# Algorithm Reference

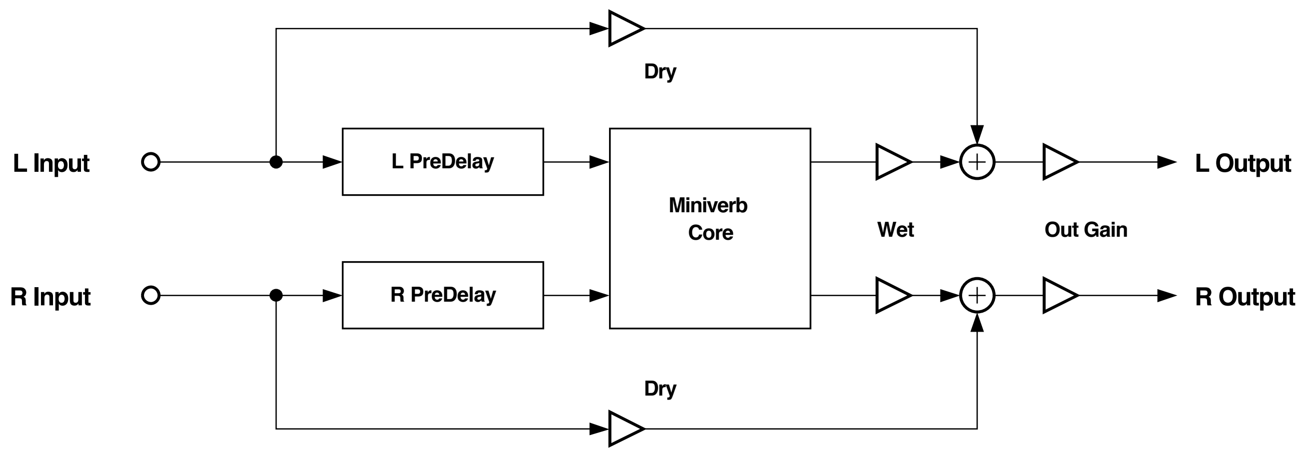


**Part Number: 910319 Rev. A**

# FXAlg #1: MiniVerb •
# FXAlg #2: Dual MiniVerb

**Versatile, small stereo and dual mono reverbs**
*Allocation Units:* 1 for MiniVerb, 2 for Dual MiniVerb

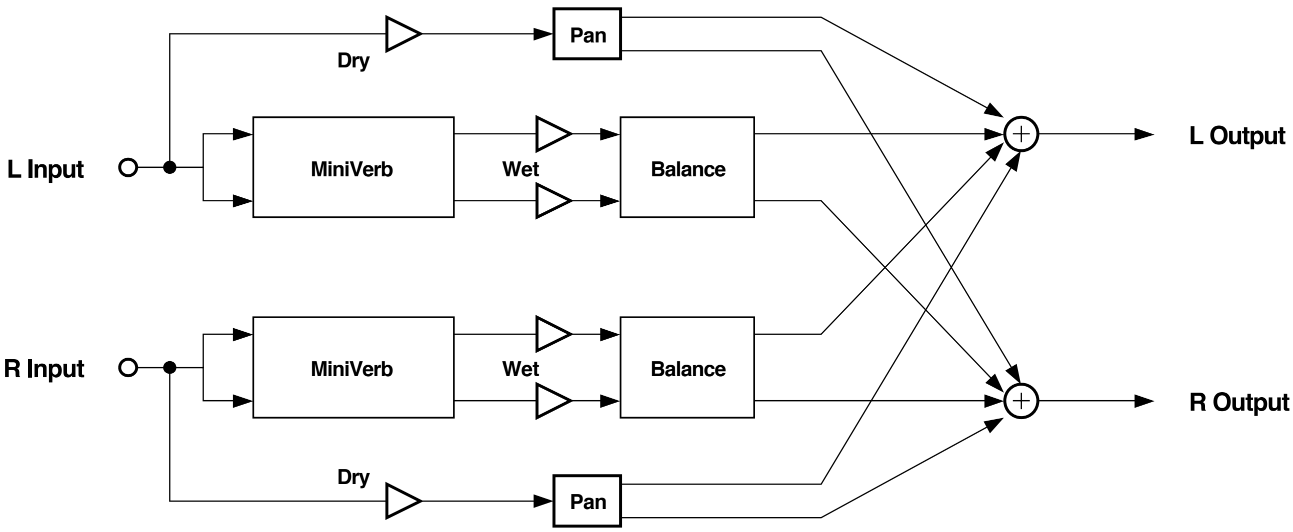
MiniVerb is a versatile stereo reverb which is found in many combination algorithms, but is equally useful on its own because of its small size. The main control for this effect is the Room Type parameter. Room Type changes the structure of the algorithm to simulate many carefully crafted room types and sizes. Spaces characterized as booths, small rooms, chambers, halls and large spaces can be selected.




**Simplified block diagram of MiniVerb**

Each Room Type incorporates different diffusion, room size and reverb density settings. The Room Types were designed to sound best when Diff Scale, Size Scale and Density are set to the default values of 1.00x. If you want a reverb to sound perfect immediately, set the Diff Scale, Size Scale and Density parameters to 1.00x, pick a Room Type and you'll be on the way to a great sounding reverb. But if you want to experiment with new reverb flavors, changing the scaling parameters away from 1.00x can cause a subtle (or drastic!) coloring of the carefully crafted Room Types.

Diffusion characterizes how the reverb spreads the early reflections out in time. At very low settings of Diff Scale, the early reflections start to sound quite discrete, and at higher settings the early reflections are seamless. Density controls how tightly the early reflections are packed in time. Low Density settings have the early reflections grouped close together, and higher values spread the reflections for a smoother reverb.




**Simplified block diagram of Dual MiniVerb**

Dual MiniVerb has a full MiniVerb, including Wet/Dry, Pre Delay and Out Gain controls, dedicated to each of the left and right channels. The two blocks in the diagram above labeled “MiniVerb” contain a complete copy of the MiniVerb on the previous page. Dual MiniVerb gives you independent reverbs on both channels which has obvious benefits for mono material. With stereo material, any panning or image placement can be maintained, even in the reverb tails. This is pretty unusual behavior for a reverb, since even real halls will rapidly delocalize acoustic images in the reverberance. Since maintaining image placement in the reverberation is so unusual, you will have to carefully consider whether it is appropriate for your particular situation. To use Dual MiniVerb to maintain stereo signals in this manner, set the reverb parameters for both channels to the same values. The Dry Pan and Wet Bal parameters should be fully left (-100%) for the left MiniVerb and fully right (100%) for the right MiniVerb.

## Parameters (MiniVerb):

**PAGE 1**

| | | | |
|---|---|---|---|
| Wet/Dry | 0 to 100%wet | Out Gain | Off, -79.0 to 24.0 dB |
| Rvrb Time | 0.5 to 30.0 s, Inf | HF Damping | 16 to 25088 Hz |
| L Pre Dly | 0 to 620 ms | R Pre Dly | 0 to 620 ms |

**PAGE 2**

| | | | |
|---|---|---|---|
| Room Type | Hall1 | Diff Scale | 0.00 to 2.00x |
| | | Size Scale | 0.00 to 4.00x |
| | | Density | 0.00 to 4.00x |

## Parameters (Dual MiniVerb):

### PAGE 1

| | | | |
|---|---|---|---|
| L Wet/Dry | 0 to 100%wet | R Wet/Dry | 0 to 100%wet |
| L Out Gain | Off, -79.0 to 24.0 dB | R Out Gain | Off, -79.0 to 24.0 dB |
| L Wet Bal | -100 to 100% | R Wet Bal | -100 to 100% |
| L Dry Pan | -100 to 100% | R Dry Pan | -100 to 100% |

### PAGE 2

| | | | |
|---|---|---|---|
| L RoomType | Hall1 | | |
| L RvrbTime | 0.5 to 30.0 s, Inf | | |
| L Diff Scl | 0.00 to 2.00x | L Density | 0.00 to 4.00x |
| L Size Scl | 0.00 to 4.00x | L HF Damp | 16 to 25088 Hz |
| L PreDlyL | 0 to 620 ms | L PreDlyR | 0 to 620 ms |

### PAGE 3

| | | | |
|---|---|---|---|
| R RoomType | Hall1 | | |
| R RvrbTime | 0.5 to 30.0 s, Inf | | |
| R Diff Scl | 0.00 to 2.00x | R Density | 0.00 to 4.00x |
| R Size Scl | 0.00 to 4.00x | R HF Damp | 16 to 25088 Hz |
| R PreDlyL | 0 to 620 ms | R PreDlyR | 0 to 620 ms |

Wet/Dry — A simple mix of the reverb sound with the dry sound.

Out Gain — The overall gain or amplitude at the output of the effect.

Rvrb Time — The reverb time displayed is accurate for normal settings of the other parameters (HF Damping = 25088kHz, and Diff Scale, Room Scale and Density = 1.00x). Changing Rvrb Time to Inf creates an infinitely sustaining reverb.

HF Damping — Reduces high-frequency components of the reverb above the displayed cutoff frequency. Removing higher reverb frequencies can often make rooms sound more natural.

L/R Pre Dly — The delay between the start of a sound and the output of the first reverb reflections from that sound. Longer pre-delays can help make larger spaces sound more realistic. Longer times can also help improve the clarity of a mix by separating the reverb signal from the dry signal, so the dry signal is not obscured. Likewise, the wet signal will be more audible if delayed, and thus you can get by with a dryer mix while maintaining the same subjective wet/dry level.

Room Type — Changes the configuration of the reverb algorithm to simulate a wide array of carefully designed room types and sizes. This parameter effectively allows you to have several different reverb algorithms only a parameter change away. Smaller Room Types will sound best with shorter Rvrb Times, and vice versa. (Note that since this parameter changes the structure of the reverb algorithm, you don’t want to assign it a KDFX Modulation that will change it in real time.)

