

# AUDITIVE GESTALTUNG

## Audio Effects

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## 1 Effect Processors

### 1.1 Hardware Effects

Hardware units can use discrete analogue components or be DSP-based; some high-end analogue units still offer a sound quality that is not matched by any DSP-based unit or software plugin, for example state-of-the-art EQs, Compressors, guitar preamps, etc. by brands such as GM Labs, Avalon Design, Focusrite, etc.

DSP-based units might use similar or even exactly the same algorithms than similar software plugins (for example: Waves L2 Ultramaximizer is available as hardware unit or software plugin); they require an additional AD/DA stage to process analogue signals (which will introduce about 2 ms latency) and/or a digital I/O interface for use within Hard Disk Recording systems.

Some effects such as Hall, Ambience, Reverb, Delay, etc. are realized nowadays exclusively using DSP-based units (for example, Lexicon, Quantec).

### 1.2 Software Plugin Effects

Software plugins can be “native” or “DSP-based”.

Native plugins use algorithms that run directly on the main CPU of the host computer; when upgrading the computer CPU, the processing power increases accordingly.

Typical native plugin standards are:

- Windows: Direct-X, VST (Virtual Studio Technology), ReWire 2
- MacOS X: AU (Audio Units), VST, ReWire 2, Logic native effects

Application specific formats:

- Emagic Logic: Logic native effects
- Pro Tools LE: RTAS (Real Time Audio Suite)
- MOTU Digital Performer: MAS (MOTU Audio System)

DSP-based plugins use algorithms that require specific DSP processors (typically Motorola DSPs, like in Pro Tools and TC Power Core, or Analog Devices SHARK DSPs, like in Creamware SCOPE). They usually offer greater stability and lower latency than most native systems, but at a higher cost, as they require special DSP-cards to run.

When using DSP-based plugins, the processing power does not depend from the host computer CPU (except for the graphical interface). However, both Native and DSP engines can be used together, like for example when using Pro Tools with TDM plugins + RTAS native plugins. Generally, on a DSP based system to increase the processing power it is necessary to purchase additional DSP-cards.

Typical DSP-based plugin standards are:

- Pro Tools TDM PT Mix, PT HD and Logic Pro + PT systems
- Crewamware SCOPE Pulsar and Scope or VST wrapper (XTX Mode)
- Universal Audio UAD1 available to the host as VST/AU plugins
- TC Electronics PowerCore available to the host as VST/AU plugins

Nowadays the processing power of modern computer CPUs (such as Intel Core2Duo, Core2Quad, dual and quad core Xeon; AMD Athlon 64 or Opteron) rivals or exceeds that of mid-sized DSP based systems such as Pro Tools TDM or SCOPE.

### 1.3 Hardware vs. Software Effects Comparison

From the conceptual point of view, it does not really matter if a specific effect is using an external hardware unit (be it analogue components or DSP based) or a software plugin: that means you will have to set parameters in a similar way, and you may get similar results with both. However there are differences in sound as well as general advantages and disadvantages in usage. Also, while some effects can be realized both with analogue components or DSP based units, there are some that can only be realized using DSP (for example, time stretching), and some where analogue processing is preferred.

<b>Hardware vs. Software Effects Comparison Table</b>	
<b>Hardware Effects</b>	<b>Software Plugin Effects</b>
Professional hardware effect units (such as Lexicon, Quantec, Avalon Design, GM Labs, Focusrite, etc.) usually deliver a higher sound quality than standard software plugins, however at a higher price.	Plugins are comparatively cheap and extremely versatile; software plugins make it possible for the first time for small budget studios to complete a professional production in the digital domain using a so called "virtual

