Inside the Yamaha 01V

Created by Cub Lea with the assistance of the online DS2416 and 01V user community.

This document is a supplement to the C-Console manual.
This is not an official part of the C-Console documentation.
It is designed to provide 01V owners with additional information about the mixer which is not found in the owner's manual.

TABLE OF CONTENTS

INTRODUCTORY AND TECHNICAL MATERIAL

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AN INTRODUCTION TO DYNAMICS PROCESSORS

What is dynamics processing?

Dynamics refers to the variation that occurs in an audio signal's level over a period of time. Dynamics processors allow you to control how the signal level changes over time by raising or lowering the level, or simply by allowing the level to remain unchanged, depending on how you want the track treated. While dynamics processors may have level controls, dynamics doesn't normally concern itself with overall output levels. Instead, it deals with the way those levels change.

Envelopes: understanding the terminology

In terms of dynamics, all audio material has an *envelope*, or a "container of time", which holds the sound. When referring to dynamics, audio engineers describe the characteristics of the container rather than the sound itself, although in many cases it is difficult to tell the difference.

Depending on the application, an envelope may be described in terms of up to a dozen separate sections of time. But in most cases, envelopes can be neatly described by division of the sound into two sections: attack and decay. Technically speaking, these are the only real divisions you need to make to any sound in order to define its envelope. Any other divisions in the sound's timeline are artificial and made purely for convenience.

The **attack** portion of the sound consists of the rise in volume when the sound is first generated. In nature, all sounds have a measurable attack. No sound begins at maximum volume, although many sounds may seem to do so.

The **decay** portion of the sound consists of the period of the sound's existence in which its volume is falling. Some sounds have practically infinite decay; others begin to decay to nothingness almost immediately after the sound begins.

All a dynamics processor really needs to be effective is a way to detect and control attack and decay. But much more control is needed in recording and engineering environments, so most envelopes used by modern dynamics processors add two other divisions to a sound's timeline: sustain and release.

Sustain is normally used to define how a sound trails off after an initial decay period. A sound might rise from 0dB to 75dB in a millisecond or two, then fall to 60dB within just 25 milliseconds or so. But the nature of the sound source might allow the level to fall from 60dB to 45dB over a period of several seconds, not milliseconds. Drums typically have this type of response curve. While attack starts at 0dB and stops where the peak level in the sound is reached, there is no fixed point at which sustain starts and stops, because sustain is an artificial division in the timeline. Depending on how sustain is applied, an engineer could specify the sustain section of a sound to begin when the sound falls to a given number of decibels, or specify it to begin after a certain length of time has elapsed. (Most dynamics processors start the sustain section based on the level of the signal.)

Release, the other common arbitrary division in the timeline, comes after the sustain section (or, in envelopes defined without sustain, after the decay section) as a way to gradually drop the sound level to 0dB or below the audible threshold. It acts as a second decay section. The release section of an envelope is normally used to mimick the effect in nature of a sound reaching a certain level and then falling below the audible level. In nature, there is a constant level of background noise. Let's say for the sake of argument that it's always 45dB. When a sound falls below 45dB, it blends with all other sounds in the environment and *seems* to disappear very rapidly. The sound is actually still there, but it's no longer discernible amid the normal background noise. In recording and engineering environments, background noise levels are usually exceptionally low, so release is used to simulate the effect of a sound dropping into the background.

Finally there's **threshold**. Most dynamics devices use this as the volume level where processing of a sound's dynamics starts to kick in. Below that level, no processing is done. (Note that in the case of noise gates, no processing means no sound at all.)

Capabilities of dynamics processors

Among the most basic and common applications of dynamics processing include:

- preventing a track with an especially harsh or "hot" attack, or a signal with uneven peak levels, from dominating the mix
- reducing, or even eliminating, the audibility of line hum, hiss, and background noise
- increasing the presence of a track by reducing its overall dynamic range, thus allowing its perceived volume to be increased without actually increasing the risk of clipping
- increasing the emotional range of a track by widening its dynamic range to accentuate peaks or attacks

But that barely scratches the surface of what modern dynamics processors can do. Over the years, engineers have discovered a wide range of more inventive applications for dynamics processors by applying fancy signal routings and unusual parameter sets. Among these less basic processes are:

- applying compression only to specific EQ bands, e.g. "de-essing" which is removing the harshness from the 5kHz region of a vocal track, or "squashing bass" by applying limiting to signals in specific low frequency bands
- "ducking", in which a rise in the level of one signal forces another signal's level to fall; often used by disc jockeys speaking over a background music track to allow the music track's level to rise during gaps in the dj's speech
- triggered gating, in which the volume envelope of one signal controls the volume envelope of the signal being sent through the gate; perhaps most commonly used in pop music in noise gates applied to bass guitar tracks as a means of insuring that bass notes have an attack which is in precise time with the kick drum
- presence boosting through the use of high amounts of compression, achievable due to the tonal effects of compression on audio material. Compressing a signal decreases its dynamic range, giving an illusion of greater volume in a compressed signal with the same actual level as the original.
- "pumping and breathing" effects through the application of large amounts of compression to a track with a relatively wide dynamic range, often used on rock guitar and more recently on percussion tracks

There are many, many more possible applications for dynamics processors than those listed above. We've only mentioned the most common recording applications here.

There's one other very common type of dynamics processing you may use every day without knowing it. Most forms of noise reduction used on cassette tape are achieved with the help of dynamics processors.

A bit of history

Dynamics processors were first developed to help provide better control over the dynamic range of an audio signal destined for recording or broadcast. And throughout the history of recorded music, the importance of dynamics processing has increased as the dynamic range of popular forms of music have risen and as the technology capable of capturing and recording, broadcasting and playing back that music has increased in sophistication.

"Play it slow and dirty": a less-than-clean past

Early dynamics processors were tube-based circuits, and tube designs are still widely used in professional sound engineering. Even today, many of the costliest, most prized and respected dynamics processors are tube-based circuits. This is because the core of any good dynamics processor is its amplification circuitry. (See **Distortion and overdrive** for a detailed discussion of what makes these tube circuits so desirable.) These early circuits tended to be slow to respond and very dirty compared to modern circuits. Very often dynamics weren't even needed because microphone and playback technology prior to the mid-1930s was unable to capture or transmit a dynamic range wide enough to make dynamics control essential.

Meeting the demands of modern music

That changed as the big-band sound arrived on the scene and microphone and amplification technology began to leap forward in sophistication. By the mid-1940s, most better recording and broadcast studios made use of at least some form of limiting either to improve sound quality or enhance presence as the dynamic range of home audio equipment increased thanks to better tube circuit designs and improved manufacturing processes.

The sharp attacks and enormous dynamic ranges of amplified guitars made control over signal envelopes even more important as rock and roll came into prominence in the mid-1950s. Without peak limiting in the signal chain, it was difficult to capture the energy and immediacy of a live rock and roll band from the recording floor without risking unacceptable amounts of clipping in the recorded signal.

Fighting to be heard: why you *have* to use dynamics today

Where most commercials in the early days of radio were performed live, prerecorded advertisements had become the norm by the mid-1950s. The widespread use of prerecorded messages gave advertisers themselves the ability to fully control the way their messages sounded. But by then there were standards in place that put limits on the volume that broadcasters were allowed to use. But broadcasters and engineers already knew that certain types of dynamics processing could actually make their messages seem louder or more urgent, giving them an edge over competitors who didn't use these techniques, and dynamics control became widely applied both by advertisers *and* broadcasters. Today virtually every ad production facility in the western hemisphere uses dynamics processing to try to make their messages *seem* louder than others without actually exceeding imposed limits. It is still easy to spot amateurish ad production because it lacks the perceived volume of most other ads. It has actually been a long time since dynamics could be used to make an audio message stand out. The fact is that it *has* to be used today because everyone *else* is using it.

Dynamics in "our generation": an effect in its own right

As experimentation became the norm in rock and roll as the 1960s wore on, engineers discovered, tried and applied routings and parameter settings considered impractical, impossible or unlistenable only a few short years earlier. Dynamics processing quickly outgrew its limited roles as a janitor for music recording and presence enhancer for broadcast audio. It became a full-fledged special effects category.

One of the most famous pioneering instances of dynamics processing as a special effect is the Who's "My Generation". The band members, who actively participated in the production (and apparently *not* at the invitation of the producer), kept decreasing the compression threshold and boosting the input levels until they achieved the song's signature "in-your-face" presence. The engineer at that session once remarked that he'd never seen, nor heard of, anyone applying limiters to a mix to such a high degree prior to that date. This type of heavy limiting is still a popular rock/dance recording effect, and is applied to individual tracks in virtually all types of popular music.

The pursuit of technical perfection: dynamics comes of age

By the mid-1970s, technical precision had become a watchword in pop recording. It was no longer enough to play it loose and sloppy and survive on feel alone. Music had become the top-grossing entertainment industry in the world, and with that came pressure to use every means available to stay on top. As the decade rolled on, de-essing, limiting and triggered gating became routine, and almost expected, even in small semi-pro studios. A mere decade earlier, only the finest studios in the world could have boasted that kind of processing power.

As the cost of reasonably high quality solid-state dynamics devices continued to fall through the 1970s and 1980s, better studios routinely acquired as many as two dozen different dynamics processors. Earlier tube designs suitable for professional recording had traditionally been extremely expensive and were often applied only to the final mix or to a particular "problem track". But by the end of the 1980s, even the average home recordist could afford - and find a use for - a half-dozen or more compressors, de-essers, limiters and noise gates.

"The digiverse...and beyond": dynamics in the digital domain

Modern digital dynamics processors have truly brought world-class dynamics processing to the desktop. Today you can buy plugins and stand-alone applications for a few hundred dollars whose response characteristics are almost an exact match for classic tube designs that might sell for several thousand dollars per channel. Dynamics circuitry is simple enough to emulate in a DSP chip that better mixing cards can run literally dozens of high-quality dynamics devices at once and still have plenty of processing horsepower left over.

The importance of dynamics is clearly evident in the way companies are including anywhere from several to *dozens* of dedicated, independent dynamics processors on cards and digital hardware mixers. If you're new to recording, take this as an object lesson in the importance of dynamics processing to modern recording. While only the most finicky engineers will ever likely find a valid application for *all* of the dynamics processors on some devices that offer one for every channel in the mixer, it's not uncommon to discover a need for as many as a dozen compressors, gates and duckers on a single mix. And digital dynamics processing is an incredible bargain, too. Even as late as 1990, achieving the type of quality and control over the dynamics of your mixes that you can achieve today even using free software plug-in dynamics effectors would literally cost you thousands of dollars.

Experimenting with dynamics

Experimentation with EQ is best done using a bland broadband signal such as pink noise. Not so with dynamics processors. To get a solid feel for what these devices can do, you need familiar real-life audio material. You need to hear how the dynamics change over time. To get a good picture of how dynamics processors affect audio signals, try them using two or three different types of material. Here are a few suggestions.

- a dramatic selection of classical music with a wide range of emotions and dynamics, perhaps a short overture
- a relatively monotonous but "heavy" pop dance track
- a basic voice-over or a reading of dramatic material
- long "found-sound" samples from nature or civilization
- a high-energy big-band or heavy metal track

Additional introductory information

At the time of this writing, a very good basic tutorial on dynamics processing for recordists was hosted by Harmony Central at *http://harmony-central.com/Effects/effects-explained.html*

Dynamics in the recording process is covered with sensitivity and depth on Digital Domain at *http://www.digido.com/compression.html*. Even longtime recordists will likely find useful information here.

Compressors and limiters

Compression

The idea of compression is to try to balance the overall volume of a signal over time by reducing the volume of loud sections. This increases the apparent volume or "fullness" of the signal at the expense of reducing its dynamic range. Compressors do this by producing a proportional decrease in volume during loud sections of the signal, acting as a type of automated volume control. Any signal louder than a given threshold level will be reduced in volume by half, three-quarters, nine-tenths, or any other fraction (or, more commonly, *ratio)* specified by the engineer. The key is that overall volume isn't reduced by the specified ratio. If the dynamic range is 90dB, the ratio is 3:1 and the threshold is -24dB, a signal of -23dB will not be reduced to a total volume of 22-2/3dB (((90-23)/3)=22.67). Instead it will be reduced two-thirds of the way to the *threshold* level. So our -23dB signal would only be reduced by two-thirds of a decibel. It's a little like having an automatic volume control that only starts working above a certain level. Compressors normally have no effect on signals below the specified threshold level.

Let's assume that you have a clean jazz guitar track with a dynamic range so wide that the pick attack overpowers the mix in certain spots. You might set the compressor to -32dB with a ratio of 2:1 to give the picking less punch. This preserves all the emotion in the soft picking, which won't be affected by the compressor. But it will *significantly* reduce the peak levels of intensely picked passages. All signals up to -32dB will pass through the compressor unaffected. But for every 2dB *above* -32dB, the signal will be attenuated by 1dB. So signals that rise to -16dB, which is 16dB above threshold, will be cut by 8dB. There's the 2:1 ratio in play. Basing gain reduction on a percentage of total volume relative to a base volume level is what makes good compressors sound so natural and transparent.

Limiting

A limiter is nothing more than a specialized compressor. It performs the same type of attenuation of signals above the specified peak level (thus the name "peak limiter"), but the ratio on a compressor is infinite for practical purposes. Nothing above the specified peak level gets through a limiter. The same jazz guitar track fed through a limiter at -32dB will definitely sound "squashed" because there will be much less dynamic range during intense passages. Compression preserves dynamics where they are needed; limiting removes them where they're unwanted.

Most better compressors can be configured to act as peak limiters. Since unused compressor features don't typically sap DSP horsepower in devices that include several dynamics processors, some manufacturers code compression and limiting into the same algorithm so they're available from the same interface. If you're using a device like this that has a compressor, chances are excellent that the compressor will also act as a limiter. Simply configure the compressor for **Infinity:1** compression and presto...it's a limiter.

Too much of a good thing: what these devices do to sound

Now that dynamics can be processed in the digital domain with virtually zero loss of fidelity,[*] compression and limiting are often applied without a second thought. 'Tweren't so just a few short years ago. As you probably know, every time you pass an audio signal through another op-amp device, it loses fidelity. Every device in an analog signal chain passes the signal through at least one op-amp, and the total loss of fidelity at the end of the chain is highly dependent upon the quality of those circuits. Due to the way signals are passed to op-amps in compressors and limiters, they are especially vulnerable to producing signal degradation and demand very high quality components to insure maximum fidelity. Expensive components add up to an expensive piece of hardware. There is no way that a $200 outboard limiter is going to sound as good as a $2,000 outboard limiter with the same response characteristics to any engineer with a trained ear, no matter what the charts and specs might say.

In the digital domain, it's simply a matter of writing an algorithm that accurately reproduces the desired response curves and has a high enough resolution to preserve the integrity of the input. As long as the input signal can take the application of a digital process without significant degradation, there's no reason why a freeware DirectX plugin can't sound as good as a dedicated DSP card assigned solely to the task of dynamics processing. In the analog domain, compromising on dynamics devices on several tracks almost always results in audible degradation of the final mix.

Myths debunked

Now that we've defined what compression and limiting *are*, let's define what they *aren't*, because there are a lot of otherwise knowledgeable musicians and recordists who still harbor misconceptions about compression and limiting.

Myth: Compressors and limiters are completely different circuits.

Fact: A compressor and a limiter can be built around the same circuit. All you need to do to turn a compressor into a limiter is to set the ratio to infinite and remove the ratio control, or allow a ratio control to be set manually to **Infinity:1.**

[*] "Zero loss of fidelity" assumes, of course, that you've done your homework and understand what is required during recording and processing to insure the maximum possible fidelity in a processed digital signal.

For a thorough discussion of this, see *http://www.digido.com/ditheressay.html*.

Myth: Compressors use a gain control; limiters do not.

Fact: Normally manufacturers assume that you don't want a gain control on a limiter, but that doesn't mean one can't be used. Cheap limiters often do omit an output gain control, but most better outboard limiters include these controls.

Myth: Compressors can increase the gain of softer signals as well as decrease the gain of loud signals.

Fact: This is probably the most pervasive myth about compressors. This myth springs primarily from a side effect of inexpensive pedal compressors. Most users set these pedals' gain controls to higher than the input level. When large amounts of compression are applied, you certainly hear the signal being attenuated, but the release time is so slow on many of these devices that you also hear a noticeable rise, or "breathing" effect, after the note trails off, and any background noise in the signal seems louder. But it only seems louder because the gain is boosted, not because the compressor itself is raising the signal level. Using any good compressor with gain set to 0 (no increase or decrease in volume), you'll hear quite clearly that the compressor does *not* increase low signal levels.

Myth: Beyond the threshold level, compressors and limiters stop processing the input signal.

Fact: Actually the exact opposite is true. Compressors and limiters can't affect the signal until it gets to the threshold level. They don't take effect *at all* before the threshold is reached. (Actually this is only partly true. Soft knee compressors and limiters start to do their work *before* threshold is reached, but they don't reach their peak attenuation ratio at levels below the threshold level.)

Compression and limiting in practice

Compression is used on almost every type of audio material you can imagine. Chances are good that the last TV commercial you heard used both compression *and* limiting to give it an "in-your-face" feel. Compression is used on practically every instrument in every variety of pop music at some point in the recording process. Applied to bass instruments, it brings the instrument forward in the mix without unduly increasing boominess. Applied to acoustic instruments, it imparts a "smoother" tone to the music, often increasing the accessibility of the music. (Compression tends to be used relatively sparingly when recording and mixing classical music, especially compared to the way it is applied in pop.) Applied to the spoken voice, it permits a much wider emotional range to be expressed without forcing the listener to strain or suffer during particularly subdued or intense passages, but of course it can't completely compensate for good mic technique on the part of the speaker. It often doesn't even need to be applied to electric instruments, especially overdriven or distorted instruments, since amplifying anything to the point of clipping results in compression of the signal.

Where compression is used primarily to insure that a relatively natural dynamic range is preserved, limiting is used to deliberately crush dynamic range. If applying a compressor will solve the same problem as a limiter, most engineers will choose to use a compressor since the peaks in the audio material will normally sound more natural than they sound after limiting. Heavy limiting can be a dramatic, in-your-face effect, but most engineers prefer to reserve the use of limiters for processing tracks by players with especially poor control over their dynamics or for fixing problem areas in mixes, input signals or individual tracks where the signal must occasionally be prevented from going a few dB into clipping.

The good, the bad and the ugly

Effective use of a quality limiter can actually help squeeze more fidelity out of analog recordings. If a track threatens to go too far into the red (clipping) in a few spots, applying limiting to the track insures that the signal can't rise above a given level (perhaps +3dB measured at the recorder's input), allowing the input gain to be boosted just a touch. Every dB of gain you can add to the lowest level of input signal lowers the noise floor of a tape recording and, in theory at least, should improve overall fidelity.

Subtle, expertly-applied limiting can sound just as transparent as compression, but it is often applied *instead* of compression, resulting in peaks that sound artificial and forced rather than smooth and natural. This is why limiting is generally applied as a problem-solver and a final touch. For example, limiting is applied to two-track stereo masters much more often than compression prior to committing the material to tape or CD, because compressing at less than an infinite ratio can result in a master that sounds "thin" and lacks sufficient punch. This sort of punching-up can easily be overdone, though. Limiting applied during the mastering process seldom removes more than a few dB of the most severe peaks, because anything more would impart a "squashed" sound to the recording.

Many devices compress whether you want them to compress or not. Tube amplification circuits are fabled for their ability to impart a subtle compression to the signal that softens peaks in a pleasing way. Any circuit driven to clipping will compress at the point of clipping. Technically, almost all loudspeakers also act as compression devices since the louder they become, the harder they have to work to push air and the more the air's resistance impedes the speaker's ability to produce a 100 percent accurate output. The air's resistance acts to impart subtle compression to the speaker's output. Microphones have subtle compression effects. And today we're even seeing commercial plugins designed to reproduce the compression effects of audio signals recorded to magnetic tape.

Pumping and breathing: dirty, yes...sexy, no

These two effects are often the results of incorrectly applied compression and limiting. But they can also be used as effects in their own right. Pumping occurs when release times are set too long for the track being processed. When loud passages give way to quiet passages, low level signals will seem to rise in volume as the gain reduction of the compressor decreases. Keyboard tracks can be especially vulnerable to pumping when heavily compressed. Breathing occurs when background noise seems to rise in volume as gain reduction drops at the end of a passage, and is most commonly heard on improperly gated overdriven guitars.

Pumping can be a valid and quite dramatic musical effect, especially when two levels of overdrive are used on an input track. The trick is to set the release time of the compressor long enough to allow a smooth-sounding rise in volume when switching from the louder of the two overdrive tones to the quieter.

Breathing has a number of interesting applications. Several popular dance tracks deliberately apply overlong release times to a compressed reverb signal, giving the reverb a feeling of rising in volume as it tails off.

A bit of trivia: "chemical compressors"

Here's a strange bit of trivia to ponder before your next mixdown. Both nicotine and caffeine can be thought of as "neural compressors". We have long known that the effects of both of these drugs include modifying our response to dynamics in environmental stimuli. This means that we tend to perceive less total variation in the ranges of sound, light, smell, taste and touch while under the influence of these substances. In these cases, though, there's no threshold control for the compression effects we experience. Both substances tend to elevate our perception of low-level stimuli and decrease our perception of high-level stimuli.

Noise gates

Discriminating against the weak: how noise gates work

At its most basic, a noise gate is nothing more or less than a switch. When input signals rise above a specified threshold level, the switch opens the gate and allows the signal to pass through to the gate's output. When it falls below that level, the switch closes the gate and the output is silenced. This allows "space" in the audio material that might contain annoying low-level background noise to be filtered out of the track, effectively increasing the track's perceived dynamic range and clarity.

In practice, noise gates are far more sophisticated. Noise gates designed for high-fidelity audio applications are severely limited in value if all they do is switch the output on and off. If that switching isn't done in a way that allows the sound to retain a natural envelope, then the gating effect will sound like a channel being switched on and off. The sound has to have appropriate attack and decay characteristics, which is why noise gates are built on the same type of circuit architecture as compressors and limiters.

Better noise gates offer control over sustain as well as decay. Stringed instruments in particular need this extra control for natural-sounding gating. An overdriven electric guitar may not open the gate until the level reaches -

20dB so that amplifier hiss and stray hand noise on the strings doesn't pass through. But as the signal decays, the note may need to be heard at levels as low as -35dB or lower, and the track in question might have dead air in a subtle passage that wouldn't pass through the gate if just attack and decay parameters were used. Adding sustain to the gate allows the gate to remain open for a prolonged period even if the signal falls below the threshold level. This allows for "space" in the track without having the track drop out completely.

Noise gating is not the same thing as noise *reduction*. When the gate is open, both the desired audio material and any accompanying background noise will pass through to the output. Noise reduction circuits will normally allow low-level passages to pass through, noise and all, with a significantly decreased volume level when signals drop below a certain level. But when a noise gate is closed, *nothing* gets through.

Not a completely transparent effect

Theoretically, noise gates should apply no coloration to the sound. In practice, a slight coloration often does creep into the sound since the gate doesn't normally close faster than the ear can hear. (Don't be too concerned about this; it's generally a *very* slight, generally inaudible coloration in quality gates.) While digital gates can be opened in a millisecond or less, it's normally impractical to close the gate at this rate unless you're working with a regular signal such as a repeating sample. In practice, the gate needs to be closed gradually to accommodate the natural decay characteristics of musical instruments. While the gate is fully closed, no signal passes so no coloration occurs. While the gate is fully open, any coloration that occurs is produced by the circuitry. This coloration might be painfully audible on cheaper compressor pedals, but should be virtually unnoticeable in quality rack-mount analog and software-based units.

"Quiet, you!": the original purpose of noise gates

As with limiters, noise gates are primarily used as problem-solvers. Gating was originally used to compensate for hissy tape tracks before (and even after) the days of noise reduction circuitry, background noise in recording studios and chambers, ambient noise in remote recordings, and other sonic artifacts which needed filtering out of the final track. It's unlikely any of those "artifacts" will ever disappear. The sound of hands settling themselves on guitar strings might be exceptionally quiet compared to the sound produced by the strings through the pickups, but as recording gear becomes more sensitive, ever-quieter sounds find ways to intrude on recordings and irritate both engineers and listeners.

Modern consumer audio equipment is now so sophisticated that noise is virtually inexcusable in a commercial recording. Gates are so universally used in modern recording that it's not unusual to find 24-track analog studios with a rack of high-quality outboard noise gates for almost every channel. In the digital domain, gating can be done so cleanly and usually with so little risk of deterioration of the input signal, that it can be applied to virtually every track that doesn't originate as an already-clean digital audio signal. When a digital gate is open, the original signal passes through completely unaffected, so the only possible risk of deterioration occurs to low-level attack and decay slopes. But when an analog gate is open, there is still at least one op-amp through which the signal must pass, so there *is* risk of fidelity loss in this case.

Gating as an effect

In modern recording, noise gates are much more than hiss-killers, style-correctors and "smother blankets" for background noise. Applying a noise gate to digital reverb effects was a favorite engineer's trick in the late 1980s to punch up snare drums and other percussion. Many newer dance and pop tracks use triggered noise gates to add a unique flavor to guitar and vocal tracks. For example, the vocal or guitar might only be allowed through the gate when snare or synthesizer volume rises past the threshold, giving the gated instrument a harsh, sampled-sounding envelope. In this application, the vocal or guitar channel's noise gate is triggered open and shut by the level of an entirely different track.

Triggered gating has a wide range of uses, some more dramatic than others. For an interesting effect on a break or refrain, set up the mixer so that a background vocal or synth pad is on the channel to the right of the snare or hi-hat track. Select **Left (Post)** or **Left (Pre)** as the **Trigger.** The hi-hat's volume envelope will then trigger the vocal or synth pad, adding an attention-grabbing percussive feel to that track.

Another common noise gate effect with a heritage of more than a generation is attack filtering. Try setting the attack on a gate from 50ms to 200ms and apply it to a stringed instrument. This softens the attack of the instrument considerably, giving it a “bowed” sound.

Ducking

Making way for the voice

Ducking is a type of triggered compression, but works in reverse from the sort of triggered gating discussed in the noise gates backgrounder. Ducking involves compressing a signal when the trigger signal reaches a certain level. It’s a relatively new type of dynamics effect that didn’t become widely used until about the mid-1960s. Radio stations began to use this trick at that time as a means of allowing disc jockeys to speak over a backing music track without having a sense of dead or thin air when the dj wasn’t speaking. When the level of the dj’s voice passes a certain level, the music track drops in volume accordingly, or “ducks underneath” the dj’s voice. When the dj isn’t talking, the music rises again to a normal broadcast level. The old joke has it that duckers were invented so that dj’s wouldn’t have to put down their cigarettes to turn down the background music when they spoke.

Duckers are now essential devices in virtually all types of spoken-word production. You’ll hear ducking used in a lot of television commercials to even out the dynamics of the audio track and allow the announcer’s message to take clear priority over the music at the times when the announcer is speaking. While ducking is far from being transparent in this type of application, the mark of effective ducking is making it *seem* transparent. With analog devices, this often meant modifying settings several times during the mix to account for changes in the levels of the background track. With newer digital devices, much of this work can be automated.

Ducking as an effect

Ducking can also be used as a “presence enhancer” for specific tracks, primarily for percussion instruments. The gain of certain submixes, e.g. pads and sound effects, could be triggered by a kick drum to duck, or drop in volume, for the short duration of the kick’s sound. This seems to give the kick drum a higher priority in the mix for the brief moments when it is heard, adding presence to the kick without applying gain or compression. It’s an effect you may have heard on a number of dance tracks. When done correctly, it creates an interesting illusion in which the ducked tracks don’t seem to fall in volume because the track controlling the ducking dominates the mix.

Effective ducking in a broadcast context will produce a similar illusion. When done *in*correctly, especially using the wrong threshold and release time values, you hear a distinct attack slope or “breathing” effect as the ducked track rises during quiet passages in the controlling signal’s track. It’s an obtrusive and annoying effect, and something you’ve probably heard watching sportscasts or small-market TV news broadcasts. But it’s an effect nonetheless and has been applied with success by many engineers in pop recordings.

Range expansion

Putting back the peak

As you might expect, expansion is the opposite of compression. What a compressor does to even out the dynamics of a signal, an expander does to...well...*expand* them. Sounds below a certain level won’t generally have their dynamic range expanded, since that would make some sections of the material too quiet. But sounds *above* a certain level will have their relative gain multiplied just as compressed signals have the gain of high-level passages divided.

Expansion is often used to compensate for the dynamic range lost when recording to tape or passing an instrument through tube circuits which compress the signal. One common use for expansion is for increasing the apparent dynamic range of legacy tape tracks or low-resolution digital samples converted to hi-res digital tracks. When applied in the right way using a high-quality expander, the resulting track will seem to have lost some of its compression and will sound more “alive”. When applied *in*correctly, it makes the expanded track sound thin.

Judicious use of expansion can also impart an illusion of enhanced fidelity even to relatively well-recorded tracks, but it has to be used in the right context. Expansion has little application in most pop music, since the goal is generally to increase the presence and perceived balance in the mix by bringing instruments forward. Since range expansion increases the maximum level of peaks in a track, most expanded signals need to be pushed *back* in the mix so that they don’t clip.

On the other hand, expansion can be a godsend for live chamber recordings or recordings made on cheap portable gear. It can help restore life to a campfire guitar performance made on a ghetto blaster, add realism to voice or instrument tracks that may have been recorded using the wrong mic at too close a range, and even accentuate a player’s dynamics on overdriven guitar tracks as long as the signal isn’t too heavily distorted.

Expansion as an effect

Expansion can also be applied as a distinctive effect to dramatically increase dynamic range. While this might sound great on high-end studio monitors, the results are often disagreeably artificial-sounding on low-end to mid-priced consumer audio systems. All too often the effect loses its impact if it’s played on anything but quality audio gear, since inexpensive consumer audio gear tends to introduce its own compression and/or limiting characteristics into the sound on playback.

Companding

Companding facts and a case of mistaken identity

Companding, at least in terms of the implementations used by some manufacturers’ devices, is actually what many people think *compression* is (see compression myth #3). Companding is, as you would expect from the name, a combination of compression and expansion. Sounds *below* the specified threshold level are expanded in volume proportionate to the threshold, while sounds *above* the specified level are compressed.

If you’re new enough to recording not to fully understand what compressors do, you’re likely to find that this effect comes closer to what you expect from a compressor than the actual compression effect. With that in mind, give it a try and see if it meets your imagined expecations of a compressor. But if you’re already familiar with compression theory and you haven’t tried the compander yet, imagine the, er, expanded range of possibilities inherent in a dynamics processor that can raise the level of quiet passages as well as reduce the level of loud ones.

Companding has useful applications virtually every type of recording and mixing. It’s most commonly used to enhance field recordings of dramatic situations where signal levels may rise and fall dramatically over short periods of time. Where ducking is ideal for forcing one track down when another needs to be prominent, companding is more effective for managing background sound in live field recording. A compander can help increase the presence of ambient street or field noise where distant mic placement is used to capture audio signals with a fairly high dynamic range.

In studio recording, it can be used to increase the presence of instruments with wide dynamic ranges. Applying companding to a piano, for example, allows pianissimo sections to retain the subtlety of tone of gentle hammer strikes while preventing fortissimo passages in the same piece from redlining the level meters. The same is true of electric and acoustic guitars. In these cases, the tonal subtlety of quiet passages can be preserved while the overall dynamic range of the track is reduced.

A little trivia: the companders you know and love

If you’ve ever been involved in CB (citizens’ band), shortwave or ham radio, or dealt with private narrow-band radio systems for cab companies, messengers or the like, you already know the term, since you’ll see “companding” in the specifications for virtually all transciever hardware. Companding the signal in narrow-band transmissions effectively allows for a much higher intelligibility, especially on the tiny speakers typically used in dispatch and dash-mount radio units. Companding has been used for decades to improve the sound quality in these types of hardware, both as a single-ended process and as a double-ended process.

It's double-ended companding with which you are likely most familiar. Because it is sophisticated double-ended analog companding which lies at the heart of most noise reduction systems used in tape recording and playback, right from the earliest Type A Dolby noise reduction to the newer Type C Dolby used in consumer recording equipment and dbx technology used in many popular cassette multitrack decks. Here's how companding works to reduce noise.

Take the average home cassette deck. The inputs themselves are noiseless (okay, let's pretend the world is perfect and *assume* they are) and the internal circuitry supports a relatively wide dynamic range. Given these conditions, the signal that goes to tape should retain all the quality of the input signal, shouldn't it? We know that doesn't happen. No matter how clean the input signal, when you play back your newly-recorded cassette you will hear low-level hiss and a reduction of dynamic range, even if you're using the finest tape formulations on the market. The hiss is an artifact of the recording process, and there's nothing you can do to eliminate it.

Or is there? What if you companded the signal before putting it on tape? This would allow you to record the input signal to tape at a hotter level since peaks would be compressed. Low-level passages would also be louder. In order to hear the material as it would recorded though, you would need to apply inverse compansion to the signal to "decode" it, meaning that the dynamic range of loud passages must be restored and low-level signals must be pushed back down to appropriate levels.

Apply this double-ended companding correctly and you can indeed get the dramatic improvement in signal-to-noise ratio that dbx and Dolby C boast. Firstly, you're recording the input at a hotter level thanks to compression, meaning that on playback the noise floor of the tape hiss is reduced as range is expanded. Secondly, you're pushing the noise floor down even further by expanding the range of low-level signals. Any quiet passages above the noise floor of tape hiss appear to be even farther above the noise floor on playback. So both loud *and* quiet passages in the recorded material benefit from this double-ended companding.

