Introduction to Music Production

Lesson 1: Sound and Signal Flow

#1 Arrival time of sound. So much of what we do in a mix is to create a sense of space, depth and a real location. All these ideas are based on the idea of **PROPAGATION**.

- **Propagation** - sounds moving through the air.
- **Amplitude** - Speed of Sound. (1k/3 sec; 340 m/sec; 1 ft/msec; 1km/3 sec; 1 mile/5 sec. (approximations)
- **Frequency / Timber**
- **Effect Categories effected in a Mix:** Delay, Reverb, Phasers, Flangers

#2 AMPLITUDE - The extent of the wave: how much the air compresses and rarifies (makes more complex) as the waveform propagates thru the air.

- **SOUND in AIR** - the direction of vibration is parallel (the same), as the direction of propagation.

SLINKY DEMO: Longitudinal Wave - push wave through without up and down waves.

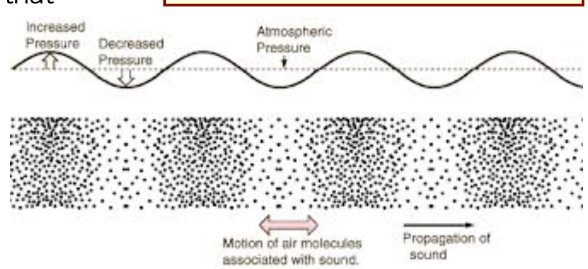
- The amount of compression is the amplitude. **A Transverse Wave:** Swing Slinky back and forth rigorously, high amplitude. Slowly, low amplitude. (Guitar String)
- **Longitudinal Wave (Air):** Push gently: low amplitude, push hard, big wave, high amplitude. It is a longetudinal rarifaction and it's moving in the same direction as the propagation.
- **Rarifaction** (less dense as the air moves by). **Compression** (more dense as the air moves by). The extent of that is our **Amplitude** (louder or quieter to our ears) is measured in decibels. (**dbspl-decibals sound pressure level**). It's a relative measure (no set point for 0). 0=Quietest to pain. Lowering: called Attenuation.
- Air is measured by dbspl (the lowest thing we can hear) 0 to PAIN.
- **Amplitude (measurable by a computer) or Loudness (our perception of amplitude): dbfs** (full scale) the loudest thing that can be represented in numbers within the computer. 0 is the LOUDEST, then negative from there. Amplitude is measureable by a COMPUTER, Loudness is our Human Perception of it.
- **Loudness: Duration** (how long) and **Frequency** (we hear high and lows differently).
- **Dynamic Plug Ins:** expanders, gates, compressors, limiters.
- **Dynamic Range of a Mic:** decibels it will reproduce to sound properly. **In Gear:** the range levels of the noise floor (quietest) up to distortion. **In Music** - range of quietest section to it's loudest.
- **Addition Look Ups—**Log Rhythms, Dynamic Range, Decibels, Fletcher Munson Curves, Equal Loudness.
- **Effects Categories in a mix:** Compressors, limiters, expanders and gates.

prop·a·ga·tion n.

1. Multiplication or increase, as by natural reproduction.

2. The process of spreading to a larger area or greater number; dissemination.

3. *Physics* The act or process of propagating, especially the process by which a disturbance, such as the motion of electro-magnetic or sound waves, is transmitted through a medium such as air or water.



Created by Dianna Burrup (diburrup@gmail.com)

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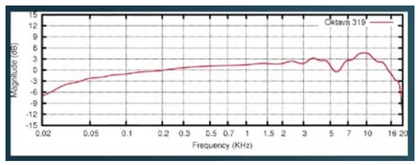
FORUM: Useful Links for Sound:

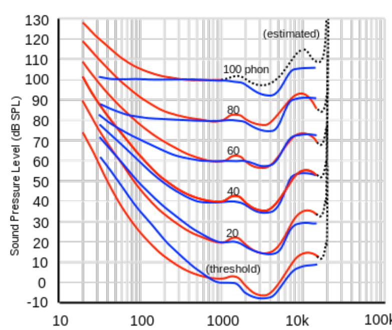
http://www.thefreedictionary.com/propagation
http://en.wikipedia.org/wiki/Sound#Propagation_of_sound
http://www.ndt-ed.org/EducationResources/CommunityCollege/Ultrasonics/Physics/modepropagation.htm

#3 FREQUENCY (measured on a computer). It's how FAST a sound is vibrating. Hertz is measure of frequency.

- Our sense of Hi and Low is Pitch. **Pitch** is something we **perceive, like Loudness.** Low - slow pulse, High - lots of pulses.
- **Transverse Wave:** Swing Slinky back and forth rigorously, high frequency. Slowly, low frequency.
- All the principals of sound are independent of each other.
- **Propagations:** are Delays and Reverbs.
- **Amplitude:** expanders, gates, compressors, limiters.
- **Frequency:** (more related to **Timbre** - collection of sound at multiple frequencies.) **Sine Wave -** energy at a Single Frequency; **Instruments** - Multiple Frequencies, i.e. Harmonics, Overtones, Spectrum, Timbre.
- **BOOST** (raise the amplitude) the **Bottom End** (Frequency). "EQ" - An Equalizer is a collection of filters. Amplitude at a specific frequency is a filter, and it's manipulating the timbre.
- **Range of Human Hearing:**
  Lowest = 20 Hertz
  Highest = 20,000 Hertz
  1 Hertz = 1/second
- We don't hear equally across the range.
- Frequency Response Curve Of your gear, acts like an EQ.



- Frequency and Timbre is important to everything we do.
- **Fletcher Munson Curves.** The first research on the topic of how the ear hears different frequencies at different levels was conducted by Fletcher and Munson in 1933. In 1937 they created the first equal-loudness curves.



Equal-loudness contours (red) (from ISO 226:2003 revision)
Fletcher–Munson curves shown (blue) for comparison

- **LOOKUP**
  Sound masking: addition of natural or artificial sound (white noise or pink noise) into an environment to cover up unwanted sound by using auditory masking.
- A sound is said to have a **missing fundamental**, **suppressed fundamental**, or **PHANTOM FUNDAMENTAL** when its overtones suggest a fundamental frequency but the sound lacks a component at the fundamental frequency itself. The brain perceives the pitch of a tone not only by its fundamental frequency, but also by the periodicity implied by the relationship between the higher harmonics; we may perceive the same pitch (perhaps with a different timbre) even if the fundamental frequency is missing from a tone.
- **FREQUENCY RESPONSE:** is the quantitative measure of the output spectrum of a system or device in response to a stimulus, and is used to characterize the dynamics of the system. It is a measure of magnitude and phase of the output as a function of frequency, in comparison to the input.
- **TIMBRE (overtones, relative levels of partials in a sound):** is the quality of a musical note or sound or tone that distinguishes different types of sound production, such as voices and musical instruments, string instruments, wind instruments, and percussion instruments. The physical characteristics of sound that determine the perception of timbre include spectrum and envelope. Timbre is what makes a particular musical sound different from another, even when they have the same pitch and loudness.
- **Manipulates Timber & Spectrum:** Filter Effects.

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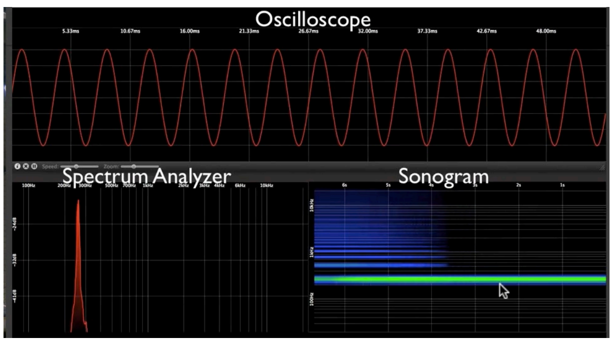
FORUM: Useful Links for Visualizing Sound

A previous students explanation of propagation, amplitude, frequency and timbre:
http://www.youtube.com/watch?v=bbT8TN-jKXI
Visual demonstration of a concept we've all heard before relating to frequency and pitch.
http://www.youtube.com/watch?v=ngk-ECb8ccQ
Awesome Frequency Chart: http://www.independentrecording.net/irn/resources/freqchart/main_display.htm

#4 VISUALIZING SOUND

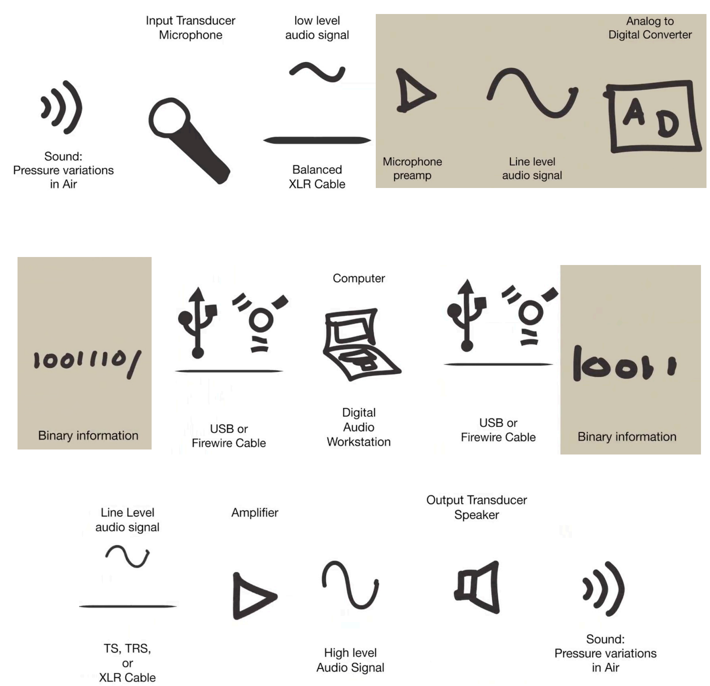
3 Different types displays to give us a good visual representation in visualizing sound.



- **OSCILLOSCOPE:** (like a wave form in a DAW) Microscopic view of the audio track. See how pressure varies in the air. Vertically (Y) is **Amplitude** and Horizontally (X) is **Time**. (Hard to see the frequency and timbre of the sounds.) Accurate representation of the speakers movement. Amplitude changes the wave but not the frequency. Timbre change shows how much the sound is changed.
- **SPECTRUM ANALYZER:** We see where the EXACT frequency it is at ... Horizontal (X) — **Frequency** and Vertically (Y) — **Amplitude.** Can tell that sound has a lot of energy at 2k, or 500 hz, etc. It doesn't give us a sense of how the sound will change over time. Momentary picture. **Spectrogram Analysis** gives us the sound changes over time. Timbre is shown as a series of peeks. Octave changes as frequency. Sawtooth Wave - harmonic series: 1st-Fundamental - Each frequency is at an integer multiples of the fundamental frequency. (You will often see a spectrum analyzer in the EQ. The role of the EQ is to manipulate the timbre-relative levels of the partials.
- **SONOGRAM:** Changes over time. (like a Spectrum Analyzer flipped on it's side) **Y-Frequency** is vertical, **X-Time** is Horizontal. **Z-Amplitude. Changing Amplitude:** Gives us a history of the how the timbre and the spectrum have changed. Change the Timbre (converting to saw tooth wave): additional harmonics are shown and how they change over time.
- **Vowel Sounds:** AEIOU: Sonogram: Fundamental Frequency is the basic C plus upper harmonics, variations in pitch is shown as changing frequencies. The mouth is a complex filter (EQ). vowel Sounds are primarily variations of Spectrum.
- **Oscillator** (a Sound Creator).
- **Rarefied:** Less Dense than atmospheric pressure.

#5 CONNECTIONS OVERVIEW

- **Signal Flow - Mic to Computer:** Mic (Input Transducer)Audio Interface: Mic Preamp, Analog to Digital Converter, then to computer and to the DAW.



#6 Mic as a Transducer

- Changes **sound** variations to **voltage** variations. Mics will color the signal in SOME way. Looking at SM58, AKG 414 Condenser: Type, Polar Pattern & Frequency Response to see how to place them when recording common signals.

Lesson 1, Con't: Sound and Signal Flow

#7 Mic Types

- **Condenser** — In Studio. So sensitive you can get feedback loop if used live. Phantom Power or +48 Volts. Med to Large Diaphragm is good one to get started with.

- **Dynamic** — On Stage because it doesn't pick up outside of it's small area very well. It's rugged. Does not require external power.
- **Acoustics of the room and the mic placement has a much bigger impact than the mic.**

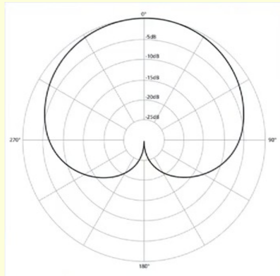
#8 Mic Frequency Response

- **Vocal Range Mic (SM58)** — Freq Response Chart comes with any mic. This has a peek at 5,000 hz which helps voice come in well.
- **Condenser Mic**-Picks up EVERYTHING Range. Flat Frequency Response. Picks up everything equally. "What you hear is what you get."

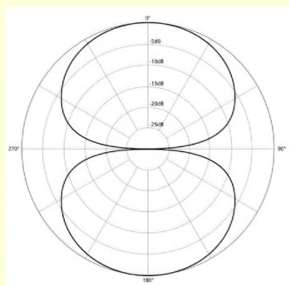
#9 Mic Polar Patterns

- **Polar Pattern -** Describes what it picks up well and what it rejects.
- Always remember when recording the choice of a polar pattern will have a major impact on how much of the space or the room that you are actually capturing.

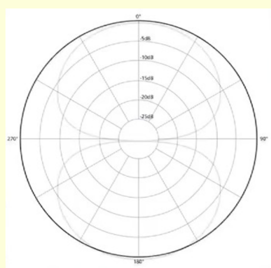
- **Cardiod Pattern (plus different styles of this pattern).**



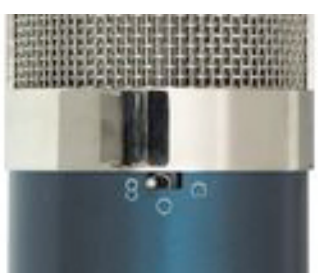
- **Figure 8**



- **Omnidirectional**



- Decide if you want to capture more of the space or the instrument.
- Some mics lets you choose the pattern with a switch, but it also changes the frequency response.

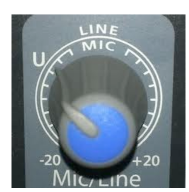


#10 Mic Placement

- **Placement is the Most Important in Recording.**
- **General Guideline:** Microphones are your ears in the studio. Walk around and listen. Notice how it sounds different when you move around.
- Front Addressed (SM58) and Side Addressed (AKG414). Point the Logo at the item you are recording.

#11 Line Level and Gain Staging

- There are 2 standard line levels: +4 Studio Level and -10 Consumer Level. Ideally you want to bring it up ONCE and leave it there.
- TRIM: **U stands for Unity** - you are not amplifying or attenuating (bringing up or down).
- **Mic Preamp:** Brings it up the the standard line level.
- The worse thing you can do in your signal flow is applify it at one point, then attenuate it at another, then amplify it again. Keep it at Line Level during the entire signal flow.



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#12 Cables

- Get high quality cables!
- **1/4" Cable:** also known as an instrument cable or a **TS** Cable. **Tip and Sleeve**, single conductor cable, outer sleeves prevents noise. Use as short a cable as possible.
- **1/4**: **TRS** Cable. **Tip Ring and Sleeve** or Stereo Cable.
- **XLR Cable:** Has 3 connectors like TRS. Only used in a balanced configuration. Use these if you need a LONG cable runs (or Balanced type cable)
- **Direct Box: IN**put TS for as a short run and out to XLR for a long run. Has a parallel output.
- **1/8":** Same as the TRS but smaller.
- **RCA:** Functions just like a 1/4" cable.

#13 Interface

- Audio Interface: MOTU - all inputs need to be TRS, Midi Input and Output, Digital Signal in and out, Firewire is good for recording multiple inputs, USB for a few at a time.
- Provides ins and outs, XLR-Mic PreAmp (Trim or Gain), 48 (Phantom Power), PAD is Attenuation (reduces the level of the signal) and headphone output.
- Functions also as an analog to digital converter.

#14 Mic Connection & Gain

- Avoid the sound of making the connection from going through your computer.
- **Connection:** Reduce the input gain all the way down on the Interface. But does not guarantee no sound will come though tho.
- Turn off Phantom Power OFF.
- XLR Cable: MIC is MALE.
- Turn ON Phantom Power.
- **Level:** Set the Level where we want it to be in our final recording. Find the loudest piece you will be in the music.
- Never let it go into the red or get the very top. Better off to keep it a little bit lower.
- **Disconnection:** Turn off the output volume (monitors); mic gain down, then turn off phantom power.

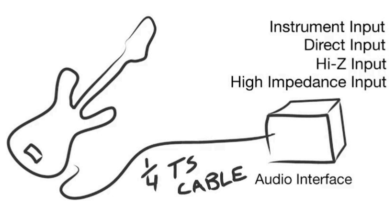
#15 Analog to Digital Converter

- Signal from Mic - analog or continuous signal. Preamp is also an a digital converter, also called a quantization or sampling. Sampling is taking the smallest pieces of audio so we can represent it in 1s and 0s.