Diff Scale — A multiplier which affects the diffusion of the reverb. At 1.00x, the diffusion will be the normal, carefully-adjusted amount for the current Room Type. Altering this parameter will change the diffusion from the preset amount.

| | |
|---|---|
| Size Scale | A multiplier which changes the size of the current room. At 1.00x, the room will be the normal, carefully-tweaked size of the current Room Type. Altering this parameter will change the size of the room, and thus will cause a subtle coloration of the reverb (since the room's dimensions are changing). |
| Density | A multiplier which affects the density of the reverb. At 1.00x, the room density will be the normal, carefully-set amount for the current Room Type. Altering this parameter will change the density of the reverb, which may color the room slightly. |
| Wet Bal | In Dual MiniVerb, two mono signals (left and right) are fed into two separate stereo reverbs. If you center the wet balance (0%), the left and right outputs of the reverb will be sent to the final output in equal amounts. This will add a sense of spaciousness. |

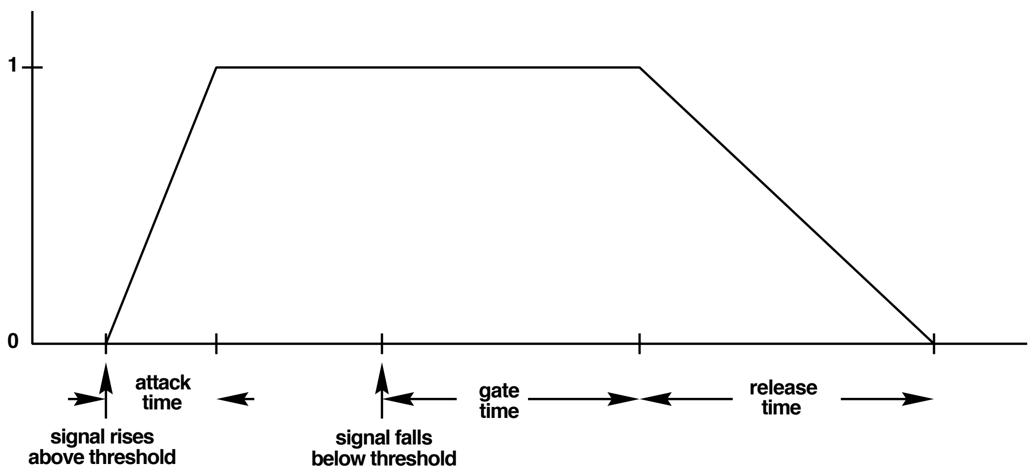
# FXAlg #3:  Gated MiniVerb

**A reverb and gate in series**
*Allocation Units:* 2

This algorithm is a small reverb followed by a gate. The main control for the reverb is the Room Type parameter. Room Type changes the structure of the algorithm to simulate many carefully crafted room types and sizes. Spaces characterized as booths, small rooms, chambers, halls and large spaces can be selected. See the previous section (FXAlg #1-2) for details on the reverb.

The gate turns the output of the reverb on and off based on the amplitude of the input signal. One or both input channels is used to control whether the switch is on (gate is open) or off (gate is closed). This on/off control is called “side chain” processing. You select which of the two input channels or both is used for side chain processing. When you select both channels, the sum of the left and right input amplitudes is used.

The gate is opened when the side chain amplitude rises above a level that you specify with the Threshold parameter. The gate will stay open for as long as the side chain signal is above the threshold. When the signal drops below the threshold, the gate will remain open for the time set by the Gate Time parameter. At the end of the Gate Time, the gate closes. When the signal rises above threshold, it opens again. What is happening is that the gate timer is being constantly retriggered while the signal is above threshold.




**Gate Behavior**

If Gate Duck is turned on, then the behavior of the gate is reversed. The gate is open while the side chain signal is below threshold, and it closes when the signal rises above threshold.

If the gate opened and closed instantaneously, you would hear a large digital click, like a big knife switch was being thrown. Obviously that’s not a good idea, so Gate Atk (attack) and Gate Rel (release) parameters are used to set the times for the gate to open and close. More precisely, depending on whether Gate Duck is off or on, Gate Atk sets how fast the gate opens or closes when the side chain signal rises above the threshold. The Gate Rel sets how fast the gate closes or opens after the gate timer has elapsed.

The Signal Dly parameter delays the signal being gated, but does not delay the side chain signal. By delaying the main signal relative to the side chain signal, you can open the gate just before the main signal rises above threshold. It’s a little like being able to pick up the telephone before it rings!

## Parameters:

### PAGE 1

| | | | |
|---|---|---|---|
| Wet/Dry | 0 to 100%wet | Out Gain | Off, -79.0 to 24.0 dB |
| Rvrb Time | 0.5 to 30.0s, Inf | HF Damping | 16 to 25088 Hz |
| L Pre Dly | 0 to 620ms | R Pre Dly | 0 to 620 ms |

### PAGE 2

| | | | |
|---|---|---|---|
| Room Type | Hall1 | Diff Scale | 0.00 to 2.00x |
| | | Size Scale | 0.00 to 4.00x |
| | | Density | 0.00 to 4.00x |

### PAGE 3

| | | | |
|---|---|---|---|
| Gate Thres | -79.0 to 0.0 dB | Gate Time | 0 to 3000 ms |
| Gate Duck | In or Out | Gate Atk | 0.0 to 228.0 ms |
| | | Gate Rel | 0 to 3000 ms |
| | | GateSigDly | 0.0 to 25.0 ms |
| Reduction<br><br>-dB 60 40 * 16 * 8 4 0 | | | |

<u>Wet/Dry</u> A simple mix of the reverb sound with the dry sound. When set fully dry (0%), the gate is still active.

<u>Out Gain</u> An overall level control of the effect's output (applied after the gate).

<u>Gate Thres</u> The input signal level in dB required to open the gate (or close the gate if Gate Duck is on).

<u>Gate Duck</u> When set to "Off", the gate opens when the signal rises above threshold and closes when the gate time expires. When set to "On", the gate closes when the signal rises above threshold and opens when the gate time expires.

<u>Gate Time</u> The time in seconds that the gate will stay fully on after the signal envelope rises above threshold. The gate timer is started or restarted whenever the signal envelope rises above threshold.

<u>Gate Atk</u> The attack time for the gate to ramp from closed to open (reverse if Gate Duck is on) after the signal rises above threshold.

<u>Gate Rel</u> The release time for the gate to ramp from open to closed (reverse if Gate Duck is on) after the gate timer has elapsed.

<u>Signal Dly</u> The delay in milliseconds (ms) of the reverb input signal relative to the side chain signal. By delaying the reverb signal, the gate can be opened before the reverb signal rises above the gating threshold.

For descriptions of the other parameters, see the previous section, FXAlgs #1-2.

# FXAlgs #4-11: Classic • TQ • Diffuse • Omni reverbs

## FXAlg #4: Classic Place
## FXAlg #5: Classic Verb
## FXAlg #6: TQ Place
## FXAlg #7: TQ Verb
## FXAlg #8: Diffuse Place
## FXAlg #9: Diffuse Verb
## FXAlg #10: OmniPlace
## FXAlg #11: OmniVerb

**More Complex Reverb algorithms**
*Allocation Units:* “Classic” 2; others 3

This set of 2 and 3 PAU sized algorithms can be divided into 2 groups: Verb and Place. Verb effects allow user friendly control over medium to large spaces. Their decay times are controlled by Rvrb Time or LateRvbTim parameters, and Room Types range from rooms to large areas. Place algorithms on the other hand are optimized for small spaces. Decay time is controlled by the Absorption parameter, and Room Types offers several booths.

Each of these reverb algorithms combines several components: a diffuser, an injector, predelay, an ambience generator with feedback, and various filters. These components provide sonic building blocks for both the early reflection portions and the body of the reverb.

The ambience generator is the heart of each reverb algorithm and creates most of the ‘late’ reverb in algorithms with an Early Reflections circuit. It is comprised of a complex arrangement of delay lines to disperse the sound. By using feedback in conjunction with the ambience generator, a reverb tail is produced. The length of this reverb tail is controlled by the Rvrb Time parameter in the “Verb” algorithms, or the Absorption parameter in “Place” algorithms.

In order to create reverbs that are smoother and richer, some of the delays in the ambience generator are moved by LFOs. The LFOs are adjusted by using the LFO Rate and LFO Depth controls. When used subtly, unwanted artifacts such as flutter and ringing that are inherent in digital reverbs can be reduced.

In the feedback loop of the ambience generator are filters that further enhance the sonic properties of each reverb. A lowpass filter is controlled by HF Damping. Its action mimics high-frequency energy being absorbed as the sound travels around a room. A low shelving filter is controlled by LF Split and LF Time, which are used to shorten or lengthen the decay time of low frequency energy.

At the beginning of each algorithm are diffusers. A diffuser creates an initial “smearing” quality on input signals usually before the signal enters the ambience generating loop. The DiffAmtScl and DiffLenScl parameters respectively change the amount and the length of time that the sound is smeared. The Diffuse reverbs, however, implement diffusion a little differently. See the section on Diffuse Verb and Diffuse Place below for detailed information.

Some algorithms use injector mechanisms when feeding a signal into the ambience generator. An injector creates copies of the input signal at different delay intervals and feeds each copy into the ambience generator at different points. This results in finer control over the onset of the reverb. By tapering the amplitudes of early copies vs. late copies, the initial build of the reverb can be controlled. Inj Build controls this taper. Negative values create a slower build, while positive values create a faster build. Inj Spread scales the length of all the copies as a group. Inj Skew (Omni reverbs) delays one channel relative to the other before injecting into the ambience generator. Negative values delay the left side while positive values delay the right side. Inj LP controls the cutoff frequency of a 1 pole (6dB/oct) lowpass filter associated with the injector.

Predelay can give the illusion that a space is more voluminous. Separate control over left and right predelay is provided which can be used to de-correlate the center image, increasing reverb envelopment.

In addition to filters inside the ambience feedback loop, there also may be filters placed at the output of the reverb including a low shelf, high shelf, and/or lowpass.

Algorithms that utilize Early Reflection circuits use a combination of delays, diffusers, and filters to create ambience that is sparser than the late portion of the reverb. These early reflections model the initial near-discrete echoes rebounding directly off of near field surfaces before the reverb has a chance to become diffuse. They add realism when emulating real rooms and halls.

The starting point when creating a new reverb preset should be the Room Type parameter. This parameter selects the basic type of reverb. Due to the inherent complexity of reverb algorithms and the sheer number of variables responsible for their character, the Room Type parameter provides condensed preset collections of these variables. Each Room Type collection has been painstakingly selected by Kurzweil engineers to provide the best-sounding combination of mutually complementary variables modeling an assortment of reverb families. When a room type is selected, an entire incorporated set of delay lengths and diffusion settings are established within the algorithm. By using the Size Scale, DiffAmtScl, DiffLenScl, and Inj Spread parameters, you may scale individual elements away from their pre-defined values. When set to 1.00x, each of these elements are accurately representing their preset values determined by the current Room Type.

Room Types with similar names in different reverb algorithms do not sound the same. For example, Hall1 in Diffuse Verb does not sound the same as Hall1 in TQ Verb.

The Size Scale parameter scales the inherent size of the reverb chosen by Room Type. For a true representation of the selected Room Type size, set this to 1.00x. Scaling the size below this will create smaller spaces, while larger scale factors will create large spaces.