Make no mistake about it, there are few sensible reasons for using noise reduction techniques like these in digital recording. You shouldn't need to do this sort of thing with your own tracks. Besides, as versatile as some manufacturers' companders may be, you simply can't duplicate Dolby circuitry using a compander. While Dolby and dbx both use companding as the basis for their noise reduction schemes, they are far from simple compressor/expander devices. Increased compression of certain frequencies, selective phase-inversion and other types of electronic wizardry all combine to produce the total package and the effects you hear on recorded material that uses noise reduction.

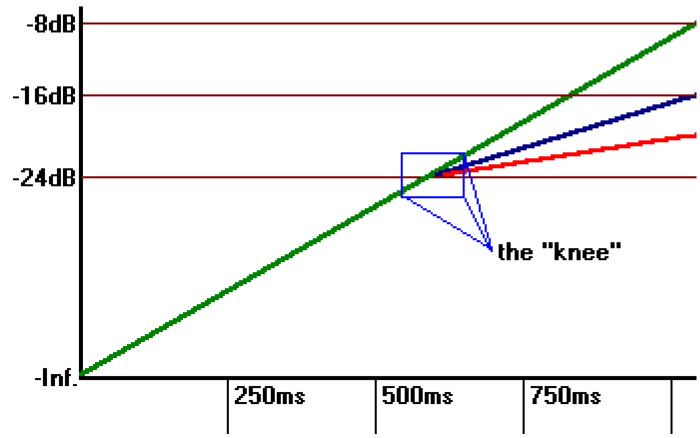
Hard knee vs. soft knee

It is often difficult for novice engineers to grasp the difference between hard knee and soft knee dynamics processing. This difficulty arises in part because the effect is rather subtle unless relatively dramatic processing is applied to the signal.

First let's review how the compressor does its job. Think of the diagrams used below as representing the time/gain curve of a slow-rising string pad that needs a full second of attack to reach its peak level. Assume for the sake of this example that its peak level is 2.5dB below clipping. Most compression diagrams you'll see use curves that represent sounds similar to our imaginary synth pad; the actual time/gain curves of non-electronic instruments look much different from these typical examples.

Hard knee

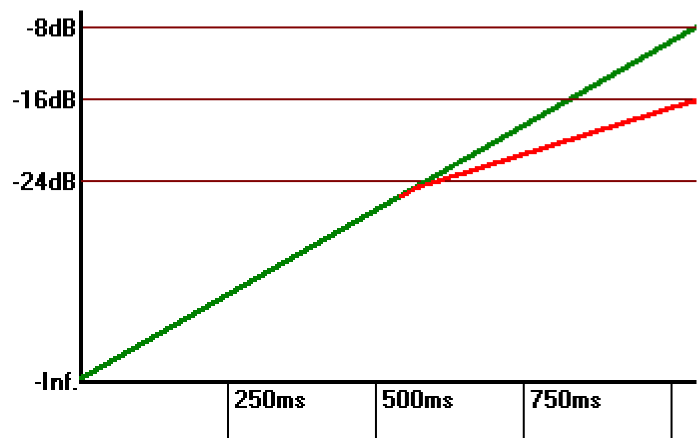
Note in the illustration below how compression begins instantly when the signal's level passes -24dB. This is the typical behavior of solid-state limiters and compressors. The blue line represents 2:1 compression, meaning that for every dB above -24dB that the input signal rises to, the compressor wil cut 1dB of gain. The green line indicates the input signal's actual volume. Note also how there is *no* blue line below the -24dB threshold level. Below this level, the compressor has zero effect on the signal and allows it to have its normal dynamic range.



Without compression, the synth patch's volume rises in a straight line to its maximum level. With 2:1 compression (the blue line), the compressor *instantly* starts allowing only 1dB of gain through the output for each dB of actual gain in the input signal as soon as threshold level is reached. (This assumes an attack time on the compressor of 0ms). The output signal with 2:1 compression only reaches -16dB when the input signal's level reaches -8dB. With 4:1 compression, the output signal is only allowed to rise to -20dB when the input signal has reached -8dB.

Notice how sharply the slope turns when it reaches the threshold level. To the trained ear, this slope will sound highly unnatural, like a two-stage attack envelope. Even those who can't hear this sudden change in the gain curve will still *feel* something artificial about the sound.

Soft knee



This simpler illustration, typical of tube-emulation-type compressors, demonstrates how soft-knee compression changes the attack slope of our string pad to make it more natural-sounding. It actually starts compressing the sound *before* it reaches the -24dB level, but it starts compressing with a *lower ratio* than normal, applying more compression as the input gain rises. It doesn't perform full compression until after the input signal has actually reached the threshold level. While the curve in this illustration is subtle, the arc in the slope is visually "softer" than the slopes in the hard knee examples.

Note that this curve shows a tube *emulation* compressor. The knee bends *before* the signal reaches the threshold level and doesn't stop bending until it is slightly above threshold. In many tube compressors, the knee actually starts to bend *after* the signal reaches the threshold level, meaning the compressor actually kicks in at threshold at less than the specified compression ratio. In some soft-knee solid-state outboard units, the knee is locate completely below the threshold level.

Also note that the length and angle of the knee's arc is variable. Some inexpensive soft-knee compressors produce a relatively short arc, especially at low threshold levels (e.g. -36dB, which is not atypical for mic'd guitars) and don't provide as gradual an increase in compression ratio.

Reshaping the knee

The width of the arc is an important consideration in the design of soft-knee compressors. There's a fine line between having the ratio increase too gradually and having it increase too rapidly. Some types of audio material, especially fast-rising signals that drop in and out of the track frequently (e.g. percussion and stabs), lend themselves to relatively short arcs (a "harder" knee) while others seem to sound more pleasing with a very gradual arc. With this in mind, some newer compressors allow you to define the width of the knee or, in some cases, even draw your own custom knee curve.

DYNAMICS PROCESSORS PARAMETER REFERENCE

Compressor

The 01V's DSP compressor is a versatile and effective digital compressor/limiter offering a range and depth of features usually found only on top-quality solid-state analog outboard units. The variable knee setting makes it well-suited to use on virtually all types of audio material. This is *accurate* compression which doesn't color the sound. It is not intended to sound exactly like high-end tube-based compressors or limiters.

The 01V compressor can also be configured to act as a peak limiter by setting the **Ratio** to **inf:1.**

Compression and limiting in the digital domain requires a little more thought than compression in the analog domain. This is because "small amounts" of clipping of digital signals is generally far nastier than a similar amount of clipping in an analog recording environment. Generally speaking, while peak limiting can be useful when recording tracks to insure higher dynamic range and more headroom while preserving "clean" sound, compression should probably be avoided until you are ready to mix the tracks. Because compression is always available on all channels, you should probably dedicate the dynamics processor on your inputs to peak limiting or noise gating.

Threshold

This parameter specifies the level at which compression will kick in. Threshold levels for dynamics processors are referenced downward from 0dB, or if you prefer, as negative values relative to the maximum level before the compressor's circuitry begins to distort (clip) the signal.

The trick to using compression (some engineers consider it an art, or at very least a skill requiring practice and refinement) is finding the ideal balance between compression ratio and threshold to produce the desired effect. Natural-sounding compression depends on matching the two parameters in a way that preserves the "feel" of a track's dynamics while reducing peaks or increasing presence to a desirable degree.

If the threshold is set too low, the signal may sound "squashed", as if it is being squeezed out of an underpowered system such as a portable stereo. (Low threshold levels combined with high ratios can be used as an effect in some circumstances.) Setting threshold too high defeats much of the purpose for compression, since little or none of the signal will be compressed.

Peak limiting, on the other hand, is relatively easy to master. It can usually be configured by watching the channel meters during an audition and making parameter decisions based on what the meters say about the input signal's maximum levels. In most cases, effective peak limiting does not consist of setting the threshold at a desired value and setting the ratio to **inf(inity):1.** To preserve a natural sound in the peaks of the signal, the threshold usually needs to be brought down several dB *below* the target maximum level and the ratio adjusted to insure that the highest peaks barely reach the maximum desired level. Setting an appropriate knee shape will further assist in generating peak limiting that doesn't sound artificial or forced.

Triggered compression/EQ effects (e.g. de-essing) are not possible with the 01V's available routing.

Range:	0dB - -54dB
Increment:	1dB
Default:	0dB

Ratio

This parameter determines how much the input signal's gain will be reduced when it passes the threshold level. For example, a ratio of 4:1 will reduce a signal's gain by 3dB if its gain is 4dB above the threshold level (i.e. allow only 1dB of output gain for every 4dB of input gain); by 2dB if gain is 8dB above threshold, etc. A ratio of **inf:1** turns the compressor into a "hard limiter", allowing zero gain above the threshold level. A ratio of **1:1** performs no gain reduction at all, essentially shutting off compression.

Range:	1:1 - Infinity:1
Increments:	1:1, 1.1:1, 1.3:1, 1.5:1, 1.7:1, 2:1, 2.5:1, 3:1, 3.5:1, 4:1, 5;1, 6:1, 8:1, 10:1, 20:1, Infinity:1
Default:	1:1

Knee

This determines the shape of the knee at the point at which compression occurs. The **Soft** settings produce a progressively wider arc in the gain curve as the input signal approaches the threshold level. As the knee's arc is widened from from **Soft 1** to **Soft 5,** the compressor generally produces a more natural-sounding gain reduction since each increase in the size of the arc results in compression beginning a little farther below the threshold level. The wider the arc, the farther below the threshold level that gain reduction will begin. The knee in Yamaha digital effectors begins to bend below threshold and completes the arc above threshold.

Options:	Hard, Soft 1, Soft 2, Soft 3, Soft 4, Soft 5
Default:	Hard

Attack

This parameter specifies how fast compression or limiting will be applied to the input signal once it passes the threshold level.

Instruments such as guitars and percussion (pianos, drums, etc) with steep attack/initial-decay curves tend to lose definition and "punch" when the attack value is set too low. On the other hand, setting the value too high may result in undesirable sharp transient peaks. Note that the width of the knee arc also plays a role in producing a natural-sounding attack in these instruments, so you can often use a slightly longer attack when applying a softer knee.

Because an 01V compressor does its job by performing calculations on a digital signal, it allows a *true* zero-attack-time setting. This means that a signal which starts at peak input gain can be compressed from the very first sample. This is useful to know when assigning compressor parameters based on a visual inspection of the WAV file or input data stream. "Instant attack" is not possible in analog compressors. By nature of their design, analog dynamics processors always have slight delays before compression or limiting can kick in, although high-quality circuit components may produce a compressor capable of very short attack times.

Range:	0ms - 120ms
Increment:	1ms
Default:	0ms

Release

This parameter specifies the total time the compressor will be allowed to continue compressing the signal when it falls below the threshold level. Long release times allow compression to be applied to the signal even if the input signal falls well below the threshold level. In other words, it specifies how long the compressor waits after the last input peak before reducing the compression ratio.

A compressor has to allow the signal to decay naturally or the result is an unnatural-sounding sustain effect. A poorly-selected release time will result in a "shelf" in the decay curve, an audible moment when the signal drops below the threshold level and the volume seems to "dive". This effect is often only audible when the channel is solo'd, but it is often "felt" by listeners even if it is not plainly audible in a mix.

Release does *not* invert the function of the device when gain falls below threshold. This means that longer release times won't produce an audible *in*crease in gain as the signal level falls farther below threshold.

Range:		8ms - 63.5s
Increment:	8ms - 264ms:	8ms
	264ms - 520ms:	16ms
	520ms - 1.03s:	32ms
	1.03s - 2.06s:	64ms
	2.06s - 4.10s:	128ms
	4.10s - 8.20s:	256ms
	8.20s - 16.4s:	512ms
	16.4s - 32.8s:	1024ms
	32.8s - 63.5s:	2048ms
Default:		8ms

Gain

The perceived loudness of the input signal may drop as the signal's gain is compressed, resulting in a need to increase gain at the output of the compressor. This parameter is normally not adjusted when simple peak limiting is required, since the goal is to control transient peaks. When actual compression is desired, you will often need to increase gain to compensate for this perceived loss of volume.

Since this gain control is part of a combined calculation within the dynamics processor, you *may* be able to reduce the risk of signal degradation on some tracks by using this gain control as the attenuator (pre gain control) for a track rather than the fader control (slider) on the channel strip. Gain adjustments count as a digital process. Whenever you can eliminate a process from the signal chain, you produce a corresponding reduction in the risk of signal degradation through repeated processing. Each channel's EQ, dynamics and gain controls are calculated as a three separate equations so that means three processes applied to your input signal.

Range:	0dB - 18dB
Increment:	0.5dB
Default:	0dB

Noise Gate

The 01V noise gate is a relatively complete gating device. The available threshold range of 54dB should be more than sufficient for all but the most demanding applications.

There are two factors you will need to consider when deciding where and when to apply noise gates. Firstly, the noise gate is not available as a separate effect; you can apply *either* compression/ducking/companding *or* gating, and not both, unless you route the channel's output signal into another channel of the mixer. Secondly, there is no knee control. While variable knee settings are available for compression, ducking and expansion, no variation in knee shape is available for the noise gate. This appears to be a hard-knee gate, which may limit its applicability to more delicate musical signals. This should not be a great problem, since noise gate plug-ins used by most hard disk recording applications use very little CPU time and have a negligible effect on potential signal degradation. When you need both types of processing on a track, you may wish to leave gating to a plug-in and allow the 01V to do other types of dynamics processing.

Threshold

This parameter specifies the level at which the gate opens. If you set the threshold value higher than a typical level for the track, then little or no signal will pass through the gate. Generally speaking, unless you intend to apply gating as a sound effect, the level should be set a few dB above the noise floor of the track (the level of the loudest background noise you are attempting to gate), or just far enough above the background noise level to give you a comfortable margin for error if the background noise level tends to rise and fall.

Careful one-time setting of a noise gate's threshold may not be enough to manage tracks with background noise that rises and falls in volume. You may need to "massage" the track in an audio editor or use C-Console's automation features to alter the threshold level at various points in the track.

On the other hand, setting a gate's threshold too high may have adverse effects on the volume envelope of the output signal, producing a "swell" effect or a loss of some of the track's attack characteristics. See the **Compressor** section for additional information.

Range: 0dB - -54dB
Increment: 1dB
Default: 0dB

Range

This parameter controls how "wide" the gate will be. While only signals above the threshold level are allowed to *open* the gate, you normally won't want to *close* the gate at the same level. Signals with long decay curves such as pianos and sustaining electric guitars might "dive" on you if the range isn't set large enough. Range control is provided to insure that as the signal decays, the gate can be closed at a much lower level than that required to open it.

Range is measured from the threshold to the point at which the noise gate closes. 70dB (the total range) + 54dB (the total range for the threshold) = 124dB, or approximately the total theoretical dynamic range of the 01V. Exceptionally high range values will tend to make decay settings inaudible. At the default value of 0dB, the hold and decay settings have full control over the decay characteristics of the output signal.

Range: 0dB - -70dB
Increment: 1dB
Default: 0dB

Attack

This parameter determines how gradually the gate is opened when the gate detects a signal above the threshold level. The default value of **0ms** is also the normal value for this parameter.

Careful selection of attack values can be used both as a corrective effect and as a sound effect. For example, at 10-16ms of attack, the initial sharp attack of a percussion or plucked instrument can be attenuated or "softened" without completely wiping out the attack. At higher values, unnatural attack effects or "swells" can be achieved.

The 01V's noise gate does its job by performing calculations on a digital signal, so it allows a *true* zero-attack-time setting. This means that a signal that starts at peak input gain can be passed through an open gate from the very first sample with no attack. This is not possible in analog gates which, by nature of their design, always have some attack when the gate first opens. If you've been annoyed by this attack effect in analog gates, you should find it blessedly lacking in the 01V.

Range: 0ms - 120ms
Increment: 1ms
Default: 0ms

Hold

The **Hold** parameter defines a specific length of time the gate will remain open, regardless of signal level, before the device checks to see if the signal has dropped below the lower limit of the gate range. This allows a degree of control over a sound's decay characteristics that the decay parameter alone cannot provide, since no gain reduction is applied during the period defined by this parameter. Increasing this parameter's value can allow you to keep the gate open long after the signal has fallen below the track's noise floor. This can actually improve some tracks with wide dynamic ranges, but in many cases it will result in unwanted noise creeping into the track as signal levels fall below the noise floor.

Range:		0.03ms - 2.94s
Increment:	0.03ms - 1.09ms:	0.03165ms
	1.09ms - 2.03ms:	0.0625ms
	2.03ms - 4.03ms:	0.125ms
	4.03ms - 8.03s:	0.25ms
	8.03ms - 16ms:	0.5ms
	16ms - 32ms:	1ms
	32ms - 64ms:	2ms
	64ms - 128ms:	4ms
	128ms - 256ms:	8ms
	256ms - 512ms:	16ms
	512ms - 1.02s:	32ms
	1.02s - 2.05s:	64ms
	2.05s - 2.94s:	128ms
Default:		0.03ms

Decay

Decay time is a measure of how long it takes for the sound to drop to 0dB from the moment the gate begins to close. This parameter allows the gate to be closed gradually, helping to impart a more natural decay to the signal. Without decay control, the gate would “slam” shut and an instant drop in level to zero would be heard as the gate closes. Imagine the effect this instant drop to zero volume could have on the decay characteristics of an overdriven guitar track and you’ll understand how important it is to select an appropriate value.

The **Decay** period begins only after the **Hold** period has expired.

Range:		8ms - 63.5s
Increment:	8ms - 264ms:	8ms
	264ms - 520ms:	16ms
	520ms - 1.03s:	32ms
	1.03s - 2.06s:	64ms
	2.06s - 4.10s:	128ms
	4.10s - 8.20s:	256ms
	8.20s - 16.4s:	512ms
	16.4s - 32.8s:	1024ms
	32.8s - 63.5s:	2048ms
Default:		8ms

Ducker

The ducker is a specialized noise gate algorithm, and uses the same parameters and default values as the gate. The primary difference between the ducker and noise gate is that the ducker depends upon a trigger signal to open and close the gate.

The signal on the channel that uses the ducker will have no effect on the ducker. If you don’t select a **KEY-IN** channel, the ducker simply won’t work. Once a **KEY-IN** channel is selected, all gating parameters will be controlled by the trigger signal which feeds the **KEY-IN.** This means that all parameter settings need to be selected with the dynamics of the trigger signal’s track in mind.

Put another way, if you are applying the ducker to Channel 3 using a **KEY-IN** signal from **AUX1**, then you are using Channel 3’s dynamics processor. But the track which you are feeding into **AUX1** (e.g. Channel 8) is the track that actually controls the gate. This makes Channel 3 the *processed* signal, while Channel 8 acts as the *trigger* signal (assuming that Channel 8 is feeding the selected **KEY-IN** channel).

Threshold

This parameter specifies the level the trigger signal's track must reach before the gate on the processed signal's track can open. See the **Noise gate** section for additional information.

Range:	0dB - -54dB
Increment:	1dB
Default:	0dB

Range

This parameter controls how "wide" the gate will be. See the **Noise gate** section for additional information.

Range:	0dB - -70dB
Increment:	1dB
Default:	0dB

Attack

This parameter determines how gradually the input signal's gate is opened when the trigger signal's level rises above the threshold level. See the **Noise gate** section for additional information.

Range:	0ms - 120ms
Increment:	1ms
Default:	0ms

Hold

The **Hold** parameter defines a specific length of time the gate will remain open, regardless of the trigger signal's level, before the device checks to see if the signal has dropped below the lower limit of the gate range. See the **Noise gate** section for additional information.

Range:		0.03ms - 2.94s
Increment:	0.03ms - 1.09ms:	0.03165ms
	1.09ms - 2.03ms:	0.0625ms
	2.03ms - 4.03ms:	0.125ms
	4.03ms - 8.03s:	0.25ms
	8.03ms - 16ms:	0.5ms
	16ms - 32ms:	1ms
	32ms - 64ms:	2ms
	64ms - 128ms:	4ms
	128ms - 256ms:	8ms
	256ms - 512ms:	16ms
	512ms - 1.02s:	32ms
	1.02s - 2.05s:	64ms
	2.05s - 2.94s:	128ms
Default:		0.03ms

Decay

Decay time is a measure of how long it takes for the sound to drop to 0dB from the moment the gate begins to close. See the **Noise gate** section for additional information.

Range:		8ms - 63.5s
Increment:	8ms - 264ms:	8ms
	264ms - 520ms:	16ms
	520ms - 1.03s:	32ms
	1.03s - 2.06s:	64ms
	2.06s - 4.10s:	128ms
	4.10s - 8.20s:	256ms
	8.20s - 16.4s:	512ms
	16.4s - 32.8s:	1024ms
	32.8s - 63.5s:	2048ms
Default:		8ms

Expander

“Expander” is short for *dynamic range expander*. The range expander in the 01V is functionally equivalent to the compressor except that it applies range *expansion* rather than compression to the input signal.

Range expansion is an increasingly overused effect as digital recordists try to take full advantage of the dynamic range available in their hardware. In most cases, use of this effect should be avoided until final mixing to insure that it is actually needed. While the concept of range expansion seems to point to its ability to improve the quality of legacy tracks recorded on less capable gear, or signals that lack dynamic range, in practice the process is often overused and unnecessary, and often results in a track which sounds excessively punchy and irritating.

Threshold

This parameter specifies the level at which range expansion will kick in. See the **Compressor** section for additional information.

Range:	0dB - -54dB
Increment:	1dB
Default:	0dB

Ratio

This parameter determines how much the input signal’s gain will be increased when it passes the threshold level. See the **Compressor** section for additional information.

Range:	1:1 - Infinity:1
Increments:	1:1, 1.1:1, 1.3:1, 1.5:1, 1.7:1, 2:1, 2.5:1, 3:1, 3.5:1, 4:1, 5;1, 6:1, 8:1, 10:1, 20:1, Infinity:1
Default:	1:1

Knee

This determines the shape of the knee at the point at which expansion starts. See the **Compressor** section for additional information.

Options:	Hard, Soft 1, Soft 2, Soft 3, Soft 4, Soft 5
Default:	Hard

Attack

This is a critical parameter in range expansion when working with plucked or percussive instruments. These instruments tend to have a fast-rising attack to a very loud peak, and this peak falls relatively rapidly (usually with a logarithmic curve) as the signal begins to decay. Applying “instant” expansion to instruments with this type of envelope can easily push the signal to clipping as the initial peak is reached. Setting the attack to a higher value can allow expansion to kick in *after* the initial peak is reached, resulting in a less obtrusive effect. Allow your ears to be your guide to whether a longer attack time actually improves the sound of a range-expanded signal. See the **Compressor** section for additional information.

Range:		8ms - 63.5s
Increment:	8ms - 264ms:	8ms
	264ms - 520ms:	16ms
	520ms - 1.03s:	32ms
	1.03s - 2.06s:	64ms
	2.06s - 4.10s:	128ms
	4.10s - 8.20s:	256ms
	8.20s - 16.4s:	512ms
	16.4s - 32.8s:	1024ms
	32.8s - 63.5s:	2048ms
Default:		8ms

Release

Release time is as critical as attack time when you need to insure sweet, listenable range expansion. Judicious setting of the release parameter can help insure a natural-sounding decay curve and expansion in the “right places” in the track’s envelope. See the **Compressor** section for additional information.

Range:		8ms - 63.5s
Increment:	8ms - 264ms:	8ms
	264ms - 520ms:	16ms
	520ms - 1.03s:	32ms
	1.03s - 2.06s:	64ms
	2.06s - 4.10s:	128ms
	4.10s - 8.20s:	256ms
	8.20s - 16.4s:	512ms
	16.4s - 32.8s:	1024ms
	32.8s - 63.5s:	2048ms
Default:		8ms

Gain

The perceived loudness of the input signal may rise as the signal’s gain is expanded. This may result in a need to attenuate the gain from the channel fader to insure that expansion does not push the signal to clipping. After expansion, you may need to fine-tune the output using this gain control. Note that while the compressor’s **Gain** control *adds* gain, this control on the expander acts as an attenuator to reduce output level. See the **Compressor** section for additional information.

Range:	0dB - -18dB
Increment:	0.5dB
Default:	0dB

Compander High/Compander Low

The compander algorithms are single-ended companders designed to compress high-level signals and expand the range of low-level ones. They are *not* noise-reduction units, although they might sound like it with some material. Above the specified threshold, they compress the track. In the middle of the output range is a “width” area in which no processing is done. Subtract the width amount from the threshold level and you have the level at which expansion occurs. The expansion of low-level signals is designed to enhance the dynamics of low-level signals by making quiet sounds appear quieter...or louder, depending on the “quiet” level...than they actually are.

This can be useful for accentuating dynamics in normally "tight" instruments such as overdriven guitars but the application of companding should always be done with attention to the entire mix. In order to get a clearer picture of how the companders behave, apply Compander High (**CMPH**) to a highly expressive synth patch or to both overdriven and clean electric guitar sounds to see how it responds to dynamics.

The **H**(igh) and **L**(ow) variants of the compander refer to the degree of expansion applied. **CMPH** applies 5:1 expansion below the threshold-minus-width level, **CMPL** applies a much more subtle 1.5:1 expansion.

No knee control is provided for the companders; none should be needed. Since the same ratio applies both above and below the threshold, no sloping of gain/attenuation is needed as the signal approaches threshold.

While the companders *can* be used as "loudness maximizers", it may be best to limit application of the compander algorithms to single channels rather than attempt to apply them as "mastering" processors to a whole mix due to the potential harm they can do to low-level passages.

Threshold

This parameter specifies the level at which compression starts. Signals below the threshold level pass through unmodified; signals above threshold are compressed. See the **Compressor** section for additional information.

Range:	0dB - -54dB
Increment:	1dB
Default:	0dB

Ratio

This parameter determines how much the input signal's gain will be decreased when it passes the threshold level or increased if it is below threshold. See the **Compressor** section for additional information. Ratio has no effect on the expander.

Range:	1:1 - Infinity:1
Increments:	1:1, 1.1:1, 1.3:1, 1.5:1, 1.7:1, 2:1, 2.5:1, 3:1, 3.5:1, 4:1, 5;1, 6:1, 8:1, 10:1, 20:1, Infinity:1
Default:	1:1

Width

The width control is provided for creating a volume range in which no compression *or* expansion occurs. While the total dB of compression depends on the total dB between threshold and 0dB, the total dB of expansion doesn't have to be referenced from -124dB (the theoretical limit of the 01V using an all-digital signal path). Instead, you can set the width so that expansion occurs only in the range of 0 output to the level defined by the equation **Threshold minus Width**. Large width values tend to impart a more subtle range expansion effect; narrower ranges will result in the output level falling off audibly more sharply as the input level drops.

As you might notice when tweaking this parameter in the **Dynamics Editor,** the range of the width parameter is related to the threshold level. The lower the threshold, the narrower the available width.

Range:	0dB - 90dB
Increment:	9dB
Default:	90dB

Attack

This is a critical parameter in range expansion when working with plucked or percussive instruments. These instruments tend to have a fast-rising attack to a very loud peak, and this peak falls relatively rapidly as the signal begins to decay. Applying "instant" expansion to instruments with this type of envelope can easily push the signal to clipping as the initial peak is reached. Setting the attack to a higher value can allow expansion to kick in *after* the initial peak is reached, resulting in a less obtrusive effect. Allow your ears to be your guide to whether a longer attack time actually improves the sound of a range-expanded signal. See the **Compressor** section for additional information.

Range:		8ms - 63.5s
Increment:	8ms - 264ms:	8ms
	264ms - 520ms:	16ms
	520ms - 1.03s:	32ms
	1.03s - 2.06s:	64ms
	2.06s - 4.10s:	128ms
	4.10s - 8.20s:	256ms
	8.20s - 16.4s:	512ms
	16.4s - 32.8s:	1024ms
	32.8s - 63.5s:	2048ms
Default:		8ms

Release

Release time is as critical a parameter as attack time when you need to insure a sweet, listenable sound in a range-expanded signal. Judicious setting of the release parameter can help insure a natural-sounding decay curve and expansion in the “right places” in the input signal’s envelope. See the **Compressor** section for additional information.

Range:		8ms - 63.5s
Increment:	8ms - 264ms:	8ms
	264ms - 520ms:	16ms
	520ms - 1.03s:	32ms
	1.03s - 2.06s:	64ms
	2.06s - 4.10s:	128ms
	4.10s - 8.20s:	256ms
	8.20s - 16.4s:	512ms
	16.4s - 32.8s:	1024ms
	32.8s - 63.5s:	2048ms
Default:		8ms

Gain

The perceived loudness of the input signal may rise as the signal’s gain is expanded. This may result in a need to attenuate the gain to insure that expansion does not push the signal to clipping. After expansion, you may need to fine-tune the output using this gain control. Note that no gain reduction is possible here. See the **Compressor** section for additional information.

Range:	0dB - 18dB
Increment:	0.5dB
Default:	0dB

EFFECTS BACKGROUNDERS

About the backgrounders

Before we begin...

Let’s be fair about this. If you’ve invested enough that you have an 01V and the equipment to make the most of it, we have to assume that you have at least *some* basic knowledge of recording and audio terminology. This documentation includes some relatively basic but *somewhat* technical background on the effect types available in the 01V. If you don’t have this basic grounding, you might find this section of the docs rather rough going. We had to strike a balance between providing concise, meaningful information and speaking to the widest possible range of 01V/C-Console users. Sadly this has to come at the expense of those who may be new to recording and audio.

It could also leave some readers feeling a bit cheated. C-Console products cover a wide range of hardware devices, and not all are capable of the same level of effects versatility. We chose to leave this section intact for all C-Console versions, since we believe engineers and recordists will be curious to know about every popular effect type even if their hardware doesn’t currently support it.

If you need additional information or background material on any of these effects or how they are used in recording, there are several superb sites on the web where you can learn about basic recording concepts, audio terminology, and engineering techniques. We recommend everyone try a few well-chosen keywords in their favorite search engine to find out just how much quality information there is out there, but we especially recommend this to novice recordists.

Digital Domain at *http://www.digido.com* has a particularly well-respected selection of essays on pro audio, and Harmony Central (*http://harmony-central.com/Effects/*) maintains an excellent all-purpose resource on effects theory and effects devices that we recommend to all recordists.

This section is aimed at those with basic knowledge, but we’re not leaving out veterans either. We’ve tried to pack this section with interesting historical tidbits, “biz gossip” and other trivia. We also peppered this section with some of our best tricks and most hard-won knowledge to make it worth your time to give it a good browse.

What’s covered here

You know what unity gain means. You don’t confuse “arming a track” with a weapons offence. You’ve never run your speedboat to ground in a patchbay. Even if you’ve got the basics of recording down pat, it’s possible - perhaps even likely - that you’re still a bit fuzzy on effects. Digital processing makes it possible to have so many versatile effects in the same device that it seems almost impossible to find the time to become familiar, let alone competent, with all of them. Have no fear...we’ll guide you at least partway through the maze.

Digital reverbs

Let’s skip the basic stuff. If you need a refresher, browse the **Reverb** parameter reference, where you should find enough information for a grounding in the basics of digital reverb tweaking.

“Oooohhh...*sweet!*”

If the 01V is your first exposure to digital reverb, you’ll likely have the same rush of glee every recordist feels when they begin to play with their first ‘verb. But after a few hours of bathing your chosen axe in ambient wetness, you may want to take a harder look at the reverb parameters dialog and see what else is possible. In addition to natural-sounding reverberation, the 01V's reverb algorithms can produce a wide range of musically valid effects, many of which sound nothing like what we normally think of as reverb.

Early reflections

"Early reflections" refers to the first set of reflected sound waves in reverberation. When a sound is produced from the center of a large enclosure such as an auditorium, it takes time for that sound to travel to the walls and ceiling and reflect back to the source. Clap your hands in the center of an empty gymnasium or warehouse and you'll clearly notice the brief delay between the clap and the reverberation.

If your ears are especially sharp, you'll even notice it in an enclosure as small as a schoolroom. You'll also notice that the first reflections are crisp, almost harsh-sounding. That's entirely natural. Early digital reverb units were often criticized as unnatural-sounding, often due to the way they handled - or completely omitted - early reflections. Accurate reproduction of these reflections is considered critical to creating a realistic-sounding reverb effect.

The first set of reflections are the most harmonically rich echoes. High- and/or low-frequency material hasn't had as much of an opportunity to lose its strength through *phase cancellation* (we'll discuss this term in a moment), and the dispersement of the first echoes also plays an important role in a reverb's psychoacoustics. It's one of those things the average listener doesn't notice, and may not even *like* if they hear these reflections isolated from the rest of the reverb. But we all *expect* it as part of natural reverberation. It is very likely a part of our genetic heritage.

To a certain degree, we all still have "bat ears", remnants of our old hunter/gatherer listening skills. It wasn't that long ago in evolutionary terms that we relied on our hearing at every moment of the day to alert us to the proximity of predators and prey. We could tell in an instant the size of the animal or object, and almost its precise distance and angle to us, based solely on a brief sound. Blind people usually discover this long-lost ability out of necessity; audio engineers tend to develop it out of desire. In any case, while the warm "noise" of a reverb tail may sound pleasing, our esthetic sensibilities are attuned to hearing well-defined early reflections. Those first reflections help to define our spatial relationship to the sound source, and reverbs just don't "feel right" without them. This makes it especially important for programmers to calculate realistic tonal colorations for early-reflections.

Early reflections are the "rock" on which the rest of the reverb is built. That's because it's these early reflection signals which are fed back into the reverb algorithm to produce subsequent sets of reflections. If the early reflections aren't natural-sounding, the whole reverb will suffer from a lack of realism.

This critical component of all quality reverb effects is also a valid effect in its own right. See the Early Reflections backgrounder for some additional information.

The role of phase cancellation in reverb decay

As you probably know, when two sound waves meet each other at angles greater than 90 degrees, any overlap in the two sound waves will result in the two waves cancelling each other out wherever they overlap.

