FH MMA SALZBURG – AUDIO

AUDIO EFFECTS

1 EFFECT PROCESSORS

1.1 HARDWARE EFFECTS

Hardware effect units can be based on *analog*, *electromechanical, electromagnetic*, *digital* or *hybrid* technology.

1.1.1 ANALOG

These effects use only discrete analog components to achieve the desired effect. They can be based on *valves* or *solid-state* electronic components (capacitors, inductors, resistors, etc.)
Examples: compressors (UREI 1176, Teletronix LA2A, Fairchild 670, etc.), EQs (Manley Massive Passive, NEVE 1073/1081 EQ, Pultec EQP-1A/MEQ-5, etc.), bucket brigade delays (moog moogerfooger MF-104, BOSS DM-2, etc.), modulation Effects (moog moogerfooger MF-108, BOSS CE-2, etc.), distortion, etc.

1.1.2 ELECTROMECHANICAL

These effects use transducers to convert analog electrical signals into physical vibrations in mechanical components (plate, spring, etc.), then contact microphones or pickups to translate the signal back into analog electrical signals.
Examples: spring reverbs (often found in guitar amps), plate reverbs (introduced in the late 1950s in studios, one of the most famous is the EMT 140).

1.1.3 ELECTROMAGNETIC (TAPE)

These effects (mainly delays) are based on a tape loop. Sound is recorded on tape magnetically and then played back by two separate read/write heads. The distance between the heads and the speed of the tape define the length of the delay. Examples: Roland Space Echo RE-201, Maestro Echoplex EP-3 and EP-4, etc.

1.1.4 DIGITAL

The signal is converted from analog to digital, processed using DSPs (Digital Signal Processors), and then converted back from digital to analog. The DSPs run special algorithms designed to either emulate the function of analog and electroacoustic units, or to produce unique digital effects. The additional AD/DA conversion can introduce a small latency to the signal – typically 1-2 ms – which prevents the use of these units for parallel processing (the processed signal would cause comb filtering when recombined with the original signal).

From the working principle point of view, there is no conceptual difference between a hardware DSP unit and a software plugin effect: DSP-based units sometimes use the same algorithms of their software plugins counterparts (for example: the Waves L2 Ultramaximizer is available both as hardware unit and as software plugin) and are not necessarily offering a better sound quality.

Examples: digital reverbs (Lexicon 480L, EMT 250, Bricasti M7), digital delays (Lexicon PCM42, Roland SDE-3000, TC Electronic M350), digital limiters (Waves L2), multi-effect units (TC Electronic M2000)

1.1.5 HYBRID

These units use a combination of several processing principles.
Example: a console using analog EQ and dynamics, with a digital multi-effect unit for delay and reverb effects.

1.2 SOFTWARE PLUGIN EFFECTS

Software plugins can be *native* or *DSP-based*.

Additional information about plugin formats can be found here: http://www.digitalnaturalsound.com/images/stories/fh_mma_courses/pdf/mg_digital_audio_formats_drivers_plugins.pdf

1.2.1 NATIVE PLUGINS

Native plugins use algorithms that run on the main CPU of the host computer; therefore, the processing power depends directly on the CPU performance (architecture, number of cores, clock speed) and RAM specifications. Upgrading the CPU to a faster model, the overall audio processing power increases as well.

Typical native plugin standards are:

- **Windows:** Direct-X, VST 2/3 (Virtual Studio Technology), ReWire 2, AAX (Pro Tools)
- **MacOS X:** AU (Audio Units), VST 2/3, ReWire 2, AAX (Pro Tools), Logic native effects (Apple LogicPro), MAS (MOTU Audio System for MOTU Digital Performer)

Figure 1: A rather large Cubase project, running over 100 high quality plugins (75 audio effects and 25 virtual instruments), with the host CPU load at around 40% (Intel i7-5930K CPU).

1.2.2 DSP-BASED PLUGINS

DSP-based plugins use algorithms that run on specific DSP processors, such as Motorola DSPs (like TC Power Core and Pro Tools TDM), Texas Instruments DSPs (like in Pro Tools HDX), or Analog Devices SHARK DSPs (like in Universal Audio UAD-2). They usually offer great stability and lower latency than most native systems, but at a higher cost, as they require special DSP-cards to run.

Figure 2: UAD-2 Octo DSP Accelerator as internal PCIe card (left) and external Thunderbolt unit.

Figure 3: ProTools HDX DSP Board

Some DSP accelerators (for example UAD-2 series) are available either as internal PCIe slot cards (that must be installed directly on the host computer motherboard) or as external units (connected through Firewire, USB2/3, or Thunderbolt).

Some other DSP systems (for example ProTools HDX) are only available as internal PCIe slot cards. Because the current Apple MacPro (since 2013) features no internal PCIe slots, the card must be installed in a PCIe expansion chassis (for example by Magma), adding more costs to an already expensive system.

When using DSP-based plugins, the processing power mainly depends on the specifications and number of DPS installed. The host computer CPU also contributes to the overall system performance, as it must shuffle a large amount of data between the DAW application and the DSP card. On a DSP-based system, to increase the processing power it is necessary to purchase and add additional DSP accelerator units.

Both Native and DSP engines can be used together, for example when using Pro Tools with AAX DSP plugins + AAX native plugins; or when using Cubase with a combination of Native and UAD plugins.

Typical DSP-based plugin standards are:

- **Universal Audio UAD2** available to the host as VST/AU/RTAS/AAX plugins
- **Pro Tools AAX DSP** Pro Tools HD
- **Pro Tools TDM** older DSP based standard for Pro Tools (discontinued)
- **TC Electronics PowerCore** available to the host as VST/AU plugins (discontinued)
- **Creamware SCOPE** (discontinued)

When DSP systems were first introduced in the ’90s (for example Pro Tools TDM), the processing power of DSP-based system exceeded that of the host CPUs and allowed running higher quality plugins (or a larger number of plugins) than on a native system. It also allowed to run a complete mix at extremely low latency, bypassing the operating system, drivers and computer hardware completely.

Nowadays (2016) the processing power of modern computer CPUs such as Intel i7 or Xeon CPUs (especially versions with 6 or more CPU cores) vastly exceeds that of DSP based systems such as Pro Tools HDX, UAD Quad/Octo, etc.

A well configured high-end PC with a professional audio interface (for example: RME Fireface UFX, UAD Apollo, MetricHalo ULN-8/LIO-8) can also achieve very short I/O latencies (less than 1,5 ms), rivalling that of DSP-based audio systems (at least as long as the CPU load is not too high).

1.3 HARDWARE VS. SOFTWARE EFFECTS COMPARISON

Until a few years ago, DSP-based units and software plugins could not match the sound quality offered by high-end analog units, for example state-of-the-art EQs, Compressors, guitar preamps, etc. by brands such as GM Labs, Manley Labs, Avalon Design, NEVE, SSL, API, Focusrite, etc.

Universal Audio (UAD) was one of the first companies achieving digital emulations of analog classics virtually indistinguishable from the originals. In 2010 they introduced an emulation of the Manley Massive Passive mastering EQ (tube based), followed by the 1176 series compressors, LA2A limiting amplifiers and Pultec EQs. Nowadays there are other companies also offering faithful emulations of analog hardware, including Slate Digital (currently offering state-of-the-art emulation of analogue consoles, tape recorders, compressors and EQs), Brainworx, SPL, etc.

All these new generation plugins are based on a technology called *component modelling*, where the algorithm does not just try to emulate the “sound” of the original unit, but the physical properties of every component of the hardware. Because of this, every aspect of the original effect can be reproduced, including non-linear behaviors, saturation, distortion, analog clipping, etc.

UAD also pioneered the 100% faithful reproduction of vintage digital effects, running the exact same exact algorithms found on the original units and then modelling the input/output amplifiers. Example: UAD Lexicon 224, EMT 250 and AMS RMX 16.