	studio environment”.
Hardware effects can be used independently from host applications (HDR sequencers); they offer higher reliability (for example for live concerts) as they do not depend on host software and computers (even DSP-based hardware units hardly ever “crash”, as they are based on a specially designed, EPROM based operating system).	Plugins are mostly used within a host application (HDR sequencer) on a computer and hence “depend” on it; if the host application crashes, so do the plugins; for this reason, they are rarely used for live concerts.
Each hardware effect unit can be used only once; to open more than an instance of an effect, more hardware units are required; alternatively, the track must be “bounced” with the effect on a new track (“print to tape”), to free up the effect for a new task.	It is always possible to open 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; so it is possible to use the same plugin on several different tracks at the same time, using different settings, all accessible and modifiable in real time.
Analog effects have absolutely 0 latency, most hardware digital ones have just AD/DA latency (less than 2 ms).	When using plugins, the latency depends on the audio subsystem and the driver settings of the host software. Latencies under 2 m/s I/O are rare (for example, Pro Tools systems, or very fast native systems based on RME cards), while 20-40 I/O ms are rather common.
Settings can rarely be saved (especially on analogue units) and in any case are not recalled automatically when loading a project in the HDR 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 host application “project file”. Additionally, for most plugin types (VST, AU, TDM etc.) it is possible to control every parameter in real time using automation from the host application; this makes it possible to program complex real time changes and “morphing” between different effect and sounds settings.
Until a few years ago, there was no real replacement for hardware based reverb effects, such as Lexicon, Quantec, etc. Still nowadays, they are the preferred reverb effects in pop productions.	Recently software reverb effects, such as Waves IR1, Altiverb, Emagic Space Designer, etc. have matched or in some cases surpassed the quality offered by hardware reverb effects.
Hardware effect units offer direct physical control on most or all parameters: especially in analogue hardware effects, each function has its own switch, knob or fader, hence more intuitive and comfortable to use than software plugins using a mouse-interface. Generally, hardware effects motivate to work using the ears and “feeling”, and not using the eyes and “thinking” (you tend to listen to sound changes instead of looking at plugin settings on screen).	Plugins offer direct visualization of all parameters (for example, EQ curves, compression ratios, etc.); however editing parameters with the mouse is not always comfortable, and just looking at parameter visualizations there is the danger of judging more with the eye, and less with the ear. Using additional MIDI controllers (such as the Mackie Control, or Digidesign Pro Control) can greatly improve the usability of plugin effects as well as plugin synthesizers.
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 cheap to add plugins, there is sometimes the tendency to use too many effects in a mix.

## 2 Effect 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, EQs, filters and dynamics (compressor, gate, 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 effected signal to the dry one; note: chorus, flanger etc. usually work best at around 50% dry + 50% effect settings;
- there are of course exceptions! For example, you can send all the instruments of a drum-kit (and if you want also the bass) to the same aux, then compress the hell out of it and add the output to the original uncompressed signal - sounds terrific!
- 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 severe sound artifacts)
- especially additional VST or Direct-X plugins in programs like Logic or Cubase do not necessarily "know" whether they are used as insert or aux send/return.

### 2.2 Using an external effect unit in a DAW

- you can of course use the aux-return (as standard) to re-insert the signal from an external effect unit (for example a reverb) in the mixer, but it might be more convenient to use a free couple of mono-channel inputs, because:
  - 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 EQ (works great on reverbs and delays!);
  - you can also send again the FX signal to another effect processor with another aux send (for example send to aux 1 = delay, and send again some of the delay return signal to aux 2 = reverb);