#16 Pickup Connections

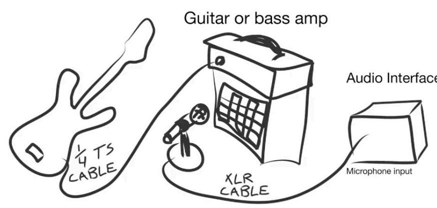
- Connecting a guitar/bass:

Pickup option 1



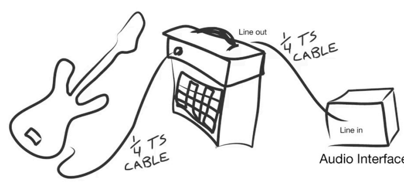
- Turn TRIM all the way down as well as the speakers. Connect bass to the device, make sure it is an Instrument connection.
- There can be a little of a delay (or latency) from bass or electric guitar coming from the speakers opposed to "in the air" as an acoustic guitar.
- Need to monitor these the signal (playback) outside of the computer. Plug into an amp then into the interface: Using a mic you get the sound of the room:

Pickup option 2



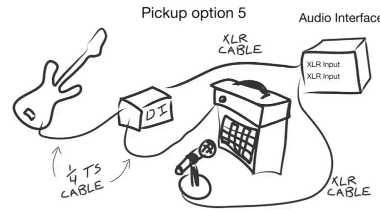
- Recording with a LINE OUT from the amp into your interface (you'll hear it in real time):

Pickup option 3



- If no line out, you can use a DIRECT BOX - using a SHORT 1/4" instrument cable to LONG XLR direct to your interface (gets impedance correct). Then input direct to your amplifier from the direct box. Preferred Method, OR:

Pickup option 5



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Lesson 2: The DAW

#1 Lesson Overview

- Defining PRODUCTION: **Pre Production** - composing song, planning. **Production** is the actual tracking/ recording. **Post Production** - Editing-Mixing-Mastering.
- All DAWs have the same basic features / functions.
- **MIDI** (Musical Instrument Digital Interface) a realtime interpretation of a score.

#2 Analog to Digital

Computer can only process Binary Info: Based on the BIT, single memory location. 1 or 0.

	1 Bit word	2 Bit word	3 Bit word	Decimal number
	0	00	000	0
	1	01	001	1
		10	010	2
		11	011	3
			100	4
			101	5
			110	6
			111	7
$2^{wordlength}$	2	4	8	

$2^8 = 256$
$2^9 = 512$
$2^{10} = 1{,}024$
$2^{11} = 2{,}048$

$2^{22} = 4{,}194{,}304$
$2^{23} = 8{,}388{,}608$
$2^{24} = 16{,}777{,}216$

- CD Standard: 16-Bit word; In the studio: 24-bit, gives you a wider dynamic range. Every Bit doubles in value.
- **Word Length** is related to **amplitude** (the longer the word length, the wider the dynamic range - resolution).
- **24-bit recording** allows you record quieter. Always do this!
- **Sampling Rate (**related to **frequency)** how often we do the measurements. We have to measure 40,000 times per second. The higher the sampling rate the more accurately it represents the higher frequencies. Higher sampling rates are used in video. **Use 48,000 Hz sampling rate.** Speeding up the rate shortens up the length and vice versa.

Audio CD Standard
16-bit wordlength
44,100 samples/sec.

#3 Buffer Size

- Computer collects a “QUE” (buffer) of samples.

Lowering buffer size reduces latency, but also reduces the number of plugins that can be used.

Raising buffer size increases latency, and increases the number of plugins that can be used.

While recording use a low buffer size, try 128 samples. During post production raise buffer size to 1024 when necessary.

- Everything is based on powers of 2.
- You always need to set your buffer size.
- Increase buffer size when computer starts choking.
- **Freezing a Track** - computer records to audio and reduces CPU Strain.
- **Buffer size** is a collection of individual samples.

Sample Rate of 48 KHz
= 48,000 samples/sec.

Buffer Size = 128 samples

Delay = 2 milliseconds

Sample Rate of 48 KHz
= 48,000 samples/sec.

Buffer Size = 1024 samples

Delay = 21.3 milliseconds

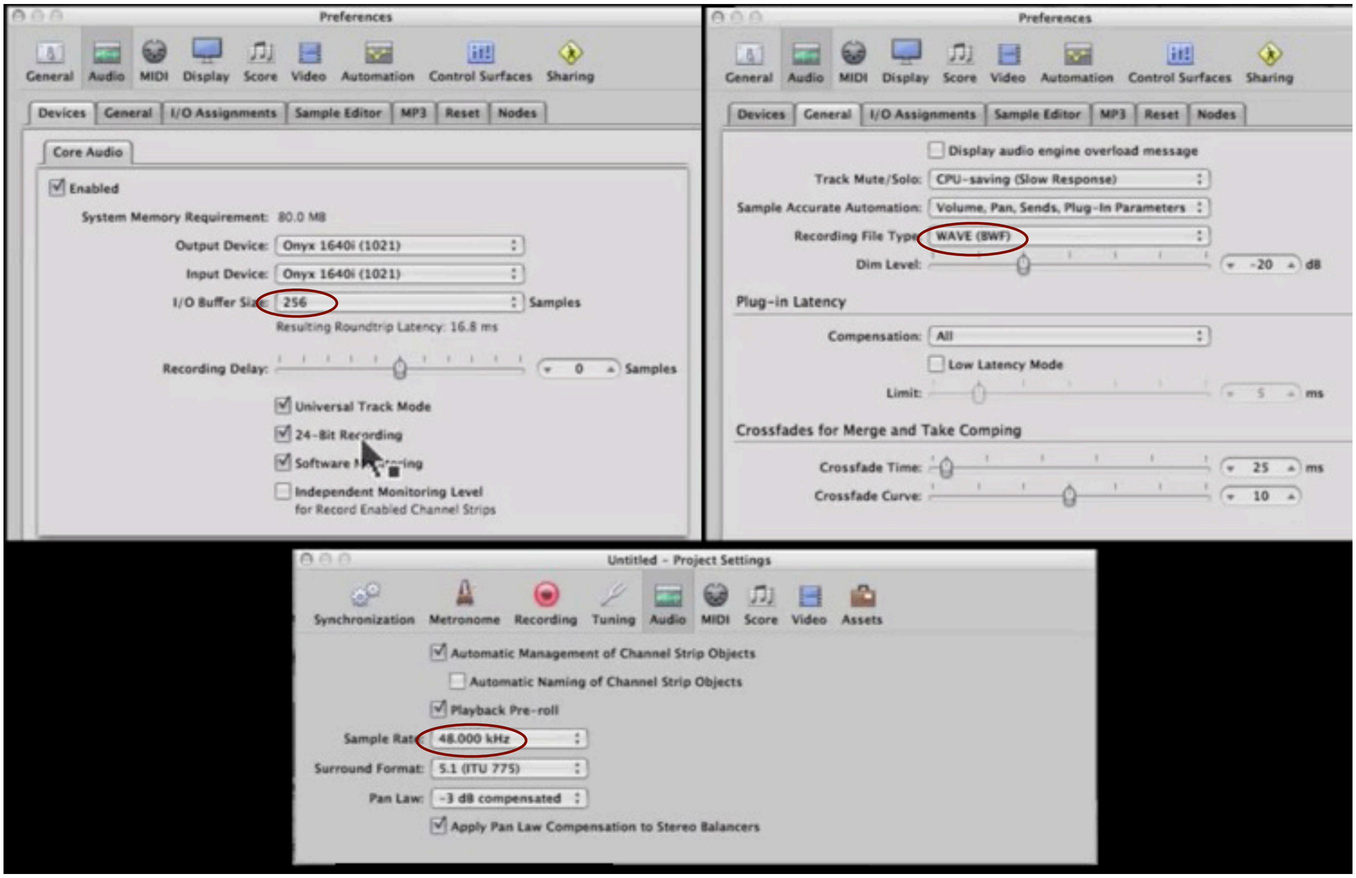
- You may to constantly adjust buffer sizes during the production process if you do not have a high powered computer.

#4 File Size

- An increase in **bit depth and sample rate** makes much larger files.
- **Data Compression** - like a zip file - lossless, keeps all data but reduces file size.
- **Audio Compression -** WAV or AIFF large, highest quality (lossless). **Broadcast Wav (BWF)** Files stores more metadata - use if possible.
- **Interleaved (iTunes)** - single file that includes both left / right audio channel - use this, along with 24 Bit and 48 kHz Sample Rate. 128 samples Buffer Size for Recording.
- **De-Interleaved** - separate files for left and right.

Lesson 2, Con't: The DAW

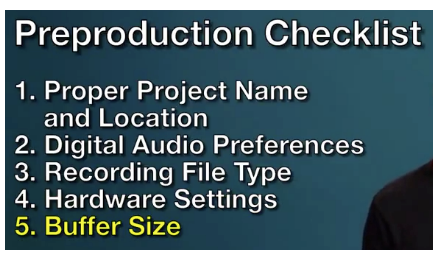
The LOGIC SETUP of PREFERENCES



#5 Project Folder (6:18)

- Contains a Proprietary File and the ASSETS: Project File. Audio Files, Crossfades, Analysis Files, Undo History (Backups) and Intermediary files for time stretching. Look at your DAWs Folder.
- **Do not save a project folder inside another project folder.**
- **To Share:** Move the entire Project Folder with all the assets. Leave things the way are saved by the DAW. Zip.
- **IN LOGIC**: 2 project files can occupy the same Project Folder, utilizing the same Audio Files Folder.

#6 Project Checklist



#7 Multitrack

- **Types of Tracks: AUDIO** (some have Mono and Stereo); **MIDI** - real time "score"; **AUX** - provides routing capabilities. submix, effects, **routing tool**. **INSTRUMENT** - inputs one type of data and outputs another. **GLOBAL** - tempo, & meter changes, markers and key changes.

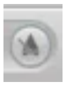
#8 Recording Audio

- **Mono or Stereo** - Input makes a BIG difference. Single Mic or pickup (Mono) - use this most of the time. Name it for organization.
- **Instrument** - buffer needs to be low, 24 bit, sample rate 48K.

- Audio - 3/4 of the way up the meter. Always trust your ears.
- Turn off the click and listen.

Lesson 2, Con't: The DAW

#8 Recording Audio (LogicX), con't



- Record another version by duplicating the last track:

- You can edit between them and pick the best performance.

#9 Trimming (LogicX)

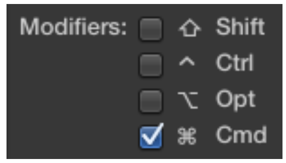
- Strive to get a great recording rather than editing right away. **Use editing for fine tuning.**
- Two things are created when you record: an **Audio File** and a **REGION (Clip)** - it's a window into the sound file, a reference. Much of the editing we do is only to the regions. Non destructing editing, not editing the underlying data audio file.
- 1st thing we can do - adjust it' to a region is **TRIMMING** - adjust it's bounds. (Play with editing the bounds of the regions in your DAW.)

#10 Separating and Cutting (Split at Playhead) (LogicX)

- Split the region: REGION - (Logic) Split in icon bar at top. **(or Cmd + T)**

#11 The Grid (LogicX)

- When Moving a Region - be able to turn off grid easily and change the resolution.
- Learn the modifier key that bypasses the grid: CNTL G.



#12 Fades (LogicX)

- Sometimes there a pops or clicks at a split region. Use a perimeter to cut out at zero crossing, hides the noise of that sound.
- Learn key command to add a fade. **CNTL I (fade in) CNTL O (fade out)**
- When you are overlapping regions, apply a **cross-fade**. 1st region dips down and 2nd comes up. Usually in the regions, but see if there's a key command. **Option X.**

Fade In	**Fade Out**
^I	^O

#13 Zooming (LogicX)

- **Select First, then Zoom.** Keeps selection right in the middle of the screen. **Under + in far right corner.**

Zoom Window
⌥⌘M

Zoom Horizontal Out
⌘←

Zoom Horizontal In
⌘→

Zoom Vertical Out
⌘↑

Zoom Vertical In
⌘↓

Recall Zoom 1
^⌥⌘1

Recall Zoom 2
^⌥⌘2

Recall Zoom 3
^⌥⌘3

#14 Cycling (LogicX)

- **(Repeat)** Learn how to cycle a section: turn off and turn on.

Cycle Mode
C

Cycle Audition
^C

#15 Merging-Join (LogicX)

- With lots of regions and lots of cross-fades - you can make a new audio file with all the edits - consolidating or merging. **This is destructive edit.**

Join Regions/Notes
⌘J

Join Regions per Tracks
J

#16 Naming and Coloring Regions (LogicX)

- Right mouse click to colorize, (Text Tool Under + in far right corner.)

Show/Hide Colors
⌥C

Color Tracks by Region Color
⌥⇧⌘C

Color Regions by Track Color
⌥⇧C

#17 Markers (LogicX)

- Specific location in your song. Logic: In Global tracks. You can also colorize - pick color wheel, colors type.

Create Marker
⌥'

Create Marker for Selected Regions
⌥⇧'

Delete Marker
⌥⌫

Go to Previous Marker
⌥,

Lesson 2, Con't: The DAW

#18 Comping (LogicX)

- Choosing the best features from all tracks or takes into one track.
- Create a new track, set the input to NONE.
- Grid set to BAR.
- Using OPTION and select cuts and move to the new track.

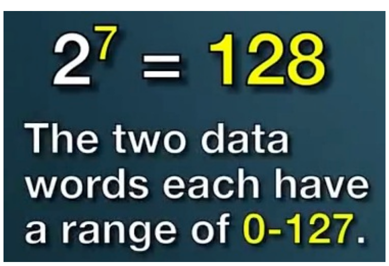
Rename Take or Comp
⇧T
Delete Take or Comp
⌥⇧⌫
Toggle Take Folder Quick Swipe Comping Mode
⌥Q

#19 Midi

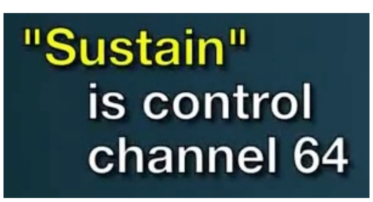
Musical Instrument Digital Interface

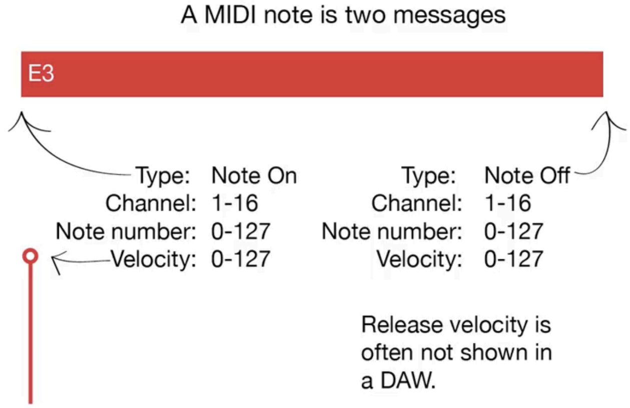
- Midi data messages are relayed in CHANNELS: 1-16 like TV Channels. Its not a direct representation of SOUND - it's like a SCORE.

- **2 data words** (7 bit numbers - see below): gives us the parameters of the message: On or Off, or what note it is, and the velocity.

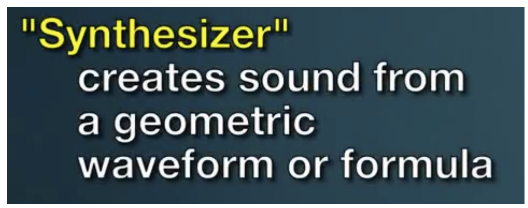


- Not all controllers send velocity.
- Stuck Note: A Panic sends a All Notes Off.





- Channel Pressure is also known as Aftertouch.
- There's a SENDING machine and a RECEIVING device and the latter determines what happens with the data.
- Midi going to a DAW is Automation, to a Midi Synthesizer it's a control change message.



#19 Midi Controllers

- **Class Compliant:** doesn't need a driver.
- Uses 5 pin Din Cable. Midi keyboards are connected through a USB, which is bi-directional.
- 8 knobs on Midi Keys controls 8 parameters. Can be configured to send any Control Change (CC). Software can also be used.

#20 Software Instruments

- **Midi Data Performance:** Plugins can be used in a DAW, virtual instruments.
- Go to your DAW and add a soft-synth.

Lesson 2, Con't: The DAW

#21 MIDI Edition: Velocity

- In Midi Data: every single note is a separate event that can be edited individually, just like audio, except that the view is different, like a BAR instead of a wave.
- Adjust the velocity in a piano roll in your DAW.

#21 MIDI Editing: Quantization

- Tightens up a performance in by pulling up to subdivisions.
- First, set the grid, 16th note is a good place to start, but adjust according to your performance notes.
- Quantization strength - how close to the grid it goes: 100% pulls all the way the he grid.

Use 20% repeatedly until it sounds good enough that it still retains the human feel.

