Temecula DSP DEEP/4 — Introduction

The DEEP/4 is a 24-bit digital effects processor built around four separate processing engines, each with its own dedicated input and output, along with comprehensive mixdown functionality.

Credits and Inspiration

The DEEP/4 is deeply inspired by the classic Ensoniq DP/4 effects processor from 1992. We wish to acknowledge and credit the brilliant original engineering team who created that legendary hardware unit, whose pioneering work defined an entire generation of digital effects processing and routing architecture:

- Jon Dattorro
- Bill Mauchly
- Dave Andreas
- John O. Senior
- Tom Metcalf
- Bill McCutcheon

The Effects

Over 40 fully programmable, high-fidelity digital effect algorithms are available in the DEEP/4. The selection spans reverbs, chorus, flanging, delays, distortion, pitch shifting, and numerous other effect types, many offering dynamic real-time control of their settings. A total of 400 effect presets are provided: 200 factory ROM (Read Only Memory) presets and 200 user-editable RAM (Random Access Memory) locations for storing your own creations.

Parallel Processing

Unlike conventional multi-effects units that handle only a single input signal, the DEEP/4's four-input, four-output architecture supports true multi-channel parallel processing. Although there is a single user interface, up to four distinct input signals can each be routed to their own dedicated internal processor. The multiple I/O design also enables specialized effect types such as vocoding and ducking.

You can operate the DEEP/4 as a single large effects system, a pair of stereo-input processors, three separate processors, or four fully independent effects units. Routing among the four processing engines is entirely programmable, supporting any mix of serial and parallel signal paths. Feedback loops and side-chain routing are also available. This flexible architecture, combined with the extensive algorithm library, enables unusual effect topologies that rigid-routing systems cannot achieve. Additionally, the built-in output mixing feature lets you combine the stereo outputs of all four units into a single stereo pair (outputs 1 and 2), conserving mixer return channels.

The Four Units

Four independent effects processors reside inside the DEEP/4, designated as Units A, B, C, and D. Each relies on an ESP (Temecula DSP Signal Processor) chip paired with 16-bit analog-to-digital and digital-to-analog converters, delivering a very high quality output signal.

The four processors can also be "ganged" in pairs or groups of four to run algorithms that demand more horsepower than a single unit provides. The Pitch Shift 2U and 3.3 sec Delay 2U algorithms are examples of these multi-unit effects.

Signal Flow and Routing

The four processing units in the DEEP/4 (A, B, C, and D) support a broad range of interconnection schemes. All routing between units is fully programmable through the Config (Configuration). The Config governs the number of input sources, the way units are wired to one another, and how outputs are combined.

Units, Sources, and Configs

Four independent effects processors reside in the DEEP/4, designated Units A, B, C, and D.

For the purpose of connecting inputs, the DEEP/4 works in terms of **Sources** — the number of distinct audio signals entering the processor. You can set the DEEP/4 to handle 1, 2, 3, or 4 Sources at once.

Important: A *source* is not equivalent to an *input*. A source may be either mono or stereo. Two audio signals plugged into Inputs 1 and 2 could serve as one stereo source or as two separate mono sources. The distinction is entirely governed by the configuration.

How inputs and outputs map to the units is dictated by the active **Config**. The Config manages every connection in the internal digital "patch bay." Switching to a different Config rewires the DEEP/4 to suit your current requirements. All routing, algorithm selections, and parameter settings can be stored in a **Config Preset**.

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1 Source Config

The 1 Source Config turns the DEEP/4 into a single large multi-effects processor, directing the same input signal through all four units.

- **Mono In:** The signal arriving at Input 1 is distributed as a mono feed to every unit.
- **Stereo In:** Inputs 1 and 2 are delivered together as a stereo feed to every unit.

The 1 Source Config works well for:

- Applying effects to a single instrument like a guitar or keyboard
- Building a chain of premium-quality effects for processing a vocal or other critical audio source

1 Source Config Parameters

Param	Name	Range	Description
00	Input Config Select	1 Source Config	Selects 1 Source mode
01	AB–CD Routing	Serial or Parallel	Routes the AB pair to/from the CD pair. Serial: AB feeds into CD. Parallel: AB and CD process in parallel and outputs are mixed.
02	AB Unit Routing	Serial, Parallel, Feedback1, Feedback2	How Units A and B are connected within the pair
03	CD Unit Routing	Serial, Parallel, Feedback1, Feedback2	How Units C and D are connected within the pair
04	AB Config Depen-dent	Dry Path / Feedback amount	If AB is serial: controls dry signal path around AB. If AB is feedback: controls feedback amount B->A. If AB is parallel: unused.
05	CD Config Depen-dent	Dry Path / Feedback amount	Same as 04 but for the CD pair
06	AB Input Select	(12) Stereo or (1) Mono	Selects mono (Input 1) or stereo (Inputs 1 and 2) input
07–10	Bypass Kill A/B/C/D	Bypass or Kill	Determines what happens when a unit is bypassed. Bypass (b): only dry signal passes through. Kill (k): no signal passes through.

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2 Source Config

The 2 Source Config splits the DEEP/4 into two independent multi-effects processors, each with two units of processing power. One signal feeds the A & B pair; the other feeds C & D. The two pairs operate as completely separate two-unit devices.

- The AB pair's stereo output is routed to Outputs 1 and 2
- The CD pair's stereo output is routed to Outputs 3 and 4

The 2 Source Config works well for:

- Independently processing signals from two keyboards, or from a guitar and a keyboard
- Studio setups requiring two distinct multi-effects chains (such as Chorus & Reverb or Flange & Delay) on two different signals simultaneously

2 Source Config Parameters

Param	Name	Range	Description
00	Input Config Select	2 Source Config	Selects 2 Source mode
01	AB Unit Routing	Serial, Parallel, Feedback1, Feedback2	How Units A and B are routed together
02	CD Unit Routing	Serial, Parallel, Feedback1, Feedback2	How Units C and D are routed together
03	AB Input Select	(12) Stereo or (1) Mono	Mono or stereo input for the AB pair
04	CD Input Select	(34) Stereo or (3) Mono	Mono or stereo input for the CD pair
05	AB Config Dependent	Dry Path / Feedback amount	Same behavior as 1 Source Config param 04
06	CD Config Dependent	Dry Path / Feedback amount	Same behavior as 1 Source Config param 05
07–10	Bypass Kill A/B/C/D	Bypass or Kill	Per-unit bypass behavior

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3 Source Config

The 3 Source Config organizes the DEEP/4 into three separate effects processors. Units A and B each operate independently as single-unit processors, while Units C & D are paired together as one two-unit effects processor.

- Input 1 feeds Unit A, Input 2 feeds Unit B
- Input 3 (or Inputs 3 and 4 for stereo) drives Units C & D
- The stereo outputs of A and B can either be blended to Outputs 1 and 2 (Mixed Stereo) or routed separately to Output 1 and Output 2 (Dual Mono)
- The CD pair sends its output to Outputs 3 and 4

The 3 Source Config works well for:

- Studio scenarios requiring two independent single effects (e.g., separate reverbs for drums and vocals) alongside a multi-effect chain running simultaneously
- Dedicating Unit A to a guitar, Unit B to one Aux Send, and Units C & D to a second Aux Send

3 Source Config Parameters

Param	Name	Range	Description
00	Input Config Select	3 Source Config	Selects 3 Source mode
01	AB Output Select	Dual Mono or Mixed Stereo	Dual Mono: A->Out1, B->Out2 independently. Mixed Stereo: A and B mixed to stereo Outs 1&2.

Param	Name	Range	Description
02	CD Unit Routing	Serial, Parallel, Feedback1, Feedback2	How Units C and D are routed together
03	CD Input Select	(34) Stereo or (4) Mono	Mono or stereo input for the CD pair
04	CD Config Dependent	Dry Path / Feedback amount	Same behavior as other configs
05–08	Bypass Kill A/B/C/D	Bypass or Kill	Per-unit bypass behavior

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4 Source Config

The 4 Source Config represents one of the DEEP/4's most versatile arrangements. Every Unit (A, B, C, and D) operates as a fully independent single-unit effects processor. All four unit inputs are mono.

- Input 1 -> Unit A, Input 2 -> Unit B, Input 3 -> Unit C, Input 4 -> Unit D
- Outputs can be mixed stereo or independent mono

The 4 Source Config is especially suited for:

- Studio environments where four entirely independent effects are needed at the same time

4 Source Config Parameters

Param	Name	Range	Description
00	Input Config Select	4 Source Config	Selects 4 Source mode
01	AB Output Select	Dual Mono or Mixed Stereo	How A and B outputs are sent to Outs 1 and 2
02	CD Output Select	Dual Mono or Mixed Stereo	How C and D outputs are sent to Outs 3 and 4
03–06	Bypass Kill A/B/C/D	Bypass or Kill	Per-unit bypass behavior

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Signal Routing Between Units

Based on the active Config, units can be interconnected in one of five distinct ways, each represented by a different symbol on the DEEP/4 display:

Symbol	Name	Meaning
->	Serial	Signal passes from the left unit into the right unit
+	Parallel	Both units receive the input simultaneously; their outputs are summed together
/+	Feedback	Serial connection with a feedback path returning the second unit's output back to the first unit's input
*	Ganged	Two units are "ganged together" running a 2-unit algorithm (e.g., Pitch Shift 2U). Routing is fixed and cannot be changed.
(space)	Independent	No link between the units; each is routed on its own

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Understanding Serial and Parallel Routing

Serial Routing

With serial routing, the input signal passes *through* the first unit *before* reaching the input of the second unit.

```
Input -> [Unit A] -> [Unit B] -> Output
```

If the first unit runs a chorus and the second runs a reverb, the signal flows through the chorus first, then into the reverb. The result is a chorused sound with reverb layered on top of it.

On the DEEP/4 display, a **->** symbol appears between any units (or unit pairs) wired together in series.

Parallel Routing

With parallel routing, the same input signal is fed independently to *both* units, and their outputs are then summed together.

```
        -> [Unit A] ->
Input ->                 (+) -> Output
        -> [Unit B] ->
```

If the first unit runs a chorus and the second a reverb, you hear the chorused sound *alongside* the reverbed sound, but the chorus output does *not* pass through the reverb, and the reverb output does not carry any chorusing.

On the display, a **+** symbol appears between units connected in parallel.

Feedback Routing

Feedback routing resembles serial routing, but adds a feedback path that returns the second unit's output back to the first unit's input.

```
Input -> (+) -> [Unit A] -> [Unit B] -> Output
          ^                     |
          +------ Feedback ----+
```

The distinction between **Feedback 1** and **Feedback 2** lies in how the dry signal merges with the wet signal:

Feedback 1: The dry signal is summed at Unit A's input (ahead of the feedback summing point) and again at the final output after Unit B.

Feedback 2: The dry signal is summed between Unit A and Unit B (at Unit A's wet output) and again at the final output after Unit B.

AB to CD Signal Routing

In a 1 Source configuration (all four units processing one input signal), you can choose either serial or parallel routing between the AB and CD unit pairs.

Serial between AB and CD

```
Input -> [A] -> [B] -> [C] -> [D] -> Output
```

Parallel between AB and CD

```
        -> [A] -> [B] ->
Input ->                  (+) -> Output
        -> [C] -> [D] ->
```

Serial and parallel routing can be freely combined both *within* each pair and *between* the pairs. This yields a vast array of distinct configurations, each producing noticeably different sonic results.

Available ABCD Routing Combinations

By combining the AB–CD Routing (serial/parallel), AB Unit Routing (serial/parallel/feedback), and CD Unit Routing (serial/parallel/feedback), a total of 32 routing permutations are possible. The 18 unique signal-path topologies (excluding dry-path variations between Feedback 1 and 2) are:

AB Routing	AB–CD Link	CD Routing	Description
Serial	-> Serial	Serial	Full four-unit chain: A->B->C->D
Serial	-> Serial	Parallel	A->B feeds into C+D parallel
Serial	-> Serial	Feedback	A->B feeds into C<->D feedback loop
Parallel	-> Serial	Serial	A+B parallel, then C->D serial
Parallel	-> Serial	Parallel	A+B parallel, then C+D parallel
Parallel	-> Serial	Feedback	A+B parallel, then C<->D feedback
Feedback	-> Serial	Serial	A<->B feedback, then C->D serial
Feedback	-> Serial	Parallel	A<->B feedback, then C+D parallel
Feedback	-> Serial	Feedback	A<->B feedback, then C<->D feedback
Serial	+ Parallel	Serial	A->B and C->D in parallel
Serial	+ Parallel	Parallel	A->B and C+D in parallel
Serial	+ Parallel	Feedback	A->B and C<->D in parallel

AB Routing	AB–CD Link	CD Routing	Description
Parallel	+ Parallel	Serial	A+B and C->D in parallel
Parallel	+ Parallel	Parallel	A+B and C+D all in parallel
Parallel	+ Parallel	Feedback	A+B and C<->D in parallel
Feedback	+ Parallel	Serial	A<->B and C->D in parallel
Feedback	+ Parallel	Parallel	A<->B and C+D in parallel
Feedback	+ Parallel	Feedback	A<->B and C<->D in parallel

Bypass and Kill

When deactivated, each unit can operate in either **Bypass** or **Kill** mode:

- **Bypass (b):** Only the dry signal passes through the unit. Functionally equivalent to setting the Mix to 00.
- **Kill (k):** No signal whatsoever passes through the unit. Functionally equivalent to setting the Volume to 00.

The Vocoder

About the Vocoder

The DEEP/4 includes a built-in vocoder. A vocoder examines the frequency content of an incoming signal (typically speech captured by a microphone) and imposes that spectral shape onto the pitched audio from a second source (such as a synthesizer or sampler).

How the Vocoder Works

The DEEP/4's vocoder engages all four units to carry out a single function. Each of the four algorithms handles a different frequency band, and they are wired in parallel so that every algorithm receives the same pair of inputs. The vocoder algorithms analyze the signal arriving at Input 1 and impose its characteristics onto the signal at Input 2. The vocoder config preset links the four specialized algorithms (Vocoder Low, Vocoder Mid1, Vocoder Mid2, and Vocoder High), which operate together to produce the complete vocoder effect.

The voice signal (vox) at Input 1 is routed to the Spectrum Analyzer. Inside the Analyzer, bandpass filters split the voice into individual frequency bands. The Analyzer then measures the amplitude within each band and passes that data to the Real Time Dynamic Multi-band EQ. This EQ section similarly divides the carrier signal (Input 2) into separate frequency bands. The output level of each carrier band is governed by the amplitude measured in the corresponding analyzer band. The net result is that the carrier signal's frequency spectrum is shaped to mirror the vox signal's spectrum. An additional internal path from the vox input bypasses the spectrum analyzer entirely, routing high-frequency sibilance (t's, p's, clicks, pops, etc.) straight to the output for better articulation.

Setting Up the Vocoder

Connect the voice signal (vox) to Input 1 (front or rear panel of the DEEP/4). Plug the synthesis signal (carrier) into Input 2 on the rear panel. For best results, use a carrier that is harmonically rich with wide bandwidth. Route Outputs 1 and 2 to your monitoring or playback system.

Selecting the Vocoder Preset

1. Press the Select button, followed by the Config button.
2. Use the Data Entry Knob or the Left/Right Arrow buttons to navigate to preset #61 Vocoder Preset.
3. Press the Select button to activate the Vocoder preset. The DEEP/4 will automatically switch to Unit A.

Using the Vocoder

Speak into the microphone while simultaneously playing suitable notes on a keyboard (or other controller) and listen to the output. Working with a vocoder takes some practice, but the musical possibilities are very rewarding. Popular techniques include producing “robot-speech” by talking into the mic while holding a single note, or generating choir-like textures by vocalizing “aah” or “ooh” into the mic while playing chords on the keyboard.

Keep in mind that the output pitch is determined solely by the carrier input and is unaffected by the pitch of your voice into the microphone. The tonal qualities of the carrier signal also influence vocoder quality. The carrier must contain enough harmonic content to span the vocoder’s frequency range, and it should be played in a pitch register that roughly matches the vox (microphone) input. For instance, speaking in a low-pitched voice while playing high notes on the keyboard will yield poor results.

Although Input 1 (vox) is optimized for speech, any audio source can be used. The vocoder will apply the spectral characteristics of whatever signal appears at Input 1 onto the carrier, which can yield some distinctive and unusual timbres.

Vocoder Parameters

Each vocoder algorithm provides a specific set of user-adjustable parameters. The first parameter identifies the algorithm name:

- Vocoder Low
- Vocoder Mid 1
- Vocoder Mid 2
- Vocoder High

Mix `00-99` Sets the balance between the dry signal and the wet signal. A value of 00 lets only the unprocessed signal through, while 99 removes the dry signal entirely, leaving only the wet (vocoded) output. High settings are recommended for this parameter.

Volume `00-99` Controls the output level. A value of 00 silences the signal completely, producing no sound.

Speech Gain `-48 to +48 dB` Sets the amount of boost or attenuation applied to the Input 1 (vox) source after pre-emphasis processing. Greater pre-emphasis levels generally call for higher speech gain values. Adjust this until the sound is satisfactory.