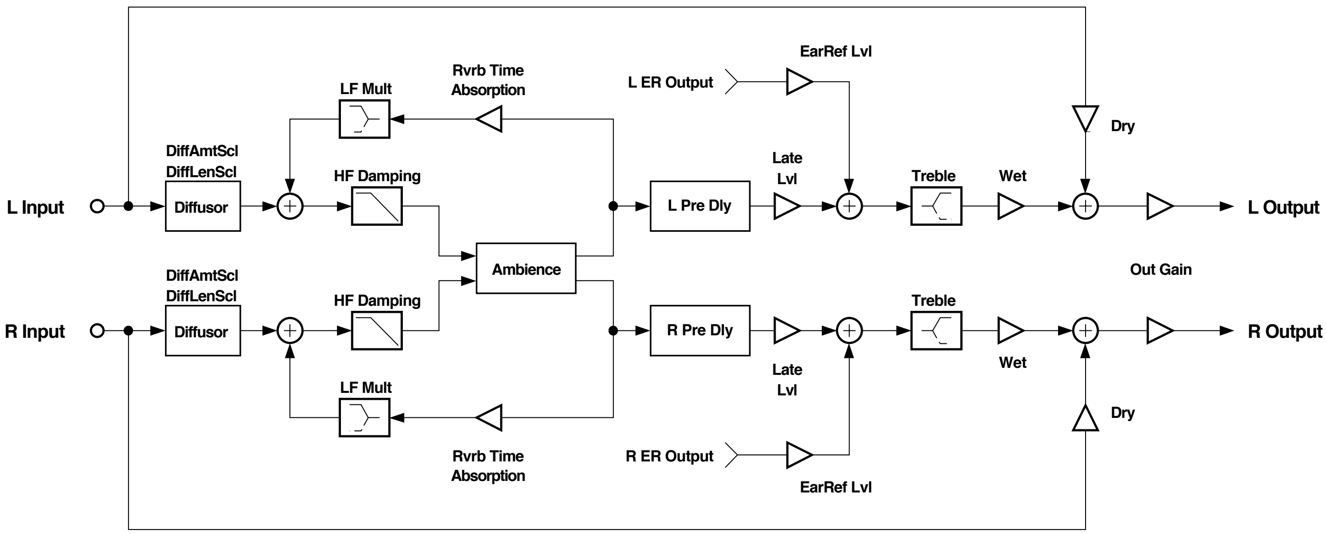
The InfinDecay switch is designed to override the Rvrb Time parameter and create a reverb tail with an infinite decay time when ‘On’. However, certain HF Damping settings may reduce this effect, and cause the tail to taper away. This parameter is an excellent candidate for a KDFX Modulation, using a switch pedal as a source.

## Classic Verb and Classic Place:

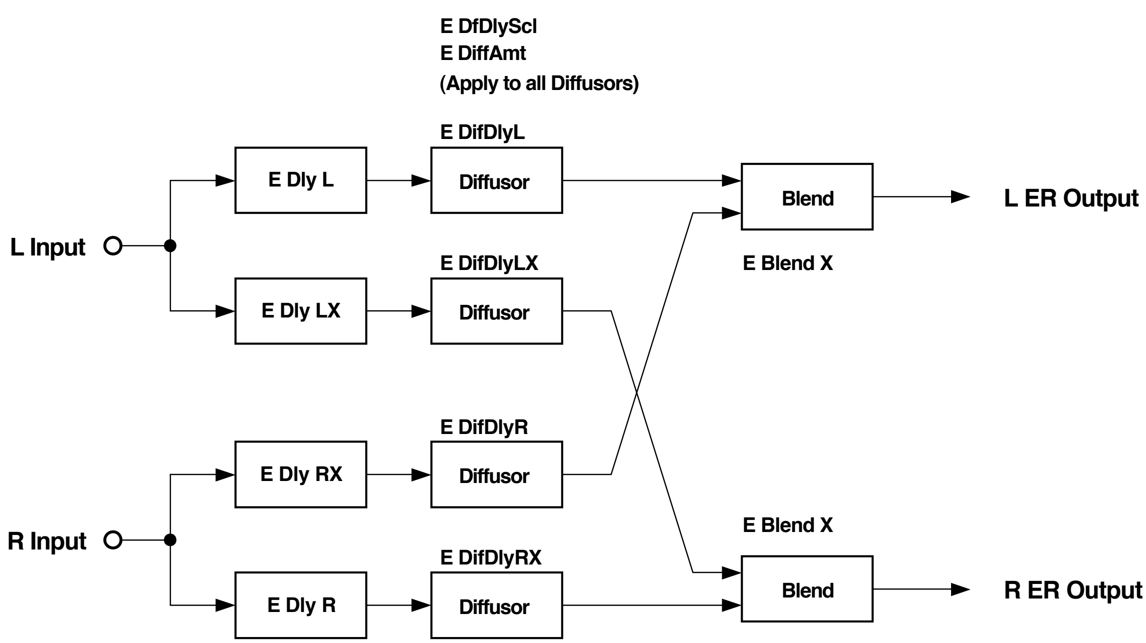
Classic reverbs are 2-PAU algorithms with early reflections. The late portion consists of an input diffuser; ambience generator with low shelving filters, lowpass filters, and LFO moving delays; and predelay.

The early reflection portion consists of one delay per channel sent to its own output channel controlled by E Dly L and E Dly R, and one delay per channel sent to its opposite output channel controlled be E Dly LX and E Dly RX. Each of these delays also use a Diffuser. Diffusion lengths are separately controlled by E DifDly L, E DifDly R, E DifDly LX, and E DifDly RX while diffusion amounts are all adjusted with E DiffAmt.

The late reverb and early reflection portions are independently mixed together with the Late Lvl and EarRef Lvl controls. The wet signal is passed through a final high-shelving filter before being mixed with the dry signal.




Signal flow of Classic Verb and Classic Place




Early reflection portion of Classic Verb and Classic Place

PAGE 1 (Classic Verb)

| | | | |
|---|---|---|---|
| Wet/Dry | -100 to 100% | Out Gain | Off; -79.0 to 24.0 dB |
| Rvrb Time | 0.00 to 60.00 s | EarRef Lvl | -100 to 100% |
| HF Damping | 0 to 25088 Hz | Late Lvl | -100 to 100% |
| L Pre Dly | 0.0 to 230.0 ms | R Pre Dly | 0.0 to 230.0 ms |

PAGE 1 (Classic Place)

| | | | |
|---|---|---|---|
| Wet/Dry | -100 to 100% | Out Gain | Off; -79.0 to 24.0 dB |
| Absorption | 0 to 100% | EarRef Lvl | -100 to 100% |
| HF Damping | 0 to 25088 Hz | Late Lvl | -100 to 100% |
| L Pre Dly | 0.0 to 230.0 ms | R Pre Dly | 0.0 to 230.0 ms |

PAGE 2 (Classic Verb)

| | | | |
|---|---|---|---|
| Room Type | Hall1, ... | DiffAmtScl | 0.00 to 2.00 x |
| Size Scale | 0.01 to 2.00x | DiffLenScl | 0.00 to 2.00 x |
| InfinDecay | On or Off | LFO Rate | 0.01 to 10.00 Hz |
| | | LFO Depth | 0.0 to 100.0 ct |
| TrebShlf F | 16 to 25088 Hz | LF Split | 16 to 25088 Hz |
| TrebShlf G | -79.0 to 24.0 dB | LF Time | 0.50 to 1.50 x |

PAGE 2 (Classic Place)

| | | | |
|---|---|---|---|
| Room Type | Booth1, ... | DiffAmtScl | 0.00 to 2.00 x |
| Size Scale | 0.01 to 2.00x | DiffLenScl | 0.00 to 2.00 x |
| | | LFO Rate | 0.01 to 10.00 Hz |
| | | LFO Depth | 0.0 to 100.0 ct |
| TrebShlf F | 16 to 25088 Hz | LF Split | 16 to 25088 Hz |
| TrebShlf G | -79.0 to 24.0 dB | LF Time | 0.50 to 1.50 x |

PAGE 3

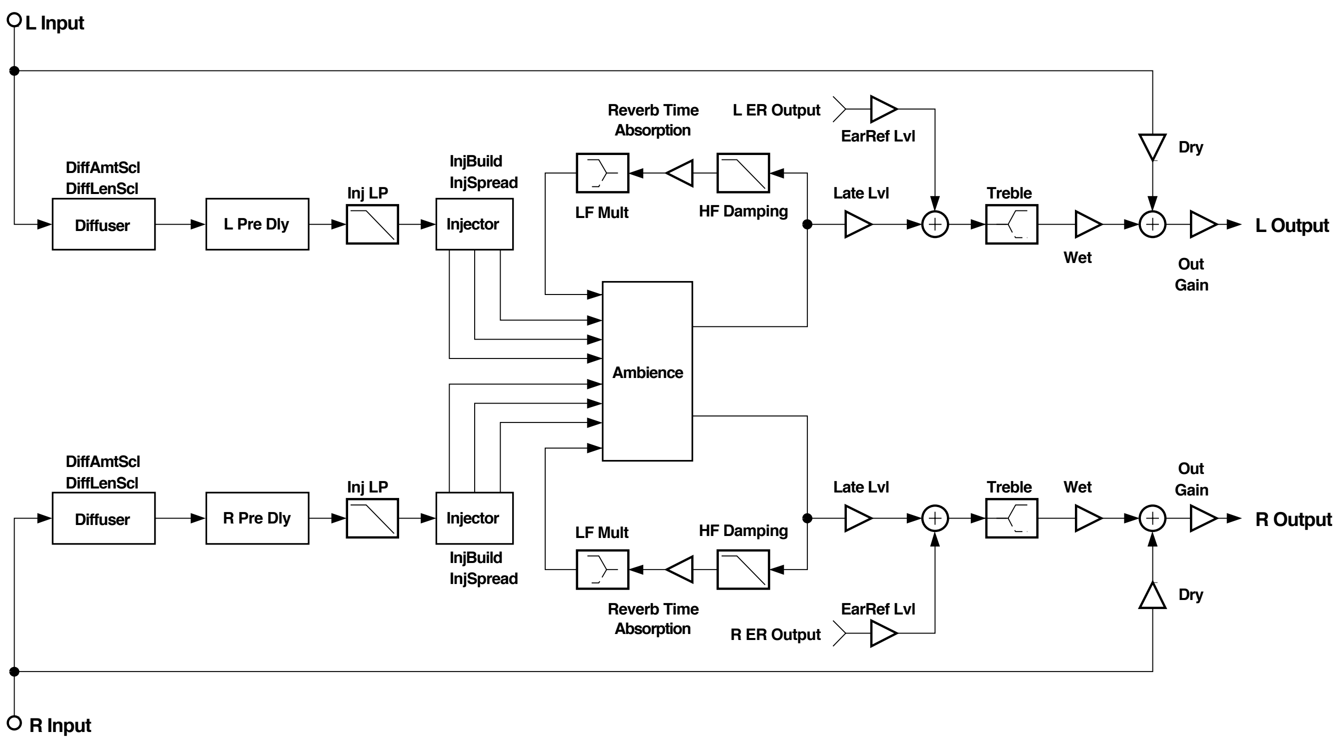
| | | | |
|---|---|---|---|
| E DfDlyScl | 0.00 to 2.00 x | E X Blend | 0 to 100% |
| E DiffAmt | -100 to 100% | | |
| E Dly L | 0.0 to 720.0 ms | E Dly R | 0.0 to 720.0 ms |
| E Dly LX | 0.0 to 720.0 ms | E Dly RX | 0.0 to 720.0 ms |
| E DifDlyL | 0.0 to 160.0 ms | E DifDlyR | 0.0 to 160.0 ms |
| E DifDlyLX | 0.0 to 230.0 ms | E DifDlyRX | 0.0 to 230.0 ms |