SlateDigital recently launched the VerbSuite Classics, a plugin that uses advanced impulse/response and convolution methods (based on licensed LiquidSonics technology) to faithfully emulate, in a single plugin, the sound of several hardware units (Lexicon 480L, Bricasti M7, AMS RMX 16, ETM 250, etc.).

Other companies do not try to emulate the sound or interface of existing hardware effect, but offer instead modern and innovative sound processing paired with a unique graphic user interface, optimized for the screen.
Among others, FabFilter with their excellent collection of Pro-Series plugins and Flux with the IRCAM Tools plugins.

From a sound design or mixing perspective, it does not really matter if a specific effect is running on an external hardware unit (based on analog, electromechanical, electromagnetic, or digital processing) or on a software plugin that emulates hardware units: the way you approach the control and setting of the parameters is similar.

The main difference is the interface: direct hardware control for the hardware units, and mouse/keyboard, touch-screen or external MIDI/USB controller for the software plugins. Ultimately, you can get great results with both.

Depending on the situation, there might still be differences in sound between hardware and software effect units, as well as general advantages and disadvantages in usage. Also, while most effects can be realized both with analog components or DSP based units (EQs, Dynamics), there are some that can only be realized using DSP (for example: time stretching, authentic sounding acoustic spaces, etc.).

HARDWARE VS. SOFTWARE EFFECTS COMPARISON TABLE		
	HARDWARE EFFECTS	**SOFTWARE PLUGIN EFFECTS**
Sound Quality & Character	Professional hardware effect units (such as Lexicon, Quantec, Avalon Design, GM Labs, NEVE, SLL, API, Focusrite, etc.) usually deliver excellent sound quality, paired with a very subtle *sound signature* or *character*, that until recently was very difficult to replicate in digital form.	While standard plugins (as included in popular DAWs such as Cubase, LIVE, Logic, etc.) are based on very simple digital algorithms and offer no particular *sound character*, there is nowadays a new generation of plugins based on component-modelling that can faithfully replicate hardware effect units in software form, including non-linear behaviors, saturation, soft-clipping, etc.
Controls & User Interface	Hardware effect units offer direct physical control on most or all parameters: especially in analog hardware effects, each function has its own switch, knob or fader, hence more intuitive and comfortable to use (and offering a pleasant tactile feedback), compared to software plugins using a traditional mouse/keyboard interface. This also allows simultaneous adjustment of several parameters at once, which not possible with traditional mouse/keyboard interface. Generally, hardware effects motivate to work using the ears and *feeling*, and not using the eyes and *thinking* (you rather *listen* to sound changes, instead of *looking* at plugin settings on screen).	Software plugins sometimes offer a very informative user interface, that can provide useful information unavailable in a hardware unit (for example, a real-time spectrum analyzer paired with an EQ curve). However, editing parameters with the mouse is not very comfortable, and looking at parameter visualizations there is the danger of judging more with the eye, and less with the ear. The usability of plugin effects and software synthesizers can be greatly improved using a multi-touch pad or screen, or an external MIDI/USB controller (such as the Mackie Control, or Avid S3). This allows tactile feedback, simultaneous adjustment of several parameters, and generally working in a more intuitive way, similar to using actual hardware effects.
Cost	Hardware units are expensive: for example, the Manley Massive Passive costs about 5.000 €, a Bricasti M7 reverb 2.800 € and a working Fairchild 670 can cost over 10.000 €. They also require maintenance for a 100% error-free operation. Servicing is expensive, as are the required replacement parts (especially valves).	Software plugins are usually considerably cheaper than their hardware counterparts: for example, the UAD Manley Massive plugin costs 300 € and the SlateDigital VerbSuite Classics reverb costs 185 €. They also require zero maintenance; however, they do require updates to maintain compatibility with the latest DAW and OS versions.
Studio Production	Running a complete mix using only hardware effects requires huge investments: a studio console with enough channels for all signals, many external compressors and EQs, several reverbs and delays, a patch-bay, etc. Because nowadays a similar sound quality can be achieved using advanced digital plugins, some traditional recording studios have a hard time to survive, because they cannot offer their services at a competitive price while maintaining all the hardware in perfect operating conditions.	Software plugins allow small budget studios to complete a professional audio production in the digital domain using a so called *virtual studio environment* for a fraction of the costs required in a traditional studio. Thanks to modern advances in plugin technology, this is possible without compromising in sound quality. The only things that cannot be easily emulated is the acoustic of a professional control room and (to a lesser extent), of a recording room. Which is why it still makes sense to produce (or at least mix and master) in a professional studio.

Reliability	Hardware effects can be used independently from host applications and computers; hence they offer higher reliability, which is an important factor for critical applications (like live concert recording and mixing). Even DSP-based hardware units are very reliable, as they are based on a custom operating system stored in an EPROM, with just the functionality required and nothing more.	Plugin effects run within of a DAW (Digital Audio Workstation: a computer running a HDR/MIDI sequencer application) and hence “depend” on it; if the host application or the OS crash, so do the plugins; for this reason, it is not recommended to use them for critical applications. On the other hand, some DAWs (such as Ableton LIVE) are optimized for live performance and offer several very reliable basic plugin effects.
Number of Instances	Each hardware effect unit can be used only once; it is not possible open another instance of a hardware effect: you need as many hardware units as the instances that are required. Alternatively, the track must be *bounced* to another track, including the effect (a process called in the early days “print to tape”), to free up the effect for a new task. However, after the bounce it is not possible to modify the effect parameters anymore.	Plugins are much more versatile: it is always possible to run several instances of the same plugin (until the DSP resources or the CPU of the host system have been maxed out), using a single plugin license. Therefore, it is possible to use the same plugin on several different tracks at the same time, with different settings, all accessible and modifiable in real time.
Latency & Monitoring	Analog effects have absolutely *zero latency*. Parallel processing (like parallel compression) is usually not a problem and there is no risk of comb filtering. onsider however that if you use an analog EQ, this will affect the signal phase, and phase cancellation/boost can occur when recombining the unprocessed (dry) and processed signals. Digital hardware effects, require AD/DA conversion and can have up to 2 ms latency. They should not be used for parallel processing.	When using plugins, the latency depends on the audio interface, the audio driver, and the DAW software settings. Latencies under 2 ms I/O are possible (for example with Pro Tools HDX systems, or very fast native systems), while 20-40 ms I/O are rather common. The latency is usually only an issue when recording/monitoring directly through the DAW, or when performing software instruments (synthesizers, samplers, etc.) in real time. A way to circumvent this is using the *Zero Latency Monitoring* option offered in some audio drivers and interfaces (for example RME) and rely on the audio interface own mixer for monitoring. During mixing, the latency can be set at a much higher value, as it is not so relevant anymore (the internal latency of plugins is compensated automatically by most DAWs).
Settings & Total Recall	Settings can rarely be saved (especially on analog units) and are not recalled automatically when loading a project in the DAW application; however, it is sometimes possible to save effects via MIDI and change settings using MIDI program change commands.	All plugin settings are saved automatically in the DAW project file. Additionally, most plugin types (VST, AU, AAX etc.) allow dynamic control of every parameter in real time using automation from the DAW; this makes it possible to execute complex dynamic mixes, *morphing* between different parameter settings, etc.
Using Too Many Effects	Eventually there are no free hardware effects left in the studio, so there is rarely the risk to use too many effects.	As it is so easy and convenient to add plugins, there is sometimes the tendency to use too many effects in a mix.