## 3 Effect Description and Parameters

### 3.1 Filters and EQs

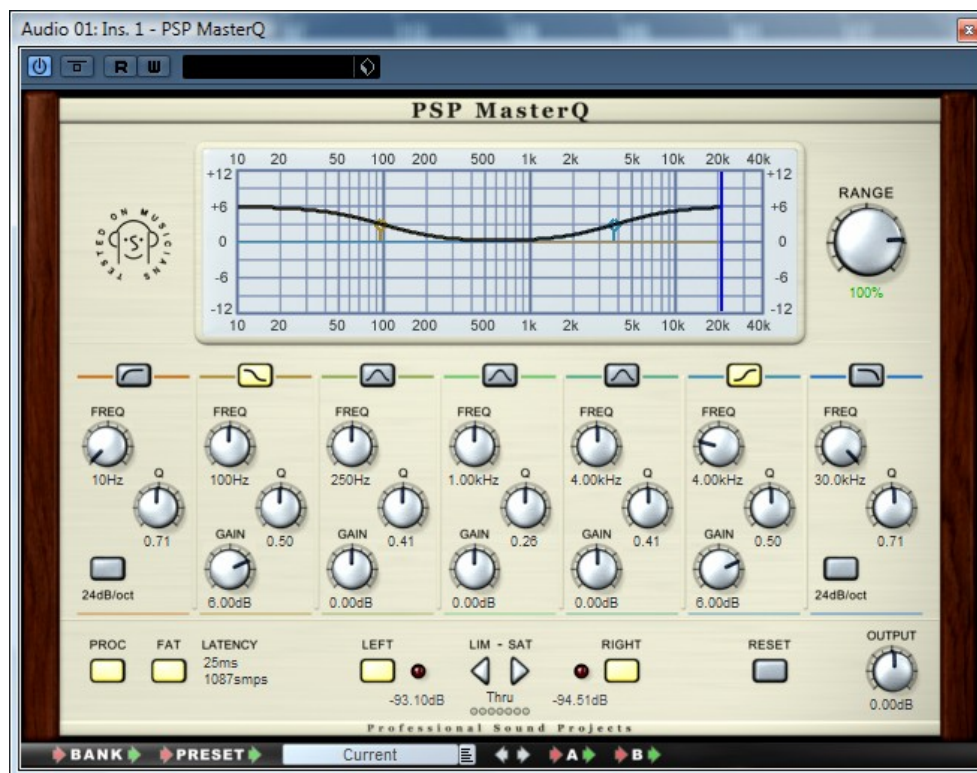
#### 3.1.1 Shelving EQ

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 amplifier or car stereo.

##### Parameters:

- usually only the gain (boost or cut) control is available; control frequencies on hardware mixers are typically set at 80 or 100 Hz for the low shelving EQ and 10 or 12 kHz for the high shelving EQ;
- in some digital versions of the shelving EQ (for example Cubase 4 channel EQ, PSP Master Q, etc.) also the frequency and slope can be adjusted, allowing a greater flexibility.

If an instrument just sounds too “thin”, “dark”, or “boomy”, it is often enough to adjust it with a shelving EQ, which will not change the character of the sound in the way a peak EQ would.



Low and High Shelving EQ (PSP Master Q)