#22 Common MIDI Recoding and Editing Functions

- **Multi-Sampling:** different velocities recorded for an instrument.
- Synthesizers will tax your CPU, calculating the waveforms in real time.
- **Piano Roll or Midi Editor** - find velocities for editing. You have to scroll vertical and horizontally and zoom.
- You can grab the edge, middle or move to a different pitch. To get an audio sound, use midi monitor or headphones icon in your DAW.
- **To quantize small portions,** right mouse click quantize, and set the grid, amount: 20% and keep using that until you like it. Some allow start/end quantize. Doing just the beginnings of notes and leaving the endings as they are make for a more musical performance.
- Overall Quantize: 50%.
- **Voicing:** Octave (also called range or register) changing. He moved the lower note up an octave.
- **Batch Processing:** Select all, move all velocities down at once.
- **Overdub or Midi Merge** to add a melody over it. (Creates it on a line above the chords)

###

L O G I C X

Lesson 2, Con't: The DAW

MIDI Control Change Messages – Continuous Controllers

MIDI CC Number	MIDI CC Purpose	MIDI CC Description
MIDI CC 0	Bank Select	Allows user to switch bank for patch selection. Program change used with Bank Select. MIDI can access 16,384 patches per MIDI channel.
MIDI CC 1	Modulation	Generally this CC controls a vibrato effect (pitch, loudness, brighness). What is modulated is based on the patch.
MIDI CC 2	Breath Controller	Often times associated with aftertouch messages. It was originally intended for use with a breath MIDI controller in which blowing harder produced higher MIDI control values. It can be used for modulation as well.
MIDI CC 3	Undefined	
MIDI CC 4	Foot Controller	Often used with aftertouch messages. It can send a continuous stream of values based on how the pedal is used.
MIDI CC 5	Portamento Time	Controls portamento rate to slide between 2 notes played subsequently.
MIDI CC 6	Data Entry Most Significant Bit(MSB)	Controls Value for NRPN or RPN parameters.
MIDI CC 7	Volume	Control the volume of the channel
MIDI CC 8	Balance	Controls the left and right balance, generally for stereo patches. 0 = hard left, 64 = center, 127 = hard right
MIDI CC 9	Undefined	
MIDI CC 10	Pan	Controls the left and right balance, generally for mono patches. 0 = hard left, 64 = center, 127 = hard right
MIDI CC 11	Expression	Expression is a percentage of volume (CC7).
MIDI CC 12	Effect Controller 1	Usually used to control a parameter of an effect within the synth/workstation.
MIDI CC 13	Effect Controller 2	Usually used to control a parameter of an effect within the synth/workstation.
MIDI CC 14	Undefined	
MIDI CC 15	Undefined	
MIDI CC 16 – 19	General Purpose	
MIDI CC 20 – 31	Undefined	
MIDI CC 32 – 63	Controller 0-31 Least Significant Bit (LSB)	
MIDI CC 64	Damper Pedal / Sustain Pedal	On/Off switch that controls sustain. (See also Sostenuto CC 66) 0 to 63 = Off, 64 to 127 = On
MIDI CC 65	Portamento On/Off Switch	On/Off switch 0 to 63 = Off, 64 to 127 = On
MIDI CC 66	Sostenuto On/Off Switch	On/Off switch – Like the Sustain controller (CC 64), However it only holds notes that were "On" when the pedal was pressed. People use it to "hold" chords" and play melodies over the held chord. 0 to 63 = Off, 64 to 127 = On

MIDI CC 67	Soft Pedal On/Off Switch	On/Off switch - Lowers the volume of notes played. 0 to 63 = Off, 64 to 127 = On
MIDI CC 68	Legato FootSwitch	On/Off switch - Turns Legato effect between 2 subsequent notes On or Off. 0 to 63 = Off, 64 to 127 = On
MIDI CC 69	Hold 2	Another way to "hold notes" (see MIDI CC 64 and MIDI CC 66). However notes fade out according to their release parameter rather than when the pedal is released.
MIDI CC 70	Sound Controller 1	Usually controls the way a sound is produced. Default = Sound Variation.
MIDI CC 71	Sound Controller 2	Allows shaping the Voltage Controlled Filter (VCF). Default = Resonance - also(Timbre or Harmonics)
MIDI CC 72	Sound Controller 3	Controls release time of the Voltage controlled Amplifier (VCA). Default = Release Time.
MIDI CC 73	Sound Controller 4	Controls the "Attack' of a sound. The attack is the amount of time it takes forthe sound to reach maximum amplitude.
MIDI CC 74	Sound Controller 5	Controls VCFs cutoff frequency of the filter.
MIDI CC 75	Sound Controller 6	Generic – Some manufacturers may use to further shave their sounds.
MIDI CC 76	Sound Controller 7	Generic – Some manufacturers may use to further shave their sounds.
MIDI CC 77	Sound Controller 8	Generic – Some manufacturers may use to further shave their sounds.
MIDI CC 78	Sound Controller 9	Generic – Some manufacturers may use to further shave their sounds.
MIDI CC 79	Sound Controller 10	Generic – Some manufacturers may use to further shave their sounds.
MIDI CC 80	General Purpose MIDI CC Controller	Generic On/Off switch 0 to 63 = Off, 64 to 127 = On
MIDI CC 81	General Purpose MIDI CC Controller	Generic On/Off switch 0 to 63 = Off, 64 to 127 = On
MIDI CC 82	General Purpose MIDI CC Controller	Generic On/Off switch 0 to 63 = Off, 64 to 127 = On
MIDI CC 83	General Purpose MIDI CC Controller	Generic On/Off switch 0 to 63 = Off, 64 to 127 = On
MIDI CC 84	Portamento CC Control	Controls the **amount of Portamento.**
MIDI CC 85 – 90	Undefined	
MIDI CC 91	Effect 1 Depth	Usually controls reverb send amount
MIDI CC 92	Effect 2 Depth	Usually controls tremolo amount
MIDI CC 93	Effect 3 Depth	Usually controls chorus amount
MIDI CC 94	Effect 4 Depth	Usually controls detune amount
MIDI CC 95	Effect 5 Depth	Usually controls phaser amount

MIDI CC 96	(+1) Data Increment	Usually used to increment data for RPN and NRPN messages.
MIDI CC 97	(-1) Data Decrement	Usually used to decrement data for RPN and NRPN messages.
MIDI CC 98	Non-Registered Parameter Number LSB (NRPN)	For controllers 6, 38, 96, and 97, it selects the NRPN parameter.
MIDI CC 99	Non-Registered Parameter Number MSB (NRPN)	For controllers 6, 38, 96, and 97, it selects the NRPN parameter.
MIDI CC 100	Registered Parameter Number LSB (RPN)	For controllers 6, 38, 96, and 97, it selects the RPN parameter.
MIDI CC 101	Registered Parameter Number MSB (RPN)	For controllers 6, 38, 96, and 97, it selects the RPN parameter.
MIDI CC 102 – 119	Undefined	
MIDI CC 120 to 127 are "Channel Mode Messages."		
MIDI CC 120	All Sound Off	**Mutes all sounding notes.** It does so regardless of release time or sustain. (See MIDI CC 123)
MIDI CC 121	Reset All Controllers	It will reset all controllers to their default.
MIDI CC 122	Local On/Off Switch	Turns internal connection of a MIDI keyboard/workstation, etc. On or Off. If you use a computer, you will most likely want local control off to avoid notes being played twice. Once locally and twice whent the note is sent back from the computer to your keyboard.
MIDI CC 123	All Notes Off	**Mutes all sounding notes.** Release time will still be maintained, and notes held by sustain will not turn off until sustain pedal is depressed.
MIDI CC 124	Omni Mode Off	Sets to "Omni Off" mode.
MIDI CC 125	Omni Mode On	Sets to "Omni On" mode.
MIDI CC 126	Mono Mode	Sets device mode to Monophonic.
MIDI CC 127	Poly Mode	Sets device mode to Polyphonic.

Lesson 3: The Mixer

#1 Lesson Overview

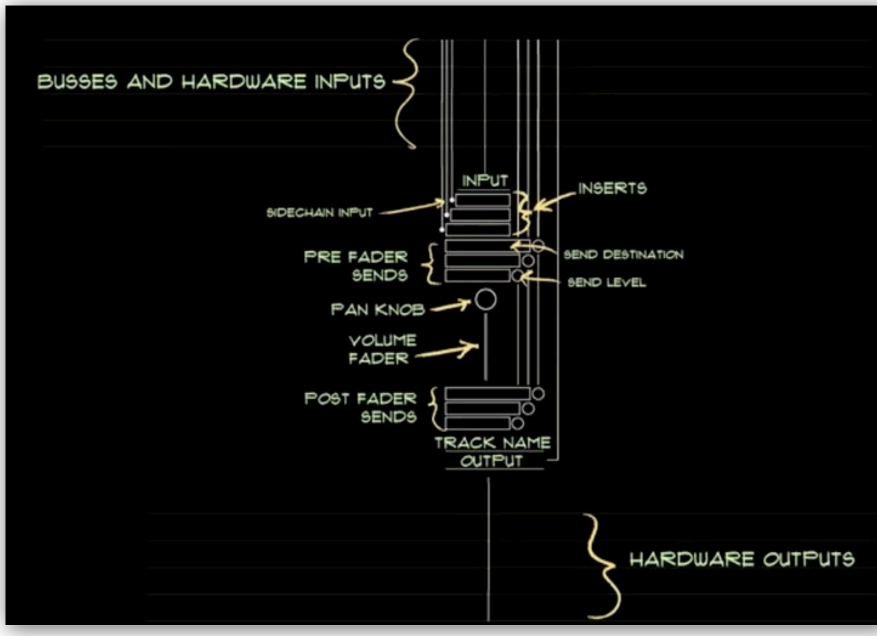
- Mix from the perspective of the listener. What will they hear and focus on when listening for the 1st time? Think about the clear path through the music. One section, highlight melody, another, bass, etc.

#2 The Channel Strip

ANALOG MIXER

- **TRIM Control** (adjusts the gain of the Mic Preamp) / Line In Button / Bal or Unbal / XLR - The is the **Mic Pre Section.**
- **Signal Flow runs from** Top to Bottom input at top and output section at bottom, w/ exceptions.
- **Insert Section - t**he sound goes OUT one side of the cable and comes back IN through another - creates a loop from the mixer to an external device of gear (compressor, EQ, etc). and back to mixer. **Insert Cable:**

- **AUX Send** - separate output for the track. Routes a track to more than one place.
- **EQ Section** (maybe)
- **Pan Knob: "**Reduces" the two levels of either left or right.
- **Fader:** U (Unity) will not amplify or attenuate the signal. Keep here as much as possible.



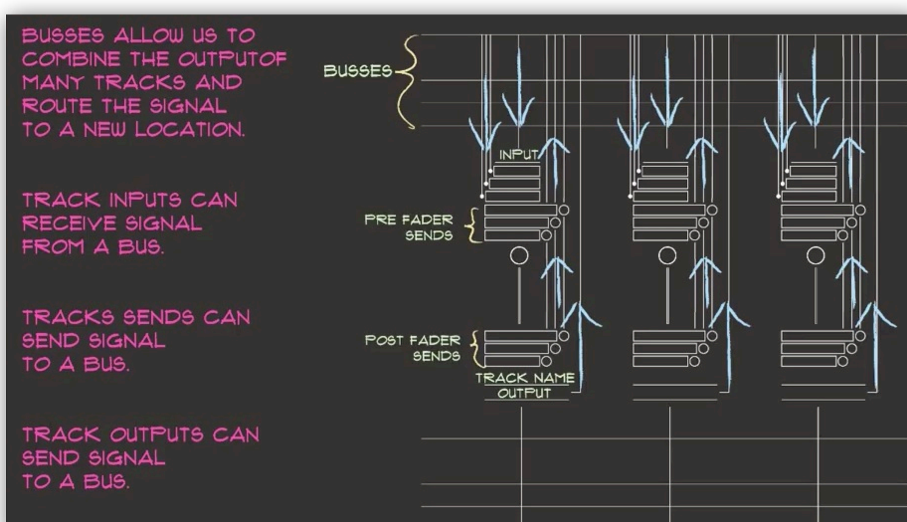
- All tracks are combined and sent to the **Main Master Bus** - Peaks in Yellow, never going into the Red.
- It's a Mono Channel Strip but Output is always Stereo.

DAW - Signal Flow

- **Inputs**
- **Outputs:** Routes to Hardware and Buses.
- **Learn the direction of the Signal Flow.**
- **Inserts:** collection of places where we can add effects - Gates, Compressors, etc.
- **SENDS:** It will always be sent to a **Bus or output.** Can exist in two different places, **pre-fader and post-fader**. Set the **Level** - how much to signal is going to the output?
- Volume and Pan

#3 The Bus Concept

- Anytime you are trying to combine multiple streams of audio (tracks), you'll need a BUS (**aka: Summing Bus**).



Lesson 3, Con't: The Mixer

Some DAWs do this automatically, others require setup and naming. A bus combines signals from several other places on a mixing board.

- Multiple drum tracks going to one location bus is also called a **Sub-Mix.**
- Another signal flow that uses busses (Aux): siphoning off a bit of sound from each one of the tracks and sending it to another effect or monitor mix, this is a **send/return signal flow. (i.e. a Monitor Mix)**
- The hardest problems to troubleshoot are when the busses are used incorrectly.
- I/O Window in the DAW allows creating, assigning and naming Busses.

#4 Effects Categories

- Setting up audio effects in the most efficient way.
- 3 categories of audio effects (also called **DSP** - Digital Signal Processing)
- **1-Dynamic Effects (related to Amplitude)**
- **2-Delay Effects (related to the Propagation principle of sound).** (i.e. creating an Illusion of 3 dimensionality.)
- **3 - Filter Effects (related to Timbre)**
- We setup our signal flow to how we want to process things. i.e. Drums as a single unit (SubMix), using one plugin. One Reverb Send and Return (Aux) that can be used for multiple tracks in different amounts.

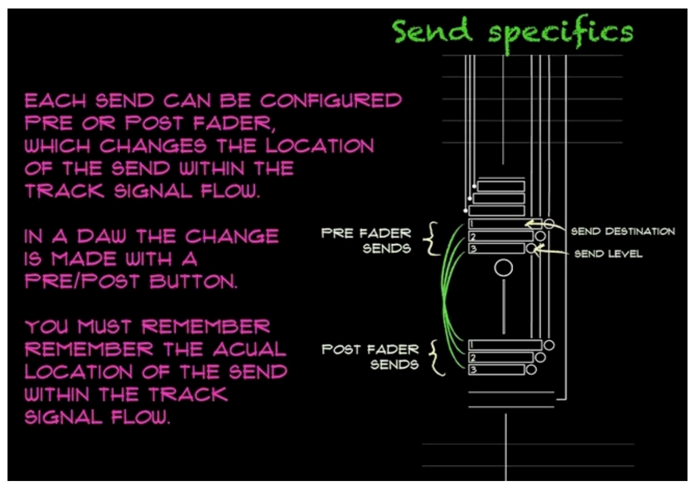
Dynamic Effects (Control Amplitude)
- Compressors
- Limiters
- Expanders
- Noise Gates

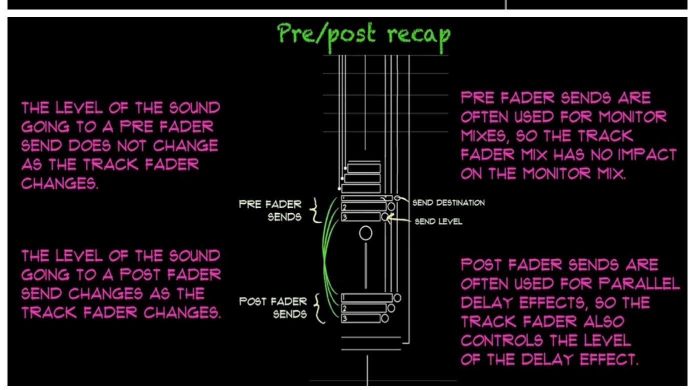
Delay Effects (Control Propogation Qualities)
- Reverbs
- Delays
- Phasers
- Flangers
- Choruses

Filter Effects (Control Timbre)
- High Pass Filter
- Low Pass Filter
- Band Pass Filter
- Parametric EQ
- Graphic EQ