## TQ Verb and TQ Place:

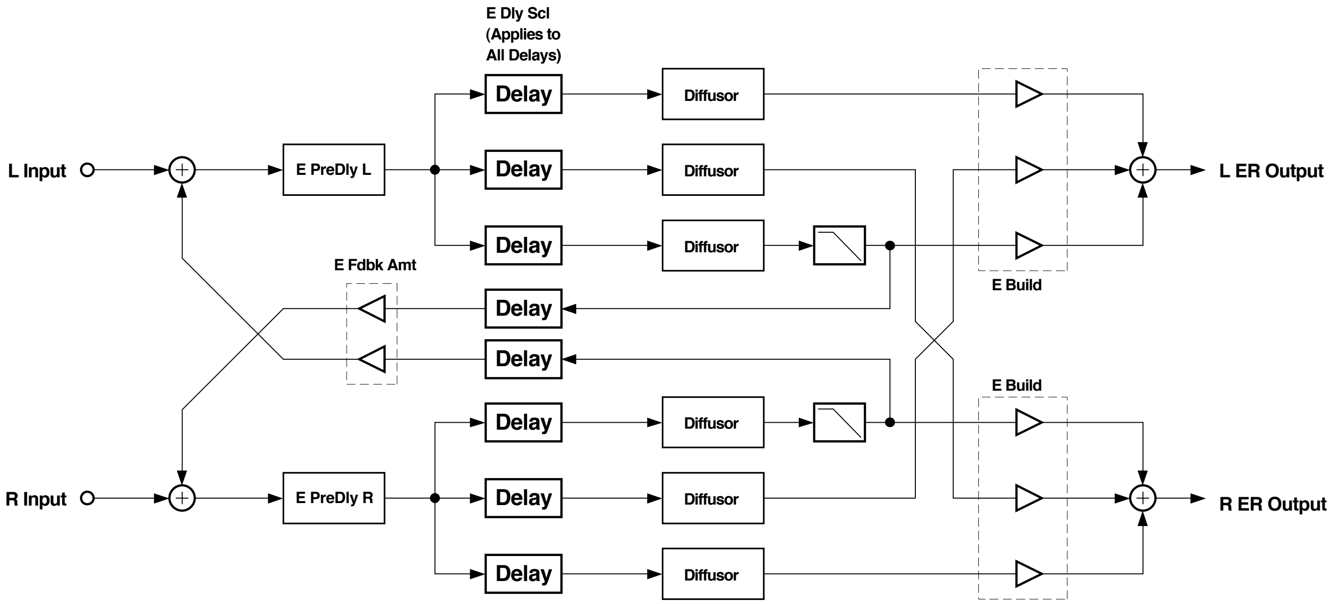
TQ reverbs are 3-PAU algorithms with early reflections. The late portion consists of an input diffuser, injector, ambience generator with a lowpass filter, low shelving filter, and LFO moving delays, and predelay.

The early reflection portion combines a combination of delays, diffusers, and feedback. The relative delay lengths are all fixed but are scalable with the E Dly Scl parameter. Relative diffusion lengths are also fixed, and are scalable with the E DfLenScl parameter. Diffusion amounts are adjusted with E DiffAmt. The E Build parameter ramps the gains associated with each delay line in a way that changes the characteristic of the onset of the early reflections. Negative amounts create a slower onset while positive amounts create a faster onset.

The late reverb and early reflection portions are independently mixed together with the Late Lvl and EarRef Lvl controls. The wet signal is passed through a final high shelving filter before being mixed with the dry signal.




**Signal flow of TQ Verb and TQ Place**




Early reflection portion of TQ Verb and TQ Place

PAGE 1 (TQ Verb)

| | | | |
|---|---|---|---|
| Wet/Dry | -100 to 100% | Out Gain | Off; -79.0 to 24.0 dB |
| Rvrb Time | 0.00 to 60.00 s | EarRef Lvl | -100 to 100% |
| HF Damping | 0 to 25088 Hz | Late Lvl | -100 to 100% |
| L Pre Dly | 0.0 to 230.0 ms | R Pre Dly | 0.0 to 230.0 ms |

PAGE 1 (TQ Place)

| | | | |
|---|---|---|---|
| Wet/Dry | -100 to 100% | Out Gain | Off; -79.0 to 24.0 dB |
| Absorption | 0 to 100% | EarRef Lvl | -100 to 100% |
| HF Damping | 0 to 25088 Hz | Late Lvl | -100 to 100% |
| L Pre Dly | 0.0 to 230.0 ms | R Pre Dly | 0.0 to 230.0 ms |

PAGE 2 (TQ Verb)

| | | | |
|---|---|---|---|
| Room Type | Hall1, ... | TrebShlf F | 16 to 25088 Hz |
| Size Scale | 0.00 to 2.50x | TrebShlf G | -79.0 to 24.0 dB |
| InfinDecay | On or Off | DiffAmtScl | 0.00 to 2.00 x |
| | | DiffLenScl | 0.00 to 2.50 x |
| LF Split | 16 to 25088 Hz | LFO Rate | 0.01 to 10.00 Hz |
| LF Time | 0.50 to 1.50 x | LFO Depth | 0.0 to 100.0 ct |

### PAGE 2 (TQ Place)

| | | | |
|---|---|---|---|
| Room Type | Booth1, ... | TrebShlf F | 16 to 25088 Hz |
| Size Scale | 0.00 to 2.50x | TrebShlf G | -79.0 to 24.0 dB |
| | | DiffAmtScl | 0.00 to 2.00 x |
| | | DiffLenScl | 0.00 to 2.50 x |
| LF Split | 16 to 25088 Hz | LFO Rate | 0.01 to 10.00 Hz |
| LF Time | 0.50 to 1.50 x | LFO Depth | 0.0 to 100.0 ct |

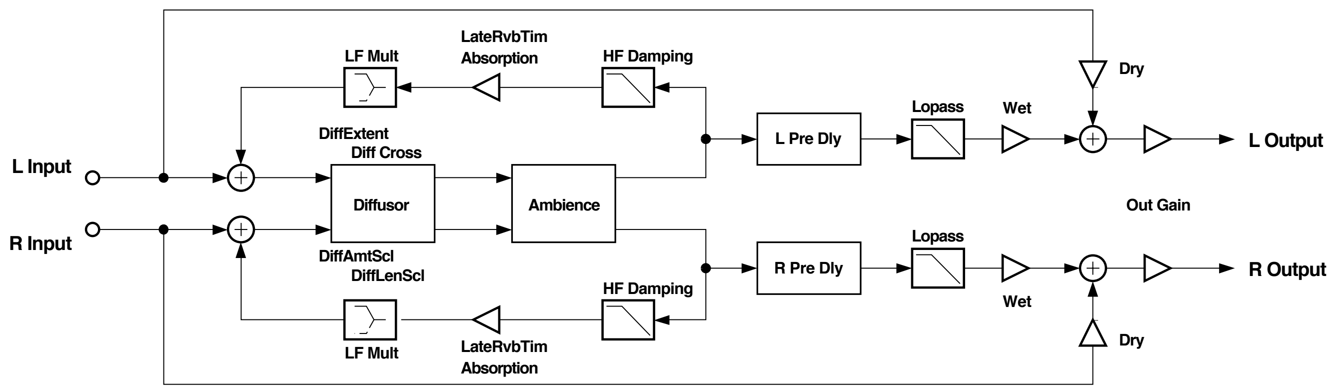
### PAGE 3

| | | | |
|---|---|---|---|
| Inj Build | -100 to 100% | Inj LP | 16 to 25088 Hz |
| Inj Spread | 0.00 to 2.50 x | | |
| E DiffAmt | -100 to 100% | E Build | -100 to 100% |
| E DfLenScl | 0.00 to 2.50 x | E Fdbk Amt | -100 to 100% |
| E DlyScl | 0.00 to 2.50 x | E HF Damp | 16 to 25088 Hz |
| E PreDlyL | 0.0 to 150.0 ms | E PreDlyR | 0.0 to 150.0 ms |

## Diffuse Verb and Diffuse Place:

Diffuse reverbs are 3-PAU algorithms and are characterized as such because of the initial burst of diffusion inherent in the onset of the reverb. Each of these algorithms consists of an input diffuser; ambience generator with a lowpass filter, low shelving filter, and LFO moving delays; and predelay.

In the Diffuse reverbs, the diffuser is implemented a little differently. The diffuser is just inside the ambience generation loop, so changes in diffusion create changes in the reverb decay. The Diffuse reverbs also offer DiffExtent and Diff Cross parameters. DiffExtent selects one of seven arbitrary gate-time lengths of the initial diffusion burst, while Diff Cross adjusts the combination of left and right channels that are diffused.




**Signal flow of Diffuse Verb and Diffuse Place**

PAGE 1 (Diffuse Verb)

| | | | |
|---|---|---|---|
| Wet/Dry | -100 to 100% | Out Gain | Off; -79.0 to 24.0 dB |
| LateRvbTim | 0.00 to 60.00 s | | |
| HF Damping | 0 to 25088 Hz | Lopass | 16 to 25088 Hz |
| L Pre Dly | 0.0 to 230.0 ms | R Pre Dly | 0.0 to 230.0 ms |

PAGE 1 (Diffuse Place)

| | | | |
|---|---|---|---|
| Wet/Dry | -100 to 100% | Out Gain | Off; -79.0 to 24.0 dB |
| Absorption | 0 to 100% | | |
| HF Damping | 0 to 25088 Hz | Lopass | 16 to 25088 Hz |
| L Pre Dly | 0.0 to 230.0 ms | R Pre Dly | 0.0 to 230.0 ms |

PAGE 2 (Diffuse Verb)

| | | | |
|---|---|---|---|
| Room Type | Hall1,... | DiffExtent | 1 to 7 x |
| Size Scale | 0.01 to 2.50x | Diff Cross | -100 to 100% |
| InfinDecay | On or Off | DiffAmtScl | 0.00 to 2.00 x |
| | | DiffLenScl | 0.01 to 2.50 x |
| LF Split | 16 to 25088 Hz | LFO Rate | 0.01 to 10.00 Hz |
| LF Time | 0.50 to 1.50 x | LFO Depth | 0.0 to 100.0 ct |

PAGE 2 (Diffuse Place)