A sound wave is a rhythmic pushing and pulling of air in a specific direction. Imagine the above-baseline part of the oscillator curve as the "push" phase and the below-baseline part as the "pull" phase of the wave. When two waves arrive at the same point in space from different directions, they can't *both* push or pull at the same time in exactly the same place, so here's what happens. If they come at each other from angles greater than 90 degrees, they cancel each other out if they're both pushing or pulling that same point in space. If they come at each other from angles of less than 90 degrees, they can still cancel each other out, but only if one wave is pushing and the other is pulling.

Here's how this works in practice. Aim a pair of PA columns, both playing the same audio material, at someone standing in between, and six feet in front of, both columns. Now reverse the output phase on one of the columns. You'll notice that when you reverse the phase, the apparent signal level actually *drops* below the level of the sound from just one column. This is because as the waves meet in the center of the audio field, both sets of waves push against each other. Flip the phase switch again, and the combined output of both columns will seem *louder* than a single column, just as you'd expect. When both columns use the same phase, the waves swing back and forth, up and down in sync with each other, and when they meet in the center, they'll push and pull air at an *in*creased rate rather than a *de*creased rate.

This leaves one loose end to be tied up. If energy can neither be created nor destroyed, what happens to the energy when waves cancel each other? Nothing. Air molecules at rest always have a certain amount of *potential* energy. When a speaker driver *pushes* the air, it adds energy momentarily to the air it is pushing. On the inward thrust of the driver, when it is *pulling* air, it temporarily *subtracts* a bit of energy from that volume of air. Eventually the air comes to rest with the same energy it had when it started, so nothing is ever truly lost or gained through phase cancellation.

Trust us...this was a *lot* harder to write than it was to read.

Phase cancellation in practice

Keep in mind that phase cancellation is far more important to reverbs than its opposite, phase complementation (or, more commonly, phase *summing),* although both play a role in the overall sound of a reverb. This is because reflected sound waves usually suffer much more cancellation than summing before they reach the listener.

Also keep in mind that *complete* phase cancellation is rare in nature. It’s easy to produce in audio circuits, but you can’t just aim a pair of speakers at one another and expect them to cancel each other out in the center. Waves almost never meet precisely head-on and at precisely opposing phases. They divide whenever they hit angled surfaces and reflect at those angles, and meet other waves at a variety of angles. The angle determines the degree of cancellation or summing.

Creation of a rich, sweet acoustic environment requires that the sound reflects at many varying angles, producing partial cancellation and complementation effects across the audible spectrum (and even outside of it). This, perhaps more than any other fact, is what makes quality reverbs so processor-intensive and so difficult to program. The number of angle and signal loss calculations involved in accurately reproducing the sound of a cathedral, or even a sports arena, is enormous. And those reflection calculations have to be repeated over and over again until the signal levels in the reflected waves have cancelled themselves to near-zero.

Programmers can use a number of tricks and shortcuts to emulate certain types of phase cancellation and complementation, but almost all of these shortcuts decrease the apparent realism of the reverb The only way to create a really sweet reverb is to duplicate the type - and the *number* - of equations used in natural reverb.

Spring reverb

Until the advent of affordable digital reverb, about the only affordable artificial reverbs you could find were those venerable old spring tanks. You gotta love ‘em...if you’ve been around since before the dawn of affordable digital, you’ve probably owned at least one spring tank. And even if you own a brand new guitar amp, it likely contains a spring device rather than a digital effector.

Nothing else sounds like a real spring reverb, and in all fairness to the vintage gear buffs, if spring ‘verbs weren’t so ingrained in our cultural memories as the de facto ambience for guitar tracks, no one would probably care about them except a few techno-heads looking for unique sounds. No other common reverb device, with the possible exception of cheap BBD analog echo devices, sound as limp or artificial as a spring tank.

Spring reverbs consist of a metal enclosure (the “tank”) housing two or more lightly-tensioned steel springs connected to transducers (pickups) to capture the reverberation. The spring is vibrated at one end by the input signal. A transducer at the other end of the spring picks up the sound and feeds it to the output. The spring acts somewhat like a very inaccurate speaker, and produces reverberation due to the fact that unlike a “real” speaker driver, the spring coil isn’t forced by surrounding magnets to come to rest on demand. Instead, once set in motion, it continues to vibrate until its energy is dissipated, thus producing a “tailing” effect. (Springs and transducers are suspended within the tank by additional springs to help increase the decay time of the spring’s vibration.)

All in all it’s a very artificial sound, but it’s not easy to recreate digitally. Spring materials vary. Spring lengths, widths, and wire gauges vary. The number of springs in the tank varies widely. The suspension springs play a role, and of course there’s also the type and quality of the transducer to consider. Even the amplitude of the input signal affects the overall reverb tone.

Most digital reverb units need to emulate a variety of enclosures and reverb types. Spring tanks are not popular choices when better-sounding ambiences are available, so most digital effectors are limited to a single, fixed algorithm for the spring reverb emulator. The tone of this emulator can vary widely from unit to unit depending on which model of reverb tank the programmer chose to emulate. Chances are good that if you have five separate spring reverb emulators from five different publishers or manufacturers, each one has its own unique tone.

Plate reverb

Even the richest spring reverbs - and some of the better ones are surprisingly sweet - sound pale in comparison to a real live plate reverb. Plate ‘verbs are constructed by suspending two long, thin sheets or “plates” of metal side by side so that the distance between the sheets can be varied at one end, and optionally, the length of the sheets can also be varied. The sheets, which are connected to transducers, act as speaker substitutes, but their construction and method of suspension allows them to vibrate freely. If you ever get a chance hold a large piece of thin galvanized sheeting in one hand and shout at it at a shallow angle and you’ll instantly grasp how plate ‘verbs do their job.

A variety of enclosure types can be emulated by varying the length and spacing of the two sheets, but as with spring reverbs, the result always has a signature plate-type sound that isn’t entirely natural. In order to accommodate reasonable frequency ranges and decay times, the sheets must also be quite large. Most plate reverbs are fixed installations, and some take up an entire room.

The appeal of plate systems persists to this day because of a unique characteristic of high-quality plate systems. These devices tend to have a relatively low degree of rolloff in upper midrange bands (they’re metal surfaces, after all), giving them a unique place even today as vocal reverbs. Nothing sweetens sibilance like a plate ‘verb, which is why some studio owners will still pay literally thousands of dollars an EMT plate system that will require a lot of valuable floor space to house.

Emulating plate ambience in a digital device is mainly an exercise in accounting for plate sizes, spacings and angles, materials and thicknesses. As you can imagine, the subtleties of plate systems aren’t likely to be well emulated by anything less than a dedicated reverb effect devoted to plate emulation, but the smoothness and resonant peak signatures we associate with plate reverbs have been reasonably well emulated by digital devices for over a decade.

Digital advantages and drawbacks

The biggest problem with digital devices has been that you are limited to the algorithms built into the device. Unless the device’s chips could be “flashed” with new algorithms or replaced with new chips, you were stuck with a single set of algorithms that had to suffice for all situations. Considering the time and effort that engineers and programmers put into developing reverb algorithms, it seemed like a fair tradeoff.

Assemble a group of ten engineers, have them perform blind auditions of ten mid-priced and high-end digital mastering reverbs, and it’s likely that you won’t get them to agree on the quality of any of them. Part of the reason is because psychoacoustics plays an enormous role in preference for particular reverb algorithms. For example, if you’ve had a practicing Catholic upbringing, it’s quite likely that you have a decided preference for hall and cathedral reverbs, because that tonality will resonate childhood memories. Given a choice of hall-type algorithms, you’ll probably prefer a medium-quality reverb that closely matches the acoustics of the church of your childhood over a high-quality reverb that doesn’t.

As with most multieffects units, the 01V contains a set of fairly generic algorithms. You might be able to *approximate* your high school gymnasium, but it’s unlikely you’ll be overjoyed with the way it *feels.* Only those folk who went to schools with the same gym design will really notice, so it shouldn’t matter to most listeners if you can’t get a *precise* match. If the lack of a precise match is irritating to the engineer, it’s mainly a psychological handicap.

Then there’s the “tale of the tail”. Digital reverbs have long been known for having “grainy” tails as the output level of the wet signal decays to low levels. There’s a good reason for that. Each set of echoes gets recycled, in whole or in part, back through the effector. Each reprocessing introduces new digital artifacts into the sound. The trick in creating realistic reverb tails is to use the available processor to its utmost to insure the lowest amount of digital distortion in the tail without using so few reflections that it kills the diffuse character of a natural reverb tail. The 01V, as with most better multieffectors sold today, uses 32 bit internal processing to insure the least possible degradation in the output signal. Compared to “vintage” 12-bit digital units such as the SPX-90 and REV-7, its wet signal is positively pristine.

But the bit giveth and the bit taketh away. It’s one thing to try to mimick the ambience of a given physical space. It’s quite another to develop your *own* ambient effects. While the 01V may not have the tweakability of a Lexicon or TC Electronics stand-alone unit, it does provide fourteen separate adjustable parameters, many of which offer a range of adjustment well beyond what’s required for “natural” ambience. We’ve hinted at a few of the more unusual applications for the reverbs in the parameter reference, but only a few. Creative use of digital ambience effects is still largely underexplored by engineers.

Digital reverbs versus physical modelling

One of the newest weapons in the engineer’s arsenal is *physical modelling*. Through the use of sophisticated algorithms, signal processing chips and CPUs can be used to emulate virtually any sound imaginable, from the nuance of a live drummer (e.g. Roland’s new V-Drum units) to orchestral instruments (Yamaha’s VL series synthesizers) to ambient spaces. Several firms now produce plugin effects designed to reproduce the response characteristics of specific enclosures, microphones, ambient spaces and speakers. If you truly need cream-of-the-crop reverb, physical modelling is *the* wave of the future. You just can’t equate the accuracy of physical modelling with a reverb device that uses best-guess algorithms...physical modelling comes out on top every time.

But ambience modelling at this time has one very serious drawback. It requires *enormous* amounts of signal processing horsepower. While Yamaha *could* have included an ambience modelling preset in the 01V (they had the software for it), it’s no surprise that they didn’t. A physical modelling preset would quite likely require the DSP horsepower of both effectors and still not compare to the quality of software plugin effects.

Early reflections

See the early reflections subsection of the digital reverbs backgrounder for basic information on the role of early reflections in reverb effects.

Sophisticated multitapping

This effect is actually part of reverberation, but the growth in popularity of digital reverbs has helped it to evolve into an in-demand effect in its own right.

“Early reflections” is actually little more than a sophisticated multitap delay. The effect consists of just the first set of reflections, or a set of delayed copies of the original signal. Early reflections differ from multitapped echoes in two ways. First, delay taps are tonally altered in accordance with the reverb algorithm used to generate the delays. While a normal multitap delay might offer high and low frequency rolloff as the only tonal colorations, early reflections might use varying degrees of both in addition to phase notching and other types of modulation. Second, the taps are clustered tightly in time, just as they would be in a natural ambient space, and each tap has its own unique coloration. Standard multitap delays usually allow you to vary the delay time on every tap, but they don’t normally allow you to modify the tone of each tap.

The resulting E/R effect is completely unnatural but highly usable, especially in modern dance and rock. When it’s done well, it sounds something like a “fat” stereo delay effect, or hearing your original track echoed by a chorus of identical instruments picked up from a distance. When it’s done poorly, it sounds like a muddy, out-of-phase echo.

Practical applications

Early reflections with a high initial delay or predelay setting (e.g. 300ms) can make for especially rich echo effects. Shorter initial delay times (<50ms) are often used as “thickening” effects. Some especially fat drum and rhythm sounds can be created by applying early reflections as 100 percent wet signal. Varying the algorithm usually affects both the tonal quality of the reflections and the timespan between the first reflection in the set and the last.

Additionally, the degree of "fatness" in the effect can usually be dialled in by increasing the number of reflections used by the effect.

Delay effects (echo-type)

Drawing the line between echo and modulation

Delay effects come in a distant but solid second place in rankings of most-used effects in modern recording. But the versatility of delay-based effects is so great that a clear distinction needs to be drawn between two very different types of delay effects. The two types are *echo* delay and *modulated* delay. The same device can produce both types of delay effects, and each type has very different applications.

Echo versus modulated delay

"Echo" shouldn't need explanation. We'll define modulated delay effects as the shorter effects which usually depend upon a low-frequency oscillator to sweep the pitch of the delayed signal. Flanging, phase shifting, chorus and "symphonic" are all modulated delay effects, and all four of them normally depend upon delay times of less than 30 milliseconds.

For the sake of this backgrounder, and in context of the cross-references in the parameter guides for each of the effects, we've drawn a line in the sand at 50 milliseconds, which is about the shortest delay normally used for doubling or slapback. Anything higher is considered to be an echo-type delay effect, and to keep things relatively well-organized, background information on modulated delays has been moved to a separate topic.

Delay and decay, and a little background on tape echoes

Let's start by exploring those long "virtual Grand Canyon" echoes we've all grown to love. These were originally achieved by using two or more tape decks and varying the distance between them. By widening or narrowing the distance between decks, you vary the time between the output of the first deck (the original signal) and the output of the second deck (the echo). If this sounds like an expensive and unwieldy way to generate echoes, you're right...it is. Engineers often built special tape guides for use between the two machines to insure reasonably accurate distances between the tape heads, and thus reasonably accurate delay times.

Eventually someone had the notion of using a tape loop to create the echoes. The input signal could be recorded onto the tape and played back from a playback head in the same device. Depending on the device, the delay time might be varied either by varying the speed of the capstan motor (the most common method) or by varying the distance between the record and playback heads. Whatever was on the tape would be erased prior to the tape loop reaching the record head again, thus allowing for realtime echo effects without the need for multiple tape machines.

Once that methodology was figured out, it didn't take much imagination to take it one step further and create multiple decaying echoes using a tape loop system. In fact, there are three ways to do this.

The least common method, used on a few rare machines that depend on capstan motor speed, involves using a lower degaussing strength on the erase head to insure that some of the original signal remained on the tape when it passed over the playback head a second time. As you might guess, this is messy, complicated, and impractical. But it does work.

The most accurate and expensive method is to string a series of playback heads together along the tape's path, evenly spaced so that each head in the chain plays back at a different time interval. Trailing echo effects are achieved by reducing the output level from each head according to where that head lies in the chain. While this produces the highest output quality of any type of tape echo effect, you can imagine the cost and inconvenience involved in using and maintaining ten quality playback heads just to produce a reasonably clean trailing echo effect.

The third method uses a single record head and a single playback head. Delay times are varied either by altering the distance the two heads or by changing the tape speed to increase travel time from record head to playback head.

Trailing echoes are achieved by mixing the playback head's output at a lower level with the input signal on the record head. When the newly-recorded tape reaches the playback head, it contains both the new audio material and an echoed copy of the old material. A tape loop is still used, but a long loop is used so that the same short length of tape isn't travelling over the record and playback heads several times per minute. This design is quick, slick, convenient, and cheap enough to produce that reasonable-quality tape echoes of this type became affordable for club bands and home studios by the mid-1970s.

While tape echo effects only require one playback head for basic effects, most better units have three to five playback heads. These are spaced in such a way that very long delay effects can be achieved without loss of fidelity. The dead center of the first playback head might be mounted as close as 20mm from the center of the record head. It might be linked to a "slapback" or "doubling" switch on the front panel and reserved for very short echo effects. The second head will normally be mounted two to four times the distance from the first playback head, with the third head mounted two or three times the distance from the *second* head. Delays on the first head might range from 50ms to 150ms, from 100ms to 300ms on the second head, and from 300ms to 900ms on the third. The actual delay time for any head would depend on the speed of the device's capstan motor. Achieving this kind of range with a single head requires a 9:1 ratio of tape speeds; this only requires a 3:1 ratio, allowing for much faster average tape speed, much higher fidelity over the full range of delay times, and much less stress on those expensive capstan motors. (Note that actual head spacings, tape speeds and delay ranges vary from model to model...the values used here are only examples.)

"Tape warmth"

I suppose we all saw it coming...with the trend to vintage gear, even old Maestro Echoplexes and Roland Space Echoes are fetching premium prices on the second-hand market from engineers who want that "tape warmth" that digital echo units can't provide.

Or *can* they? Let's explore where this "warmth" comes from.

We can start by ruling out tubes. Virtually none of the best-selling tape echoes used tube circuits. Most of these devices predate inexpensive Dolby B circuits and certainly didn't run anywhere near the 15ips "pro" tape speed standard so they were often *very* noisy. Sorry, Arboretum...sorry, Steinberg... tape hiss is an interesting effect, but it isn't "warm". The "warmth" comes primarily from the effects of tape compression or saturation. Recording a signal to tape produces an acoustically pleasing compression effect on the peaks, similar to the tonal coloration of natural echo. An additional characteristic of tape echoes is their frequency response. Tape echoes are rarely flat above 12kHz, and many begin to roll off high end at 8kHz or less depending on tape speed and actual tape quality. HF rolloff increases as succeeding echoes are fed back through the record head. This has the effect of enhancing the perceived presence of the source signal, hence an illusion of warmth. Virtually all newer delay lines, including the 01V **Echo** effect, try to emulate this effect to some degree.

Analog delay: echoes of a past best forgotten

Okay, maybe you *could* foresee a day when a used Roland Space Echo might fetch US$500. (You could buy them by the truckload for under $100 each at the dawn of the 1990s.) But who could have foreseen *analog* delay units fetching premium prices?

Vintage gear? Maybe. A vintage effect? Let us pray not. Solid state analog delay lines are little more than specially-designed filter devices engineered to generate exceptionally long delay times. A special type of filter, called a bucket brigade device (BBD), is used to more or less keep the electric signal "busy" in the circuitry until the desired delay time has elapsed. Then the sound is mixed back with the original signal. BBDs don't require a temporary storage medium such as RAM or magnetic tape to store the signal, which made them very inexpensive to produce in comparison to early digital devices.

The problem with BBDs is that they are very low fidelity devices. Even the best BBD-based analog delay lines can't manage clean frequency response past 5kHz at any delay time longer than a slapback. And the longer the delay, the lower the fidelity. BBDs designed for much lower delay times can produce pretty good frequency response, making them quite acceptable for use in devices such as choruses and flangers which only need very short delay times. But they sound so poor in echo effects that it's difficult to find more than a handful of valid applications for their use...so difficult, in fact, that we couldn't think of a single popular song that uses BBD analog delay as a "signature" effect.

That hasn't stopped some from *trying* to find valid applications for it, though. Remember the infamous Radio Shack solid-state reverb? (If you heard it, it wasn't something you'd soon forget. It was horrible.) This effect was included on a couple of their PA amplifiers and line mixers. It was nothing more than a cheap (*very* cheap), relatively low-fidelity analog echo that lacked even the basic warmth of a $10 two-spring guitar amp reverb.

Bottom line: if you really want analog delay tone, buy an old Ibanez/Boss/Korg/Yamaha delay pedal or someone's old disused rack-mount analog delay. Don't expect it to be emulated by a DSP card or plug-in. Programmers have better things to do with their time.

Multitapping the lines: only an offence in conservative states

It doesn't take a genius to figure out how multitapping came about. As an afterthought of tape echo design, an innovative manufacturer decides to allow two or more heads' outputs to be mixed together or fed to separate outputs. This allows the first head in the chain to "tap into" the signal before it reached the most distant active playback head. The result is two echoes occurring at different intervals, each of which can be recorded or mixed separately. Better tape delays even allowed head spacings to be varied to take full advantage of double, triple or quad tapping from the different heads.

Multitapped delays with feedback (repeating echoes) can be applied to percussion effects to give them the sound of objects dropped on a hard surface, bouncing ever faster and more quietly until it comes to rest. A well-known example would bethe sound of a drumstick tip dropped on a snare head and allowed to bounce down to silence. Tapping vocal lines with beat-matched delay intervals is used to produce unusual variations on chants and rounds. One famous, and very dramatic, Lexicon multitap effect "sprays" a series of tapped delays across the stereo field rapid-fire. These are only three of the better-known effects possible using multitapped delays.

Multitapping is often used to enhance the stereo field of a delay. Varying the delay times between left- and right-channel echoes by a few milliseconds creates a noticeable ambience in the delay (and noticeable phase summing/cancellation problems when the signal is mixed to mono). It is occasionally applied in mixing to time-align tracks which may have slight push. But its most dramatic application is for generating series of delays which add interesting dimensions to the rhythm of the track. The "standard" multitap preset is 250ms for the first tap, 500ms for the second tap and 750ms for the third. Try jamming any lead instrument through a delay with this configuration and you'll learn fairly quickly how multitapping can add complex rhythmic elements to a track.

Multitapping has come to be expected from any good digital delay or multieffector. But multitapped delays are not as simple to create as you might think. You need *at least* two and preferably between five and thirteen completely independent delay circuits. Each circuit has to have the same capabilities and memory requirements. Early multitapped delays were far costlier than standard digital delay lines precisely because of the multiplied requirements of the digital circuitry. Fortunately, processor power has increased enough and memory costs have dropped enough that virtually any device capable of reasonable reverb will have enough horsepower for tapped delay.

The multitapping capabilities of Yamaha's effectors in the 01V aren't really comparable to those of most newer delay units and plugins. Only the **Echo** and **Stereo Delay** are suitable for use as multitap delays. Even as two-tap delays they both have their limitations. Considering that software multitap delays are relatively inexpensive and are among the least destructive and least CPU-intensive effects on delicate digital delay signals, you shouldn't miss this capability much.

By the way, tapping is something every delay device does by "tapping into" the signal at a certain point in time to create the echoed signal. Multitapping only applies to tapping at more than one point in time.

A short, sharp slap(back)

Slapback is a short delay effect, usually consisting of just one or two mono or stereo echoes of 50ms to 125ms in length. It is often used to "thicken" or add presence to a track. At the lower end of the scale (50ms), it acts as a doubling effect. One common application, heavily used in hobby studios but seldom at the pro level, is to make a standard 6-string guitar sound like a 12-string. At the upper end, it is commonly used as a short echo effect and is sometimes referred to as “the Elvis echo”. In fact, Sam Phillips’ studio, where Elvis recorded Heartbreak Hotel and immortalized slapback forever - used a stairwell as an echo chamber to produce this effect. Slapback was once so popular that some metropolitan rock and roll stations in the US gutted whole rooms for use as echo chambers or set up microphones in locked-off stairwells at night just to add slapback echo to their broadcast signal. These physical echo chambers also added their own unique ambience and reverberation effects, which is why applying a simple single-repeat digital delay to a track won’t actually give you that “fifties feel”. Slapback also becomes highly annoying to the listener after a while, which is why many radio stations invested in more acoustically pleasing plate reverb units in the 1960s for use as presence enhancers for their broadcast signals.

Use and overuse

As with reverbs, echo effects produce a remarkable amount of acoustic satisfaction. Be aware that the same addictive use of reverb that plagues so many amateur mixes can also be a problem with echo effects. Very faint echoes can be remarkably effective at adding a perception of space, preserving a track’s musicality without drowning it in gimmickry.

Varying the type and equalization of an echo effect can also help prevent that “dissociated feeling” one gets when listening to track after track of heavily-echoed guitar or keyboard. For variety, try using the **Early Reflections** effect with long initial delay times and you might find that you can get by with much lower output levels on delay effect channels without sacrificing the impact of the effect.

Digital delay and MIDI

When working with digital multitrack and MIDI, it’s a good idea to know when digital delay effects are inappropriate or unnecessary. That knowledge could free up an 01V effector or give you a few extra percent of CPU usage for a plugin, and result in a higher fidelity recording in the bargain.

MIDI-controlled signals such as synthesizer or sampler output can often be delayed via MIDI itself rather than by a digital delay device. The application of digital delay to a digital audio signal introduces additional processing to the delayed sound. At the very least, this reduces the number of additional processes you can perform on that signal before deterioration begins to occur. MIDI delay has no such drawback, because the only things that are delayed are MIDI volume, noteon/noteoff messages and possibly filter controller messages. If you’re new to mixing pop or rock music, you may not have faced the problem virtually every pop engineer on a budget encounters on a regular basis: too many ideas and too few effectors to realize them. Any time you can use one less effector by working around the need for it, it’s worth the effort.

So that begs the critical question...when should you *not* use MIDI delay on a track instead of an external effector? Generally speaking, you won’t get adequate results if the patch is pressure or velocity sensitive. If the delay is accomplished through a less sophisticated MIDI processor, the patch’s timbre will not respond appropriately when echo is applied. Pressure, velocity and real-time modulators won’t always be taken into account by the delay. You’ll also find MIDI delay unacceptable if the track is already short of polyphony, since each echoed voice is another voice of polyphony taken from the available pool.

Delay with modulation

See **Delay effects (echo type)** for an explanation of how we distinguish modulated delay from echo-type delay.

A little history

Until digital devices began to proliferate, most modulated delay effects were rarely (if ever) referred to as such. Check out ads and spec sheets for chorus, flange and phase shift effects from the early 1970s and you’ll be hard-pressed to find “delay” anywhere in the copy. Yet they’re all effects that depend on a delayed signal fed through a modulator.

Most early studio and stage delays were tape-based units which used a tape loop, a single record head and one or several playback heads to achieve their effects. Modulation was a tricky effect to achieve on these units since it required accelerating and decelerating the capstan motor at a regular interval, or performing some other trick such as moving a pulley to rhythmically vary the distance between tape heads. That effect was usually achievable by using cams on the motor to rhythmically accelerate and decelerate the capstan (pulleys weren't as commonly used), but these weren't part of early tape delay designs. It wasn't until about 1972/73 that tape echoes offering modulation effects first began to appear. Even then they were most commonly used for special effects such as sweeping a vocal echo of several hundred milliseconds up or down into infinity as a "gimmick" effect. It wasn't until better analog delay circuits began to appear in rack and pedal units that "modulated delay" became associated with the three most common modulated delay effects: chorus, flanging and phase shifting.

Four effects, one common source

With the advent of digital effectors in the late 1970s, fine control over delay times and modulation oscillators made it possible for manufacturers to combine echo *and* modulation delay effects in the same device. By that time "chorus" and "flanging" had become part of the audio engineer's lexicon, so marketers began to advertise the full range of the device's capabilities. Sophisticated buyers have expected since then that *any* modulated delay device should be able to produce echo, slapback, doubling, chorus, flanging and simple phasing, and perhaps multitap delay as well.

While phase shifting, flanging and chorusing can all be achieved using a modulated delay device (provided that it can handle the delay range...many older digital delays don't generate delays less than 10 or 20 milliseconds), technicians and musicians have since found ways to enhance each of these effects using parameters and physical modifications that aren't usually found in your garden-variety modulated delay. Most standard modulated delays use a triangle wave as the oscillator for pitch modulation. But both chorus and flange effects can sound less harsh and artificial when a *sine* wave is used in its place. (The 01V provides this option for these effects.)

Play with with the **Modulated Delay** effect in one of the 01V's effects units and you'll hear fairly quickly how delay time relates to effect type. With delay times of less than two milliseconds and a fair amount of feedback gain, you can approximate phase shifting effects. From two to 14 milliseconds with feedback gain, you'll achieve virtually any flanging effect you might need. From 15 to 50 milliseconds with little or no feedback gain, you can duplicate most chorus effects.

Chorus

The classic chorus effect is little more than modulated delay using a delay time ranging from 4ms to 35ms, a modulator set to relatively low depth (less than a semitone of total sweep) and medium speed (0.2-0.75Hz), and either zero feedback (a single delay) or a specified number of repeats at or near the same volume ("thick" chorusing). The delayed signal is usually mixed with the dry input signal at unity gain or slightly below the level of the input signal, and some EQ may be used to cut the low or midrange band to prevent the fundamental of the instrument from sounding too wobbly. Chorus effects for bass guitar tend to roll off the low end on the wet signal at or below 200Hz, and many engineers roll off the wet signal in the 200-800Hz range when adding chorus to guitars to create a "shimmering" effect.

The result, depending on the settings selected, ranges from a "watery" effect something like a 12-string guitar or an out-of-tune piano, to a clean thickening of the sound similar to two instruments playing in unison. Add multiple repeats and the sound thickens further, giving the impression of a "chorus" of identical instruments, hence the effect's name.

Perhaps no single artist is more responsible for the popularity of chorus effects in popular music than Andy Summers, former guitarist for the Police. In the early 1980s, the "watery" tone he showcased on virtually all of the Police's hits from Regatta De Blanc onward became a trademark sound. Within a couple of years, chorused guitar was heard on virtually every pop track to hit the charts. By about 1984, chorus pedals had become as popular as distortion devices with guitarists, no small feat for an effect that really has only one sound. (But hey, it's a *great* sound!)

If you’ve worked with synthesizers, you probably know how to detune stacked oscillators to fatten up a synth patch. Chorusing and detuning sound similar but are actually worlds apart in the way they work. Detuning mixes a second, slightly pitch-shifted copy with the source sound to give the impression of multiple synths/samples/instruments playing in unison.

Chorusing, like detuning, applies a second copy of the source sound to the original. But instead of shifting the pitch down slightly, a chorus effect *delays* the signal by several milliseconds (usually from 15ms to 40ms) and adds pitch modulation to the delayed signal. If you configure a chorus effect so that you hear only the effect output with no dry signal mixed in, all you’ll hear is a slow vibrato. Dial in the dry signal and you’ll start to hear the vibrato’d signal sweep over and under the original signal. Typically the effected signal is detuned from the dry signal during half of the modulator’s cycle, and the dry signal is detuned from the effected signal during the other half. The range and length of the detune/overtune cycle is variable in better chorus effects, and a normal value for the maximum pitch variation of the vibrato is about +/-15 to 25 cents, or one-third to one-half of a semitone. You need to gauge the pitch variation by ear when using non-dedicated chorus units since they don’t normally display the modulator depth as a measure of cents.

Where’s that $&%@! fundamental?

Chorus is one of the most emotionally pleasing effects you can apply in pop music, but it has one serious drawback. Modulating the pitch creates something of a hypnotic effect due in large part to the slow vibrato applied to the delayed portion of the signal. This hypnotic effect tends to distract the listener from the fundamental pitch of the instrument. Too many amateur recordists have tracked guitars, synths and basses “wet” with chorus only to discover that they couldn’t produce a solid vocal performance over the effected bed tracks. Most singers don’t have perfect pitch; they need to hear at least one instrument in precise tune as a reference pitch to sing to. Always try to preserve at least one dry, unchorused bed track in any project for use as a pitch reference for any singers who might need it.

Flanging

Myth and magic

More mythology has built up around the origins of flanging than around virtually any other popular effect. Here’s the straight goods on how this effect got its name.

Back when the Beatles were recording one of their “experimental” tracks for either Revolver or Rubber Soul, they were experimenting with vocal doubling by running the same tape over two sets of playback heads set at a fixed distance from one another. The effect was actually discovered by accident when someone in the studio accidentally leaned against the reel, temporarily slowing down the tape. (Legend has it that the culprit was Ringo; we can’t confirm or deny this but it *does* sound a bit too much like the setup for another bad drummer joke.)

As the playback slowed slightly and then sped up, modulation was heard between the signals playing back on the two sets of tape heads. The effect was more like what we know today as chorusing, since it didn’t have the characteristic feedback we associate with flanging. (Feedback was introduced later by feeding the playback head’s signal back into the record head mixed with dry signal.)

In any case, producer George Martin was later asked about the effect by a reporter. Instead of revealing the true story, he made up a name for the effect on the spot: “flanging”. Apparently it was dumb luck that he chose a name which could be justified with a little rewriting of history. So as the *legend* goes, the effect was achieved by “riding”, or manually braking, the spooling reel’s flange to varying degrees with the bare hand. (The “flange” is the fastener that holds a tape reel in place on the recorder.) This slows down and speeds up playback as pressure is applied and released. But when this was repeated at Martin’s Abbey Road studio, and later at studios around the world, manual braking of the reel was done *not* by manually braking the flange, but by braking the *outer rim* of the reel with the bare hand or a piece of cloth.

Martin’s impromptu choice of names stuck, and by 1967 engineers on both sides of the Atlantic had begun to routinely apply manual or motorized modulation to recorded signals. When transistorized effects modules became justifiable as mass-produceable products later in the 1960s, feedback was added to the effect to mimick a long row of playback heads, each playing back the signal at a different delay interval.

The sound and the metal fury

Flanging is achieved by combining a dry signal with a signal which is delayed by a few milliseconds, frequency-modulated, and fed back through the delay line. The resulting tone is metallic, and its frequency response curve appears to have a large number of notches in it at various frequencies, giving it a comb-like appearance on an oscilloscope. (Perhaps you’ve heard of “comb filters”.) An oscilloscope will also show the comb “dancing” from side to side on the screen as the delayed signal’s pitch is modulated, mirroring the way the filter notches sweep up and down the frequency range specified by the modulator’s depth control. The feedback control varies the audible depth of the notches; the more feedback, the deeper and wider the notches.

At lower delay time settings (around 2ms) the effect imparts a breathy tone of a type usually associated with phase shifting. At the upper end of the normal range (around 10ms and higher), the tone becomes harshly metallic. In the middle of that range (2ms - 10ms), the tone is a mix of breathy and metallic. “Signature” flange effects can usually be attributed to the right mix of feedback and correct delay time setting.

Flanging is a favorite effect of rock guitarists due to its metallic character. It has never been widely used in anything *but* rock recordings precisely because the sound is considered to be so grating and unpleasant in anything but high-energy music. Used judiciously, flanging adds enormous bite to an already gritty sound (e.g. the lead guitar on Judas Priest’s “You’ve Got Another Thing Coming”) and can impart a novel sound to dance tracks (e.g. the rhythm track effects in Tom Tom Club’s “Genius Of Love”). Used carelessly, flanging is as annoying as an extended fart.

Phase shifting

Demystifying the wind

Phase shifting is one of the most misunderstood effects in popular music. It just plain scares a lot of untrained engineers who don’t understand how it works, and it even baffles relatively well-schooled amateurs who can’t figure out why their delay units and software plugins can’t emulate their treasured MuTron Bi-Phase and MXR Phase 100 pedals. There’s no mystery to it once you understand the mechanics, but as we discovered in our own research, it’s not exactly easy to find a clear, concise explanation of the effect.

