About this help file

This Guide file contains information about the new Logic Audio Plug-Ins. Not all Plug-Ins described in this file are available in Logic Audio Gold or Silver. However, we included them to fully inform you about the range of the Emagic Plug-Ins.

Control Features

We developed new control features for Version 4.0 plug-ins; the sliders are pretty much self-explanatory but you have two options for manipulating the new rotary knobs:

1.) Click on the center and drag the mouse up and down. This option lets you access all available value increments.

2.) Click on the knob and move the mouse in a circular motion. Here the program responds to the angle of rotation. You'll find it easier to fine-tune values when the circular motion has a greater radius - in other words, drag it in bigger circles.

Fat EQ

Similar to the bass and treble controls on a car stereo, equalizers let you shape or voice the tone of a signal. The difference is, of course, that an EQ gives you much more precise and powerful options for influencing a sonic image.

The extremely high-quality Fat EQ offers up to 5 fully parametric bands - buttons 1 through 5 let you activate these individually; inactive bands will otherwise drain your computer's resources.
The icons above the graphic display let you determine whether Band 1 responds like a high-pass filter or a low shelving EQ. Similarly, for Band 5 you can switch back and forth between a low-pass filter and a high shelving EQ. Bands 2 and 4 can be switched from their normal operating mode as fully parametric bell EQs to low or high shelving EQs. You're stuck with the center band (No. 3) as is - it always operates as a fully parametric bell EQ.

Go to the area directly below the graphic display depicting the frequency response curve to select the frequency for the individual bands. You don't have to click on the arrows to change values in increments. Simply click on the number and change the value via your mouse. You'll be able to hear and thus audition a frequency better if you rotate the Cut/Boost knob located below it clockwise to turn it up. The same holds true for any annoying frequency that you want to attenuate. Once you've located the frequency that you're hunting for, you can back off the Cut/Boost knob and set it to the desired value. Use the Q (quality) parameter located in the bottom display to determine to which extent the band influences neighboring frequencies. At low Q values, the EQ influences a wider frequency band, and at high Q values, the effect of the EQ band is limited to a very narrow frequency range. Please bear in mind that your perception of an attenuated or boosted frequency depends on the Q factor: If you're working with a narrow frequency band, generally you'll have to cut or boost it more drastically to notice a difference.

General hints about the Controls

Noise Gate

Ordinarily, a noise gate suppresses unwanted noise that may become audible during a lull in the signal. You can, however, also use it as a creative sound-sculpting tool.
Here's the basic principle behind a noise gate: Signals that lie above the threshold are allowed to pass unimpeded - hence the term "open" gate. Anything below the defined threshold (e.g. background noise, crosstalk from other signal sources, etc) is fully muted, which audiophiles call a "closed" gate. In other words, the Threshold slider determines the lowest level that a signal must have to open the gate - it separates the wanted or useful signal from the unwanted or noise signal.

Actually, an open gate lets a mix of the wanted and unwanted signals pass. Fortunately, we all benefit from a nifty effect called “masking” - the noise signal is camouflaged by the louder useful signal so that the listener is no longer able to perceive the background noise. You'll find that in practice this generally does the trick - the undesirable noise is rendered inaudible.

The Reduction slider lets you control the intensity of noise suppression. As a rule, you should set it to the lowest value and leave it there to make sure that the gate closely completely. If you prefer, you can dial in other values so that the noise signal is reduced less dramatically. Alternatively, you can even boost the signal by up to 20 dB.

The three rotary knobs at the top influence the dynamic response of the noise gate. If you want the gate to open extremely quickly, say, for percussive signals such as drums, set the Attack knob to the lowest value by turning it as far as it will go counter-clockwise. If the signal fades in a bit more softly, as is the case with string pads and the like, a noise gate that opens too quickly can wreak havoc with the signal, causing it to sound unnatural. For this type of sonic scenario, set the Attack knob so that the gate emulates the attack of the original signal. Much the same holds true for the Release phase of signals. When you're working with signals that fade out gradually or have longer reverb tails, you should turn the Release knob up so that the noise gate allows the signal to fade naturally. The Hold knob determines the minimum amount of time that you want the gate to stay open. This knob lets you avoid the dreaded "chattering" effect caused by a rapidly opening and closing noise gate. The Hysteresis slider gives you another option for avoiding chatter without dialing in a minimum Hold time. Let's back up a bit for a brief explanation: Noise gates often begin chattering when - during the attack or release phase - the level of a signal fluctuates slightly but very rapidly. Instead of clearly exceeding or falling short of the threshold value, the level hovers around the threshold instead. The noise gate then rapidly switches on and off to compensate, producing the undesirable chattering effect. If, however, you were able to tell the noise gate to open at the determined threshold level and remain open until the level drops below another lower predefined threshold, you'd be able to avoid chatter as long as the sonic window formed by these two thresholds is large enough to contain the fluctuating level of the signal. This is exactly what this feature enables you to do - the value determined by the Hysteresis slider is actually the difference between the Threshold values that open and close the gate. This value is always negative and, generally, -6dB is a good place to start.
If you're dealing with audio material featuring extremely sensitive transients or attack phases that are critical to the overall sound, you may find it beneficial to have the noise gate open up a tad before the useful signal fades in. This is what the Lookahead slider is designed for. The program analyses the signal level ahead of time and anticipates the point at which it can open the gate before the signal actually attains the threshold value. When you chose to use this feature, please make sure you set the Attack, Hold or Hysteresis controls to appropriate values.

As a rule, Lookahead only works with digital noise gates. Actually, at this point in time, the feature only works properly - i.e. without causing undesirable latency of the useful signal - in a native processing environment.

When you're working with noise gates, you'll run across scenarios where the useful signal and the noise signal have levels that are near enough to be perceived as identical. A typical example is the crosstalk of

a hi-hat - its signal tends to bleed into the snare drum track when you're recording a drum set. If you're using a noise gate to isolate the snare, in many cases the hi-hat will also open the gate. To avoid this effect, the on-board noise gate ships with sidechain filters.
A sidechain is pretty much what it sounds like - the signal is tapped, routed out and analyzed to find out whether or not it falls below or exceeds the threshold. In other words, the sidechain signal is used to control the noise gate.