2 EFFECT SIGNAL ROUTING

2.1 WHEN TO USE INSERT AND WHEN AUX SEND/RETURN

Depending on the effect category, the use of channel **inserts** (effect used only within one channel, ev. with *mix* control for *dry/wet balance*) or **aux send/return** (effect available from more channels, effect mixed with the *dry* sound) might be preferred. The following are not strict "rules", just general recommendations:

- Generally, **Preamp or Console emulations, EQs, Filters and Dynamics** (compressor, expander/gate, limiter, etc.). are used as channel **insert** (if stereo, also as master insert), to use specific settings for each track.
- **Delays and Reverbs** are usually used as **aux send/return**, to spare processing power and/or to make the same effect available on more channels.
- **Modulation** (Chorus, Flanger, Phaser, etc.) and **Distortion effects** can be used in both ways; if they are used as insert, you use the *mix* parameter to adjust the amount of dry and effect signal; otherwise you use the *aux send* to add processed signal to the unprocessed one.
  Note: Chorus, Flanger etc. usually work best at around 50/50 % dry/wet settings. Also, remember that if you use them as send/return, adding more effect will also make the track louder.
- There are of course exceptions, for example, *Parallel Compression*: you can send all the instruments of a drum-kit to the same group channel, then compress the hell out of it and add the output to the original unprocessed signal – it sounds great!
- Important: when you use any effect as aux send/return, you should always make sure to have the *mix* parameter set to 100% *wet* (only effect signal): no dry signal should be added back to the original signal (in case of systems without *bus delay compensation*, this can cause unpleasant comb filtering coloration)
- Most third-party plugins in programs like Logic or Cubase do not necessarily "know" whether they are used as insert or aux send/return, so you might have to check that the mix parameter is set correctly (100% *wet* when used as send/return)

2.2 INTEGRATING EXTERNAL HARDWARE EFFECT UNITS IN A DAW

Cubase and other DAWs allow you to define a set of inputs and outputs to use as send/return channels for an external hardware effect unit. This can then be integrated in the mix like any other plugin (but of course can be used only once).

- You can adjust the Delay amount, Send and Return gain
- In the Return channel, you can finely adjust the stereo width with the separate pan controls on the L and R signal
- You can change the color of the effect sound with the internal EQ (works great on reverbs and delays!)

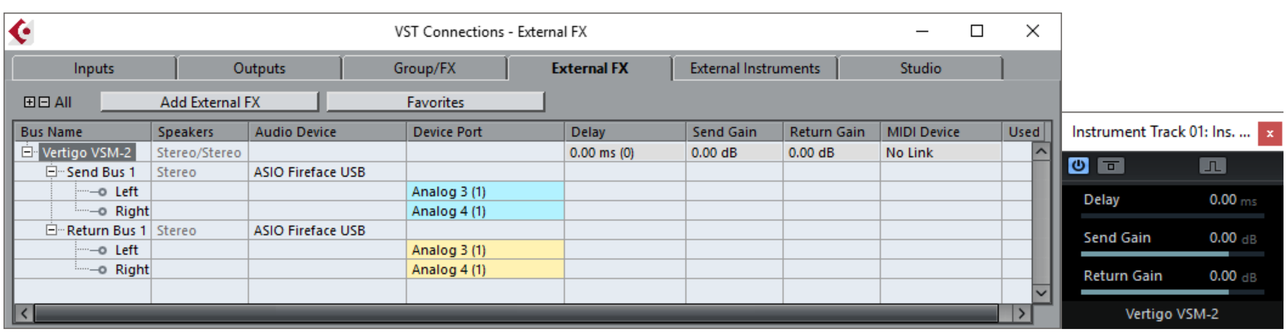

Figure 4: Cubase 9: setup of an external hardware effect in the VST Connections; Delay, Send and Return gain controls.

3 EFFECT GROUPS, DESCRIPTION AND PARAMETERS

3.1 FILTERS AND EQS

3.1.1 SHELVING EQ

Low and **High Shelving EQs** are used for general tone correction, to adjust the balance of the low and high frequency range; they work like the bass and treble controls on a standard hi-fi amplifier, or car sound system.

PARAMETERS

- **Type**: *Low* or *High Shelving*.
- *Fixed-Frequency Shelving EQs* only feature a *Gain* control (often found in budget mixers).
- *Semi-Parametric Shelving EQs* feature *Gain* and *Frequency* controls (often found in advanced mixer EQs).
- *Full-Parametric Shelving EQs* feature *Gain*, *Frequency* and *Slope* (Q) controls (often found in digital shelving EQs and special analog EQs, such as the Manley Massive Passive).
- **Gain**: controls the boost or cut amount (in dB).
- **Frequency**: sets the control frequency of the EQ (in Hz).
- **Slope** or **Q**: sets the steepness of the curve.
- **Curve Type**: simulates different types of shelving EQs (available for example in the Cubase Channel EQ).
- Many hardware mixers feature fixed control frequencies, typically set at 80 or 100 Hz for the low shelving EQ and 10 or 12 kHz for the high shelving EQ. In this case, the only available parameter is *Gain*.

Figure 5: Low Shelving EQ set at 80 Hz (+6 dB boost) and High Shelving EQ set at 12 kHz (+6 dB boost) – FabFilter ProQ2

If an instrument just sounds too *dark* and *muddy*, or too *thin* and *harsh*, it is often enough to adjust it with a low or high shelving EQ, rather than using a peak EQ (that could also change the basic character of the sound).

3.1.2 PEAK EQ

A **Peak EQ** is used for accurate tone shaping, to remove or emphasize specific formants, to change the character of a sound, etc. It is the “Swiss Knife” of the sound engineer.

PARAMETERS

- *Fixed-Frequency Peak EQ* only feature a Gain control (often found in budget mixers)
- *Semi-Parametric Peak EQ* feature *Frequency* and *Gain* controls.
- *Full-Parametric Peak EQ* feature *Frequency*, *Gain* and *Bandwidth* (Q) controls
- Some mixers have separate *low-mid* and *hi-mid* peak EQs, that cover different frequency ranges.
- **Gain**: the boost or cut amount (in dB).
- **Frequency**: the center control frequency of the EQ (in Hz).
- **Bandwidth** or **Q**: the steepness of the EQ curve.

Figure 6: two Full Parametric Peak EQs set at 100 Hz (-6 dB cut) and 2 kHz (+6 dB boost), both with Q set to “1” – FabFilter ProQ2

Tip: when you look for *formants* in an instrument or vocal part, first set a peak EQ gain to *boost*, medium Q, and sweep around searching for the desired frequency, until you can identify it.
Now you can set the gain to *cut*, if you wish to remove the undesired formant, or to moderate *boost*, if you want to emphasize this format and help the instrument cut better through the mix, without altering the channel volume.

3.1.3 LOW-CUT AND HI-CUT FILTERS

A **Low-Cut Filter** (also called High-Pass Filter) can be used to eliminate low frequency noise or vibrations (for example, mechanical noise transmitted through the floor to the microphone stand, or traffic noise).

A **High-Cut Filter** (also called Low-Pass Filter) can be used on bass range instruments (such as bass drum or bass guitar), in order to remove high frequency noise (*hiss*); it can be used to soften an otherwise aggressive instrument, for example an electric guitar with distortion; finally, it can be used for special effects: low pass filters with resonance are very popular in dance and electronica styles (techno, trance, drum'n'bass, etc.).

PARAMETERS

- **Type**: *High-Cut* or *Low-Cut*
- **Cutoff Frequency**: the control frequency in Hz, defined as the position at which the attenuation is -3 dB
- **Slope** (DE: *Flankensteilheit*): the steepness of the filter curve, expressed in dB/octave (6, 12, 18, 24 dB /oct)
- **Resonance** (synthesizer type filter) or **Q** (EQ): this parameter emphasizes the frequency range in proximity of the cutoff frequency, causing an artificial formant effect.

Figure 7: Low Cut filter set at 50 Hz and High Cut filter set at 10 kHz – FabFilter ProQ2

3.1.4 NOTCH FILTER

A **notch filter** is used to cut single frequencies out, for example a 50 Hz noise from a bad ground loop, without affecting the rest of the spectrum. It is extremely steep and must be used with caution.