Phase shifting is typically used as a highly dramatic musical effect, and it endured a period of agonizing overuse in “psychedelic” recordings circa. 1968/69. When it’s applied as the last effect in a chain, it tends to overtake the track on which it’s used to the point where the phaser sweep grabs more attention than the instrument’s fundamental.

But it doesn’t have to be dramatic to be highly effective. One of the most interesting (and most sought-after) phase shifting effects is the Eddie Van Halen “swoosh” or “wind effect” heard throughout the first Van Halen album. (Yes, you heard right...it’s a *phase shifter,* not a flanger. And yes, we felt foolish too when we heard this.) This effect was achieved using an MXR Phase 90 stomp box - a very simple 4-stage phase shifting circuit - set to its slowest possible sweep rate. What made this phasing effect so effective was that it was applied to the *dry* guitar signal *before* the amp input. The phaser had no effect on the overdrive harmonics of the guitar amplifier. The effect was not without drama but it was far more subtle than typical applications of phase shifting of that period.

How it works...

At its most basic, phase shifting is just flanging with a shorter delay time. Unlike flangers, which have no tonal range without a delay time control, most phaser pedals will not allow you to adjust the delay time. The theory is that you shouldn’t *have* to adjust it. Phase shift delay times are typically below 1ms, which is usually well below the lower limit of dedicated flange effects, but as you increase or decrease the delay time, you change the character of the sound.

Many first-time owners of modulated digital delays with delay times ranging down to (or below) 1ms have been surprised to discover that the right parameter selection allowed them to mimic, and often duplicate, the sounds of dedicated chorus, flange and phase shift devices...but in the case of phase shifting it's not a perfect emulation. While transistorized chorus and flange effects rely upon solid-state components known as BBDs, or bucket-brigade devices, to produce the required delays, emulation of phase shifting often came as a surprise because BBDs are not normally used in phasers. Instead, phasers rely upon all-pass filters (equalization filters which apply no boost or cut to any part of the signal) to produce the delay. Filters of any type produce a delay in the signal; it's only an accident of convenience that phase shifters can to use these circuits to generate the needed delay times. And it's a circumstance of *in*convenience that these fixed-state filter circuits also produce fixed delay times.

...and why you can't get an MXR Phase 90 out of your digital delay

That might explain *how* they work, but it doesn't explain why your rack-mount multieffector's modulated delay doesn't sound like your favorite phaser pedal. There's an added element to phase shifting that gives it its unique character: *stages.* Stages in a phase shifter roughly correspond to taps in a multitap delay. By adding more stages, or taps, the "windy" effect of a phase shifter becomes smoother and more pronounced, since each stage in a phase shifter adds another phase notch to the frequency range. You can apply feedback to your heart's content to a phase shifting effect created by the average modulated delay, but you can't easily duplicate the true sound of the stage multiplier control found on some of the best-loved classic phase shifters unless you dial in four, six, or eight ultra-short tapped delays.

Most chorus and flange effects only require a single delay tap to achieve their signature sound, which is why modulated delays can produce these effects so easily. Phasers require four or more taps to recreate classic effects, which is why a flanger set to an identical delay time to a phaser will never sound as smooth as a classic phase shifting circuit. But a single tap may be quite sufficient for no-frills, light-duty phasing effects.

Phasing in practice: phasing out gimmickry

Until flanger pedals hit the market with a heavy metal vengeance in the mid-1970s, phase shifters had the dubious distinction of being the most overused effects in pop and rock. On some occasions, they've generated truly breathtaking and memorable effects; on others, they literally ruined some otherwise fine pop and rock tracks.

Setting a phaser to minimum of feedback (assuming the phaser has a feedback control; many "vintage" pedals do not), imparts a tone that one of our metalhead friends has termed "sissy flanging"...a breathy resonant sweep that lacks the metallic character of a flanger. As more stages are added, the windiness becomes more pronounced. As stages are removed, the effect takes on more of the character of a chorus effect. The phase-effect-to-end-all-phase-effects, the "jet takeoff" effect, is generated by adding more stages (taps) and feedback than typical pedal units can provide. Apply enough stages and a phaser can impart that jet-takeoff sound to anything from a soft baritone vocal to a screaming overdriven lead guitar.

After spending most of the 1980s in a south Florida jail after being found guilty on several counts of poor taste, phase shifting finally acquired a new publicity agent in the mid-1990s and has found a whole new audience with no memory of its crimes of bad taste during the 1970s. The effect has been reborn as a technique for adding drama to breakbeats and a unique feel to psy, hip-hop, trance and techno. It still does the odd psychedelic-revival gig on pop and metal tracks, but fortunately its days of rampant "taste crime" seem to be over.

Digital phasing, or "spot the hidden jet"

The 01V's phaser is quite versatile and complete, and should effectively mimick virtually any "classic" phasing effect. The offset parameter, which corresponds to delay time, opens up the final frontier in phase shifting, and gives it a full range of features. If there is one potentially irritating limitation in the 01V phaser, it is the maximum of 16 stages in the effect.

Symphonic

A Yamaha exclusive

One effect you can't duplicate in most modulated delays is Yamaha's **Symphonic** effect. This is the oddball in our assortment of modulated delay effects...odd in that it is virtually a Yamaha exclusive. Symphonic first appeared as a Yamaha mainstay in the SPX-90 and REX-50 effectors released in the mid-1980s. Today it is found in virtually every multieffects device made by Yamaha.

Chorus goes downhill

It's a long-standing joke among musicians and engineers that Yamaha invented the symphonic effect because they didn't know how to program a proper chorus effect. While it *is* true that the choruses on vintage Yamaha gear have been criticized as lacking depth and smoothness, it is *not* true that Yamaha expects the symphonic effect to be used as a replacement for chorusing. Instead, symphonic is provided as an *alternative* to chorusing when a thicker sound is desired.

Symphonic takes a standard chorus effect and adds optional feedback to thicken the sound, then beefs it up a bit by pitch-shifting the input signal down a few cents prior to feeding it through the delay line (detuning). In the process they've created a rather unique effect, because unless a modulated delay or dedicated chorus device includes a harmonizer or detune parameter, it won't be able to faithfully reproduce "symphonic".

Feedback is not available in the 01V's stock **Chorus** effect, so if you like your choruses rich, thick and high in cholesterol, try symphonic with zero detuning instead to see if it doesn't meet your needs.

It should be noted that while we know of no other manufacturer that uses an effect with the name "symphonic" to produce a similar sound, there *are* multieffectors that can provide variable detuning and feedback with chorus-type effects. While the implementation of the effect and the name "symphonic" are unique to Yamaha, it's probably unfair to sell this as a *truly* unique sound.

Auto-panning

This is one effect which should be bone obvious from its name alone. Auto-panning is a way to rhythmically pan a signal across the stereo field. Basic auto-panning can be achieved by twisting a pan pot back and forth on a mixer, but most auto-pan effects, including the 01V's, go much further.

Cycles, circles and ping pong

There are three common applications of auto-panning: cyclic, circular or "3D", and ping-ponging.

Cyclic panning pans the signal from one channel to the other, then starts the pan again from the direction where it started. This gives the impression of a sound travelling in and out of the stereo field to and from the same points.

Ping-pong panning is cyclic panning in both directions. When the sound has completed its travel in one direction (e.g. right to left), it travels back in the original direction again (left to right) at the same rate, making the sound appear to bounce back and forth in the stereo field at a fixed distance from the listener. This is the type of panning you get when you twist a pan pot back and forth.

Circular, or "3D" panning, is ping-pong panning with amplitude (volume) modulation. Cyclic panning and ping-pong panning create the effect of sound travelling in a straight line, but circular panning uses volume changes to impart the illusion of sound travelling in a circle in front of the listener. For example, as it passes from left to right, it may appear to move closer to the listener as it reaches the center of the field; as it returns from right to left again, it seems to move farther from the listener toward the center of the field, thus giving the impression of circular motion. This is a dramatic spatial effect when applied in the correct context, but beware of overuse. Circular panning can become tiresome when heard for long periods, and it can even be downright startling when used in the wrong place.

3D or not 3D...

While circular panning is often referred to as 3D panning, there is no third dimension involved here. (In fact, there's not even a third dimension in 3D "surround sound" since all sound is generated on the same two-dimensional plane, unless you want to include time as the third dimension...but that's another story.) While ping-ponging and cyclic panning generates a one-dimensional stereo field consisting of a straight line from left to right, circular panning can only expand the perceived field to a two dimensions. In order to emulate actual 3D sound, there would also have to be an additional illusion of up-and-down motion.

Warning: this product may be harmful if used excessively

One way to reduce the potential for overapplication of auto-pan effects is to follow the "back-off rule"[*] . Take the effect to where you like it, then "back off" (narrow) the stereo field using the panpots on the effect outputs. Auto-panning from 10:00 to 2:00 on a track that weaves through an entire four-minute composition is considerably less tiring to the listener than the same track panned 9:00 to 3:00.

Tremolo and vibrato

Knowing the difference could save your life

Speak the word "tremolo" in front of a singer with Bel Canto training and you'll likely see them cringe. Ask a particularly temperamental vocalist to *sing* in tremolo and you might be in mortal danger. Tremolo, or "trembling", is the curse of the classically-trained singer and a kiss of professional death should it ever manifest itself in the voice. Vibrato, on the other hand, is the singer's friend. A clear 6 to 7Hz vibrato is considered the hallmark of a well-trained classical voice.

So get this straight once and for all time. Tremolo is the cyclic modulation of *volume* in an audio signal. Vibrato is the cyclic modulation of *pitch*. Knowing the difference could save your life.

It doesn't help at all that *volume* and *vibrato* both begin with "v". And the industry doesn't make it any easier to remember the difference. Take for instance the consistent misuse of "tremolo arm" as the common term for vibrato tailpieces and bridges used on guitars. These devices are used to modulate the *pitch* of notes and chords, not the volume, yet they're so ingrained in our consciousness as tremolo units that most players simply call them "trems". Bigsby had it right with their "vibrato tailpiece", but regrettably they've fallen to the bottom of the whammy bar heap. Kahler, Floyd Rose and Steinberger are just confusing the issue by calling their vibrato systems "trem" systems, because they are *not* true tremolo effects. They are *vibrato* effects.

Take this as an object lesson on the value of being right in the music industry.

A mysterious secret

It was the Bel Canto masters of the Renaissance who first decreed that a well-defined vibrato in the 6Hz to 7Hz range was the mark of vocal mastery. Those old Bel Canto masters must have known something. 6.14Hz just happens to be the vibrational frequency of the Earth's own standing electromagnetic wave, a wave that has been found repeatedly

[*] The "back off" rule: set the sound to where your ears, then *back off* one notch on the dial and you'll probably produce a better result. This is a useful rule of thumb if you tend to be overly infatuated with certain effects or EQ curves, or if you often set input levels too hot or trim levels too high. It's nothing more than a mental trick to bring yourself down out of your own private cloud and into the realm of "real world" audio.

to be essential to good health in virtually all organisms. This is not mere trivia. This principle has been applied in non-vocal material in new age and meditational music for assisting listeners in “grounding” themselves.

What it doesn’t fully explain is why vibrato is so popular while tremolo is so rare. You can doubtless remember dozens of songs that rely on vibrato effects; you may be hard-put to think of even a handful of songs that use tremolo. (Need a hint? Think of the signature riffs in “I Walk The Line” by Johnny Cash, or “How Soon Is Now” by The Smiths.) Most guitarists never get tired of their vibrato arms, but the tremolo controls on vintage tube guitar amps tend to be the least-used controls on the amp.

The reason for this disparity in preference probably has its roots in genetic memory. It’s hard to imagine anything in nature which is able to produce a strong tremolo effect which is not also a potential threat. Anything able to move toward you and away from you, getting softer and louder in an audible cycle several times a second, is likely to be a light, winged creature, i.e. an insect. If it is close enough that you truly notice it, and it produces a fundamental within the range of most musical instruments, it’s also likely to be a fairly *large* winged creature such as a wasp, bee or hornet. And if it repeatedly advances and retreats, there’s a strong likelihood that it’s also potentially aggressive. If we are in fact attuned to tremolo in this way, then it’s no wonder that trem effects are so unpopular.

Rotary speaker effect

“There’s no substitute for a Leslie.”

If you’ve never heard that line, you’ve never known a Leslie owner. Just ask any serious engineer about rotating speaker effects and one of the first five words out of their mouths is bound to be “Leslie”. The Leslie brand of rotating-speaker cabinets is still considered the world standard, so standard in fact that most musicians and engineers refer to rotating speakers as Leslie’s the way we refer to cola drinks as Cokes.[*]

Rotary speaker devices are constructed by mounting a speaker driver (in the case of better cabinets, two drivers: a woofer and tweeter) on a rotating axle in the center of a speaker cabinet. As the speaker rotates, the sound you hear from the front of the cabinet changes from the direct sound of a driver facing toward you to a muffled tone as the driver spins to a 180-degree angle from you, and the only sound you hear is the reflected sound from the baffles.

Five effects in one: what makes up the effect

The rotating speaker “effect” is actually combination of five effects. There’s a gentle phase shifting from the rhythmic variation of the sound’s location and distance. There’s also a slight vibrato created by the Doppler effect that occurs as the distance to the sound source alternates between the distance between listener and the speaker to the distance between listener and the rear wall of the cabinet. Mild tremolo is added in from the rhythmic volume change that occurs as the sound source alternates between speaker and cabinet wall. And then of course there’s equalization, since a rotating speaker does not, of course, have the flat frequency response curve of a studio monitor. Finally there’s tube tone. The amplifiers used in most rotating speaker cabinets have traditionally been, and are to this day, vacuum tube circuits.)

[*] Note if you will how we threw caution to the wind and chose to refer to Coke in the plural as “Cokes”. If this doesn’t sound risky or unwise to you, then allow us to explain.

For more than a generation, Coca-Cola have been vigorous media watchdogs, diligently scanning almost every document that flows from the public prints to insure that its name is spelled correctly. It’s the rare small-town newspaper in North America that hasn’t received a friendly notice from Coca-Cola after seeing its flagship product’s name misspelled as “coke” or “Cocacola”, or when its plural has been referred to as “Cokes”, as we referred to it here to enhance the consistency of the text. No doubt we’ll soon be receiving *our* reminder from Coca-Cola’s media relations department, and it’s anyone’s guess how long we’ll be “watched” for similar offences in the future.

It was worth the risk, though. Brazenly facing the scrutiny of this corporate giant demonstrates yet again the lengths to which we’ll go to serve our users. We hope you appreciate the sacrifice.

The origins of "classic Leslie tone"

Leslie is far from the only rotary speaker manufacturer, and not even the only one to use tube amplification. But the signature sounds of their most popular designs have become pop institutions. As with Marshall amps and Moog filters, there truly *is* no substitute for a Leslie if you must have "that sound". But while a Leslie is a Leslie sounds like a Leslie must be a Leslie, there's much more to "classic Leslie tone" than just the device's construction. The overall tone of the effect depends a great deal on how the cabinet is engineered and baffled, how microphones are selected and placed to pick up the cabinet sound, what's fed into the amp, and how the mic'd signal is mixed.

The rotating-speaker effect produced by a Leslie cabinet, the sound that you likely think of as "classic" rotary speaker tone first came to prominence in pop music when applied to the Hammond B3 organ in late-1960s hard rock. The B3-Leslie combination is as signature to late-60s/early-70s hard rock as the Stratocaster/Marshall or Les Paul/Vox AC combinations. And just as most distortion pedals are attempts to mimic the sound of an overdriven Marshall cabinet and amp, most transistorized or digital rotary speaker effects are attempts to mimic the Leslie speaker cabinet and amp.

Aural frustration in one simple step

High-quality Leslie emulation is *very* difficult to achieve in software and/or circuitry, a fact which a lot of semipro engineers and circuit designers have discovered at a high cost in equipment and frustration. Sure, any halfway respectable Leslie emulator can duplicate the sound of a clean B3/C3 series church-type organ fed through a two-speaker Leslie at medium volume. Some can even mimick the varispeed switch that alternates the rotor between slow and fast rotation. But that's not what the average rock engineer wants to hear. "*Real* Leslie", that distinctive breathy-grindy modulation made famous by the Allman Brothers, Deep Purple, ELP and so many more, is actually a combination of both the rotary speaker effects, mic placement and selection, *and* up to *three* stages of tube amplification.

Here's the problem. The signal from a vintage B3/C3 is preamplified by tube circuits housed within the organ. Hammond preamps are often modified to produce clipping or generate an especially hot signal for clipping the Leslie inputs. That signal is then fed into the Leslie, which uses more tubes for both the preamp and power amp stages. The classic heavy rock Leslie sound depends on *at least* two and very often a third tube stage, and typically at least one stage is driven to clipping. Mix this overdriven, tube-saturated Leslie output with a second direct or mic'd signal from the organ, and *finally* you approach Jon Lord's "Machine Head" tone or Gregg Allman's "Whipping Post" sound.

Accurate reproduction of "classic Leslie tone", even using a sophisticated digital multieffects unit, is a tall order. Unless you're willing to shell out the big bucks for a specialized Leslie emulator designed to duplicate this tone, you're not likely to be happy with the simulations in *any* digital effectors.

If you can be content with clean, no-frills "soap opera" or "church-organ" rotating speaker sound, most digital devices will produce a relatively faithful reproduction of the effect, and that includes the 01V. In fact, the 01V tries to go considerably farther than clean rotation, implementing a fair degree of overdrive and tone control as well.

But if you crave that real "memories of Woodstock" grinding wheeze, then nothing short of a dedicated Leslie emulator will satisfy...and even that might fall short. The only sure way we know at this time to get "classic Leslie tone" is to pay the price - in cost, maintenance and experimentation - for an actual Leslie cabinet and Hammond preamp that you can mic and mix to your personal satisfaction.

A brief note to Hendrix fans

Many guitarists use Leslie units to generate a sense of motion in their music, but Hendrix fans take note: Jimi used an emulator. As likely as not, the Leslie-like device you hear on his recordings was an Electro-Harmonix effects pedal called the Bee-Bah. Hendrix' influence is still so strong that the Bee-Bah has actually been reissued by Electro-Harmonix using the original circuitry. This pedal is a very primitive unit by today's standards, but with a little experimentation, most rotating speaker effects can produce a sort of Bee-Bah emulation. But here again, if you crave that true classic tone, you're probably going to have to invest in the real goods.

Fortunately a Bee-Bah doesn’t weigh 80 kilos or replace a 1500-watt heater.

Ring modulation

Ah, sweet dissonance: a little history

Ring modulation is one of the most bizarre effects in the engineer’s bag of tricks. Its powerful, dissonant character tends to make it pop out of almost any mix. As a result, this effect tends to be used almost faddishly. Ring modulators enjoyed an initial period of popularity in the 1970s when Moog/Arp-type synthesizers practically required by law in heavy rock and pop. You’ll hear ring modulation effects in British progressive rock from artists such as Emerson, Lake and Palmer and King Crimson, Edgar Winter’s “Frankenstein”, and in early all-synthesizer recordings from Wendy (then Walter) Carlos and Isao Tomita.

The effect fell from fashion as the 70s wore on. As polyphonic synths became commonplace, fewer and fewer keyboards and modules included ring modulators. Players and producers tended to prefer more sweeping, quasi-orchestral textures from synthesizers, and when the Yamaha DX series arrived with built-in capability for emulating almost any imaginable ring modulation effect, no one seemed to give this device a second thought. Where ring modulators *were* used, they were often applied in context of emulating other sounds such as bells and metallic effects rather than in generating unique tones. The relatively limited applicability of this effect as a general instrument processor meant that ring modulator pedals were rarely used in live performance. When they *were* used, their signature dissonance tended to sound more irritating than surprising.

Ring modulators regained popularity in the 1980s due in part to their “rediscovery” by industrial/no-wave composers as a means of generating interesting dissonant tonalities. By the mid-1990s it had found a new popularity in dance/techno tracks as a trick for generating uniquely dissonant percussion sounds and for distorting vocal tracks in unusual ways. And with the recent resurgence of interest in “vintage analog” synth tones, classic ring modulation effects of the late 60s and early 70s are appearing more and more often in pop, rock and dance music, and have once again become staple components of synthesizers.

How the effect works

Ring modulation is created by modulating an input signal with a second signal in a way that deliberately ignores rules of harmony. The input signal is combined with the modulating signal to produce an output with harmonics which are the sum and difference of the original signals. *Sum and difference*...not evenly-divided or multiplied harmonics. The actual effects can range from unusual bell-like effects to metallic tones to outlandish noise.

Ring modulators were first applied outside of acoustics labs as modules in early analog synthesizers. Arp 2600s and Minimoogs both employed ring modulators as standard built-in modules. Early ring modulators typically used either an internal oscillator as the modulating signal or, if the synth or stand-alone device allowed patching, virtually any input signal. The “signature” sound of a ring modulator is a sine wave used as a modulator for a sine or triangle wave oscillator. The dissonance this creates is clear but the harmonic structure of the sound is relatively simple. When more complex signals are used as the input or as the modulator, the complexity of the output signal’s harmonics rises accordingly and the more dissonant and noise-like the output sound becomes. Even relatively simple forms such as sawtooth and square waves become non-musical when fed through a ring modulator.

The effect’s application is limited because the architecture of the device doesn’t normally allow for mixing of input and modulator levels. Regardless of what kind of signal you feed into the effect, you’ll hear a very different sound at the output.

Modern ring modulators

Most ring modulator devices are patterned after the early Arp and Moog modules. They contain only one input for the instrument. The modulating signal is usually housed within the unit as a hardware oscillator, often nothing more than a simple sine wave. Some devices have only a single control: a frequency dial for the modulator. Some allow the frequency to be swept by an LFO to partially mimick the capabilities of analog synths. Some devices, including ring modulators used in many high-end modular synths, have external inputs allowing any signal to be used as the modulator.

Ring modulation in the 01V

The 01V's ring modulator is a relatively simple design. It doesn't allow a second signal to be used as the modulator; instead it uses a fixed internal sine wave oscillator. It does, however, provide an LFO for sweeping the harmonics of the output. It also offers the ability to modulate the input signal with itself. (Ring modulation might be a misnomer for this type of routing; *balanced* modulation might be a more accurate term.)

This is actually one of the more usable implementations of ring modulation, since modulating a signal with itself produces overtones which are half and double the overtones of the original signal. Harmonics occurring in exact multiples tend to sound a lot less dissonant and more harmonically interesting than the overtones you might get by modulating one complex signal with another equally complex signal. But it certainly doesn't make the effect universally applicable. Try running an overdriven guitar, piano or organ through the modulator effect and you'll likely find the effect grating and difficult to hear for long periods. Ring modulation is generally most pleasing when applied to very simple harmonic structures such as monophonic sine and and triangle wave oscillators.

Beyond ring modulation

When you experiment with ring modulation using relatively simple signals such as analog-style synth tones, you may discover a number of familiar tones emerging from the device. This may be because arithmetically-based modulation is the basis for at least one very popular type of synthesis: Yamaha's own DX synthesis (frequency modulation or FM synthesis). One basic two-operator FM algorithm is virtually identical in its routing to frequency modulation.

But don't expect to be able to use the 01V's ring modulator as a quasi-FM synth to any great degree, because the effect doesn't allow nearly the range of control over the modulating signal's envelope or routing as a true FM synthesizer.

Harmonizer

Cheaper than hiring a chorus

It has been a cherished dream of vocalists, guitarists and players of acoustic instruments to have a device which allows them to record a single track and create a copy of that track at a precise harmonic interval, closely matching the timbre and tempo of the first track. Until very recently that device was a pipe dream. Today it is commonplace.

The dream of affordable harmonizer technology is coming very close to reality with the latest generation of programmable harmonizers and pitch-correctors. Newer harmonizers can allow you to play a flute melody in G and have the harmonizer actually generate a pitch-shifted *harmony* of that melody...not just a copy of the original signal shifted up a fifth and time-aligned with the original signal, but a *precise harmony*. Even more amazing, physical modelling algorithms in high-end harmonizers will actually allow the pitch-shifted signal to sound almost exactly like the original instrument played in a higher register.

Pitch-shifting versus true harmonization

We've come a long way from days when the best you could expect from a $1,000 harmonizer was an exact interval of the original signal that sounded like the Chipmunks Orchestra when pitch-shifted up, or like a band of demons when pitch-shifted down. Pitch shifters, which are not true harmonizers, were one of the last "staple" effects to find their way into the average multieffector, and have traditionally been one of the costliest effects to purchase as stand-alone hardware. The reason? Horsepower. Pitch shifters require a *ton* of it, and harmonizers require even more.

Consider the logistics involved in creating an accurate digital harmonizer. First, the input signal has to be multiplied or divided by the appropriate factor to create the harmony. That multiplication needs to happen extremely rapidly so that the output doesn't seem to have an audible delay from the original signal. Then it has to be *time-aligned.* When a signal is *pitched* up by simple multiplication, it is also *sped* up. The trick is to apply an algorithm which smoothly recreates the original signal's time alignment without introducing jitter or graininess. As if that wasn't tough enough, a *true* harmonizer has yet another job to do: pitch *tracking*, or analysis of a proper harmony based on the harmonic content of the input signal. It's not enough for modern harmonizers to shift up or down by a given number of cents or semitones. The interval needs to vary as the fundamental pitch of the input signal varies, and it needs to vary *smoothly* or the result will sound like a musician struggling to find the correct note.

Add it all up and you've got the kind of engineering challenge that makes programmers and circuit designers wish they'd taken dentistry. By 1985, almost anyone could afford a reasonably good digital reverb. Quality pitch shifters, on the other hand, were still luxury items, and signal processors weren't yet fast enough to justify mass-producing true harmonizer devices.

This led to a bit of abuse of terminology by manufacturers fighting for market share. When pitch-shifters first appeared on the market, many of them were labelled as harmonizers even though they could only perform simple pitch-shifting...if they could even do *that* well. Most early digital pitch shifters, even professional 16 bit models, were extremely limited in their scope. Beyond a reasonable pitch shift range, usually seven semitones and often much less, the pitch-shifted signal suffered from digital distortion during time-alignment processing which resulted in jitter and graininess. Even untrained ears notice this distortion, and it's plainly audible in recordings of that era that use pitch shifters.

Today, even relatively inexpensive devices can outperform top-of-the-line devices of the 1980s which might have sold for as much as a luxury car. Processor speeds have allowed devices to accurately track fundamentals and correctly time-align signals with negligible delay (latency) in the output signal. And the jitteriness and grainy character of early units has largely disappeared. Harmonizing has come of age.

Harmonizing in current multieffects units

But it has only come of age thanks to the low cost of extremely powerful signal processing engines. Until multiple-DSP devices become a lot more inexpensive, you may still find yourself in need of a dedicated device or a processor-intensive plugin to perform true harmonization. Current multieffects units by and large have too much to do to support true harmonization without compromising other functions.

You need only look at the 01V's architecture to understand the hardware demands of harmonizers. Only effects unit 2 is permitted to use the **HQ Pitch Shift** effect, because even with all of the DSP horsepower on the card, all that the 01V's effectors can handle is one high-quality pitch-shifted and time-aligned output.

While a high-quality harmonizer with pitch tracking seems theoretically possible using the DSPs available on the card, one wonders how much processing time it would need to borrow from the EQ and dynamics sections in order to do its job well. We can all wish that the 01V's effectors included a true harmonizer, but considering that state-of-the-art hardware harmonizers/pitch-correctors sold for well over US$2,000 at the time the 01V was released, it is probably an unreal expectation. We'll have to settle for simple pitch-shifting.

Distortion/overdrive

Okay, who started it?

Not so very long ago, if your music didn't use guitars, you didn't care about distortion or overdrive. But no longer. The styles and sounds of modern dance and electronic composition have changed the way composers and arrangers look at distortion and overdrive. Today you're just as likely to fuss over the drive level and tone coloration of the clipping on the snare, string pad, or even the vocal line as you are over a guitar tone, even if you record in a genre that wouldn't *dream* of using overdrive ten years ago.

So who started it? The public demands that the offenders be identified.

Our forensic analysts believe that the culprits acted as a gang, even if they weren't well-organized. We'll leave the direct accusations to the Taste Crimes division...let *them* argue over whether Ray Davies of the Kinks, Jeff Beck of the Yardbirds, Pete Townsend of the Who or any of a dozen other guitarists of the period were most responsible for the first rush of popularity of overdrive/distortion. All we will say with authority is that the first "British invasion" of the early 1960s put overdrive on the pop music map as a staple effect for guitars. So if you need a scapegoat, you may as well blame the whole UK.

It's even more unclear who is responsible for the more recent epidemic of crimes against clean tone. Once again, we decline to name names. All we will say with authority is that since the early 1980s, distortion and overdrive have become increasingly acceptable on everything from drums to orchestral instruments, and it appears that yet another UK gang is largely responsible. (Jim Marshall is said to know who's behind this, but he wanted cash up-front for names.)

Distinguishing distortion from overdrive

Regardless of who history chooses to praise or condemn as pioneers, drive and distortion are with us for a long, long time to come as fixtures of high-energy music. In fact, they contribute so much to the overall feel of intense musical forms that sometimes seems that more time and effort is spent studying how musical signals can be distorted than how they are generated in the first place.

A good part of this study came as a result of a happy accident of history: the widespread use of valves, or vacuum tubes, as amplification devices. It's common knowledge that tube distortion just plain sounds better than transistor distortion. But until we truly understand *why* this is so, we're not well-equipped to make intelligent decisions about the use of distortion in professional environments. If you want a clear picture of the limitations and challenges involved in using digital circuits and software to emulate tube tone, it will help greatly to understand just what you're hearing.

Let's start by making a clear distinction, albeit an arbitrary one, between distortion and overdrive. Technically speaking, *any* type of unnatural coloration is distortion. Overdrive is just one form of distortion, and overdrive is not limited to the clipping sound generated by an overdriven tube. Transistor distortion can be created by overdriving a transistor. Practically speaking, though, we have to narrow these definitions for the sake of clear communication. So throughout this document, *distortion* refers to broadband clipping effects generally associated with solid-state devices ("transistor overdrive", if you like), while *overdrive* refers to a more resonant, psychologically pleasing clipping effect generally associated with vacuum tube devices.

An important difference between overdrive and distortion

The difference between tube and transistor distortion may be subtle in terms of what you hear. The difference in terms of what you *feel* is enormous, especially if you have the hyperalertness of youth on your side. Transistors driven to clipping tend to add even harmonic series' to the distorted sound. Tubes, on the other hand, tend to add *odd* harmonic series'. Human preference for odd harmonics is a well-known principle of psychoacoustics and extremely important to our discussion, because psychoacoustic preference translates to that elusive "warmth" that we all seem to seek.

...and another...

The type and intensity of the harmonic series generated by a clipped circuit also plays an important role. Some series' are simply more pleasing to some people, especially at a given relative volume to the signal's fundamental frequency. Seventh and ninth harmonics in particular* are known to be widely preferred over all others.

* This preference for sevenths and ninths has implications far beyond designing distortion/overdrive circuits. Some types of audio compression, including some MP3 encoders, incorporate synthesis routines to increase the levels of these harmonics in high frequency content. This helps to compensate for lost high frequency data in the compression process. In fact, some engineers actually apply MP3 compression to some tracks as a means of artificially generating these harmonics.

In other words, some tube designs clip more warmly than others by nature of the series' they generate and the intensity at which those series' are generated. For example, we know that the clipping produced by power amp tubes is generally perceived as warmer than the clipping produced by preamp tubes.

Even transistors vary in their "warmth". We tend to think in terms of silicon transistors as the standard, but 'twern't always so. The first transistor was made from germanium, and that was the standard material used well into the 1960s. As any fan of fuzz-tone can tell you, there's a clear difference between the clipping effects of a germanium transistor and a silicon one.

...and another...

You may know about MOSFETs, the field-effect transistors used in many guitar amplifiers to duplicate the clipping characteristics of tubes. Haul one of those MOSFET amps into a laboratory and you might be surprised to see how close it actually *does* come to vacuum tubes on an oscilloscope. At first glance, it may be hard to fathom why MOSFETs couldn't completely replace tubes in drive circuits.

The fact is that it can't. Tube overdrive consists of far more than just added harmonics. Feedback also plays a critical role. It's feedback that makes those harmonics resonate. We're not talking about hold-your-axe-in-front-of-the-speaker-cabinet feedback, either. We're talking about feedback that occurs inside the circuit as the clipped signal bleeds and recirculates through the amplifier stages. Well-designed modern tube circuits generate feedback in precisely those areas that our ears like most. A well-designed transistor circuit can *try* to duplicate this effect, but it is much harder to achieve in solid state because of the difficulty involved in managing the even harmonics that creep into the signal during feedback.

A transistor and a tube may produce a similar overdrive tone in a "clean" circuit, but tubes offer the distinct advantage of a design which allows for selective enhancement of acoustically pleasing harmonics. Tube devices of all kinds take advantage of this characteristic of tube circuitry by providing *saturation* controls. These allow you to boost or cut the natural feedback and harmonic resonance produced by the circuit, effectively increasing or decreasing the "saturation" of harmonics in the output signal.

...and another...

While saturation effects may be barely noticeable to untrained ears at normal output levels, they become increasingly important as you induce more clipping from the circuit. Another analog circuit, the transformer, plays a critical role in insuring a warm tone. Transistorized circuits usually employ durable output transistors to step voltage up or down or translate voltage from line levels to needed internal levels. Tube circuits, on the other hand, typically employ heavy, expensive wire-wound isolation transformers. These devices more than any other tend to account for the high relative weight of tube devices...and the high cost. A quality isolation transformer for pro audio components can cost as much as $250.