When you press and hold the Monitor button, you can audition the sidechain signal. Then you can set the filters so that only those frequencies at which the useful signal is particularly loud are allowed to pass. For our example, we'll use a high-cut filter that only allows the bottom end and mids of the snare to pass and cuts the higher frequencies of the hi-hat. Now when you switch sidechain monitoring off, it will be much easier to dial in a suitable threshold. This will be a value that is only exceeded by the level of the louder useful signal - in our example the frequencies that make up the snare's fundamental tone. Put simply, the noise gate only allows the sound of the snare to pass. Should the need arise, you can follow much the same procedure to isolate a kick or snare drum within an entire mixdown.

General hints about the Controls

Compressor

A compressor tightens up the dynamics of a signal. This means that the difference in levels between loud and soft passages is reduced. With the peak remaining constant, the overall loudness - the perceived volume - of a track is increased. Next to an EQ, a compressor is your most valuable sound-shaping tool when you're mixing. A compressor is a universal effect, it has a virtually unlimited range of applications. You should definitely exploit it for vocal tracks but a compressor can also often work wonders for master signals. When you use a compressor, be sure to route the entire signal through it by inserting into tracks. It is only used in a bus when you want to compress a group of tracks (e.g. drums) simultaneously, by the same amount. Again, these should be routed to the bus in their entirety, as opposed to using Send knobs to route just part of each signal to the bus.

The Emagic Compressor was designed to emulate the response of the finest analog compressors. The principle behind it is as follows: When a signal exceeds the defined threshold level, the compressor actually alters the response so that is no longer linear. Instead, all levels that exceed the threshold are attenuated by the value that you set via the Ratio slider. A ratio of 4:1 means that an incoming level that lies 4 dB above the threshold is dampened so that it comes out the other end of the compressor with a level just 1 dB above the threshold. On the flip side, if you route in a signal that is loud enough to double the output level of the compressor (+ 6dB), this input signal would have to have a level 24 dB greater than the threshold level. This tells us that a compression ratio of 4:1 is a fairly drastic manipulation of the original signal's dynamics. Since the compressor lowers levels, the volume of its output signal is normally lower than that of the input signal. To compensate for this decrease in levels, the output of the compressor is equipped with a Gain slider. Auto Gain automatically sets the level of amplification to a value equivalent to the "sum of the threshold value minus the threshold value divided by the ratio" or put less confusingly T - (T/R). This function ensures that a normalized input signal is amplified so that the output signal is also normalized, regardless of the values that you set for threshold and ratio - provided you are dealing with relatively static signals. Use the Attack and Release knobs to shape the dynamic response of the compressor. Attack determines the amount of time it takes for the compressor to react to signals that exceed the threshold. At higher values, the compressor does not fully dampen a signal until after it runs through its attack phase. This type of setting ensures the original attack, for example the sound of a pick or finger striking a guitar string, remains intact or clearly audible. If, on the other hand, you want to maximize the level of a master signal, set the Attack knob to low values; i.e. so that the compressor responds more swiftly. Release determines the amount of time it takes for the compressor to stop dampening louder passages once the signal level falls below the threshold. If the compressor generates an ugly pumping sound, adjust the Release knob accordingly.

When you have configured a compressor so that it dampens the signal at and higher than the threshold value in the predetermined ratio, while the level just below the threshold is routed through at a 1:1 ratio, an audiophile would say that the compressor is working with a "hard knee." However, in many cases you'll come up with a better sounding track by using a more gradual transition from the 1:1 ratio below the threshold to the ratio that you entered for levels above the threshold. Here the characteristic curve is not as radical - it rises gradually from the bottom left to the top right, as seen in the graphic display. This type of compressor characteristic is called a "soft knee." The Knee slider lets you dial in anything from hard to soft, and provides a useful number of increments. This wide range of options gives you the tools to shape the sound to fit the sonic scenario, depending on if you want to radically maximize loudness with absolutely no regard for the original dynamics (hard) or are going for the more musical compression that acoustic recordings thrive on (soft). Keep in mind that Knee controls solely the characteristics of the compression, not its intensity; use the Threshold and Ratio sliders for this purpose.

Incidentally, the Gain Reduction Meter indicates the intensity of compression with which you're tightening up the original signal. This feature is a great help, particularly for the less experienced mix maestro. Keep an eye on it to make sure you're not doing an overkill on the compression.

When the compressor has to decide whether or not the level exceeds the threshold (or for soft knee compression, if the level is getting close to the threshold), it can analyze either the peak or RMS level. The latter value is a better indication of how we perceive loudness. When you use the compressor primarily as a limiter, select the Peak button. When you're compressing individual signals, use of the RMS button will often deliver better, more musical results.

If you activate Auto Gain and RMS simultaneously, the signal may be saturated! If you hear any distortion, switch Auto Gain off and enter a suitable gain level manually.
Despite all of these handy tips for tweaking sounds, you should always keep one thing mind -there are no hard and fast rules. Whether or not something is to your liking is determined solely by your taste and your ears.

General hints about the Controls

Expander

The Expander works along the lines of the Compressor, with one huge difference - it increases, rather than reduces, the dynamic range above the threshold. The Ratio slider features a value range of 1:1 to 0.5:1. This means that the Emagic Expander is a genuine "upward expander" (as opposed to a "downward" expander that increases the dynamic range below the threshold). You can use this effect to emphasize the transients of highly compressed signals. This spices up the sonic image, making it sound livelier and fresher. Please bear in mind that you will perceive the signal as being softer even when the peak level remains the same. In other words, the expander decreases loudness. If you manipulate the dynamics of a signal fairly radically (depending on the threshold and ratio values), you'll find that you'll have to back off the level via the Gain slider to avoid distortion. In most cases, Auto Gain will take care of this for you.

