Chapter 14

Plugins - software utilities offer virtually every processing function imganable with little or no reduction in quality, capabilities or automation features. These programs are designed to be intergrated into an editor or DAW production environment in order to perform a particular real-time or non-real-time processing function.

Most popular Standards - VST (PC/MAC) / DirectX (PC) / AudioSuite (MAC) / Audio Units (MAC) / MAS (MOTU for PC/MAC) / TDM + RTAS (PC/MAC)

Signal Paths in Effects Processing - signal processing device can be inserted into an analog or digital chain by inline or parallel routing.

Inline Routing

- a device can be plugged into an inputs' insert point. This approach is often used to insert an outboard device directly into an input strip's signal path
- a console's main output bus could be run through a device (stereo compressor) to affect an overall mix or submix grouping.
- an effects stomp box could be placed between a mic preamp and console input to create a grungy distortion effect.
- a DAW plugin could be inserted into an input path to process only the signal on that channel.

External Control over AN Inline Effects Signal Path

Certain in line effects processors allow for an external signal source to act as a control for affecting a signals as it passes from the input to the output of a device.

- a gate (an infinite expander that can control the passing of audio through a gain device) might take its control input from an external "key" signal that will determine when a signal will or will not pass inorder to reduce leakage.
- a vocal track could be inserted into a vocorder's control input, so as to synthetically add a robot-like effect to a track
- a voice track could be used for vocal ducking at a radio station, to fade out music when a narrator is speaking
- an external keyed input can be used to make a mix "pump" or "breathe" in a dance poduction

Send routing - is often used to augment a signal (delay, reverb)
It occurs whenever a portion of the original signal is allowed to pass through the chain while a side signal is simaltaneously fed to an effects device.

- a signal source (or mix grouping of effects send sources) can be sent to an effects device, which is then mixed back in with the original source at a console input or effects return bus.
- a signal source can be sent to an effects device that has an internal "MIX" control, which serves as a side-chain mix control for varying the amount of "dry" (original) signal to be mixed with the "wet" (effected) signal.

Effect Processing

- the spectral content of a sound: in the form of EQ or intelligent EQ + bandpass filtering.
- Amplitude level processing: in the form of dynamic range processing
- Time based effects: Augmentation or recreation of room ambience, delay, time / pitch alterations and tons of other special effects that can range from being sublimely subtle to "in your face"

Hardware + Virtual effects in action

The important rule to remember is that there are no rules; however, there are a few general guidlines.

- most important asset you can have is experiance and you own sense of artistry.
- gaining experiance takes time, a willingness to learn and lots of patience
- a device or plugin that lets us exert control over the harmonic or timbral content of a recorded sound.

Equalization refers to the alteration in freq response of an amplifier so that the relative levels of certain freq are more or less pronounced than others. EQ is specified as either plus or minus a certain number of decibels at certain freq.

Peaking Filters - its name implies, a peak-shaped bell curve can either be boosted or cut around a selected center freq.

Quality Factor (Q) - the width of its bell-shaped curve. Higher the Q, the fewer the freq are affected.

Bandwidth - is a measure of the range of freq that lie between the upper and lower -3dB (half power) points on the curve

Shelving Filters - a rise or drop in freq response at a selected freq, which tapers off to a preset level and continues at that level to the end of the audio spectrum. Shelving can be inserted at either high or low end of the audio range and is the curve type that's commonly found on home stereo bass + treble controls.

High-pass + Low-pass filters - this EQ type allows certain freq bandwidth to be passed at full level while other sections of the audible spectrum are attenuated

Passband - freq that are attenuated by less than 3dB are said to be inside the passband; those attenuated by more than 3dB are in the STOPBAND. The freq at which the signal is attenuated by exactly 3dB is called the Turnover or Cutoff FREQ and is used to name the filter freq.

EQ types
- Parametric EQ
- Selectable Freq EQ
- Graphic EQ
- Notch Filter

Parametric EQ - lets you adjust most or all of its freq parameters in a continously variable fashion. Control over the center freq and Q can be either selectable or continuous variable.

Selectable freq EQ - has a set number of freq from which to choose. These EQ usually allow a boost or cut to be performed at a number of selected freq. with a predetermined Q

Graphic EQ - provides boost + cut level control over a series of center freq that are equally spaced. The various EQ band controls generally use vertical sliders that are arranged side by side so that the physical positions of these controls could provide a "graphic" readout of the overall freq response curve at a glance.

Notch Filter - often used to zero in on and remove 60- or 50-Hz hum or other undesirable discrete-freq noises. They use a very narrow band-width to fine tune and attenuate a particular freq in such a way as to have little effect on the rest of the audio program.