Saturation occurring from the transformer's electromagnetic emanations tends to be smooth and true to the original signal. That low frequency radiation modulates high frequency content in the rest of the circuit in a psychoacoustically pleasing way. Feedback effects from power transistors tend to sound "buzzy" in comparison.

...and another...

Then there's dynamic compression. Compression may come third in our discussion of overdrive characteristics, but it is at least as important to the overall feel of overdriven sound as any other characteristic. There's not a huge amount of difference between the way tubes and transistors compress signal levels. At least, not on paper. All we'll say here is that tubes have a soft knee compression curve while transistors tend to use a hard knee curve. This characteristic of tube circuits is also one of the easiest for transistor circuits to emulate, but the fact remains that the natural tendency for transistors to compress with a hard knee is yet another strike against them in the ear of the listener.

...and it doesn't stop here

While harmonic series', saturation effects and compression curves are the three most important distinguishing characteristics between tube overdrive and transistor distortion, they are by no means the only differences between tube and transistor tone. The speed of a circuit's response to changes in dynamics also plays a role in determining the warmth of the output, as does the manner and degree of internal bleeding of signals not related to transformers and output transistors. The subtle differences in any or all of these characteristics is what gives different manufacturers' tube circuits their unique sounds.

While you might be able to imagine the logistics involved in overcoming most of the shortcomings of transistorized circuits, imagine the difficulty in emulating the relatively subtle differences between circuits made by Marshall and Mesa Boogie, Fender and Vox. Perhaps now you can begin to understand why bassists will happily lug around 90-pound Ampeg SVT tube heads rather than equally powerful, far less expensive solid-state amps that might be one fifth the weight and size...or digital PWM amplifiers that could theoretically fit in a purse. No...let's not go there....

The choice is yours: drive or distortion

Perhaps you can understand now why tube circuits and tube emulation have been the consistent choice of engineers practically since the first transistorized circuits appeared. That's not to say that distortion doesn't have a valid musical place alongside overdrive. Compared to overdrive, distortion effects tend to sound distorted over a wider bandwidth and sound especially grating at the upper end of the frequency range due to their feedback (saturation) response. The grating effect is made even more annoying by sideband harmonics occurring in the wrong series for emotional satisfaction. And when distortion kicks in, it usually kicks in *hard* with little or none of the respect for dynamics typical of tube circuits.

Overdrive has been around long enough that the average engineer intuitively knows when drive is musically valid. A few years ago, the average amateur or semipro recordist was lucky to have one or two tube amps or preamps. Today firms like Twelve Tone and Line 6 provide plugins and hardware that actually *can* generate reasonably faithful emulations of many different well-known tube circuits, and all-around amp simulators are commonplace.

So given the harshness of distortion, why on earth would anyone still want to use it? Quite simply, there may not be any other valid choice.

It's almost an unwritten law that application of drive and distortion is nearly always done in direct relation to the energy level of the music being recorded. You don't routinely hear chamber ensembles running flutes and celloes through Marshall stacks.[*] So why should death-metal and industrial limit itself to tube drive? You've gotta *know* that anything as annoying as transistor distortion has a certain future in pop music. If it weren't so, would Roland have so many transistorized drive pedals?

Drive in the digital domain

While it's a damnably difficult task to design a transistorized circuit capable of overcoming the deficiencies inherent in the transistors themselves. It becomes one heck of a lot easier when you've got a few *million* transistors that you can manipulate. And when you apply a DSP or a computer's CPU to the task, that's precisely what you have at your disposal: millions of transistors.

With those kinds of resources, reasonably accurate emulation of virtually any overdrive or distortion effect becomes nothing more than an exercise in mathematics and modelling. Ask a user of Line 6's POD module, Triaxis amplifier or Amp Farm plugin...it *is* possible to get relatively faithful tube tone out of well-designed digital circuitry. It's just not *easy*, and it's not quite perfect. While some of the newer physical modelling devices and dedicated amp simulators come *very* close, you're just not going to get the full versatility and tonal range of a Marshall JCM-100, a Vox AC30, a Fender Bandmaster and a HiWatt 100 in a plugin, a peripheral card or pedal for a while. There are simply too many

[*] Don't take this to mean that tube circuits aren't used in classical recordings. They're actually used quite extensively to enhance warmth in classical and chamber recordings. But they're used most often in microphone preamplification, equalization and compression, where "tube tone" results from natural characteristics of the circuit. The effect is not nearly as dramatic as deliberate clipping.

variables. At this time, it's a matter of auditioning the tools you have and determining whether you can live with the results that they give you.

EFFECTS UNITS PARAMETER REFERENCE

This section provides an extensive and relatively detailed guide to the parameters in all available effects contained in the 01V's two effects units.

The parameter guide attempts not merely to describe the control, but to offer suggestions for use. In conjunction with the effects backgrounders, this guide should provide even novice recordists with a solid basic understanding of what these units can and cannot do.

Note that the "panels" in the top frame of each section are "live". Selecting a parameter name from this frame takes you directly to its description.

Reverb (Hall, Room, Stage, Plate)

The 01V's built-in reverb algorithms might not sound as sweet to your ears as a $500 dedicated plug-in, an exact model of your favorite cathedral, or a $2,000 outboard unit. But with a little tweaking you're sure to find them serviceable for a wide range of applications, and that includes application as a mastering reverb.

As an added bonus, the real-time response of the reverbs to parameter changes makes them superb for learning the ins and outs of digital reverb tweaking and for experimenting with new applications of reverb-type effects. The potential range of effects possible from these units is far beyond the typical application of ambience, and it's worth spending an hour or two just playing with the parameters to discover the full range of unusual effects lurking within these algorithms.

If you can spare both 01V effects for a single combined effect, and you have an ear for subtle tonal colorations, you might find that routing the output of the **Amp Simulator** effect into a reverb effect offers you a whole new tonal palette for reverbs. In order to make optimum use of this combination, the signal should be routed through the amp simulator before being fed into the reverb. This sets up the amp simulator as a "pre-EQ" to the reverb, which offers a much greater initial control over the color of the input signal than the reverb alone. The effect might seem subtle, but it is definitely noticeable.

Reverb Time (not remote-controllable)

This is the total time elapsed between the moment at which a single sample enters the processor and the moment at which reverb output for that sample decays to zero. This parameter cannot be controlled by C-Console

Range:		0.3 - 99s
Increment:	0.3s - 5s:	0.1s
	5s - 10s:	0.5s
	10s - 20s:	1.0s
	20s-99s:	5.0s
Default:	Hall:	3.2s
	Room:	1.0s
	Stage:	1.8s
	Plate:	2.6s

Initial Delay

This parameter determines the amount of time from when the audio signal arrives at the reverb processor to when the first reflections are generated. An additional delay can be applied to early reflections if needed. (Initial delay is occasionally referred to as "predelay".)

The exceptionally wide range available in this parameter allows for some interesting reverberation effects. For example, in a 120bps composition, an initial delay of 250ms will make first reflections (almost always the loudest portion of the reverb) occur on the downbeat. If the needed delay time is short enough, this allows the reverb effect itself to perform the delay so that a plug-in, external device or the second effects unit doesn’t have to be used as a delay line.

Range:		0 - 500ms
Increment:		0.1ms
Default:	Hall:	36ms
	Room:	15ms
	Stage:	46ms
	Plate:	42ms

High frequency decay ratio

This specifies the rate at which high-frequency content will be removed from the reverb signal in relation to total reverb time. Smaller values produce shorter decay times, giving the impression of the reverb “darkening” in tone at a more rapid rate. The audible effect is similar to that of the decay or release parameter in the filter envelope of a synthesizer.

High (or low) frequency content does not decay *completely* from the signal. As a reflecting sound wave’s volume drops, a smaller percentage of high frequency content disappears from the signal with each reflection. There is no direct control over the base cutoff frequency or the shape of the decay slope for either the high or low frequency decay ratios.

Low and high frequency decay ratio parameters are used to mimick the effects of different types of floor, wall and ceiling materials. Structures with hard, smooth surfaces such as concrete or polished marble floors, vaulted concrete or rock walls or steel ceilings tend to have lower HF decay ratios than movie theatres and concert halls which are designed to prevent this. Theatres often lay carpet on the walls to insure high HF decay ratios and greater intelligibility from movie sound systems. Cathedrals are often designed to *prevent* unwanted HF decay, since the structure is often designed to produce pleasing series’ of reflections across a wide range of the audible spectrum.

The rate at which *low* frequencies decay in physical enclosures is determined partly by the type of surface from which the sound reflects, but the resonance of the structure itself also plays an important role. Structures designed with acoustics in mind tend to produce very little unnatural resonance at low frequencies. Sports arenas and auditoriums might absorb low frequencies at an annoyingly slow rate, producing strong resonants. Cathedrals are often designed to allow heavy bass from pipe organs to reflect well but not to resonate the entire structure. The 01V gives you more control over LF decay than HF decay to help account for the resonant effects of low frequencies.

In pop music, low and high frequencies should be tailed off relatively quickly using relatively high ratios for all but the dominant instruments in a mix. This helps prevent rhythm instruments from taking presence away from lead instruments. Filtering upper and lower frequencies from reverbs becomes more crucial as the density of the mix increases.

Range:		0.1 - 1.0
Increment:		0.1
Default:	Hall:	0.3
	Room:	0.4
	Stage:	0.7
	Plate:	0.3

Low frequency decay ratio

See above (**High frequency decay ratio**) for background information.

Range:		0.1 - 2.4
Increment:		0.1
Default:	Hall:	1.4
	Room:	1.2
	Stage:	0.8
	Plate:	1.0

Diffusion

This parameter determines the perceived width of the reverb's stereo field. The amount of diffusion in a reverb relates to how the reflections "spray" themselves across the stereo field. High diffusion values tend to sound more like reverberation from a sound source in the center of a large enclosure; lower values narrow the reverb image similar to the sound heard from the back of a long room or concert hall. Higher values tend to be used for room reverbs; lower values to mimic the sound of a concert hall or auditorium. The overall impact of diffusion on the 01V's reverbs is relatively subtle compared to many other reverb units you may encounter. Diffusion values on some plug-ins and outboard effects can literally range from unbearably narrow to widths so great that they almost sound like 3D spatial enhancement effects. The 01V's diffusion values seems to lie within a realistic range, usable and relatively natural at both extremes.

Range:		0 - 10
Increment:		1
Default:	Hall:	8
	Room:	8
	Stage:	6
	Plate:	6

Density

Reverb density generally refers to the perceived "richness" of the reverb and roughly translates into the total number of reflections processed within the reverb. Appropriate densities depend upon the reverb algorithm and enclosure you want to emulate. A poorly-insulated, regularly-shaped room is better emulated with a lower density value, which corresponds with fewer angles of reflection. Concert halls, as you can see from the default values, tend to be designed and built to produce a higher reverberation density.

In terms of "real life" ambience, density tends to be a function of both the complexity of the inner surface of a structure and the regularity of angles from which sound can reflect. Cathedrals tend to produce "dense" reverberation, even though the inner surfaces tend to be hard and smooth, because there are more angles from which the sound waves can bounce and collide. Auditoriums and warehouses tend to produce sparse, harsh-sounding reverberation because the surfaces are regular and symmetrical, and sound waves have fewer angles and surfaces from which to reflect.

Range:		0% - 100%
Increment:		1%
Default:	Hall:	100%
	Room:	98%
	Stage:	65%
	Plate:	70%

High Pass Filter

The high pass and low pass filters are "tone controls" for the reverb signal, but don't be fooled by this simple description. These controls play a very important role in generating realistic reverb effects.

Signals reflecting off of softer surfaces tend to lose a lot, if not all, of certain upper frequency ranges after just one or two reflections. Unfinished wood, for example, tends to be a good at rapidly absorbing high frequencies thanks both to its softness and the small irregularities in its surface. Polished, finished wood tends to be less effective at high-frequency absorption. Polished stone tends to “absorb” even less high end than polished wood, but rough stonework will rapidly eat up the top end of the audible spectrum. (“Absorption” is perhaps not the best description of this effect; in fact, the upper frequencies tend to be phase-cancelled into oblivion rather than “eaten” by the reflecting surface.)

Enclosure size also has a significant effect on low-frequency rolloff (high pass). Small rooms just can’t reflect those long low frequency waves as well as larger rooms, so lows tend to be absorbed after just a few reflections.

The high and low pass filters in the 01V’s reverbs are designed to mimic the effect that various materials have on high frequency content. You may notice that they do not completely remove all frequency content above the target frequency. Instead they apply a rolloff filter, similar in slope to the tone control EQ, to give the rolloff a more natural sound. The slope appears to be 6dB of gain reduction per octave (first-order).

Range:		Through - 8kHz
Increment:		1 semitone
Default:	Hall:	Through
	Room:	83Hz
	Stage:	Through
	Plate:	Through

Low Pass Filter

See above **(High Pass Filter)** for background information.

Range:		50Hz - Through
Increment:		1 semitone
Default:	Hall:	6.72kHz
	Room:	8kHz
	Stage:	10.6kHz
	Plate:	5.33kHz

Early Reflection Delay

This specifies the elapsed of time from the moment a sample enters the reverb algorithm to the production of the first reflected sound from that sample. This parameter is normally set to indicate the desired enclosure size; i.e. larger spaces require longer early reflection times. The larger the ambient space, the longer it takes for sound to travel from its source to a distant surface and reflect back to the source again.

This parameter specifies the delay before the *first* reflection is heard. Early reflection delays arrive in a cluster, not all at once. The total time required for the early reflection cluster varies widely depending on which reverb algorithm is used. Early reflection delay doesn’t begin counting down until the initial delay time has expired.

Range:		0 - 100ms
Increment:		0.1ms
Default:	Hall:	2.0ms
	Room:	6.0ms
	Stage:	12.0ms
	Plate:	16.0ms

Early Reflection Balance

The first set of reflected sound waves are by far the loudest, clearest part of the reverb. A correct balance of early reflections with decaying reflections is critical to the realism of the reverb sound. But realism isn’t always the goal. Loud early reflections might impart a useful metallic feel or “thickening” to tracks in high-energy recordings. Moody or hypnotic music might actually benefit from having early reflections set to a low level or removed completely, resulting in a soft, diffuse, “dreamy” ambience. Keep in mind that both extremes are unnatural and should be considered effects rather than natural ambiences. If realistic ambience is the goal, you should get the best results from values at or near the 01V defaults.

If the reverb sounds like it is detracting from the clarity of the dry signal, you can usually correct this problem by reducing the early reflection balance. The resulting ambient effect may sound slightly unnatural, but you can look upon this as no different from the effect that airbrushing of blemishes has on a photograph. Audiophiles might cringe at the sound of unnaturally low early reflection balance; the average listener with an untrained ear, on the other hand, usually finds the effect more pleasing than an acoustically accurate balance.

Range:		0% - 100%
Increment:		1%
Default:	Hall:	44%
	Room:	50%
	Stage:	39%
	Plate:	50%

Gate Level

Gate level, often referred to as “threshold”, is the level at which the noise gate allows reverb signals to pass. Signals above this level pass through the gate; signals below this level are prevented from passing through. How this parameter behaves in practice depends on the values of the **Hold** and **Decay** parameters.

Gated Reverb is available as a unique effect with its own unique algorithms. That effect uses an optimized algorithm for gate effects, but all four of the dedicated reverb algorithms in the 01V can be applied as gated ‘verbs if you want something a little different. The gate section of the reverb has four parameters for fairly wide control over the reverb envelope.

The gate is applied to the reverb, *not* to the input signal. Gate parameters should normally be left alone when you are trying to achieve natural-sounding ambient effects.

Gated reverb is generally considered to be a one-trick pony, an effect with only one sound: the “Phil Collins snare” sound or accent reverb heard so often on breakbeat loops. But it’s not nearly as limited as it might seem. For example, applying maximum attack to a hall reverb on a percussion track such as a snare drum, or to stabs or hits, can create interesting “swell” or “bloom” effects.

Gating can also be applied to reverb to accentuate the beat or create “rhythmic rheverb”. Try using the **Hold** and **Decay** parameters to create a sharp decay curve just before the downbeat on a dance track. If it’s done correctly, the sharp drop in reverb volume before the downbeat will create a brief moment of silence that seems to make the downbeat appear to jump out of the mix. Setting **ATTACK** to **0** and **GATE LVL** (gate threshold) to about **-12** will chop off a few early reflections and produce a reverb that has zero perceived attack. Combined with a long **INI. DLY** and a long **E/R DLY,** this produces an unusual single-repeat echo effect up to 350ms long that gives the echo a very sharp attack with little or no perceived early rise in volume.

Range:		Off (all pass) - 0dB (lowest threshold level: -60dB)
Increment:		1dB
Default:	Hall:	Off
	Room:	Off
	Stage:	Off
	Plate:	Off

Attack

Attack refers to the rise time in the gate envelope. This is the amount of time it takes for the sound to rise from 0dB (or -Infinity) to full volume. Low attack time values preserve a natural sound in the reverb. Longer attack times impart unusual attenuation effects to the early reflections. At the upper end of the range, you’ll begin to notice a swell or “bloom” effect.

Range:		0 - 120ms
Increment:		1ms
Default:	Hall:	4ms
	Room:	4ms
	Stage:	4ms
	Plate:	4ms

Hold (not remote-controllable)

The reverb's noise gate can optionally be "held open" for a specified period of time. This insures that reverb signal can pass through for this period of time even if the reverb signal falls below the gate threshold level. If you want to insure a natural-sounding tail, the **Hold** time should be at least as long as the total decay time for the longest decay period in the track. Shorten the **Hold** time and adjust **Decay** if you want to generate a more artificial-sounding effect.

Range:		0.2ms - 2.13s
Increment:		variable throughout the range
Default:	Hall:	325ms
	Room:	174ms
	Stage:	179ms
	Plate:	197ms

Decay (not remote-controllable)

This specifies how long it will take for the signal to fall to zero output when the reverb output level drops below the gate's threshold level. Smaller values impart a sharp, highly artificial-sounding tail to the reverb. Larger values will produce a more natural-sounding tail.

This parameter can occasionally be used in dense mixes to help prevent multiple reverbs on different tracks or submixes from muddying the mix. Shortening the duration of a reverb often thins out the reverb a bit too much since it reduces the reverb level for the entire duration of the decay. But when working with tracks that have a lot of musical space, a bit of decay slope in the reverb can allow the first second or two of reverberation to remain full and clear while reducing the duration of lower-level ambient noise as the reverb decays below a certain level. This is often enough to "clear" a bit of ambient mud.

Range:		6ms - 46s
Increment:		variable throughout the range
Default:	Hall:	75ms
	Room:	6ms
	Stage:	6ms
	Plate:	93ms

Early Reflections/Gated Reverb/ Reverse Gated Reverb

The 01V's early reflection effects sound somewhat gimmicky when applied on their own. But the pleasing sounds they produce make them gimmicks with a lot of staying power and a wide range of potential applications.

Gated and reverse reverbs have been so heavily used in many dance genres, dating back from the present to the mid-1980s, that listeners often find them boring or cliché unless they're applied in a particularly novel way. With this in mind, it may be wise to use particularly careful judgement when applying these effects to commercial pop and dance tracks.

If you can afford to dedicate both effects units to a single effect, you will find that adding the **Early Reflections** effect to **Reverb** can considerably "thicken" the overall sound of a reverb, especially if the early reflections of the reverb alone sound too thin or metallic in a given application.

The implementation of feedback gain in all three of these effects make them especially well suited for use as novel-sounding echo/delay effects.

Type (not remote-controllable)

Specifies the reverb algorithm which will define how the cluster of early reflection echoes will sound. The various reflection clusters are described as follows.

Effect	Type	Description
Early Reflections:	Small Hall:	A tight cluster of reflections with a sound similar to flanging. Somewhat breathy or metallic sounding but tight.
	Large Hall:	A looser cluster of reflections spread over a longer period. This cluster type can often be used as a novel-sounding alternative to gated reverb when a large number of reflections is specified.
	Random:	Randomly delays the reflections in the cluster. This cluster has a longer total period than the **Large Hall** type. “Clickety”-sounding when few early reflections are specified (especially effective with <8 reflections); smoother-sounding when more reflections are used. When combined with a small **ROOMSIZE** value, it produces an almost scratchy effect.
	Reverse:	Natural early reflection sets are brightest and loudest at the beginning of the cluster, reflecting the fact that the first reflections travel the shortest distance and suffer the least loss of volume and timbre. The **Reverse** type gives the cluster an unnatural, rising volume envelope. When a pure wet signal is used, this type can significantly soften a track’s attack. The base algorithm for the reverse type sounds like a medium between the large and small hall algorithms.
	Plate:	Plate reverbs tend to produce a relatively regular set of reflections, meaning that reflections return to the source at regularly-spaced intervals. This option applied to early reflections imparts a “flangey” sound, but the tonal range is wider. The “static flange” effect is highly metallic with larger room sizes; much less metallic than other algorithms when smaller room sizes are dialled in.
	Spring:	Emulates the reverb effect produced by a spring reverb tank. The cluster’s period is longer than that of any other algorithm except large hall, and the variation in volume between first and last reflection appears to be the lowest of any of the algorithms. This results in a rather pleasing, breathy tone when other values are set to their defaults.
Gated reverb:	Type A/B:	Non-functional in C-Console for this reverb type.
Reverse gated reverb:	Type A:	A relatively slow-rising reflection cluster. Larger room sizes produce audibly more “brittle” results a noticeably longer “tail” than Type B.
	Type B:	A faster-rising, tighter cluster than Type A, expressing a set of reflections from a regularly-shaped (near-cube) enclosure.

Room Size

Specifies the size, or total volume, of the virtual enclosure used to produce the reflections. The size of the enclosure determines the period (total elapsed time) from the first reflection in the cluster to the last.

Range:		0.1 - 20
Increment:		0.1 (**Note:** we were unable to determine the unit of measurement used for this parameter. We assume it is a measurement of cubic volume, but the audible results do not correspond with cubic metres.)
Default:	Early reflections:	2.4
	Gated reverb:	1.6
	Reverse gated reverb:	2.8

Liveness

The "liveness" of an ambient space is a measure of how well it reflects sound from multiple surfaces without losing volume or timbre. Since early reflections include both direct reflections (one bounce from a remote surface) and multiple indirect reflections (e.g. off the right wall, off the back wall, off the left wall and back to the source), the degree of liveness can be used to fine-tune the period (total elapsed time) for all reflections.

Range:		0 - 10
Increment:		1
Default:	Early reflections:	7
	Gated reverb:	10
	Reverse gated reverb:	10

Initial Delay

This is the length of time it takes before the first reflection is generated. The exceptional range of this parameter makes this effect useful as a single-repeat delay effect. For example, delayed early reflections can be beat-matched with the dry signal using an appropriate value (e.g. 250 or 500ms for a 120bpm tempo). When combined with the feedback gain parameter, any of these three effects can be used as an echo.

Range:		0 - 500ms
Increment:		0.1ms
Default:	Early reflections:	5ms
	Gated reverb:	3ms
	Reverse gated reverb:	35ms

Diffusion

This parameter defines the perceived width of the reverb's stereo field. The effect of diffusion on early reflections is more pronounced than it is on a standard reverb effect. See the **Reverbs** section for additional information.

Range:		0 - 10
Increment:		1
Default:	Early reflections:	7
	Gated reverb:	8
	Reverse gated reverb:	8

Density

Generally defines the "richness" of the reverb or early reflections. See the **Reverbs** section for additional information.

Range:		0% - 100%
Increment:		1%
Default:	Early reflections:	85%
	Gated reverb:	96%
	Reverse gated reverb:	96%

Early Reflection Number

This specifies the total number of early reflections generated by the effect. The more reflections produced, the less brittle the sound, although "thickness" tends to be perceived as a flanging/phase cancellation effect when smaller room sizes are used. Numbers significantly larger or smaller than the default tend to sound unnatural as ambient reflections, but this shouldn't deter experimentation since none of these three effects are intended to emulate natural ambiences.

This parameter is normally adjusted in conjunction with the room size parameter.

Range:		1-19
Increment:		1
Default:	Early reflections:	11
	Gated reverb:	18
	Reverse gated reverb:	12

High Frequency Decay Ratio

When applied to early reflections, this plays the same role as HF decay ratio plays in delay and echo effects. See the **Reverbs** section for additional information on HF decay in natural reverbs; see the **Mono Delay** section for additional information on HF decay in echo effects.

Range:		0.1-1
Increment:		0.1
Default:	Early reflections:	0.4
	Gated reverb:	0.8
	Reverse gated reverb:	0.8

Feedback Gain

This has the same function as the feedback control on a delay line. This specifies the percentage of original volume the signal should have with each successive echo.

There is no way to specify an exact number of repeats for the delay without automating this parameter so that it is reduced to **0** at appropriate points in the track. Feedback gain of **0** produces a single repeat. See the **Mono Delay** section for additional information.

Range:		-99% - +99%
Increment:		1%
Default:	Early reflections:	+2%
	Gated reverb:	+9%
	Reverse gated reverb:	+7%

High Pass Filter

The high and low pass filter dials act as a shelving EQs or "rolloff" filters for the gated/reverse reverb or early reflection cluster. These dials are reasonably effective "simple tone controls" for these effects because careful adjustment is less critical with these effects than it is when attempting to create natural-sounding reverbs. The filters appear to be 6dB per octave (first order) shelving controls. See the **Reverbs** section for additional information.

Range:		Through - 8kHz
Increment:		1 semitone
Default:	Early reflections:	Through
	Gated reverb:	Through
	Reverse gated reverb:	Through

Low Pass Filter

See above.

Range:		Through - 8kHz
Increment:		1 semitone
Default:	Early reflections:	11.9kHz
	Gated reverb:	8.98kHz
	Reverse gated reverb:	12.6kHz

Mono Delay

This is a standard one-in, one-out digital delay line comparable in versatility to early 16-bit rack units. It lacks the ability to apply multiple taps to the delay, but it does allow very fine adjustment of delay time (tenths of a second) and a reasonably long maximum delay time to compensate for this lack.

Delay Time

Specifies the delay time and the time between repeats of the delayed signal.

Range:	0 - 2730ms
Increment:	0.1ms
Default:	250.0ms

Feedback Gain

The delayed signal can optionally be fed back into the delay line to generate additional echoes. Feedback gain reduces the volume of the output signal prior to each pass through the delay, so that each echo has the specified percentage of volume of the preceding echo. This indirectly determines the total number of echoes produced, but there is no specific way to determine how many echoes will be produced unless this parameter is reduced to **0%** using automation at a certain point in track playback.

When set to 0%, only one echo is produced. Values higher or lower than 0% produce progressively more echoes as the input signal drops in volume more slowly with each repeat.

This effect provides positive and negative feedback gain values to allow the echoes to have inverse phase to the input signal. This offers additional tonal control which you'll find especially useful when using short delays. Values greater than 0% produce echoes in phase with the input signal; values less than 0% produce echoes in inverse phase with the input.

Range:	-99% - +99%
Increment:	1%
Default:	+38%

High Frequency Decay Ratio

This parameter controls the rate at which high-frequency content will be removed from the delayed signal. Each succeeding echo will contain less and less high frequency content.

This parameter has no practical equivalent in “real life” acoustics. Instead, it tends to mimic the effect of tape echo devices which tend to lose high frequency content with each succeeding echo. The effect of this parameter is relatively subtle at low levels, but it may be useful in preventing digital distortion or aliasing noise from creeping into delay effects which require a large number of repeats.

The HF gain reduction per echo is relatively low. Our guess, based on testing, is that the reduction is between 3 and 4dB per echo at the default 0.6 using a shelving algorithm at 1kHz.

Range:	0.1 - 1.0
Increment:	0.1
Default:	0.6

High Pass Filter

This applies a 2nd order high pass filter to the input signal prior to feeding it through the delay. This allows you to "chop" all low frequency content out of the delayed signal *before* it passes through the delay line. Unlike the HF decay control described above, this is a one-shot filter performed before the input signal reaches the delay unit; it doesn't filter the signal over and over each time it passes through the delay.

Combined with the low pass filter, these dials act as effective tone controls which can help insure that relatively loud echoes don't overpower the mix.

Range:	Through - 8kHz
Increment:	1 semitone
Default:	Through

Low Pass Filter

This applies a 2nd order *low* pass filter to the input signal prior to feeding it through the delay. This allows you to "chop" all high frequency content out of the delayed signal. As with the high pass filter described just above, it does not have any effect on reduction of high frequency content from the delayed sound through progressive feedback echoes.

Range:	50Hz - Through
Increment:	1 semitone
Default:	Through

Stereo Delay

This is a one-in, two-out digital delay line comparable in versatility to early 16-bit stereo-output rack units. It lacks the ability to apply multiple taps to the delay. Delay times are relatively short compared to newer stand-alone rack units, but with over a second of delay this should not be a significant limitation. As with the **Mono Delay,** it allows for very fine adjustment of delay time (0.1ms), and faithfully preserves the quality of the input signal. The **Echo** effect offers more control over routing without compromising flexibility.

Delay Time Left/Delay Time Right

Specifies the delay time and the time between repeats (if any) of the delayed signal. In this effect, delay times can be set independently for the left and right output channels of the delay. Because this effector is two independent delays, maximum delay times for this effect are half the maximum delay time for **Mono Delay** and **Modulation Delay.**

Range:		0 - 1350ms
Increment:		0.1ms
Default:	Left:	250.0ms
	Right:	375.0ms

Feedback Gain Left/Feedback Gain Right

The delayed signal can optionally be fed back into the delay line to generate additional echoes. Feedback gain reduces the volume of the output signal prior to each pass through the delay, so that each echo has the specified percentage of volume of the preceding echo. Feedback can be adjusted independently for the left and right channels. See the **Mono Delay** section for additional information.

Range:		-99% - +99%
Increment:		1%
Default:	Left:	+40%
	Right:	+25%

High Frequency Decay Ratio

This parameter controls the rate at which high-frequency content will be removed from the delayed signal. Each succeeding echo will contain less and less high frequency content. See the **Mono Delay** section for additional information.

Range: 0.1 - 1.0
Increment: 0.1
Default: 0.5

High Pass Filter

This applies a 2nd order high pass filter to the input signal prior to feeding it through the delays. This allows you to "chop" all low frequency content out of the delayed signal. See the **Mono Delay** section for additional information.

This parameter cannot be set independently for left and right channels.

Range: Through - 8kHz
Increment: 1 semitone
Default: Through

Low Pass Filter

This applies a 2nd order low pass filter to the input signal prior to feeding it through the delays. This allows you to "chop" all high frequency content out of the delayed signal. See the **Mono Delay** section for additional information.

This parameter cannot be set independently for left and right channels.

Range: 50Hz - Through
Increment: 1 semitone
Default: 10.0kHz

Modulation Delay

This is a basic one-in, two-out digital delay line comparable in versatility to the early 16-bit stereo rack units with modulators. It lacks the ability to apply multiple taps to the delay and long delay times, but it does allow very fine adjustment of delay time (0.1ms), permitting a wide range of modulation delay effects especially at shorter delay times, offers phase inversion using the **FB GAIN** dial for additional tonal control over swept delay effects, and faithfully preserves the quality of the input signal.

Delay Time

Specifies the time between repeats of the delayed signal. The range is identical to that of the **Mono Delay.** Most modulated delay effects will not require more than a small fraction of this device's delay buffer.

Range: 0 - 2725ms
Increment: 0.1ms
Default: 250.0ms

Feedback Gain

The delayed signal can optionally be fed back into the delay line to generate additional echoes. Feedback gain reduces the volume of the output signal prior to each pass through the delay, so that each echo has the specified percentage of volume of the preceding echo. See the **Mono Delay** section for additional information.

Range: -99% - +99%
Increment: 1%
Default: +40%

High Frequency Decay Ratio

This parameter controls the rate at which high-frequency content will be removed from the delayed signal. Each succeeding echo will contain less and less high frequency content. See the **Mono Delay** section for additional information.

Range: 0.1 - 1.0
Increment: 0.1
Default: 0.6

Frequency

Sets the oscillation rate for the modulator. Low settings sweep the pitch modulator slowly; higher settings sweep it more rapidly. (While it is the delay time which is modulated, the audible effect sounds like pitch modulation.) In this effect, the modulator is applied to the delay time *as the signal passes through the delay*. This means that the signal sounds pitch-modulated at the output, and each successive pass through the delay (feedback) applies more modulation.

While this effect provides a wide range of frequencies, actual modulation *depth* is relatively low. This doesn't prevent the implementation of dramatic modulator sweeps, but it can make them rather tricky to achieve.

This unit has a relatively unusual feature which may be useful in matching oscillator sweeps to beats. It resets the modulator's wave to the start of a wave when the delay detects no input for a certain period of time, indicating that a gate trigger is used. What this means is that the modulator doesn't start to oscillate until it detects an input signal to be processed. This may make the effect particularly difficult to control in some cases, but it also makes modulation very easy to predict if you are aware of significant gaps in the input track.

Range: 0.05Hz - 40Hz
Increment: 0.05Hz
Default: 0.85Hz

Depth

This specifies the modulation depth possible in the delayed signal. In terms of what you hear, modulation depth corresponds to the range of pitch variation in the output as the modulator sweeps through a full cycle. The range for this parameter is sufficient for vibrato and chorus effects, but with a maximum depth of approximately 2-3 semitones, you may find it insufficient for dramatic pitch sweeps comparable to what you can achieve using a synthesizer's modulators.

Range: 0 - 100%
Increment: 1%
Default: 60%

High Pass Filter

This applies a 2nd order high pass filter to the input signal prior to feeding it through the delays. This allows you to "chop" all low frequency content out of the delayed signal. See the **Mono Delay** section for additional information.

This filter can be especially useful with bass instruments. Modulated delays tend to make bass instruments sound as though they are constantly diving in and out of tune, especially when doubling or chorus-type effects are used. But when the high pass filter is set higher than the fundamental frequency for the highest note in the instrument's track, it can help to insure that the effect doesn't create problems caused by "wobbly fundamentals".