General hints about the Controls

Enveloper

The Enveloper is an unusual tool that lets you shape transients, i.e. the attack and release phases of signals. No other type of dynamic effect (compressor, expander) can achieve similar results - and they can be quite impressive indeed.

The most important Enveloper controls are the two Gain sliders that govern Attack (left) and Release (right). In the center position, the signal remains unprocessed. If you turn the gain up, the attack or release phase is emphasized, if you turn it down, the corresponding phase is toned down. For example, boosting the attack phase lends a drum sound more snap or amplifies the sound of a guitar string being plucked or picked. When you cut the attack, percussive signals fade in more softly. You can also mute the attack so that is virtually inaudible. This lets you come up with all kinds of interesting manipulations. Another handy application for this is maintaining friendships - it allows you to mask the poor timing of accompanying instruments rather than tell your pals that they have all the groove of a horde of accountants at the office Christmas party.

Emphasizing the release phase also boosts the amount of reverb. Conversely, when you tone down the release phase, tracks originally drenched in reverb end up sounding drier. This effect is particularly useful when you're working with drumloops, but of course there are many more application options. Let your imagine be your guide.

When using the Expander, you should set the threshold to the minimum value and leave it there. Only when you seriously crank the release phase, thus boosting the noise level of the original recording to an excruciating level, should you turn the Threshold slider up a tad to limit the Enveloper so that it influences the useful signal only.

Drastic boosting or cutting of the release or attack phase may change the overall level of the signal. The Out Level slider allows you to compensate for this effect manually.
The Time parameters for the attack and release phase (2 knobs below the graphic display) enable you to access the time-based intervals that the program interprets as the attack and release phases. Generally, you'll find values of some 20ms (attack) and 1500ms (release) are fine for starters. Adjust them according to the type of signal that you're processing.

Similar to its counterpart on the noise gate, the Lookahead slider lets you enter values that tell the Enveloper to anticipate what the signal will be up to in the very near future. Normally you won't need this feature, except possibly for signals with extremely sensitive transients. If you do decide to employ this feature, you may have to adjust the attack time accordingly.

To give you better insight into true nature of the Enveloper, here's a quick look at how it works: It is equipped with two internal envelope followers. One follows the amplitude of the input signal directly, whereas the other follows all changes generated by the variable delays (individually adjustable for attack and release). The difference between the two envelope followers is used to boost or cut the original signal by way of the corresponding Gain sliders (also individually adjustable for attack and release).
In contrast to a compressor or expander, the Enveloper operates independently of the absolute level of the input signal - provided the Threshold slider is set to the lowest possible value).

Phase Gain

This plug-in enables you to influence the level and phase of the signal.
"Norm" unchanged Phase
"Inv" inverted Phase

The stereo version of this plug-in lets you invert the phase position of the two stereo sides separately:

"L- R+" left inverted, right unchanged
"L+ R-" right inverted, left unchanged

When you invert the phase of a signal, it sounds identical to the original. Only when the signal is heard in conjunction with other signals does phase inversion have an audible effect. For example, if you mic a snare drum from the bottom as well as from the top, you should invert the phase of the bottom microphone's signal so that it is in phase with the top mic signal. Keep in mind that the phase of condenser microphones is generally opposite that of dynamic microphones. If you are using one of each of these two breeds of microphone to record a snare, you won't have to invert the phase of one of them. On the other hand, if you mic a sound source from the same direction using two different types of microphones, you will have to invert the phase of one of the mics. This is done to avoid cancellations when the signals are mixed together.

Influencing levels via "P-Gain makes sense when you're working with automated tracks during post-processing and you want to quickly adjust master levels. This could be the case, for example, when you've inserted an additional plug-in that does not feature a dedicated gain control or you want to change the basic level of a track for a remix version.

General hints about the Controls

Distortion

This distortion effect simulates the lo-fi dirt generated by a bipolar transistor. Move the Drive slider up to increasingly saturate the transistor. Generally, the distortion created by the plug-in tends to increase the signal level, an effect that you can counter via the Output slider. The Tone knob lets you filter the harmonics-laden distortion signal, which then delivers a somewhat less grating, softer tone. The Emagic Eye is watching - it visually represents the settings of the Drive and Tone parameters.

General hints about the Controls

Overdrive

The Overdrive effect emulates the distortion of a field-effect transistor (FET). When saturated, FETs generate warmer-sounding distortion than bipolar transistors. Here too the Drive slider pushes the transistor over the edge and into overdrive. Generally, the distortion created by the plug-in tends to increase signal levels, an effect that you can compensate for via the Output slider. The Tone knob lets you filter the harmonics-laden distortion signal, which delivers an even warmer sound. Once again, the Emagic Eye visually represents the settings of the Drive and Tone parameters.

General hints about the Controls

BitCrusher

BitCrusher is the ultimate digital distortion box. You can do all kinds of wild stuff with it, such as revive the 8-bit sound of the pioneering days of digital audio, create artificial aliasing by dividing the sample rate, or distort signals so radically that they are rendered unrecognizable. Warning: The BitCrusher can damage your hearing when operated at high volumes!

The Drive slider lets you boost the level at the input of the BitCrusher. Please note that this tends to excite the clipping stage located at the output of the BitCrusher as well.

The Resolution knob lets you reduce the resolution from 24 bits down to 1 bit.

The number of bits is always an exponent of two. The range of available values is equivalent to the exponents of two that a given sample rate can handle. For example, whereas 65,536 different values are possible for 16 bits, at 8 bits, you're left with just 256. The sonic image becomes ever more ragged as the values decrease because the number of sampling errors increases, thus generating more distortion. At extremely low bit resolutions, the amount of distortion can be greater than the level of the useful signal.
The Downsampling slider lets you lower the sample rate. For example, at a value of 2, the original 44.1 kHz signal is sampled at a rate of just 22.05 kHz. With a factor of 10, the rate is knocked all the way down to 4.41 kHz.