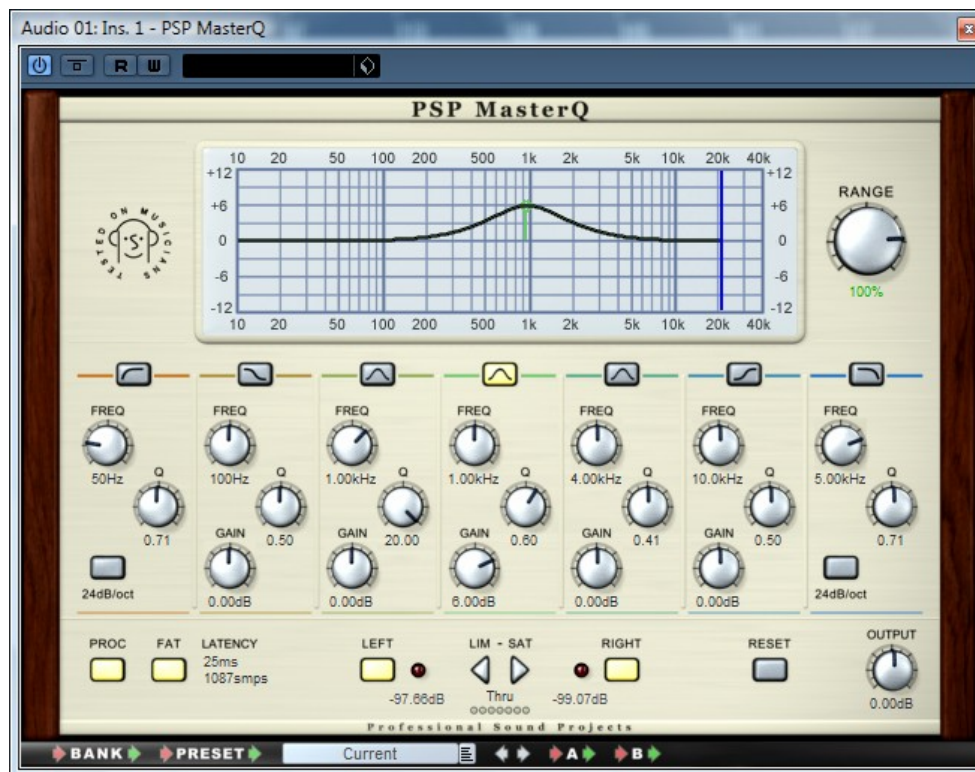
### 3.1.2 Peak (Bell) EQ

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:

- a semi-parametric peak EQ has only frequency and gain controls;
- a full-parametric peak EQ has frequency, gain and bandwidth (Q) controls;
- some mixers have separate "low-mid" and "hi-mid" peak EQs, that cover different frequency ranges.

Tip: when you look for formants in an instrument, set the peak filter to "boost", medium Q, and sweep around searching the desired frequency until you can spot it: then you can set the filter to "cut", if you wish to remove the undesired formant, or to moderate "boost", if you want the instrument to come better through the mix without altering the channel volume.



Full Parametric Peak EQ (PSP Master Q)

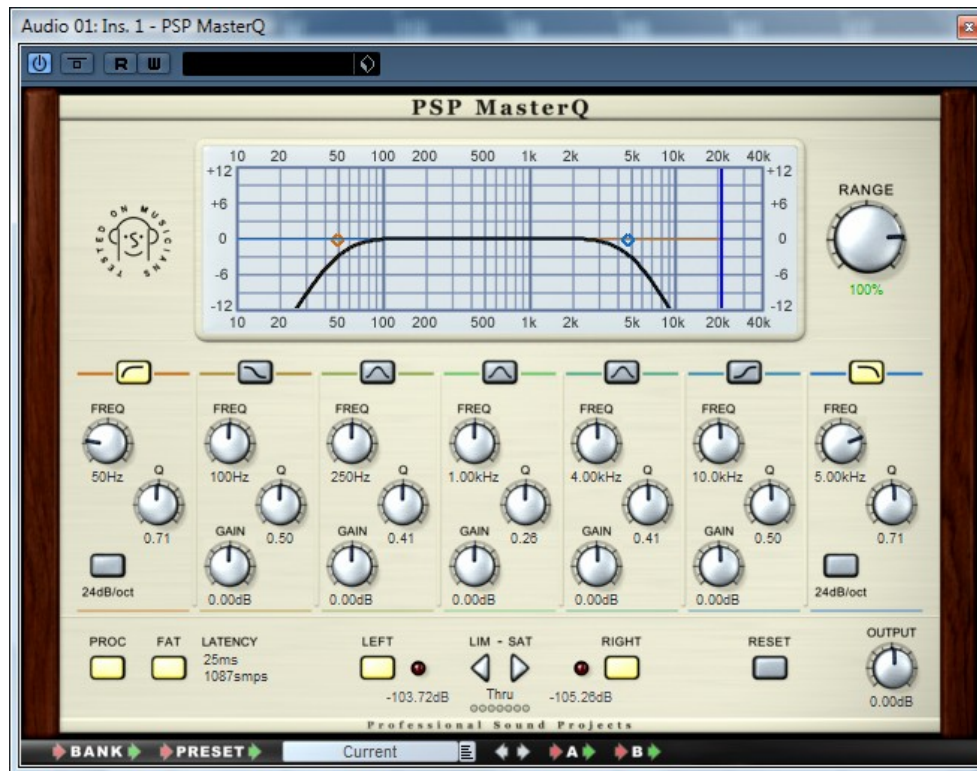
### 3.1.3 Low Cut / High Pass and High Cut / Low pass filters

A low cut filter can be used to eliminate low frequency noise or vibrations (for example, mechanical noise transmitted through the floor to the microphone stand).

