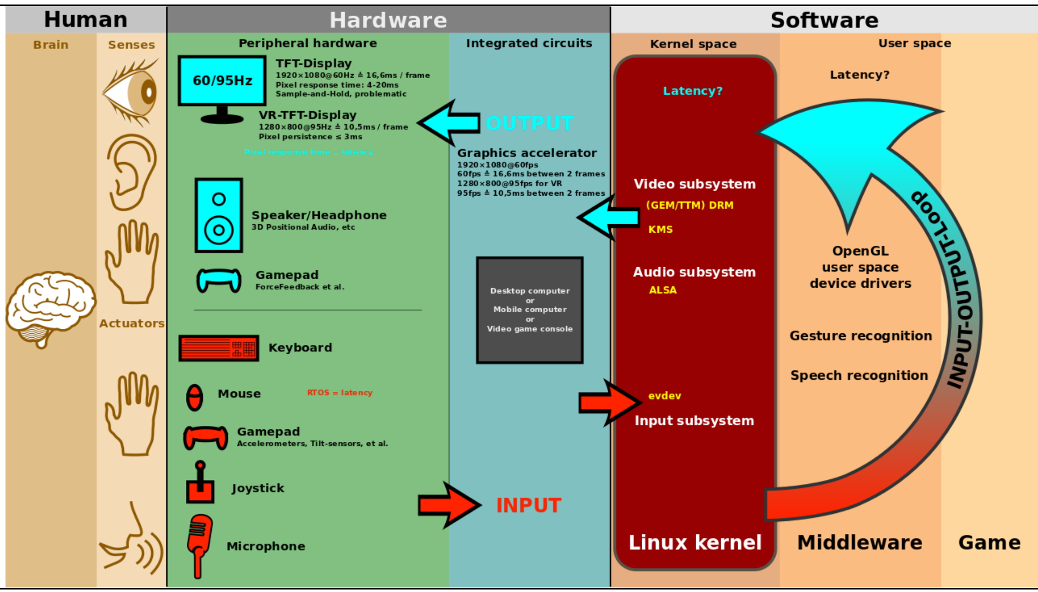


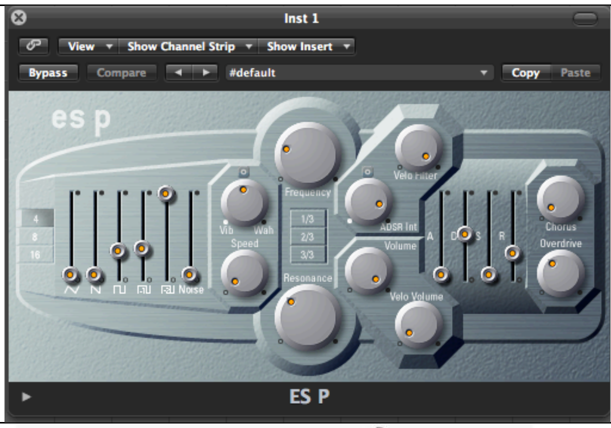
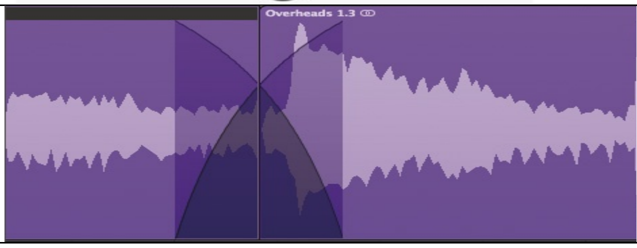
This chart illustrates the comparative scales and nominal relative alignments of the most common legacy analogue audio meter types. It should be noted that the actual alignment levels vary with different applications, and that modern Loudness with True Peak metering (based on the ITU-R BS-1770 specification) supersedes all of these historical meter forms. Some common analogue sine-wave tone alignment levels associated with different applications include:

US Broadcast (traditional):	+8dBu
US Music, Film and modern broadcast:	+4dBu
UK Broadcast:	0dBu
German Broadcast:	-3dBu
Other Territories:	Local Preferences

Digital recording specs	
Audio compression	A reduction in the number of bits needed to represent data to save storage capacity and speed up file transfer.
PCM (Pulse Code Modulation)	Method for sampling analogue signals. Umbrella term for forms like WAV and AIFF (See below)
WAV/AIFF	Uncompressed audio files. AIFF (audio interchange file format) is Apple's version.
Mp3	Compressed *audio* file. Short for MPEG-1 audio later 3! Bit rate of 96 – 320kbps. The most common format for compressed digital audio. Originated in year 1990s.
Mp4	Stores audio, video and still images. MPEG-4 part 14. Invented in 2000s.
CODEC	Hardware or software device that compresses/decompresses data.
FLAC/ALAC	FLAC = free lossless audio codec. ALAC used by *Apple* as lossless audio codec. The file types ending in .m4a
WMA	'Windows media audio'. Windows version of MP3
Ogg Vorbis	A streaming alternative to MP3 currently used by Spotify.
lossy/vs Lossless	

DEVELOPMENT OF RECORDING AND PRODUCTION TECHNOLOGY

<table>
<tr><th colspan="2">3.1 Digital equipment</th></tr>
<tr><th colspan="2">Digital hardware/software attributes</th></tr>
<tr><td><u>Digital</u> Equipment/recording Characteristics
</td><td>Frequency Response:
•
•
Signal to noise Ratio:
•
•
Headroom:
•
•
Clipping:
•
•</td></tr>
<tr><td>Graphical User interface(GUI)</td><td>
</td></tr>
</table>

Digital sequencing and digital audio workstations (DAW)	
Advanced functions of a DAW	**See 1.1**
Native (or real time processing)	
Software instruments	
Disruptive editing	
Non-disruptive editing (also known as non-linear)	
Convolution Reverb	See Reverb (1.12)

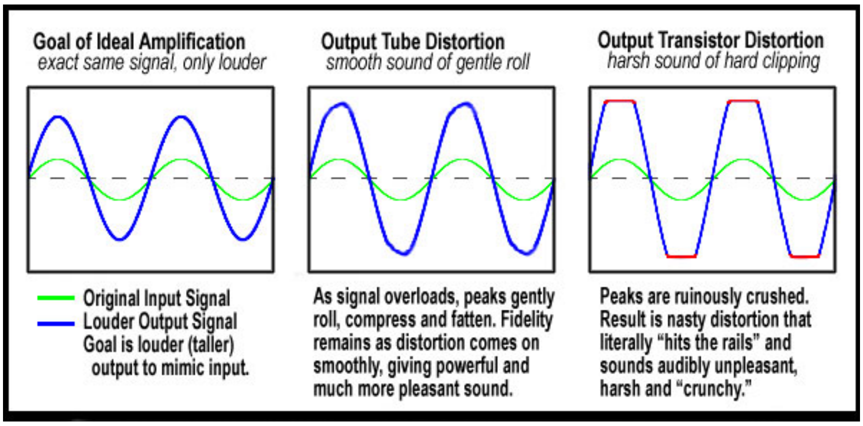
Digital consumer formats	
MQA	
M4a	
CD	
pits	
Digital recording and sampling hardware	
Digital Multi-tracker	
Digital Samplers	
Synclavier (1978)	
Fairlight CMI	
E-MU ESI Series (1994)	

Akai S1000	
Digital Drum Machines	
Linn LM-1 (sampling with limited memory!)	
Roland V-drums	
AKAI MPC60	