Vocoder Sibilance Level `00-99` Governs how much high-frequency sibilance is sent to the output. This filter passes all vox frequencies above roughly 3500 Hz directly to the synthesized output. Higher values generally improve articulation. A recommended starting point is approximately 20 on a single unit (A, B, C, or D), or 5 on each of the four units.

Vocoder Response Time `Slow, Normal, or Fast` Determines the speed at which the carrier follows the vox signal. A fast setting analyzes and synthesizes the signal quickly. A slow setting analyzes and synthesizes with greater accuracy. The default value is Normal.

Vocoder Pre-emphasis `00-99` Boosts high frequencies and attenuates low frequencies in the vox signal (Input 1). A value of 99 applies maximum emphasis, while 00 applies none.

Vocoder Modulators

Like all other algorithms, the vocoder supports real-time parameter control and shares the standard modulation parameters:

Mod1 Source / Mod2 Source `Off / Controller 1-8` Chooses the modulation sources that drive the parameter destinations. Each algorithm offers two independent mod sources. Any of the controller sources defined in the System MIDI mode may be selected.

Mod1 Destination / Mod2 Destination `00-34` Determines which algorithm parameters are modulated by the chosen sources. Any parameter in the algorithm may be targeted (except the algorithm name). Each vocoder algorithm provides two independent mod destinations.

Mod1/Mod2 Param Range Min / Param Range Max `00-99` These four parameters define the minimum and maximum modulation depth (expressed as a percentage of the target parameter's range) applied by the Mod Source to the Mod Destination.

Common Algorithm Parameters

All algorithms in the DEEP/4 share a set of common parameters. They are documented here once rather than being repeated on every individual algorithm page.

Mix (Parameter 01)

Range: 00 to 99

The Mix parameter sets the balance between the original (dry) signal and the fully processed (wet) signal. A value of 00 passes only the unprocessed signal, while 99 removes the dry signal entirely and leaves only the wet output. Certain algorithms work best with a blend of wet and dry, while others perform optimally at a setting of 99.

Volume (Parameter 02)

Range: 00 to 99

The Volume parameter governs the output level of the signal. A value of 00 silences the output entirely, meaning any algorithms and/or configs downstream will also receive no signal and produce no sound.

Algorithm Modulators

Every algorithm supports real-time parameter control and shares a common set of modulation parameters. Their exact position varies by algorithm, but they always occupy the final eight parameter slots.

Mod1 Source / Mod2 Source

Range: Off, Controller 1–8

These parameters choose the mod sources that drive the parameter Destinations. Each algorithm provides two independent mod sources. Any of the eight DEEP/4 System Controller sources configured in System/MIDI mode may be selected.

Mod1 Destination / Mod2 Destination

Range: Off, 01 to 34 (depending on the algorithm)

This parameter determines which algorithm parameters are modulated by the mod sources. The available choices depend on the active algorithm. Any parameter within an algorithm may be targeted (except the algorithm name). Each algorithm provides two independent mod destinations.

Mod1/Mod2 Param Range Min / Param Range Max

Range: 00 to 99

These four parameters define the minimum and maximum modulation depth (expressed as a percentage of the target parameter's range) applied by the Mod Source to the Mod Destination. Swapping the min and max values will invert the modulation effect.

Algorithms: Reverb

Gated Reverb

Gated Reverb delivers an outstanding gated reverb effect. Cutting off a reverb tail mid-decay produces the characteristic gated sound. The Gated Reverb's internal architecture is based on the Plate Reverb topology.

Description

To produce this gating behavior, both the Gated and Reverse reverb algorithms must gate several internal parameters rather than simply shaping the output amplitude envelope. Nevertheless, it is the output amplitude envelope that the user directly manipulates. The DEEP/4 provides a highly flexible gated reverb, tailored for percussive sources but effective with any input material.

The gate opens when the incoming signal crosses the trigger threshold, which is intentionally set as low as practical so that no part of the input signal is lost. What distinguishes the gated reverb from the reverse reverb is its retriggering behavior: each time the input signal crosses a user-adjustable retrigger threshold, the gate resets. The gate remains open while the input stays above the retrigger threshold, accumulating all incoming signal under a single gate window until the total input level drops below that threshold. At that point, the Hold Time countdown begins. Two separate thresholds exist to prevent false retriggers and to guarantee accurate hold time durations. For individual gating on every single note, consider using the Non Lin reverb algorithms instead.

Parameters

01 — Mix `00-99` Blend between dry and wet signal. See Common Parameters.

02 — Volume `00-99` Output volume. See Common Parameters.

03 — Attack `1 ms-10.0 s` Determines how quickly the gated reverb rises once the incoming signal hits the trigger level. Typically the attack should be brief and should not exceed the Hold Time. Avoid using this parameter to emulate a reverse reverb envelope, since here the volume increases linearly whereas the Reverse reverb features an accelerating volume curve.

04 — Hold Time `1 ms-10.0 s` Specifies how long the reverb sustains after the retrigger point and before the release phase begins. If retriggered, the Hold Time resets and starts over.

05 — Decay `0.20-100.0 sec` Adjusts the decay rate in a manner similar to the Reverse Reverb algorithm. Typically this is set quite high. Although the Reverse Reverb does not expose a decay rate parameter, the DEEP/4 makes it available here for creative uses at lower settings.

06 — Release Time `1 ms-10.0 s` Defines the duration after the Hold Time expires during which the gated reverb fades out. These times are usually kept very short.

07 — Trigger Threshold `-96 to +00 dB` Establishes the signal level that activates the gated reverb. When the incoming signal reaches this value, the gated reverb starts. Higher settings demand a louder input signal. Choose the lowest practical value for your source material without introducing false triggers.

08 — Retrigger Threshold `-96 to +00 dB` Determines the level at which the gated reverb reactivates. For a precise Hold Time that starts at the beginning of your source, set this above the incoming signal level to prevent retriggering. Once the trigger threshold has been crossed, the gated reverb is running. Each time the signal reaches the retrigger threshold, the Hold Time restarts. If this level sits below the incoming signal, the gated reverb will continuously retrigger. Combined with a high Decay Rate (parameter 05), this lends a cavernous character to percussion.

09 — HF Damping `00-99` Governs how rapidly high frequencies attenuate during the reverb decay. Raising this value progressively removes more high frequency content. A setting of 00 is recommended.

10 — Diffusion 1 `00-99` Spreads transients to soften and diffuse the sound. At lower values, percussive sounds appear as a sequence of distinct echoes, while higher values increase the smearing for a smoother result. A starting value around 50 is suggested.

11 — Diffusion 2 `00-99` Works in series with Gated Diffusion 1 and operates the same way, but targets lower frequency content. A starting value around 50 is suggested.

12 — Decay Definition `00-99` Governs how quickly echo density accumulates within the reverb tail. Setting this too high may cause echo density to build faster than the decay rate allows. As a guideline, Definition should remain below the Decay Rate. Values between 25 and 50 are recommended.

13 — Slapback `0-500 ms` Sets the delay time of an internal dry stereo signal to produce a slapback effect. Typically the slapback is set equal to or longer than the Hold Time (parameter 04) to create a reverse-style effect.

14 — Slapback Level `00-99` Sets the volume of the slapback (internal dry) signal. A value of 00 silences the slapback entirely.

15 — Early Reflections 1 `-99 to +99` Adjusts the level of the first early reflection. Reducing these levels produces a wetter character.

16 — Early Reflections 2 `-99 to +99` Adjusts the level of the second early reflection.

17 — Early Reflections 3 `-99 to +99` Adjusts the level of the third early reflection.

18 — Early Reflections 4 `-99 to +99` Adjusts the level of the fourth early reflection.

19 — Left/Right Balance `-99 to +99` Positions the gated reverb signal in the stereo field. A value of -99 pans fully left, +99 pans fully right, and +00 centers the reverb in the stereo image.

20-27 — Mod Parameters `--` See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Hall Reverb

Hall Reverb simulates a spacious acoustic environment, producing dense, rich reverberation. It uses the same signal routing architecture as the Small Room and Large Room algorithms, but is voiced for broader, more expansive spaces.

Description

The input signal passes through a low pass filter and then enters diffusers that spread the signal. From there it feeds into a larger decay diffuser called Definition, where it is dispersed over time to form the reverb tail. Left and right taps from the Definition are sent to the output, generating a synthesized stereo field. A feedback path from the Definition runs through a low pass filter and into a low frequency decay stage that shapes how quickly low frequencies diminish. An additional parameter at this point governs the overall decay time for both left and right channels, and these signals are fed back into the Definition. Two echo taps situated between the diffuser and the Definition can be routed either to the output directly or back through the Definition. A separate dry signal path runs straight from the input to the output, governed by the mix parameter (01).

These reverbs tend to sound best when blending wet and dry signal together.

Parameters

01 — Mix `00-99` Blend between dry and wet signal. See Common Parameters.

02 — Volume `00-99` Output volume. See Common Parameters.

03 — Decay `0.70-250.0 sec` Sets how long the reverberation takes to fade to near-silence after the input stops. Longer values are well-suited to the hall reverb character.

04 — Predelay Time `0-450 ms` Determines the delay before the input signal reaches the reverb engine. Larger values introduce a longer gap before reverberation begins.

05 — LF DecayTime `-99 to +99` Functions as a tonal shaping control. Positive values emphasize low frequencies and extend their decay; negative values attenuate them.

06 — HF Damping `00-99` Determines how quickly high frequencies are absorbed during the decay. In natural reverberant spaces, high frequency energy is typically absorbed by surrounding surfaces. Raising this value progressively filters out more treble content.

07 — HF Bandwidth `01-99` Operates as a low pass filter on the reverb input, determining how much high frequency material enters the effect. Higher values permit more treble to pass through. Behaves much like a tone knob on a guitar.

08 — Diffusion1 `00-99` Spreads input transients to create a smoother, more diffuse sound. Lower settings cause percussive inputs to emerge as individual echo repetitions, while higher settings blend them together for a more seamless result. A value of 50 is a good starting point.

09 — Diffusion2 `00-99` Functions like Diffusion1 and is placed in series with it, but targets lower frequency content. Try varying the balance between the two diffuser controls to find the ideal setting for your source.

10 — Decay Definition `00-99` Determines the rate of echo density buildup over time. Pushing this too high can cause density to accumulate faster than the decay dissipates it. As a guideline, keep Definition at or below the sum of LF Decay Time and Decay Time.

11 — Detune Rate `00-99` Sets the LFO speed of the pitch modulation applied within the reverb tail. This detuning introduces a gentle oscillating pitch shift that makes the decay sound more organic by disrupting standing resonances.

12 — Detune Depth `00-99` Sets the intensity of the detuning, determining how far the pitch shifts. Low settings produce a metallic quality. Certain sources benefit from minimal values, while others sound more natural with deeper modulation.

13 — Primary Send `-99 to +99` Sets the level of the diffused input signal feeding into the reverb Definition stage.

14 — Ref 1 Time `0-120 ms` Sets the delay for the first pre-echo (early reflection). Pre-echoes represent the initial sound bouncing off walls or other reflective surfaces. Higher values push this diffused signal further out in time.

15 — Ref 1 Level `00-99` Sets the amplitude of the first pre-echo. This level governs the echo feed into the Definition.

16 — Ref 1 Send `00-99` Sets the amplitude of the first pre-echo routed straight to the output.

17 — Ref 2 Time `0-120 ms` Sets the delay for the second pre-echo.

18 — Ref 2 Level `00-99` Sets the amplitude of the second pre-echo. Since sound loses energy with each surface reflection, avoid setting this too high if you want a realistic echo character.

19 — Ref 2 Send `00-99` Sets the amplitude of the second pre-echo routed straight to the output.

20 — Position Balance (1) `-99 to +99` Emulates the perceived depth of the room/hall. Imagine a microphone positioned near the front of the space. Higher values bring the sound closer to the front; raising the deeper parameters (21, 22) instead suggests a more distant, wetter room/hall.

21 — Position Balance (2) `-99 to +99` Second depth microphone position, mid-distance.

22 — Position Balance (3) `-99 to +99` Third depth microphone position, farthest from the front.

23–30 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Large Plate

Large Plate emulates a larger-format plate reverb unit. Plate reverbs capture vibrations from a suspended metal sheet to generate a characteristically metallic reverberation. Large plate effects are frequently employed to enrich vocal performances.

Description

The input signal passes directly through diffusers that spread the audio. It then feeds into a larger decay diffuser called Definition, where it disperses over time to form the reverb tail. The signal routes to the output and through a low pass filter. A parameter governs the decay duration for both left and right channels, and the resulting signal feeds back into the Definition. A separate dry signal path runs from input to output, managed by the mix parameter (01).

Despite the "small" or "large" naming, the parameters perform the same role in both plate algorithms. What distinguishes the large and small plate reverbs are the fixed internal component values (not user-accessible). Extended decay settings produce excellent results with these algorithms.

Parameters

01 — Mix `00-99` Blend between dry and wet signal. See Common Parameters.

02 — Volume `00-99` Output volume. See Common Parameters.

03 — Decay `0.40-140.0 sec` Sets how long the reverberation takes to fade away. Extended decay values produce particularly good results with these algorithms.

04 — Predelay Time `0-430 ms` Determines the delay before the input signal reaches the plate reverb. A value of 0 means no delay is applied.

05 — HF Damping `00-99` Raising this value progressively removes high frequency content. Higher settings produce a sharper decay cutoff. This parameter drives a low pass filter placed in series with the decay path inside the Definition.

06 — HF Bandwidth `01-99` Functions as a low pass filter on the plate reverb output, shaping the treble content present in the signal. Higher values let more high frequencies through, yielding a brighter, more ringing tone. Routing a mod controller across a wide range can produce noteworthy effects.

07 — Diffusion 1 `00-99` Spreads the input signal to produce a smoother sound. At lower values, percussive inputs emerge as individual discrete echoes, while higher values increase the smearing and make those echoes less distinct.

08 — Diffusion 2 `00-99` Works in series with Diffusion 1 and targets lower frequency content. Plate reverbs tend toward a metallic character, and the diffusers help to spread the signal and reduce that metallic quality.

09 — Decay Definition `00-99` Determines how quickly echo density accumulates over time. Excessive values may cause density to build faster than the decay allows. Aim for the highest setting that produces good results with your particular source material.

10 — Early Ref Level 1 `-99 to +99` Adjusts the amplitude of the first early reflection. Reducing these levels yields a wetter character. All four reflection taps are positioned near the Definition input.

11 — Early Ref Level 2 `-99 to +99` Adjusts the amplitude of the second early reflection.

12 — Early Ref Level 3 `-99 to +99` Adjusts the amplitude of the third early reflection.

13 — Early Ref Level 4 `-99 to +99` Adjusts the amplitude of the fourth early reflection.

14 — Left/Right Balance `-99 to +99` Positions the plate reverb signal in the stereo field. A value of -99 pans fully left, +99 pans fully right, and +00 centers the reverb in the stereo image.

15-22 — Mod Parameters `--` See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Large Room

Large Room offers a bigger acoustic space than Small Room, while still delivering natural ambience. It uses the same signal routing architecture as the Small Room and Hall Reverb algorithms, but is voiced for a larger room environment.

Description

The input signal passes through a low pass filter and then enters diffusers that spread the audio. It feeds into a larger decay diffuser called Definition, where it disperses over time to form the reverb tail. Left and right taps from the Definition feed the output to produce a synthesized stereo field. A feedback path from the Definition passes through a low pass filter into a low frequency decay stage that shapes how quickly the bass frequencies diminish. An additional parameter at this point governs the overall decay duration for both channels, and these signals feed back into the Definition. Two echo taps between the diffuser and the Definition can route either to the output directly or back through the Definition. A separate dry signal path runs from input to output, managed by the mix parameter (01).

These reverbs perform best when blending wet and dry signal. For the room algorithms, high Decay values are generally not advisable since they tend to produce an unrealistic, endless sustain. Because actual rooms rarely exhibit long decay times, keeping this value modest yields the most convincing results.

Parameters

01 — Mix `00-99` Blend between dry and wet signal. See Common Parameters.

02 — Volume `00-99` Output volume. See Common Parameters.

03 — Decay `0.20-150.0 sec` Sets how long the reverberation takes to fade to near-silence after the input stops. For room algorithms, lower values are advisable to prevent an unrealistic, endless sustain.

04 — Predelay Time `0-450 ms` Determines the delay before the original signal reaches the reverb. Larger values introduce a longer gap.

05 — LF DecayTime `-99 to +99` Functions as a tonal shaping control. Positive values emphasize low frequencies and extend their decay; negative values attenuate them.

06 — HF Damping `00-99` Determines how quickly high frequencies are absorbed during the decay. In natural reverberant environments, treble energy tends to be soaked up by surrounding surfaces. Raising this value progressively filters out more high frequency content.

07 — HF Bandwidth `01-99` Operates as a low pass filter on the reverb input, determining how much treble enters the effect. Higher values let more high frequencies through. Behaves much like a tone knob on a guitar.

08 — Diffusion1 `00-99` Spreads input transients to create a smoother, more diffuse sound. Lower settings cause percussive inputs to emerge as individual echo repetitions, while higher settings merge them for a more seamless result. A value of 50 is a good starting point.

09 — Diffusion2 `00-99` Functions like Diffusion1 and is placed in series with it, but targets lower frequency content. Try varying the balance between the two diffuser controls to suit your source material.