A high cut filter can be used on bass range instruments (such as bass drum or bass guitar), in order to remove high frequency noise ("hiss"); it could also be used to "repair" a take where clipping distortion occurred, provided there is not much energy in the high freq. range; 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:**

- frequency, sometimes slope (= flankensteilheit);
- resonance (synthesizer type) control may also be available on some filters.



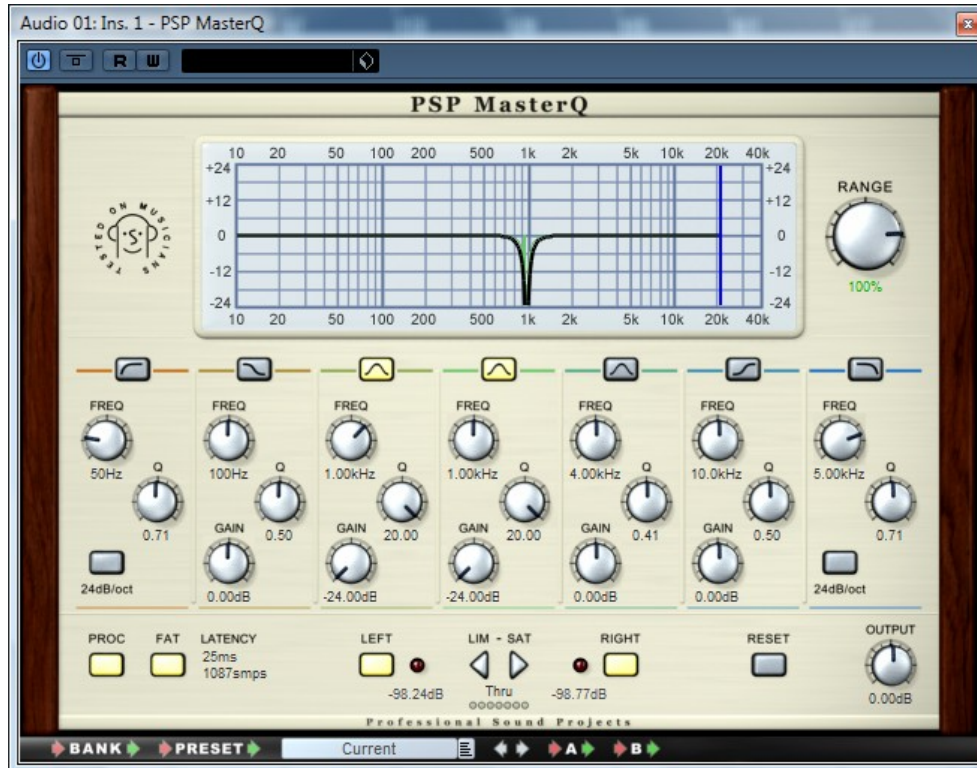
Low Cut and High Cut Filters (PSP Master Q)

### 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.

Parameters:

- 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.



Simulated Notch Filter using 2 Peak EQs with narrow Q setting (PSP Master Q)

## 3.2 Dynamic Processors

### 3.2.1 Compressor

A compressor can be used to:

- control volume changes (for example on vocals, to help "cut through the mix", making quiet parts of the signal louder)
- 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 radically change the character of the sound (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 squashing the dynamics (works nice on drums).