The theory of digital signal processing states that, as a rule, frequency data up to half the value of the sample rate remains intact. Any frequencies above this value are lost. However, the BitCrusher is not equipped with a low-pass filter designed to filter frequencies that lie above the value equivalent to half the sample rate. Indeed, you could use a simple low-pass filter for this purpose. Nonetheless, there is method behind the madness - it is precisely the distortion generated by these alias tones (hence the name aliasing) that makes this effect so unusual. "What does it sound like," you ask? Something similar to a short-wave radio with a station dialed in just a tad off the mark required for decent reception.
The Clip Level slider lets you define at which point below the normal threshold you want the signal to start clipping. The Mode buttons are used to determine whether the signal peaks that exceed the clip level are either “Folded,” “Cut,” or “Displaced” (check out the graphics on the buttons and the resulting waveform in the display). The kind of clipping that occurs in standard digital systems is usually closest to that of the center mode (Cut). Internal distortion may generate clipping similar to the types generated by the other two modes.

General hints about the Controls

Tape Delay

The Emagic Tape Delay is what back in the pioneering days of electronic audio was called an echo device. The first good news is it's variable in musical increments. It is equipped with a high- and low-pass filter in the feedback circuit as well as a circuit that simulates tape saturation effects. This plug-in is ideal for the dub delays invented by Jamaican toast masters and used in many styles of music today.
Here you won't be wasting your time computing delays and typing in the values in milliseconds. Instead, all you have to is enter the desired note value directly. Simply click on the desired button for half, quarter, eighth or sixteenth notes.

When it is computing the desired delay, the plug-in uses the internal tempo of the sequencer. Tempo information is updated in the plug-in window when you open it and later, every time you execute a mouse operation. Please note that at this point the plug-in can't handle tempo changes. The reason for this is because we wanted to avoid the noise that is generated when a delay is switched. The Tempo box serves solely to display the current bpm value - you can't use it to change the tempo of the sequencer.
When you want to create dotted note values, move the Groove slider all the way to the right to "75%"; for triplets, select the setting "33.33%". Note that all intermediate values are possible. Conveniently, you can view the current delay value in the "Delay display.

Unsurprisingly, the Feedback slider determines feedback intensity; in other words, the amount of delayed and filtered signal that is routed back to the input of the Tape Delay. When you set it to the lowest possible value, the Tape Delay generates just a single echo. Turned all the way up, the echoes are repeated ad infinitum. Keep in mind that the levels of the original signal and its taps (echo repeats) tend to add up and may cause distortion. Here's where the internal tape saturation circuit comes to the rescue - it can be used to ensure that these overdriven signals sound good.

You can shape the sound of the echoes via the on-board high-cut and low-cut filters. Although these filters are fairly flat, they're not located post-output. Instead, they call the feedback circuit home, meaning that the effect achieved by these filters increases in intensity with each repeat. If you're in the mood for an increasingly muddy tone, move the high-cut filter slider towards the left. For ever thinner echoes, move the low-cut filter slider towards the right.

The Mix slider determines the balance between the original (dry) signal and effects (wet) signal. If you've inserted the Tape Delay to an individual track, you'll generally find that settings of up to 50% are desirable. If the Tape Delay is patched to the insert of a bus object and you're routing the signals of a track to the plug-in via the Send controls, you should set the Mix slider to "100%" and leave it there.
If you're unable to hear the effect even though you've set up a suitable configuration, be sure to check out not only the Mix knob, but also the filter settings: Move the high-cut filter slider to the far right and the low-cut filter slider to the far left.

Currently, there is only one Tape Delay (m/m), hence the monaural input and output. A Stereo Delay is your best bet for stereo tracks or busses.

General hints about the Controls

Stereo Delay

The Stereo Delay works much like the Tape Delay, which is why we'll skip the general info and take a closer look at the differences between the two. There is just one Stereo Delay (s/s), hence the stereo input and output. Of course, you are free to use the Stereo Delay for monaural tracks or busses when you want to create independent delays for the two stereo sides. Please bear in mind that if you chose this option, the track or bus has two channels from the point of insertion forward. In contrast to the Tape Delay, the Stereo Delay does not feature a circuit that replicates tape saturation.

You can set the Delay (via Note buttons and Groove sliders), Feedback and Mix values separately for the two sides. In contrast, the settings for the high-cut and low-cut filters apply to both sides equally. In addition, the plug-in features a Cross Feed knob for each stereo side - it determines the feedback intensity; i.e. the level at which each signal is routed to the opposite stereo side.

General hints about the Controls

Modulation Delay

As its name implies, the Modulation Delay generates effects such as flanging or chorus based on modulated short delays. It can also be used - without modulation - to create resonator or doubling effects. The modulation section consists of two LFOs with variable frequencies (0-20 Hz). The balance between these two is determined by the LFO Mix slider. Use the Width slider to enter the desired modulation width. When the Width slider is set to the far right position, delay modulation is switched off completely. The Volume Modulation (Vol. Mod.) slider lets you determine the intensity of the amplitude modulation (Tremolo). The Constant Modulation (Constant Mod.) button lets you do just that - ensure that the modulation width remains constant regardless of the modulation rate. When this feature is switched off, higher modulation frequencies reduce the modulation width. In simple delay circuits, a delay modulation normally also modulates the pitch of the signal. Use the Anti Pitch button to ensure the pitch of the modulated signal remains constant. This is exactly how high-end chorus and flanger effects work.
Set the basic delay time via the Flanger-Chorus knob. Set to the far left position, the Modulation Delay puts on its flanger cap. As you move towards the center position, it thinks it's a chorus. As you move the knob closer to the far right position, you will hear clearly audible delay taps. This type of setting is generally used without modulation (Width = 0) for doubling effects.

The Stereo Phase knob defines the phase of the modulation between the left and right stereo sides. At 0°, the extreme values of the modulation are achieved simultaneously on both side, at 180°, the extreme values opposite each other are reached simultaneously.