EQ is all about compensating for deficiences in a sound pickup or about reducing extraneous sounds that make their way ~~pickup or about reducing~~ into a pickup signal. Whenever possible, EQ should not be used as a Band-Aid. Don't rely on alway being able to "fix-it-in the mix". EQ is best used in situations where:

- there's no time or money left to redo the track.
- the existing take was simply magical and shouldn't be re-recorded
- the track was already recorded during a previous session.

Tab

Instrument	Frequencies of Interest
Kick drum	Bottom depth at 60–80 Hz, slap attack at 2.5 kHz
Snare drum	Fatness at 240 Hz, crispness at 5 kHz
Hi-hat/cymbals	Clank or gong sound at 200 Hz, shimmer at 7.5 kHz to 12 kHz
Rack toms	Fullness at 240 Hz, attack at 5 kHz
Floor toms	Fullness at 80–120 Hz, attack at 5 kHz
Bass guitar	Bottom at 60–80 Hz, attack/pluck at 700–1000 Hz, string noise/pop at 2.5 kHz
Electric guitar	Fullness at 240 Hz, bite at 2.5 kHz
Acoustic guitar	Bottom at 80–120 Hz, body at 240 Hz, clarity at 2.5–5 kHz
Electric organ	Bottom at 80–120 Hz, body at 240 Hz, presence at 2.5 kHz
Acoustic piano	Bottom at 80–120 Hz, presence at 2.5–5 kHz, crisp attack at 10 kHz, honky-tonk sound (sharp Q) at 2.5 kHz
Horns	Fullness at 120–240 Hz, shrill at 5–7.5 kHz
Strings	Fullness at 240 Hz, scratchiness at 7.5–10 kHz
Conga/bongo	Resonance at 200–240 Hz, presence/slap at 5 kHz
Vocals	Fullness at 120 Hz, boominess at 200–240 Hz, presence at 5 kHz, sibilance at 7.5–10 kHz

Note: These frequencies aren't absolute for all instruments, but are meant as a subjective guide.

Dynamic Range - the gain reduction can be accomplised either by manually riding the fader's gain or through the use of a dynamic ranger processor that can alter the range between the signal's softest and loudest passages.

- Saturation
- Average Signal Level
- System/ambient noise

Saturation - occurs when an input signal is so large that an amp's supply voltage isn't large enough to produce the required output current or is so large that a digital converter reaches full scale.

Average Signal Level - is where the overall signal level of a mix resides.

<u>Metering</u> - audio engineers need a basic standard to help determine whether the signals they're working with will be stored or transmitted without distortion. (VISUAL LEVEL DISPLAY)

- AVERAGE
- PEAK

<u>Peak-to-Peak value</u> - the total amplitude measurement of the positive and negative peak signal levels.

<u>Peak-indicating Meter</u> - a readout that measures the maximum amplitude fluctuations of a waveform.

<u>Headroom</u> - the difference between the maximum level that can be handled without incurring distortion and the average operating level of the system.

Compression - in effect, can be thought of as an automatic fader. It is used to proportionately reduce the dynamics of a signal that rises above a user-definable level (threshold) to a lesser volume

- the dynamics can be managed by the electronics and/or amplifiers in the signal chain
- the range is appropriate to the overall dynamics of a playback or broadcast medium.
- an instrument better matches the dynamics of other recorded tracks within a song or audio program.

The most common controls on a compressor include; input gain, ~~thresh~~ threshold, output gain, slope ratio, attack, release and meter display.

- Input Gain - this control is used to determine how much signal will be sent to the compressor's input stage.
- Threshold - this setting determines the level at which the compressor will begin to proportionately reduce the incoming signal.
- Output Gain - this control is used to determine how much signal will be sent to the device's output. It's used to boost the reduced dynamic signal into a range where it can best match the level of a medium or be better heard in a mix.
- Slope ratio - this control determines the slope of the input-to-output gain ratio. In simpler terms, it determines the amount of input signal (dB) that's needed to cause a 1-dB increase at the compressor's output.
- Attack - this setting (calibrated in milli-seconds) determines how fast or how slowly the device will turn down signals that exceed the threshold.

- release - similar to the attack setting, release (ms) is used to determine how slowly or quickly the device will restore a signal to its original dynamic level once it has fallen below the threshold point.

  meter display - this control changes the compressor's meter display to read the device's output or gain reduction levels.