#### Parameters:

- 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 1dB 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; more moderate on drums, unless you are looking for a special sound;
- soft or hard knee: hard knee is more efficient, but can make compression hearable (pumping); soft knee works more "musically", as it starts compressing lightly before the threshold level and reaches the set ratio much after that, but is not as efficient at compressing sudden peaks (clipping might occur in some cases);
- attack time: how fast does the compressor reacts after the signal has crossed the threshold (usually set shorter for percussive sounds, however "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 ms;
- 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;
- make-up gain, or output level: boosts the overall signal level and is used to compensate for the gain "lost" through the compression.



Compressor (Cubase 4)

### 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 for:

- avoiding signal “clipping” – usually the last effect in the master output (for example in mastering);
  - reducing in an undetectable way the level of short signal peaks and make the whole track louder without losing perceived dynamics like with a compressor;
- Beware of too short attack/release times, as they can create distortion.

Parameters:

- input: it basically control the limiting amount (higher input levels cause more limiting);
- release time: like in a compressor, only usually much shorter; sometimes “auto” release is available.



Limiter (UAD Precision Limiter)

### 3.2.3 Transient designer

It is a kind of combination of upward and downward compressor and expander; it can be used to change the shape and character of percussive material such as drums and percussion loops. (as a side-effect. it can be also used to give the impression that there is less or more ambience)



Transient Designer (Digital Fish Phones Dominion)

### 3.2.4 De-esser

It is a special kind of compressor that reacts only to the frequencies in the specific range of "S", "T" and other consonants (normally, 5 to 7 kHz).

The "single band" versions compress the complete signal, while "multiband" versions only compress the selected frequency range, leaving the rest of the signal unaffected.

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:

- freq: on which frequency should the control range be centered;
- side chain: what kind of filter is used on the "control" signal (low cut or peak)
- threshold: from which level should the signal 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 match the right frequencies).



DeEsser (Waves)

### 3.2.5 Expander/Gate

It is used to remove undesired, low dynamic level parts of a signal; example: to reduce "leaking" from the different microphones when recording a drum-kit, or to remove low level noise between vocal takes.

#### 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 -∞ when closed, while an expander reduces the signal level by a set ratio (similar to a compressor, but inverted);
- attack time: how fast does the gate/expander opens after a signal reaches the threshold;
- hold time: how long does the gate/expander remain fully open;

- release time: how long does it take for the gate/expander to gradually close/reduce the gain, after the signal has dropped under the threshold level again;
- high cut and low cut in the "side chain": to 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).



Expander (Cubase 4)



Gate (Cubase 4)

### 3.3 Modulation Effects

#### 3.3.1 Chorus, Flanger, Phaser, Rotary Speaker (Leslie)

Modulation effects can be used to "thicken up" and add motion to sounds that are flat, static and / or uninteresting; for example, a simple synth pad with just one oscillator per voice sounds much nicer 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" 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 greatly change the original sound.

The principle behind Chorus and Flanger is similar: the signal is delayed by one or more delay lines, then modulated in pitch by a LFO (low frequency oscillator). The effect signal is mixed with the dry signal about 50/50, producing typical coloration due to "Comb Filtering".

Chorus uses longer delay times (typically, 5-40 ms), while Flanger uses shorter delay times (typically, less than 5 ms) and has the additional parameter "feedback", which "feeds" part of the effect signal back into the effect unit, causing additional coloration. The Phaser uses no delay, just phase shift (it is like a very short delay, that affects just the waveform phase), and hence works best on drums/percs.

##### Parameters:

- delay (chorus and flanger only): how much are the modulated lines delayed from the dry signal; the
- speed or rate: the speed of the LFO(s) that controls the pitch modulation; it can be synced to the song tempo
- depth or width (chorus and flanger only): how much does the signal modulates above and under the reference pitch;
- feedback (flanger and phaser only): sends parts of the effect signal back into the effect unit, creating the typical flanger and phaser "sweep", and "liquid" sounds;
- phase: defines the phase of the delayed lines;
- color, sweep floor and ceiling (on phaser only): control the color of the phaser effect;
- mix: defines the balance between dry and effect signal.