PARAMETERS

- **Frequency** and **Bandwidth** (**Q**)
- It is similar to a full-parametric peak EQ, but it always works in *cut* mode; the gain reduction can be as much as -36 dB and the Q is generally much narrower than in a peak EQ.

Figure 8: a Notch EQ set at 100 Hz (with Q set to a moderate 5) and another Notch EQ set at 2 kHz (with a much narrower Q of 40) – FabFilter ProQ2

SOLID-STATE FULL PARAMETRIC EQ

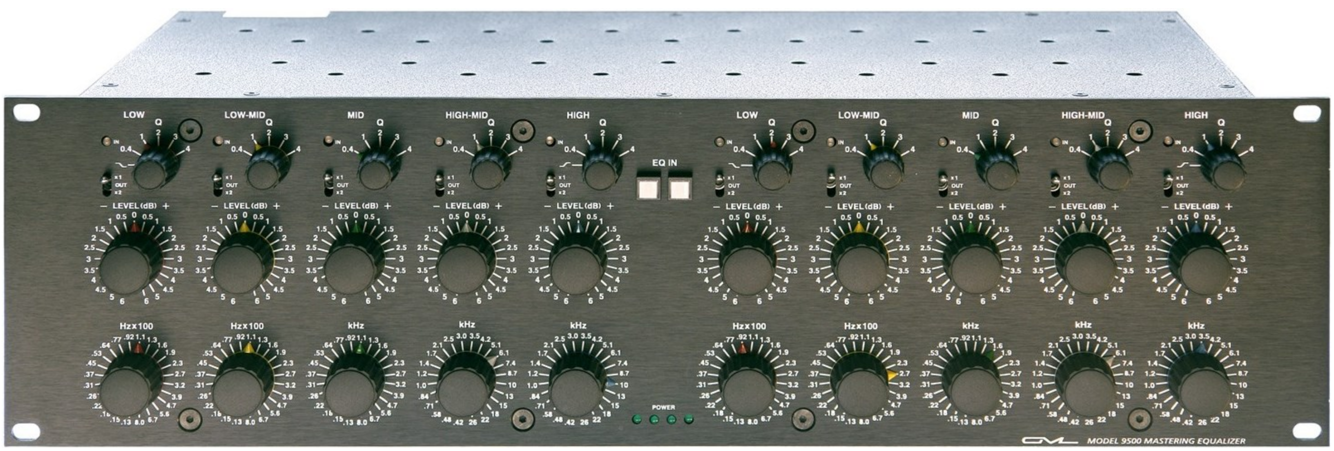

Figure 9: GML Model 9500 Mastering Equalizer (analog hardware)

SOLID-STATE SEMI-PARAMETRIC DUAL-CHANNEL EQ

Figure 10: API 550 Dual Equalizer (analog hardware)

Figure 11: NEVE 1073 DPX Equalizer (analog hardware)

VALVE-BASED PASSIVE STEREO EQ

Figure 12: Manley Massive Passive Stereo Equalizer (analog hardware)

Figure 13: UAD Manley Massive Passive (plugin, emulating the original Manley hardware)

3.2 DYNAMIC PROCESSORS

All dynamic processors are basically automated volume controls, where the output level is dependent on either the input signal level, or a different control signal (*side-chain*).

3.2.1 COMPRESSOR

The most common **Compressor** type is the *upward compressor*, that reduces the level of the signal by a set **ratio**, once the signal level crosses beyond a defined **threshold**. Further parameters (**attack**, **release**) control how fast the compressor reacts to changes in signal level.

A compressor can be used to:

- *control volume changes* (for example on vocals, to help "cut through the mix", reducing the differences in level between the loud and quiet parts)
- *limit the overall dynamic range of the signal* by reducing the peaks (in mastering, using moderate threshold and low compression ratios, so that it fits better on a medium with limited dynamic range such as CD)
- as a "sound design" device, to *change the character of the sound* from subtle to extreme (for example on bass drum, or electric bass, with extreme threshold and high compression ratios), making it *thicker*, *punchier*, *compact* and/or *louder*, or completely *crushing* the dynamics.

PARAMETERS

- **Input**: controls the level of the input signal.
- **Threshold**: defines which part of the dynamic range should be compressed (all signals louder than the set threshold); the lower you set the threshold, the larger portion of the dynamic range gets compressed; usually set to about -10 to -20 dB for most signals.
- **Compression Ratio**: the amount of compression relative to input (for example, *2:1* means you need *2 dB* at the input for *1 dB* more at the output); ratios of *4:1* or more can be used for vocals; sometimes *8:*1 or more for e-bass or e-guitar; the ratio used on single drum instruments can also be high, but it is best to keep that on the drum group moderate (2:1), unless you are looking for a special sound.
- **Soft** or **Hard Knee switch**: controls whether the compression starts immediately after the signal has gone over the threshold (at the set ratio), or whether the compression starts progressively already before the threshold and only reaches the set ratio well beyond the threshold (the soft knee usually sounds more musical).
- **Soft Knee dB value**: defines how many dBs before the signal hits the Threshold does the compression start.
- **Attack Time**: how fast does the compressor reacts after the signal has crossed the threshold (usually set shorter for percussive sounds, however a setting of *0 ms* should be avoided on digital compressors as this would completely remove the attack portion of the signal); typical settings are between *2* and *20 m*s.
- **Release Time**: how long does the compressor take to return to the original gain, after the signal has dropped again under the threshold level; typical settings are *20* to *200 ms*, or even longer for guitars and bass (up to 1 s).
- **Make-Up Gain**, or output level: boosts the overall signal level and is used to compensate for the gain "lost" through the compression.
- **Mix** or **Dry Mix**: mixes the original, uncompressed signal with the compressed signal, achieving *parallel compression* directly in the compressor, with the requirement of routing to a separate bus.

Digital Compressors just *compress*, they usually do not add any specific “coloration” (distortion, saturation, etc.) to the sound, like for example valve or FET based analog hardware. They might offer very fine control over the compression, at the cost of sometimes offering too many parameters to tweak. Sometimes a simpler design can yield better results, as less time is spent tweaking, and more actually making music.

Opto-Compressors are based on analog hardware and typically feature rather slow attack and release times. One of the most popular Opto-Compressors is the Teletronix LA-2A (later re-issued as hardware by Universal Audio).

VCA-Compressors are also based on analog hardware and usually feature very fast attack and release times. A typical example is the DBX 160, or the compressors included in the dynamic section of large format analog consoles by SSL, NEVE, etc.

FET-Style Compressors are also based on analog hardware and feature very fast attack and release times. A typical example is the UAD 1176 compressor.

DIGITAL COMPRESSOR

Figure 14: TDR Kotelnikov Digital Compressor (digital plugin)

Figure 15: WAVES C1 Compressor (digital plugin)

FET-STYLE COMPRESSOR

Figure 16: UAD 1176LN hardware FET-style compressor (analog hardware)

Figure 17: UAD 1176AE Limiting Amplifier (plugin, emulating Universal Audio original hardware)

Figure 18: SlateDigital FG-116 Compressors: standard, vintage and modern (plugin, emulating three versions of the 1176LN original hardware)

VCA COMPRESSOR

Figure 19: DBX 160 VCA-Compressor (analog hardware)

Figure 20: UAD DBX 160 (plugin, emulating the DBX original hardware)

OPTO COMPRESSOR

Figure 21: UAD Teletronix LA-2A opto-compressor (plugin, emulating the original hardware)

3.2.2 LIMITER

The **Limiter** is a special type of compressor that works with a hard-knee curve, infinite compression ratio and very fast (theoretically undetectable) attack and release times.
You can use it to:

- prevent *signal clipping* – usually placed as the last effect in the master chain (for example in mastering);
- reduce in an undetectable way the level of short signal peaks and make the whole track louder without losing perceived dynamics like with a compressor.