10 — Decay Definition `00-99` Determines the rate of echo density buildup over time. Pushing this too high can cause density to accumulate faster than the decay dissipates. As a guideline, keep Definition at or below the sum of LF Decay Time and Decay Time.

11 — Detune Rate `00-99` Sets the LFO speed of the pitch modulation introduced into the reverb tail. This detuning adds a gentle oscillating pitch shift that makes the decay sound more organic by disrupting standing resonances.

12 — Detune Depth `00-99` Sets the intensity of the detuning, determining how far the pitch shifts. Low settings produce a metallic quality. Certain sources benefit from minimal values, while others sound more natural with deeper modulation.

13 — Primary Send `-99 to +99` Sets the level of the diffused input signal feeding into the reverb Definition stage.

14 — Ref 1 Time `0-120 ms` Sets the delay for the first pre-echo (early reflection). Pre-echoes represent the initial sound bouncing back from walls or other reflective surfaces. Higher values push the diffused signal further out in time.

15 — Ref 1 Level `00-99` Sets the amplitude of the first pre-echo. This level governs the echo feed into the Definition.

16 — Ref 1 Send `00-99` Sets the amplitude of the first pre-echo routed straight to the output.

17 — Ref 2 Time `0-120 ms` Sets the delay for the second pre-echo.

18 — Ref 2 Level `00-99` Sets the amplitude of the second pre-echo. Since sound loses energy with each surface bounce, avoid setting this too high if you want a realistic echo character.

19 — Ref 2 Send `00-99` Sets the amplitude of the second pre-echo routed straight to the output.

20 — Position Balance (1) `-99 to +99` Emulates the perceived depth of the room. Imagine a microphone near the front of the space. Higher values bring the sound closer to the front; raising the deeper parameters (21, 22) instead suggests a more distant, wetter room.

21 — Position Balance (2) `-99 to +99` Second depth microphone position, mid-distance.

22 — Position Balance (3) `-99 to +99` Third depth microphone position, farthest from the front.

23-30 — Mod Parameters `--` See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Non Lin Reverb I

Non Lin I is tailored for shorter-duration effects (roughly 0.5 seconds). Non Lin reverbs are versatile enough to produce blooming reverb, gated reverb, reverse reverb, and early reflection effects.

Description

Non linear reverbs do not generate a conventional exponentially decaying reverb tail. Unlike the Hall, Room, and Plate reverbs, Non Lin I, II, and III route the input signal through the reverb diffusers just once. Because of this single-pass behavior, these diffusers are labeled Density rather than Definition (the term used in the other reverb types). Density governs the total amount of echo density, in contrast to the rate at which echo density increases. Other reverb algorithms offer only limited early reflection shaping. For greater flexibility, try running these algorithms in series or parallel with other reverbs to emphasize early reflections. The Non Lin reverbs intentionally add coloration to the processed sound.

The input signal passes directly through a diffuser that spreads the audio. It then enters a decay diffuser called Density, where it disperses over time. Inside the Density stage, the signal passes through a high frequency damper before routing to the output. After the Density, the signal also passes through a low pass filter. Two echo taps sit between the diffuser and the Density. A separate dry signal path runs from input to output, managed by the mix parameter (01).

Parameters

01 — Mix `00-99` Blend between dry and wet signal. See Common Parameters.

02 — Volume `00-99` Output volume. See Common Parameters.

03 — Envelope Level 1 `00-99` Sets the output tap level at sequential points in time across the Density, from input to output. Envelope Level 1 is tapped immediately after the diffusers and before the echoes. If this tap is unwanted, set Envelope Level 1 to 00. Keeping the average Envelope Level below 45 is recommended to avoid overdriving the reverb.

04 — Envelope Level 2 `00-99` Sets the second output tap level.

05 — Envelope Level 3 `00-99` Sets the third output tap level.

06 — Envelope Level 4 `00-99` Sets the fourth output tap level.

07 — Envelope Level 5 `00-99` Sets the fifth output tap level.

08 — Envelope Level 6 `00-99` Sets the sixth output tap level.

09 — Envelope Level 7 `00-99` Sets the seventh output tap level.

10 — Envelope Level 8 `00-99` Sets the eighth output tap level. Envelope Levels 8 and 9 sit at the very tail of the Density; pushing these too high may produce unwanted ringing. These taps also tend to sound quite dry.

11 — Envelope Level 9 `00-99` Sets the ninth output tap level. Adjust all nine tap levels together to sculpt the envelope shape for your application.

12 — NonLin HF Damping `00-99` Positioned inside the Density stage, this parameter determines how much high frequency energy gets filtered out.

13 — NonLin HF Bandwidth `01-99` Functions as a low pass filter on the output signal, governing how much treble is audible. Higher values allow more high frequencies through. This behaves like a tone knob on a guitar.

14 — NonLin Diffusion1 `00-99` Spreads higher-frequency input transients. Higher values are recommended for smoother percussion sounds. Very low settings produce a strongly repetitive, echo-like character. Diffusion1 and 2 operate within each diffuser block.

15 — NonLin Diffusion2 `00-99` Works like Diffusion1 but targets lower frequencies. A value of 50 roughly represents an equal blend of dry and diffused signal and serves as a useful starting point.

16 — NonLin Density 1 `00-99` Determines the echo count.

17 — NonLin Density 2 `00-99` Determines the echo count in a lower frequency range. For the smoothest results, Density 2 is typically set lower than Density 1.

18 — NonLin Primary Send `-99 to +99` Sets the level of the diffused input signal, which arrives nearly instantaneously relative to the input. This signal feeds directly into the Density at the specified level.

19 — Reflection 1 Time `0-600 ms` Determines the delay before the first pre-echo enters the Density. Pre-echoes are reflections bouncing back from walls or other hard surfaces.

20 — Reflection 1 Send `-99 to +99` Sets the amplitude of the first pre-echo.

21 — Reflection 2 Time `0-600 ms` Determines the delay before the second pre-echo enters the Density.

22 — Reflection 2 Send `-99 to +99` Sets the amplitude of the second pre-echo. Try both positive and negative values on all echoes to alter the tonal character of the output.

23 — Left/Right Balance `-99 to +99` Positions the reverb signal in the stereo field. A value of -99 pans fully left, +99 pans fully right, and +00 centers the reverb in the stereo image.

24-31 — Mod Parameters `--` See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Non Lin Reverb II

Non Lin II provides a longer duration of approximately 1.5 seconds. Non Lin reverbs are versatile enough to produce blooming reverb, gated reverb, reverse reverb, and early reflection effects.

Description

Non linear reverbs do not generate a conventional exponentially decaying reverb tail. Unlike the Hall, Room, and Plate reverbs, Non Lin I, II, and III route the input signal through the reverb diffusers just once. Because of this single-pass behavior, these diffusers are labeled Density rather than Definition (the term used in the other reverb types). Density governs the total amount of echo density, in contrast to the rate at which echo density increases. Other reverb algorithms offer only limited early reflection shaping. For greater flexibility, try running these algorithms in series or parallel with other reverbs to emphasize early reflections. The Non Lin reverbs intentionally add coloration to the processed sound.

The input signal passes directly through a diffuser that spreads the audio. It then enters a decay diffuser called Density, where it disperses over time. Inside the Density stage, the signal passes through a high frequency damper before routing to the output. After the Density, the signal also passes through a low pass filter. Two echo taps sit between the diffuser and the Density. A separate dry signal path runs from input to output, managed by the mix parameter (01).

Non Lin II shares its parameter set with Non Lin I and Non Lin III, but uses different Reflection Time ranges (see parameters 19 and 21 below).

Parameters

01 — Mix `00-99` Blend between dry and wet signal. See Common Parameters.

02 — Volume `00-99` Output volume. See Common Parameters.

03 — Envelope Level 1 `00-99` Sets the output tap level at sequential points in time across the Density, from input to output. Envelope Level 1 is tapped immediately after the diffusers and before the echoes. If this tap is unwanted, set Envelope Level 1 to 00. Keeping the average Envelope Level below 45 is recommended to avoid overdriving the reverb.

04 — Envelope Level 2 `00-99` Sets the second output tap level.

05 — Envelope Level 3 `00-99` Sets the third output tap level.

06 — Envelope Level 4 `00-99` Sets the fourth output tap level.

07 — Envelope Level 5 `00-99` Sets the fifth output tap level.

08 — Envelope Level 6 `00-99` Sets the sixth output tap level.

09 — Envelope Level 7 `00-99` Sets the seventh output tap level.

10 — Envelope Level 8 `00-99` Sets the eighth output tap level. Envelope Levels 8 and 9 sit at the very tail of the Density; pushing these too high may produce unwanted ringing. These taps also tend to sound quite dry.

11 — Envelope Level 9 `00-99` Sets the ninth output tap level. Adjust all nine tap levels together to sculpt the envelope shape for your application.

12 — NonLin HF Damping `00-99` Positioned inside the Density stage, this parameter determines how much high frequency energy gets filtered out.

13 — NonLin HF Bandwidth `01-99` Functions as a low pass filter on the output signal, governing how much treble is audible. Higher values allow more high frequencies through. This behaves like a tone knob on a guitar.

14 — NonLin Diffusion1 `00-99` Spreads higher-frequency input transients. Higher values are recommended for smoother percussion sounds. Very low settings produce a strongly repetitive, echo-like character. Diffusion1 and 2 operate within each diffuser block.

15 — NonLin Diffusion2 `00-99` Works like Diffusion1 but targets lower frequencies. A value of 50 roughly represents an equal blend of dry and diffused signal and serves as a useful starting point.

16 — NonLin Density 1 `00-99` Determines the echo count.

17 — NonLin Density 2 `00-99` Determines the echo count in a lower frequency range. For the smoothest results, Density 2 is typically set lower than Density 1.

18 — NonLin Primary Send `-99 to +99` Sets the level of the diffused input signal, which arrives nearly instantaneously relative to the input. This signal feeds directly into the Density at the specified level.

19 — Reflection 1 Time `0-85 ms` Determines the delay before the first pre-echo enters the Density. Pre-echoes are reflections bouncing back from walls or other hard surfaces.

20 — Reflection 1 Send `-99 to +99` Sets the amplitude of the first pre-echo.

21 — Reflection 2 Time `0-85 ms` Determines the delay before the second pre-echo enters the Density.

22 — Reflection 2 Send `-99 to +99` Sets the amplitude of the second pre-echo. Try both positive and negative values on all echoes to alter the tonal character of the output.

23 — Left/Right Balance `-99 to +99` Positions the reverb signal in the stereo field. A value of -99 pans fully left, +99 pans fully right, and +00 centers the reverb in the stereo image.

24-31 — Mod Parameters `--` See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Non Lin Reverb III

Non Lin III is sonically close to Non Lin I but features reduced stereo motion, making it particularly well-suited for drum tracks. Non Lin reverbs are versatile enough to produce blooming reverb, gated reverb, reverse reverb, and early reflection effects.

Description

Non linear reverbs do not generate a conventional exponentially decaying reverb tail. Unlike the Hall, Room, and Plate reverbs, Non Lin I, II, and III route the input signal through the reverb diffusers just once. Because of this single-pass behavior, these diffusers are labeled Density rather than Definition (the term used in the other reverb types). Density governs the total amount of echo density, in contrast to the rate at which echo density increases. Other reverb algorithms offer only limited early reflection shaping. For greater flexibility, try running these algorithms in series or parallel with other reverbs to emphasize early reflections. The Non Lin reverbs intentionally add coloration to the processed sound.

The input signal passes directly through a diffuser that spreads the audio. It then enters a decay diffuser called Density, where it disperses over time. Inside the Density stage, the signal passes through a high frequency damper before routing to the output. After the Density, the signal also passes through a low pass filter. Two echo taps sit between the diffuser and the Density. A separate dry signal path runs from input to output, managed by the mix parameter (01).

Non Lin III uses the same parameters as Non Lin I, including identical Reflection Time ranges (0-600 ms).

Parameters

01 — Mix `00-99` Blend between dry and wet signal. See Common Parameters.

02 — Volume `00-99` Output volume. See Common Parameters.

03 — Envelope Level 1 `00-99` Sets the output tap level at sequential points in time across the Density, from input to output. Envelope Level 1 is tapped immediately after the diffusers and before the echoes. If this tap is unwanted, set Envelope Level 1 to 00. Keeping the average Envelope Level below 45 is recommended to avoid overdriving the reverb.

04 — Envelope Level 2 `00-99` Sets the second output tap level.

05 — Envelope Level 3 `00-99` Sets the third output tap level.

06 — Envelope Level 4 `00-99` Sets the fourth output tap level.

07 — Envelope Level 5 `00-99` Sets the fifth output tap level.

08 — Envelope Level 6 `00-99` Sets the sixth output tap level.

09 — Envelope Level 7 `00-99` Sets the seventh output tap level.

10 — Envelope Level 8 `00-99` Sets the eighth output tap level. Envelope Levels 8 and 9 sit at the very tail of the Density; pushing these too high may produce unwanted ringing. These taps also tend to sound quite dry.

11 — Envelope Level 9 `00-99` Sets the ninth output tap level. Adjust all nine tap levels together to sculpt the envelope shape for your application.

12 — NonLin HF Damping `00-99` Positioned inside the Density stage, this parameter determines how much high frequency energy gets filtered out.

13 — NonLin HF Bandwidth `01-99` Functions as a low pass filter on the output signal, governing how much treble is audible. Higher values allow more high frequencies through. This behaves like a tone knob on a guitar.

14 — NonLin Diffusion1 `00-99` Spreads higher-frequency input transients. Higher values are recommended for smoother percussion sounds. Very low settings produce a strongly repetitive, echo-like character. Diffusion1 and 2 operate within each diffuser block.

15 — NonLin Diffusion2 `00-99` Works like Diffusion1 but targets lower frequencies. A value of 50 roughly represents an equal blend of dry and diffused signal and serves as a useful starting point.

16 — NonLin Density 1 `00-99` Determines the echo count.

17 — NonLin Density 2 `00-99` Determines the echo count in a lower frequency range. For the smoothest results, Density 2 is typically set lower than Density 1.

18 — NonLin Primary Send `-99 to +99` Sets the level of the diffused input signal, which arrives nearly instantaneously relative to the input. This signal feeds directly into the Density at the specified level.

19 — Reflection 1 Time `0-600 ms` Determines the delay before the first pre-echo enters the Density. Pre-echoes are reflections bouncing back from walls or other hard surfaces.

20 — Reflection 1 Send `-99 to +99` Sets the amplitude of the first pre-echo.

21 — Reflection 2 Time `0-600 ms` Determines the delay before the second pre-echo enters the Density.

22 — Reflection 2 Send `-99 to +99` Sets the amplitude of the second pre-echo. Try both positive and negative values on all echoes to alter the tonal character of the output.

23 — Left/Right Balance `-99 to +99` Positions the reverb signal in the stereo field. A value of -99 pans fully left, +99 pans fully right, and +00 centers the reverb in the stereo image.

24-31 — Mod Parameters `--` See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Reverse Reverb

Reverse Reverb generates a reverberation that swells upward over time, emulating a backwards sound with a maximum length of several seconds. Its internal architecture is derived from the Plate Reverb topology.

Description

When audio enters this algorithm, the underlying plate reverb engine activates almost instantly, and the output volume then ramps upward. This algorithm fires only once per trigger event. The Reverse Reverb is activated when the input signal exceeds a user-defined threshold. After triggering, the reverse envelope runs to completion and ignores any further trigger events. For a reverse effect that responds to repeated triggers, use the Reverse Reverb II instead.

Parameters

01 — Mix `00-99` Blend between dry and wet signal. See Common Parameters.

02 — Volume `00-99` Output volume. See Common Parameters.

03 — Envelope Hold Time `1 ms-10.0 s` Determines how long the reverse effect sustains after being triggered. As a guideline, avoid setting the hold time significantly longer than the attack time.

04 — Envelope Attack `1 ms-10.0 s` Defines the duration over which the reverb swells to full level. Setting this shorter than the hold time (parameter 03) is recommended.

05 — Envelope Release `1 ms-10.0 s` Sets the fade-out duration once the hold time has elapsed. This is usually kept very short. Lower values produce a sharper cutoff.

06 — Trigger Threshold `-96 to +00 dB` Choose the lowest practical value for your source material. Avoid going so low that false triggers occur. Once the input signal rises above this threshold, the reverse envelope begins.

07 — HF Damping `00-99` Works best at low settings. Performs the same role as in the Plate Reverb, progressively filtering out high frequency content. For the most natural-sounding reverse effect, a setting of 00 is recommended.

08 — Diffusion 1 `00-99` Spreads the input signal to produce a smoother reverb. This parameter targets the high frequency range. Higher values are recommended for percussive sources.

09 — Diffusion 2 `00-99` Works in series with Diffusion 1 and targets lower frequency content.

10 — Decay Definition `00-99` Determines how quickly echo density accumulates over time. Excessive values may cause density to build faster than the decay rate allows. This can also be exploited as a creative effect.

11 — Slapback `0-530 ms` Sets the delay time of an internal dry signal to produce a slapback effect. This helps simulate a backwards reverb by placing the dry signal at the tail end. For best results, set Mix (parameter 01) fully wet (99). As a guideline, match this roughly to the Envelope Hold Time (parameter 03).