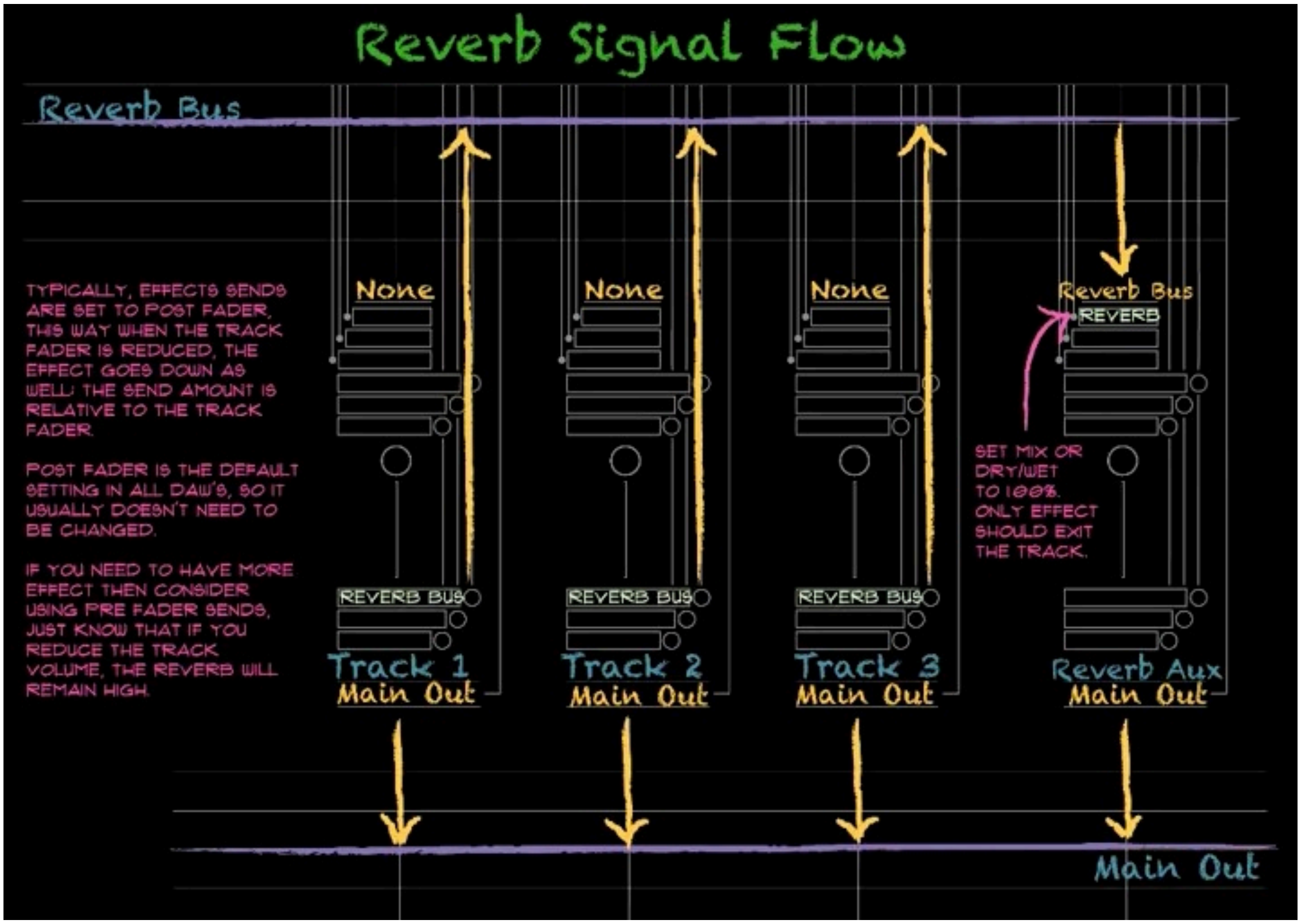
#5 Inserts

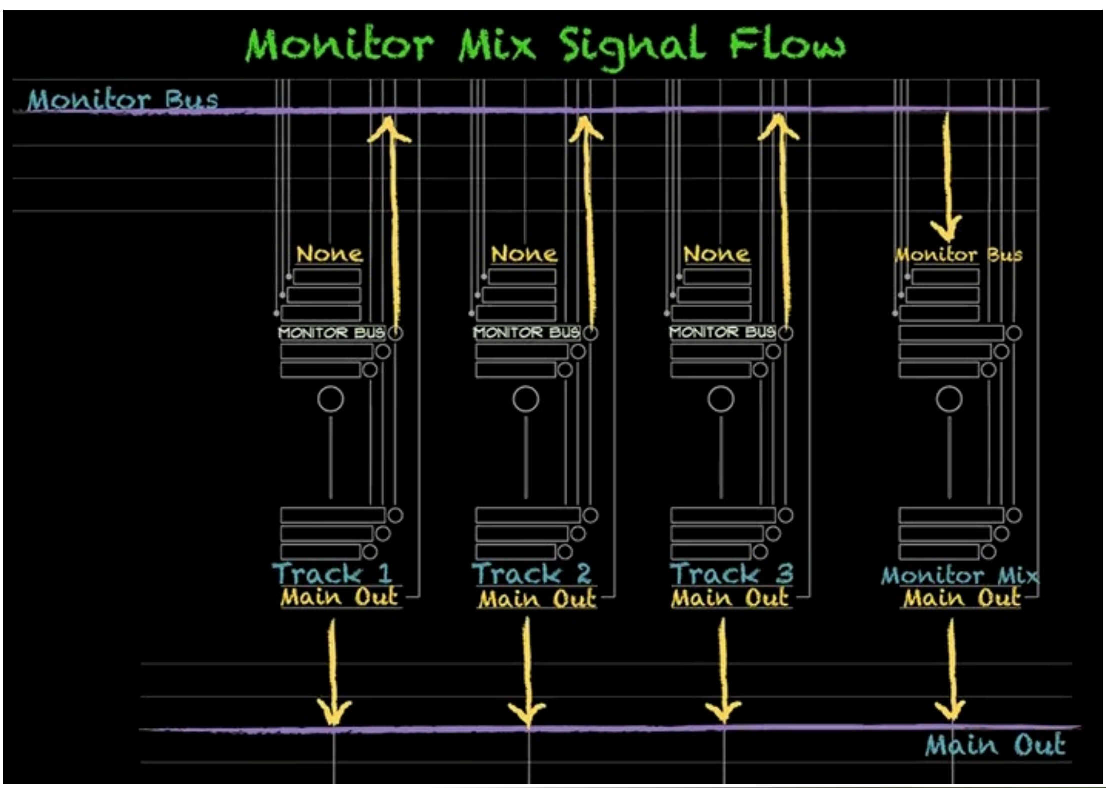
- Easy to do Inserts in a DAW, unlike the cabling in an analog board. You can buy 3rd party effects. **It's common to use dynamic and filter effects as inserts in a track.**
- The Insert Effects process one by one from top to bottom (i.e. serial processing).
- Common to add delay and reverb into an aux track (which is a SEND structure).
- Processing power is ALWAYS a concern, use as few effects as possible, and be aware of resources.
- **Inserts:** add an EQ. SAVE SETTINGS for other projects, becoming modular. Learning to BYPASS (mute) to audition with and without. Learn to Use a Preset and Save a Preset.
- Correct order for the signal flow in a mixing board: **Input trim —> Inserts —> Pre Fader Sends —> Track Fader —> Post Fader sends**
- **It makes a difference which ORDER the inserts are in.** Top to Bottom or Left to Right. The sound enters the top of the channel strip, gets processed by the first plugin, then moves down to the next one and so forth. It can be useful to change their order. Usually use of a modifier key and dragging.





Lesson 3, Con't: The Mixer





Lesson 3, Con't: The Mixer

- Copy plugins between tracks.
- MIDI data cannot use the same plugins.
- LEARN THESE in your DAW:

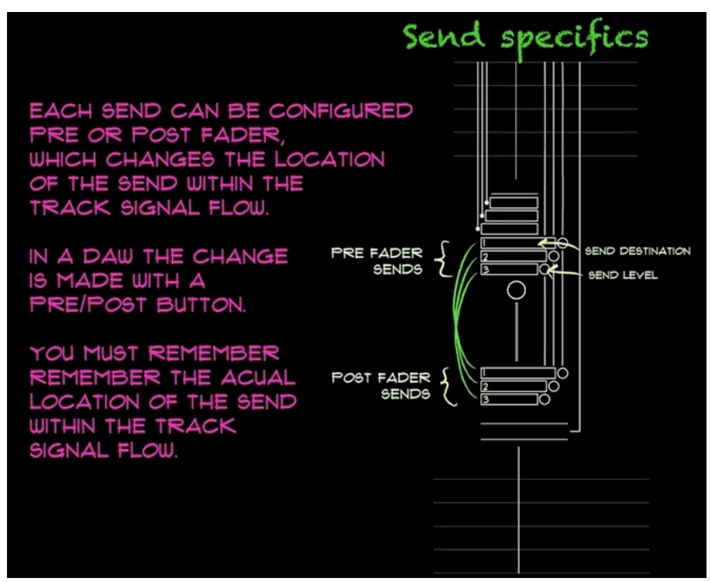
Necessary Insert Skills:
-Adding inserts
-Bypassing inserts
-Choosing presets
-Saving presets
-Changing order of inserts
-Copy / pasting inserts

#6 The Submix (a mix within a mix - **Bus**)

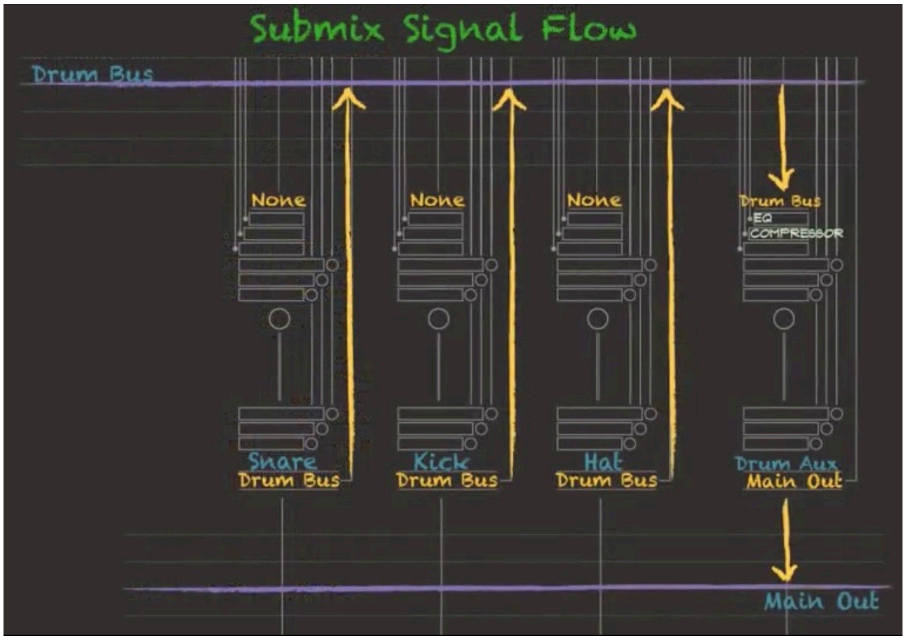
- **Create a single Bus** (name it well!) **Route the outputs** of the individual tracks to the **Drum Submix** Bus. Then, create an stereo **Aux Track** (New Track) whose **INPUT** is that **Drum Submix** Bus. Use a single insert like an EQ or compressor, effecting all the drums as a single unit. A single fader now represents the entire Drum Kit.
- **Creating a New Bus:** Make it Stereo, Route Tracks to New Buss. Create an AUX Track, set input to the Drum Submix Bus. You can then add an EQ insert on that bus, controlling all the drums simultaneously.

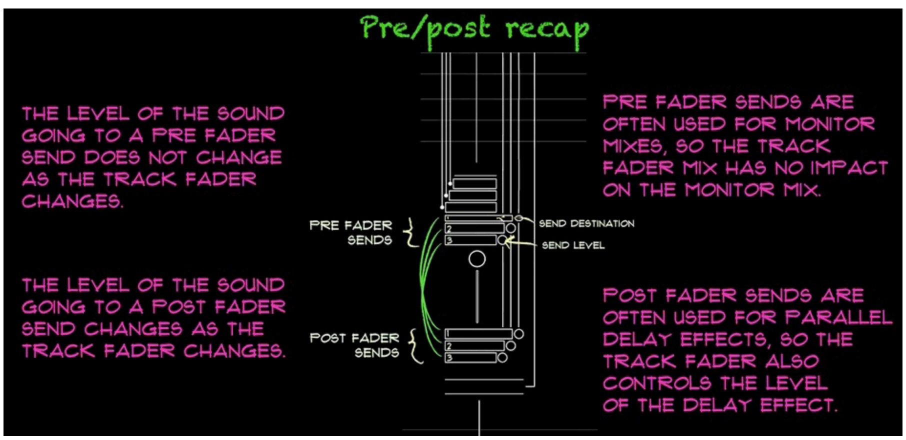
#7 Sends (WATCH AGAIN!)

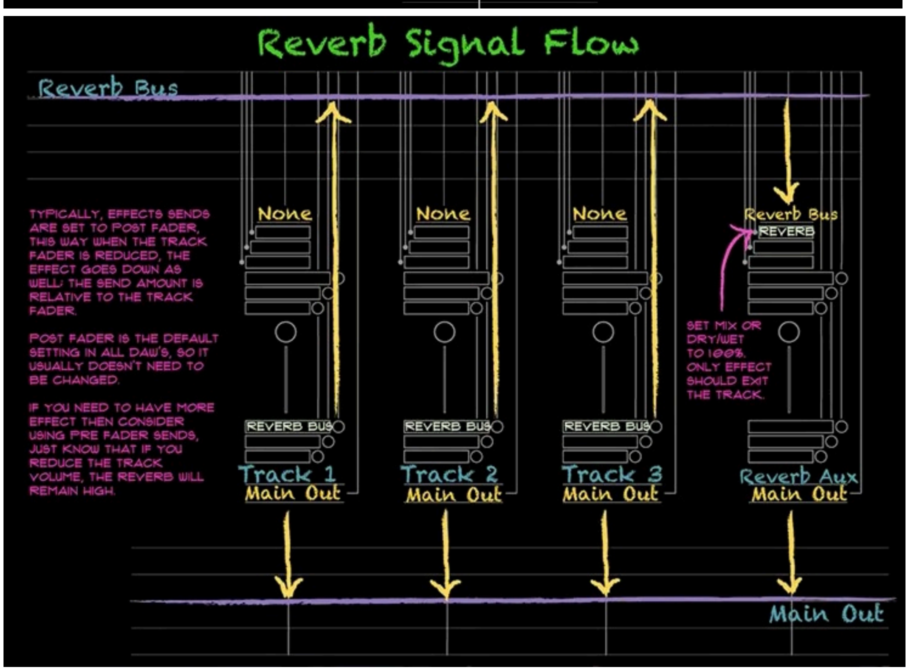
- **AUX SENDS -** Routing the same signal to a different place but with a different amount or pan position.
- Send Section is the most complicated on the mixer.
- Tricky part is they are not where they look like they are—knob might be in one place but the send might be happening in a different location. It can be moved between two different locations: before the fader (PRE - totally **in-dependent** of the track fader) or after the fader (POST - totally **dependent** on the track fader, where changing the level of the fader, changes the amount that's going to that SEND.)
- **Good use of a Track SEND is a Monitor Mix** (PRE Aux Send). Band members can hear only what they want to.
- **House Mix** - faders control the what the audience hears.
- **When setting up Effects on Aux Sends, always use POST Send. If set to PRE, you will still hear the effects after fader is brought down. (I need to do this on the Roland 1680!)**
- An **Aux Track** is also called a **RETURN TRACK.**
- **Setting up a Reverb:** Create an AUX using a single plugin.
- **#1 Create a BUS** in your audio assignments (I/O Labels): **Reverb Bus, Bus 2** (Bus 1 is the Drum Submix.)
- **#2 Create an Aux Track**, in Stereo. **Input is set to Bus 2.**
- **#3** Rename **Aux 2** to **Reverb Aux.**



Lesson 3, Con't: The Mixer







- **#4 Route all track SENDS to Reverb Bus, POST Fader (Dry/Wet).**
- **#5 on the Reverb Aux**, INSERT needs to be set to **Reverb**. Platinum Verb, Stereo. **Set WET to 100%**. Set TYPE of Reverb (#default) to Big Room or something else.
- The reverb on ALL tracks can be adjusted simultaneously now with the Reverb Aux.
- We use **Post Fader Sends** when setting up **Parallel Effects (reverb/delays).**

#8 Automation

- Automation (a 'moving' mix) sounds soother (at a higher resolution) than CC midi data.
- Channel strip has automation: **Read, Write, Latch, Touch and Overwrite** Modes for Automation.
- **Any fader, knob or button can be made to change** automatically over the course of the music.
- **LATCH (most useful):** record enable, whenever you touch any fader or button it will start to record your changes to any knobs. Great for fading a track that is too long. You can then adjust the wave. Turn OFF Rec. Stays at the same level you left it at when you hit stop.
- **Pencil Tool:** draw the fade out.
- **TOUCH:** An Overdub for Automation. As soon as you let go of the fader or button, it will snap to the previous it was at at the point you let go of the fader or stop.

###

Lesson 4: Dynamic Effects (Post Prod)

#1 Lesson Overview

- **Dynamics:** As a Performer - Musical Control of Volume over the course of the song, includes intensity. **Equipment Dynamics:** Has a dynamic range where it operates properly - called the Linear Area - outside of that we'll have issues. Dynamic Range in amplitude is between noise and distortion. Air Measured in dBSPL, threshold of human hearing.

> **0dB SPL(Sound Pressure Level)** is the level of the quietest audible sound, the threshold of hearing.
>
> dB SPL measurements will generally be positive, going up from this level.

- iPhone App for **Sound Pressure Level Meter**. Get a feel for the general level of SPL in your environment. Use it in mixing, put in front of your monitors.

- Our **human EQ** changes with amplitude. When things get **Quiet**, we notice the midrange more and more, highs and lows drop off; **Louder**, it flattens out a bit. Above that, PAINFUL!
- It's best to mix at the SAME level on your monitors every time.

#2 Noise

ANALOG MIXER

- **There is Always SOME kind of noise.** In recording we try to eliminate two kinds of noise: Acoustic (Room).

> **Reducing Acoustic Noise:**
> 1. Listen carefully to the "silence" or "room tone" to identify noise.
> 2. Move away from noisy sources like fans and windows.
> 3. Create an isolated space for recording.
> 4. Turn off noisy sources like A/C, fans, heating, TV, and appliances.

- **Self-Noise:** electrical noise that gear produces.

> **Reducing Electrical Noise:**
> 1. Use fewer pieces of gear.
> 2. Use shorter cables.
> 3. Use balanced cables.
> 4. Turn off appliances and dimmers.
> 5. Use high quality gear.

> **Avoid Unnecessary Gain:**
> 1. Boost level electrically as little as possible.
> 2. Try to move mic closer to the sourse instead of increasing gain.
> 3. Choose a directional microphone to isolate the source from a noisy environment.

- The cleaner the recording the more options you have. **Don't try and fix it in the mix.**

#3 Distortion

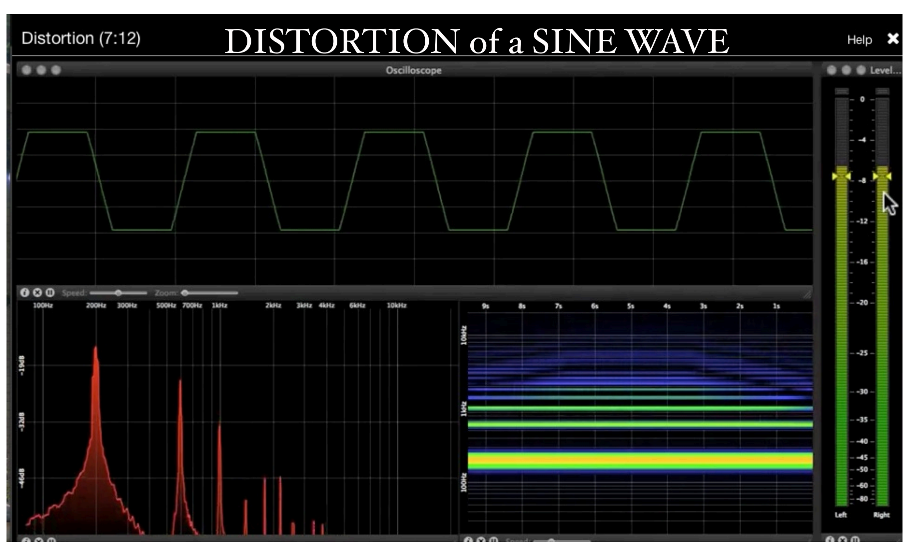
- **Non Linear range of equipment** - as you get really loud, it causes distortion.
- **Distortion turns volume variations into timbre variations.** Output gets brighter and brighter.
- **"Drive" is a Gain Control before a distortion stage.**

> **Avoid distortion at every point in your signal flow:**
> Careful mic placement
> Use of a pop filter.
> Careful gain staging: stay in the green at every point in the signal flow.

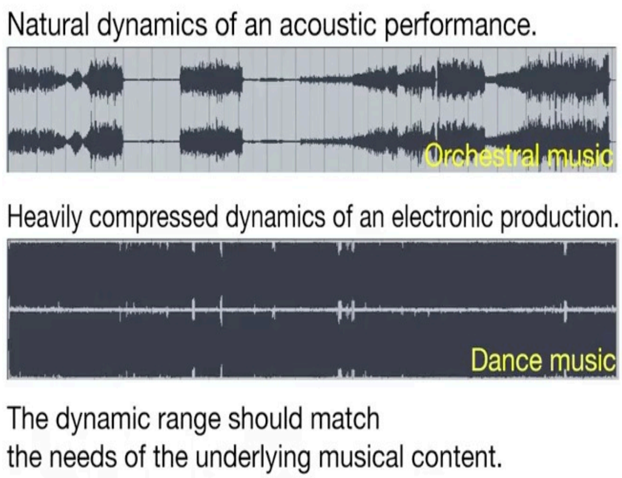
- Avoid distortion all the way through your signal flow. NOTHING should touch the red - avoid at ALL costs.
- **SINE WAVE** - energy at a single frequency.