If you want the limiting to be more effective, without reducing the perceived loudness, you should try to use short attack and release times; however, be careful: at extreme short settings, you might get hearable distortion on the signal transients. Too long attack/release times should be avoided too: the limiting would not be very effective, as you would just get a reduced overall level and perceived loudness.

PARAMETERS

- **Input**: controls the limiting amount (higher input levels cause the limiter to work harder).
- **Release Time**: like in a compressor, only usually much shorter; sometimes an *auto release* option is available.
- **Output** or **Ceiling**: defines the maximum level of the output signal (typically -0.1 dBFS).
- Some limiters might also include a **Saturator**: it compresses and smooths the transients in a similar way to sound recorded to a tape recorder. At extreme settings, this can cause some distortion, but it is musically more pleasing than hard clipping the transients.

DIGITAL LIMITER

Figure 22: brainworx bx_limiter (digital plugin)

Figure 23: UAD Precision Limiter (digital plugin)

ULTRAMAXIMIZER

Figure 24: WAVES L1 Ultramaximizer with dithering section (digital plugin)

FET PEAK LIMITER

Figure 25: Pendulum Audio JFET/MOSFET Analog Peak Limiter (analog hardware)

PASSIVE LIMITER/CLIPPER/SATURATOR

Figure 26: Gyraf Audio Gyratec Infundibulum Limiter/Clipper/Saturator (analog hardware)

3.2.3 TRANSIENT DESIGNER

The **Transient Designer** is a combination of upward and downward compressors and expanders, coupled with very easy and intuitive controls.

It can be used to change the shape and character of percussive material such as drums and percussion loops, at the same time also affecting the amount of perceived ambience in the recording.

PARAMETERS

- **Attack**: controls the level of the *attack* portion of the sound (more/less punch).
- **Sustain**: controls the level of the *decay* portion of the sound (more/less ambience).
- **Output Gain**: controls the level of the output signal.

TRANSIENT DESIGNER

Figure 27: SPL Transient Designer | Twin Tube Processor (analog hardware)

Figure 28: UAD SPL Transient Designer (plugin, emulating SPL original hardware)

3.2.4 DE-ESSER

The **De-Esser** is a special kind of compressor that reacts only to the frequencies in the specific range of "S", "T" and other consonants (usually, a range between 5 to 10 kHz).

The *single band* versions compress the complete signal, while *multiband* versions split the signal and only compress the selected frequency range, leaving the rest unprocessed.

The *multiband* version can also be used to control specific instruments in a mix (for example hi-hat, or snare) by setting the frequency to the specific *formants* of that instrument.

PARAMETERS

- **Frequency**: the center of the control range frequency.
- **Side Chain**: the type of filter is used on the *control* signal (low cut or peak).
- **Threshold**: the level from which the signal should be compressed.
- **Mode**: single band or multiband.
- **Monitor**: *audio* selects the standard effect output, while *side chain* is used to listen to the filtered control signal (to check if it matches the right frequencies).

ANALOG DE-ESSER

Figure 29: SPL Dual Channel Auto Dynamic De-Esser (analog hardware)

DIGITAL DE-ESSER

Figure 30: UAD Precision DeEsser (digital plugin)

3.2.5 EXPANDER/GATE

The **Expander/Gate** is used to remove undesired, low dynamic level parts of a signal; example: to reduce “spilling” from the different microphones when recording a drum-kit, or to reduce/remove noise between vocal parts.

PARAMETERS

- **Threshold**: the gate only opens when the signal level is higher than the set threshold, else no signal is sent to the output.
- **Gain Reduction**: a gate reduces the level to -oo when closed, while an expander reduces the signal level by a set ratio (like a compressor, but inverted).
- **Attack Time**: how long does it take for the gate/expander to open/increase the gain, after a signal reaches the threshold.
- **Hold Time**: how long the gate/expander remains fully open.

- **Release Time**: how long does it take for the gate/expander to gradually close/reduce the gain, after the signal level has dropped under the threshold again.
- **High Cut** and **Low Cut** in the *side chain*: define which frequency range should control the gate/expander (example: if you have a drum loop and want to isolate the snare only, you set the gate to react only do mid-high frequencies, and very high peaks; if you want to isolate only the bass drum, it should react only to very low frequencies, and high peaks).

DIGITAL EXPANDER

Figure 31: Steinberg Expander (digital plugin)

DIGITAL GATE

Figure 32: Steinberg Gate (digital plugin)

ANALOGUE MODELED EXPANDER/GATE

Figure 33: Nomad Factory Blue Tubes Logical Gate Expander 622 (plugin, emulating analog hardware sound)

3.3 MODULATION EFFECTS

Modulation effects can be used to *thicken up* and add *movement* and *texture* to sounds that are otherwise *flat*, *static* and/or *uninteresting*; for example, a simple synth pad with just one oscillator per voice sounds much nicer and lively with added chorus or flanger.

Modulation effects can be used on acoustic and electric guitars, on vocals etc. to make them sound *thicker* (a sort of *doubling effect* without re-recording the part). They do not sound very good on complex acoustic instruments (like piano) or ensembles (like strings, brass, etc.).

With extreme settings, you can create special effects (especially with the flanger and phaser) that substantially change the character of the original sound.

3.3.1 CHORUS

Figure 34: Nomad Factory Blue Tubes Dual-Analog Chorus CH-2S (plugin, emulating analog hardware sound)

A **Chorus** is based on the following principle: the signal is delayed by one or more delay lines, then modulated in pitch by a LFO (low frequency oscillator); finally, the effect signal is mixed together with the dry signal (usually about 50/50), producing coloration due to *Comb Filtering*. A Chorus uses longer delay times than the Flanger (typically, 5-40 ms).

PARAMETERS

- **Delay**: the delay amount from the dry signal to the modulated lines.
- **Speed** or **Rate**: the speed of the LFO(s) that controls the pitch modulation; it can be synced to the song tempo.
- **Modulation Depth**: how much does the signal modulates above and under the reference pitch.
- **Shape** or **Waveform**: defines the type of the LFO waveform (sine, triangle, sawtooth, etc.)
- **Stereo Width**: in a stereo chorus/flanger, it controls the spread of the delay lines in the stereo field (how far left and right are they panned).
- **Mix**: controls the balance between dry and effect signal.

Figure 35: UAD Roland Dimension D Chorus (plugin, emulating Roland original hardware)

3.3.2 FLANGER

Figure 36: UAD MXR Flanger (plugin, emulating original hardware)

The **Flanger** principle is similar to that of a Chorus: the signal is delayed by one or more delay lines, then modulated in pitch by a LFO. However, a Flanger uses shorter delay times than the chorus (typically, less than 5 ms), which causes more pronounced *Comb Filtering* artifacts and has the additional parameter *feedback*, which feeds part of the effect signal back into the effect unit, causing additional coloration.

PARAMETERS

- **Sweep Width**: controls the delay modulation amount from the LFO, which in turns affects how high and low in the frequency spectrum the comb filtering effect sweep will be hearable
- **Sweep Speed**: controls the speed of the LFO modulation
- **Regen** or **Feedback**: controls how much of the processed signals is fed back into the effect unit (causing additional coloration and a more *electronic*, *liquid* or *blubby* sound)
- **Phase** or **Invert**: defines the phase of the delayed lines
- **Mix**: controls the balance between dry and processed signal

3.3.3 PHASER

The **Phaser** differs from the Chorus and Flanger, as it does not use traditional delay-lines but all-pass filters to just shift the phase of the signal. When recombined with the original, unprocessed signal, it causes coloration due to *Comb Filtering* (similar to the chorus and flanger, but more pronounced). Because there is no delay, the Phaser is better suited to process sounds with strong percussive character, like drums and percussion.