The Feedback slider controls the intensity at which the effects signal feedback is routed to the input. If you're going for radical flanging effects, enter a high Feedback value, if simple doubling is what you're after, you don't want any feedback at all. The Mix slider determines the balance between dry and wet signals.

General hints about the Controls

Phaser

The Emagic Phaser emulates the effect of analog phaser circuits in four to 12 orders (as in 4th order, 5th order etc.) Use the Order slider to set the desired number of orders. Note that you are free to select odd-numbered settings (5, 7, 9, 11), which, strictly speaking, don't generate actual phasing. We simply felt that the more unspectacular comb filtering effects can come in handy on occasion. As a rule, the more orders a phaser has, the heavier the effect. The settings "4, 6, 8, 10 and 12" put five different phaser algorithms at your fingertips, all of which replicate the analog circuits that they are modeled on, each designed for a specific application.

The modulation section comes complete with two LFOs featuring individually variable frequencies and freely variable mix options. Additionally, the frequency of LFO 1 can be modulated via the level of the input signal. Use the Envelope Modulation slider to set the desired modulation intensity. By staking out the limits of the modulation via its highest and lowest values, you can determine the modulation width and range. These high/low limits are controlled by the Sweep Ceiling and Sweep Floor sliders - you can enter values for them directly in the form of the desired frequency. This value also determines the maximum intensity of the comb filtering created by the phasing effect.

The Stereo Phase knob is used to define the phase for the left and right channels of a stereo phaser (s/s). When you're using a monaural phaser, this parameter is, of course, meaningless.
As the icing on the phasing cake, you can tweak the Color slider to add just that to the effect. Here the comb filtering effect is amplified via feedback.

General hints about the Controls

Pitch Shifter

The first thing you may notice about the Pitch Shifter is its minimalist approach - just a few parameters are visible. Semi Tones is used to set the transposition in semi-tone increments within a range of one octave upwards or downwards. Cents controls detuning in increments equivalent to 1/100th of a semi-tone step. Use the Mix slider to control the desired balance between the original and processed signals.

The Drums, Speech and Vocals buttons are used to set internal parameters so that the program delivers the best results for a given application. When you select Drums, the groove of the original track remains intact, with Vocals, the intonation of the original is retained unaltered. Hence Vocals is well-suited for all other signals that are inherently harmonic or melodious; for instance, string pads. Speech mode is a compromise between the two - the program attempts to retain both the rhythmic and harmonic aspects of the signal, which is desirable for complex signals such as spoken-word recordings or rap music. Speech is thus also suitable for other "hybrid" signals, such as rhythm guitar.

When in doubt, Speech is a good place to start. A/B the options to compare them and find the one that suits a given signal best. When auditioning settings and judging them for quality, it's a good idea to temporarily turn the Mix knob up to "100%". Keep in mind that Pitch Shifter artifacts are a lot harder to hear when you mix a smaller percentage of a transposed vocal to the overall signal.

General hints about the Controls

Ensemble

The Emagic Ensemble is like a Pitch Shifter on steroids - it consists of eight internal, modulatable Pitch Shifters. Two standard LFOs and one random LFO enable you to come up with fairly complex pitch modulations, which - much like a natural chorus effect - conjure up the impression of an incremental or vocal ensemble. A big Emagic Eye visually represents the number of voices and their modulations.
Use the Voices slider to determine how many voices (1-8) are generated in addition to the original.

Please note that the plug-in's appetite for computer resources increases proportionally to the number of voices: When you activate eight voices, the Ensemble requires roughly eight times the performance of a Pitch Shifter.

The two conventional LFOs and the random LFO (for the record, it generates random modulations) each feature a Rate knob that controls frequency and an Intensity slider to determine the modulation width.
The Phase knob controls the phase relationship between the modulations of the individual voices. The value that you select here depends on the number of voices, which is why it is indicated in percentages rather than degrees. The value 100 (or -100) is equal to the greatest possible distance between the modulation phase of all voices. Here the voices are distributed an equal distance apart over the full 360°.
The Stereo Base slider serves to distribute the voices across the stereo panorama. When you set a value of 100%, the stereo base is expanded artificially. Please bear in mind that monaural compatibility may suffer.

In addition to the familiar Mix slider that determines the balance of dry and wet signals, the Ensemble also features an Effect Volume knob. This lets you determine the level of the effects signal separately. This feature allows you to compensate for changes in volume caused by manipulating the Voices parameter.

General hints about the Controls

AutoFilter

The AutoFilter is an extremely versatile, resonance-capable low-pass filter that comes with a couple of truly unique features. Its important parameters are located on the right side: The Cutoff Frequency knob determines the point at which the filter kicks in. Higher frequencies are attenuated, lower frequencies are allowed to pass unhindered.

(This seems a good opportunity to share some good news: In case you were unaware of this, be advised that you can control all parameters of a plug-in via MIDI controllers. Depending on whether you insert the AutoFilter to Slot 1, 2, 3 or 4, the cutoff frequency is controlled via Controller Number 64, 80, 96 or 112. The MIDI channel must, of course, correspond to the settings of the track or bus object that you're dealing with. For live control applications, the given track of a mix object must be activated in the arrangement; e.g. "A-Playback").

The Resonance knob emphasizes the frequency range bordering the cutoff frequency. When you turn the resonance well up, the filter itself begins oscillating (with the cutoff frequency). Just like on the legendary Minimoog, self-oscillation is initiated before you max out the resonance parameter. When you're working with resonance, ordinarily the manner in which the low-pass filter allows frequencies to pass changes: Higher resonance values cause the filter to cull out the bottom end-the signal ends up sounding thinner. The Fatness parameter compensates for this audio version of anorexia. When you turn Fatness up to maximum, the resonance setting has no effect on the response of the frequencies below the cutoff frequency.