#4 Dynamic Range Manipulation

- Adjusting the energy Heightened highs and lows for an emotional journey.



Lesson 4, Con't: Dynamic Effects



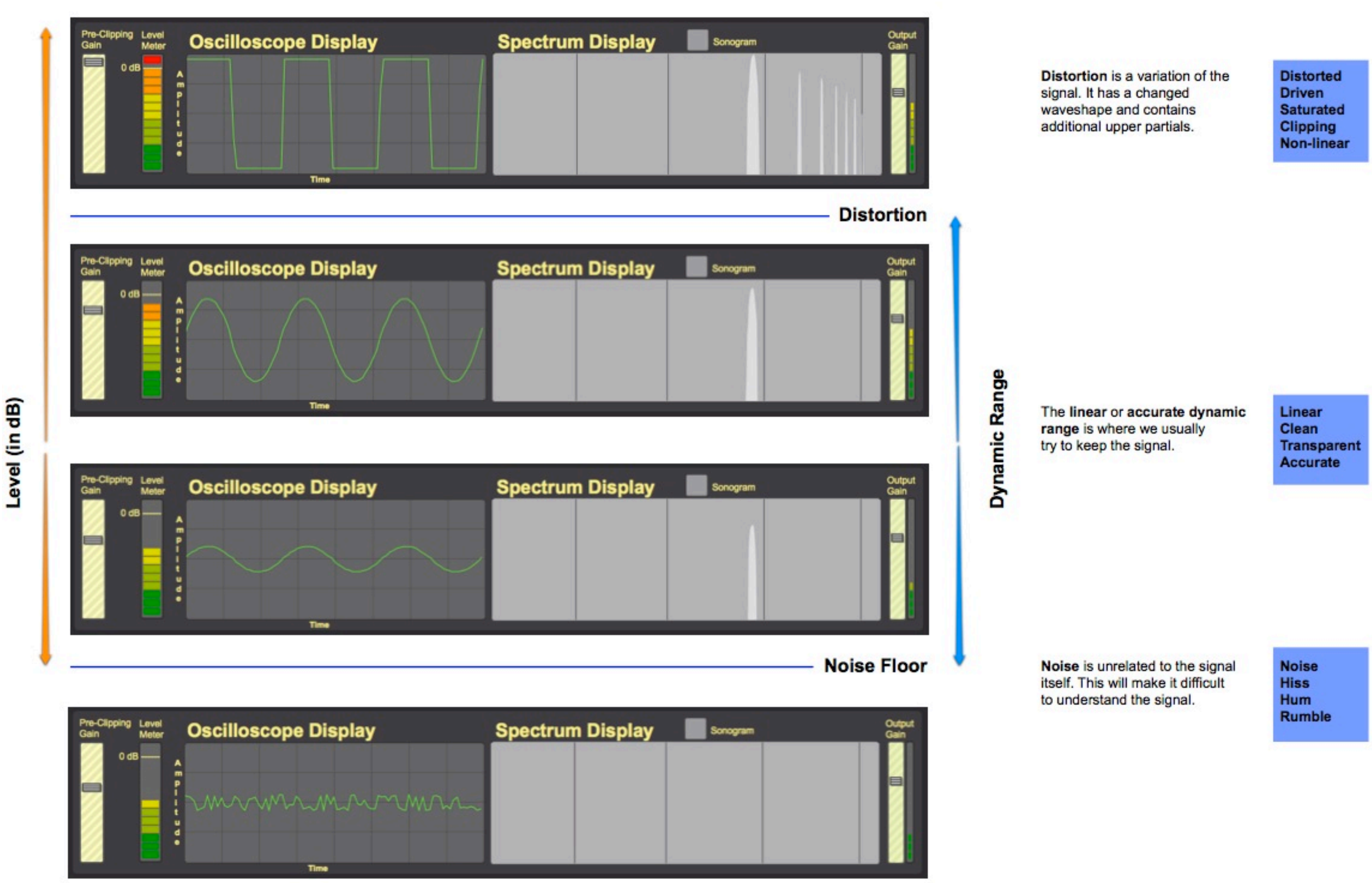
DYNAMICS CONTROL:

- **Macro Scale** - zooming out and look at the relative levels between the sections.
- **Micro Scale -** adjusting volume of verse or chorus manually with volume automation:

- Manual dynamic range control with volume automation on vocal to bring up and down.

Dynamic Range

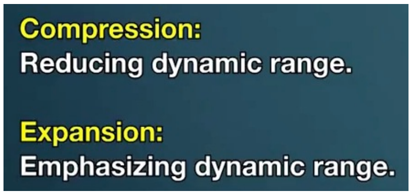
Sound exists along this spectrum, and it impacts everything we do with audio.



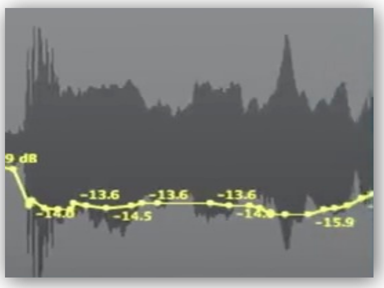
Lesson 4, Con't: Dynamic Effects

- **Micro Dynamics** also control the **TRANSIENT (moment when the amplitude changes a lot in a short amount of time.)** Clap or drum hit. We are controlling the punchiness.
- Use dynamics to create a focus for the listener at any point in time.

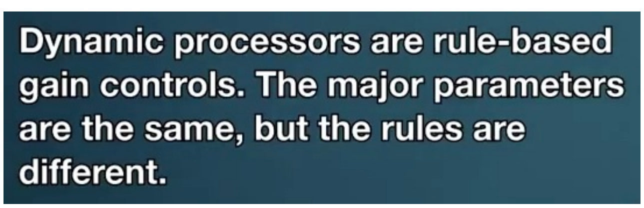
#5 Dynamic Processors Overview



- **Adjusting level of vocal:** chop up the vocal, changing word by word. **Riding a fader (algorhythmic):** playback and move the fader opposite of the performance. You can use a gain or trim plugin, using the main fader for overall volume. We are doing what a compressor does.



- **Use a Gain or Trim plugin** - then use main fader for mixing the vocal.

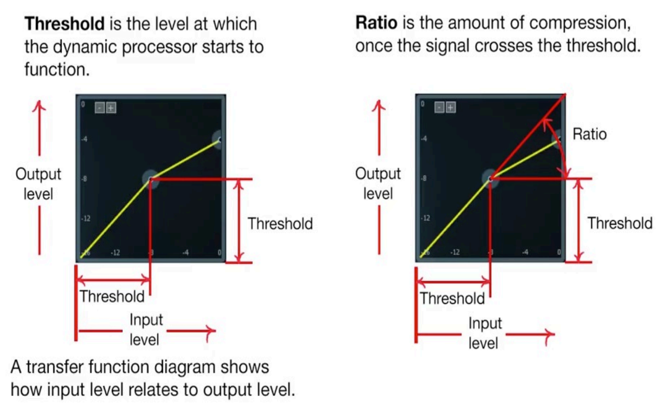


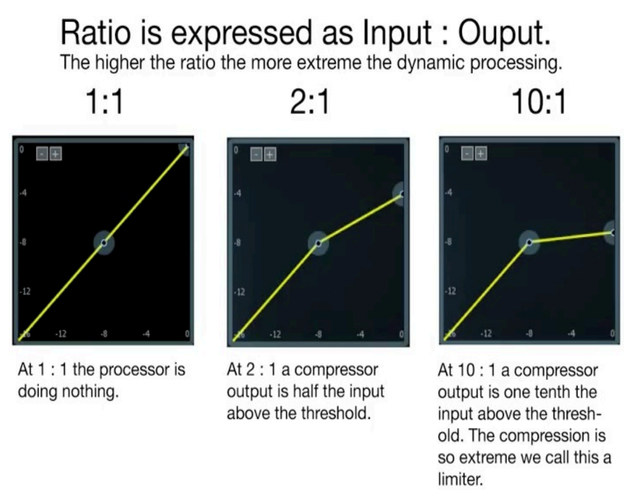
- All these dynamic processors are just changing the volume. These are Non Linear Devices - they react differently at different amplitudes.

#6 Dynamic Processor Parameters

EXPANDERS, GATES, COMPRESSORS, LIMITERS

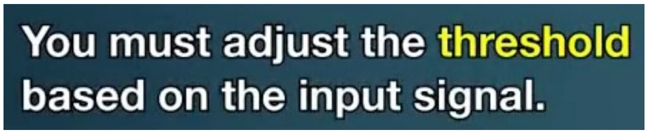
- **#1 Side Chain or Key Section -** analyze the input signal and calculate the envelope of the audio (average level over time). It's roll is to analyze the signal and calculate the envelope of the audio. The envelope is the signal that represents the average level in any point in time. Uses Root Means Squared (RMS) - Average Level. Take every signal and multiply by itself - squares the signal, gives a positive version of the signal, then takes an average.
- **#2 Volume Fader.**
- **SIDE CHAIN** -





- **Above 10:1 Ratio is considered a LIMITER.**
- How fast the volume fader moves is based on **Attack** (how fast the volume fader comes DOWN when the sound goes ABOVE the threshold). **Release** (how fast the fader comes back up after the sound has gone BELOW the threshold. It is expressed in milliseconds, the lower they are the faster the volume fader will move. They function at the beginning of the sound, the transient. The can be set differently.

- All these controls are interrelated. Understand all of their functionality.

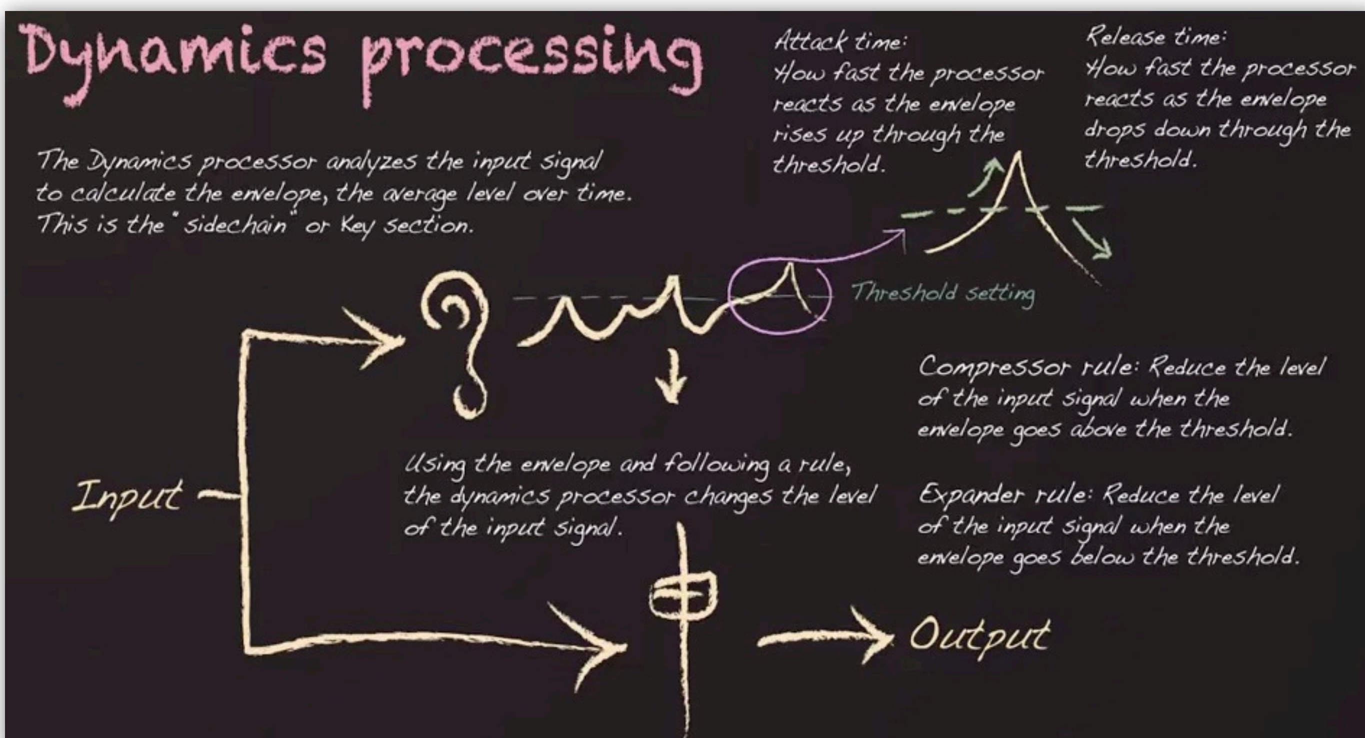


OVERVIEW:

- **THRESHOLD** - the point at which the dynamic effect **STARTS FUNCTIONING**.
- **RATIO** - **how much it functions** once it crosses the threshold. **Input : Output.**
- **ATTACK** - How **fast the volume will change at the beginning of the sound** when it crosses the threshold.
- **RELEASE** - How fast the fader moves when the signal crosses BACK through the threshold.
- **KNEE** - as we approach the threshold it will soften the effect.
- **LOOK AHEAD** - a delay imposed on the signal but not on the envelope.

#7 Gate

- (i.e. Noise Gate) To remove the hiss and noise between musical moments.
- Gate Full Featured Dynamics Plugin: Compression, Threshold, Gain Reduction (GR).
- **Transfer Function** (below) - Input Horizontally, Output Vertically. The lines shows us what the output level will be for every input level.
- 
- **Guitar Cleanup:** Bring Gate all the way up, play track, **Adjust Gate while playing**. Too high and gate will start chattering and miss important notes. Adjust the Threshold as it is playing to adjust it perfectly.
- Impact of **Attack** (Beginning how fast the gate opens up and **Release** (Ending - how fast it closes down).
- Increasing the attack made the gate move slower.



Lesson 4, Con't: Dynamic Effects

- With Release too high, the gate never really brings the level down fast enough. Keep these pretty fast, but based on the sound itself.
- Get into the FINER POINTS of the gate in your DAW. **Know what everything DOES.**

#8 Downward Compressor

- Referring in any way to a "compressor" means the **downward compressor**.
- Goal reducing the level of the loud stuff. We set a threshold, reducing the level above the threshold.
- **Compressors are trying to control the transient**. All the controls are interrelated.
- Choosing a different compressor can have a major impact on the sound it.
- **Fundamentals of Compression:**
- **Limiting Mode:** Ratio of 30:1 Hear what an effect is doing at an extreme, then back off for more musical sound.
- Lower the **Threshold** to adjust the LOUD parts. We have a quieter overall sound.
- To bring up the perceived loudness of a sound: **Increase the Output Gain.**
- The hard part of compression is handling the **Attack and Release**. They can seem backwards at times. Higher Attack lets transients through.
- Often we keep attack time quite low so the transient is not allowed through.
- You have to CRAFT the Attack, Release, Ratio and Threshold controls everytime you are working with a Compressor.
- **Compressing a Bass Line:** Uneven Waveforms. 1st, set the Threshold where you are crossing it pretty regularly. Used a HEAVY Compression of 4:1 (lower settings for a guitar); then increased Gain. Much more consistent performance. Used a ow Attack and Release, compressor will react very quickly allowing no transients. Use a little for a more natural feel.
- Get comfortable with the **Compressor and Attack Settings**. You may want some of the dynamics and transients in there.
- Put in a Drum Hit and see if you can manipulate the sound and remove transients.

#9 Limiter

- Is an **Extreme Downward Compressor**, with a ratio set VERY high, above 10:1 with a low attack time.
- Used so that the Level NEVER gets above a certain point, a protective roll, but doesn't sound great. **(Brick Wall Limiter**)
- Evolved now in a more contemporary use called a "Loudness Maximizer." Making things as Loud as possible without increasing the volume or amplitude.
- Pushing compressors and limiters (and even gates) too hard Increases the chance of distortion. You don't want to add upper harmonics. Most commonly a problem with low frequency notes. To remove this, you have to manipulate the attack and release times. The tradeoff is a little of the transient will come through.
- **As you lower the threshold, a Limiter automatically brings up the output Gain automatically.**
- Increasing the **Speed Control in the Limiter** can reduce the distortion.
- We have to train ourselves to hear compression before we can actually use it in a musical manor.
- **Experiment:** Load drum groove by real drummer. Find out how you can manipulate it compressing it heavily. Increase the ratio quite high, set Attack quickly, Release in ms range. Then start reducing threshold. The bring up Gain a bit.
- Describe the change you hear differently.
- Use Bypass to compare.
- What if we increase attack? Transients come through more. Increase or Decrease Release?
- Try and recognize the "sound of compression"when you listen to records.