Figure 37: Nomad Factory Blue Tubes Analog Phaser APH-2S (plugin, emulating analog hardware sound)

PARAMETERS

- **Rate** and **Mod. Depth**: defines the speed and amount of LFO modulation
- **Shape** or **Waveform**: defines the type of the LFO waveform (sine, triangle, sawtooth, etc.)
- **Resonance** or **Feedback**: controls the amount of signal fed back into the effect
- **Color**, **Sweep Floor, Width** and **Ceiling**: control the color of the phaser effect
- **Mix**: sets the balance between dry and effect signal
- **Output**: controls the output level

Figure 38: SoundToys PhaseMistress (plugin, emulating analog hardware sound)

3.3.4 ROTARY SPEAKER (LESLIE)

Before the advent of analog and digital effects, Donald Leslie invented a special type of speaker cabinet with rotating bass and horn elements. These speakers were mainly used in combination with Hammond Organs (an electromechanical instrument).

The speed of the rotating elements can be set between about 40-50 rpm (chorale) and 340-400 rpm (*tremolo*). The rotation causes the speakers to move closer and farther away from the listener, and hence a modulation of the pitch and color due to the resulting *Doppler Effect*. The effect is somewhat similar to that of a Chorus, with an added modulation of the high frequency range (due to the horn alternatingly pointing to – and away from – the listener).

There have been several attempts to recreate the sound of a Leslie Speaker through analogue as well as digital effects, but (so far) almost none can faithfully capture the unique character of the original device, because it also changes character depending on the acoustic of the room the speakers are placed in.

PARAMETERS

- **Overdrive**: overloads the input section, adding color and distortion to the signal.
- **Slow / Fast**: defines the speed of the rotation, separately for the Bass and Horn speaker elements.
- **Acceleration**: the time the modulation takes to change in speed.
- **Phase**, **Angle**, **Distance**: simulate the characteristics and positioning of the original speaker.
- **Mix**: sets the balance between the dry and processed signal.

Figure 39: Hammond Leslie 3300 rotary speaker controls (original hardware)

Figure 40: Steinberg Rotary Speaker (digital plugin, emulating rotary speaker sound)

3.4 AMBIENCE EFFECTS (REVERB, DELAY)

Ambience effects are mainly used to add *depth of field*, *dimension* and *space* to a mix, placing every instrument and vocal element in a virtual acoustic space that fits its character for a specific music style and piece. They can also be used as special effects and sound design devices.

3.4.1 ROOM/HALL REVERB

Digital Reverbs can work on the principle of a *mathematical algorithm*, or be based on *impulse responses*.

Impulse Response Reverbs (also called **Convolution Reverbs**) generate the reverb from impulse responses recorded from real acoustic spaces (as well as from plate and spring reverbs, or other digital reverb units).
The simulation of real acoustic spaces (such as rooms, studios, halls and churches) can be so convincing that it can be almost undistinguishable from an actual recording done in the simulated space. Therefore, Impulse Response Reverbs are the main choice when mixing classical music productions.
They can also be used to simulate hardware reverb units, such as digital algorithm reverbs, plate or spring reverbs; only the most sophisticated impulse response algorithms include parameters to recreate the modulation aspects of the algorithm used in the original hardware (for example: SlateDigital VerbSuite Classics).
The main drawback is that the reverb parameters (reverb time, room size etc.) can only be adjusted within certain limits: the basic *character* of the impulse used is determined by the acoustic character of the original space or device recorded.
Examples: Steinberg REVeleation, AudieEase Altiverb, Waves IR1, SlateDigital VerbSuite Classics.

Algorithm Based Reverbs use complex mathematical formulas to simulate the reverberation in a real acoustic space. The model usually uses combination of early reflections and reverb tail algorithms. These reverbs are usually very flexible, because all parameters can be adjusted freely to fit the mix requirements.
They might not always sound as authentic as impulse response reverbs, which is why they are mainly used for pop/rock/electronic productions, but not much for classical ones.
Examples: Lexicon 224/300/480L, EMT250, Quantec Room Simulator, Quantec Yardstick, Steinberg RoomWorks, FabFilter Pro-R, Valhalla Room, etc.

Plate Reverbs are based on a resonating metal plate. The analog electric signal is applied using transducers (electromagnets) to one end of the plate, then the resonances are captured by contact microphones on the other end. Using multiple microphones, a stereo reverb can be simulated. Within certain limits, a plate reverb can come quite close to the character of a "real" room. They are mainly used on vocals and drum/percussion parts. Example: EMT 140

Spring Reverbs are based on a resonating spring. The analog electric signal is applied using a transducer to one end of the spring, then the resonance is re-captured using a pickup at the other end. This type of reverb, often found in guitar pre-amps or cabinets, does not sound very realistic and has a strong characteristic "ringing" in the reverb tail, but it might still fit well the sound of specific instruments (such as an electric guitar, or a Hammond organ, etc.). Example: ALG BX20

REVERB PARAMETERS

- **Pre-Delay**: the time between the dry signal impulse and the first *early reflections*; it simulates the distance between sound source and reflective surfaces.
- **Reverb Time**: the average decay time of the reverb, often defined as *RT60*, or the time it takes for the sound pressure level to decay 60 dB (= 1/1000 of its former value).
- **Room Size:** the size of the emulated virtual space; it affects the basic frequencies of the room resonances; usually larger spaces tend to sound more "diffuse" than small ones.

- **Room Shape**: the shape of the emulated virtual space; it affects the pattern (ratio and spectral distribution) of the room resonances.
- Both *size* and *shape* affect the *character* of the reverb (smooth, rough, ringing, diffuse, etc.).
- **Stereo Spread** or **Width**: affects the spread of the reverb delay lines within the stereo field.
- **Density**: the amount of reverb delay lines used; usually higher values produce a smoother decay.
- **High Cut** and **Low Cut Filters**: affect the bandwidth in the high and low frequency range; if you are trying to emulate a realistic acoustic space, the reverb should already be cut at around 4-5 kHz.
- **Low** and **High Frequency Damping**: affect the decay time of low and high frequencies in relation to the defined reverb time; these parameters simulate the different absorption factors of typical construction materials such as wood, brick, concrete, curtains, etc.; in some reverbs the *material* type can be selected directly.
- **ER** and **Reverb Level**: the level of the *early reflection* and the *tail* part of the reverb.
- **Dry/Wet Balance**: adjusts the reverb level in relation to the dry signal.

DIGITAL ALGORITHM REVERB

Figure 41: UAD EMT250 Digital Reverb (plugin, emulating the original hardware unit using the same algorithms)

Figure 42: Lexicon 300 Digital Reverb (hardware)

IMPULSE RESPONSE REVERB

Figure 43: SlateDigital VerbSuite Classics (plugin, emulating several classic hardware reverb units including the Lexicon 480L)

Figure 44: Steinberg REVerence Impulse Response Reverb (plugin, based on impulse responses recorded in real acoustic spaces)

PLATE REVERB

Figure 45: UAD EMT 140 Plate Reverb (plugin, emulating original EMT plate reverb hardware)

SPRING REVERB

Figure 46: UAD AKG BX20 Spring Reverb (plugin, emulating the original AKG hardware unit)

SOME TIPS ABOUT USING REVERBS IN A MIX

- It is recommended not to use too much reverb, as it can make the track sound "muddy" (make the vocals or instruments unclear), or just sound rather "old fashioned" (like some late seventies /early eighties productions, that used a lot of reverb).
- Avoid using many long reverbs: sometimes shorter reverbs work better; use also different reverb widths, do not leave all in full stereo; generally, try to use different sound spaces with different characters (long, short, bright, dark, wide, narrow, etc.); you can also use EQ on the reverb output (aux return) to further modify the reverb character
- Sometimes a delay may be used in place of a reverb: the track will sound more "transparent", and the delays can be set to be in "sync" to the song beat.