The Slope buttons determine how steep you want your low-pass filter to be: Frequencies above the cutoff frequency are dampened by 6, 12, 18 or 24 dB per octave (in audio jargon, these are called filters of the 1st, 2nd, 3rd and 4th order). Even if the 24 dB filter is largely the component of choice for synthesizer designers, be sure to experiment with the other options, they can also deliver pretty hip results. Distortion Input and Output lets you separately control the two distortion units - one pre-input and the other post-output. Although the two distortion modules are identical, their respective positions in the signal chain - before and after the filter, repectively - enable them to generate remarkably different sounds.
All other parameters of the AutoFilter are used to dynamically modulate the cutoff frequency. These come in two sections: Envelope (ADSR, Envelope Generator) and LFO (Low Frequency Oscillator, Modulation Generator).

The Threshold parameter applies to both sections and analyzes the level of the input signal. If the input signal level exceeds that of the variable Threshold level, the envelope and LFO are retriggered. The Modulation slider of each section determines the intensity of control signal's effect on the cutoff frequency. Envelope: When the Threshold level is exceeded, the control signal is triggered at the minimum value. After a variable interval, the length of which is determined by the Attack parameter, the signal reaches its maximum value. It drops off during the time it takes to run through an interval defined by the Decay value, and ends up at the predetermined Sustain value. Once it drops below the threshold, it falls all the way to the minimum value in the amount of time determined by the Release parameter. If the input signal again falls below the Threshold level before the control signal has reached the Sustain level, the Release phase is triggered. The Dynamic Modulation parameter lets you modulate the peak value of the Envelope section using the level of the input signal.

LFO: The LFO oscillates in the waveform that you can determine via one of the Waveform buttons: descending sawtooth (saw down), ascending sawtooth (saw up), triangle, pulse wave or random (random values, "Sample & Hold"). Once you've selected a waveform, you can shape the curve via the Pulse Width knob. Use the Frequency knobs to dial in the desired LFO frequency: Coarse sets a value between 0.1 and 1000 Hz, Fine lets you adjust it in smaller increments. The Speed Modulation (Speed Mod.) knob is used to modulate the frequency of the LFO independently of the input signal level. If the input signal exceeds the threshold level, the modulation width of the LFO increases from zero to the value that you have entered for Modulation. You can also define the amount of time this process takes by entering the desired value with the Delay knob. If the Sync button is activated, the waveform is started at 0° as soon as the threshold is exceeded.

Whenever you use the AutoFilter as a stereo plug-in, you can determine the phase relationship between the LFO modulation on the two stereo sides via the Stereo Phase knob.

General hints about the Controls

Spectral Gate

The Spectral Gate makes the signals above and below the threshold level audible, separately and within a modulatable frequency range. It works with a Fourier transformation of the entire signal. Technical jargon aside, it is a tool that lets you come up with pretty wacky filtering effects, stuff that, until this plug-in saw the light of day, hadn't been possible.

The frequency range that you want to process is defined by the Center Frequency and Bandwidth knobs. This frequency band is separated by steep slopes. Within this band, you can use the Threshold slider to determine a level that separates the frequencies above and below it. The frequencies above the threshold are made audible via the Super Energy knob, the frequencies below it via Sub Energy. Additionally, the original signal outside the defined frequency band can be added to the mix: Low Level blends in the frequencies that lie below the frequency band (bass frequencies) and High Level, the higher frequencies that lie above the defined frequency band.

The actual frequency band can be modulated via three parameters: Speed determines the modulation frequency, Center Frequency Modulation (CF Mod.) defines the intensity of the center frequency modulation and Band Width Modulation (BW Mod.) controls the bandwidth modulation.
The Gain slider lets you adjust the level of the generated effects signal.

We suggest you use a drumloop when you begin experimenting with this plug-in. Set Center Frequency to the minimum and Band Width to the maximum values (the entire frequency range is processed). Turn up the Super Energy or Sub Energy knob one at a time and fiddle with the Threshold knob. You'll soon get a feel for how different threshold levels affect the sound of Super Energy and Sub Energy. When you've come across a sound that you consider particularly nifty (or even just useful), you can narrow down the bandwidth drastically, gradually increase the center frequency and use the Low Level and High Level sliders to mix in some treble and bass from the original signal or, at lower Speed settings, turn up the CF Mod. or BW Mod. knobs. Enjoy!

General hints about the Controls

Reverb

Reverb is the bread-and-butter effect, the most prevalent of them all and the one you can least do without. In nature, every sonic event that reaches our ears is accompanied by varying amounts of reverb. As a rule, you could say that early reflections off walls and the floor are more significant than the actual reverb tail (diffuse reverberation). When you're planning a mix, be aware that reverb is perceived as a natural environment for a signal to hang out it, whereas the total absence of reverb is perceived as an "effect". The absence of reverb creates the impression of the kind in-your-face spatial immediacy or nearness that is considered desirable for just a few types of signals. A mix in which all signals are perceived an equal distance from the listener lack spatial depth - which is an involved way of saying that it sounds flat or two-dimensional.

Reverb plug-ins should always be patched into busses. Route signals from individual tracks to the plug-in via the respective Send knobs. With this method, you're making the most your machine's resources by using it to compute the reverb for several tracks at once. You can vary the Send knob setting to stagger the signals so they appear to have different spatial depths.

If, however, you want to slap a special type of reverb on just one track, you are, of course, free to insert the plug-in directly to it. In this case, use the Mix parameter found on every reverb plug-in to determine the signal's spatial depth or, put simply, how far back in a room it seems to be. The balance between the dry and wet signals is the single most important parameter when you're working with reverb.
Tip: Often you'll find it beneficial to insert an equalizer (Fat EQ) before or after a reverb plug-in. This type of setup lets you adapt the sound of the reverb to the requirements of the mix. Some audio engineers like to compress the reverb signal to create a tighter sonic image.

When you're trying come up with a great reverb sound using a plug-in, we recommend that you first try out the factory presets. Once you locate a reverb that sounds somewhat similar to what you're after, you can edit it to suit your taste.