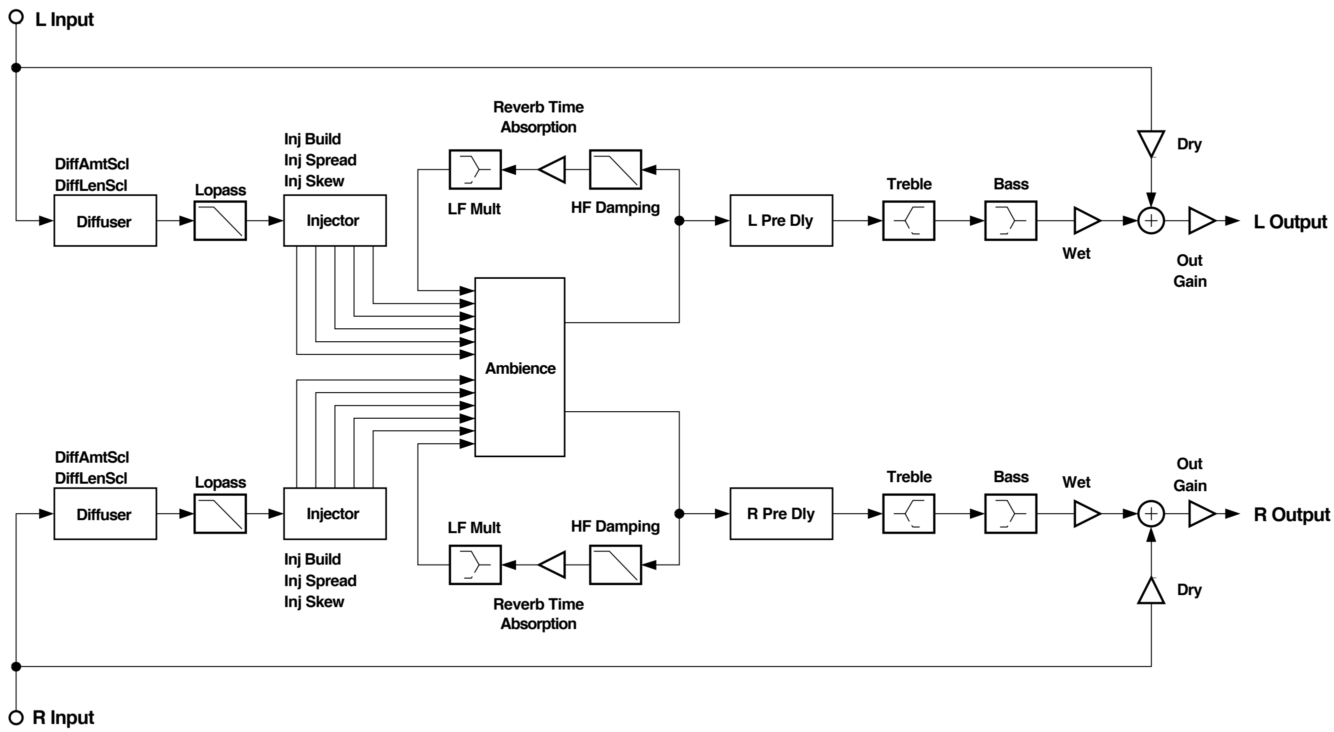
| | | | |
|---|---|---|---|
| Room Type | Booth1, ... | DiffExtent | 1 to 7 x |
| Size Scale | 0.01 to 2.50x | Diff Cross | -100 to 100% |
| | | DiffAmtScl | 0.00 to 2.00 x |
| | | DiffLenScl | 0.01 to 2.50 x |
| LF Split | 16 to 25088 Hz | LFO Rate | 0.01 to 10.00 Hz |
| LF Time | 0.50 to 1.50 x | LFO Depth | 0.0 to 100.0 ct |

## OmniVerb and OmniPlace:

Omni reverbs are 3-PAU algorithms that consist of an input diffuser; injector; ambience generator with a lowpass filter, low shelving filter, and LFO moving delays; and predelay.

The Expanse parameter adjusts the amount of reverb energy that is fed to the edges of the stereo image. A value of 0% will concentrate energy in the center of the image, while non-zero values will spread it out. Positive and negative values will impose different characteristics on the reverb image.

At the output of the reverb are a pair each of low-shelving and high-shelving filters.




Signal flow of OmniVerb and OmniPlace

### PAGE 1 (OmniVerb)

| | | | |
|---|---|---|---|
| Wet/Dry | -100 to 100% | Out Gain | Off; -79.0 to 24.0 dB |
| Rvrb Time | 0.00 to 60.00 s | | |
| HF Damping | 0 to 25088 Hz | Lopass | 16 to 25088 Hz |
| L Pre Dly | 0.0 to 230.0 ms | R Pre Dly | 0.0 to 230.0 ms |

### PAGE 1 (OmniPlace)

| | | | |
|---|---|---|---|
| Wet/Dry | -100 to 100% | Out Gain | Off; -79.0 to 24.0 dB |
| Absorption | 0 to 100% | | |
| HF Damping | 0 to 25088 Hz | Lopass | 16 to 25088 Hz |
| L Pre Dly | 0.0 to 230.0 ms | R Pre Dly | 0.0 to 230.0 ms |

**PAGE 2 (OmniVerb)**

| | | | |
|---|---|---|---|
| Room Type | Hall1, ... | Expanse | -100 to 100% |
| Size Scale | 0.00 to 2.50x | | |
| InfinDecay | On or Off | DiffAmtScl | 0.00 to 2.00 x |
| | | DiffLenScl | 0.00 to 4.50 x |
| LF Split | 16 to 25088 Hz | LFO Rate | 0.01 to 10.00 Hz |
| LF Time | 0.50 to 1.50 x | LFO Depth | 0.0 to 100.0 ct |

**PAGE 2 (OmniPlace)**

| | | | |
|---|---|---|---|
| Room Type | Booth1, ... | Expanse | -100 to 100% |
| Size Scale | 0.00 to 2.50x | | |
| | | DiffAmtScl | 0.00 to 2.00 x |
| | | DiffLenScl | 0.00 to 4.50 x |
| LF Split | 16 to 25088 Hz | LFO Rate | 0.01 to 10.00 Hz |
| LF Time | 0.50 to 1.50 x | LFO Depth | 0.0 to 100.0 ct |

**PAGE 3**

| | | | |
|---|---|---|---|
| | | TrebShlf F | 16 to 25088 Hz |
| Inj Build | -100 to 100% | TrebShlf G | -79.0 to 24.0 dB |
| Inj Spread | 0.00 to 4.50 x | BassShlf F | 16 to 25088 Hz |
| Inj Skew | -200 to 200 ms | BassShlf G | -79.0 to 24.0 dB |

<u>Absorption</u> This controls the amount of reflective material that is in the space being emulated, much like an acoustical absorption coefficient. The lower the setting, the longer it will take for the sound to die away. A setting of 0% will cause an infinite decay time.

<u>Rvrb Time</u> Adjusts the basic decay time of the late portion of the reverb.

<u>LateRvbTim</u> Adjusts the basic decay time of the late portion of the reverb after diffusion.

<u>HF Damping</u> This controls the amount of high frequency energy that is absorbed as the reverb decays. The values set the cutoff frequency of the 1-pole (6dB/oct) lowpass filter within the reverb feedback loop.

<u>L Pre Dly, R Pre Dly</u> These control the amount that each channel of the reverb is delayed relative to the dry signal. Setting different lengths for both channels can de-correlate the center portion of the reverb image and make it seem wider. This only affects the late reverb in algorithms that have early reflections.

<u>Lopass</u> Controls the cutoff frequency of a 1-pole (6dB/oct) lowpass filter at the output of the reverb. This only affects the late reverb in algorithms that have early reflections.

<u>EarRef Lvl</u> Adjusts the mix level of the early reflection portion of algorithms offering early reflections.

<u>Late Lvl</u> Adjusts the mix level of the late reverb portion of algorithms offering early reflections.

Room Type — This parameter selects the basic type of reverb being emulated, and should be the starting point when creating your own reverb presets. Due to the inherent complexity of reverb algorithms and the sheer number of variables responsible for their character, the Room Type parameter provides condensed preset collections of these variables. Each Room Type preset has been painstakingly selected by Kurzweil engineers to provide the best sounding collection of mutually complementary variables modeling an assortment of reverb families. When a room type is selected, an entire incorporated set of delay lengths and diffusion settings are established within the algorithm. By using the Size Scale, DiffAmtScl, DiffLenScl, and Inj Spread parameters, you may scale individual elements away from their preset value. When set to 1.00x, each of these elements is accurately representing its preset value determined by the current Room Type.

Room Types with similar names in different reverb algorithms do not sound the same. For example, Hall1 in Diffuse Verb does not sound the same as Hall1 in TQ Verb.

Size Scale — This parameter scales the inherent size of the reverb chosen by Room Type. For a true representation of the selected Room Type size, set this to 1.00x. Scaling the size below this will create smaller spaces, while larger scale factors will create large spaces.

InfinDecay — Found in “Verb” algorithms. When turned “On”, the reverb tail will decay indefinitely. However, certain HF Damping settings may reduce this effect, and cause the tail to taper away. When turned “Off”, the decay time is determined by the “Rvrb Time” or “LateRvbTim” parameters. This parameter is an excellent candidate for a KDFX Modulation, using a switch pedal as a source.

LF Split — Used in conjunction with LF Time. This controls the upper-frequency limit of the low-frequency decay time multiplier. Energy below this frequency will decay faster or slower depending on the LF Time parameter.

LF Time — Used in conjunction with LF Split. This modifies the decay time of the energy below the LF Split frequency. A setting of 1.00x will make low-frequency energy decay at the rate determined by the decay time. Higher values will cause low-frequency energy to decay slower, and lower values will cause it to decay more quickly.

TrebShlf F — Adjusts the frequency of a high-shelving filter at the output of the late reverb.

TrebShlf G — Adjusts the gain of a high-shelving filter at the output of the late reverb.

BassShlf F — Adjusts the frequency of a low-shelving filter at the output of the late reverb.

BassShlf G — Adjusts the gain of a low-shelving filter at the output of the late reverb.

DiffAmtScl — Adjusts the amount of diffusion at the onset of the reverb. For a true representation of the selected Room Type diffusion amount, set this to 1.00x.

DiffLenScl — Adjusts the length of the diffusion at the onset of the reverb. For a true representation of the selected Room Type diffusion length, set this to 1.00x.

DiffExtent — Adjust the onset diffusion duration. Higher values create longer diffuse bursts at the onset of the reverb.

Diff Cross — Adjusts the onset diffusion cross-coupling character. Although subtle, this parameter bleeds left and right channels into each other during onset diffusion, and also in the body of the reverb. 0% setting will disable this. Increasing this value in either the positive or negative direction will increase its effect.

Expanse — Amount of late reverb energy biased toward the edges of the stereo image. A setting of 0% will bias energy towards the center. Moving away from 0% will bias energy towards the sides. Positive and negative values will have a different character.