3.4.2 ECHO / DELAY

Delays produce one or several repeating *echo-like* reflections, with variable repetition patterns, amount and times. There are several different types of delays:

- **Mono or Stereo Delay**
- **Ping-Pong Delay**: the delay lines are cross-fed L to R and R to L, creating a *bouncing* effect across the panorama.
- **Modulation Delay**: can be used to create Chorus or Flanger type effects.
- **Tape Delay**: based on a tape loop, nowadays emulated in digital plugins.

DELAY PARAMETERS

- **Delay Time**: usually separate values for L and R channels in stereo delays; time can be set in milliseconds, or in note values related to the song BPM (beat per minute) value;
- **Feedback**: feeds the effect output back into the delay line and controls the number of delay repetitions;
- **Cross Feedback**: feeds the output of the L delay into the input of the R delay, and vice versa, creating a so-called *ping-pong* delay;
- **High-Cut** and **Low-Cut** filters: affect the bandwidth and hence the color of the delayed signal; it generally sounds more interesting if the delay is not a 100% accurate copy of the dry signal (which was always the case using "vintage" equipment)
- **Mix**: adjusts the balance between delay and dry signal.

Some delays (like the PSP 84 shown in Fig. 48) feature additional filters (with cutoff and resonance) to change the character of the delay, or modulation parameters to create chorus- and flanger-type effects, as well as a reverb section.

DIGITAL MONO DELAY

Figure 47: PSP Lexicon 42 (plugin, emulating the original Lexicon 42 hardware)

DIGITAL STEREO DELAY

Figure 48: PSP 84 Stereo Delay (plugin, inspired by the Lexicon 42 hardware delay)

VINTAGE DELAY

Figure 49: SlateDigital Repeater Vintage Modelled Delay (plugin, emulating several classic hardware delay units)

TAPE DELAY

Figure 50: UAD RE-201 (software plugin, emulating original Roland hardware)

Figure 51: UAD EP-34 Classic Tape Echo (plugin, emulating the original Echoplex EP-3 and EP-4 hardware units)

3.5 SATURATION AND DISTORTION EFFECTS

3.5.1 OVERDRIVE, DISTORTION

Figure 52: SoundToys devil-loc distortion (plugin, inspired by the sound of analog hardware)

Overdrive and **Distortion** plugins simulate the saturation and distortion of analog preamps, tube guitar amplifiers and speaker cabinets.

OVERDRIVE AND DISTORTION PARAMETERS

- **Input**, **Drive** or **Boost**: controls the input stage, and how hard the amp is driven; higher levels produce more distortion.
- **Output**: at high drive levels, it is used to compensate the output level to avoid clipping.
- **Tone** or **Color**: adjusts the tonal character of the distortion; often this is just a low pass filter, or a simple combination of *bass* and *treble* controls.
- **Feedback: simulates the feedback occurring between the speaker cabinet and the guitar strings;**
- **Spatial**: controls the stereo width.
- **Mix**: balance between *distorted* and *clean* signal.

Figure 53: Native Instruments Driver Overdrive/Distortion (plugin, inspired by analog hardware)

3.5.2 GUITAR AMP + SPEAKER

Guitar amplifiers can be valve or sold-state based. Traditionally, the sound of an overdriven or distorted electric guitar is achieved overloading the input of the valve stage of the preamp.

PARAMETERS

- **Gain**: controls the input stage drive; higher values create more distortion.
- **Distortion Type**: Clean, Overdrive, Distortion, Crush, etc.
- **Bass**: low shelving EQ.
- **Lo-/Hi-Mid**: low-mid and high-mid peak EQ.
- **Treble**: high shelving EQ.
- **Presence**: a sort of “Air” filter, gently boosting the mid-high range to achieve more presence.
- **Volume**: output level.

Figure 54: Brainworx RockRack Guitar/Bass Amp (plugin, emulating hardware preamp and speaker cabinet)

Figure 55: Native Instruments Guitar Rig 5 with Rammfire Preamp+Cabinet (plugin, emulating hardware preamp and speaker cabinet)

3.5.3 SATURATORS

These processors change the color and character to the sound, by adding different types of harmonics. The effect can go from very subtle (adding just some *warmth* and *character*) to extreme distortion. As the implementation of controls and effects differs greatly depending on the unit, the parameters will be described for each effect separately.

VERTIGO VSM-3 MIX SATELLITE

Figure 56: brainworx Vertigo VSM-3 (plugin, emulation the original hardware from Vertigo)

This units exists both as hardware (Vertigo VSM-2) and software emulation (Vertigo VSM-3). It is a *creative analog mix and mastering tool,* designed to add color and character to the mix.

It offers M/S (Mid/Side) processing, blending and two sophisticated coloration generators.

PARAMETERS

- **Input** and **Output Level**: used to careful level match the signal before and after processing and allow A/B comparison without being fooled by differences in level.
- **2nd Harmonic FET Crusher** section: the controls for the FET type section, that emulates a Class A and triode tube stage. It creates pure 2nd harmonics (octave), adding warmth and richness to the sound.
- **3rd Harmonic Zener Blender** section: the controls for the valve type overdrive section, that emulates a pentode valve into overload. It creates 3rd harmonics, but also some higher order odd harmonics, depending on the amount of drive. It makes thing sound brighter, brings out detail and adds a subtle compression effect.
- **Drive**: controls the amount of drive and harmonic generation.
- **Level**: sets the output level of this section.
- **Input Filter**: defines what frequency range is passed into the harmonic generator.
- **Shape**: this is a high cut filter designed to get rid of distortion byproducts and prevent a harsh and gritty result.
- **THD Mix**: this is a dry/wet control where you can mix between the unprocessed signal and the output of the harmonic generator
- **Style**: defines the attitude and style of the harmonic generator, with a software or harder characteristic.
- **M / LR / S** Selector: this switches defines if either M, LR or S is sent to the harmonic generator.
- **Parallel/Serial** Switch: In Parallel mode, the two distortion units work in *parallel* and the blend between 2nd and 3rd harmonic is controlled by the **THD Mixer** control. *Serial* overrides the mix pot setting and routes the signal the 2nd Harmonic stage first followed by the 3rd Harmonic stage.

SOUNDTOYS DECAPITATOR

Figure 57: SoundToys 5 Decapitator Saturator (plugin, emulating the saturation of analog hardware consoles such as API, SSL, Neve, etc.)

The Decapitator is not really emulating a specific hardware unit, but rather the saturation and distortion characteristics of several analog devices. While not as flexible as the DSM-3 (it does not offer M/S processing or selective frequency filters), it can offer a wide array of coloration characteristics, from subtle saturation up to crushing distortion.

PARAMETERS

- **Style**: chooses the type of saturation distortion. A is modeled after the *Ampex 350 tape drive preamp*; E is modeled after the *Chandler/EMI TG channel*. N is modeled after the *Neve 1057 input channel*. T and P are modeled after *Thermionic Vulture Culture* triode and pentode settings, respectively.
- **Drive**: it acts like an input gain control in an analog circuit. The harder the signal is pushed, the more it will saturate. At higher settings, it will start producing hearable distortion. The manner how the audio will distort depends on the circuit modelling style chosen.
- **Punish**: adds an extra 20 dB of gain to the input signal, producing more loud, distorted and even brutal results.
- **Low Cut**: removes low frequencies *before* they hit the saturation section. **Thump** adds a few dB of boost right at the Low Cut frequency (similarly to the resonance of an analog filter)
- **Tone**: an overall control for the color of the sound, from dark to bright – a sort of tilt filter operating *before* the saturation section.
- **High Cut**: removes high frequencies from the distorted sound, operating *after* the saturation section. The **Steep** switch alters the slope of the High Cut filter: when it is OFF the filter is a very gentle 6 dB /oct rolloff, when it is ON it becomes a super-steep 30 dB /oct rolloff.
- **Output**: controls the output level. Higher drive settings can considerably increase the audio level emanating from the Decapitator, the output control can be used to compensate for that and bring down the volume again.
- **Mix**: determines the balance between the original audio and the signal processed by the Decapitator.