SilverVerb
GoldVerb
PlatinumVerb
Enverb

SilverVerb

Although the Emagic SilverVerb is based on a simple reverb algorithm, it delivers remarkably good results.
The actual reverb algorithm is controlled by just two parameters: Like the name says, Reflectivity defines how reflective the imaginary walls, ceiling and floor will be. Room Size challenges your architectural skills - use it define the dimensions of simulated rooms. The graphic display visually represents these parameter settings.

Pre-delay determines the delay between the original signal and the reverb tail. Whereas high pre-delay settings tend to generate something similar to an echo, low values often muddy the original signal. Ideally, you should go for as a high a setting as possible before the plug-in begins generating something that sounds like a delay tap.

Low Cut and High Cut let you filter bass and treble frequencies out of the reverb tail.
In most cases this will unclutter your mix. The reason for this is that a long reverb with a great deal of bottom end generally makes for a flabby mix and high frequencies in the reverb usually sound somewhat unpleasant, hamper speech intelligibility, or mask the overtones of the original signals.

General hints about the Controls
General Hints about Reverb

GoldVerb

The GoldVerb consists of two sections: Early Reflections and Reverb (diffuse reverberations). The balance between these two sections is controlled via the Balance ER/Reverb slider located above the graphic. When you set this Balance slider to either of its far positions, the unused section is deactivated to maximize performance.

Occasionally, you should try using solely the Early Reflections sections of the GoldVerb for ambient effects. It will be well worth your effort!

Early Reflections

This section emulates the original signal's first reflections when they bounce off the walls, ceiling and floor of a natural room. These early reflections are essential to how we perceive a room. All information about the size and shape of a room capable of being handled by the human ear is contained in these early reflections.

Pre-delay

Pre-delay is the amount of time that elapses between the original signal and the arrival of the early reflections. In any given room size and shape, Pre-delay determines the distance between the listener and the walls, ceiling and floor. In conjunction with artificial reverb, it has proven an advantage when this parameter can be manipulated separately from, and over a greater range than, what is considered natural for Pre-delay. In practice, too short a pre-delay tends to make it difficult to pinpoint the position of the signal. It can also color the sound of original signal. On the other hand, too long a pre-delay can be perceived as an unnatural echo. It can also divorce the original signal from its early reflections, which leaves an audible gap. Unfortunately, the ideal pre-delay setting depends on the properties or, more accurately, the envelope of the original signal. Percussive signals generally require shorter pre-delays than signals in which the attack fades in gradually. The rule of thumb is: Use the longest pre-delay possible before the undesirable side effects such as an audible echo begin materializing.

Room Shape

Use this slider to define the geometric form of the room. The numeric value (3-7) represents the number of corners it has.

Room Size

Unsurprisingly, Room Size determines the dimensions of the room. The numeric value indicates the length of its walls, i.e. the distance between two corners.

Stereo Base

The Stereo Base parameter enables you to define the distance between the two virtual microphones that you are using to audition the simulated room. Spacing the microphones slightly further apart than the distance between two human ears generally delivers the best results. Of course, more realistic results can be obtained if you choose to use the distance between two ears located on opposite sides of the same head.

Reverb

This section generates diffuse reverberation.

Initial Delay

This is the delay between the original signal and the diffuse reverb tail. If you're going for a natural-sounding, harmonic reverb, the transition between the early reflections and the reverb tail should be as smooth and seamless as possible. Basically, what we said about the pre-delay (see the section: Early Reflections) holds true for this parameter:

Set the initial delay so that it is as long as possible without there being a perceptible gap between the early reflections and the reverb tail.

Density

This parameter controls the density of the diffuse reverb. Ordinarily, you want the signal to be as dense as possible. However, less density means the plug-in eats up less computing power. Moreover, in rare instances, too great a density can color the sound, which you can fix simply by backing off the density knob. If you select a density value that is too low, the reverb tail will sound grainy.

Reverb Time

Reverb Time is commonly considered the amount of time it takes for the level of a reverb signal to drop by 60 dB. This why the time is often indicated as RT60. Most natural rooms have a reverb time somewhere in the range of 1 to 3 seconds, a value which absorbent surfaces and furniture reduces. Large empty halls or churches have reverb times of up to eight seconds, some cavernous or cathedral-like venues even beyond that.

High Cut

Uneven or absorbent surfaces (wallpaper, wood paneling, carpets etc.) tend to reflect lower frequencies better than higher frequencies. The High Cut filter lets you replicate this effect. If you set the High Cut filter so that it is wide open, the reverb will sound as if it is reflecting off of stone or glass.

Spread

This parameter controls the stereo image of the reverb. At 0%, the plug-in generates a monaural reverb, at 100%, the stereo base is artificially expanded - which, of course, makes the reverb sound monumental, but collapses in a monaural playback.

Yes, the 21st century is looming, but if you're mixing tracks that should still sound fairly decent when they're piped through tiny short-wave receivers, antiquated kitchen radios, bargain-basement clock radios, simple (monaural) TVs, hearing aids etc., be sure to keep monaural compatibility in mind.

General hints about the Controls
General Hints about Reverb

PlatinumVerb

The difference between the PlatinumVerb and the GoldVerb is the former's enhanced Reverb section. The Early Reflections sections of the two plug-ins are identical. For more info, please read the section covering the GoldVerb, here we'll focus on the additional features of the PlatinumVerb.

The Reverb section of the PlatinumVerb is based on a genuine dual-band concept. This is to say that the on-board frequency crossover splits the incoming signal into two bands which are then treated with reverb in two separate modules.

Crossover

This is the frequency at which the two frequency bands are split for separate processing.

Low Ratio

This parameter factors the reverb time of the bass band. It deviates from the setting for the Reverb Time parameter, which applies to the high band. At 100%, the reverb times for the two bands are identical. At lower values, the reverb time of the frequencies below the crossover frequency is shorter. At values greater than 100%, the reverb time for low frequencies is longer.