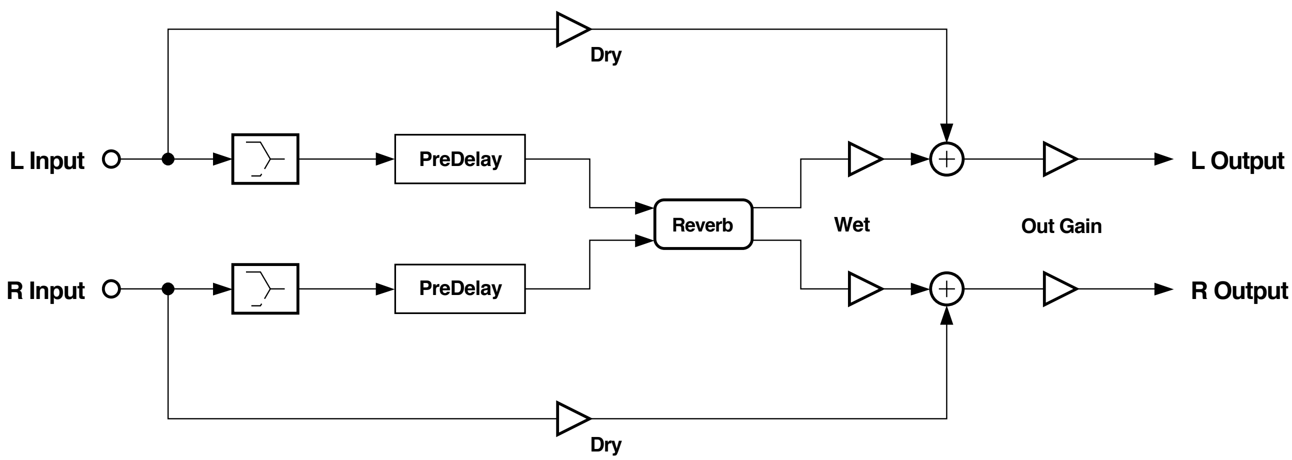
| | |
|---|---|
| LFO Depth | Adjusts the detuning depth in cents caused by a moving reverb delay line. Moving delay lines can imitate voluminous flowing air currents and reduce unwanted artifacts like ringing and flutter when used properly. Depth settings under 1.5ct with LFO Rate settings under 1.00Hz are recommended for modeling real spaces. High depth settings can create chorusing qualities, which won't be suitable for real acoustic spaces, but can nonetheless create interesting effects. Instruments that have little or no inherent pitch fluctuation (like piano) are much more sensitive to this LFO than instruments that normally have a lot of vibrato (like voice) or non-pitched instruments (like snare drum). |
| LFO Rate | Adjusts the rate at which the moving reverb delay lines move. |
| Inj Build | Used in conjunction with Inj Spread, this adjusts the envelope of the onset of the reverb. Specifically, it tapers the amplitudes of a series of delayed signals injected into the body of the reverb. Values above 0% will produce a faster build, while values below 0% will cause the build to be more gradual. |
| Inj Spread | Used in conjunction with Inj Build, this scales the length of the series of delays injected into the body of the reverb. For a true representation of the selected Room Type injector spread, set this to 1.00x. |
| Inj LP | This adjusts the cutoff frequency of a 1 pole (6dB/oct) lowpass filter applied to the signal being injected into the body of the reverb. |
| Inj Skew | Adjusts the amount of delay applied to either the left or right channel of the reverb injector. Positive values delay the right channel while negative values delay the left channel. |
| E DiffAmt | Adjusts the amount of diffusion applied to the early reflection network. |
| E DfLenScl | Adjusts the length of diffusion applied to the early reflection network. This is influenced by E PreDlyL and E PreDlyR. |
| E Dly Scl | Scales the delay lengths inherent in the early reflection network. |
| E Build | Adjusts the envelope of the onset of the early reflections. Values above 0% will create a faster attack while values below 0% will create a slower attack. |
| E Fdbk Amt | Adjusts the amount of the output of an early reflection portion that is fed back into the input of the opposite channel in front of the early pre-delays. Overall, it lengthens the decay rate of the early reflection network. Negative values polarity-invert the feedback signal. |
| E HF Damp | This adjusts the cutoff frequency of a 1-pole (6dB/oct) lowpass filter applied to the early reflection feedback signal. |
| E PreDlyL, E PreDlyR | Adjusts how much the early reflections are delayed relative to the dry signal. These are independent of the late reverb predelay times, but will influence E Dly Scl. |
| E Dly L, E Dly R | Adjusts the left and right early reflection delays fed to the same output channels. |
| E Dly LX, E Dly RX | Adjusts the left and right early reflection delays fed to the opposite output channels. |
| E DifDlyL, E DifDlyR | Adjusts the diffusion delays of the diffusers on delay taps fed to the same output channels. |
| E DifDlyLX, E DifDlyRX | Adjusts the diffusion delays of the diffusers on delay taps fed to the opposite output channels. |
| E X Blend | Adjusts the balance between early reflection delay tap signals with diffusers fed to their same output channel, and those fed to opposite channels. 0% allows only delay taps being fed to opposite output channels to be heard, while 100% allows only delay taps going to the same channels to be heard. |

# FXAlg #12: Panaural Room

### Room reverberation algorithm

*Allocation Units:* 3

The Panaural Room reverberation is implemented using a special network arrangement of many delay lines that guarantees colorless sound. The reverberator is inherently stereo with each input injected into the “room” at multiple locations. The signals entering the reverberator first pass through a shelving bass equalizer with a range of +/-15dB. To shorten the decay time of high frequencies relative to mid frequencies, lowpass filters controlled by HF Damping are distributed throughout the network. Room Size scales all the delay times of the network (but not the Pre Dly or Build Time), to change the simulated room dimension over a range of 1 to 16m. Decay Time varies the feedback gains to achieve decay times from 0.5 to 100 seconds. The Room Size and Decay Time controls are interlocked so that a chosen Decay Time will be maintained while Room Size is varied. A two-input stereo mixer, controlled by Wet/Dry and Out Gain, feeds the output.




**Simplified block diagram of Panaural Room.**

The duration and spacing of the early reflections are influenced by Room Size and Build Time, while the number and relative loudness of the individual reflections are influenced by Build Env. When Build Env is near 0 or 100%, fewer reflections are created. The maximum number of important early reflections, 13, is achieved at a setting of 50%.

To get control over the growth of reverberation, the left and right inputs each are passed through an “injector” that can extend the source before it drives the reverberator. Only when Build Env is set to 0% is the reverberator driven in pure stereo by the pure dry signal. For settings of Build Env greater than 0%, the reverberator is fed multiple times. Build Env controls the injector so that the reverberation begins abruptly (0%), builds immediately to a sustained level (50%), or builds gradually to a maximum (100%). Build Time varies the injection length over a range of 0 to 500ms. At a Build Time of 0ms, there is no extension of the build time. In this case, the Build Env control adjusts the density of the reverberation, with maximum density at a setting of 50%. In addition to the two build controls, there is an overall Pre Dly control that can delay the entire reverberation process by up to 500ms.

## Parameters:

### PAGE 1

| | | | |
|---|---|---|---|
| Wet/Dry | 0 to 100%wet | Out Gain | Off, -79.0 to 24.0 |
| Room Size | 1.0 to 16.0 m | | |
| Pre Dly | 0 to 500 ms | Decay Time | 0.5 to 100.0 s |
| HF Damping | 16 to 25088 Hz | | |

### PAGE 2

| | | | |
|---|---|---|---|
| Bass Gain | -15 to 15 dB | Build Time | 0 to 500 ms |
| | | Build Env | 0 to 100% |

Wet/Dry — The amount of the stereo reverberator (wet) signal relative to the original input (dry) signal to be output. The dry signal is not affected by the Bass Gain control. The wet signal is affected by the Bass Gain control and by all the other reverberator controls. The balance between wet and dry signals is an extremely important factor in achieving a good mix. Emphasizing the wet signal gives the effect of more reverberation and of greater distance from the source.

Out Gain — The overall output level for the reverberation effect, and controls the level for both the wet and dry signal paths.

Decay Time — The reverberation decay time (mid-band "$RT_{60}$"), the time required before the reverberation has died away to 60dB below its "running" level. Adjust decay time according to the tempo and articulation of the music and to taste.

HF Damping — Adjusts lowpass filters in the reverberator so that high frequencies die away more quickly than mid and low frequencies. This shapes the reverberation for a more natural, more acoustically accurate sound.

Bass Gain — Shapes the overall reverberation signal's bass content, but does not modify the decay time. Reduce the bass for a less muddy sound, raise it slightly for a more natural acoustic effect.

Room Size — Choosing an appropriate room size is very important in getting a good reverberation effect. For impulsive sources, such as percussion instruments or plucked strings, increase the size setting until discrete early reflections become audible, and then back it off slightly. For slower, softer music, use the largest size possible. At lower settings, Room Size leads to coloration, especially if the Decay Time is set too high.

Pre Dly — Introducing predelay creates a gap of silence that allows the dry signal to stand out with greater clarity and intelligibility against the reverberant background. This is especially helpful with vocal or classical music.

Build Time — Similar to predelay, but more complex, larger values of Build Time slow down the building up of reverberation and can extend the build-up process. Experiment with Build Time and Build Env and use them to optimize the early details of reverberation. A Build Time of 0ms and a Build Env of 50% is a good default setting that yields a fast arriving, maximally dense reverberation.

Build Env

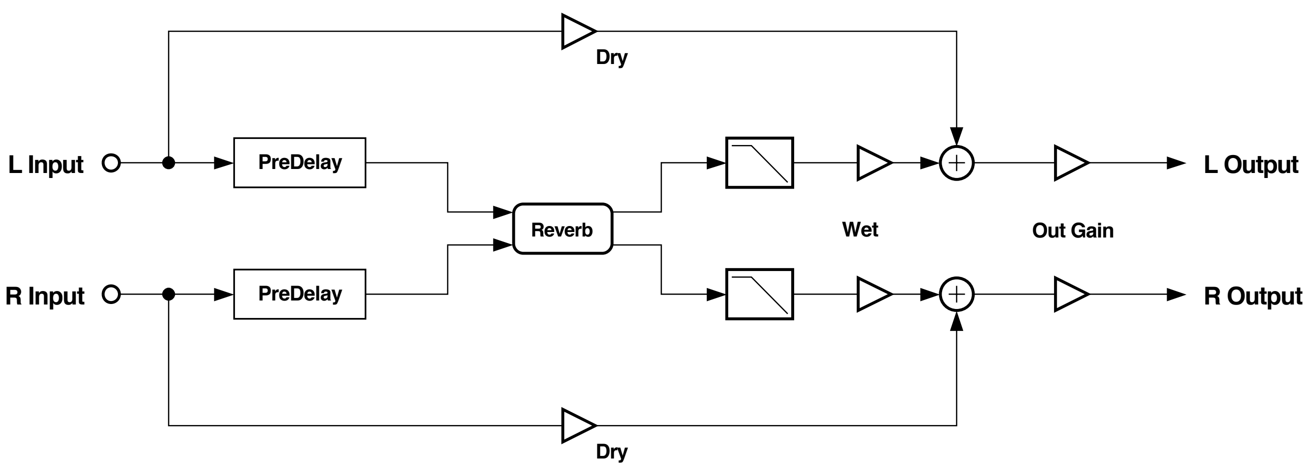
When Build Time has been set to greater than about 80ms, Build Env begins to have an audible influence on the early unfolding of the reverberation process. For lower-density reverberation that starts cleanly and impulsively, use a setting of 0%. For the highest-density reverberation, and for extension of the build-up period, use a setting of 50%. For an almost reverse reverberation, set Build Env to 100%. You can think of Build Env as setting the position of a see-saw. The left end of the see-saw represents the driving of the reverberation at the earliest time, the pivot point as driving the reverberation at mid-point in the time sequence, and the right end as the last signal to drive the reverberator. At settings near 0%, the see-saw is tilted down on the right: the reverberation starts abruptly and the drive drops with time. Near 50%, the see-saw is level and the reverberation is repeatedly fed during the entire build time. At settings near 100%, the see-saw is tilted down on the left, so that the reverberation is hit softly at first, and then at increasing level until the end of the build time.

# FXAlg #13: Stereo Hall

### A stereo hall reverberation algorithm

*Allocation Units:* 3

The Stereo Hall reverberation is implemented using a special arrangement of all pass networks and delay lines, which reduces coloration and increases density. The reverberator is inherently stereo with each input injected into the “room” at multiple locations. To shorten the decay time of low and high frequencies relative to mid frequencies, bass equalizers and lowpass filters, controlled by Bass Gain and by HF Damping, are placed within the network. Room Size scales all the delay times of the network (but not the Pre Dly or Build Time), to change the simulated room dimension over a range of 10 to 75m. Decay Time varies the feedback gains to achieve decay times from 0.5 to 100 seconds. The Room Size and Decay Time controls are interlocked so that a chosen Decay Time will be maintained while Room Size is varied. At smaller sizes, the reverb becomes quite colored and is useful only for special effects. A two-input stereo mixer, controlled by Wet/Dry and Out Gain, feeds the output. The Lowpass control acts only on the wet signal and can be used to smooth out the reverb high end without modifying the reverb decay time at high frequencies.