###

LOGIC X DYNAMIC EFFECT PROCESSORS

EXPANDER

ENVELOPER

DE-ESSER

DUCKER

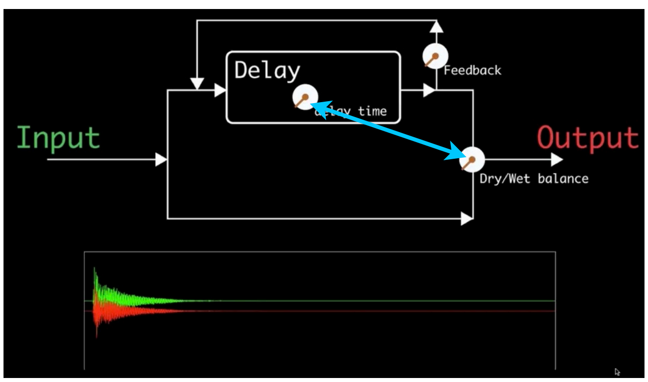
Lesson 5: Filter and Delay Effects

#1 Lesson Overview

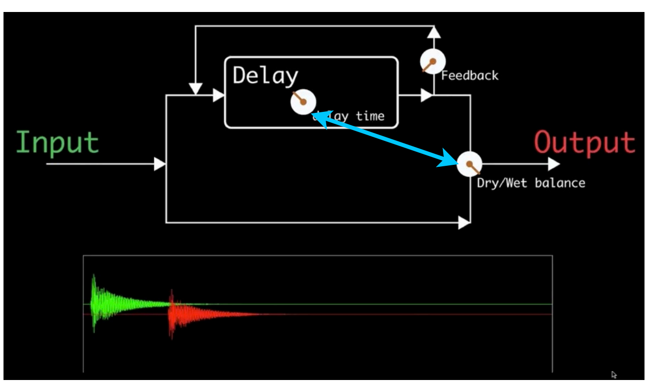
- Filter and Delays Effects are in same category: all are based on single sample delays which form all the different kinds of EQs.
- We categorize the delay effects by the LENGTH of the delay. We have really short delays where we actually move the delay around, these are Modulated Delay Effects and form Chorus, Phasers and Flangers, and create WIDTH in the mix.
- Long delays are like an echo. Used often to sync to the tempo.
- Reverb are 100s and 100s of very short delays that represent all the surfaces in a room.

#2 The Delay Concept

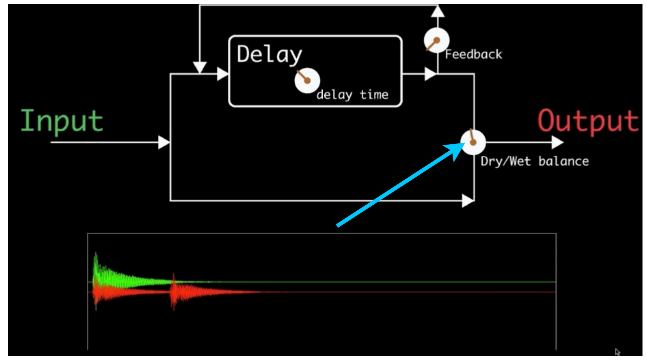
- Crafted around one delay building block and includes **Delay Time**, **Dry** (undelayed signal) / **We**t (delayed signal), and **Feedback** - How much the Wet Signal is routed back to itself.
- **Green: Input, Red: Output, Wet** - They are Exactly the same. We are only hearing the DRY side:



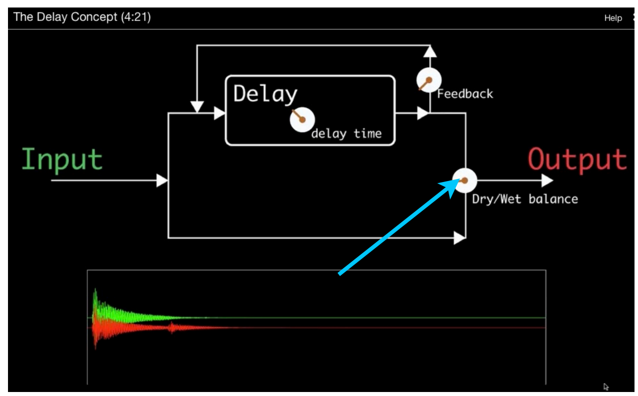
- Moved all the way to WET and increase Delay Time:



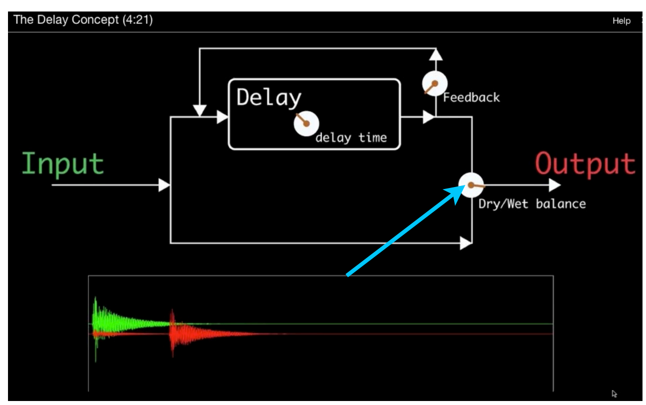
- We are only hearing the Wet Path, none of the Dry path. We can mix them together in any portion.
- In the middle, we will hear both the dry and the wet side at the output of the delay block.



- Typically, we will put the dry/wet to the left so that the delayed signal is quieter than the input dry:



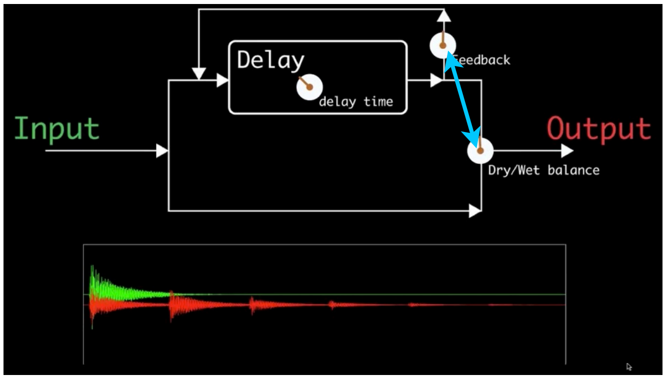
- If we turned the **Dry/Wet balance very high**, the wet delayed signal is louder than the input Dry signal:



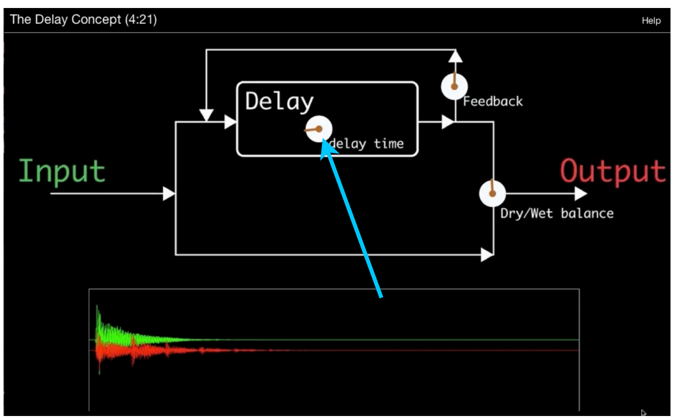
The **FEEDBACK control** routes some of the output of the delay block back to the input of the delay block. It is only a **Gain Stage**. It's increasing the level of the signal going back to the input of

Lesson 5, Con't: Filter and Delay Effects

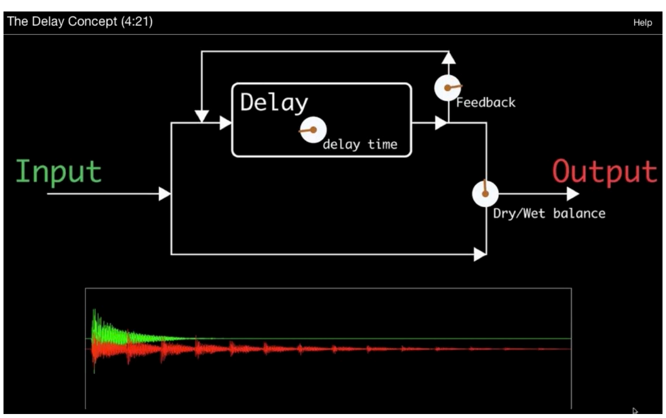
the delay. If you put the **Feedback too high**, the **sound will continue on forever**. **Low, the delays will fade away quickly**:



- **Lowering the Delay** time down, the **repeats will be closer together**:



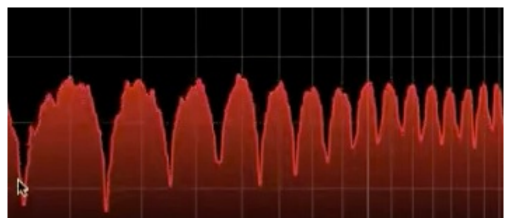
- If we **increase Feedback, the delays will continue for a very long time**:



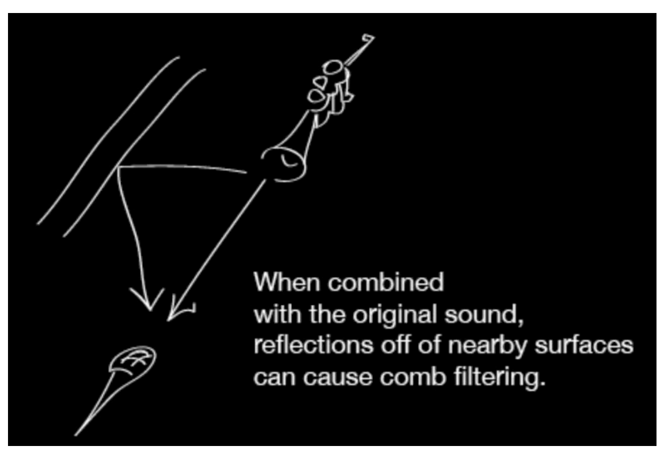
This basic functionality will translate to ALL the other delay devices that are built from this simple delay block: chorus, phasers, flangers, short and long delays, reverbs and filter devices.

#3 The Delay Spectrum

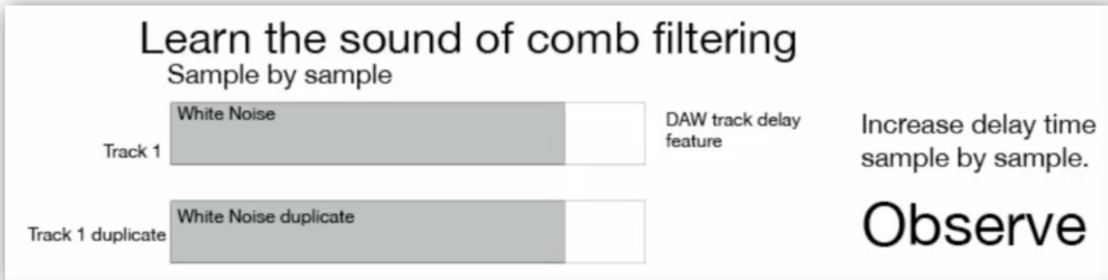
- We'll start with white noise, which is equal amplitude (energy) across the frequency spectrum (range) and we'll add slight delays mixed back with itself. To see how the spectrum changes based on that.
- **Comb Filtering:** a series of related notches which are harmonically related. It's always giving us trouble when we are recording. You have to be careful of other surfaces around you, because the sound from that object or source will bounce off of all these other objects and hit that microphone.
- **Phase Cancellation in Comb Filtering** - It's called a comb filter because of the series of deep knotches:

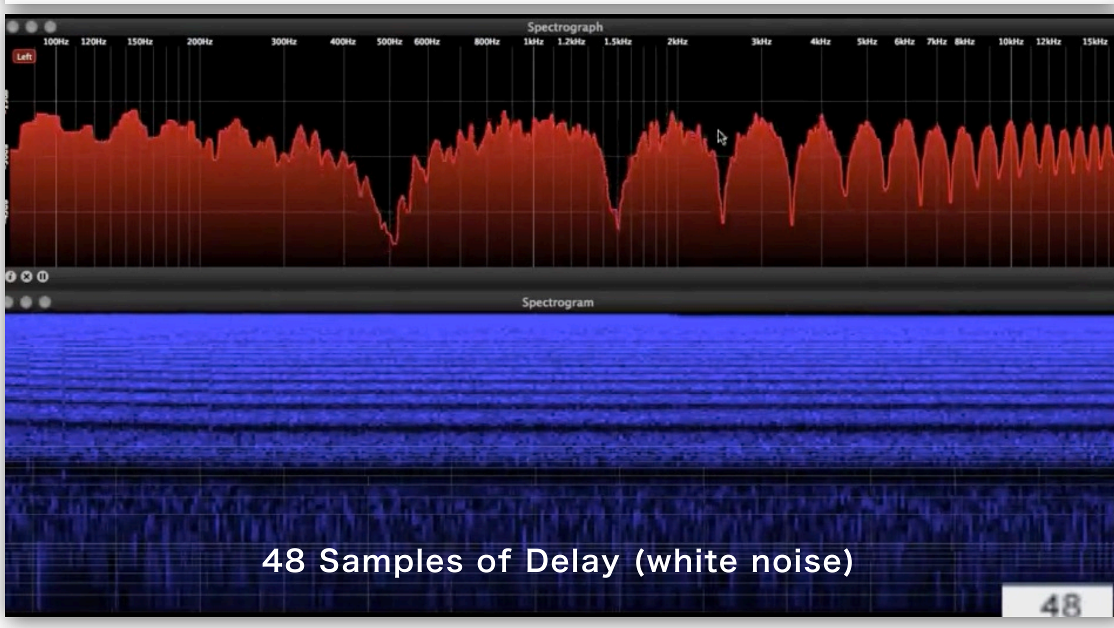


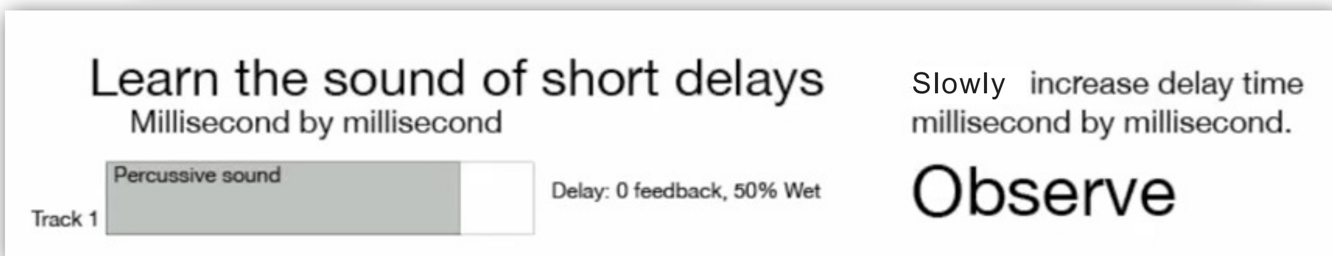
- It's important be be able to HEAR the sound of comb filtering because it's all around you. -
- This demonstration shows the impact of combining a sound with a delayed copy of itself, and this happens when you are recording. You have to be careful of nearby FLAT surfaces when you are recording.

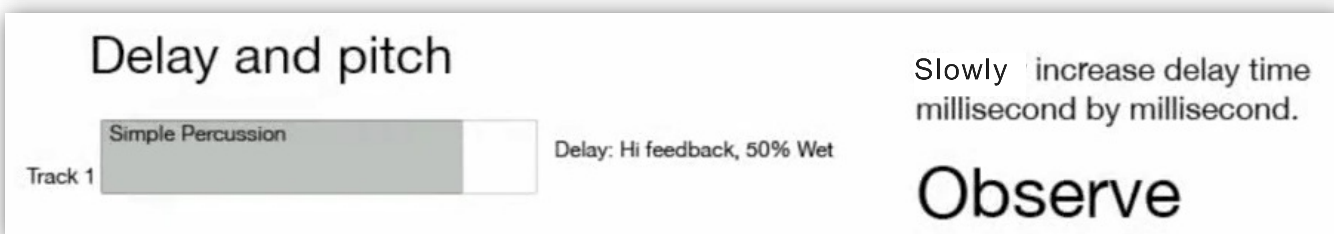


Lesson 5, Con't: Filter and Delay Effects







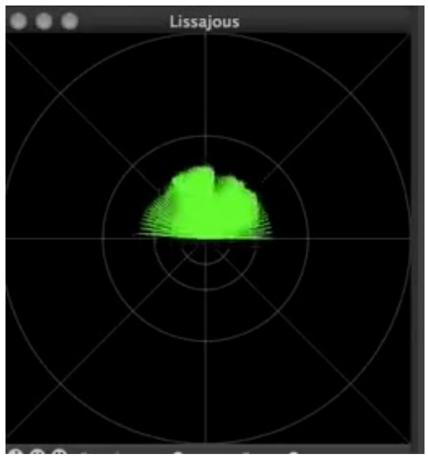


- **Snare Hit:** Combining a snare with a delayed copy of itself - Starts sound Buzzy and a high pitch is heard. 18 mil sec and it's sound like a flange. @ 36 milliseconds, it sounds like 2 different hits.
- **Clicky Sound with feedback all the way up.** Very short delay times actually create a pitch. 100% Wet, Feedback is all the way down, but start increasing it. tone goes DOWN with each milsec increase. Eventually we hear it as separate attacks, a separate rhythm.

Lesson 5, Con't: Filter and Delay Effects

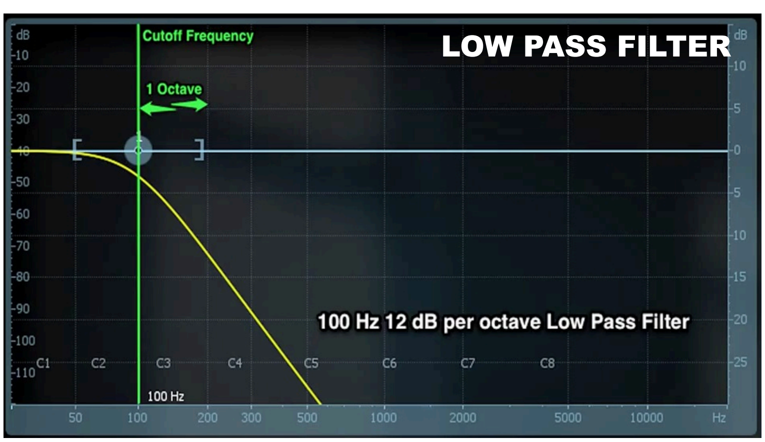
#4 Modulated Short Delays

- **Modulated:** Putting the notches in motions.
- **The effects that make up the Short Delays are choruses, phasers and flangers.**
- **FLANGER is a comb filter (a slight delay) in motion** by a low frequency oscillator. Often used differently in each speaker.
- **PHASER: Deep notches in motion across the spectrum**, and put in motion very often differently, in left and right. It's not a comb filter but has much the same sound. You can change the number of notches there are and modify how they are related to each other—with more control over where you want to emphasize.
- **Chorus: Multiple detuned copies.** Seems like there are multiple performers doing the same part. Changing the delay time can have a great impact on the pitch of the sound. Makes copies of the same signal and then slightly shifts the pitch on all of them, and changes the pitch differently in Left and Right. **Effective at making something sound wide and out to the side.**
- Adding Chorus to a Guitar: Using a **Lissajous Meter -** shows the stereo width of an element.