Range:	Through - 8kHz
Increment:	1 semitone
Default:	Through

Low Pass Filter

This applies a 2nd order low pass filter to the input signal prior to feeding it through the delays. This allows you to "chop" all high frequency content out of the delayed signal. See the **Mono Delay** section for additional information.

Range:	50Hz - Through
Increment:	1 semitone
Default:	12.6kHz

Left/Right/Center Delay

This is the 01V's closest approximation of a multitap delay, although it may not meet your definition of a typical multitap delay. While it is useful for creating stereo delay effects, it cannot produce effective ping-pong, spray or "bounce" delays.

Compare this effect with the **Echo** effect before deciding on this effect for stereo echoes. The **Echo** effect is more suited to ping-pong and bounce effects.

Delay Time Left/Delay Time Right/Delay Time Center

Specifies the delay time and the time between repeats of the delayed signal. In this effect, delay times can be set independently for the left, right and center outputs of the delay. A value of **0** effectively shuts off the delay output for that channel, so you can use only two of the three delay channels if you like.

While delay times for the **Stereo Delay** effect are half the maximum delay time for **Mono Delay** and **Modulation Delay,** the delay times here for all three delays are the same as the **Mono/Modulation Delay** maximums. This is because when feeding back the three echos, they are treated as a single unit rather than processed separately. (See **Feedback delay** below.)

Range:		0 - 2730ms
Increment:		0.1ms
Default:	Left:	250.0ms
	Center:	750.0ms
	Right:	500.0ms

Feedback Gain Left/Feedback Gain Right/Feedback Gain Center

The delayed signal can optionally be fed back into the delay line to generate additional echoes. Feedback gain reduces the volume of the output signal prior to each pass through the delay, so that each echo has the specified percentage of volume of the preceding echo. Feedback gain is adjustable independently for left, center and right channels, and can also be adjusted as a whole for all three channels (see below). See the **Mono Delay** section for additional information.

Some stereo echo effects may suffer from phase cancellation problems when auditioned in mono (outputs panned dead center). Negative feedback gain values for one or more delay channels can often help to minimize or overcome this problem without significantly altering the sound of the full stereo output.

Range:		-99% - +99%
Increment:		1%
Default:	Left:	+80%
	Center:	+80%
	Right:	+80%

Feedback Delay

Feedback in this delay effect is handled somewhat differently from the way it is handled in the **Stereo Delay.** The algorithm for this effect treats the three delays as a single unit. Using the default values, the first set of delays occur at 250ms, 500ms and 750ms. But if feedback is applied, the left channel doesn't repeat every 250ms, nor does the center channel repeat every 500ms. Instead, feedback echoes repeat based on the time value specified in the **Feedback delay** parameter.

This parameter cannot be set independently for left, center and right channels.

The audible effects of this parameter can be tested rather easily by applying a short single-shot sample (perhaps a snare drum, "Test!" or short keyboard stab) to the default delay times for all three delays. Vary the feedback delay between 500ms and 1500ms to get a clear picture of how this parameter behaves.

Range:	0 - 2730ms
Increment:	0.1ms
Default:	750ms

Feedback Gain

After setting feedback gain for each of the three delays independently, you can also apply an additional feedback using this dial, which feeds back the output of all three delays at once.

This parameter cannot be set independently for left, center and right channels. See the **Mono Delay** section for additional information.

Range:	-99% - +99%
Increment:	1%
Default:	+40%

High Frequency Decay Ratio

This parameter controls the rate at which high-frequency content will be removed from the delayed signal. Each succeeding echo will contain less and less high frequency content. See the **Mono Delay** section for additional information.

This parameter cannot be set independently for left and right channels.

Range:	0.1 - 1.0
Increment:	0.1
Default:	0.6

High Pass Filter

This applies a 2nd order high pass filter to the input signal prior to feeding it through the delays. This allows you to "chop" all low frequency content out of the delayed signal. See the **Mono Delay** section for additional information.

This parameter cannot be set independently for left and right channels.

Range:	Through - 8kHz
Increment:	1 semitone
Default:	Through

Low Pass Filter

This applies a 2nd order low pass filter to the input signal prior to feeding it through the delays. This allows you to "chop" all high frequency content out of the delayed signal. See the **Mono Delay** section for additional information.

This parameter cannot be set independently for left and right channels.

Range:	50Hz - Through
Increment:	1 semitone
Default:	Through

Echo

This is a duplication of the **Stereo Delay** effect, but it provides more sophisticated routing for producing unusual delay effects. Some of the more sophisticated effects possible with this algorithm are difficult to describe and best understood through experimentation and discovery.

Try removing the right or left channel by reducing its delay time, feedback gain and right-to-left feedback gain to zero. Gradually mix this channel back in, varying the delay and feedback parameters, to hear how the two delays interact with each other.

When attempting to create stereo echo effects, set one channel to a few milliseconds above the target delay time for a perfect echo and the other channel to a few milliseconds below the target delay time. When auditioning the effect, remember to solo the track in mono (panned dead center) as well as stereo to insure that phase cancellation is not resulting in lost fidelity in mono playback or in a narrow stereo field. Using negative feedback gain for one or both channels can often overcome phase cancellation problems that appear only when auditioning echo effects in mono.

Delay Time Left/Delay Time Right

Unlike the delay time controls for the other 01V delay units, these dials specify the time before the *first* repeat of the delayed signal. Additional controls are provided for subsequent delay times. In this effect, delay times can be set independently for the left and right output channels of the delay. Because this effector is two independent delays, maximum delay times for this effect are half the maximum delay time for **Mono Delay** and **Modulation Delay.**

Range:		0 - 1350ms
Increment:		0.1ms
Default:	Left:	176.0ms
	Right:	246.0ms

Feedback Delay Left/Feedback Delay Right

Echoes (feedback delays) can be set to trigger at a different interval from the initial delayed signal. Any additional echoes after the first echo will trigger at the interval specified in this parameter.

Range:		0 - 1350ms
Increment:		0.1ms
Default:	Left:	246.0ms
	Right:	176.0ms

Feedback Gain Left/Feedback Gain Right

This specifies the percentage of original volume the signal should have with each successive echo, and indirectly determines the total number of echoes produced. Feedback gain is adjustable independently for left and right channels. See the **Mono Delay** section for additional information.

Range:		-99% - +99%
Increment:		1%
Default:	Left:	+50%
	Right:	-50%

Left-to-Right Feedback Gain/Right-to-Left Feedback Gain

Routing in this delay can be configured to allow one channel's delayed signal to be fed into the other channel's delay. This parameter controls how much of the specified channel's output volume will be fed into the other channel's delay. When different feedback delay times are used for each channel, a range of unusual delay effects ranging "ping-pong" (back-and-forth stereo echoes) to "bounce" (the effect you get from dropping a coin or steel ball on a hard surface) can be generated.

Range:		-99% - +99%
Increment:		1%
Default:	Left:	+35%
	Right:	+35%

High Frequency Decay Ratio

This parameter controls the rate at which high-frequency content will be removed from the delayed signal. Each succeeding echo will contain less and less high frequency content. See the **Mono Delay** section for additional information.

This parameter cannot be set independently for left and right channels.

Range:	0.1 - 1.0
Increment:	0.1
Default:	0.4

High Pass Filter

This applies a 2nd order high pass filter to the input signal prior to feeding it through the delays. This allows you to "chop" all low frequency content out of the delayed signal. See the **Mono Delay** section for additional information.

This filter is applied prior to the initial delay, so it does not have any effect on reduction of high frequency content from the delayed sound through progressive feedback echoes. This parameter cannot be set independently for left and right channels.

Range:	Through - 8kHz
Increment:	1 semitone
Default:	Through

Low Pass Filter

This applies a 2nd order low pass filter to the input signal prior to feeding it through the delays. This allows you to "chop" all high frequency content out of the delayed signal. See the **Mono Delay** section for additional information.

This filter is applied prior to the initial delay, so it does not have any effect on reduction of high frequency content from the delayed sound through progressive feedback echoes. This parameter cannot be set independently for left and right channels.

Range:	50Hz - Through
Increment:	1 semitone
Default:	5.99kHz

Chorus

This is a relatively complex chorus effect with enough tweakability to accommodate most requirements. While it lacks a feedback or "multiplier" control for mixing multiple chorused signals with the input, it's still quite serviceable. The lack of feedback/multiplier parameters could be related to the fact that virtually all Yamaha effectors which include a chorus effect also contain the **Symphonic** effect. Chorus effects should be compared to comparable symphonic effects prior to actual use; in many cases symphonic is more appropriate.

The perceived quality of this effect depends a great deal on the balance between the dry and effected signal. Output from the chorus is pure wet signal, so an actual chorus effect is not audible unless the wet and dry levels are fairly close to one another.

No high pass filter is available in this effect, making it unsuitable for many bass instrument tracks where the fundamental must be preserved. Chorusing can be emulated for bass instruments in the **Modulation Delay** using delay times of 12-40ms, and its high pass filter can be applied to preserve the fundamental. While the **Modulation Delay** does not contain an amplitude modulator, emulated chorus effects using this device are often quite serviceable for special requirements such as bass instruments.

Frequency

The chorus effect has two modulators: one for pitch (delay time) and one for amplitude (volume). This dial sets the oscillation rate for both modulators simultaneously. Lower settings sweep the pitch and amplitude modulators slowly; higher settings sweep it more rapidly. Settings between 0.25 and 1.0 provide typical "thickening" chorus effects; higher settings impart more of a vibrato and/or tremolo type of effect to the output, depending on the depth of the pitch and amplitude modulators.

Range:	0.05Hz - 40Hz
Increment:	0.05Hz
Default:	0.75Hz

Pitch Modulation Depth

This dial modulates the delay time of the chorus effect. Modulation of the delay time is what varies the pitch in the delayed signal. Low settings produce a subtle thickening of the sound; higher values impart a "watery" feel to the track. The total range of the pitch modulator is relatively narrow, about +/-2 to 3 semitones from the input signal.

Range:	0% - 100%
Increment:	1%
Default:	28%

Amplitude Modulation Depth

Amplitude modulation can be applied to the delayed signal in addition to pitch modulation. This modulator decreases the volume as the modulation wave approaches baseline. This is a relatively important parameter for effective chorusing, as it helps prevent phase cancellation ("wow") effects produced by chorus algorithms which do not modulate amplitude.

Range:	0% - 100%
Increment:	1%
Default:	22%

Modulation Delay Time

This sets the delay time for the "wet" signal. While the range is exceptionally wide for a chorus effect, effective delay times for chorus effects range from 3ms to 35ms. Delays above 15ms tend to provide a more pleasing chorus effect than shorter delay times, but at the expense of a risk of audible delay (a short slapback effect) in the combined wet/dry signal.

Range:	0ms - 500ms
Increment:	0.1ms
Default:	4.7ms

Wave Type (not remote-controllable)

There are two wave types which can be applied to the modulation oscillator: sine and triangle. Typical chorus effects apply a triangle wave to the modulated delay; sine waves tend to impart a more natural sound to the chorus effect, especially when used with a judicious amount of amplitude modulation.

The triangle wave option in this parameter might not sound like you would expect it to sound. In virtually all Yamaha chorus, flange and symphonic effects since the venerable SPX-90, the triangle wave sounds more like a square wave, not a triangle wave. There is no audible rise in the modulated signal from baseline to peak pitch and vice versa in the wet signal; instead, there is a very clear and *harsh* oscillation between peak and minimum pitch when triangle wave is specified. The harshness of this oscillation is plainly audible on solo tracks with wet/dry signal at the same level and more than 30% pitch modulation.

Options:	Sine, Triangle
Default:	Sine

Flanger

There's not much more you could ask for from a flange effect, except perhaps built in triggered oscillation for beat-matched flanging. This effect should meet all but the most specialized requirements for a flanger.

Frequency

Sets the oscillation rate for the modulator. Low settings sweep the pitch modulator slowly; higher settings sweep it more rapidly. (The pitch modulator actually oscillates the delay time; pitch modulation is the audible effect of delay time oscillation.) Settings below 1Hz are the norm for flanging (note the default value), but actual implementations vary widely.

Range:	0.05Hz - 40Hz
Increment:	0.05Hz
Default:	0.25Hz

Modulation Depth

Specifies how much the modulator will vary the pitch (delay time) of the input signal. Low values produce subtle pitch variations. "Static" flanging, which applies no pitch modulation to the effected signal, is generally not as pleasing an effect as flanging applied with a small amount of modulation. Higher values impart a swooshing sound. The total range of the pitch modulator is +/-2 to 3 semitones from the input signal.

The depth, or "sweep range", might seem much wider than five or six semitones because feedback of short delays results in a perceived sense of radical variation in the resonant peaks of some frequency ranges. You'll only hear the true range of the modulator by setting feedback gain to **0.**

Range:	0% - 100%
Increment:	1%
Default:	60%

Feedback Gain

The delayed signal can optionally be fed back into the delay line to generate additional echoes. Feedback gain reduces the volume of the output signal prior to each pass through the delay, so that each echo has the specified percentage of volume of the preceding echo. See the **Mono Delay** section for additional information.

When set to 0%, only one delay will be produced. This results in a very subtle flanging effect. Values higher or lower than 0% produce progressively more echoes as the input signal drops in volume more slowly with each repeat. High values are normal for flange effects.

Values greater than 0% produce echoes in phase with the input signal. Values less than 0% produce echoes in inverse phase with the input. Inverse phase settings produce a phase cancellation effect which gives the perception of narrowing the frequency range (filtering) of the input signal.

Range:	-99% - +99%
Increment:	1%
Default:	-83%

Modulation Delay Time

This is the delay time between feedback echoes. The range is exceptionally wide for a flange effect. Delay times for flange effects normally range from 1ms to 15ms. Delays above 15ms tend to sound more like thick chorus or rotating-speaker effects; delay times below 1ms are in the range of phase shifters. In practical terms, the delay time dictates the resonant peak in the flanged sound.

The higher the delay time, the lower the frequency of the resonant peak generated by feedback delays.

Range:	0ms - 500ms
Increment:	0.1ms
Default:	7.5ms

Wave Type (not remote-controllable)

There are two wave types which can be applied to the modulation oscillator: sine and triangle. Typical flange effects apply a triangle wave to the modulated delay. Sine waves tend to impart a more natural sound to the flange effect, especially when used with a judicious amount of amplitude modulation. “Natural-sounding” flange may defeat the purpose of flanging, however, because flanging is normally used to create a decidedly artificial metallic tone. See the **Chorus** section for additional information.

Options:	Sine, Triangle
Default:	Sine

Symphonic

Think of symphonic as “chorus with mayonnaise.” In addition to delay, this effect adds a fixed detune (a slight downward pitch shift) to the input to further thicken the sound. Many veteran users of Yamaha effects find the detuning gives symphonic a more pleasing character than Yamaha’s usual chorus algorithms. This effect includes a feedback control for multiplying the number of “voices” in the “symphony”, but this means that there is no precise way to fix the number of repeats, except by turning off feedback gain which limits repeats to one.

Frequency

Sets the oscillation rate for the modulator. Low settings sweep the pitch modulator slowly; higher settings sweep it more rapidly. (The pitch modulator actually oscillates the delay time; pitch modulation is the audible effect of delay time oscillation.) Settings between 0.25 and 1.0 provide typical “thickening” chorus-type effects; higher settings impart vibrato effects depending on modulation depth.

Range: 0.05Hz - 40Hz
Increment: 0.05Hz
Default: 0.75Hz

Depth

Sets the amount of pitch variation in the delayed signal. Low settings produce a subtle thickening of the sound; higher values are often used to impart a "watery" feel to the track. The total range of the pitch modulator is between +/-2 and +/-3 semitones from the input signal.

Range: 0% - 100%
Increment: 1%
Default: 28%

Modulation Delay Time

This is the delay time for the "wet" signal. While the range is exceptionally wide for this type of effect, normal delay times range from 3ms to 35ms. Delays above 35ms impart an audible short echo (slapback) to the combined wet/dry signal.

Range: 0ms - 500ms
Increment: 0.1ms
Default: 4.7ms

Wave Type (not remote-controllable)

There are two wave types which can be applied to the modulation oscillator: sine and triangle. Typical chorus/thickening effects apply a triangle wave to the modulated delay; sine waves tend to impart a more natural sound to the effect, especially when used with a judicious amount of amplitude modulation. While this produces a harsh effect with chorus effects, application of the "triangle" wave in symphonic effects is far less irritating (see the note in the chorusing topic).

Options: Sine, Triangle
Default: Sine

Phaser

Until the late 1970s, phase shifting was associated almost exclusively with "psychedelic swoosh" and "watery" effects and was considered little more than a gimmick. Its use by Ted Templeman and Eddie Van Halen throughout the first Van Halen album gave it a new mainstream respectability, and this "wind effect" has been widely duplicated since then, especially in hard rock. (Billy Dufty's use of this effect in earlier Cult recordings is one of the better-known "copycat" examples.)

The 01V's phase shifter is remarkably versatile and sweet-sounding, capable of reasonably accurate emulations of virtually all classic pedal-type phasers. Only the most sophisticate rack units and dedicated phase shifter plugins can outperform this effect.

The secret to emulating a favorite pedal phaser with this effect lies in patient trial-and-error with particular attention to the **STAGE** dial. For example, the MXR Phase 90 (the "Van Halen phaser") is a four-stage phase shifter; the Phase 100 allows selection of four, six or eight stages. Single-control phase shifters generally derive their "unique" tone from an offset trimpot which is often accessible only by removing the case. Fine-tuning of the **OFFSET** dial can usually approximate the sweep range of a given pedal; as long as the stage multiplier is set properly it's simply a matter of adjusting feedback gain and the emulation should be quite accurate.

Perhaps the best way to audition the wide range of sonic possibilities afforded by the 01V's phaser is to apply the effect to a pink or white noise signal. The use of a broadband reference signal allows you to hear precisely how the various parameters affect the harmonics and resonance of the input signal. If you prefer something more musical, overdriven guitar chords should be harmonically rich enough to show off the phaser's potential. (Note, however, that the "Van Halen phaser" effect was achieved by running the phaser pedal *before* the amp.)

Frequency

Sets the oscillation rate for the modulator. Low settings sweep the pitch modulator slowly; higher settings sweep it more rapidly. (The pitch modulator actually oscillates the delay time; pitch modulation is the audible effect of delay time oscillation.) Typical "psychedelic" effects tend to sweep at a rate near the default. Typical heavy metal guitar phasing effects tend to sweep at the slowest possible rate.

Range: 0.05Hz - 40Hz
Increment: 0.05Hz
Default: 0.35Hz

Depth

Sets the amount of pitch variation in the delayed signal. (The modulator actually oscillates the delay time; pitch modulation is the audible effect of delay time oscillation.) Due to the way phase shift effects behave, this parameter doesn't necessarily correspond with what you hear. Because the extremely short delay times used in phasers tend to create resonant peaks in the effected signal, the relatively small range of this parameter actually translates into a sweep of the resonant peak well beyond the audible range when high values are used. Values under 10% correspond with "vintage" phase shifting effects.

The depth control only affects the upper limit of the frequency shift in the resonant band. This is a result of the way phase shift effects modify the resonance of the input signal. As you increase the depth using a high amount of feedback gain, you'll notice that the lowest resonant frequency doesn't change; only the upper resonant limit changes.

Range: 0% - 100%
Increment: 1%
Default: 28%

Feedback Gain

The feedback gain dial in the 01V effector corresponds to the **Regeneration** dial found on many phaser pedals. The higher the feedback gain, the more pronounced the phase shift effect. Values in the range of 30-70% (plus or minus) are the norm. Values greater than 0% produce a wet signal in phase with the input signal; values less than 0% produce a wet signal in inverse phase with the input.

Set this value near the upper limit if you want to accentuate the resonant peak generated by the effect. If you maximize feedback gain while setting up the effect, you should find it easier to determine the frequency range of the most resonant band produced by the phaser.

Range: -99% - +99%
Increment: 1%
Default: +78%

Offset

This dial defines the level of phase offset of the wet signal to the input. In terms of what's happening in the 01V, offset corresponds with delay time. (Delay times are rarely used as parameters for phase shifters. Standard phaser parameters were defined long before delay lines became popular.) In terms of what you hear, this corresponds with the location of the center of the phaser's most resonant frequency band. At the upper end of the available range, the most resonant center frequency is actually beyond audible limits. At the lower end, the resonant's range actually produces an interesting and musically useful effect that sounds like the pitch modulator folding back on itself.

Range: 0 - 100
Increment: 1
Default: 46

Stages

This parameter specifies the number of stages, or tapped delays, which will be drawn from the input signal. The more stages, the thicker and "breathier" the effect. Stages in a phase shifter correspond to the number of taps in a multitap delay.

Range: 2 - 16
Increment: 2
Default: 8

Auto Pan

This is a reasonably good auto-pan effect providing both one- and two-dimensional panning effects.

Frequency

Sets the modulator's oscillation rate. The modulator controls the rate at which the sound appears to sweep across the stereo field. Low settings sweep the modulator slowly; higher settings sweep it more rapidly.

Range: 0.05Hz - 40Hz
Increment: 0.05Hz
Default: 0.35Hz

Depth

In terms of the panning effect, "depth" may be an unfortunate choice of words. "Width" may be more appropriate, because this parameter controls the width of the stereo field in which the sound source is panned. At 100%, the sound source pans from 9:00 to 3:00. Lower values produce a narrower stereo image.

Range: 0% - 100%
Increment: 1%
Default: 100%

Direction (not remote-controllable)

Five options are available with this parameter. **L<->R** produces the same effect as twisting a pan pot back and forth. The **L->R/R->L** options sweep the sound from one side of the stereo field to the other. At higher depth settings, there is a noticeable gap between cycles and an audible attack slope when using the triangle and sine wave types. The two **Turn** options create an illusion of the sound source travelling in a circle in front of the listener.

Options: L<->R, L->R, L<-R, Turn Left, Turn Right
Default: L<->R

Wave Type (not remote-controllable)

Three options are available here: sine, triangle and square wave oscillation. Sine wave oscillation approximates the effect of a sound source passing fairly close to you. The triangle wave approximates the effect of a sound source travelling in an arc from the left side of the field to the right. Square wave oscillation "ping-pongs" the sound from one side of the field to the other.

Options: Sine, Triangle, Square
Default: Sine

Tremolo

This is basic, no-frills tremolo effect. It cannot be synchronized with a vibrato effect and cannot add vibrato; it's a simple one-trick pony. Synchronized tremolo/vibrato, or vibrato alone, can be simulated using a pure wet output signal from the **Chorus** effect with no feedback gain and the lowest possible delay time.

Frequency

Sets the oscillation rate for the modulator. Low settings sweep the amplitude (volume) of the signal slowly; higher settings sweep it more rapidly.

Range: 0.05Hz - 40Hz
Increment: 0.05Hz
Default: 4.05Hz

Depth

Sets the amount of amplitude variation in the effected signal. At 100% the tremolo envelope sweeps from -Infinity (no audible output) to unity gain.

Range: 0% - 100%
Increment: 1%
Default: 91%

Wave Type (not remote-controllable)

Three options are available here: sine, triangle and square wave oscillation. Sine wave oscillation approximates the tremolo of a human voice or a rotating sound source. Triangle wave oscillation imparts a more machine-like oscillation to the effect. Square wave oscillation produces instantaneous volume changes as the oscillator cycles.

Options: Sine, Triangle, Square
Default: Triangle

High Quality Pitch Shift

This is a real-time pitch-shift effect which maintains the tempo of the input signal while increasing or decreasing its pitch. It produces only one pitch-shifted output, but boasts less of the graininess and instability of the **Dual Pitch Shift** effect since the 01V's effects chip can be devoted entirely to producing a single pitch-shifted signal.

This effect can be controlled in real time using automation (e.g. to pitch-correct a wobbly vocal track or produce a true harmony rather than just a pitch-shifted output) but as noted in the harmonizers backgrounder, a device or plug-in designed solely for this purpose might be required to achieve optimum output quality with the least amount of effort.

Pitch

This dial is the coarse adjustment for the pitch shifted output.

Range: -12 - +12
Increment: 1 semitone
Default: -2

Fine Adjust

This dial permits finer adjustment of pitch shifted output, useful for retuning slightly out-of-tune tracks or sections, or for “detune” effects.

Range: -50 - +50
Increment: 1 cent (1/50 semitone)
Default: +14

Delay Time

Specifies the delay to apply to the pitch-shifted signal.

While a delay time of **0** is selectable, a slight latency delay will occur between the input of the dry signal and the output of the pitch-shifted signal. This delay is inevitable due to the complexity of pitch shift computations. The documentation is unclear about the delay produced by the effect; if the processor is efficient enough, the delay may be as low as 1 sample (less than 3ms) at 44.1kHz input sampling rate. The higher the precision specified on the **MODE** dial, the longer the delay will likely be, although this is not confirmed by the documentation.

Range: 0 - 1000ms
Increment: 0.1ms
Default: 400.0ms

Feedback Gain

This control has an identical effect to feedback gain in a standard delay. By adding feedback to the pitch-shifted signal, you can achieve a “trailing” effect in which the signal’s pitch either rises or falls away as the echoes fade. See the **Mono Delay** section for additional information.

Range: -99% - +99%
Increment: 1%
Default: +54%

Mode (not remote-controllable)

This specifies the precision of the pitch shift effect. Higher values offer more precision. The effect is relatively subtle on most tracks, however higher mode values produce a small amount of latency (delay) in the pitch-shifted signal. Low values are appropriate for pre-mastering applications; maximum values should probably be used when applying pitch shift effects during mastering or rendering.

Options: Modes 1 through 10
Default: Mode 7

Dual Pitch Shift

The dual pitch shift effect apparently divides the effects chip’s horsepower by four in order to produce two pitch-shifted outputs rather than just one, and to allow both effects units to use this effect. Naturally the resulting output won’t be as accurate as that produced by the **High Quality Pitch Shift.** Generally speaking this effect should only be applied to monitor mixes or scratch tracks. The **High Quality Pitch Shift** can (and probably should) be applied twice and the output rendered to disk as a separate track when working with mission-critical tracks that require dual pitch-shifting.

There is one possible application of this effect that shouldn't leave you feeling cheated if you can't or won't use the HQ pitch shifter: detuning. Thickening a sound by adding two slightly detuned signals to the original will not result in significant loss of signal quality, because time and timbre alignment of slightly detuned signals can be done with a lot more accuracy using a lot less DSP horsepower. This makes the **Dual Pitch Shift** a useful alternative to **Symphonic** in applications where detuning is needed but modulation and/or feedback are undesirable.

Pitch 1/Pitch 2

These are the coarse adjustments for the pitch shifted outputs for "harmonizer" effects. Each output can be adjusted independently.

Range:		-12 - +12
Increment:		1 semitone
Default:	P/S 1:	-12
	P/S 2:	+7

Fine Adjust 1/Fine Adjust 2

These dials permit fine adjustment of pitch shifted output.

Range:		-50 - +50
Increment:		1 cent (1/50 semitone)
Default:	P/S 1:	-13
	P/S 2:	-3

Pan 1/Pan 2

As the name suggests, these parameter allows you to define a position in the stereo field for each of the two pitch-shifted signals. The panning increments are identical to the panning increments for the main channel pan.

Want to create a single harmony of your input signal and give that harmony a stereo effect? Pan each output to an appropriate position, use identical coarse adjustments, detune the outputs by a few cents, and add short delays. The right output is normally detuned a few cents more than the left and also delayed ten to twenty milliseconds more than the left, and if it's done correctly, you should hear a very clear stereo harmony effect that still sounds thick and rich when played back in mono.

Range:		Left 16 - Right 16
Increment:		1 (one increment of panning is equal to 5.625 degrees in the stereo field)
Default:	P/S 1:	Left 16
	P/S 2:	Right 16

Delay Time 1/Delay Time 2

Specifies the delay times to apply to the pitch-shifted signals. Each pitch-shifted signal can have a separate delay time. See **High Quality Pitch Shift** for more information.

Range:		0 - 1000ms
Increment:		0.1ms
Default:	P/S 1:	60.0ms
	P/S 2:	30.0ms

Feedback Gain 1/Feedback Gain 2

These controls have an identical effect to feedback gain in a standard delay. By adding feedback to the pitch-shifted signals, you can achieve a "trailing" effect in which the signal's pitch either rises or falls away as the echoes fade. Since this effect is used primarily for harmonizer-type effects, feedback is rarely used with it since it has a non-musical impact on the signal. See the **Mono Delay** section for additional information.

Range:		-99% - +99%
Increment:		1%
Default:	P/S 1:	0%
	P/S 2:	0%

Level 1/Level 2

When using this effect as a harmonizer, it's important to have independent level control over the two harmonies. Since the Aux faders control the main output of the effect, independent gain controls are required for each of the pitch-shifted signals. Negative values produce a pitch-shifted signal of inverse phase to the input.

Range:		-100 - +100
Increment:		1%
Default:	P/S 1:	+100%
	P/S 2:	+100%

Mode (not remote-controllable)

This determines the precision of the pitch shift effect. See the **High Quality Pitch Shift** section for additional information.

Options:	Modes 1 through 10
Default:	Mode 6

Rotary Speaker Simulator

While it's not as close to a Leslie cabinet simulation as you might like (i.e. it won't turn a DX-7 organ patch into Gregg Allman or Jon Lord), it's a versatile and well-thought-out effect with creative applications well beyond "organ grinding". The effect allows three separate oscillation levels (**NONE, SLOW** and **FAST**), emulates the clipping of overdriven tube preamps used in classic rotary cabinets such as the Leslie models, and allows separate mixing of the emulated bass cabinet and "tweeter horn" outputs typical of two-way rotary cabinets.

Rotate

This parameter is actually a switch designed for real-time control. Its function is similar to the motor footswitch for a rotary speaker cabinet and either starts or stops the rotation effect.

Options:	Start/Stop
Default:	Start *(on)*

Speed

Better rotary speaker cabinets have two preset rotation speeds in addition to a basic on/off. Changing the speed generates a degree of drama when the emotional tone of the music changes. This is a switch designed for real-time control for switching between the high and low speed settings.

Options:	Fast/Slow
Default:	Fast

Drive

This dial controls the level of overdrive applied to the signal before it is input into the rotation algorithms. The drive type corresponds with the **Overdrive 1** algorithm in the **Distortion** effect, intended to mimic the clipping of a 12-AX7 preamp tube. This doesn't adequately mimic the tone of a late-60s rock organ, since that classic tone depended on pushing the *power* amp to its limits as well and possibly overdriving the organ's onboard preamp. The drive algorithm seems to mix well though with signals which are already overdriven at the effect's input.

Range:	0 - 100
Increment:	1
Default:	80

Acceleration

Rotary speaker cabinets do not just "drop into gear" at the preset high or low speed. It takes anywhere from a few hundred milliseconds to a few seconds for the motor to spin up to the preset speed. In actual rotating speakers, the spin-up rate depends upon the condition of and quality of the motor and the weight of the drivers, axles and mounts that need to be accelerated. This parameter can be set to a full range of acceleration rates, ranging from rapid speed change to a relatively gradual rise. Extremely slow spin-up and instant spin-up are not achievable; there will always be a noticeable spin-up period for any significant speed changes.

Range:	0 - 10
Increment:	1
Default:	5

Low Gain/High Gain

These controls mix the relative levels from the two "virtual drivers" in the effect. Better rotary cabinets contain not one but two rotating speakers: a bass driver and a horn tweeter, both mounted on the same axle assembly. In some high-end Leslie models, the two drivers can actually rotate at different speeds, but this cannot be duplicated with the 01V's effectors.

The apparent harsh resonance of the high end driver emulation in this effect is not unintentional. Some owners of Leslie cabinets find the tweeter so annoying, especially when the amp is driven to clipping, that they disconnect it completely. Others find it so appealing for guitar that they disconnect the *woofer,* mixing the Leslie horn with mic'd guitar amp output.

Range:		0 - 100
Increment:		1
Default:	Low:	100
	High:	95

Slow Speed/Fast Speed

These parameters govern the rotation speeds for slow and fast rotation of the emulated speakers. Ranges are identical for both speeds, so it is possible to have the slow setting faster than the fast setting.

Range:		0.05Hz - 10Hz
Increment:		0.05Hz
Default:	Slow:	0.60Hz
	Fast:	2.05Hz

Ring Modulator

This is a relatively basic ring or "balanced" modulator effect offering a choice of self-modulation and sine wave modulation.

Source

In order to generate ring modulation effects, a second signal is needed since the effect depends on calculating sum and difference between two signals. Either the input signal itself, or a sine wave oscillator, can be selected as the second signal. In "vintage" ring modulation effects such as those generated by analog synthesizers, a sine wave is used as the second frequency.

When the input signal is applied as the modulating signal (self-modulation), the effect is static. This means that no oscillation is possible and the remaining three controls are nonfunctional.

Options: Oscillator/Self
Default: Oscillator

Oscillator Frequency

This specifies the frequency of the sine wave oscillator when it is chosen as the modulation source. The frequency can be modulated in real time using C-Console's automation for some relatively dramatic effects.

Range: 0 - 5000Hz
Increment: .01Hz
Default: 905.5Hz

Frequency Modulator (LFO) Frequency

When the sine wave oscillator is used as the modulation source, pitch modulation (vibrato) can be applied to it in the same way that an LFO can be applied to the modulating oscillator in a synthesizer. This dial controls the speed of that oscillator

Range: 0 - 40Hz
Increment: .05Hz
Default: 1.80Hz

Frequency Modulator (LFO) Depth

The depth of the pitch modulator's oscillator is controlled using this dial. The range is the same as the modulator depth for delay effects, or approximately +/-2 to 3 semitones.

Range: 0 - 100%
Increment: 1%
Default: 60%

Modulated Filter

If you create popular dance music or commercial soundtrack, this might become one of the most-used effects in your toolbox. This algorithm produces an effect similar to that of the resonant filters found on synthesizers. The effect is somewhat similar to what you would hear if you set a parametric EQ to a high gain/medium-low Q setting and twisted the frequency knob back and forth. But instead of being limited to filtering an oscillator or sample, the filter can be applied to an entire track.