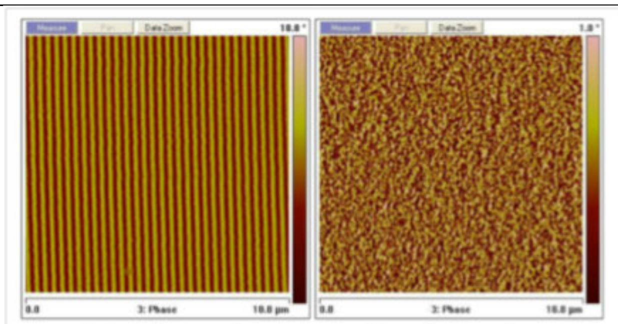
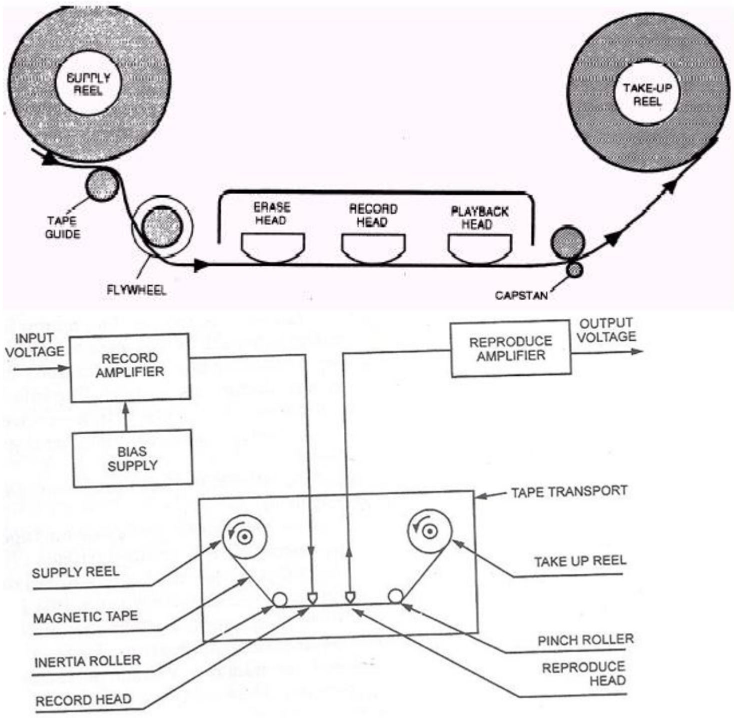
3.2 Analogue equipment