Both of these phenomena occur in nature. In most mixes, a shorter reverb time for bass frequencies is preferable. For example, if you're using the PlatinumVerb to put reverb on a drumloop featuring kick drum and snare, a short reverb for the kick drum lets you dial in substantially more of the wet signal.

Low Level

This knob determines the level of the Bass reverb. At 0 dB, the volume of the two bands is equal. The Bass reverb level can be boosted by up to 12 dB and attenuated by up to 100 dB.
In the vast majority of mixes, your best bet is to dial in a lower level for the low-frequency reverb signal. This lets you turn up the level of the bass instrument - it will sound punchier. This also helps counter bottom end masking effects.

General hints about the Controls
General Hints about Reverb

EnVerb

The Emagic EnVerb is based on a rather unusual and innovative reverb algorithm. It has a unique feature - you can adjust the envelope of the diffuse reverb tail as you see fit. This gives you options well above and beyond that of a conventional gated reverb.
The algorithm of the EnVerb requires quite a bit of computing power!

Time Parameters

With a concept as sophisticated as that of the EnVerb, you can well imagine that a single parameter for reverb time just won't do the trick.

Original Delay

This parameter enables you to delay the original signal. Of course, this delay is only perceptible when the Mix parameter is set to a value other than 100%. The starting point of the diffuse reverb tail is not influenced in any manner.
A delayed original signal is particularly handy when you want to generate reverse reverb: Set all envelope parameters to zero with the exception of Attack and Original Delay, which you should set to approximately the same value by which you want to pre-delay the given region or track.

Pre Delay

This is the delay between the (undelayed) original signal and the starting point of the reverb attack phase.

Attack

This is the amount of time it takes for the reverb to climb to its peak level.

Decay

This is the amount time it takes for the level of the reverb to drop from its peak to the sustain level.

Sustain

This is the level of the reverb that remains constant throughout the sustain phase.

Hold

This is the duration of the sustain phase.

Release

This is the amount of time that the reverb takes to fade out completely after it has run through its sustain phase.

Sound Parameters

The following parameters shape the sound of the reverb. (For more information on these parameters, check out the in-depth descriptions of the GoldVerb or PlatinumVerb).

Density

Reverb density. Higher values generally sound better.

Spread

Stereo base of the reverb.

High Cut

High-frequency attenuation for the reverb.

Crossover

The crossover frequency for the following parameter...

Low Level

Relative reverb level of frequencies below the crossover-frequency. Although you are free to turn the level of these frequencies up, in most cases, you'll get better-sounding results when you set negative values for this parameter.

General hints about the Controls
Allgemeine Hinweise zum Hall

Oscillator

The Oscillator generates a static frequency as determined by the Frequency parameter. The level of the oscillator is controlled by the Oscillator parameter, whereas the level of the original signal of the audio object is controlled by the Input parameter. An oscillator wouldn't be much fun if all it could do was produce an interminable sine tone. One of the redeeming features that spares us equally interminable boredom is the ring modulator - it produces so-called side bands. The frequencies of these side bands depend on the original signal of the audio object and the signal of the oscillator. In technical terms, the ring modulated signal is the product of real-time multiplication of the two signals. If you analyzed the resultant ring modulated signal, you'd find that this signal consists of the original signal as well as all the sum and difference of its frequencies.

Here's an example: Let's say the audio object plays back a 200 Hz sine wave and the oscillator generates a frequency of 1000 Hz. Ring modulating these signals generates side bands with 1200 Hz (sum) and 200 Hz (difference). The generated spectrum is not necessarily what most people would call harmonic. This is why ring modulators are often considered good tools for shaping enharmonic metallic sounds, bells being one of the more conventional sounds that they can produce. Bear in mind that the signal of your audio object will be much more complex than a simple sine wave. Moreover, the oscillator is no slouch either - it can do more than just generate sine waves. Consider an oscillator an additional source of distortion rather than a simple perpetual note generating component.

Here's a quick run-down on the parameters: Frequency determines the frequency of the oscillator. Sawtooth sets the level of the sawtooth wave, the spectrum of which contains all harmonics. Sine Wave defines the level of the sine wave and Pulse the level of the pulse/square wave. PulseWidth controls the ratio of positive and negative pulses. A value of 50% generates a square wave, the spectrum of which consists solely of odd harmonics. Noise adds low-frequency noise. Input controls the level of the original signal of the audio object, Ring Modulation the level of the ring modulator, and Oscillator the level of the pure oscillator signal.

General hints about the Controls

General Informationen

The real-time effects of Version 4.0 have been fundamentally redesigned and now supplant the real-time effects found in earlier versions. The sole exception is the earlier version's reverb plug-in, still available under the appropriate name "3.0 Reverb." Although the quality of the new reverb algorithms is better by any objective measure, we nevertheless left the old reverb in the program's arsenal of sound-sculpting tools simply to give you more options to chose from.
If you want to revisit the basics of working with effects plug-ins, please check out the chapter entitled "Mixer and Audio Objects" in the reference manual.

Update customers take note

To maintain the sonic integrity of your songs, the updated version does not tamper with your old songs. They remain fully intact, and are opened under Version 4.0 and played back with the old effects. However, when you want to insert plug-ins, only the new effects appear in the selection list. "Why?" you ask: Simply because they sound better and significantly enhance performance.

Autoload Song

If you have preconfigured effects plug-ins for your autoload song, we recommend that you replace these with the new algorithms. If you have positioned equalizers on the interface, be sure to select the new algorithm for each EQ band or insert a Fat EQ instead. Your best bet for replacing old flanger and chorus effects is the new Modulation Delay (ModDelay). Choose the new Tape Delay (m/m) or Stereo Delay (s/s) in place of the old delay, replace your basic reverb with a SilverVerb, and your main reverb with a GoldVerb or PlatinumVerb. Hint: If your machine is not one of the most muscular on the market, you can be assured of good results even when you use a SilverVerb as your main reverb.