Since the modulator cannot be turned off in this effect, it is not suitable for adding "fixed wah" or resonant peaking effects. If you need this type of effect, try the **Dynamic Filter** instead.

Frequency

Sets the oscillation rate for the modulator. Low settings sweep the resonant peak slowly across the specified range; higher settings sweep it more rapidly.

Range: 0.05Hz - 40Hz
Increment: 0.05Hz
Default: 1.25Hz

Depth

This specifies the modulation depth, or the range of filter variation possible in the effected signal. The range is not as high as you might expect from a dedicated resonant filter plug-in or "vintage" analog synthesizer, but it should be sufficient for a fairly wide range of effects.

Range: 0 - 100%
Increment: 1%
Default: 60%

Type (not remote-controllable)

Specifies the type of filter to be applied to the signal: band pass, low pass or high pass. Changing the filter type doesn't significantly affect the sound of the resonant sweep. Instead it modifies the overall frequency range of the effected signal. This is particularly important to keep in mind since the effect is usually applied as a pure wet signal.

Options: LPF, HPF, BPF
Default: BPF (**b**and**p**ass **f**ilter)

Offset

This sets the low end of the frequency range where the filter sweep will start. Low values start the sweep at the lower end of the audible range; higher values start it at higher frequencies. Values higher than 50 tend to lie beyond the range of the fundamental of nearly all musical content, and are best suited for sweeping the upper end of the audible spectrum. Values near the upper limit are beyond the audible spectrum for most tracks.

Range: 0 - 100
Increment: 1
Default: 10

Resonance

This parameter controls the amount of gain applied to the resonant frequency band. Higher settings produce a more pronounced resonant peak in the effected signal.

Range: 0 - 20
Increment: 1
Default: 10

Phase

This defines the phase alignment of the resonant filter in relation to the input signal. Modifying the phase alignment produces an effect similar to mild phase shifting. At 180 degrees, phase cancellation makes the effect sound as though the sweep frequency has been doubled and the depth has been cut. The effect sounds less like a phase shifter on signals with relatively simple harmonic content than it does on more harmonically complex signals such as vocal tracks. This parameter lends itself well to use as a real-time expression control.

Range: 0 - 354.38 degrees
Increment: 5.625 degrees (equivalent to 1/32nd increments)
Default: 0.00 degrees

Level

Since resonant peaking changes the volume of the signal (generally increasing it), output level control is needed to insure the effect doesn't clip. This dial controls the output level of the effected signal.

Range:	0 - 100
Increment:	1
Default:	70

Distortion

This effect is the digital equivalent of a bank of guitar-type overdrive and distortion pedals. While these distortion algorithms won't duplicate the sound of a driven Marshall stack or Fender combo to the satisfaction of sophisticated ears, they're serviceable distortion/drive effects for techno/dance music, scratch tracking or notepadding, or for applying a second layer of drive or distortion to already-overdriven guitar tracks. (Chaining two or more overdrive/distortion devices is an old home recordists' trick for thickening up the tone of guitars when the available choice of amps is limited or unsuitable for a certain track.) It also includes a simple but useful noise gate, mirroring the noise gate feature found in Yamaha's better overdrive pedals.

Distortion Type (not remote-controllable)

Five distortion algorithms are available, each with its own particular character.

Distortion 1 and **Distortion 2** are the kind of hard, broadband distortion effects which you normally associate with heavy metal guitar. **Distortion 1** tends to produce a wider bandwidth than **Distortion 2.** It sounds a lot like Yamaha's own analog distortion pedals from the 1970s and 1980s and is probably best suited to non-guitar tracks. **Distortion 2** has resonance at the upper midrange of the frequency spectrum, similar to that produced by heavily overdriven guitar amplifiers, and sounds less "buzzy", somewhat reminiscent of Boss' famous Heavy Metal distortion box.

Overdrive 1 and **Overdrive 2** are similar to the pedal effects of standard overdrive pedals produced by Roland, Ibanez, Yamaha, et. al. since the late 1970s. Where the **Distortion** algorithms produce even harmonics, imparting a grating tone to the effect, the **Overdrive** algorithms tend to produce more of the odd harmonics associated with tube clipping. **Overdrive 1** is similar in tone to heavy overdrive pedals such as Yamaha's own drive pedals, and has the same "buzz" in the upper midrange as **Distortion 1.** It appears to be intended to mimic the tone of an overdriven preamp tube. **Overdrive 2** is a creamier, smoother tone reminiscent of the classic Boss OD1/SD1 or Ibanez Tube Screamer effects. Both overdrive algorithms have resonant peaks around 1kHz and the low-end rolloff typical of overdrive stomp boxes.

Crunch is more broad-band overdrive effect with a tone reminiscent of a guitar preamp using the typical 12-AX7 preamp tube. It doesn't seem to have the "body" of overdriven power amp tubes, but it has more low-end body than the overdrive algorithms. While it retains more of the input signal's original tonal range than the overdrive algorithms, it doesn't produce the harsh even harmonics of the distortion effects. This makes **Crunch** useful for driving both percussion and guitar tracks. Those looking for a relatively accurate tube drive reproduction won't find this especially exciting, but it sounds to us like the most versatile drive effect of the lot.

Both the **Distortion** and **Overdrive** algorithms apply heavy compression to the signal. **Crunch** doesn't apply nearly the degree of compression as the other four algorithms.

Our descriptions of these effects are derived from our experience with a fairly wide range of amps, pedals and rack devices. For Yamaha's own descriptions of the algorithms, see the 01V Owner's Manual.

Options:	Distortion 1, Distortion 2, Overdrive 1, Overdrive 2, Crunch
Default:	Distortion 1

Drive

This controls the amount of distortion or overdrive applied to the input signal, as well as how the distortion/overdrive effect responds to the input signal's dynamics.

The two **Distortion** algorithms produce little audible variation throughout the **Drive** range and don't respond well to dynamics in the input signal. This is typical of stand-alone distortion devices, so this should come as no surprise. The **Overdrive** algorithms, on the other hand, are fairly responsive to dynamics, giving this control similar behavior to the drive control on virtually every other manufacturer's analog overdrive pedals except (you guessed it) Yamaha's. The **Crunch** algorithm is the most dynamically sensitive and produces the widest variation in drive, with almost no clipping at all at the lower end of the range.

Range:	0 - 100
Increment:	1
Default:	90

Master Gain

Just as gain or level controls are required on overdrive/distortion stomp boxes, a gain control is needed here to insure that the clipping generated by the effect, which is desirable, doesn't produce a signal that clips the output, which is highly *un*desirable. **Overdrive** and **Crunch** algorithms in particular require reductions in gain as drive is increased.

Reducing this parameter will not reduce the amount of drive or distortion produced relative to the track's dynamics. Adjustment of drive response to dynamics must be done by adjusting the input gain from the instrument, the channel's input attentuator the channel's auxiliary send control.

Range:	0 - 100
Increment:	1
Default:	40

Tone

This is functionally equivalent to the tone control of a pedal-type distortion/overdrive. It appears to use the tone EQ type. High frequencies can either be boosted or cut. Highs are often cut when using the **Distortion** algorithms to reduce the harshness of the tone. The tone control occurs post-drive.

Range:	-10 - +10
Increment:	1
Default:	-5

Noise Gate

Analog-type distortion/overdrive effects usually require noise gating to prevent noise from filling the track when there is no musical content to be processed or when the dynamic range drops below a usable level. Guitars in particular require gating to prevent fretboard scratch, percussive sounds from the guitar body and low-level pickup hum from "leaking" into the recording due in part to the heavy compression applied by drive/distortion devices.

The 01V's digital algorithms are "clean", meaning that gating is not necessary to quiet the effect itself. However it *is* necessary with input tracks which are not gated and which may contain background noise, 60-cycle hum, or other artifacts, and especially useful since the dynamics processor for the input track will very often be used to compress the signal prior to overdriving it. The actual parameter value does not appear relative to the threshold levels used on dedicated noise gates, although it serves the same purpose. Higher values are only needed with especially "hot" input signals containing a great deal of ambient noise.

Range:	0 - 20
Increment:	1
Default:	2

Amp Simulator

This is much more than just an afterthought. Although it doesn't have the same accuracy and character of dedicated units such as Line 6's POD or better amp simulator plug-ins, this is a serviceable amp emulator capable of producing usable reproductions of mic'd amplifier tones. It won't fool trained ears into thinking you used a Marshall, Fender or Mesa Boogie on the track, but it is definitely an improvement over direct injection and easily the equal of many name-brand pedal units.

This effect doesn't have to be limited to the guitars, basses and "classic keyboards" either. It can add a unique treatment to virtually any instrument, and is probably the quickest, easiest and most sonically pleasing way to push an overly-dominant instrument back in the mix. In testing with another "amp simulator" (the Ibanez VA3 Virtual Amplifier), we found that using one amp sim on the VA3 and combining it with another algorithm on the 01V produced several very sweet effects covering a relatively wide tonal range.

When recording guitar or bass tracks the use of a direct box, mixer pream or some other preamp device is recommended prior to feeding the instrument into the 01V's input. Even a guitar pedal with the effect turned off will do the job in a pinch.

Amp type

This dial defines the base amplifier type you wish to emulate. Each algorithm attempts to emulate the tonal colorations and phase cancellation effects of that particular type of amplifier. The five **Combo** settings appear to use little more than a phase notch swept over the lower midrange as a means of emulating amps with various speaker/cabinet configurations.

Stack M1, Stack M2: By the initials, you'd guess that these algorithms are intended to emulate the sound of a Marshall stack, and they're reasonably effective for that purpose, provided you don't have the Real Thing handy or a dedicated amp emulation device. The first algorithm appears to emulate a single-cabinet Marshall stack with close mic'ing. The second algorithm includes phase cancellation effects and sounds similar to angled (off-axis) mic'ing of a dual Marshall stack.

Stack M2 is the default.

Thrash: This might be a misnomer. The mid-cut effect in this algorithm reminds one of the HiWatt tone heard on early Black Sabbath and Who recordings. If it truly was based on the HiWatt, this is the only HiWatt emulation we could find in *any* device on the market at the time of publication.

Mid Boost: This sounds much like the standard flat tone of an inexpensive solid-state combo guitar amp such as a Roland Cube, Fender Squier or Stage series amplifier. As you dial in more drive, the algorithm begins to take on Vox- or Fender-like characteristics.

Combo PG, Combo VR, Combo DX, Combo TW The depth of the bottom end on the Combo PG algorithm is reminiscent of California "boutique" amps, ultra-heavy combos such as Soldanos, Mesa Boogies or Carvins. The DX algorithm is reminiscent of the Roland Jazz Chorus.

Mini: The frequency response of this algorithm is reminiscent of close-mic'ing a Fender Champ-type amplifier.

Flat: This tends to sound more like a solid-state guitar rig or a simple 12-AX7-based tube preamp rather than a particular amplifier.

Distortion Type (not remote-controllable)

Five distortion algorithms are available, each with its own distinct character. See **Distortion** for more information.

Options: Distortion 1, Distortion 2, Overdrive 1, Overdrive 2, Crunch
Default: Distortion 1

Noise Gate

Analog-type distortion/overdrive effects usually require noise gating to prevent noise from filling the track when there is no musical content to be processed or when the dynamic range drops below a usable level. See **Distortion** for more information.

Range: 0 - 20
Increment: 1
Default: 2

Drive

This controls the amount of distortion/overdrive applied to the input signal. See **Distortion** for more information.

Range: 0 - 100
Increment: 1
Default: 90

Master Gain

This controls the output gain of the effect. See **Distortion** for more information.

Range: 0 - 100
Increment: 1
Default: 60

Cabinet Depth

This is not a measure of the physical depth of the cabinet being emulated. Instead, it defines the degree of “cabinet effect” applied to the overall sound. Close-mic’ing a guitar amp produces relatively little cabinet effect since what is being recorded consists primarily of the speaker’s output. As the mic is drawn back from the speaker cone, or moved away from it toward a neutral position facing the cabinet, more and more cabinet effect is heard. At the upper end, you’ll notice a considerable amount of resonance in the tone typical of what you hear standing a few feet from a guitar or bass amplifier.

Circuitry defines harmonic content, but much of the tonal coloration that gives guitar tracks their distinctiveness is derived from the effects of the mic, driver, and design of the cabinet. For example, both ported and unported closed cabinets will actually affect the movement of the speaker since the sound reflecting inside the cabinet applies a degree pressure to the cone. Cabinet *type* is determined by the amp simulation selected; there is no separate control for matching a cab type to an amp type.

Range: 0 - 100
Increment: 1
Default: 40

Bass, Middle, Treble

These controls appear to have no parallels in the 01V’s EQs. They behave like the passive tone controls on most tube guitar amps. All they do is *cut* the corresponding frequency range; they cannot boost. The response slopes sound similar to those of the 01V’s tone EQ type.

Range:		0 - 100
Increment:		1
Default:	Bass:	95
	Middle:	100
	Treble:	91

EQ Gain, Frequency, Q

In addition to the three tone controls described above, this effect provides a band of parametric or **Peaking** equalization for additional tone control.

The default settings appear to have been chosen to mirror the usual 1kHz parametric EQ that engineers so often apply to guitars and electric pianos.

Dynamic Filter

Although the parameters are similar, in practice this effect has relatively little in common with the **Modulated Filter** effect and is usually used in very different applications. It has a sound similar to "auto-wah/auto-filter/envelope filter" pedals such as the Ibanez AutoFilter/AutoWah, Boss Touch-Wah or MuTron III. This effect is also referred to as an envelope filter or envelope follower.

The dynamic filter is designed to be a "wah-wah"-type sound in which the depth of the wah effect is dependent on the dynamics of the input signal. This is a dramatic effect with its own unique place in a musician's sonic arsenal; it won't normally replace or improve upon any other specific effect.

But that's by no means the limit of its talents. It can also serve as a fixed-frequency resonant filter of the type found on analog synthesizers or as a "fixed wah". Many classic rock tracks use a fixed wah effect (wah pedal set to an unchanging position) to create a thin, almost vocal tone with guitar tracks. This effect was normally achieved by running the guitar through a wah-wah pedal and not using the pedal to vary the resonant frequency. To emulate this effect, set **SENSE** to **0**, select **LPF** as the filter **TYPE,** and vary the **OFFSET** to the desired resonant frequency. The resulting effect very closely mirrors "fixed" or "standing wah" effects. This effect is also used on percussion tracks and "found sound" samples in dance/techno music and was popular for a time in jazz/rock fusion.

Among the better-known applications of dynamic filtering in pop music are the clavinet track in Stevie Wonder's "Higher Ground" and several guitar tracks from Jeff Beck's collaborations with Jan Hammer.

If you have used simple dynamic filters in the past such as the Boss Touch Wah or Ibanez AutoWah, you'll find this to be a dramatically more versatile implementation. If you're a MuTron III or AutoFilter fan, you'll find this effect lacks a touch of the warmth of these pedals, but hey, for noise-free operation, it's worth the trade-off. The high pass filter option in particular opens up sonic possibilities not possible with the Touch Wah.

Sense

This control determines how sensitively the filter will react to the input signal. The lower the sensitivity, the less effect produced from a given change in volume. Ideally, input gain should be adjusted from the channel's auxiliary send or input attenuator prior to setting this control.

Range:	0 - 100
Increment:	1
Default:	60

Filter Type (not remote-controllable)

This control defines the type of filtering to be applied to the input signal. For traditional "auto-wah" effects, the BPF (bandpass filter) is used. This emulates the low frequency cut effect of an older or inexpensive wah-wah pedal. However, in most cases the LPF (low pass filter) option will produce a much more pleasing effect since it doesn't chop off all the low end. LPF has virtually the same resonant frequency range as the BPF. HPF implementations are probably most useful for mixing effect with the dry signal, since it dramatically cuts low and midrange frequencies.

Options:	LPF, BPF, HPF
Default:	BPF (bandpass filter)

Offset

This dial sets the frequency of the filter's resonant peak. (Keep in mind that the resonant peak shifts upward as input gain increases.) Lower values produce resonants at the lower end of the audible spectrum; higher values sweep the resonant to high frequencies. This control does not specify the frequency of the resonant, so its frequency range must be discovered through experimentation.

Range:	0 - 100
Increment:	1
Default:	9

Resonance

This controls the amount of gain applied to the resonant frequency. Resonance remains constant across the range of the filter sweep; it does not increase in intensity as the resonant frequency rises.

Range:	0 - 20
Increment:	1
Default:	13

Decay time (not remote-controllable)

Once the input signal falls from its loudness peak, the frequency of the resonant peak decreases (**DIR.** set to **UP**) or increases (**DOWN**) as the input signal decays. Values lower than 250ms generally correspond with decay characteristics of "vintage" envelope filter effects. Higher values are more suited to sparser tracks with fewer notes; lower values may be more appropriate for allegro tempos and percussion tracks.

This parameter has an exceptionally wide range for effects of this type. At the upper end of the scale, the resonant sweep can be applied as a "one-shot" effect, allowing the sweep to be heard with the very first attack of the track. Unless there are huge gaps in the track, the remainder of the track will have the same fixed resonant filtering.

Range:	6ms - 46s
Increment:	variable throughout the range
Default:	145ms

Direction

When this dial is set to **UP,** the frequency of the effected signal's resonant peak rises as the effect processes the attack portion of the envelope. When set to **DOWN,** the resonant frequency falls as attack is processed. In terms of what you'd hear from a wah pedal, the **UP** sound corresponds to a rapid "toeing", while **DOWN** corresponds with rapid "heeling".

Options:	Up, Down
Default:	Up

Level

This controls the output gain of the effect. Output gain control is important since this effect can dramatically alter the total volume of the input signal.

Range: 0 - 100
Increment: 1
Default: 90

Dynamic Flanger

This is a standard flanging effect whose delay time parameter sweeps with the dynamics of the input signal, resulting in a resonant peak that sweeps in a similar fashion to the **Dynamic Filter.** The flanger has far less resonance than the filter and retains the metallic flavor typical of flange effects. Feedback gain is not dynamically controllable, meaning that below the sensitivity threshold the effect takes on the character of a “static flange” or short feedback delay.

The effect itself is difficult to control with input signals from delicate analog sources such as guitars, basses or vocals, especially when the input track’s notes do not occur at regular intervals (e.g. every quarter-note).

Sense

This control determines how sensitively the modulator reacts to the input signal. The lower the sensitivity, the less effect produced from a given change in volume. Ideally, input gain should be adjusted from the channel’s auxiliary send or input attenuator prior to setting this control.

Range: 0 - 100
Increment: 1
Default: 59

Feedback Gain

Flanging as it is normally applied in pop music depends a great deal upon feedback to impart the signature “metallic” sound to the signal. When feedback is applied, the delayed signal is fed back into the delay as a percentage of the volume of the last delay output. This specifies the percentage of original volume that the signal should retain with each successive pass through the delay. In terms of what you hear, this dial controls the degree of perceived resonance in the effected signal, and thus plays a major role in determining the depth of the dynamic effect.

When this dial is set to 0%, only one delay will be produced, resulting in a very subtle flanging effect. Values higher or lower than 0% produce progressively more echoes as the input signal drops in volume with each repeat. High values are normal for flange effects.

Values greater than 0% produce echoes in phase with the input signal; values less than 0% produce echoes in inverse phase with the input. Inverse phase settings produce a phase cancellation effect which gives the perception of narrowing the frequency range of the effected signal.

Range: -99% - +99%
Increment: 1%
Default: -88%

Offset

This dial sets the flanger’s highest resonant peak in the output signal’s frequency range. Lower values produce resonants at the lower end of the audio spectrum at maximum input gain; higher values sweep the resonant to the upper range of the spectrum.

OFFSET is a misnomer for this control, since this parameter actually represents the initial delay time of the flanger. Lower values represent *higher* delay times; higher values represent shorter delays.

Range: 0 - 100
Increment: 1
Default: 67

Direction

When this dial is set to **UP,** the frequency of the effected signal's resonant band rises as the effect processes the attack portion of the envelope. When set to **DOWN,** the resonant band's frequency falls as the input signal's attack is processed and rises again as it decays. Technically this parameter determines whether delay time rises (**DOWN**) or falls (**UP**) as the input signal's amplitude rises.

Options:	Up, Down
Default:	Up

Hold

Once the maximum sweep value has been reached, the effect will optionally "hold" the delay time (or freeze the center frequency of the resonant band if you prefer) for a specified period of time.

The perceived stability of the effect can be improved by boosting this value to the 75ms - 300ms range depending on the requirements of the input signal. Lower values tend to increase the "wobbliness" of the effect when applied to input signals with delicate dynamics.

Range:	0.02ms - 2.13s
Increment:	variable throughout the range
Default:	0.79ms

Decay time

Once the input signal's loudness peak has passed, the frequency of the resonant peak, or the delay time if you prefer, returns to baseline as the input signal decays. When **DIR.** Is set to **UP,** the resonant band's frequency drops; set to **DOWN** it rises. See **Dynamic Filter** for additional information.

Range:	6ms - 46s
Increment:	variable throughout the range
Default:	46ms

Dynamic Phaser

This is actually a more interesting effect than you might think. On the surface it looks like little more than a phase shifter variant of the **Dynamic Flanger**, but there's a little surprise lurking under the hood. The nature of the phase shift effect, when applied dynamically with the right tweaking of parameters, imparts more of a formant-like resonance to the effected signal than either the **Dynamic Flanger** or **Dynamic Filter**. The result is an effect similar to a "talk box". (Well-known examples of talk box effects include the lead guitar lines on Peter Frampton's "Do You Feel Like I Do" and Joe Walsh's "Rocky Mountain Way".) While unconfirmed, we believe that Don "Buck Dharma" Roeser used a dynamic phase shift effect on Blue Öyster Cult's semi-classic "E.T.I. (Extra Terrestrial Intelligence)" to achieve its unique guitar tone.

Unfortunately the effect is just as difficult to control as the **Dynamic Flanger** with delicate analog input signals such as guitars or vocals. Narrowing the dynamic range with compression or limiting should help bring the effect under control for many instruments.

The effect may be too "twitchy" for guitars and vocals, but it is certainly stable enough to deal with input signals such as sampled percussion and synthesizers which have consistent levels. This effect, when applied to an analog synth pad, imparts an unusual vowel-like quality to it. The resonant band can be "fixed" (prevented from sweeping) by setting the **SENSE** parameter to 0.

Sense

This control determines how sensitively the modulator reacts to the input signal. The lower the sensitivity, the less effect produced from a given change in volume. Ideally, input gain should be adjusted from the channel's auxiliary send or input attenuator prior to setting this control.

Range: 0 - 100
Increment: 1
Default: 59

Feedback Gain

The feedback gain setting in the 01V effector normally corresponds to the depth dial on pedal-type phasers. The higher the feedback gain, the thicker the phase shift effect. (See the **Phaser** section for additional information.)

Range: -99% - +99%
Increment: 1%
Default: -88%

Offset

This sets the level of phase offset of the wet signal to the input. In terms of dynamic phasing, this corresponds with the delay time at the beginning of the time sweep. When **DIR.** is set to **UP,** the delay time sweeps *down* from this level with the attack of the input signal; vice versa when **DIR.** is set to **DOWN.** (See the **Phaser** section for additional information.)

Range: 0 - 100
Increment: 1
Default: 29

Direction

This control defines how the resonant effect "travels". When set to **UP,** the frequency of the effected signal's resonant band rises as the effect processes the attack portion of the envelope and falls again as it decays. When set to **DOWN,** the resonant frequency falls as attack is processed and rises again during decay. Technically speaking, this parameter determines whether delay time rises (**DOWN**) or falls (**UP**) as the input signal's amplitude rises.

Options: Up, Down
Default: Up

Stages

Specifies the number of stages (delay taps) applied to the input signal. For more information, see the **Phaser** section.

Range: 2 - 16
Increment: 2
Default: 10

Hold (not remote-controllable)

Once the maximum sweep value has been reached, the effect will optionally "hold" the delay time (which translates into the resonant band) at that level for a certain period of time. (See the **Dynamic Flanger** section for additional information.)

Range: 0.02ms - 2.13s
Increment: variable throughout the range
Default: 0.09ms

Reverb+Chorus / Reverb>Chorus

These two effects combine digital reverberation with chorusing. The only difference between the two is routing. **Reverb+Chorus** applies chorusing to the input signal before it is sent to the reverb. **Reverb>Chorus** applies chorusing to the reverb itself. There appears to be little audible difference between the overall quality of this reverb and the main reverb algorithms but the reverb used here has fewer parameters than the dedicated algorithms. Note the lack of a reverb type selector.

Reverb Time (not remote-controllable)

This is the total time elapsed between the moment when a sample passes through the processor and the moment at which that sample's reverb level drops to zero.

Range:		0.3 - 99s
Increment:	0.3s - 5s:	0.1s
	5s - 10s:	0.5s
	10s - 20s:	1.0s
	20s-99s:	5.0s
Default:	Rev+Chorus	2.2s
	Rev>Chorus	1.8s

Initial Delay

This is the total elapsed time between the moment when the audio signal arrives at the processor and the first reflections are produced. (See the **Reverb** section for additional information.)

Range:		0 - 500ms
Increment:		0.1ms
Default:	Rev+Chorus	30ms
	Rev>Chorus	22ms

High Frequency Decay Ratio

This dial specifies how rapidly high-frequency content will be removed from the reverb signal in relation to total reverb time. See the **Reverb** section for additional information.

Range:		0.1 - 1.0
Increment:		1
Default:	Rev+Chorus	0.3
	Rev>Chorus	0.3

Diffusion

This parameter determines the perceived width of the reverb's stereo field. See the **Reverb** section for additional information.

Range:		1 - 10
Increment:		1
Default:	Rev+Chorus	10
	Rev>Chorus	10

Density

This dial defines the "richness" of the reverb. See the **Reverb** section for additional information.

Range:		0% - 100%
Increment:		1%
Default:	Rev+Chorus	90%
	Rev>Chorus	60%

High Pass Filter

The **HPF/LPF** dials act as "tone controls" for the reverb, but these parameters play a more significant role than the term "simple tone control" might imply. See the **Reverb** section for additional information.

Range:		Through - 8kHz
Increment:		1 semitone
Default:	Rev+Chorus	Through
	Rev>Chorus	Through

Low Pass Filter

See **High Pass Filter** above.

Range:		Through - 8kHz
Increment:		1 semitone
Default:	Rev+Chorus	9.51kHz
	Rev>Chorus	5.65kHz

Reverb/Chorus Balance/Reverb Balance

For **Reverb+Chorus,** this dial acts as a mixing control for the two effects. For **Reverb>Chorus,** it acts as a master gain for the combined effect.

Range:		0-100%
Increment:		1%
Default:	Rev+Chorus	30%
	Rev>Chorus	30%

Frequency

Sets the oscillation rate for the chorus modulator. See the **Chorus** section for additional information.

Range:		0.05Hz - 40Hz
Increment:		0.05Hz
Default:	Rev+Chorus	0.55Hz
	Rev>Chorus	1.95Hz

Pitch Modulation Depth

Sets the amount of pitch variation (delay time) in the chorus' delayed signal. See the **Chorus** section for additional information.

Range:		0% - 100%
Increment:		1%
Default:	Rev+Chorus	60%
	Rev>Chorus	43%

Amplitude Modulation Depth

In addition to pitch modulation, amplitude modulation can be applied to the delayed signal. See the **Chorus** section for additional information.

Range:		0% - 100%
Increment:		1%
Default:	Rev+Chorus	48%
	Rev>Chorus	34%

Modulation Delay Time

This is the delay time for the “wet” signal in the chorus effect. See the **Chorus** section for additional information. The modulator can be “frozen” by specifying a value of 0, but this changes the algorithm into Reverb+Delay or Reverb>Delay.

Range:		0ms - 500ms
Increment:		0.1ms
Default:	Rev+Chorus	3.0ms
	Rev>Chorus	7.0ms

Wave Type

There are two wave types which can be applied to the modulation oscillator: sine and triangle. See the **Chorus** section for additional information.

Options:		Sine, Triangle
Default:	Rev+Chorus	Sine
	Rev>Chorus	Triangle

Reverb+Flange / Reverb>Flange

These two effects combine digital reverberation with flanging. The only difference between the two is routing. **Reverb+Flange** applies flanging to the input signal before it is sent to the reverb. **Reverb>Flange** applies flanging to the reverb itself. There appears to be little audible difference between the overall quality of this reverb and the main reverb algorithms but the reverb used here has fewer parameters than the dedicated algorithms. Note the lack of a reverb type selector.

Reverb Time (not remote-controllable)

This is the total time elapsed between the moment when a sample passes through the processor and the moment at which that sample’s reverb level drops to zero.

Range:		0.3 - 99s
Increment:	0.3s - 5s:	0.1s
	5s - 10s:	0.5s
	10s - 20s:	1.0s
	20s-99s:	5.0s
Default:	Rev+Flange	1.6s
	Rev>Flange	2.7s

Initial Delay

This is the total elapsed time between the moment when the audio signal arrives at the processor and the first reflections are produced. (See the **Reverb** section for additional information.)

Range:		0ms - 500ms
Increment:		0.1ms
Default:	Rev+Flange	28ms
	Rev>Flange	15.1ms

High Frequency Decay Ratio

This dial specifies how rapidly high-frequency content will be removed from the reverb signal in relation to total reverb time. See the **Reverb** section for additional information.

Range:		0.1 - 1.0
Increment:		0.1
Default:	Rev+Flange	0.3
	Rev>Flange	0.5

Diffusion

This parameter determines the perceived width of the reverb's stereo field. See the **Reverb** section for additional information.

Range:		1 - 10
Increment:		1
Default:	Rev+Flange	10
	Rev>Flange	10

Density

This dial defines the "richness" of the reverb. See the **Reverb** section for additional information.

Range:		0% - 100%
Increment:		1%
Default:	Rev+Flange	95%
	Rev>Flange	80%

High Pass Filter

The **HPF/LPF** dials act as "tone controls" for the reverb, but these parameters play a more significant role than the term "simple tone control" might imply. See the **Reverb** section for additional information.

Range:		Through - 8kHz
Increment:		1 semitone
Default:	Rev+Flange	Through
	Rev>Flange	Through

Low Pass Filter

See above.

Range:		Through - 8kHz
Increment:		1 semitone
Default:	Rev+Flange	9.51kHz
	Rev>Flange	3.17kHz

Reverb/Flange Balance

For **Reverb+Flange,** this dial acts as a mixing control for the two effects. For **Reverb>Flange,** it acts as a master gain for the effect.

Range:		0% - 100%
Increment:		1%
Default:	Rev+Flange	24%
	Rev>Flange	16%

Frequency

Sets the oscillation rate for the flange modulator. Low settings sweep the pitch modulator (delay time) slowly; higher settings sweep it more rapidly. Settings below 1Hz are the norm for flanging (note the default value).

Range:		0.05Hz - 40Hz
Increment:		0.05Hz
Default:	Rev+Flange	0.75Hz
	Rev>Flange	1.95Hz

Modulation Depth

Specifies how widely the pitch modulator can vary the pitch (delay time) of the delayed signal. See the **Flanger** section for additional information.

Range:		0% - 100%
Increment:		1%
Default:	Rev+Flange	61%
	Rev>Flange	36%

Feedback Gain

This specifies the percentage of original volume the signal should retain with each successive repeat. See the **Flanger** section for additional information.

Range:		-99% - +99%
Increment:		1%
Default:	Rev+Flange	+82%
	Rev>Flange	+43%

Modulation Delay Time

This is the delay time between feedback echoes. See the **Flanger** section for additional information.

Range:		0ms - 500ms
Increment:		0.1ms
Default:	Rev+Flange	3.0ms
	Rev>Flange	7.0ms

Wave Type

There are two wave types which can be applied to the modulation oscillator: sine and triangle. See the **Flanger** and **Chorus** sections for additional information.

Options:		Sine, Triangle
Default:	Rev+Flange	Triangle
	Rev>Flange	Sine

Reverb+Symphonic / Reverb>Symphonic

These two effects combine digital reverberation with flanging. The only difference between the two is routing. **Reverb+Symphonic** applies symphonic thickening to the input signal before it is sent to the reverb. **Reverb>Symphonic** applies thickening to the reverb itself. There appears to be little audible difference between the overall quality of this reverb and the main reverb algorithms but the reverb used here has fewer parameters than the dedicated algorithms. Note the lack of a reverb type selector.

Simple thickening of the reverb becomes quite noticeable and free of "wobble" when the **DEPTH** dial is set to **0** to remove modulator effects. This turns the effect into a detune-thickened reverb.

Reverb Time (not remote-controllable)

This is the total time elapsed between the moment when a sample passes through the processor and the moment at which that sample's reverb level drops to zero.

Range:		0.3 - 99s
Increment:	0.3s - 5s:	0.1s
	5s - 10s:	0.5s
	10s - 20s:	1.0s
	20s-99s:	5.0s
Default:	Rev+Symph:	2.8s
	Rev>Symph:	2.6s

Initial Delay

This is the total elapsed time between the moment when the audio signal arrives at the processor and the first reflections are produced. (See the **Reverb** section for additional information.)

Range:		0ms - 500ms
Increment:		0.1ms
Default:	Rev+Symph:	19.0ms
	Rev>Symph:	12.0ms

High Frequency Decay Ratio

This dial specifies how rapidly high-frequency content will be removed from the reverb signal in relation to total reverb time. See the **Reverb** section for additional information.

Range:		0.1 - 1.0
Increment:		0.1
Default:	Rev+Symph:	0.3
	Rev>Symph:	0.3

Diffusion

This parameter determines the perceived width of the reverb's stereo field. See the **Reverb** section for additional information.

Range:		1 - 10
Increment:		1
Default:	Rev+Symph:	10
	Rev>Symph:	10

Density

This dial defines the “richness” of the reverb. See the **Reverb** section for additional information.