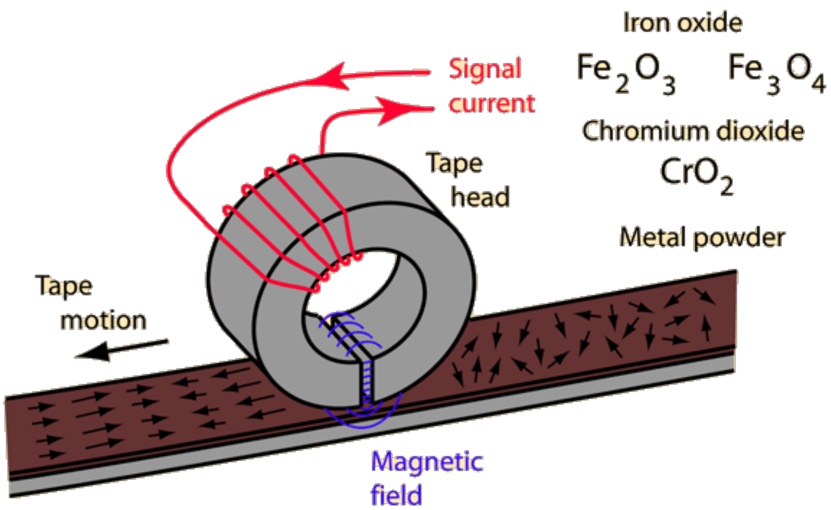
Analogue hardware attributes

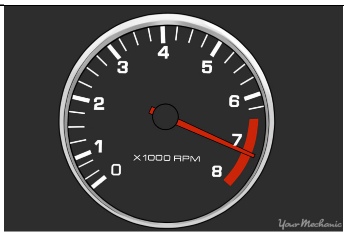
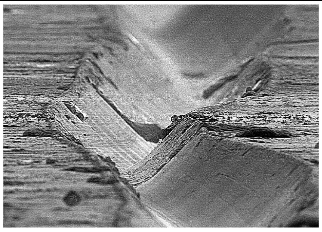
Analogue Equipment /recording Characterstics 	Frequency Response: • Narrow • 35Hz – 10Hz Signal to noise Ratio: • Low ratio – lots of hiss • Headroom: • Less head room • Clipping: • Clipping easier – more likely to clip • Quieter overall
Valve	  A valve works by… passing a current through a tube with a vacuum (like a light bulb). It affects the sound by…causing soft clipping (distortion/ overdrive) of the wave form.

Transistor	 A Transistor works by… It affects the sound by…
Fidelity	Hi-fi: high fidelity – high quality reproduction of sound Lo fi: low quality (normally digital)
Transistor v. valve distortion.	
Tape Saturation	 Analogue distortion in tape**.**
Wow	**Variations in pitch**
Flutter	

Analogue pitch problems.	 Oscillators need to be tuned manually by ear.
Crackle /hiss/ jumps	Needle jump out of the groove. Crackel/hiss = dust on the groove and bad low signal to noise ratio.
Warping	when vinyl gets too hot and cannot be played.

Tape machines	
A tape Machine	 stores and records tape.
Degaussing	 Before degaussing After degaussing Removes any unwanted magnetism. Used for re-recording.
Tape machine set up	

Supply Reel	unrecorded tape stored and fed out from here.
Take up Reel	Stores the recorded tape.
Play head	 Opposite mechanism to tape head. Creates a current based on the tape magnetism.
Tape Head (Recording)	A magnet wrapped in copper coil.
Pinch Roller	Directs the tape back to tape up reel and controls the tape speed
Iron Oxide	
How tape works	 **The** magnetic ring carries a current which replicates the sound waves. This produces a magnetic field in the ring proportional to the audio signal. The tape then becomes magnetised and the iron oxide powder order to replicate the sound waves on the tape.
Splicing	 Cutting tape to physically truncate the audio.
Print Through	Also known as bleed-through. Undesirable effect where unwanted music from other tracks on the tape can be heard.

Analogue consumer formats	
vinyl	**Popular consumer format from 1920s – 1980s.**
7/10/12 inch Vinyl	 12" single @ 45 rpm 10" single @ 78 rpm 7" single @ 45 rpm
RPM	rotations per minute. The slower the rotation the more music can fit on a disc.
Groove	**Carved into the disc. A physical representation of the sound wave.**
Stylus	Holds the cartridge which holds the needle. Induces an electrical current corresponding to the wveform.
Needle	Diamond tip, picks up the wave form. 

Cassette Tape	 Dates from 1970s. Consumer way of listening to tape. Portable with protective casing.
C60/C90 Tape	
Walkman	 Portable tape player.
Mixing/ mastering principles of Analogue	• Light EQ • Light compression • Removing extreme frequencies.
RIAA Curve (vinyl)	**Recording Industry association of America:** Makes grooves smaller by applying EQ to reduce low frequencies so the stylus doesn't move so much. EQ then applied during playback to compensate for the loss of bass.