**Simplified block diagram of Stereo Hall.**

Within the reverberator, certain delays can be put into a time varying motion to break up patterns and to increase density in the reverb tail. Using the LFO Rate and Depth controls carefully with longer decay times can be beneficial. But beware of the pitch-shifting artifacts which can accompany randomization when it is used in greater amounts. Also within the reverberator, the Diffusion control can reduce the diffusion provided by some all pass-networks. While the reverb will eventually reach full diffusion regardless of the Diffusion setting, the early reverb diffusion can be reduced, which sometimes is useful to help keep the dry signal “in the clear”.

The reverberator structure is stereo and requires that the dry source be applied to both left and right inputs. If the source is mono, it should still be applied (pan centered) to both left and right inputs. Failure to drive both inputs will result in offset initial reverb images and later ping-ponging of the reverberation. Driving only one input will also increase the time required to build up reverb density.

To gain control over the growth of reverberation, the left and right inputs each are passed through an "injector" that can extend the source before it drives the reverberator. Only when Build Env is set to 0% is the reverberator driven in pure stereo by the pure dry signal. For settings of Build Env greater than 0%, the reverberator is fed multiple times. Build Env controls the injector so that the reverberation begins abruptly (0%), builds immediately to a sustained level (50%), or builds gradually to a maximum (100%). Build Time varies the injection length over a range of 0 to 500ms. At a Build Time of 0ms, there is no extension of the build time. In this case, the Build Env control adjusts the density of the reverberation, with maximum density at a setting of 50%. In addition to the two build controls, there is an overall Pre Dly control that can delay the entire reverberation process by up to 500ms.

## Parameters:

### PAGE 1

| | | | |
|---|---|---|---|
| **Wet/Dry** | **0 to 100%wet** | **Out Gain** | **Off, -79.0 to 24.0 dB** |
| **Room Size** | **2.0 to 15.0 m** | **Diffusion** | **0 to 100%** |
| **Pre Dly** | **0 to 500 ms** | **Decay Time** | **0.5 to 100.0 ms** |
| **HF Damping** | **16 to 25088 Hz** | | |

### PAGE 2

| | | | |
|---|---|---|---|
| **Bass Gain** | **-15 to 0 dB** | **Build Time** | **0 to 500 ms** |
| **Lowpass** | **16 to 25088 Hz** | **Build Env** | **0 to 100%** |
| **LFO Rate** | **0.00 to 5.10 Hz** | | |
| **LFO Depth** | **0.00 to 10.20 ct** | | |

Wet/Dry — The amount of the stereo reverberator (wet) signal relative to the original input (dry) signal to be output. The dry signal is not affected by the Bass Gain control. The wet signal is affected by the Bass Gain control and by all the other reverberator controls. The balance between wet and dry signals is an extremely important factor in achieving a good mix. Emphasizing the wet signal gives the effect of more reverberation and of greater distance from the source.

Out Gain — The overall output level for the reverberation effect, and controls the level for both the wet and dry signal paths.

Decay Time — The reverberation decay time (mid-band "$RT_{60}$"), the time required before the reverberation has died away to 60dB below its "running" level. Adjust decay time according to the tempo and articulation of the music and to taste.

HF Damping — Adjusts lowpass filters in the reverberator so that high frequencies die away more quickly than mid and low frequencies. This shapes the reverberation for a more natural, more acoustically accurate sound.

Room Size — Choosing an appropriate room size is very important in getting a good reverberation effect. For impulsive sources, such as percussion instruments or plucked strings, increase the size setting until discrete early reflections become audible, and then back it off slightly. For slower, softer music, use the largest size possible. At lower settings, RoomSize leads to coloration, especially if the DecayTime is set too high.

Bass Gain — Adjusts bass equalizers in the reverberator so that low frequencies die away more quickly than mid and high frequencies. This can be used to make the reverberation less muddy.

Lowpass — Used to shape the overall reverberation signal's treble content, but does not modify the decay time. Reduce the treble for a softer, more acoustic sound.

Pre Dly — Introducing predelay creates a gap of silence that allows the dry signal to stand out with greater clarity and intelligibility against the reverberant background. This is especially helpful with vocal or classical music.

Build Time — Similar to predelay, but more complex, larger values of BuildTime slow down the building up of reverberation and can extend the build up process. Experiment with BuildTime and BuildEnv and use them to optimize the early details of reverberation. A BuildTime of 0ms and a BuildEnv of 50% is a good default setting that yields fast arriving, natural reverberation.

Build Env — When BuildTime has been set to greater than about 80ms, BuildEnv begins to have an audible influence on the early unfolding of the reverberation process. For lower-density reverberation that starts cleanly and impulsively, use a setting of 0%. For the highest-density reverberation, and for extension of the build-up period, use a setting of 50%. For an almost reverse reverberation, set BuildEnv to 100%. You can think of BuildEnv as setting the position of a see-saw. The left end of the see-saw represents the driving of the reverberation at the earliest time, the pivot point as driving the reverberation at mid-point in the time sequence, and the right end as the last signal to drive the reverberator. At settings near 0%, the see-saw is tilted down on the right: the reverberation starts abruptly and the drive drops with time. Near 50%, the see-saw is level and the reverberation is repeatedly fed during the entire build time. At settings near 100%, the see-saw is tilted down on the left, so that the reverberation is hit softly at first, and then at increasing level until the end of the build time.

LFO Rate and LFO Depth — Within the reverberator, the certain delay values can be put into a time varying motion to break up patterns and to increase density in the reverb tail. Using the LFO Rate and Depth controls carefully with longer decay times can be beneficial. But beware of the pitch-shifting artifacts which can accompany randomization when it is used in greater amounts.

Diffusion — Within the reverberator, the Diffusion control can reduce the diffusion provided by some of the all-pass networks. While the reverb will eventually reach full diffusion regardless of the Diffusion setting, the early reverb diffusion can be reduced, which sometimes is useful to help keep the dry signal "in the clear."

# FXAlg #14: Grand Plate

### A plate reverberation algorithm

*Allocation Units:* 3

This algorithm emulates an EMT 140 steel plate reverberator. Plate reverberators were manufactured during the 1950s, 60s, 70s, and perhaps into the 80s. By the end of the 1980s, they had been supplanted in the marketplace by digital reverberators, which first appeared in 1976. While a handful of companies made plate reverberators, EMT (Germany) was the best known and most popular.

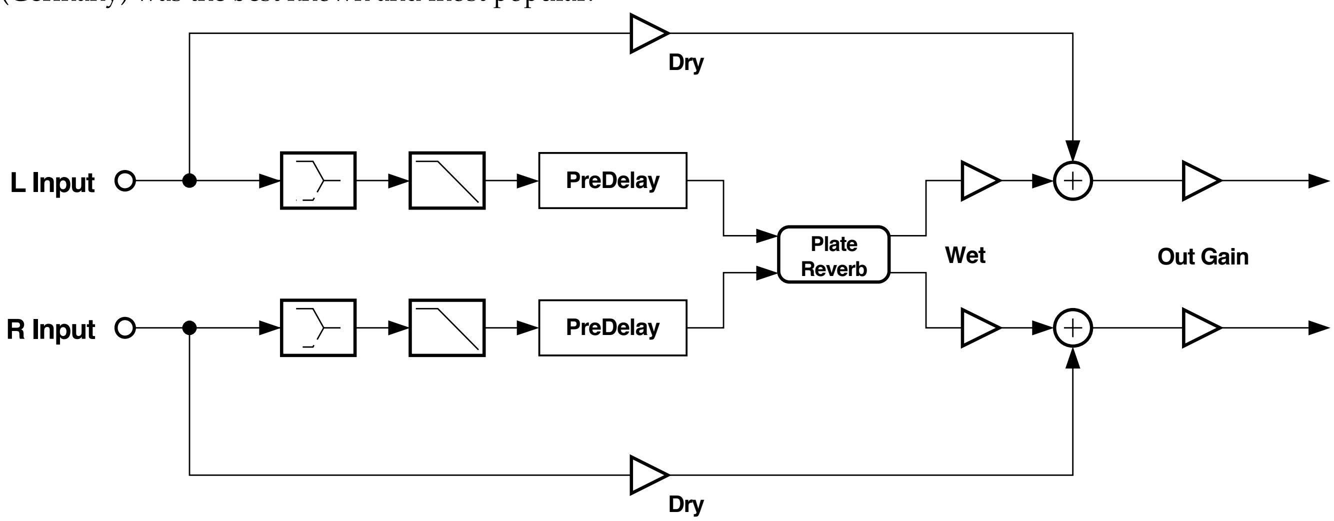



**Diagram of Grand Plate reverb**

A plate reverberator is generally quite heavy and large, perhaps 4 feet high by 7 feet long, and a foot thick. They were only slightly adjustable, with controls for high frequency damping and decay time. Some were stereo in/stereo out, others mono in/mono out.

A plate reverb begins with a sheet of plate steel suspended by its edges, leaving the plate free to vibrate. At one (or two) points on the plate, an electromagnetic driver (sort of a small loudspeaker without a cone) is arranged to couple the dry signal into the plate, sending sound vibrations into the plate in all directions. At one or two other locations, a pickup is placed, sort of like a dynamic microphone whose diaphragm is the plate itself, to pick up the reverberation.

Since the sound waves travel very rapidly in steel (faster than they do in air), and since the dimensions of the plate are not large, the sound quickly reaches the plate edges and reflects from them. This results in a very rapid build-up of the reverberation, essentially free of early reflections and with no distinguishable gap before the onset of reverb.

Plates offered a wonderful sound of their own, easily distinguished from other reverberators in the pre-digital reverb era, such as springs or actual "echo" chambers. Plates were bright and diffused (built up echo density) rapidly. Curiously, when we listen to a vintage plate today, we find that the much vaunted brightness is nothing like what we can accomplish digitally; we actually have to deliberately reduce the brightness of a plate emulation to match the sound of a real plate. Similarly, we find that we must throttle back on the low frequency content as well.

The algorithm developed for Grand Plate was carefully crafted for rapid diffusion, low coloration, freedom from discrete early reflections, and "brightness". We also added some controls that were never present in real plates: size, predelay of up to 500ms, LF damping, lowpass roll off, and bass roll off. Furthermore, we allow a wider range of decay time adjustment than a conventional plate. Once the algorithm was complete, we tuned it by listening to the original EMT reverb on one channel and the Grand Plate emulation on the other. A lengthy and careful tuning of Grand Plate (tuning at the micro detail level of each delay and gain in the algorithm) was carried out until the stereo spread of this reverb was matched in all the time periods—early, middle, and late.

The heart of this reverb is the plate simulation network, with its two inputs and two outputs. It is a full stereo reverberation network, which means that the left and right inputs get slightly different treatment in the reverberator. This yields a richer, more natural stereo image from stereo sources. If you have a mono source, assign it to both inputs for best results.

The incoming left source is passed through predelay, lowpass (Lowpass), and bass-shelf (Bass Gain) blocks. The right source is treated similarly.