Range:		0% - 100%
Increment:		1%
Default:	Rev+Symph:	96%
	Rev>Symph:	100%

High Pass Filter

The **HPF/LPF** dials act as “tone controls” for the reverb, but these parameters play a more significant role than the term “simple tone control” might imply. See the **Reverb** section for additional information.

Range:		Through - 8kHz
Increment:		1 semitone
Default:	Rev+Symph:	Through
	Rev>Symph:	63Hz

Low Pass Filter

See above.

Range:		50Hz - Through
Increment:		1 semitone
Default:	Rev+Symph:	8.98kHz
	Rev>Symph:	10.00kHz

Reverb/Symphonic Balance

For **Reverb+Symphonic,** this dial acts as a mixing control for the two effects. For **Reverb>Symphonic,** it acts as a master gain for the effect.

Range:		0% - 100%
Increment:		1%
Default:	Rev+Symph:	53%
	Rev>Symph:	55%

Frequency

Sets the oscillation rate for the symphonic effect’s modulator. See the **Symphonic** section for additional information.

Range:		0.05Hz - 40.00Hz
Increment:		0.05Hz
Default:	Rev+Symph:	0.95Hz
	Rev>Symph:	1.45Hz

Depth

Sets the amount of pitch variation in the delayed signal in the symphonic effect. See the **Symphonic** section for additional information.

Range:		0% - 100%
Increment:		1%
Default:	Rev+Symph:	84%
	Rev>Symph:	67%

Modulation Delay Time

This sets the base delay time for the symphonic effect's modulator. See the **Symphonic** section for additional information.

Range:		0ms - 500ms
Increment:		0.1ms
Default:	Rev+Symph:	9.8ms
	Rev>Symph:	2.5ms

Wave Type

There are two wave types which can be applied to the modulation oscillator: sine and triangle. See the **Symphonic** section for additional information.

Options:	Sine, Triangle
Default:	Sine

Reverb>Pan

While auto-panning is often applied in an overly-dramatic fashion, this combination effect can be quite subtle and effective since it only auto-pans the reverb signal, leaving the track's dry signal alone. The effect is particularly subliminal and other-worldly when tracks using an auto-panned reverb are also fed through the master reverb.

Reverb Time (not remote-controllable)

This is the total time elapsed between the moment when a sample passes through the processor and the moment at which that sample's reverb level drops to zero.

Range:		0.3 - 99s
Increment:	0.3s - 5s:	0.1s
	5s - 10s:	0.5s
	10s - 20s:	1.0s
	20s-99s:	5.0s
Default:		6.0s

Initial Delay

This is the total elapsed time between the moment when the audio signal arrives at the processor and the first reflections are produced. (See the **Reverb** section for additional information.)

Range:	0 - 500ms
Increment:	0.1ms
Default:	30ms

High Frequency Decay Ratio

This dial specifies how rapidly high-frequency content will be removed from the reverb signal in relation to total reverb time. See the **Reverb** section for additional information.

Range:	0.1 - 1.0
Increment:	0.1
Default:	0.8

Diffusion

This parameter determines the perceived width of the reverb's stereo field. See the **Reverb** section for additional information.

Range: 1 - 10
Increment: 0.1
Default: 10

Density

This dial defines the "richness" of the reverb. See the **Reverb** section for additional information.

Range: 0% - 100%
Increment: 1%
Default: 82%

High Pass Filter

The **HPF/LPF** dials act as "tone controls" for the reverb, but these parameters play a more significant role than the term "simple tone control" might imply. See the **Reverb** section for additional information.

Range: Through - 8kHz
Increment: 1 semitone
Default: Through

Low Pass Filter

See above.

Range: 50Hz - Through
Increment: 1 semitone
Default: 11.3kHz

Reverb/Pan Balance

Sets the mix of reverb to panned reverb. Lower values increase the panning depth relative to unpanned reverb.

Range: 0% - 100%
Increment: 1%
Default: 30%

Frequency

Sets the oscillation rate for the panning modulator, i.e. the rate at which the sound appears to sweep across the stereo field. Low settings sweep the pitch modulator slowly; higher settings sweep it more rapidly.

Range: 0.05Hz - 40Hz
Increment: 0.05Hz
Default: 0.65Hz

Depth

This parameter controls the *width* of the stereo field in which the sound source is panned; depth may be a misnomer. At 100%, the sound source pans from 9:00 to 3:00; at lower values the width of the stereo image narrows.

Range:	0% - 100%
Increment:	1%
Default:	98%

Direction (not remote-controllable)

Five options are available with this parameter. See the **Auto Pan** section for details on the options.

Options:	L<->R, L->R, L<-R, Turn Left, Turn Right
Default:	L<->R

Wave Type (not remote-controllable)

Three options are available here: sine, triangle and square wave oscillation. See the **Auto Pan** section for details on the options.

Options:	Sine, Triangle, Square
Default:	Sine

Delay+Early Reflections / Delay>Early Reflections

As mentioned in the **Early Reflections** parameter guide, the E/R effect is inherently well-suited for use as an echo-type delay effect. Combining the two effects adds even more flexibility to the range of this effect. With the low delay times available, "static flanging" and slapback effects can be applied using early reflections as the delayed signal.

Delay Time Left, Delay Time Right

Specifies the time between repeats of the delayed signal. In this effect, delay times can be set independently for the left and right output channels of the delay. Because this effector is two independent delays *plus* E/R, maximum delay times for this effect are *less* than half the maximum delay time for **Mono Delay** and **Modulation Delay.**

Range:		0 - 1000ms
Increment:		0.1ms
Default:	D+E/R Left:	27.5ms
	D+E/R Right:	26.0ms
	D>E/R Left:	250ms
	D>E/R Right:	500ms

Feedback Delay

Sets an initial "predelay" time before delays or feedback echoes begin. See **Left/Right/Center Delay** for a full description of this parameter's capabilities.

Range:		0 - 1000ms
Increment:		0.1ms
Default:	Delay+E/R:	28.0ms
	Delay>E/R:	500.0ms

Feedback Gain

This specifies the percentage of original volume the signal should have with each successive echo, and indirectly determines the total number of echoes produced. Feedback cannot be adjusted independently for the left and right channels. See the **Mono Delay** section for additional information.

Range:		-99% - +99%
Increment:		1%
Default:	Delay+E/R:	+8%
	Delay>E/R:	+48%

High Frequency Decay Ratio

This parameter, when applied to delays, controls how rapidly high-frequency content will be removed from the delayed signal in relation to the number of feedback echoes. This parameter cannot be set independently for left and right channels. See the **Mono Delay** section for additional information.

Range:		0.1 - 1.0
Increment:		0.1
Default:	Delay+E/R:	0.8
	Delay>E/R:	0.4

Type (not remote-controllable)

Specifies the reverb algorithm used to define the sound of the early reflection cluster. For a complete description of the available types, see the **Early Reflections** section.

Options:		Small Hall, Large Hall, Random, Reverse, Plate, Spring
Default:	Delay+E/R:	Small Hall
	Delay>E/R:	Large Hall

Room Size

Specifies the size of the virtual enclosure used to produce the reflections. See **Early Reflections** for a additional information.

Range:	0.1 - 20
Increment:	0.1
Default:	2.4

Liveness

The “liveness” of an ambient space is a measure of how well it reflects sound from multiple surfaces without losing volume or timbre. See the **Early Reflections** section for additional information.

Range:		0 - 10
Increment:		1
Default:	Delay+E/R:	4
	Delay>E/R:	4

Initial Delay

Specifies the length of time before the first reflection is heard.

Range:		0 - 500ms
Increment:		0.1ms
Default:	Delay+E/R:	25.0ms
	Delay>E/R:	10.0ms

Diffusion

This parameter determines the perceived width of the reverb’s stereo field. Higher values produce a wider perceived stereo field. See the **Reverb** section for additional information.

Range:		0 - 500ms
Increment:		0.1ms
Default:	Delay+E/R:	25.0ms
	Delay>E/R:	10.0ms

Density

This dial defines the "richness" of the early reflections. See the **Reverb** section for additional information.

Range:		0% - 100%
Increment:		1%
Default:	Delay+E/R:	80%
	Delay>E/R:	90%

Early Reflection Number

This specifies the total number of early reflections that the effector will generate. See the **Early Reflections** section for additional information.

Range:		1-19
Increment:		1
Default:	Delay+E/R:	16
	Delay>E/R:	18

Delay/Reflections Balance

In the **Delay+Early Reflections** algorithm, this control determines the mix of delay effect with E/R effect. At 0% the effect uses E/R only; at 100% it uses delay only. In **Delay>Early Reflections** it determines the relative level of the two effects. At 0% it uses delay only; at 100% it ignores delay parameters and no effect is produced since no signal reaches the early reflections section of the effect.

Range:		0%-100%
Increment:		1%
Default:	Delay+E/R:	58%
	Delay>E/R:	30%

Delay+Reverb Reverb>Delay

Delay+Reverb is a parallel effect in which both echo and reverb can be applied to the same track. **Reverb>Delay** applies echo delay to the reverb signal. These are useful combo effects for non-critical tracks, but due to the quality loss in the reverb algorithms and the reduced degree of control over reverb parameters compared to the dedicated reverb effects, they may not be suitable for mission-critical applications.

Delay Time Left, Delay Time Right

Specifies the time between repeats of the delayed signal. In this effect, delay times can be set independently for the left and right output channels of the delay. Because this effector is two independent delays *plus* reverb, maximum delay times for this effect are *less* than half the maximum delay time for **Mono Delay** and **Modulation Delay.**

Range:		0 - 1000ms
Increment:		0.1ms
Default:	D+Rev Left:	410.0ms
	D+Rev Right:	398.0ms
	Rev>D Left:	43.0ms
	Rev>D Right:	50.0ms

Feedback Delay

Sets the initial delay before feedback echoes begin. See **Left/Right/Center Delay** for a full description of this parameter's capabilities.

Range:		0 - 1000ms
Increment:		0.1ms
Default:	Delay+Rev:	410.0ms
	Rev>Delay:	50.0ms

Feedback Gain

This specifies the percentage of original volume the signal should have with each successive echo, and indirectly determines the total number of echoes produced. Feedback cannot be adjusted independently for the left and right channels. See the **Mono Delay** section for additional information.

Range:		-99% - +99%
Increment:		1%
Default:	Delay+Rev:	+47%
	Rev>Delay:	+12%

Delay High Frequency Decay Ratio

This parameter, when applied to delays, controls how rapidly high-frequency content will be removed from the delayed signal in relation to the number of feedback echoes. This parameter cannot be set independently for left and right channels. See the **Mono Delay** section for additional information.

Range:		0.1 - 1.0
Increment:		0.1
Default:	Delay+Rev:	0.1
	Rev>Delay:	0.8

Reverb Time (not remote-controllable)

This is the total time elapsed between the moment when a sample passes through the processor and the moment at which that sample's reverb level drops to zero.

Range:		0.3 - 99s
Increment:	0.3s - 5s:	0.1s
	5s - 10s:	0.5s
	10s - 20s:	1.0s
	20s-99s:	5.0s
Default:	Delay+Rev:	3.6s
	Rev>Delay:	1.4s

Initial Delay

This is the total elapsed time between the moment when the audio signal arrives at the processor and the first reflections are produced. (See the **Reverb** section for additional information.)

Range:		0 - 500ms
Increment:		0.1ms
Default:	Delay+Rev:	25ms
	Rev>Delay:	16ms

Reverb High Frequency Decay Ratio

This dial specifies how rapidly high-frequency content will be removed from the reverb signal in relation to total reverb time. See the **Reverb** section for additional information.

Range:		0.1 - 1.0
Increment:		0.1
Default:	Delay+Rev:	0.7
	Rev>Delay:	0.6

Diffusion

This parameter determines the perceived width of the reverb's stereo field. See the **Reverb** section for additional information.

Range:		1 - 10
Increment:		1
Default:	Delay+Rev:	10
	Rev>Delay:	10

Density

This dial defines the "richness" of the reverb. See the **Reverb** section for additional information.

Range:		0% - 100%
Increment:		1%
Default:	Delay+Rev:	94%
	Rev>Delay:	70%

High Pass Filter

The **HPF/LPF** dials act as "tone controls" for the reverb, but these parameters play a more significant role than the term "simple tone control" might imply. See the **Reverb** section for additional information.

Range:		Through - 8kHz
Increment:		1 semitone
Default:	Delay+Rev:	Through
	Rev>Delay:	Through

Low Pass Filter

See above.

Range:		50Hz - Through
Increment:		1 semitone
Default:	Delay+Rev:	3.17kHz
	Rev>Delay:	5.33kHz

Delay Balance

In the **Delay+Reverb** algorithm, this control determines the mix of delay effect with reverb effect. At 0% the effect uses reverb only; at 100% it uses delay only. In **Reverb>Delay** it determines the relative level of the two effects. At 100% it ignores reverb parameters and no effect is produced since no signal reaches the delay section of the effect.

Distortion>Delay

This is a fairly typical rock/techno combination effect. The algorithm applies distortion or overdrive to the signal prior to feeding it to the delay. Since this effect provides a modulator and offers tight control over short delay times, functional equivalents of drive-plus-chorus, drive-plus-flange, drive-plus-light-phasing and drive-plus-slapback can be achieved in addition to the expected drive-plus-echo effect.

Distortion Type (not remote-controllable)

Five distortion algorithms are available, each with its own particular character. See the **Distortion** section for additional information about the distortion types.

Options: Distortion 1, Distortion 2, Overdrive 1, Overdrive 2, Crunch
Default: Distortion 1

Drive

This controls the amount of distortion applied to the input signal and indirectly determines how the distortion/overdrive section responds to the input signal's dynamics. See the **Distortion** section for additional information.

Range: 0 - 100
Increment: 1
Default: 90

Master Gain

This controls the output gain of the effect. See the **Distortion** section for additional information.

Range: 0 - 100
Increment: 1
Default: 40

Tone

This is functionally equivalent to the tone control of a pedal-type distortion/overdrive and appears to use the same algorithm as the tone EQ type. High frequencies can either be boosted or cut. Highs often need to be cut when using the **Distortion** algorithms to reduce the harshness of the tone.

Range: -10 - +10
Increment: 1
Default: -6

Noise Gate

This parameter specifies the threshold level for the noise gate. See the **Distortion** section for additional information.

Range: 0 - 20
Increment: 1
Default: 3

Delay Time

Specifies the delay time and the time between repeats of the delayed signal.

Range: 0 - 2725ms
Increment: 0.1ms
Default: 385.0ms

Feedback Gain

This specifies the percentage of original volume the signal should have with each successive echo, and indirectly determines the total number of echoes produced by the delay section. See the **Mono Delay** section for additional information.

Range: -99% - +99%
Increment: 1%
Default: +30%

High Frequency Decay Ratio

This parameter, when applied to delays, controls how rapidly high-frequency content will be removed from the delayed signal in relation to the number of feedback echoes. This parameter cannot be set independently for left and right channels. See the **Mono Delay** section for additional information.

Range: 0.1 - 1.0
Increment: 0.1
Default: 0.3

Frequency

Sets the oscillation rate for the pitch (delay time) modulator in the delay unit. See the **Modulated Delay** section for additional information.

Range: 0.05Hz - 40Hz
Increment: 0.05Hz
Default: 0.35Hz

Depth

This specifies the modulation depth, or the range of pitch variation possible in the delayed signal. See the **Modulated Delay** section for additional information.

Range: 0 - 100%
Increment: 1%
Default: 78%

Distortion/Delay Balance

Since **Master** controls the output gain of the drive effect, this parameter acts as a gain control for the delay effect.

Range: 0 - 100%
Increment: 1%
Default: 78%

Multi Filter

This effect acts as three *parallel* resonant filters. (Parallel meaning that they can't be "stacked". If you're not familiar with the concept, think of these as similar to the resonant filters on synthesizers or wah pedals.

These are remarkably versatile filters offering fourth-order filtering (24dB/octave) with resonance covering almost the entire frequency range of the 01V, and can be used as very precise EQ devices or as resonant peaking devices similar to resonant synth filters. You'll hear the results instantly when you pass a signal through the filters.

This effect does not use an oscillator to "sweep" the resonant peak. When used without resonance, these act as "semi-parametric" EQs offering gain and center frequency control but no Q (frequency range) control.

{bmlt note.shg}Note that this effect has a significant drawback in automated mixing. The filters cannot be "swept" across the frequency range in real time during an automated mix because the frequency values do not shift smoothly.

These effects won't replace costly dedicated resonant filters, but they are useful algorithms nonetheless since they offer precise fourth-order filtering and effectively three bands of "EQ" if the effect is used this way. These should probably not be used to replace or supplement filters in software or hardware synths because they may not be well-matched to the synth design, but they are of most use in synth-based pop and rock, and for special effects.

Type 1, 2, 3 (not remote-controllable)

There are three options: high pass, low pass and band pass. The typical application is band pass. High and low pass both act as harsh filters, trimming off all low or high frequencies respectively and creating a very thin output sound.

Options:	HPF, LPF, BPF
Default:	LPF

Frequency 1, 2, 3 (not remote-controllable)

This sets the center frequency of the filter. Functionally equivalent to the center frequency of a "sweep" or parametric EQ.

Range:	37Hz - 22kHz
Increment:	Variable throughout the range
Default:	37Hz

Level 1, 2, 3

This controls the output gain of the specified filter.

Range:	0 - 100
Increment:	1
Default:	0

Resonance 1, 2, 3

Resonance controls in filters are functionally equivalent to feedback controls in delay lines. This determines how much of the EQ'd signal is fed back into the filter. The more feedback, or resonance, the more pronounced the effect around the center frequency.

Range:	0 - 20
Increment:	1
Default:	0

Freeze

Perhaps you've heard of the "hidden sampler" embedded in the 01V...well, here it is. This is a single-shot sample record/playback effect similar to the sampling effects found in many outboard delay lines and reverbs. It offers up to three seconds of mono sampling, and the use of unusual phase/predelay-plus-feedback settings on the output can add even more drama to the effect.

This effect is particularly useful for multimedia, stage and pop/dance production, and can also be applied to live performance with a little practice. Familiarity with the parameter set will open up the full range of creative opportunities inherent in the effect.

Record mode

This determines how recording should be triggered. **Manual** triggers recording when the **Enter** button is pressed on the 01V. **Input** triggers recording when **Trigger level** reaches the specified trigger volume.

Options: Manual, Input
Default: Manual

Start

Determines where the sample playback will begin. Sample playback is measured in samples, not in time. This is an important distinction, but it shouldn't be difficult to grasp. 44.1kHz sample rate gives you approximately 131,000 samples in a 3-second sample, or 11,025 samples in a 250ms sample. Each increment of start time is approximately 1,300 samples, giving you a total of 100 options from this dial.

Range: 0 - 131,070 samples
Increment: 1,310.7 samples
Default: 0

Play mode

There are three ways that samples can be triggered for playback. Momentary playback plays a single-shot sample when the sampler is triggered from the **Enter** button or a remote controller that activates this control. Continuous playback will continually repeat the sample as long as the trigger is active (especially useful when triggering from a MIDI keyboard or sequencer). Input triggering plays the sample only when the input trigger level reaches the trigger level threshold.

Options: Momentary, Continuous, Input
Default: Momentary

End

Determines where the sample playback will end. The difference between the **Start** and **End** period of the sample is the total length of sample playback when the sample is triggered.

Range: 0 - 131,070 samples
Increment: 1,310.7 samples
Default: 0

Loop number

Determines the number of repeats of a given section of a sample. Typical single-shot sampling uses no loops, just a single playback. When a loop number greater than zero is used, the **End** parameter is ignored; the sample is instead looped a specified number of times using a loop length specified here.

Multiplying the number of loops increases the number of repeats. Note that continuous-play playback should probably not be used with multiple loops since the number of loops will likely be determined by a MIDI trigger.

Range: 0 - 100
Increment: 1
Default: 0

Loop length

Specifies the length of the loop in samples (see above).

Range: 0 - 131,070 samples
Increment: 1,310.7 samples
Default: 0

Trigger level

When loop recording or playback is set to **Input,** the trigger level is the input volume that must be reached before the signal is recorded or the loop is played back.

Range: 0dB - -60dB
Increment: 1dB
Default: 0dB

Record Delay

This determines the delay time between triggering and the beginning of sample record. This parameter is typically used to allow for any preparation that might be needed prior to generating or acquiring the sample, for compensating for "reaction time" delays, or for insertion or elimination of short periods of silence at the beginning of the sample.

Negative values are possible because the signal fed through this effect is stored in memory. For example, when a -500ms value is selected, the recorded sample will actually include 500ms of audio from *before* the recording was actually triggered.

Range: -1000ms - 1000ms
Increment: 1ms
Default: -1000ms

Trigger Master

Range: -1000ms - 0ms
Increment: 1ms
Default: -1000ms

MIDI Trigger (not remote-controllable)

When a MIDI device such as a keyboard or sequencer is used to trigger the sampler, any single key can be used as the trigger key, or all keys in the range C1-C6 can trigger the sample. This parameter should be left in the default **OFF** position if the sample is intended to be triggered by a threshold input signal.

Range: OFF, C1 - C6, ALL
Increment: 1 semitone
Default: OFF

Pitch

The output pitch of the sample can be shifted up or down by up to an octave in semitone increments using this control. A value of **0** outputs the sample at the pitch at which it was recorded.

Range: -12 - 12
Increment: 1 semitone
Default: -12

Fine Pitch

The output pitch of the sample can be further refined in cents using this control. A value of **0** applies no fine pitch correction to the sample.

Range:	-50 - 50
Increment:	1 cent
Default:	-50

EQs IN THE 01V

EQ types overview

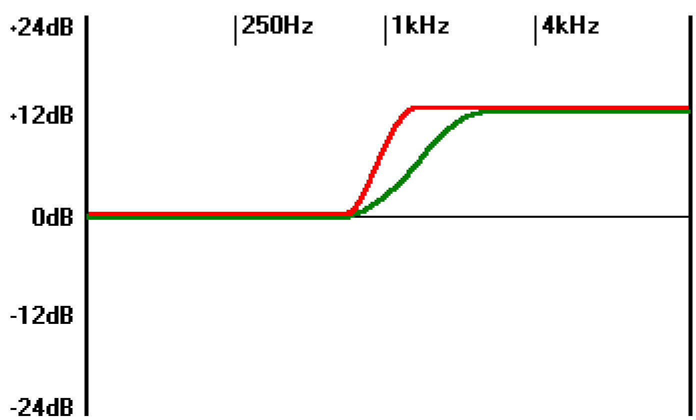
While we can tell you what each of the EQ types do, we can't tell you how they sound. One of the best ways to get a strong sense of how the various EQ types actually affect your sound is to experiment with them on a sample of white noise. (Pink noise is preferable if you have a synthesizer or software tone generator that can produce it.) EQing a noise sample affords you a much more dramatic demonstration of how the selected EQ affects the entire audible spectrum than EQing a musical track, since a noise sample will (or at least it *should*) contain the complete audible spectrum.

As of this writing, Harmony Central hosted a good basic tutorial on EQ for recordists at *http://harmony-central.com/Effects/Articles/Equalization/*. The page includes several graphics which help illustrate the types of response curves discussed below.

Who gave the orders?

"Order", by the way as it is used in reference to audio filters, refers to the order of magnitude of the filter's ability to refine its frequency response. As previously suggested, each order of magnitude steepens the response curve of a filter, allowing for more precise control over boost or cut of the desired frequency range.

This illustration gives you a general idea of the difference in frequency response between a 1st order (green) and 2nd order (red) filter. A 1st order filter can only produce a frequency response slope of 12dB per octave. A 2nd order filter, on the other hand, has a slope of 6dB per octave. A 1st order filter set for 12dB of gain at 1kHz doesn't reach the full 12dB of gain until 2kHz, where a 2nd order filter reaches full boost at 1.5kHz.



The difference between 1st and 2nd order filters is an important one once you begin to develop a sense of how the response curves color the sound. The precision of a 2nd order filter is critical when you want to filter specific frequency bands without affecting neighboring bands.

How critical? Well, think for a moment about the typical 30-band graphic EQ. Graphic EQs are simply a set of tuned bandpass filters. Now imagine needing to cut one band by 12dB to compensate for an annoying harmonic. Since the next band is only 1/3 octave away, you need to use a very precise filter to insure that:

a) you don't radically affect the neighboring frequency bands, and...

b) that the filter is tuned precisely enough that the response curve between two bands is smooth

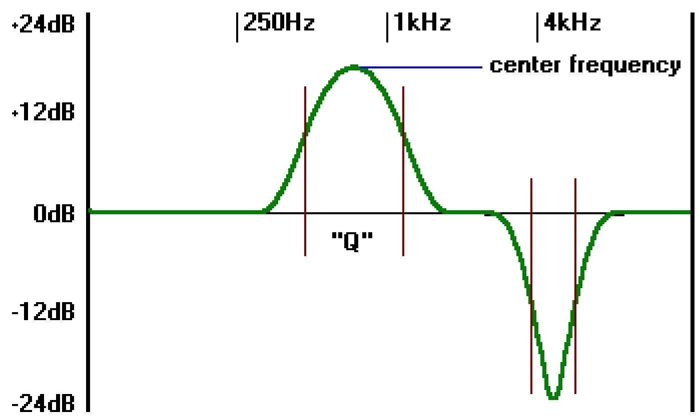
If the filters are not matched correctly to the type of equalization you want to perform, you wind up with a response curve with notches and sharp peaks where you would probably prefer a more natural-sounding response curve where gain rises and falls less dramatically over the frequency range.

Do a little mental calculation and you'll quickly realize that in the case of graphic EQs, filter magnitude has to be matched correctly to the number of bands available or the resulting response won't be "natural". A 10-band EQ requires 1st order filters, a 20-band EQ requires 2nd order filters, and 30-band EQs require (but...buyer beware...do *not* always use) 3rd order filters to insure that the response curve is smooth from one band to the next.

Peaking

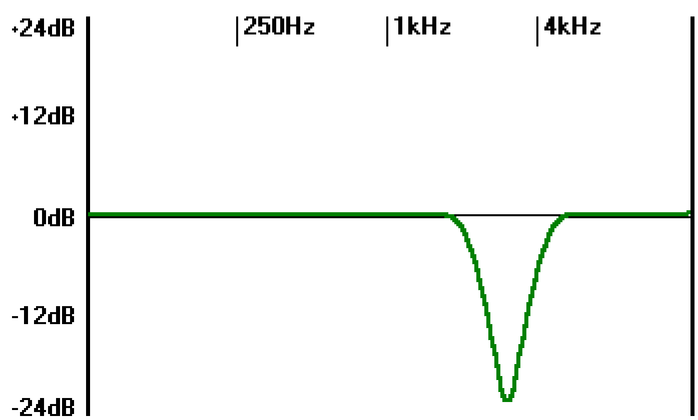
This is typical parametric equalization. Parametric EQ is so named because it requires parameters in which to work, unlike a graphical EQ or tone control which has only a boost/cut control. Parametric EQ applies a bell-shaped curve to the signal when boosting, or a dish-shaped curve when cutting. The **Gain** control adjusts the height or depth of the bell curve, and the Q control adjusts the width of the curve. **Freq** sets the center frequency of equalization, where the maximum gain or cut occurs.

In this illustration, we've used the **Peaking** EQ type to apply two bands of equalization. The first adds about 17dB of gain at 750Hz using a medium Q value. The second band is just as radical, applying 18dB of cut at 5kHz using a low Q value. Note the difference in the frequency response curve that the Q value makes. At 750Hz, the larger Q value used by our first band of EQ affects the frequency response for nearly an octave and a half on either side of the center frequency. The 5kHz cut band, on the other hand, using a lower Q value, only affects frequencies for less than an octave either side of the center frequency.



By modifying the Q value, you can affect very large frequency ranges in the sound or narrow the frequency range to a "notch" in the response curve, as shown in the illustration below.

What you *can't* do with parametric EQ, not without the most sophisticated and expensive models, is raise the Q so high that you're only affecting one frequency. Yes, you can apply two or three bands of -18dB high-Q cut at 60Hz to try to remove an annoying 60-cycle hum, but in doing so you'll also be cutting audible harmonics from the 45-80Hz range. That's nearly a full octave which will be affected by an attempt to eliminate a single frequency. Don't assume that this is a problem with the software. "Natural" EQ has a very similar effect, and the gradual arcs in the parametric bands are necessary -- even with notch filtering, to maintain natural-sounding equalization.



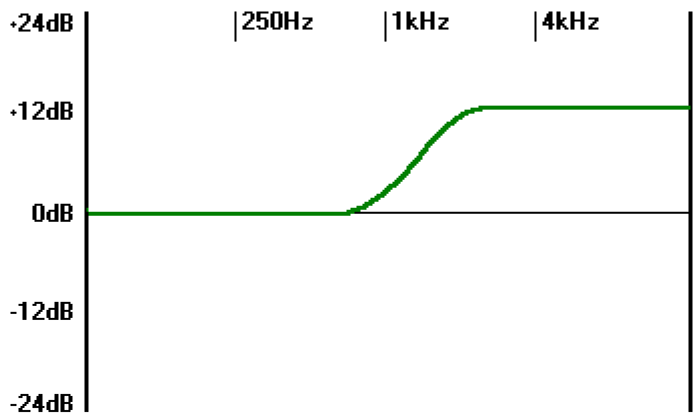
The **Bandpass** EQ type, when used to isolate specific frequency ranges, tends to have a much sharper slope, and appears to be at least a 2nd order filter able to handle much narrower frequency ranges without coloring adjacent bands. But it still has an effect on frequencies surrounding the center frequency. If you need *ultra*-narrow EQ bands applied to your tracks, ask about software solutions to the problem. Several specialty EQs and sample modifiers are available which have the ability to completely remove specific frequency ranges without affecting neighboring frequencies, a trick that even the costliest analog EQs can't perform.

Low and High Shelving

This might better be named ***2nd Order* Low/High Shelving.** The audible effect is similar to **1st Order Low/High Shelving,** but the gain boost or reduction slope between 0dB and the target frequency is much steeper than **1st Order Low/High Shelving.** The EQ type is so named because after applying boost or gain, you see a sharp curve beginning at the selected **Frequency** (also known as the "knee") and extending for a couple of octaves, and the remainder of the equalization curve is as flat as the frequencies below the boost/cut frequency.

In practical terms, this EQ type's effect is somewhat like a very sharp bass or treble control, but with one critical difference: you can control the frequency at which bass or treble boost/cut begin. The sample frequency response curve shown below illustrates a shelving boost of 13dB at 800Hz. Such an EQ might be used to compensate for sound recorded in a bassy and resonant building or enclosure, although other types of EQ would probably be more suitable to this task. A more practical application of shelving is to reduce the presence of one or more obtrusive male background vocal tracks by applying a shelf cut at 200Hz. This would reduce the presence of the vocal without completely eliminating its bass response.

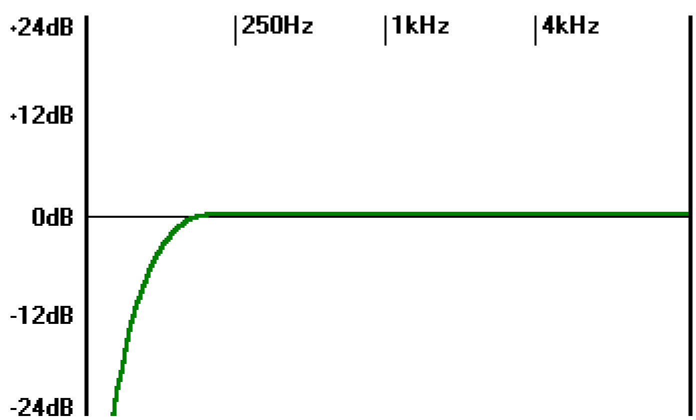
Whenever using this effect, you may want to compare the result with similar settings using the **1st Order Low/High Shelving** and the **1st Order Low/High Tone** types. In many cases a gentler slope, or a continuous slope such as that produced by the tone type EQ, will be more effective for a given track than this EQ type.



The difference between low and high shelving is that low shelving is designed to "shelve" (boost or cut) frequencies *below* the specified starting frequency and high shelving shelves frequencies *above* the specified frequency.

Low and High Pass

The shelving EQ types (described just above) apply a particular amount of boost or cut to all frequencies above or below the frequency specified on the slider. Low and high pass filters, on the other hand, literally chop off all frequencies above or below the specified frequency, allowing only the low or high frequencies to *pass* through the EQ's output. This allows you to remove all low-frequency content from a track or, conversely, limit the track to nothing but low and midrange frequencies.



The illustration below shows a representation of a high pass filter applied at about 200Hz. As you can see, the curve drops sharply beginning at 200Hz, filtering all low-frequency content in the signal. A similar low-pass filter curve at, for example, 4kHz would show the same dip, only it would extend from 0Hz to 4kHz on the baseline and drop sharply at 4kHz, allowing all signals below the 4kHz level to pass through.

Only the **Frequency** control is active for this EQ type. The rolloff curve is fixed for both low and high pass filters.

A bit of trivia...

By the way, if the word “rolloff” sounds familiar, it should. While today’s recordings can contain bass as low as 4 to 8Hz, and newer systems can actually *handle* frequencies this low, for decades the Record Industry Association of America prescribed a specific type of high pass filter, known as the “RIAA rolloff curve”, for application to all recordings produced with RIAA logos. A high pass filter which removed frequencies below 32Hz in a steep curve was used to help to insure that both consumer and broadcast audio products could safely handle a relatively full frequency range at high volumes without significant loss of fidelity or risk of damage to the speakers of the day.

While RIAA filters are still in use, newer hardware is significantly more tolerant of low frequency information. Newer pop, modern dance, soundtrack and experimental music often completely ignores the RIAA rolloff because the application of this filter significantly impairs the output of subwoofer frequencies.