3.5.4 BIT-CRUSHER, BUFFER OVERRIDE, ETC.

Figure 58: Steinberg Bit-Crusher (digital plugin)

A Bit-Crusher artificially reduces the bit resolution and sampling frequency of a signal, making it sound “lo-fi” (low fidelity); at extreme settings, the dynamic and frequency range are extremely reduced, while quantizing noise and “alias” distortion become dominant factors in the sound character.

“Lo-Fi” sound is often used in many contemporary musical styles (for example drum'n bass, chillout, electronica, etc.)

BIT-CRUSHER PARAMETERS

- **Bit Depth** or **Resolution**: removes the information in the LSB (least significant bits) from the digital signal: for example, setting the depth to “8” limits the resolution to 8-bit and the dynamic range to about 48 dB; as a result, the level of the “quantizing noise” is increased;
- **Downsampling** or **Sample Divider**: simulates a reduction in sampling frequency, which causes a loss in “bandwidth”; as this is done without using antialiasing filters, the sound also deteriorates in quality due to "aliasing" artifacts;
- **Clip Mode** and **Level**: set level and type of clipping distortion;
- **Output** and **Mix**: set the output level and the balance between effect and dry signal.

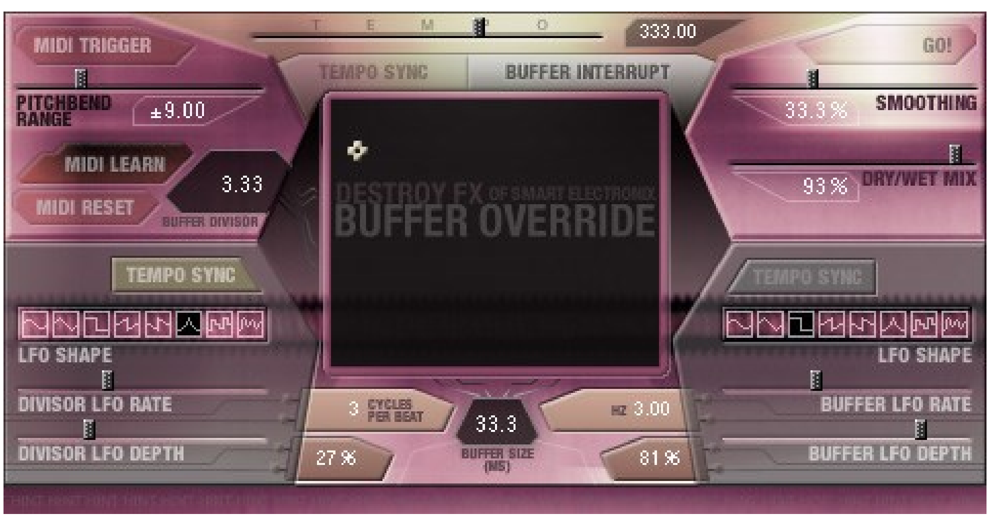

Figure 59: Destroy-FX Buffer Override, simulating a sound driver not working properly and producing several unique lo-fi effects (digital plugin)

3.6 PREAMPLIFIERS AND VIRTUAL CONSOLES

3.6.1 PREAMPLIFIERS

The **preamplifier** is one of the most important elements in the analogue signal chain – today more than ever, as it is one of the few remaining elements that *are* analogue.

A preamp can be super-clean, transparent and linear and offer virtually zero coloration to the signal (just boosting a mic or line input signal as required), or it can feature a *vintage design* and add some specific color and character to the sound (in form of subtle saturation, compression and change in frequency response).

Contrary to popular belief, a super-clean and linear preamp does can capture all the information from the original signal (nothing is "lost"), while the typical characteristic of vintage preamps can be emulated quite faithfully post-recording, through digital plugins. On the contrary, once the signal has been processed by a vintage design preamp, the coloration cannot be "removed" anymore, as it becomes part of the signal (some of the information has been "lost").

Figure 60: Audient ASP-880 8-Channel Mic/Line Preamp, with variable input impedance (hardware)

Figure 61: UAD NEVE 1073 Preamp and EQ, UAD UA 610-A Preamp and UAD UA 610 B Preamp (software plugins, emulating original hardware)

Figure 62: SlateDigital FG-73 (emulating Neve 1073 solid state preamp), and FG-76 (emulating Telefunken V76 tube preamp)

3.6.2 VIRTUAL CONSOLES

A **Virtual Console** is the accurate emulation of an analog console sound character, including frequency response, saturation, distortion, clipping, etc.

PARAMETERS

- **Input**: controls the input level; higher input levels cause more coloration.
- **Output**: controls the output level (used to compensate for the set input level).
- **Calibration**: defines the reference level (usually -18 dBFS). Saturation and Coloration becomes more pronounced when the signal is beyond the reference level.
- **Model**: defines the type of console being emulated (SSL, NEVE, Trident, Tube, etc.).
- **Drive**: higher values cause more coloration, saturation and eventually distortion and clipping.
- **Noise Reduction**: when turned ON, the electrical noise and *hiss* from the original console are NOT added to the console model (only the frequency response and saturation/distortion).

Figure 63: SlateDigital FGH-73 plugin (emulating NEVE 1073 preamp),
Virtual Channel and Virtual MixBuss (emulating SSL, NEVE, Trident and Tube consoles)

3.7 TAPE SIMULATION

Figure 64: SlateDigital Virtual Tape Machines (plugin, emulating real tape recorder hardware)

The sound of high quality studio **tape recorders** is often described as being *pleasant*, *round*, *rich*, *warm* and *fat*.

Most of these attributes are the result of:

- High frequency loss due to HF tape saturation (the magnetic particles hit a sort of “ceiling”, beyond which HF content cannot be committed to tape) and therefore more *bass-biased* sound (hence *fatter*, *warmer*)
- Smoothing of the transients (hence *rounder*)
- Mild saturation/distortion, adding pleasant harmonics to the original sound (hence *richer*)

Modern plugins attempt to faithfully recreate the sound attributes of a studio tape recorder, with the option to tune down or even turn off completely side effects such as *tape hiss* (HF noise) and *wow & flutter* (fluctuations in tape speed)

Figure 65: UAD ATR-102 tape recorder (plugin, emulating original tape recorder hardware)

3.8 CHANNEL STRIPS

A **channel strip** is a complete audio channel from an analog or digital console, including Gain, Dynamics (compressor, expander, gate), EQ and output fader.

Several plugin manufactures (including UAD, Waves, Brainworx, etc.) offer realistic emulations of original hardware consoles by NEVE, SSL, API, Trident and others. The emulation includes not only the faithful modeling of the dynamic and EQ sections, but also the subtler characteristic of the console (frequency response, saturation, noise, etc.).

If you open a channel strip on each audio channel of your DAW, you can mix almost like if you were on a real hardware console.

Some manufactures attempt to not only recreate faithfully the sound of a single console channel, but the sound of a complete console, emulating also the subtle differences in sound from channel to channel (due to tolerances in audio component manufacturing). One such example is the bx_console by Brainworx, emulating faithfully a Neve VX analog console.

Figure 66: Brainworx bx_console (emulating analog NEVE VXS console)

Figure 67: UAD API Vision Channel Strip (emulating API Vision console)

Figure 68: UAD Precision Channel Strip (inspired by analog hardware)

RECOMMENDED LITERATURE

- PIEPER, Frank: *das Effekte Praxisbuch* – CG Carstensen (ISBN 3-910098-16-9)
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