12 — Slapback Level `00-99` Sets the volume of the slapback (internal dry) signal. A value of 00 silences the slapback entirely.

13-20 — Mod Parameters `--` See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Reverse Reverb II

Reverse Reverb II is functionally the same as Reverse Reverb, with one key distinction: this algorithm will retrigger based on a user-defined input signal threshold. After triggering, the reverse envelope plays through to completion unless a new input signal causes it to restart. For a reverse effect that does not retrigger, use the original Reverse Reverb algorithm.

Description

Like the original Reverse Reverb, this algorithm generates a reverberation that swells upward, emulating a backwards sound. The critical difference is retriggering support: whenever the input signal crosses the trigger threshold again, the envelope resets and starts over. This variant also introduces a Pre-Trigger Memory parameter that captures transients occurring just before the trigger point, feeding them into the reverse reverb tank for improved sonic quality.

Parameters

01 — Mix `00-99` Blend between dry and wet signal. See Common Parameters.

02 — Volume `00-99` Output volume. See Common Parameters.

03 — Envelope Hold Time `1 ms-10.0 s` Determines how long the reverse effect sustains after being triggered. As a guideline, avoid setting the hold time significantly longer than the attack time.

04 — Attack `1 ms-10.0 s` Defines the duration over which the reverb swells to full level. Setting this shorter than the hold time (parameter 03) is recommended.

05 — Release `1 ms-10.0 s` Sets the fade-out duration once the hold time has elapsed. This is usually kept very short. Lower values produce a sharper cutoff.

06 — Trigger Threshold `-96 to +00 dB` Choose the lowest practical value for your source material. Avoid going so low that false triggers occur. Once the input signal rises above this threshold, the reverse envelope begins.

07 — Pre-Trigger Memory `0-530 ms` Captures transients that arrive before the trigger event. This parameter significantly affects sound quality. The user sets how much pre-trigger audio gets fed into the reverse reverb tank.

08 — HF Damping `00-99` Works best at low settings. Its role is to progressively filter out high frequency content. For the most natural-sounding reverse effect, a setting of 00 is recommended.

09 — Diffusion 1 `00-99` Spreads the input signal to produce a smoother reverb. This parameter targets the high frequency range. Higher values are recommended for percussive sources.

10 — Diffusion 2 `00-99` Works in series with Diffusion 1 and targets lower frequency content.

11 — Decay Definition `00-99` Determines how quickly echo density accumulates over time. Excessive values may cause density to build faster than the decay rate allows. This can also be exploited as a creative effect.

12-19 — Mod Parameters `--` See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Small Plate

Small Plate delivers a compact, focused plate reverb. Plate reverbs capture vibrations from a suspended metal sheet to generate a characteristically metallic reverberation. Small plate effects are most commonly used in studio settings on drums and percussion.

Description

The input signal passes directly through diffusers that spread the audio. It then feeds into a larger decay diffuser called Definition, where it disperses over time to form the reverb tail. The signal routes to the output and through a low pass filter. A parameter governs the decay duration for both left and right channels, and the resulting signal feeds back into the Definition. A separate dry signal path runs from input to output, managed by the mix parameter (01).

Despite the “small” or “large” naming, the parameters perform the same role in both plate algorithms. What distinguishes them are the fixed internal component values (not user-accessible). Percussion works particularly well with the Small Plate. Extended decay settings produce excellent results with these algorithms.

Parameters

01 — Mix `00-99` Blend between dry and wet signal. See Common Parameters.

02 — Volume `00-99` Output volume. See Common Parameters.

03 — Decay `0.20-100.0 sec` Sets how long the reverberation takes to fade away. Percussion works particularly well with the Small Plate. Extended decay values produce excellent results with these algorithms.

04 — Predelay Time `0-500 ms` Determines the delay before the input signal reaches the plate reverb. A value of 0 means no delay is applied.

05 — HF Damping `00-99` Raising this value progressively removes high frequency content. Higher settings produce a sharper decay cutoff. This parameter drives a low pass filter placed in series with the decay path inside the Definition.

06 — HF Bandwidth `01-99` Functions as a low pass filter on the plate reverb output, shaping the treble content present in the signal. Higher values let more high frequencies through, yielding a brighter, more ringing tone. Routing a mod controller across a wide range can produce noteworthy effects.

07 — Diffusion 1 `00-99` Spreads the input signal to produce a smoother sound. At lower values, percussive inputs emerge as individual discrete echoes, while higher values increase the smearing and make those echoes less distinct.

08 — Diffusion 2 `00-99` Works in series with Diffusion 1 and targets lower frequency content. Plate reverbs tend toward a metallic character, and the diffusers help to spread the signal and reduce that metallic quality.

09 — Decay Definition `00-99` Determines how quickly echo density accumulates over time. Excessive values may cause density to build faster than the decay allows. Aim for the highest setting that produces good results with your particular source material.

10 — Early Ref Level 1 `-99 to +99` Adjusts the amplitude of the first early reflection. Reducing these levels yields a wetter character. All four reflection taps are positioned near the Definition input.

11 — Early Ref Level 2 `-99 to +99` Adjusts the amplitude of the second early reflection.

12 — Early Ref Level 3 `-99 to +99` Adjusts the amplitude of the third early reflection.

13 — Early Ref Level 4 `-99 to +99` Adjusts the amplitude of the fourth early reflection.

14 — Left/Right Balance `-99 to +99` Positions the plate reverb signal in the stereo field. A value of -99 pans fully left, +99 pans fully right, and +00 centers the reverb in the stereo image.

15-22 — Mod Parameters `--` See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Small Room

Small Room generates natural ambience by simulating a compact acoustic space. It uses the same signal routing architecture as the Large Room and Hall Reverb algorithms.

Description

The input signal passes through a low pass filter and then enters diffusers that spread the audio. It feeds into a larger decay diffuser called Definition, where it disperses over time to form the reverb tail. Left and right taps from the Definition feed the output to produce a synthesized stereo field. A feedback path from the Definition passes through a low pass filter into a low frequency decay stage that shapes how quickly the bass frequencies diminish. An additional parameter at this point governs the overall decay duration for both channels, and these signals feed back into the Definition. Two echo taps between the diffuser and the Definition can route either to the output directly or back through the Definition. A separate dry signal path runs from input to output, managed by the mix parameter (01).

These reverbs perform best when blending wet and dry signal. For the room algorithms, high Decay values are generally not advisable since they tend to produce an unrealistic, endless sustain. Because actual rooms rarely exhibit long decay times, keeping this value modest yields the most convincing results.

Parameters

01 — Mix `00-99` Blend between dry and wet signal. See Common Parameters.

02 — Volume `00-99` Output volume. See Common Parameters.

03 — Decay `0.20-100.0 sec` Sets how long the reverberation takes to fade to near-silence after the input stops. For room algorithms, lower values are advisable to prevent an unrealistic, endless sustain.

04 — Predelay Time `0-450 ms` Determines the delay before the original signal reaches the reverb. Larger values introduce a longer gap.

05 — LF DecayTime `-99 to +99` Functions as a tonal shaping control. Positive values emphasize low frequencies and extend their decay; negative values attenuate them.

06 — HF Damping `00-99` Determines how quickly high frequencies are absorbed during the decay. In natural reverberant environments, treble energy tends to be soaked up by surrounding surfaces. Raising this value progressively filters out more high frequency content.

07 — HF Bandwidth `01-99` Operates as a low pass filter on the reverb input, determining how much treble enters the effect. Higher values let more high frequencies through. Behaves much like a tone knob on a guitar.

08 — Diffusion1 `00-99` Spreads input transients to create a smoother, more diffuse sound. Lower settings cause percussive inputs to emerge as individual echo repetitions, while higher settings merge them for a more seamless result. A value of 50 is a good starting point.

09 — Diffusion2 `00-99` Functions like Diffusion1 and is placed in series with it, but targets lower frequency content. Try varying the balance between the two diffuser controls to suit your source material.

10 — Decay Definition `00-99` Determines the rate of echo density buildup over time. Pushing this too high can cause density to accumulate faster than the decay dissipates. As a guideline, keep Definition at or below the sum of LF Decay Time and Decay Time.

11 — Detune Rate `00-99` Sets the LFO speed of the pitch modulation introduced into the reverb tail. This detuning adds a gentle oscillating pitch shift that makes the decay sound more organic by disrupting standing resonances.

12 — Detune Depth `00-99` Sets the intensity of the detuning, determining how far the pitch shifts. Low settings produce a metallic quality. Certain sources benefit from minimal values, while others sound more natural with deeper modulation.

13 — Primary Send `-99 to +99` Sets the level of the diffused input signal feeding into the reverb Definition stage.

14 — Ref 1 Time `0-120 ms` Sets the delay for the first pre-echo (early reflection). Pre-echoes represent the initial sound bouncing back from walls or other reflective surfaces. Higher values push the diffused signal further out in time.

15 — Ref 1 Level `00-99` Sets the amplitude of the first pre-echo. This level governs the echo feed into the Definition.

16 — Ref 1 Send `00-99` Sets the amplitude of the first pre-echo routed straight to the output.

17 — Ref 2 Time `0-120 ms` Sets the delay for the second pre-echo.

18 — Ref 2 Level `00-99` Sets the amplitude of the second pre-echo. Since sound loses energy with each surface bounce, avoid setting this too high if you want a realistic echo character.

19 — Ref 2 Send `00-99` Sets the amplitude of the second pre-echo routed straight to the output.

20 — Position Balance (1) `-99 to +99` Emulates the perceived depth of the room. Imagine a microphone near the front of the space. Higher values bring the sound closer to the front; raising the deeper parameters (21, 22) instead suggests a more distant, wetter room.

21 — Position Balance (2) `-99 to +99` Second depth microphone position, mid-distance.

22 — Position Balance (3) `-99 to +99` Third depth microphone position, farthest from the front.

23-30 — Mod Parameters `--` See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Algorithms: Delay

3.3 sec Delay 2U

3.3 sec Delay 2U is a 2-Unit algorithm leveraging two ESP chips to deliver a pristine delay exceeding 3 seconds in length. Additionally, this algorithm supports signal capture and looped playback. Through this “Instant Replay” capability, you can perform alongside a repeating passage for creative layered results. For optimal results, blend wet and dry signals using the Mix control.

Instant Replay

Parameter 07 (Delay Mode) engages the “instant replay” functionality. In Continuous mode, all incoming audio is delayed normally. In Loop mode, an “instant replay” loop can be triggered via any modulation source assigned in parameter 08 (Delay Set).

Delay Set specifies which modulation source governs the loop behavior (this parameter has no effect when Delay Mode is Continuous). A controller value above 64 initiates recording; a value below 64 switches to playback. Playback enters a muted state whenever a recording shorter than 300 ms is captured.

For infinite sustain during playback, set the regen parameter to 71. Lower regen values cause repeats to gradually decay. Higher values introduce runaway feedback and distortion, which may also be desirable when using damping. The delay pan setting also influences the effective regen amount.

Note: If the unit remains in a given state for more than 3.6 seconds, switching back to playback will restore the last valid setting (from the most recent 3.6-second window).

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — Delay Time `0-3668 ms` Configures the delay duration. Assigning a real-time modulation controller to this parameter opens up a range of creative possibilities.

04 — Delay Regen `00-99` Controls how much of the output signal is routed back to the input, which increases the number of delay repeats. At 99, the delay sustains indefinitely.

05 — Delay Pan `-99 to +99` Positions the delayed signal within the stereo field. A setting of -99 places it fully left, while +99 places it fully right.

06 — Delay Regen Damping `00-99` Governs the cutoff frequency of a low-pass filter applied to the feedback path, attenuating high frequencies in the regenerated signal. Increasing this value produces greater damping.

07 — Delay Mode `Continuous, Loop/Muted, Loop/Record, Loop/Replay` Engages the "instant replay" functionality. Continuous mode delays all incoming audio normally. Loop mode enables creation of an "instant replay" loop through any modulation source assigned in parameter 08.

08 — Delay Set `Off, Controllers 1-8` Specifies the modulation source that triggers and controls loop behavior. Has no effect when parameter 07 is set to Continuous. Controller values above 64 activate recording; values below 64 activate playback.

09–16 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Dual Delay

Dual Delay delivers a studio-grade, high-fidelity stereo digital delay. The algorithm divides available memory into two matched delay lines, preserving an accurate stereo image throughout the delay path. For best results, use a blend of wet and dry signal via the Mix control.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — Left Input Delay Time `0-840 ms` Sets the delay interval between the source signal and the left input delay.

04 — Left Input Delay Time (fine) `0.00-0.99 ms` Provides millisecond-level fine adjustment for the left input delay interval.

05 — Left Input Delay Regen `00-99` Controls how much of the left delay output is routed back to the input, adding successive repeats to the delay.

06 — Left Input Delay Pan `-99 to +99` Positions the left input delay within the stereo field. At -99 the signal sits fully left; at +99 it sits fully right.

07 — Right Input Delay Time `0-840 ms` Sets the delay interval between the source signal and the right input delay.

08 — Right Input Delay Time (fine) `0.00-0.99 ms` Provides millisecond-level fine adjustment for the right input delay interval.

09 — Right Input Delay Regen `00-99` Controls how much of the right delay output is routed back to the input, adding successive repeats to the delay.

10 — Right Input Delay Pan `-99 to +99` Positions the right input delay within the stereo field. At -99 the signal sits fully left; at +99 it sits fully right.

11 — Dual Delay Cross Regen `-99 to +99` Routes each delayed signal into the opposite channel's feedback path (when both delay pans are set to opposing values); the left voice feeds into the right, and the right feeds into the left. A value of +99 or -99 produces infinite delay.

12 — Dual Delay Regen Damping `00-99` Governs the cutoff frequency of a low-pass filter on the feedback path, attenuating high frequencies in the regenerated signals. Higher values apply more damping.

13–20 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Multi Tap Delay

Multi Tap Delay occupies only a single ESP chip, leaving the remaining three units available for other algorithms. It generates four independently controllable delay voices. For best results, blend wet and dry signals using the Mix control.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — MultiTap 1 Time `0-1834 ms` Configures the delay duration for the first independent voice. Try various settings to discover the ideal combination for your source material and use case. Assigning a real-time modulation controller to these parameters enables creative possibilities.

04 — MultiTap 1 Level `00-99` Sets the volume of the first delayed voice relative to the unprocessed dry signal. A value of 00 renders the delay inaudible.

05 — MultiTap 1 Regen `00-99` Controls how much of the output signal is routed back to the input, adding successive repeats. A value of 99 produces infinite delay.

06 — MultiTap 1 Pan `-99 to +99` Positions the first delay voice within the stereo field. A value of -99 places it fully left, and +99 places it fully right.

07 — MultiTap 2 Time `0-1834 ms` Configures the delay duration for the second independent voice.

08 — MultiTap 2 Level `00-99` Sets the volume of the second delayed voice relative to the unprocessed dry signal. A value of 00 renders the delay inaudible.

09 — MultiTap 2 Regen `00-99` Controls how much of the output signal is routed back to the input, adding successive repeats. A value of 99 produces infinite delay.

10 — MultiTap 2 Pan `-99 to +99` Positions the second delay voice within the stereo field. A value of -99 places it fully left, and +99 places it fully right.

11 — MultiTap 3 Time `0-1834 ms` Configures the delay duration for the third independent voice.

12 — MultiTap 3 Level `00-99` Sets the volume of the third delayed voice relative to the unprocessed dry signal. A value of 00 renders the delay inaudible.

13 — MultiTap 3 Regen `00-99` Controls how much of the output signal is routed back to the input, adding successive repeats. A value of 99 produces infinite delay.

14 — MultiTap 3 Pan `-99 to +99` Positions the third delay voice within the stereo field. A value of -99 places it fully left, and +99 places it fully right.

15 — MultiTap 4 Time `0-1834 ms` Configures the delay duration for the fourth independent voice.

16 — MultiTap 4 Level `00-99` Sets the volume of the fourth delayed voice relative to the unprocessed dry signal. A value of 00 renders the delay inaudible.

17 — MultiTap 4 Regen `00-99` Controls how much of the output signal is routed back to the input, adding successive repeats. A value of 99 produces infinite delay.

18 — MultiTap 4 Pan `-99 to +99` Positions the fourth delay voice within the stereo field. A value of -99 places it fully left, and +99 places it fully right.

19 — Regen Damping `00-99` Governs the cutoff frequency of a low-pass filter on the feedback path, attenuating high frequencies in the regenerated signals. Higher values apply more damping.

20–27 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Tempo Delay

Tempo Delay provides a stereo digital delay (comparable to MultiTap) whose timing is governed by an assignable modulation source, such as a foot switch.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — Tempo Delay Time `various` Chooses from twelve rhythmic subdivisions that define the delay rate: 1/32 note, 1/16 triplet, 1/16 note, 1/16 dotted, 1/8 triplet, 1/8 note, 1/8 dotted, 1/4 triplet, 1/4 note, 1/4 dotted, 1/2 triplet, and 1/2 note.

04 — Internal Clock Tempo `050-250 bpm` Sets the beats per minute (bpm) when the internal clock drives the tempo. This parameter has no effect if MIDI Clocks or Footswitch1 Tapping is selected in parameter 06.