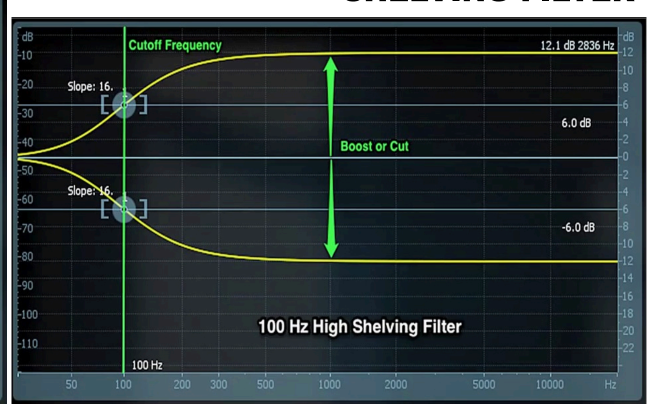


#5 Filters Overview

- **EQ** has a series of bands and each band is an individual filter and can be changed through filter TYPES.
- Low, Lo and Band Filters. **Tell is what the filter letting through** — Not what it's filtering out.
- **LOW:** Will allow the low frequencies to pass through, attenuate the highs.
- **HIGH:** Will allow the high frequencies to pass through attenuate the lows.
- **BAND:** Will allow a range of frequencies to pass through, attenuates the low and high end.
- Use the **terminology** correctly… it's **not a HIGH CUT** Filter, it's a **LOW PASS**.
- **Low Pass Filter (Removal Filter):** Establish a Cut Off Frequency. They are very drastic.
- **SHELVING FILTER: Reduction (or boost) Filter** - it will reduce the frequencies above the cutoff frequency and keep it at a set point. Used all the time in a MIX context.
- **PARAMETRIC EQ** - Bell shape notch in the Mid Range.
- All of these filters work in conjunction to create a **modern parametric EQ** plugin.
- By disabling unused filters, you are saving CPU resources.
- **Hi Pass Filter:** Removes rumble and excess noise.
- We tend to use **Shelving Filters** on the low and high ends instead of high and low pass filters in a mixing context.

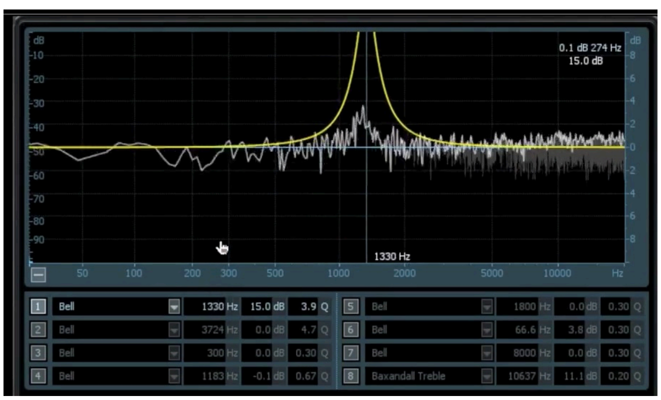


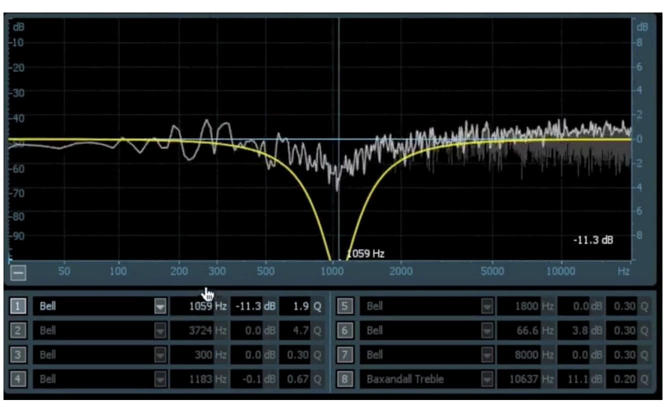
SHELVING FILTER



Lesson 5, Con't: Filter and Delay Effects

- **LOW SHELVING FILTER**. In a Shelving filter, If we boost or cut, it will not progressively reduce, the further you get away from the cutoff frequency. Instead it will reduce to a point, then stay that level, continuing on. There is a parameter for setting for the AMOUNT we are boosting and cutting. Cutting is generally more useful in than boosting can be more drastic, whereas with boosting you should be more gentle. Some give the ability to change the slope of the transition.
- HIGH SHELF, same as LOW only on high end. This is a TREBLE control on most consumer radios. MAX: Boosting 6 dBs, Cut up to around 12 dB.
- **BELL SHAPE or Parametric EQ:** Use carefully in the middle range. Q - Bandwidth Control - Width of the Boost or Cut.
- Train your ears with the connection of numbers/ sounds.
- Use white noise and use a Bell Filter, fairly high (then low) boost / narrow boost and sweep across the spectrum, watching the number:



- 

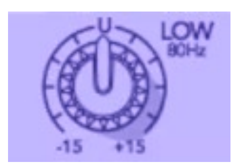
#6 Mixing EQ

- How to use in a channel strip.
- Most important is a High Pass Filter: takes out most of the noise underneath the fundamental frequency which is noise/ rumble.
- Hi Shelving Filter and one or more mid-range parametric filter: Used to control the brightness guiding the listeners focus.
- Mid-Range Parametric EQ are great for removing unwanted
- Low Shelving is good for BASS. Hi pass cutting the very LOWS, Low Shelving Filter Boosting a big at 100 hz.
- Mid-Range Parametric EQ: A boost can sound very unnatural, but a cut can be very nice. A common use is removing unwanted resonances, removing 1 frequency that is really loud (i.e.: clave).
- **We'll configure the same way a Large Format Mixing Board is configured:**

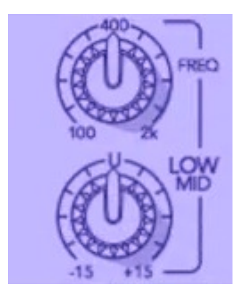
SOURCE: Loudon Stearn's Lecture.

- **Hi Pass: Remove Rumble:**

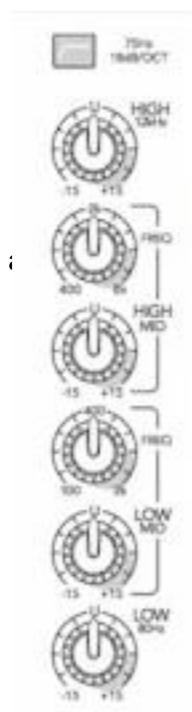
- **Low Shelving Filter:**



- **Low-Mid Range EQ (Bell Shaped**
Sweepable:

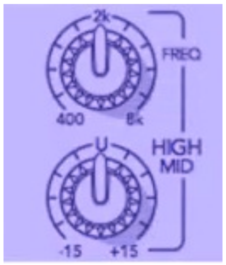


- Max of 6 dB, but cut quite a bit.

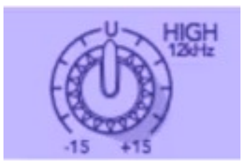


Lesson 5, Con't: Filter and Delay Effects

- **High Mid Range**



- **High Shelving Filter**



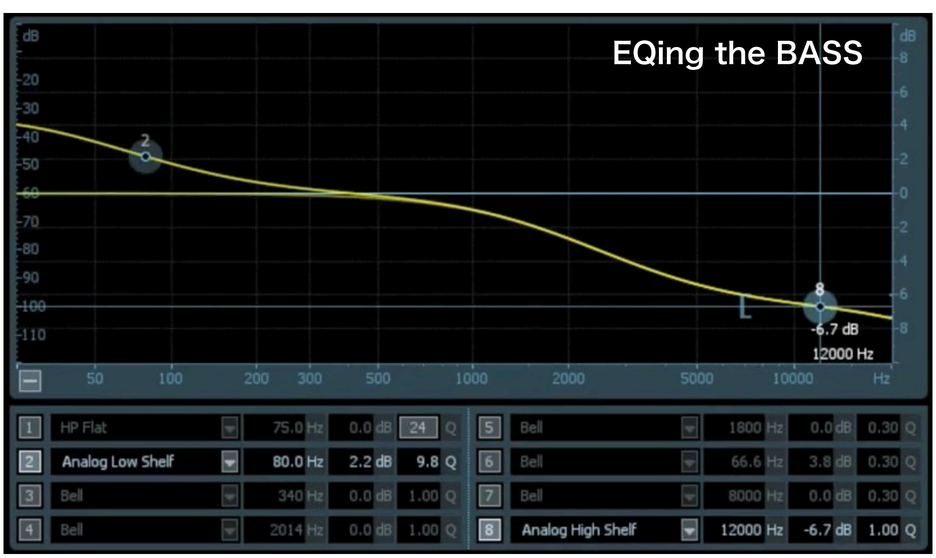
- Use this at the starting point and saving it as the default Preset.
- **EQing the BASS** - Low Shelf - increase low by a few dBs (2.2). Reduce the High Shelf (-6.7)
- EQing the Keyboard and Guitar - its the interaction between tracks that really matters.
- It's important not to think of instruments in isolation. Sometime you need to cut frequencies of one instrument to let the another one come through.

#7 Medium Delay: SlapBack (real space emulator)

- This is a Static Delay - a short delay with no feedback, giving a sense that it's in the room.
- Sounds great on guitars and vocals and is better than reverb to get more width. Often reverb can feel the mix with a noise, slap back delay works well.
- Too much wet and it starts to sound artificial.

#8 Long Delay

- Increasing delay times MORE: hear a very distinct echo.
- Using in time with music: "tempo set." Delay defines the underlying music content.
- We don't have to have the delay in the same place as the signal... can be off to the side; ping-pong.



- They can tend to get in the way of harmony.
- With Long Delays, it helps to make the copies of the delays sound different than the original dry: use a filter (cutting high end or and more interesting.
- Right sync to 1/4 notes, Left sync to 1/8 notes.

#9 Reverb

- Last Delay Plugin - Use one reverb on the entire mix -

creates the same sense of space for all the instruments - in a single space in a single location. **Or** create a more exotic collage - putting things in their own location.

- Common to use one reverb for the vocal.
- **Dry/Wet Control** - Reverb on an Aux should be 100% wet. On an individual track, dial it down.
- **Algorithmic Reverb** (GoldVerb) like synthesizers - a mathematic representation. More opportunities to manipulate the sound of the space.

Lesson 5, Con't: Filter and Delay Effects

- They divide the room into two separate components:
- **EARLY Reflections** - a set of short delays (sounds like slapback modules at different delay times). Pre Delay, Room Size, etc. Keep Reverb under 2 secs.
- **Diffuse Reflections**
- **Convulsion Reverb** (Space Designer): like a sampler, recorded the sound of a **real space**. Limited in the type of manipulations you can do to it.
- There's a large number of audio file choices representing REAL spaces. SOLO just the reverb to audition.
- Try things out. Be subtle, bring it up until you barely hear it, then bring it down again...

#10 Width in Mixing

Central Column of the MIx:

- **Kick**
- **Bass**
- **Snare Drum**
- **Vocals**
- **High Hat**

- Keep out of the way of the Central Column.

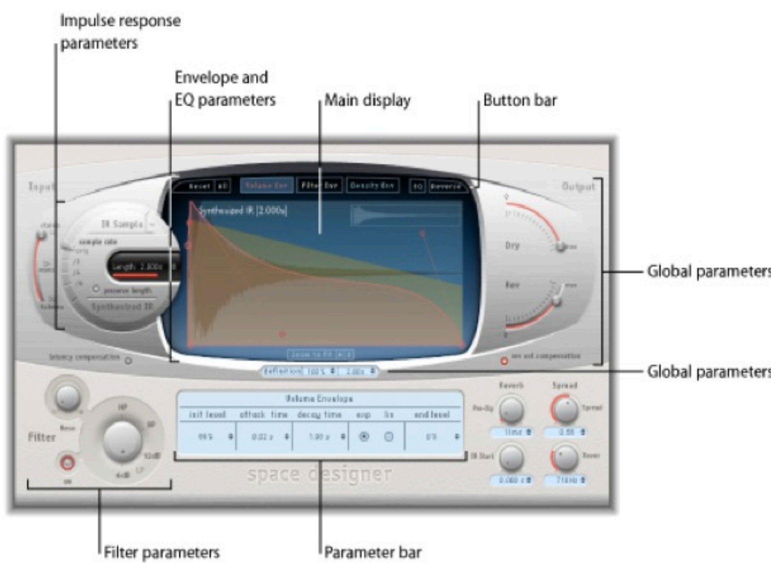
Frequency Masking

Frequency masking is where the frequencies of two (or more) instruments are battling it out for space - i.e.... they **share the same frequencies** and is probably one of the most common problems encountered when mixing.

Basically, a sound may have (in addition to its root sound) other harmonic sounds that contribute to its overall timbre. If two sounds (timbres) share similar frequencies you could easily find yourself in the position where some of these harmonics are being masked in the mix; meaning that the instruments sound different than they do in isolation. **(Source Below)**

Getting to Know the Space Designer Interface

The Space Designer interface consists of the following main sections:



- *Impulse response parameters:* Used to load, save, or manipulate (recorded or synthesized) impulse response files. The chosen IR file determines what Space Designer will use to convolve with your audio signal. See Working with Space Designer's Impulse Response Parameters
- *Envelope and EQ parameters:* Use the view buttons in the button bar to switch the main display and parameter bar between envelope and EQ views. Use the main display to edit the displayed parameters graphically, and use the parameter bar to edit them numerically. See Working with Space Designer's Envelope and EQ Parameters.
- *Filter parameters:* Used to modify the timbre of the Space Designer reverb. You can choose from several filter modes, adjust resonance, and also adjust the filter envelope dynamically over time. See Working with Space Designer's Filter
- *Global parameters:* After your IR is loaded, these parameters determine how Space Designer operates on the overall signal and IR. Included are input and output parameters, delay and volume compensation, predelay, and so on. See Working with Space Designer's Global Parameters

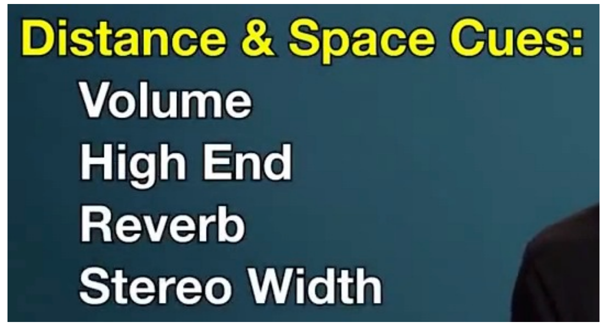
SOURCE: Apple's Logic Pro Manual

Lesson 5, Con't: Filter and Delay Effects

- **SOURCE: http://www.thewhippinpost.co.uk/mixing-music/frequency-masking.htm**
- Where there is a conflict, we can make the Secondary elements WIDE. (Stereo Width) Opening space for the Vocal.
- If you put something off to the left, put something off to the right to balance.

#11 Space in Mixing

- A mix is an illusions, trying to make a sense of a 3D environment.
- Thinking in dimensions.
- Hearing:usually we hear something in one ear and the other ear hears a delayed copy.
- Common listening environment - HeadPhones.
- DEPTH - perspective idea. All we have control of is the relative things. **4 ways to give this impression:**



- **Reverb** - far away WETTER, closer, DRYER. Closer: Brighter, Further Away - Duller.
- Listen to the space around you. The key to mixing is OBSERVING. **Listen, Listen, Listen...**

The Haas Effect

Haas Effect *Also called the precedence effect*, describes the human psychoacoustic phenomena of correctly identifying the direction of a sound source heard in both ears but arriving at different times. Due to the head's geometry (two ears spaced apart, separated by a barrier) the direct sound from any source first enters the ear closest to the source, then the ear farthest away. The Haas Effect tells us that humans localize a sound source based upon the first arriving sound, if the subsequent arrivals are within 25-35 milliseconds. If the later arrivals are longer than this, then two distinct sounds are heard. The Haas Effect is true even when the second arrival is louder than the first (even by as much as 10 dB!). In essence we do not "hear" the delayed sound. This is the hearing example of human sensory inhibition that applies to all our senses. Sensory inhibition describes the phenomena where the response to a first stimulus causes the response to a second stimulus to be inhibited, i.e., sound first entering one ear cause us to "not hear" the delayed sound entering into the other ear (within the 35 milliseconds time window). Sound arriving at both ears simultaneously is heard as coming from straight ahead, or behind, or within the head. The Haas Effect describes how full stereophonic reproduction from only two loudspeakers is possible.

SOURCE: The University of Texas at Austin.

Have fun playing with this psycho-acoustic effect!

The Haas Effect is a cool way to create a stereo effect even when we don't have a stereo source - or even two distinct sources to start with. Typically, if you didn't record in stereo *(with two mics or a stereo mic)*, multiple takes will provide the best results when you're looking for a nice, thick stereo sound. Usually I record two mono takes of a guitar part, put each on its own track, and pan those tracks left and right.I actually prefer this doubling method to *real* stereo (recording a single performance with two mics facing different directions to create a difference in each channel).

This effect is part of the process for sound localization. The short version of that link is that with multiple sources for data collection (nerves in our fingers, ears, etc.), the brain kind of ignores the duplicates to concentrate on the strongest signal, and uses the other information to tell us about the origin of the signal.

This is usually never the first method of choice for producing a stereo effect. But when you're out of other options, it sure does beat plain old mono. Try experimenting with the delay to find what sounds best, as well as playing with how far each track is panned. You can also try adjusting the sound level of each track to see what effect that gives.