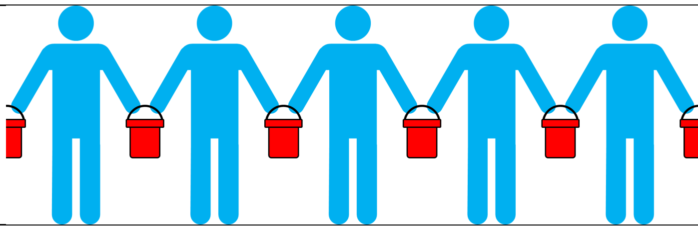
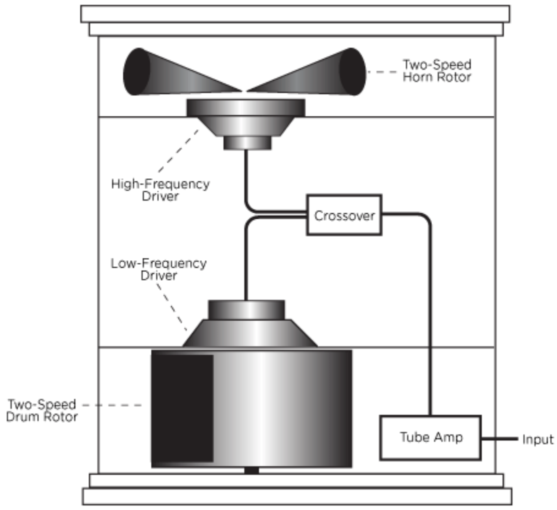
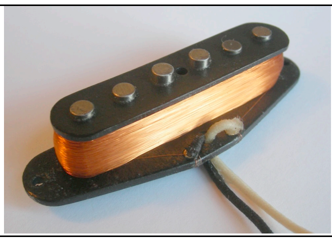
<table>
<tr><th colspan="2">Analogue effects</th></tr>
<tr><td>Tape Delay</td><td>See 1.12</td></tr>
<tr><td>Bucket Brigade</td><td></td></tr>
<tr><td>Plate/Spring Reverb</td><td>See 1.12</td></tr>
<tr><td>Rotary speaker/’leslie cabinet’</td><td>

FIG 1. SCHEMATIC DIAGRAM OF LESLIE® TWIN-ROTOR SPEAKER SYSTEM</td></tr>
<tr><td>Vinyl Scratching</td><td></td></tr>
<tr><td>Tape/vinyl ‘reverse’</td><td></td></tr>
</table>

Analogue synthesizers	
Patching	
module	
Minimoog (1969)	
Roland Jupiter-8 (1981)	
Yamaha DX-7 (1983)	
Sequential Circuit Prophet 5	
Hammond Organ	

TB-303	
Korg M1 (1988)	

Electric instruments	
Theremin	
Mellotron (1962)	
Electric Organ	
Hammond Organ	
Electric Piano	
Clavinet	

Guitars	
Electric Guitar	
Bass Guitar	
Parts of the Guitar	
Head	
Tuning pegs	
Neck	
Frets (see above)	
Pick ups	
Pole-piece (See above)	
Electro-magnetic induction	
Whammy Bar/Tremolo Arm	

Truss Rod	
Copper Coil	

The History of Recording and Production Technology

Students are required to develop knowledge and understanding of the history and development of recording and production technology, from current digital technologies back to the mono, analogue recording technologies in the 1950s, through the following eras:

Direct to tape mono recording (c.1950 – 1963)

Notes:

Early multitrack recording (c.1964 – 1969)	

Large-scale analogue multitrack (c.1968 – 1995)	

Digital recording and sequencing (c.1980 – present day)	

DAW and emerging technologies (c.1996 – present day)	

Music Styles	
Students should have knowledge and understanding of the instruments, the sounds associated with them and the combination of instruments and voices used in the following styles:	
Jazz	
Blues	
Rock 'n' Roll	

Rock	
Metal	
Punk	
	Melanie and Hannan and Lara.

Soul	
Disco and Funk	
Reggae	
	Hannah and Vince

Acoustic & Folk	
Commercial Pop	
	Chris and Martha and Gabriela
Urban	
	Hiphop/Rap: Teodor Zia and Will **Garage and Grime.** **Instrumentation** **At least 3 artists** **Artists** **(History)** **Influences** **What it led to** **Music tech equipment**
Electronic and Dance	
	Drew and Joe. **Early house and some garage.** cover House (and all it's many sub-genres), Techno, Trance, Drum 'n'

	Bass/Jungle, Big Beat, UK Garage, triphop, Dubstep, Reggaeton)