05 — TempoDelay Fine Tune `-99 to +99` Provides fine adjustment of the delay timing. Lower values yield a faster rate.

06 — Tempo Control `various` Selects the tempo source: Internal clock, MIDI clocks, or FootSwitch 1 Tapping. For FootSwitch 1 to function as a controller, it must first be assigned as a DEEP/4 Controller under System*MIDI mode (parameter 45). Tap the foot switch twice (tapping quarter notes) to establish the tempo. Subsequent taps will update the tempo, since the DEEP/4 continuously averages the interval between the last two presses. This behavior is particularly useful for performances or arrangements with fluctuating tempos.

07 — Tempo Delay Regen `00-99` Controls how much of the output signal is routed back to the input, adding successive repeats to the delay.

08 — Tempo Delay Pan `-99 to +99` Positions the delayed signal within the stereo field.

09 — Tempo Delay Regen Damping `00-99` Governs the cutoff frequency of a low-pass filter on the feedback path, attenuating high frequencies in the regenerated signals. Higher values apply more damping. Lower settings are recommended.

10–17 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Algorithms: Modulation

8 Voice Chorus

8 Voice Chorus produces a rich, symphonic texture by generating eight distinct chorus voices, each driven by its own independently randomized LFO. The algorithm includes a programmable stereo delay with cross-coupled routing between the left and right chorus outputs. It excels at transforming a single instrument into the sound of a full ensemble, as no internal filtering is applied to any of the chorus voices. A Mix value of approximately 50 is a good starting point for this algorithm.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — 8V Chorus LFO Rate `00-99` Sets the linked modulation speed across all eight voices. This modulation creates a combined effect resembling simultaneous vibrato and tremolando.

04 — 8V Chorus LFO Width `00-99` Adjusts how far the vibrato pitch deviates for each individual voice.

05 — 8V Chorus Stereo Spread `00-99` Creates a synthetic stereo image. At maximum, the output is fully stereo. Middle values blend the left and right signals across both outputs. At minimum, only the left input channel appears at both outputs. Interesting stereo motion can be achieved by assigning a modulation source to this parameter.

06 — 8V Chorus Regen `00-99` Sets the level of signal routed from the chorus output back into the chorus input. Setting this to 00 removes the regeneration entirely.

07 — 8V Chorus Left Regen Time `0-800 ms` Specifies the delay duration applied to the dry (non-chorused) signal on the left channel.

08 — 8V Chorus Right Regen Time `0-800 ms` Specifies the delay duration applied to the dry (non-chorused) signal on the right channel.

09 — 8V Chorus Delay Regen `00-99` Sets the level of signal routed from the delay output back into the chorus input, producing more repeats at higher values. Setting this to 00 silences the delay entirely.

10–17 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

EQ Chorus DDL

EQ Chorus DDL pairs a parametric EQ with a chorus and digital delay. This is the classic chorus effect, built around extended delay times to produce a modulated detuning. It works beautifully on guitar, but is well worth trying on any input source.

Description

The input signal first passes through an input level trim (parameter 17) and then into a programmable EQ. From the EQ, the signal feeds the chorus, which appears directly at the output. An unchorused copy of the signal shares the same delay lines and feeds back into the chorus. A second tap from the delay line is sent to the right output. Two separate echo taps are drawn before the chorus delay line, producing unchorused echoes routed straight to the outputs. A dry path also runs directly from the input to the output, governed by the mix parameter (01).

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — Chorus LFO Rate `00-99` Governs the speed of pitch modulation that forms the chorus. This rate should be kept very slow to produce a true chorus sound.

04 — Chorus LFO Width `00-99` Governs the depth of the pitch modulation. Because the rate is typically very slow, the width is generally set quite high.

05 — Chorus Center `00-99` Sets the base delay time around which the modulation sweeps. Changing this value alters the tonal character of the chorus. This delay time is independent of the regen and echo delays.

06 — Left/Right LFO `Out-of-Phase or In-Phase` In-phase mode causes both channels to detune in unison. Out-of-phase mode makes the left channel pitch rise while the right channel pitch falls, and vice versa.

07 — Chorus Left Delay Time `0-1500 ms` Sets the left channel regen delay duration. This is unrelated to the chorus effect itself.

08 — Chorus Right Delay Time `0-1500 ms` Sets the right channel regen delay duration. This is unrelated to the chorus effect itself.

09 — Chorus Delay Regen `-99 to +99` Governs the regeneration level applied to the delay taps. The sign sets the feedback polarity.

10 — Chorus Left Echo Time `0-1500 ms` Sets the left-side echo duration. Longer times produce a more pronounced echo. Two independent echoes serve the left and right channels.

11 — Chorus Right Echo Time `0-1500 ms` Sets the right-side echo duration. Longer times produce a more pronounced echo.

12 — Chorus Echo Level `00-99` Adjusts the loudness of both left and right echoes. Raising the value increases echo volume; setting it to 00 silences the echo. At moderate levels with sustained input, this can approximate a basic reverb tail.

13 — Bass Fc `0-1000 Hz` Defines the corner frequency of the low-band shelving filter.

14 — Bass EQ Gain `-48 to +24 dB` Determines how much boost or attenuation the low shelving filter applies.

15 — Treble Fc `01KHz-16KHz` Defines the corner frequency of the high shelving filter.

16 — Treble EQ Gain `-48 to +24 dB` Determines how much boost or attenuation the high shelving filter applies.

17 — EQ Input Level Trim `-24 to +00 dB` Lets you reduce the signal level entering the EQ section to avoid clipping when boosting frequencies.

18–25 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

EQ/DDL/with LFO

EQ/DDL/with LFO provides a stereo digital delay (comparable to Dual Delay) with LFO-driven modulation across a broad range of delay times. It sounds particularly good on electric piano, but works well with any input source.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — DDL+LFO Left Delay Time `0-845 ms` Sets the interval between the input signal and the left delay output.

04 — DDL+LFO Delay Time `0-845 ms` Sets the interval between the input signal and the right delay output. Using a different value from parameter 03 can produce dotted eighth-note style rhythms.

05 — DDL+LFO LFO Rate `00-99` Governs the speed of pitch modulation generated by the LFO. Keep this rate very slow to achieve a chorus-like quality.

06 — DDL+LFO LFO Width `00-99` Governs the depth of pitch modulation. Since the rate is typically slow, the width is usually set high.

07 — Left/Right LFO `Out-of-Phase or In-Phase` In-phase mode causes both channels to detune together. Out-of-phase mode raises the left channel pitch while lowering the right, and vice versa.

08 — DDL+LFO Delay Regen `-99 to +99` Governs the regeneration level applied to the delay taps. The sign sets the feedback polarity.

09 — DDL+LFO Delay Cross Regen `-99 to +99` Routes the delayed signal from each channel to the opposite side: left feeds right, and right feeds left. At +99 or -99, the delay repeats indefinitely. Use caution – if the delay regen is already high, extreme cross regen values may cause runaway feedback.

10 — DDL+LFO Regen Damping `00-99` Adjusts the cutoff of a low-pass filter in the feedback path, controlling how much high-frequency content is removed from successive repeats. Higher values apply more damping.

11 — DDL+LFO Right Delay Input `Off or On` Turns off the direct input feed to the right delay line. The right line will still receive signal through cross regen, enabling a ping-pong delay pattern.

12 — DDL+LFO Right Output Level `00-99` Sets the output level for the right channel.

13 — Bass Fc `0-1000 Hz` Defines the corner frequency of the low-band shelving filter.

14 — Bass EQ Gain `-48 to +24 dB` Determines how much boost or attenuation the low shelving filter applies.

15 — Treble Fc `01KHz-16KHz` Defines the corner frequency of the high shelving filter.

16 — Treble EQ Gain `-48 to +24 dB` Determines how much boost or attenuation the high shelving filter applies.

17 — EQ Input Level Trim `-24 to +00 dB` Lets you reduce the signal level entering the EQ section to avoid clipping when boosting frequencies.

18–25 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

EQ/Flanger/DDL

EQ/Flanger/DDL pairs an EQ with a flanger and digital delay. Flanging produces the classic "jet aircraft swoosh" sound.

Description

The input signal passes through an input level trim (parameter 20) into a programmable EQ, then into the flanger. The flanger output goes directly to the main output. On the left channel, the signal continues through the delay and feeds back into the flanger. A separate delay tap is sent to the right output. A single feedback parameter (12) governs both delay levels. Two independent echo taps feed the left and right outputs, with a shared level control (parameter 15). A dry path also runs from the input straight to the output, governed by the mix parameter (01). A Mix setting of 99 is recommended.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — Flanger LFO Rate `00-99` Sets the modulation speed of the flanger.

04 — Flanger LFO Width `00-99` Sets the breadth of the frequency sweep in the flanger.

05 — Flanger Center `00-99` Determines the midpoint of the flanger sweep. Increasing this value extends the available sweep range.

06 — Flanger Feedback `-99 to +99` Governs the amount of output signal fed back to the flanger input. The sign sets the feedback polarity.

07 — Flanger Notch Depth `-99 to +99` Adjusts how deep the flanging notches are. At 00, flanging is disabled, though a doppler-like effect remains audible at wide, moderately slow LFO rates.

08 — Left/Right LFO `Out-of-Phase or In-Phase` Selects whether the left and right flanger channels modulate in phase or in opposition.

09 — Flanger Sample & Hold Rate `001-100, or Off` Sets the sampling frequency of a sample-and-hold circuit applied to the flanger LFO. During each hold interval, the flanging notches freeze at their current spectral position (provided the notch depth is nonzero). A value of 001 produces the longest gaps between samples. Raising the value increases the sample frequency, resulting in smoother flanging motion. This function can also be disabled.

10 — Flanger Left Delay Time `0-1500 ms` Sets the left channel regen delay duration. This is the "ping."

11 — Flanger Right Delay Time `0-1500 ms` Sets the right channel regen delay duration. This is the "pong."

12 — Flanger Delay Feedback `-99 to +99` Governs the level of the delay taps. The sign sets the feedback polarity.

13 — Flanger Left Echo Time `0-1500 ms` Sets the left-side echo duration. Longer values produce a more prominent echo.

14 — Flanger Right Echo Time `0-1500 ms` Sets the right-side echo duration.

15 — Flanger Echo Level `00-99` Adjusts the volume of both discrete echoes. A value of 00 silences the echo entirely.

16 — Bass Fc `0-1000 Hz` Defines the corner frequency of the low-band shelving filter.

17 — EQ Gain `-48 to +24 dB` Determines how much boost or attenuation the low shelving filter applies.

18 — Treble Fc `01KHz-16KHz` Defines the corner frequency of the high-band shelving filter.

19 — EQ Gain `-48 to +24 dB` Determines how much boost or attenuation the high shelving filter applies.

20 — EQ Input Level Trim `-24 to +00 dB` Lets you reduce the signal level entering the EQ section to prevent clipping when boosting frequencies.

21–28 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

EQ/Panner/DDL

EQ/Panner/DDL pairs an EQ with an auto-panning effect and a digital delay. If the panning doesn't seem audible, check parameter 05 – a mono input signal requires the in-phase setting to produce noticeable movement.

Description

The input signal passes through an input level trim (parameter 17) into a programmable EQ, then into the panner. The panner output goes directly to the main output. On the left channel, the signal continues through the digital delay and feeds back into itself. A separate delay tap is sent to the right output. A single regen parameter (09) governs both delay levels. Two independent echo taps feed the left and right outputs, with a shared level control. This delay and echo arrangement produces the "ping-pong" effect. A dry path also runs from the input straight to the output, governed by the mix parameter (01).

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — Panner Rate `00-99` Governs the speed of the left-to-right panning motion. Higher values produce faster movement. Combining a high rate with the Sample & Hold parameter yields distinctive staccato patterns.

04 — Panner Width `00-99` Governs the breadth of the stereo panning sweep. Higher values widen the separation between left and right.

05 — Left/Right LFO `Out-of-Phase or In-Phase` Chooses between in-phase (windshield-wiper style, both channels move together) and out-of-phase (opposing motion) LFO behavior. In-phase sweeps both signals left, then both right. Out-of-phase sweeps left to the left and right to the right, then reverses, collapsing a stereo signal to mono at the midpoint. Toggle between settings to find what suits your routing.

06 — Panner Sample & Hold Rate `001-100, or Off` Sets the sampling frequency of a sample-and-hold circuit applied to the panner LFO. During each hold interval, the stereo position freezes (provided the width is nonzero). A value of 001 produces the longest gaps between holds. Raising the value increases the sample frequency, resulting in smoother panning. This function can also be disabled.

07 — Panner Left Delay Time `0-1500 ms` Sets the left channel regen delay duration. This is unrelated to the panning effect.

08 — Panner Right Delay Time `0-1500 ms` Sets the right channel regen delay duration.

09 — Panner Delay Regen `-99 to +99` Governs the regeneration level applied to both delay taps. The sign sets the feedback polarity. A value of +00 silences the delay entirely.

10 — Panner Left Echo Time `0-1500 ms` Sets the left-side echo duration. Longer times produce a more pronounced echo. Two independent echoes serve the left and right channels.

11 — Panner Right Echo Time `0-1500 ms` Sets the right-side echo duration.

12 — Panner Echo Level `00-99` Adjusts the loudness of both left and right echoes. Raising the value increases echo volume; setting it to 00 silences the echo.

13 — Bass Fc `0-1000 Hz` Defines the corner frequency of the low-band shelving filter.

14 — Bass EQ Gain `-48 to +24 dB` Determines how much boost or attenuation the low shelving filter applies.

15 — Treble Fc `01KHz-16KHz` Defines the corner frequency of the high-band shelving filter.

16 — Treble EQ Gain `-48 to +24 dB` Determines how much boost or attenuation the high shelving filter applies.

17 — EQ Input Level Trim `-24 to +00 dB` Lets you reduce the signal level entering the EQ section to avoid clipping when boosting frequencies.

18–25 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

EQ/Tremolo/DDL

EQ/Tremolo/DDL pairs a tremolo – a rhythmic fluctuation in volume – with an EQ and a digital delay.

Description

The input signal passes through an input level trim (parameter 17) into a programmable EQ, then into the tremolo. The tremolo output goes directly to the main output. On the left channel, the signal continues through the digital delay and feeds back into itself. A separate delay tap is sent to the right output. A single Regen parameter (09) governs the left and right delay levels, creating the "ping-pong" effect. Two independent echo taps feed the left and right outputs, sharing a single level control. A dry path also runs from the input straight to the output, governed by the mix parameter (01).

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — Tremolo Rate `000-200` Governs the modulation speed. Moderate values produce a noticeable wavering. At high values the frequency enters the audible range, producing ring modulation (amplitude modulation). Combining this parameter with the Sample & Hold Rate can generate distinctive staccato textures.

04 — Tremolo Depth `00-99` Governs the intensity of the amplitude modulation.

05 — Left/Right LFO `Out-of-Phase or In-Phase` Selects whether the left and right stereo tremolo channels modulate in phase or in opposition.

06 — Tremolo Sample & Hold Rate `001-100, or Off` Sets the sampling frequency of a sample-and-hold circuit applied to the tremolo LFO. During each hold interval, the instantaneous amplitude freezes (provided the depth is nonzero). A value of 001 produces the longest gaps between holds. Low values create a choppy, staccato character, while higher values increase sample density for smoother tremolo. This function can also be disabled.

07 — Tremolo Left Delay Time `0-1500 ms` Sets the left channel regen delay duration. This is unrelated to the tremolo effect.

08 — Tremolo Right Delay Time `0-1500 ms` Sets the right channel regen delay duration.

09 — Tremolo Delay Regen `-99 to +99` Governs the regeneration level applied to the delay taps. The sign sets the feedback polarity. A value of +00 silences the delay entirely.

10 — Tremolo Left Echo Time `0-1500 ms` Sets the left-side echo duration. Longer times yield a more pronounced echo.

11 — Tremolo Right Echo Time `0-1500 ms` Sets the right-side echo duration.

12 — Tremolo Echo Level `00-99` Adjusts the loudness of both left and right echoes.

13 — Bass Fc `0-1000 Hz` Defines the corner frequency of the low-band shelving filter.

14 — Bass EQ Gain `-48 to +24 dB` Determines how much boost or attenuation the low shelving filter applies.

15 — Treble Fc `01KHz-16KHz` Defines the corner frequency of the high-band shelving filter.

16 — Treble EQ Gain `-48 to +24 dB` Determines how much boost or attenuation the high shelving filter applies.

17 — EQ Input Level Trim `-24 to +00 dB` Lets you reduce the signal level entering the EQ section to avoid clipping when boosting frequencies.

18–25 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

EQ/Vibrato/DDL

EQ/Vibrato/DDL pairs a vibrato – a pitch shifter oscillating over a narrow range – with EQ and digital delay. Classic guitar amplifiers frequently included vibrato, but this algorithm is equally effective on other instruments. The sample & hold parameter does not freeze the instantaneous pitch; instead, when configured appropriately, it introduces a distinctive “chirping” quality to the input signal.