SOURCE: http://www.homebrewaudio.com/the-haas-effect/

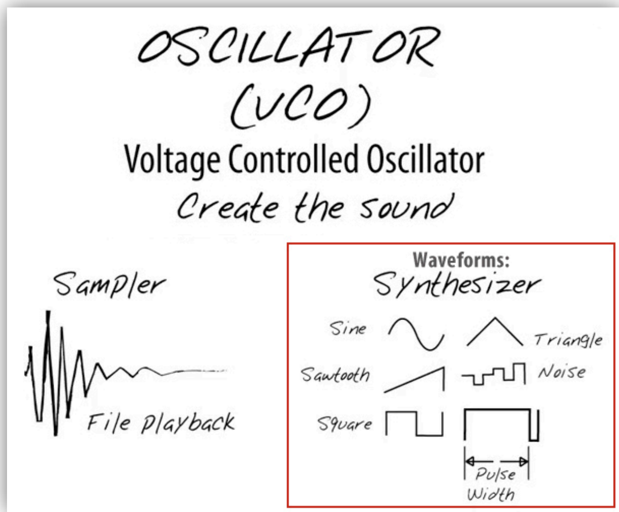
Lesson 6: Synthesis

#1 Lesson Overview

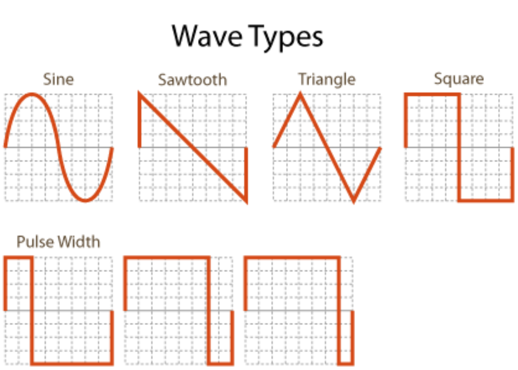
- Synthesis gives you a **language** for timbre and sound and a way to memorize and describe it. It's a contemporary tool for music production.

#2 Oscillator

- **This is the SOUND CREATOR.**



- It generates sound based on a geometric waveform. They get their name from their shape. Another name is a **VCO**. The synthesis oscillator can be modulated (where the pitch changes over time.)
- **Waveforms: Square** (hollow sound - like a 'clarinet') harmonic structure is missing the even partials 1, 3, 5, etc.; **Sawtooth** (bright, brilliant sound) has all the partials and sounds full. The shape of the waveform is related to how the sound is initially created.



- **Saw Wave** type of shape would be used for string

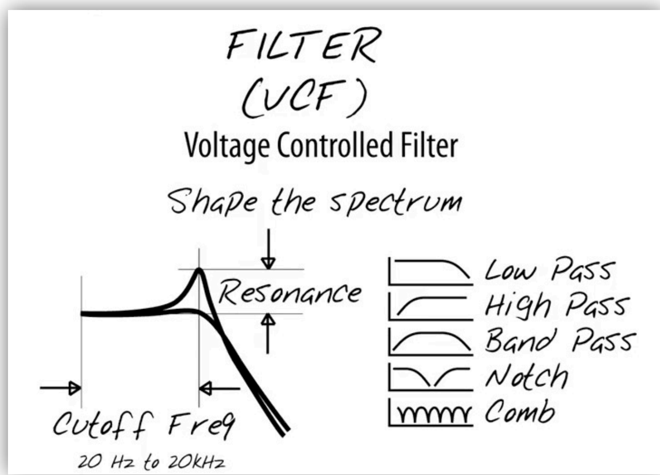
From: http://artificialtunes.tumblr.com/

instruments, called slip and stick.

- **Become familiar with these sections in your DAW with several Synth Plugins: Oscillator, Filter, Amplifier, LFO and Envelope.**

#3 Filters

- After the sound leaves the oscillator, it enters the filters. Can have simple or many filters. Unlike in an EQ, **the most important filter in a Synth is the Low-Pass Filter** (24db), referred to as "The Filter", because we want to remove the high end DRASTICALLY. You **can** switch them over to High Pass and Band Pass, but low pass used mostly.
- The most important modulated thing in the filter is the **Filter Cutoff:**



- When you hear a sweeping or wah-wah sound, that's the filter moving over time.
- **Human Voice comparison:** Oscillators - Vocal Folds (pulse wave); Filter - Mouth.
- Filters in a synth tend to be resonant. The sound of the filter becomes an important characteristic.
- Resonance is related to Feedback.
- You want the filter to move up and down along with the keyboard position. You can also have envelopes and LFO's modulate the filter frequency.
- If a patch is too bright or too dull, go to the BIG filter cutoff freq knob. It's very dramatic.
- **FILTER also named: FIL or VCF (Voltage Control Filter).**
- **A Stop Band Filter is also called a Notch.**

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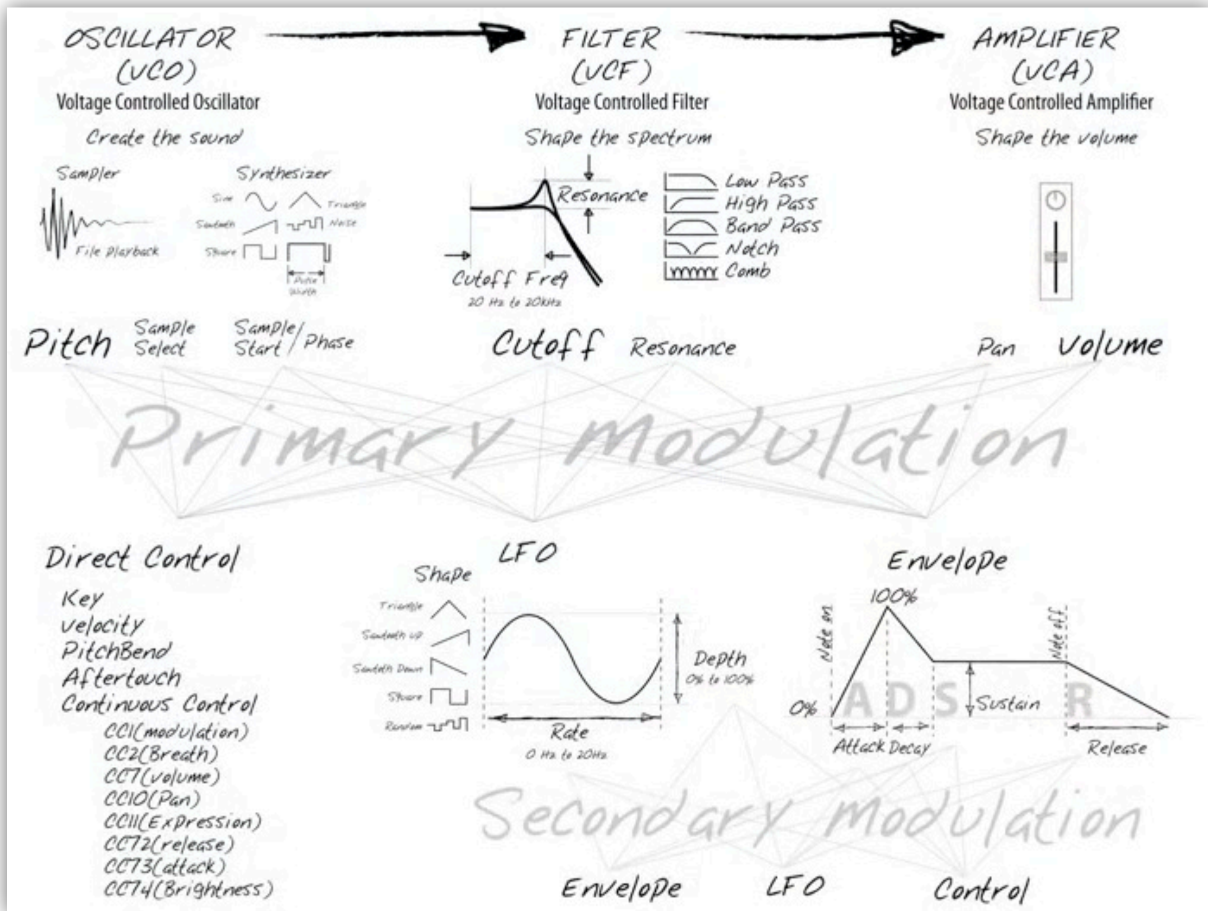

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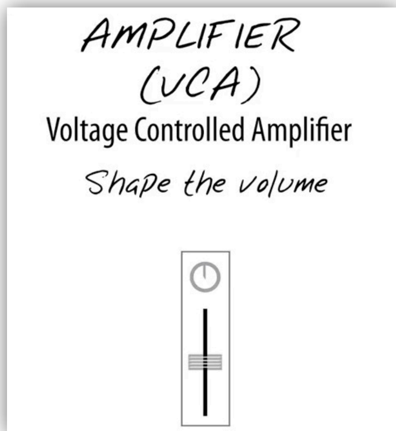
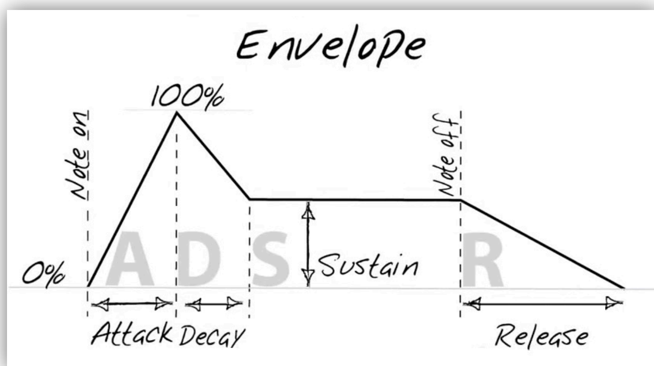
- Begin to notice what Hi Resonance sounds like. It is useful when you want to trying to emphasize the sound of the filter.
- **Apply these concepts to your synth. Manuals are great.**

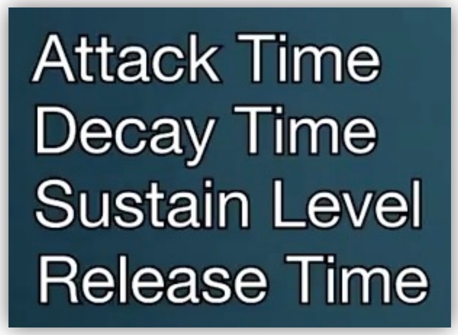
#4 Amplifiers



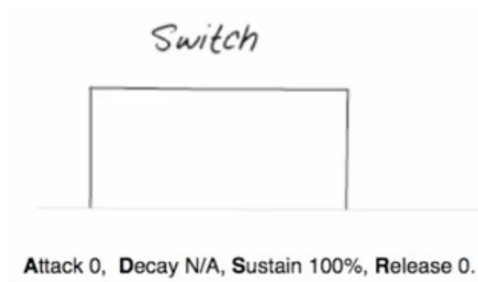
- Known as a **VCA** (Voltage Control Amplifier) - designed to move and change all the time and quite rapidly. There is a specific modulator attached known as an Modulator or Envelope - which is a set path that runs every time a key is pressed. It runs up and down, stays a while then goes away at the end of the note (release):

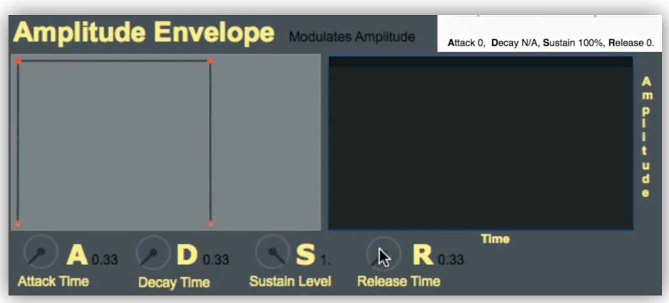


- With Compressors - in the Sidechain, that calculated an **envelope** which was an average of the time of the amplitude of sound going through it. That could be an Envelope "Follower". But, then it controls the Amplitude, so the volume part of the compressor is actually a VCA. The amp was creating a voltage that was controlling the VCA in the Compressor (or Gate, or Limiter, etc.)
- The Envelope in a Synth creates a path that we define with 4 Controls: **ADSR -** see Envelope Chart.



- Words are used in a different "context": Attack in Synthesis is very different than talking about say, a drum, if we said, "It has more attack", we would mean it was 'sharper', louder at the beginning. In a synth, Attack actually DULLS the beginning of the note.
- **OVERVIEW:** There is ALWAYS an envelope attached to the main amplifier. The envelope controls how te volume changes over time. Controlled by ADSR.
- General Purpose Modulators. You can have many different envelopes controlling very different things in your synth.
- Very often the LAST envelope in your synth will be your main Amplitude envelope.





- A little bit of Attack and Release time will often remove any clicks.

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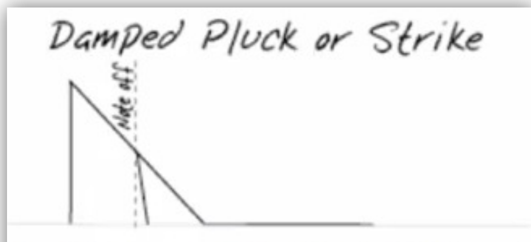
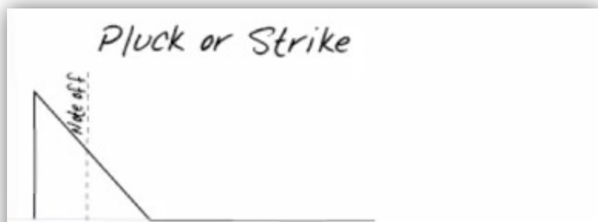
- Attack is very often set quickly.
- Hi Attack: Fade in; Hi Release: Fade out.
- A sustaining synth patch on an instruments blown or bowed will have a **Sustaining Envelope**.
- If Plucked or Hit, bring sustain level to 0, us a **Decaying Envelope.**
- The **decay time controls the length of the note.** The **Release Time** starts at the level where you release the key, no matter where that is.



Attack 0, **D**ecay 10-80%, **S**ustain 0, **R**elease 0

- For a percussive sound, where the length of the note doesn't matter, use these settings:

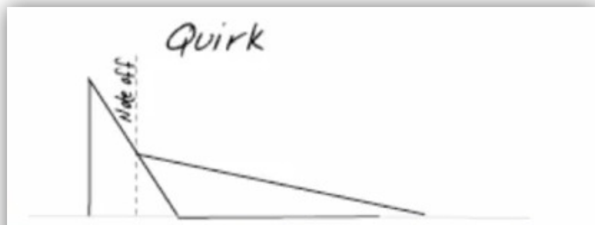


Attack 0, **D**ecay 10-80%, **S**ustain 0, **R**elease same as decay.

- **Sustaining Envelope with a Strong Attack:** (Bowing or Blowing)

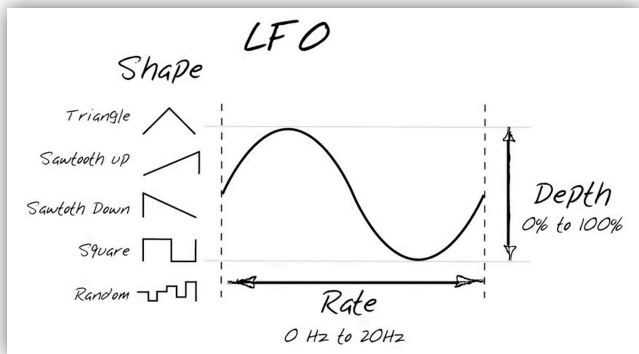
Attack 0-10%(almost 0), **D**ecay 0-10%, **S**ustain 50-90%, **R**elease 0-10%.

- For Punchier, reduce the decay time. Go away quicker, increase release time.
- **The Quirk Envelope:** Long notes: short blip. Quick Notes: long decay.



- **A**ttack 0, **D**ecay 10-30%, **S**ustain 0, **R**elease 50-100%
- Go to DAW and work with AMP Envelope.

#5 LFO (Low Frequency Oscillator)



- Cyclic in Nature (up and down) like Vibrato. LFO output is connected to the VCO input.
- **With Modulation you have 3 items:** Source (LFO); Destination (VCO); Amount of Modulation. You always have to set these things up in a Matrix.
- LFO (lower than the range of human hearing, below 20 Hz) is a Voltage Control Oscillator. The difference between it and an Oscillator is the rate that they run at.
- We can't really hear directly the output of the LFO. Tends to move in the 0 - 20 Hz Range.
- In a human singer, vibrato is a change in pitch, but also accompanied by a change in volume and timbre that runs right with that variation.
- In the simple synthesizer in the screen movies, I've hardwired the LFO to control the oscillator only. But for a more complex synthesizer, to represent the amplitude variations and timbre variations of a real vibrato, I could have the same the oscillator control the cut-off frequency of the filter, which would give a little bit of those timbre variations. The oscillator can control the

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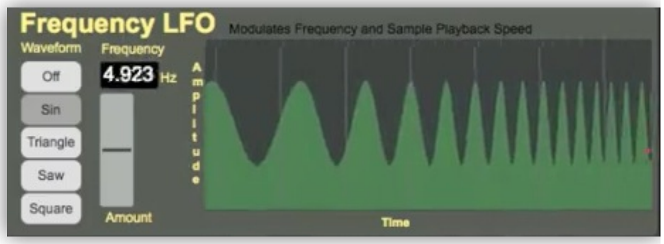
amplitude of the VCA (voltage controlled amplifier) to give those Amplitude variations.

- One modulation source can often control multiple destinations.



- Determine how to configure the modulation (vibrator) in your Synth: Source, Destination, Amount and Direction.

#6 A Language of Timbre

- Timbre is the difference between an oboe and a violin playing the same note.
- Listen differently! Synthesis gives us a language a collection of descriptors: Sustaining? Bright or Dull? Fast or Slow Attack? Cyclic Variations?
- Auto Pan - LFO controlling a Pan Knob. A lot of things can be described with the language of Synths.

Direct Control

Key
velocity
PitchBend
Aftertouch
Continuous Control
CC1(modulation)
CC2(Breath)
CC7(volume)
CC10(Pan)
CC11(Expression)
CC72(release)
CC73(attack)
CC74(Brightness)