Compression should be used with care for any of the following reasons:

- Minimized changes in volume that occur whenever the dynamics of an instrument or vocal are too great for the mix.
- Smooth out momentary changes in source-to-mic distance.
- Balance out the volume ranges of a single instrument. Can be used to "smooth out" a bass line by matching their relative volumes. Often useful for EQ volume changes. Overcompression should be avoided to avoid pumping effects.
- Reduce other freq bands by inserting a filter into the compression chain that causes the circuit to compress freq in a specific band (freq-selective compression)
- Reduce the dynamic range and/or boost the average volume of a mix so that it appears to be significantly louder.

Multi-band Compression - works by breaking up the audible spectrum into various freq bandwidths through the use of mutible bandpass filters. This allows each of the bands to be isolated and processed in ways that strictly minimize the problems or maximize the benefits in a particular band.

- Dynamic upper range of a slap bass could be lightly compressed, while heavier amounts of compression could be applied to the intrument's lower register.
- An instrument's high end can be brighted simply by adding a small amount of compression. This can act as a treble boost while accentuating some of the lower-level high freq.

Limiting - is used to keep signal peaks from exceeding a certain level in order to prevent the overloading of amplifier signals, recorded signals onto tape or disc, broadcast transmission signals, and so on.

- To prevent signal levels from increasing beyond a specified level.
- To prevent short-term peaks from reducing a program's average signal.
- To prevent high-level, high freq peaks from distorting analog tape.

Expansion - is the process by which the dynamic range of a signal is propotionately increased. Depending on the system's design, an expander can operate either by decreasing the gain of a signal or by increasing the gain.

Noise Gate - this device allows a signal above a selected threshold to pass through to the output at unity gain and without dynamic precessing; however, once the input signal falls below this threshold level, the gate acts as an infinite expander and effectively mutes the signal by fully attenuating it.

- To reduced leakage between instruments. Often, parts of a drum kit fall into this catagory; ~~for example~~
- to eliminate noise from an instrument or vocal track during silent passages.

Time based effects - another important effects category that can be used to alter or ~~a~~ augment a signal revoles around delays and regeneration of sound over time. These time-based effects often add a preceved ~~delay~~ depth to a signal or change the way we percive the dimentional space of a recorded sound.

- Time delay or regenerated echoes, chorus + flanging
- Reverb.

Delay - alters the parameter of time by introducing various forms of delay into the signal path

- whenever delay fall below the 15 ms RANGE are slowly varied over time and then are mixed with the orignal undelying signal, an effect known as combing is created. (Less 15 ms)
- By combining two identical (often slightly delayed) signals that are slightly detuned in pitch from one another, an effect known as CHORUSING can be created. (15 to 35 ms) Increasing delay times into the 15-35 ms range will create signal that are spaced too closely together to be perceived by the listner as being discrete delays. these closely spaced delays create a doubling effect
- When the delay time is increased beyond the 35-40 ms point, the listner will begin to precieve the sound as being a discrete echo. (more than 35 ms)

Reverb - consist of closely spaced and random multiple echos that are reflected from one boundry to another within a determined space. This effect helps give us perceptible cues as to the size, density and nature of a space.

- Direct Signal
- Early Reflections
- Reverberation

Direct Signal - is heard when the orignal sound wave travels directly from the source to the listner.

Early Reflections - the term given to those first few reflections that bounce back to the listner from large, primary boundaries in a given space

Reverberations - these sounds are comprised of zillions of random reflections that travel from boundary to boundary within the confines of a room.

Reverb Types.

- HALL - simulates the acousicts of a concert hall.
- Chamber
- Room
- Live (STAGE)
- Spring
- Plate
- Reverse
- Gate

Pitch Shifting - can be used to vary the pitch of a signal or soundfile (upward or downwards) in order to traspose the relative pitch of an audio program without affecting its duration

Time + Pitch Changes

- Time change - a program's length can be alterd, without affecting its pitch, by rasing or lowering its playback sample rate
- Pitch change - a program's lngth can be remain the same while pitch is shifted either up or down.
- Both - both a program's pitch and length can be altered by means of simple resampling techniques

Multiple-Effects Devices

- A single device might offer a wide range of processing functions but allow only one effect to be called up at a time.
- A single device might offer a range of processing functions that can be "STACKED" to perform a number of simultaneous effects.
- An effects device might have multiple ins + outs, each of which can perform several processing functions (effectivly giving you multiple processors that can be used in a multichannel mixdown environment)

Dynamic Effects Automation + Editing
the ability to manipulate and vary effects parameters in real time over the duration of a song or audio program..
The ability to dynamically automate effects setting can be accomplished in any number of ways -
- Via MIDI controls and parameter change messages
- Via external hardware controller
- Via DAW or other form of automation control.