Description

The input signal first passes through an input level trim (parameter 17) and then into a programmable EQ. From the EQ, the signal feeds the vibrato, whose output appears directly at the main output. The vibrato signal also enters the delay, which feeds back into itself. A separate delay tap is sent to the right output, forming a “ping-pong” delay. A Regen parameter governs the delay feedback level. Two additional echo taps feed the left and right outputs, sharing a single level control. A dry path also runs from the input to the output, governed by the mix parameter (01). This algorithm works best with a blend of wet and dry signal.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — Vibrato Rate `00-99` Governs the modulation speed. Higher values produce faster pitch oscillation.

04 — Vibrato Width `00-99` Governs the depth of the pitch modulation.

05 — Left/Right LFO `Out-of-Phase or In-Phase` Selects the pitch-change relationship between left and right channels. In out-of-phase mode, the left channel’s quadrature pitch shift trails the right by 90 degrees. In-phase mode locks both channels together.

06 — Vibrato Sample & Hold Rate `001-100, or Off` Sets the sampling frequency of a sample-and-hold circuit applied to the vibrato LFO. At low values, it produces rhythmic pitch chirps in the audio signal. Raising the value increases the sample density, yielding smoother vibrato. This function can also be disabled.

07 — Vibrato Left Delay Time `0-1500 ms` Sets the duration of the left regenerated delay.

08 — Vibrato Right Delay Time `0-1500 ms` Sets the duration of the right non-regenerated delay.

09 — Vibrato Delay Regen `-99 to +99` Governs the positive or negative feedback applied to the regenerated delay. The sign sets the feedback polarity. A value of +00 removes all feedback. This parameter affects both left and right levels.

10 — Vibrato Left Echo Time `0-1500 ms` Sets the left-side echo duration. Longer times produce a more pronounced echo. Two independent echoes serve the left and right channels.

11 — Vibrato Right Echo Time `0-1500 ms` Sets the right-side echo duration.

12 — Vibrato Echo Level `00-99` Adjusts the loudness of both left and right echoes. A value of 00 silences the echo entirely.

13 — Bass Fc `0-1000 Hz` Defines the corner frequency of the low shelving filter.

14 — Bass EQ Gain `-48 to +24 dB` Determines how much boost or attenuation the low shelving filter applies.

15 — Treble Fc `01KHz-16KHz` Defines the corner frequency of the high-band shelving filter.

16 — Treble EQ Gain `-48 to +24 dB` Determines how much boost or attenuation the high shelving filter applies.

17 — EQ Input Level Trim `-24 to +00 dB` Lets you reduce the signal level before the EQ section to avoid clipping when boosting frequencies.

18–25 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Flanger

Flanger delivers a thick digital flanging effect. The DEEP/4 includes two distinct flanger algorithms; this one produces deeper notches and needs less feedback than the EQ/Flanger/DDL variant. The perceived flanging intensity can also be adjusted somewhat by varying the Mix level.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — Flanger LFO Rate `00-99` Sets the modulation speed of the flanger notches.

04 — Flanger LFO Width `00-99` Sets the breadth of the frequency sweep around the flanger center.

05 — Flanger Center `00-99` Determines the midpoint of the flanger sweep.

06 — Flanger Regen `-99 to +99` Governs the amount of output signal fed back to the flanger input. The sign sets the feedback polarity.

07–14 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Phaser/DDL

Phaser/DDL pairs a phaser with a digital delay. Unlike a flanger, which produces harmonically spaced notches, a phaser generates non-harmonically spaced moving notches in the frequency spectrum. This implementation uses a stereo twelve-pole phase-shifting network to create frequency-dependent time delay – the key distinction between phasing and flanging. The phasing effect is generated entirely within the phaser topology, so it does not rely on the external mix control. A delay is tapped from the left phaser output and regenerated back into the phaser; setting the phaser delay feedback parameter to 00 disables this delay. The feedback parameter also governs a feed-forward tap routed to the right channel. This delay structure achieves a

1.5-second ping-pong effect and serves well as a basic reverb substitute. A Mix setting of 99 is recommended for this algorithm.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — Phaser LFO Rate `00-99` The LFO operates within the phaser network. This parameter governs how fast the phaser poles are modulated. Higher values increase the speed. Slower rates tend to work best with sustained tones.

04 — Phaser LFO Width `00-99` Governs the range of the notch sweep. Setting this to 99 produces the widest excursion, sweeping from very high to very low frequencies.

05 — Phaser Center `-99 to +99` Sets the nominal spectral position of the phaser poles. Positive values shift the characteristic “woosh” upward in frequency; negative values shift it downward. The phaser width determines how far the sweep extends from this center point.

06 — Phaser Feedback `-99 to +99` Governs the feedback level applied to both left and right phaser channels. The sign sets the feedback polarity.

07 — Phaser Notch Depth `-99 to +99` Adjusts how deep the phasing notches are. At 99, the notches are at their most pronounced. At 00, phasing is absent, though a doppler-like effect can be heard at higher LFO rates.

08 — Left/Right LFO `Out-of-Phase or In-Phase` Selects whether the left and right phaser channels modulate in phase or in opposition.

09 — Phaser Sample & Hold Rate `001-100, or Off` Sets the sampling frequency of a sample-and-hold circuit applied to the phaser LFO. During each hold interval, the notches freeze at their current spectral position (provided the notch depth is nonzero). A value of 001 produces the longest gaps between samples. Raising the value increases the sample frequency, resulting in smoother phasing. This function can also be disabled.

10 — Phaser Left Delay Time `0-1600 ms` Sets the left-side delay duration. This is the “ping.”

11 — Phaser Right Delay Time `0-1600 ms` Sets the right-side feed-forward delay duration. This is the “pong.”

12 — Phaser Delay Feedback `-99 to +99` Governs the delay feedback level. The sign sets the feedback polarity. A value of +00 disables the delay entirely. This parameter also controls the feed-forward level.

13–20 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Rotating Speaker

Rotating Speaker reproduces the iconic sound of a mechanical rotary speaker cabinet on any keyboard instrument, though it works equally well with other sources. A configurable distortion stage drives the input signal and passes through the rotors alongside the clean signal. Higher Mix settings are recommended for this algorithm.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — Rotating Speaker Slow Speed `01-55` Sets the rotor speed when in slow mode. This parameter only takes effect when Speaker Speed (parameter 05) is set to Slow. Assigning a modulation controller allows real-time speed adjustment.

04 — Rotating Speaker Fast Speed `01-55` Sets the rotor speed when in fast mode. This parameter only takes effect when Speaker Speed (parameter 05) is set to Fast. Assigning a modulation controller allows real-time speed adjustment.

05 — Rotating Speaker Speed `Slow or Fast` Selects the current rotor speed mode.

06 — Rotating Speaker Inertia `00-99` Governs how quickly the rotor transitions between slow and fast speeds. Adjust this to replicate the gradual spin-up behavior of a physical rotary cabinet.

07 — Distortion Level In `-48 to +48 dB` Sets the gain feeding the amplifier simulation, producing tube-style overdrive. Higher values yield heavier distortion.

08 — Distortion Level Out `00-99` Determines how much of the distorted signal is blended with the clean internal path. Setting this to 00 removes all distortion from the output.

09 — Rotating Speaker Distortion Tone `000-127` Shapes the character of the distortion. High values produce a harsh, raspy tone; mid-range values deliver a warm amplifier growl. When set to Off, distortion is bypassed entirely.

10 — Rotating Speaker Stereo Spread `00-99` Adjusts the perceived width of the stereo image generated by the rotating speaker. A value of 99 creates a right-to-left synthetic spread, 00 creates a left-to-right synthetic spread, and 50 collapses the output to mono.

11–18 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Algorithms: Pitch

Fast Pitch Shift

FastPitchShift is a single-unit pitch shifter built for pitch correction tasks. Its transport delay is just 10 ms, and its maximum detune range spans one semitone. For a thickened, fat tone, experiment with small positive and negative detuning on both voices (parameters 03 and 06). This algorithm is ideal for pitch correction applications — for example, consider mapping a MIDI mod wheel to control it.

For optimal results, blend wet and dry signals using the Mix control. A modulation controller assigned to Mix works well for smoothly introducing or removing the shifted signal.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — Vc 1 Fine `-99 to +99` Provides fine pitch adjustment for Voice 1.

04 — Vc 1 Level `00-99` Sets the output level of Voice 1. At 00, the pitch-shifted signal becomes inaudible.

05 — Vc 1 Pan `-99 to +99` Positions Voice 1 within the stereo image. A value of -99 places it hard left, while +99 places it hard right.

06 — Vc 2 Fine `-99 to +99` Provides fine pitch adjustment for Voice 2.

07 — Vc 2 Level `00-99` Sets the output level of Voice 2. At 00, the pitch-shifted signal becomes inaudible.

08 — Vc 2 Pan `-99 to +99` Positions Voice 2 within the stereo image. A value of -99 places it hard left, while +99 places it hard right.

09 — LFO Rate `00-99` Governs the speed of pitch modulation, producing a chorus-like effect. For true chorusing, keep this rate very low.

10 — LFO Width `00-99` Sets the depth of pitch modulation sweep. Because the rate is typically quite low, the width is generally set quite high.

11–18 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Pitch Shift/DDL

Pitch Shift/DDL pairs a pitch shifter with a digital delay line. It employs a continuous crossfade approach to pitch shifting that preserves the stereo image precisely. Among single-unit pitch shifters, this one can yield superior results for large pitch intervals in certain situations. An additional capability is its digital delay, which feeds back into the pitch shifter to produce spiraling pitch ascents or descents.

For optimal results, blend wet and dry signals using the Mix control. A modulation controller assigned to Mix works well for gradually introducing or removing the shifted signal.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — Vc 1 Semi `-12 to +12` Shifts the pitch of Voice 1 by up to one octave higher or lower in semitone increments.

04 — Vc 1 Fine `-99 to +99` Provides fine pitch adjustment for Voice 1.

05 — Vc 1 Level `00-99` Sets the output level of Voice 1.

06 — Vc 1 Pan `-99 to +99` Positions Voice 1 within the stereo image. A value of -99 places it hard left, while +99 places it hard right.

07 — Vc 2 Semi `-12 to +12` Shifts the pitch of Voice 2 by up to one octave higher or lower in semitone increments.

08 — Vc 2 Fine `-99 to +99` Provides fine pitch adjustment for Voice 2. Minor detuning here produces a chorus-like quality.

09 — Vc 2 Level `00-99` Sets the output level of Voice 2.

10 — Vc 2 Pan `-99 to +99` Positions Voice 2 within the stereo image. A value of -99 places it hard left, while +99 places it hard right.

11 — Dry Level to DDL `00-99` Routes the unprocessed signal around the pitch shifter directly into the digital delay. Higher values increase the dry signal fed to the delay. This allows proper balancing of the dry signal against the pitch-shifted delay output.

12 — Left Delay Time `0-1500 ms` Sets the delay duration for the pitch-shifted signal originating from the left input.

13 — Right Delay Time `0-1500 ms` Sets the delay duration for the pitch-shifted signal originating from the right input.

14 — Delay Mix `00-99` Balances the delay output against the direct pitch-shifted signal. At 00, only the pitch shifter is heard with no delay. At 99, only the delayed signal is heard with no direct pitch shift.

15 — Delay Regen `-99 to +99` Governs how much of the delay output is fed back into the pitch shifter input. This enables ascending or descending pitch spiral effects.

16–23 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Pitch Shifter

Pitch Shifter is a single-unit splice-based pitch shifter capable of transposing a signal up to one octave in either direction. It operates by removing or inserting small segments of the original audio to achieve the desired effect. This algorithm excels at doubling applications. At small shift intervals, splicing occurs infrequently. Rich stereo imaging can be achieved by panning the two voices independently, aided by the algorithm's inherent time-delay modulation. The left channel input feeds Voice 1, and the right channel input feeds Voice 2.

For optimal results, blend wet and dry signals using the Mix control. A modulation controller assigned to Mix works well for smoothly introducing or removing the shifted signal.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — Vc 1 Semi `-12 to +12` Shifts the pitch of Voice 1 by up to one octave higher or lower in semitone (half-step) increments.

04 — Vc 1 Fine `-99 to +99` Provides fine pitch adjustment for Voice 1.

05 — Vc 1 Level `00-99` Sets the output level of Voice 1. At 00, the pitch-shifted signal becomes inaudible.

06 — Vc 1 Pan `-99 to +99` Positions Voice 1 within the stereo image. A value of -99 places it hard left, while +99 places it hard right.

07 — Vc 2 Semi `-12 to +12` Shifts the pitch of Voice 2 by up to one octave higher or lower in semitone increments.

08 — Vc 2 Fine `-99 to +99` Provides fine pitch adjustment for Voice 2.

09 — Vc 2 Level `00-99` Sets the output level of Voice 2. At 00, the pitch-shifted signal becomes inaudible.

10 — Vc 2 Pan `-99 to +99` Positions Voice 2 within the stereo image. A value of -99 places it hard left, while +99 places it hard right.

11 — Delay vs Quality `Long/Smoother or Short/Coarser` Selects between a long/smooth mode and a short/coarse mode. The smooth setting is best for slow, sustained chords, while the coarse setting suits fast musical passages. This parameter governs the effect's transport delay: smooth produces a longer transport delay, coarse produces a shorter one. (Pitch Shifter 1U Only)

12 — LFO Rate `00-99` Governs the speed of pitch modulation, producing a chorus-like effect. For true chorusing, keep this rate very low.

13 — LFO Width `00-99` Sets the depth of pitch modulation sweep. Because the rate is typically quite low, the width is generally set quite high.

14–21 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Pitch Shift 2U

Pitch Shift 2U is a two-unit splice-based pitch shifter that incorporates zero-crossing detection. It can transpose a signal up to one octave in either direction. Like the single-unit Pitch Shifter, it functions by removing or inserting small segments of the source audio, but Pitch Shift 2U dedicates one ESP chip to zero-crossing (pitch) detection for splice synchronization, with an optimal detection range of 55 to 555Hz. Rich stereo imaging can be achieved by panning the two voices independently, aided by the algorithm's inherent time-delay modulation. The left channel input feeds Voice 1, and the right channel input feeds Voice 2.

For optimal results, blend wet and dry signals using the Mix control. A modulation controller assigned to Mix works well for smoothly introducing or removing the shifted signal.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — Vc 1 Semi `-12 to +12` Shifts the pitch of Voice 1 by up to one octave higher or lower in semitone (half-step) increments.

04 — Vc 1 Fine `-99 to +99` Provides fine pitch adjustment for Voice 1.

05 — Vc 1 Level `00-99` Sets the output level of Voice 1. At 00, the pitch-shifted signal becomes inaudible.

06 — Vc 1 Pan `-99 to +99` Positions Voice 1 within the stereo image. A value of -99 places it hard left, while +99 places it hard right.

07 — Vc 2 Semi `-12 to +12` Shifts the pitch of Voice 2 by up to one octave higher or lower in semitone increments.

08 — Vc 2 Fine `-99 to +99` Provides fine pitch adjustment for Voice 2.

09 — Vc 2 Level `00-99` Sets the output level of Voice 2. At 00, the pitch-shifted signal becomes inaudible.

10 — Vc 2 Pan `-99 to +99` Positions Voice 2 within the stereo image. A value of -99 places it hard left, while +99 places it hard right.

11 — LFO Rate `00-99` Governs the speed of pitch modulation, producing a chorus-like effect. For true chorusing, keep this rate very low.

12 — LFO Width `00-99` Sets the depth of pitch modulation sweep. Because the rate is typically quite low, the width is generally set quite high.

13–20 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Algorithms: Dynamics

De/esser

De-esser is a stereo algorithm that attenuates sibilant frequencies (such as “ess” sounds) when they exceed a certain loudness. Originally intended for vocal processing, it can also tame boomy guitar tones or the ringing quality of drums when the side-chain is configured appropriately.

A Volume setting of 99 is recommended.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — Output Gain `-48 to +48 dB` Applies cut (negative values) or boost (positive values) to the de-esser’s output level. A starting value of 00 dB is recommended.

04 — Comp Ratio `1:1 to 40:1, infinity` Determines the degree of compression. This ratio operates in decibels (dB) above the threshold. At a 4:1 setting, for instance, signal levels above the threshold are reduced to one quarter of their original change. At infinity, the compressor behaves as a limiter.

05 — Threshold `-96 to +00 dB` Establishes the threshold level. Signals exceeding this point undergo compression; signals remaining below it are left unchanged.

06 — Gain Change `N/A` A read-only display showing the current gain reduction.

07 — Comp Attack `50us to 100ms` Sets how quickly compression engages once the signal is detected, before attenuation takes hold.

08 — Comp Release `1ms to 10.0s` Sets the time required for compression to fully disengage once the signal drops beneath the threshold. This is typically set longer than the attack time (parameter 06).

09 — Noise Gate Off Below `-96 to +00 dB` Defines the lower threshold at which the noise gate silences the audio.

10 — Noise Gate On Above `-96 to +00 dB` Defines the upper threshold at which the noise gate allows audio through. This second, higher threshold guards against spurious triggering.

11 — Sidechain EQ HighPass Fc `4 to 8000 Hz` Sets the high-pass filter frequency for the side-chain EQ. Particularly useful for de-essing applications.

12 — Bass Fc `0 to 1000 Hz` Specifies the cutoff frequency of the low-frequency shelving filter.