Chorus and Flanger (Cubase 4)



Phaser (Cubase 4)



Rotary Speaker (Cubase 4)

### 3.4 Ambience Effects

#### Reverb, Echo, Delay, etc.

Ambience effects are used to add “depth of field”, dimension and space to a mix, but can also be used simply as “special effects”. Digital reverbs often simulate real acoustic spaces, such as rooms, halls and churches, but can also be used to simulate artificial devices, such as “plate” or “spring” reverbs.

Delays produce a more or less artificial type of “echo” reflections, with variable repetition patterns, amount and times. There are several different type of delays, such as “stereo delay”, “ping pong delay”, “modulation delay”, “tape delay”, etc.

#### 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 and low cut filters: affect the bandwidth in the high and low frequency range; “realistic” reverbs should already be cut at around 4-5 kHz;
- low and high damping, or “material”: 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.;
- 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.



Reverb (Cubase 5 RoomWorks)



Reverb (UAD RealVerb Pro)

Some tips about using reverbs in a mix:

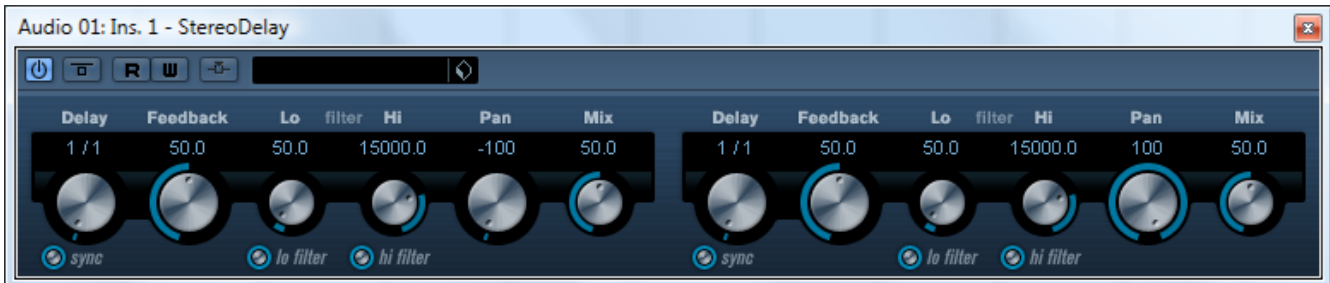
- it is recommended not to use too much reverb, as it can make the track sound "muddy" (make the vocals or instruments unclear), or just sound rather "old fashioned" (like some late seventies / early eighties productions, that used a lot of reverb).
- avoid using many long reverbs: sometimes shorter 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.

**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 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 here) might have additional filters (with cutoff and resonance) to change the character of the delay, or modulation parameters to create chorus and flange-type effects.



Stereo Delay (PSP 84)



Stereo Delay (Steinberg Cubase 4)

## 3.5 Distortion Effects

### 3.5.1 Overdrive, Distortion

Overdrive and Distortion plugins can be used to simulate the saturation of analogue tube guitar amplifiers and speaker cabinets; however, “real” guitar amps have a very distinctive character that cannot be easily reproduced using just digital plugins.

#### 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: simulated stereo width;
- mix: balance between distortion and “clean” signal.



Distortion (Cubase 4)



Speaker Simulator including Compressor, Distortion, Phaser, Tremolo, Modulation Filter, Delay and Echo (UAD Nigel)

### 3.5.2 Bit-Crusher, etc.

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.



Overdrive and Bit Crusher (Cubase 4)

**Bibliography / Further Reading:**

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**Website:**

- [www.digitalnaturalsound.com](http://www.digitalnaturalsound.com) or [www.dns-studios.com](http://www.dns-studios.com)  
> Medialab & MMA > Auditive Gestaltung