13 — Bass Gain (loShv) `-48 to +24 dB` Controls the amount of boost or cut on the low shelving filter.

14 — Mid1 Fc `100 to 9999 Hz` Specifies the center frequency of the first mid-range parametric band. Higher frequencies yield a brighter character.

15 — Mid1 Gain `-48 to +24 dB` Controls the amount of boost or cut on this parametric band.

16 — Mid1 Q `01 to 18` Adjusts the bandwidth of the resonant peak centered on this frequency band. Raising the value narrows the bandwidth.

17 — Mid2 Fc `100 to 9999 Hz` Specifies the center frequency of the second mid-range parametric band. These three parameters (17-19) mirror the previous three and allow independent control over a separate mid-range region.

18 — Mid2 Gain `-48 to +24 dB` Controls the amount of boost or cut on this parametric band.

19 — Mid2 Q `01 to 18` Adjusts the bandwidth of the resonant peak centered on this frequency band.

20 — Treble Fc `01KHz to 16KHz` Specifies the cutoff frequency of the high-frequency shelving filter.

21 — Treble Gain (HiShv) `-48 to +24 dB` Controls the amount of boost or cut on the high shelving filter.

22 — Sidechain EQ Input Trim `-48 to +00 dB` Trims the input level feeding the side-chain EQ, which analyzes the incoming signal to apply selective compression.

23–30 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Ducker / Gate

Ducker / Gate is a compressor that automatically reduces the level of one signal (such as background music) whenever a second signal (like a voice-over) is present. Once the voice-over stops, the original signal returns to its normal level. This algorithm is well suited for voice-over work, rap, and DJ applications. For correct operation, the music (the audio to be ducked) must be connected to Input 1 (left), and the voice-over goes into Input 2 (right). In this configuration, Input 2 serves as the side chain of a conventional compressor. An internal mixer combines the left and right inputs to a mono output.

At high compression ratios, the algorithm functions as a gate. In this mode, a transient source such as a snare drum can gate another music signal on Input 1 on and off, producing an externally controlled staccato effect.

The Mixer operates the same as in all other algorithms and is separate from the ducker output mixer.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — Output Gain `-48 to +48 dB` Applies cut (negative values) or boost (positive values) to the ducker's output level. A starting value of 00 dB is recommended.

04 — Ducker Output Mix `00-99` Blends the music signal output (Input 1) with the ducker's voice-over signal output (Input 2) into a mono sum. This is the internal mixer.

05 — Comp Ratio `1:1 to 40:1, infinity` Determines the degree of compression. This ratio operates in decibels (dB) above the threshold. At a 4:1 setting, for instance, signal levels above the threshold are reduced to one quarter of their original change. At infinity, the compressor behaves as a limiter.

06 — Threshold `-96 to +00 dB` Establishes the threshold level. Signals exceeding this point undergo compression; signals remaining below it are left unchanged.

07 — Gain Change `N/A` A read-only display showing the current gain reduction.

08 — Comp Attack `50us to 100ms` Sets how quickly compression engages once the signal is detected, before attenuation takes hold.

09 — Comp Release `1ms to 10.0s` Sets the time required for compression to fully disengage once the signal drops beneath the threshold. This is typically set longer than the attack time (parameter 06).

10 — Noise Gate Off Below `-96 to +00 dB` Defines the lower threshold at which the noise gate silences the audio.

11 — Noise Gate On Above `-96 to +00 dB` Defines the upper threshold at which the noise gate allows audio through. This second, higher threshold guards against spurious triggering.

12 — Bass Fc `0 to 1000Hz` Specifies the cutoff frequency of the low-frequency shelving filter.

13 — Bass Gain (loShv) `-48 to +24 dB` Controls the amount of boost or cut on the low shelving filter.

14 — Mid1 Fc `100 to 9999 Hz` Specifies the center frequency of the first mid-range parametric band. Higher frequencies yield a brighter character.

15 — Mid1 Gain `-48 to +24 dB` Controls the amount of boost or cut on this parametric band.

16 — Mid1 Q `01 to 18` Adjusts the bandwidth of the resonant peak centered on this frequency band. Higher values narrow the bandwidth.

17 — Mid2 Fc `100 to 9999 Hz` Specifies the center frequency of the second mid-range parametric band. These three parameters (17-19) mirror the previous three and allow independent control over a separate mid-range region.

18 — Mid2 Gain `-48 to +24 dB` Controls the amount of boost or cut on this parametric band.

19 — Mid2 Q `01 to 18` Adjusts the bandwidth of the resonant peak centered on this frequency band.

20 — Treble Fc `01KHz to 16KHz` Specifies the cutoff frequency of the high-frequency shelving filter.

21 — Treble Gain (HiShv) `-48 to +24 dB` Controls the amount of boost or cut on the high shelving filter.

22 — Side Chain EQ Input Trim `-48 to +00 dB` Trims the input level feeding the side-chain EQ, which analyzes the incoming signal to apply selective compression.

23–30 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

EQ/Compressor

EQ/Compressor pairs equalization with a fully featured compressor. At high compression ratios, this algorithm serves as a limiter. It works by attenuating signals that rise above the threshold while leaving signals below the threshold untouched. With elevated ratios and reduced thresholds, it can generate sustain. EQ is present in both the audio signal path and the side chain path, unlike the Expander which provides filtering only in the side chain.

A Volume setting of 99 is recommended.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — Compressor Gain `-48 to +48 dB` Increases the level of the compressed signal.

04 — Compressor Ratio `1:1 to 40:1, infinity` Determines the degree of compression. This ratio operates in decibels (dB) above the threshold. At a 4:1 setting, for instance, signal levels above the threshold are reduced to one quarter of their original change. At infinity, the compressor behaves as a limiter.

05 — Compressor Threshold `-96 to +00 dB` Establishes the threshold level. Signals exceeding this point undergo compression; signals remaining below it are left unchanged. Setting the level to 00 dB disables the compressor.

06 — Gain Change `N/A` A read-only display showing the current gain reduction.

07 — Comp Attack `50us to 100ms` Sets how quickly compression engages once the signal is detected, before attenuation takes hold.

08 — Comp Release `1ms to 10.0s` Sets the time required for compression to fully disengage once the signal drops beneath the threshold. This is typically chosen to be longer than the attack time (parameter 06).

09 — Noise Gate Off Below `-96 to +00 dB` Defines the lower threshold at which the noise gate silences the audio.

10 — Noise Gate On Above `-96 to +00 dB` Defines the upper threshold at which the noise gate allows audio through. This second, higher threshold guards against spurious triggering.

11 — Gate Release Time `1ms to 10.0s` Sets the time required for the gate to fully release once the signal drops beneath the threshold. Shorter values produce a snappier gate.

12 — Bass Fc `0 to 1000 Hz` Specifies the cutoff frequency of the low-frequency shelving filter.

13 — Bass EQ Gain `-48 to +24 dB` Controls the amount of boost or cut on the low shelving filter.

14 — Treble Fc `01KHz to 16KHz` Specifies the cutoff frequency of the high-frequency shelving filter.

15 — Treble EQ Gain `-48 to +24 dB` Controls the amount of boost or cut on the high shelving filter.

16 — EQ Input Level Trim `-24 to +00 dB` Trims the input level feeding the EQ section, preventing clipping when signals are boosted.

17–24 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Expander

Expander performs downward expansion on incoming signals. At high expansion ratios, it functions as a gate. The algorithm attenuates signals falling below the user-defined threshold while passing signals above it unaltered. This makes it effective for noise elimination. No EQ is present in the audio path; high-pass and low-pass filtering are available on the side chain only.

Two notable features distinguish this algorithm:

1. The ADSR envelope includes Attack, Sustain, and Release stages (the sustain stage is referred to as Hold Time).
2. A trigger mask function is provided, primarily intended for extracting a click track from drum recordings. When activated, this function forces a zero signal level into the side chain detector for a user-specified duration. The function engages when Trigger Mask is enabled and the side chain signal drops below the Trigger Mask Threshold.

A Volume setting of 99 is recommended.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — Exp Ratio `1:1 to 1:40, infinity` Determines the degree of expansion. This ratio operates in decibels (dB) below the threshold. At a 1:4 setting, for instance, signals below the threshold are expanded by a factor of four. At infinity, the expander behaves as a gate. A 1:1 setting applies no expansion.

04 — Exp Threshold `-96 to +00 dB` Establishes the threshold level. Signals above this point pass through unaffected; signals below it undergo expansion. Setting the level to -96 dB disables the expander.

05 — Gain Change `N/A` A read-only display showing the current gain reduction in real time.

06 — Exp Attack `50us to 100ms` Sets how quickly expansion engages once the signal is detected, before attenuation takes hold.

07 — Exp Release `1ms to 10.0s` Sets the release speed once the signal is detected below the threshold. This is typically chosen to be longer than the attack time (parameter 06).

08 — Expander Gate Hold Time `1ms to 10.0s` The sustain duration within the ADSR envelope, spanning the attack, sustain, and release stages.

09 — Sidechain EQ Gain `-48 to +48 dB` Adjusts the boost applied to the high/low pass filter output signal. This compensates for any insertion loss introduced by those filters.

10 — HighPass Fc `4 to 8000 Hz` Specifies the cutoff frequency of the high-pass shelving filter on the lower frequency band.

11 — LowPass Fc `100 Hz to 16 KHz` Controls the boost or cut applied to the low-pass filter.

12 — Trigger Mask `Off or On` Activates the trigger mask function. Once engaged, the side chain detector receives no input signal for the duration set by parameter 13.

13 — Trigger Time `1ms to 10.0s` Specifies how long the side chain detector remains blacked out. This is particularly useful for isolating the opening bar of a drum track.

14 — Trig Mask Lower Threshold `-96 to +00 dB` Establishes the trigger mask threshold. When the signal falls below this level, the mask function activates. The Expander Threshold (04) serves as upward hysteresis for the trigger mask, so the Trigger Mask Threshold must always be set below the Expander Threshold.

15 — Expander Output Gain `-48 to +48 dB` Applies cut (negative values) or boost (positive values) to the expander's output level. A starting value of +00 dB is recommended.

16–23 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Inverse Expander

InversExpander generates sustain by expanding the signal so that levels above the threshold pass through normally while levels below the threshold are boosted to produce a more uniform output. A conventional expander does the opposite, attenuating signals that fall below the threshold. In practice, an inverse expander resembles a compressor: both can sustain signals and diminish transient peaks. EQ is present in both the audio signal path and the side chain path, unlike the Expander which provides filtering only in the side chain.

A Mix setting of 99 is recommended.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — Expnd Ratio `1:1 to 40:1, Infinity` Determines the degree of expansion. This ratio operates in decibels (dB) below the threshold. At a 3:1 setting, for instance, signals below the threshold are expanded by a factor of three. Starting with values near 1:1 is recommended (a setting of exactly 1:1 disables expansion).

04 — Threshold `-96 to +00 dB` Establishes the threshold level. Signals falling below this point are boosted; signals remaining above it are left unchanged. As the input decays below the threshold, the expander progressively increases signal gain. Setting the threshold to -96dB disables the inverse expander.

05 — Gain Change `N/A` A read-only display showing the current gain increase.

06 — Exp Attack `50us to 100ms` Sets the time between initial signal detection and the onset of expansion.

07 — Exp Release `1ms to 10.0s` Sets the time required for expansion to fully disengage once the signal rises above the threshold. This is typically longer than the attack time.

08 — Exp Noise Gate Off Below `-96 to +00 dB` Defines the lower threshold at which the noise gate silences the audio.

09 — Comp Noise Gate On Above `-96 to +00 dB` Defines the upper threshold at which the noise gate allows audio through. This second parameter provides hysteresis.

10 — Bass Fc `0 to 1000 Hz` Specifies the cutoff frequency of the low-frequency shelving filter.

11 — Bass EQ Gain `-48 to +24 dB` Controls the amount of boost or cut on the low shelving filter.

12 — Treble Fc `01KHz to 16KHz` Specifies the cutoff frequency of the high-frequency shelving filter.

13 — Treble EQ Gain `-48 to +24 dB` Controls the amount of boost or cut on the high shelving filter.

14 — EQ Input Level Trim `-24 to +00 dB` Trims the level of the signal entering the EQ section, preventing clipping when signals are boosted.

15–22 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Keyed Expander

Keyed Expander functions identically to the standard Expander. The sole distinction is that the left signal (Input 1) undergoes expansion as controlled by a key signal. The key is derived from the right channel signal (Input 2).

The Mixer operates the same as in all other algorithms and is separate from the output mixer.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — Exp Ratio `1:1 to 1:40, infinity` Determines the degree of expansion. This ratio operates in decibels (dB) below the threshold. At a 1:4 setting, for instance, signals below the threshold are expanded by a factor of four. At infinity, the expander behaves as a gate. A 1:1 setting applies no expansion.

04 — Exp Threshold `-96 to +00 dB` Establishes the threshold level. Signals above this point pass through unaffected; signals below it undergo expansion. Setting the level to -96 dB disables the expander.

05 — Gain Change `N/A` A read-only display showing the current gain reduction in real time.

06 — Exp Attack `50us to 100ms` Sets how quickly expansion engages once the signal is detected, before attenuation takes hold.

07 — Exp Release `1ms to 10.0s` Sets the release speed once the signal is detected below the threshold. This is typically chosen to be longer than the attack time (parameter 06).

08 — Expander Gate Hold Time `1ms to 10.0s` The sustain duration within the ADSR envelope, spanning the attack, sustain, and release stages.

09 — Sidechain EQ Gain `-48 to +48 dB` Adjusts the boost applied to the high/low pass filter output signal. This compensates for any insertion loss introduced by those filters.

10 — HighPass Fc `4 to 8000 Hz` Specifies the cutoff frequency of the high-pass shelving filter on the lower frequency band.

11 — LowPass Fc `100 Hz to 16 KHz` Controls the boost or cut applied to the low-pass filter.

12 — Trigger Mask `Off or On` Activates the trigger mask function. Once engaged, the side chain detector receives no input signal for the duration set by parameter 13.

13 — Trigger Time `1ms to 10.0s` Specifies how long the side chain detector remains blacked out. This is particularly useful for isolating the opening bar of a drum track.

14 — Trigger Mask Threshold `-96 to +00 dB` Establishes the trigger mask threshold. When the signal falls below this level, the mask function activates. The Expander Threshold (04) serves as upward hysteresis for the trigger mask, so the Trigger Mask Threshold must always be set below the Expander Threshold.

15 — Expander Output Mix `00-99` Blends the left signal output (Input 1) with the right signal output (Input 2). This is the internal output mixer.

16 — Expander Output Gain `-48 to +48 dB` Applies cut (negative values) or boost (positive values) to the expander's output level. A starting value of +00 dB is recommended.

17–24 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Algorithms: EQ / Filter

Parametric EQ

Parametric EQ provides a minimum phase, four-band parametric equalizer. A Mix setting of 99 is suggested for optimal results with this algorithm.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — Bass Fc `0-1000 Hz` Defines the center frequency for the low band parametric.

04 — Bass Gain (loShv) `-48 to +24 dB` Adjusts the level of boost or attenuation for the low frequency parametric band.

05 — Mid1 Fc `100-9999 Hz` Defines the center frequency for the first mid band parametric.

06 — Mid1 Gain `-48 to +24 dB` Adjusts the level of boost or attenuation for the first mid frequency parametric band.

07 — Mid1 Q `01-18` Controls the bandwidth, shaping how wide or narrow the resonant peak is around the center frequency. Increasing this value narrows the affected frequency range.

08 — Mid2 Fc `100-9999 Hz` Defines the center frequency for a second mid band parametric. Functions identically to Mid1 Fc (05), targeting an independent region of the mid spectrum.

09 — Mid2 Gain `-48 to +24 dB` Adjusts the level of boost or attenuation for the second mid frequency parametric band. Functions identically to Mid1 Gain (06).

10 — Mid2 Q `01-18` Controls the bandwidth for the second mid frequency parametric. Functions identically to Mid1 Q (07), targeting an independent region of the mid spectrum.

11 — Treble Fc `01-16 KHz` Defines the center frequency for the high band parametric.

12 — Treble Gain (HiShv) `-48 to +24 dB` Adjusts the level of boost or attenuation for the high frequency parametric band.

13 — EQ Input Level Attenuation `-24 to +00 dB` Provides an input level trim ahead of the EQ stages, preventing clipping when signals are boosted.

14–21 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Rumble Filter

Rumble Filter pairs a high pass filter with a low pass filter in series, both operating at fourth order (24dB per octave). The high pass stage is well suited to removing turntable rumble, while the low pass stage effectively suppresses hiss. These filters can also be placed in a feedback loop alongside any other effect.

Mid-range Mix values are suggested for this algorithm.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — HighPass Fc `4-8000 Hz` Governs the cutoff frequency of the high pass filter acting on the input signal.

04 — LowPass Fc `100 Hz-16 KHz` Governs the cutoff frequency of the low pass filter acting on the input signal.

05 — Filter Gain `-48 to +48 dB` Compensates for the insertion loss introduced by the cascaded high pass and low pass stages, letting you raise the filtered output level.

06–13 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

VanderPol Filter

VanderPol Filter generates synthetic upper harmonics that are blended with the input, producing a brighter overall tone. Originally conceived for vocal work in the studio, this algorithm is equally rewarding when applied to instruments of all kinds. It delivers pronounced transient enhancement, making it particularly effective for plucked or percussive sounds. The filter stage operates on the signal before harmonic enhancement is applied. Dial in the filter to emphasize the frequency region you want enriched, then blend the enhanced output with the dry signal.

Mid-range Mix values are suggested for this algorithm.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — HighPass Fc `4-8000 Hz` Governs the cutoff frequency of the high pass filter acting on the input signal.

04 — LowPass Fc `100 Hz-16 KHz` Governs the cutoff frequency of the low pass filter acting on the input signal.

05 — Filter Gain `-48 to +48 dB` Compensates for the insertion loss introduced by the cascaded high pass and low pass stages, letting you raise the filtered output level.

06–13 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

VCF/Distortion

VCF/Distortion pairs a voltage controlled filter with a gritty distortion stage. Three distinct effects are available: Distortion, Wah-wah, and Auto-wah.

Description

The wah-wah and auto-wah share the same VCF. These filters can be switched off or repurposed as an EQ if needed. When using the distortion mode, cascading a speaker cabinet emulation (such as Tunable Speaker) after this effect is recommended. A second VCF follows the distortion stage. It can serve as a basic speaker simulator, or it can be modulated alongside the pre-distortion VCF.

With this algorithm, Volume governs the distortion output level. When using high distortion input gains, keep the volume low.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — Distortion Level In `00-99` Sets the gain feeding into the distortion stage, boosting the signal by up to 48 dB. For heavier distortion, drive the input gain high and reduce Distortion Level Out (04) to manage volume. For lighter distortion, keep the input gain low and raise the output volume.

04 — Distortion Level Out `00-99` Sets the gain at the distortion output. When Distortion Level In (03) is high, reduce this parameter to keep the overall volume manageable.

05 — Pre-Distortion VCF Fc `01-99` Establishes the filter cutoff frequency ahead of the distortion. Raising this value brightens the tone. This parameter accepts modulation from a CV Pedal to create a wah-wah pedal effect. Setting it to 99 disables the distortion filter. For EQ use, choose your desired value and ensure the envelope follower (parameter 07) is at 00. For auto-wah, set this near 01 (low end) and engage parameter 07.

06 — Pre-Distortion VCF Q `01-25` Shapes the height and width of the resonant peak at the filter cutoff. While Fc (filter cutoff) sets the frequency where this peak sits, Q governs how pronounced it is. This is especially important for achieving effective auto-wah sounds.

07 — Envelope Follower to Pre VCF `-99 to +99` Controls how strongly the input signal's amplitude modulates the distortion filter cutoff. At 00, no modulation occurs. With moderate positive values, Fc sweeps upward and then settles back to its resting position. With moderate negative values, Fc dips downward before returning. The speed of this recovery depends on parameters

11 and 12. This produces the auto-wah effect: positive values emphasize upper frequencies for an “oww” character, while negative values roll off highs for a “dweep” character.

08 — Post-Distortion VCF Fc `01-99` Same function as Pre-Distortion VCF Fc, but governs the second VCF located after the distortion stage.

09 — Post-Distortion VCF Q `01-25` Same function as Pre-Distortion VCF Q, but governs the second VCF located after the distortion stage.

10 — Envelope Follower to Post VCF `-99 to +99` Same function as Envelope Follower to Pre VCF, but governs the second VCF located after the distortion stage.

11 — Envelope Follower Attack `50us-10.0s` Configures the envelope follower’s attack time, determining how tightly it tracks the onset of incoming signals. Shorter attack times are generally preferred.

12 — Envelope Follower Release `1ms-10.0s` Configures the decay duration after the input signal stops, controlling how long the envelope follower takes to return to zero. These values are typically longer than the attack time.

13 — Distortion Bypass `Off or On` Lets you bypass the distortion stage entirely.

14 — Pre-EQ High Pass Cutoff `0-1000 Hz` Removes low frequency content ahead of the EQ. Higher values attenuate more of the low end.

15–22 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Algorithms: Amp / Speaker

Guitar Amp I

Guitar Amp I reproduces the rich, warm character of a guitar amplifier through tube distortion emulation. Suitable for any stringed instrument, Guitar Amp I delivers more distortion than Guitar Amp II and is tailored for hard rock tones.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — Amp Preamp Gain `-48 to +48 dB` Sets the boost or attenuation applied to the incoming signal. A setting of 00 dB is recommended, as the emulations were calibrated for optimal distortion at that point. Reducing preamp gain yields less distortion; increasing it introduces clipping. At low preamp gain settings, low tube bias values may produce more natural results.

04 — Output Level `00-99` Governs the main amp’s output level ahead of the output EQ stage.

05 — Amp Tube Bias `00-99` At preamp gains near 00 dB, this shapes the balance between even and odd harmonics, defining the amplifier’s tonal character. Mid-range values favor even harmonics for a warmer, “glowing tube” quality, while extreme settings can mimic the sound of failing tubes. Tube bias operates independently from preamp gain. Using low tube bias values at low preamp gains more faithfully replicates how a physical amplifier behaves.

06 — Pre-EQ Input Level Trim `-24 to +00 dB` Trims the signal level entering the preamp EQ, preventing clipping from boosted frequencies.

07 — Pre-EQ High Pass Cutoff `4-1000 Hz` Attenuates low frequencies ahead of the preamp. Raising this value removes more low-end content.

08 — Pre-EQ Fc `100-9999 Hz` Sets the center frequency of the preamp parametric filter. Higher frequencies yield a brighter character.

09 — Pre-EQ Gain `-48 to +24 dB` Sets the boost or attenuation applied by the preamp parametric filter.

10 — Pre-EQ Q `01-18` Shapes the width of the resonant peak around the parametric filter's center frequency. The Fc parameter selects the peak's frequency location, while Q governs how prominent the peak is.

11 — Noise Gate Off Below `-96 to +00 dB` Establishes the lower threshold at which the noise gate silences the audio output.

12 — Noise Gate On Above `-96 to +00 dB` Establishes the upper threshold at which the noise gate allows audio to pass. This second, higher threshold guards against spurious triggering.

13 — Gate Release Time `1ms-10.0s` Determines how long the noise gate remains open after the signal drops away. Increase this value to preserve longer sustain.

14 — Speaker High Pass Cutoff `4-1000 Hz` Attenuates low frequencies from the main amp before reaching the speaker. Raising this value removes more low-end content.

15 — OutEQ1 Fc `100-9999 Hz` Sets the center frequency of the first parametric filter in the main amp output stage. Higher frequencies yield a brighter character.

16 — OutEQ1 Gain `-48 to +24 dB` Sets the boost or attenuation applied by the main amp's first parametric filter.

17 — OutEQ1 Q `01-18` Shapes the width of the resonant peak for the first output parametric. The Fc parameter selects the peak's frequency location, while Q governs how prominent the peak is.

18 — OutEQ2 Fc `100-9999 Hz` Sets the center frequency of the second parametric filter in the main amp output stage. Higher frequencies yield a brighter character.

19 — OutEQ2 Gain `-48 to +24 dB` Sets the boost or attenuation applied by the main amp's second parametric filter.

20 — OutEQ2 Q `01-18` Shapes the width of the resonant peak for the second output parametric.

21 — Speaker Low Pass Cutoff `2.0-16.0 KHz` Attenuates high frequencies at the speaker output. Lowering this value removes more high-end content. This filter is less precise than the dedicated speaker cabinet emulation algorithms available in the DEEP/4.

22–29 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Guitar Amp II

Guitar Amp II reproduces the rich, warm character of a guitar amplifier through tube distortion emulation. Suitable for any stringed instrument, Guitar Amp II provides less distortion than Guitar Amp I and is optimized for blues-style tones.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — Amp Preamp Gain `-48 to +48 dB` Sets the boost or attenuation applied to the incoming signal. A setting of 00 dB is recommended, as the emulations were calibrated for optimal distortion at that point. Reducing preamp gain yields less distortion; increasing it introduces clipping. At low preamp gain settings, low tube bias values may produce more natural results.

04 — Output Level `00-99` Governs the main amp's output level ahead of the output EQ stage.

05 — Amp Tube Bias `00-99` At preamp gains near 00 dB, this shapes the balance between even and odd harmonics, defining the amplifier's tonal character. Mid-range values favor even harmonics for a warmer, "glowing tube" quality, while extreme settings can mimic the sound of failing tubes. Tube bias operates independently from preamp gain. Using low tube bias values at low preamp gains more faithfully replicates how a physical amplifier behaves.

06 — Pre-EQ Input Level Trim `-24 to +00 dB` Trims the signal level entering the preamp EQ, preventing clipping from boosted frequencies.

07 — Pre-EQ High Pass Cutoff `4-1000 Hz` Attenuates low frequencies ahead of the preamp. Raising this value removes more low-end content.

08 — Pre-EQ Fc `100-9999 Hz` Sets the center frequency of the preamp parametric filter. Higher frequencies yield a brighter character.

09 — Pre-EQ Gain `-48 to +24 dB` Sets the boost or attenuation applied by the preamp parametric filter.

10 — Pre-EQ Q `01-18` Shapes the width of the resonant peak around the parametric filter's center frequency. The Fc parameter selects the peak's frequency location, while Q governs how prominent the peak is.

11 — Noise Gate Off Below `-96 to +00 dB` Establishes the lower threshold at which the noise gate silences the audio output.

12 — Noise Gate On Above `-96 to +00 dB` Establishes the upper threshold at which the noise gate allows audio to pass. This second, higher threshold guards against spurious triggering.

13 — Gate Release Time `1ms-10.0s` Determines how long the noise gate remains open after the signal drops away. Increase this value to preserve longer sustain.

14 — Speaker High Pass Cutoff `4-1000 Hz` Attenuates low frequencies from the main amp before reaching the speaker. Raising this value removes more low-end content.

15 — OutEQ1 Fc `100-9999 Hz` Sets the center frequency of the first parametric filter in the main amp output stage. Higher frequencies yield a brighter character.

16 — OutEQ1 Gain `-48 to +24 dB` Sets the boost or attenuation applied by the main amp's first parametric filter.

17 — OutEQ1 Q `01-18` Shapes the width of the resonant peak for the first output parametric. The Fc parameter selects the peak's frequency location, while Q governs how prominent the peak is.

18 — OutEQ2 Fc `100-9999 Hz` Sets the center frequency of the second parametric filter in the main amp output stage. Higher frequencies yield a brighter character.

19 — OutEQ2 Gain `-48 to +24 dB` Sets the boost or attenuation applied by the main amp's second parametric filter.

20 — OutEQ2 Q `01-18` Shapes the width of the resonant peak for the second output parametric.

21 — Speaker Low Pass Cutoff `2.0-16.0 KHz` Attenuates high frequencies at the speaker output. Lowering this value removes more high-end content. This filter is less precise than the dedicated speaker cabinet emulation algorithms available in the DEEP/4.

22–29 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Guitar Amp III

Guitar Amp III pairs an inverse expander with a bright distortion circuit, producing amp lead tones. The inverse expander functions like a compressor that boosts all signals falling below the threshold. This algorithm is designed for heavy metal guitar solos.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — Preamp Gain `-48 to +48 dB` Sets the boost or attenuation applied to the EQ'd input signal. High gain settings produce lead tones.

04 — Output Level `00-99` Governs the output level ahead of the output EQ stage.

05 — PreEQ Input Level Trim `-24 to +00 dB` Trims the signal level entering the preamp EQ, preventing clipping from boosted frequencies.

06 — Pre-EQ Fc `100-9999 Hz` Sets the center frequency of the preamp-stage parametric filter. Higher frequencies yield a brighter character.

07 — Pre-EQ Gain `-48 to +24 dB` Sets the boost or attenuation applied by the preamp parametric filter.

08 — Pre-EQ Q `01-18` Shapes the width of the resonant peak around the filter center frequency. The Fc parameter selects the peak's frequency location, while Q governs how prominent the peak is.

09 — ExpndRatio `1:1 to 40:1, infinity` Controls the degree of inverse expansion applied to signals below the threshold. At a 3:1 ratio, for instance, level changes below the threshold are amplified threefold, pushing the signal amplitude toward the threshold level.

10 — Threshold `-96 to +00 dB` Defines the inverse expander's threshold. Signals below this level receive expansion, while those above it remain unchanged. As the input decays below the threshold, the expander progressively increases signal gain.

11 — Gain Change `N/A` A read-only display showing the current signal level.

12 — Noise Gate Off Below `-96 to +00 dB` Establishes the lower threshold at which the noise gate silences the audio output.

13 — Noise Gate On Above `-96 to +00 dB` Establishes the upper threshold at which the noise gate allows audio to pass. This second, higher threshold guards against spurious triggering.

14 — Gate Release Time `1ms-10.0s` Determines how long the noise gate remains open after the signal drops away. Increase this value to preserve longer sustain.

15 — Speaker High Pass Cutoff `4-1000 Hz` Attenuates low frequencies from the main amp before reaching the speaker. Raising this value removes more low-end content.

16 — OutEQ1 Fc `100-9999 Hz` Sets the center frequency of the first parametric filter in the main amp output stage. Higher frequencies yield a brighter character.

17 — OutEQ1 Gain `-48 to +24 dB` Sets the boost or attenuation applied by the main amp's first parametric filter.

18 — OutEQ1 Q `01-18` Shapes the width of the resonant peak around the filter center frequency. The Fc parameter selects the peak's frequency location, while Q governs how prominent the peak is.

19 — OutEQ2 Fc `100-9999 Hz` Sets the center frequency of the second parametric filter in the main amp output stage. Higher frequencies yield a brighter character.

20 — OutEQ2 Gain `-48 to +24 dB` Sets the boost or attenuation applied by the main amp's second parametric filter.

21 — OutEQ2 Q `01-18` Shapes the width of the resonant peak for the second output parametric.

22 — Speaker Low Pass Cutoff `2.0-16.0 KHz` Attenuates high frequencies at the speaker output. Raising this value allows more highs through. Dedicated speaker emulation algorithms are available as separate effects.

23–30 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Speaker Cabinet

Speaker Cabinet reproduces the warm tone of an open-back speaker enclosure. Excellent for guitar, bass, and other stringed instruments, it is especially valuable in studio settings when recording direct to the console. This algorithm captures both the resonant characteristics and nonlinear behavior of a real musical instrument speaker. Take care not to overdrive this cabinet with VCF/Distortion; lower the volume on that effect and compensate using the output gain here.

For a brighter cabinet tone, consider using Tunable Speaker instead.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — Speaker Output Gain `-48 to +24 dB` Speaker cabinets inherently attenuate signal level, so output gain is provided to recover perceived loudness. Pushing this gain too high will clip the output signal.

04–11 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Tunable Speaker

Tunable Speaker delivers an EQ-adjustable speaker tone that is brighter than Speaker Cabinet. With three parametric filters at your disposal, you can sculpt a wide variety of speaker cabinet characteristics suited to any musical genre.

Parameters

01 — Mix `00-99` See Common Parameters.

02 — Volume `00-99` See Common Parameters.

03 — Mid1 Fc `100-9999 Hz` Defines the center frequency of the first mid-range parametric. Higher frequencies yield a brighter character.

04 — Mid1 Gain `-48 to +24 dB` Applies attenuation (negative values) or boost (positive values) to the first mid-frequency parametric band.

05 — Mid1 Q `01-18` Controls the bandwidth, shaping how wide or narrow the resonant peak is around the center frequency. Increasing this value narrows the affected frequency range.

06 — Mid2 Fc `100-9999 Hz` Functions the same as Mid1 Fc, but targets a separate region within the mid-range.

07 — Mid2 Gain `-48 to +24 dB` Functions the same as Mid1 Gain, but targets a separate region within the mid-range.

08 — Mid2 Q `01-18` Functions the same as Mid1 Q, but targets a separate region within the mid-range.

09 — Mid3 Fc `100-9999 Hz` Functions the same as Mid1 Fc, but targets a third independent region within the mid-range.

10 — Mid3 Gain `-48 to +24 dB` Functions the same as Mid1 Gain, but targets a third independent region within the mid-range.

11 — Mid3 Q `01-18` Functions the same as Mid1 Q, but targets a third independent region within the mid-range.

12 — Speaker Input Attenuation `-24 to +00 dB` Trims the input level ahead of the EQ stages, preventing clipping when signals are boosted.

13 — Speaker Output Gain `-48 to +24 dB` Speaker cabinets inherently attenuate signal level, so output gain is provided to recover perceived loudness. Pushing this gain too high will clip the output signal.

14–21 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.

Algorithms: Utility

No Effect

No Effect bypasses the processing unit, applying no signal processing. Whether this utility algorithm passes audio through (bypass) or mutes it (kill) is determined by the Edit/Config parameters.

Parameters

01 — Mix `00-99` Blends a dry signal with silence. This algorithm produces two signals: one audible and one silent. At 00, the audible signal is selected. At 99, the silent signal is selected. In practice, this Mix parameter behaves like an inverted volume control.

02 — Volume `00-99` Sets the loudness of the dry external signal. 00 is silent and 99 is full volume.

03–10 — Mod Parameters – See Common Parameters for Mod1/Mod2 Source, Destination, and Range parameters.