There are lowpass filters (HF Damping) and high pass filters (LF Damping) embedded in the plate simulation network to modify the decay times. The reverb network also accommodates the Room Size and Decay Time controls.

An output mixer assembles dry and wet signals.

## Parameters:

### PAGE 1

| | | | |
|---|---|---|---|
| Wet/Dry | 0 to 100%wet | Out Gain | Off, -79.0 to 24.0 dB |
| Room Size | 1.00 to 4.00 m | | |
| Pre Dly | 0 to 500 ms | Decay Time | 0.2 to 5.0 s |
| HF Damping | 16 to 25088 Hz | LF Damping | 1 to 294 Hz |

### PAGE 2

| | | | |
|---|---|---|---|
| Lowpass | 16 to 25088 Hz | Bass Gain | -15 to 0 dB |

Wet/Dry — The amount of the stereo reverberator (wet) signal relative to the original input (dry) signal sent to the output. The dry signal is not affected by the Lowpass or Bass Gain controls. The wet signal is affected by the Lowpass and Bass Gain controls and by all the other reverberator controls. The balance between wet and dry signals is an extremely important factor in achieving a good mix. Emphasizing the wet signal gives the effect of more reverberation and of greater distance from the source.

Out Gain — The overall output level for the reverberation effect and controls the level for both the wet and dry signal paths.

Room Size — Choosing an appropriate room size is very important in getting a good reverberation effect. For impulsive sources, such as percussion instruments or plucked strings, increase the size setting until discrete reflections become audible, and then back it off slightly. For slower, softer music, use the largest size possible. At lower settings, Room Size leads to coloration, especially if the Decay Time is set too high. To emulate a plate reverb, this control is typically set to 1.9m.

Pre Dly — Introducing predelay creates a gap of silence that allows the dry signal to stand out with greater clarity and intelligibility against the reverberant background. This is especially helpful with vocals or classical music.

| | |
|---|---|
| Decay Time | The reverberation decay time (mid-band "$RT_{60}$"), the time required before the reverberation has died away to 60dB below its "running" level. Adjust decay time according to the tempo and articulation of the music. To emulate a plate reverb, this control is typically set from 1 to 5 seconds. |
| HF Damping | Adjusts lowpass filters in the reverberator so that high frequencies die away more quickly than mid and low frequencies. This shapes the reverberation for a more natural, more acoustically accurate sound. To emulate a plate reverb, this control is typically set to 5920Hz. |
| LF Damping | Adjusts hipass filters in the reverberator so that low frequencies die away more quickly than mid and high frequencies. This shapes the reverberation for a more natural, more acoustically accurate sound. To emulate a plate reverb, this control is typically set to 52 Hz. |
| Lowpass | Shapes the overall reverberation signal's treble content, but does not modify the decay time. Reduce the treble for a duller, more natural acoustic effect. To emulate a plate reverb, this control is typically set to 3951Hz. |
| Bass Gain | Shapes the overall reverberation signal's bass content, but does not modify the decay time. Reduce the bass for a less muddy sound. To emulate a plate reverb, this control is typically set to -12dB. |

# FXAlg #15: Finite Verb

### “Enveloped” reverberation algorithm

*Allocation Units:* 3

In this algorithm, the left and right sources are summed before being fed into a tapped delay line, which directly simulates the impulse response of a reverberator. The taps are placed in sequence from zero delay to a maximum delay value, at quasi-regular spacings. By varying the coefficients with which these taps are summed, one can create the effect of a normal rapidly building/slowly decaying reverb or a reverse reverb which builds slowly then stops abruptly.

A special tap is picked off the tapped delay line and its length is controlled by Dly Length. It can be summed into the output wet mix (Dly Lvl) to serve as the simulated dry source that occurs after the reverse reverb sequence has built up and ended. It can also be fed back for special effects. Fdbk Lvl and HF Damping tailor the gain and spectrum of the feedback signal. Despite the complex reverb-like sound of the tapped delay line, the Feedback tap is a pure delay. Feeding it back is like reapplying the source, as in a simple tape echo.

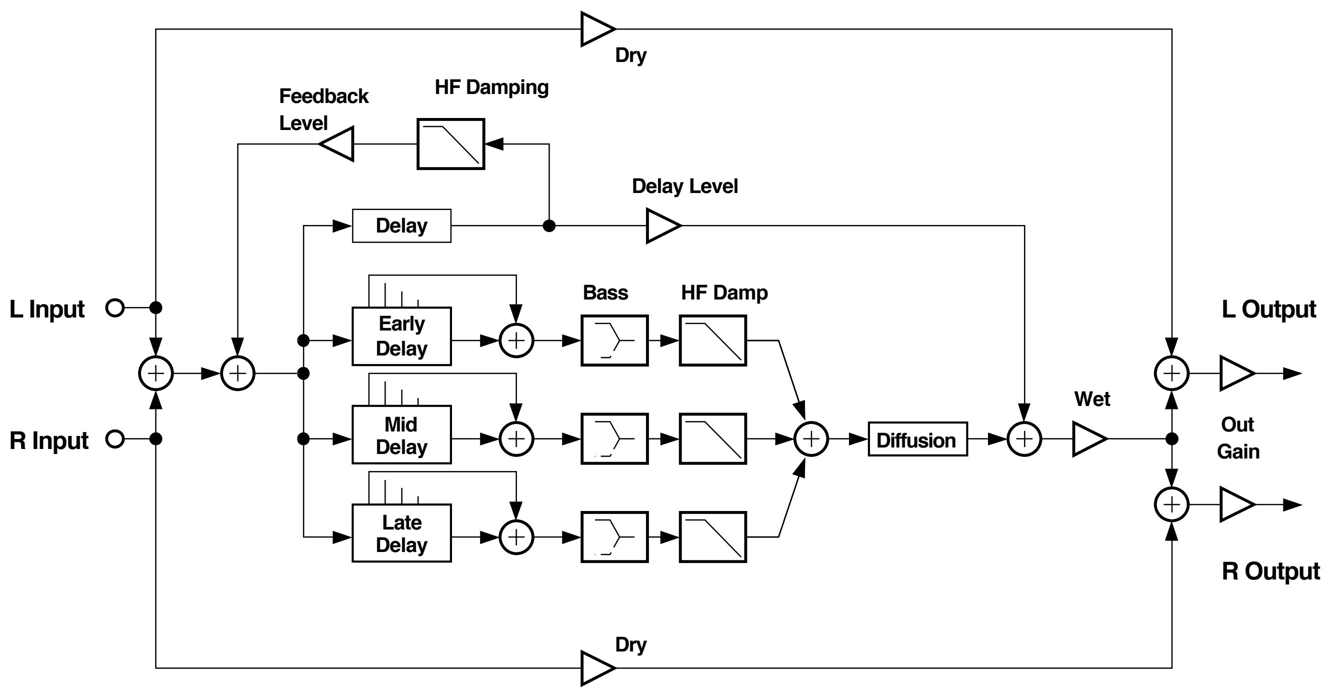



**Diagram of Finite Verb**

Dly Length and Rvb Length range from 300 to 3000 milliseconds. With the R1 Rvb Env variants, Rvb Length corresponds to a decay time ($RT_{60}$).

To make things a little more interesting, the tapped delay line mixer is actually broken into three mixers: an early, middle, and late mixer. Each mixes its share of taps and then applies the submix to a lowpass filter (cut only) and a simple bass control (boost and cut). Finally, the three equalized sub mixes are mixed into one signal. The Bass and Damp controls allow special effects such as a reverb that begins dull and increases in two steps to a brighter sound.

The Rvb Env control selects 27 cases of envelope gains for the taps. Nine cases emulate a normal forward-evolving reverb, but with some special twists. Cases FWD R1xx have a single reverb peak, with a fast attack and slower decay. The sub cases FWD R1Sx vary the sharpness of the envelope, from dullest (S1) to sharpest (S3). The sub cases FWD R2xx have two peaks; that is, the reverb builds, decays, builds again, and decays again. The sub cases FWD R3xx have three peaks.

The sub cases SYM have a symmetrical build and decay time. The cases R1 build to a single peak, while R2 and R3 have two and three peaks, respectively.

The sub cases REV simulate a reverse reverb effect. REV R1xx imitates a backward running reverb, with a long rising "tail" ending abruptly (followed, optionally, by the "dry" source mixed by Dly Lvl). Once again, the number of peaks and the sharpness are variable.

The usual Wet/Dry and Output Gain controls are provided.

## Parameters:

**PAGE 1**

| | | | |
|---|---|---|---|
| **Wet/Dry** | **0 to 100%wet** | **Out Gain** | **Off, -79.0 to 24.0 dB** |
| **Fdbk Lvl** | **0 to 100%** | | |
| **HF Damping** | **16 to 25088 Hz** | | |

**PAGE 2**

| | | | |
|---|---|---|---|
| **Dly Lvl** | **0 to 100%** | **Rvb Env** | **REV R1S1** |
| **Dly Length** | **300 to 3000 ms** | **Rvb Length** | **300 to 3000 ms** |

**PAGE 3**

| | | | |
|---|---|---|---|
| **Early Bass** | **-15 to 15 dB** | **Early Damp** | **16 to 25088 Hz** |
| **Mid Bass** | **-15 to 15 dB** | **Mid Damp** | **16 to 25088 Hz** |
| **Late Bass** | **-15 to 15 dB** | **Late Damp** | **16 to 25088 Hz** |

Wet/Dry — Wet/Dry sets the relative amount of wet signal and dry signal. The wet signal consists of the reverb itself (stereo) and the delayed mono signal arriving after the reverb has ended (simulating the dry source in the reverse reverb sequence). The amount of the delayed signal mixed to the Wet signal is separately adjustable with the Dly Lvl control. The Dry signal is the stereo input signal.

Out Gain — This controls the level of the output mix, wet and dry, sent back into KDFX.

Fdbk Lvl — This controls the feedback gain of the separate (mono) delay tap. A high value contributes a long repeating echo character to the reverb sound.

HF Damping — HF Damping adjusts a lowpass filter in the late delay tap feedback path so that high frequencies die away more quickly than mid and low frequencies.

Dly Lvl — This adjusts the level of the separate (mono) delay tap used to simulate the dry source of a reverse reverb effect. This same tap is used for feedback.

Dly Length — Sets the length (in milliseconds), of the separate (mono) delay tap used to simulate the dry source of a reverse reverb effect. This same tap is used for feedback.

Rvb Env The Rvb Env control selects 27 cases of envelope gains for the taps. Nine cases emulate a normal forward evolving reverb, another nine emulate a reverb building symmetrically to a peak at the mid point, while the last nine cases emulate a reverse-building reverb. For each major shape, there are three variants of one, two, and three repetitions and three variants of envelope sharpness.

Rvb Length Sets the length (in milliseconds), from start to finish, of the reverberation process. This parameter is essentially the decay time or $RT_{60}$ for the Rvb Env cases..R1.. where there is only one repetition.

Early Bass, Mid Bass, and Late Bass These bass controls shape the frequency response (boost or cut) of the three periods of the finite reverb sequence. Use them to tailor the way the reverb bass content changes with time.

Early Damp, Mid Damp, and Late Damp These treble controls shape the frequency response (cut only) of the three periods of the finite reverb sequence. Use them to tailor the way the reverb treble content changes with time.

# FXAlg #130: Complex Echo

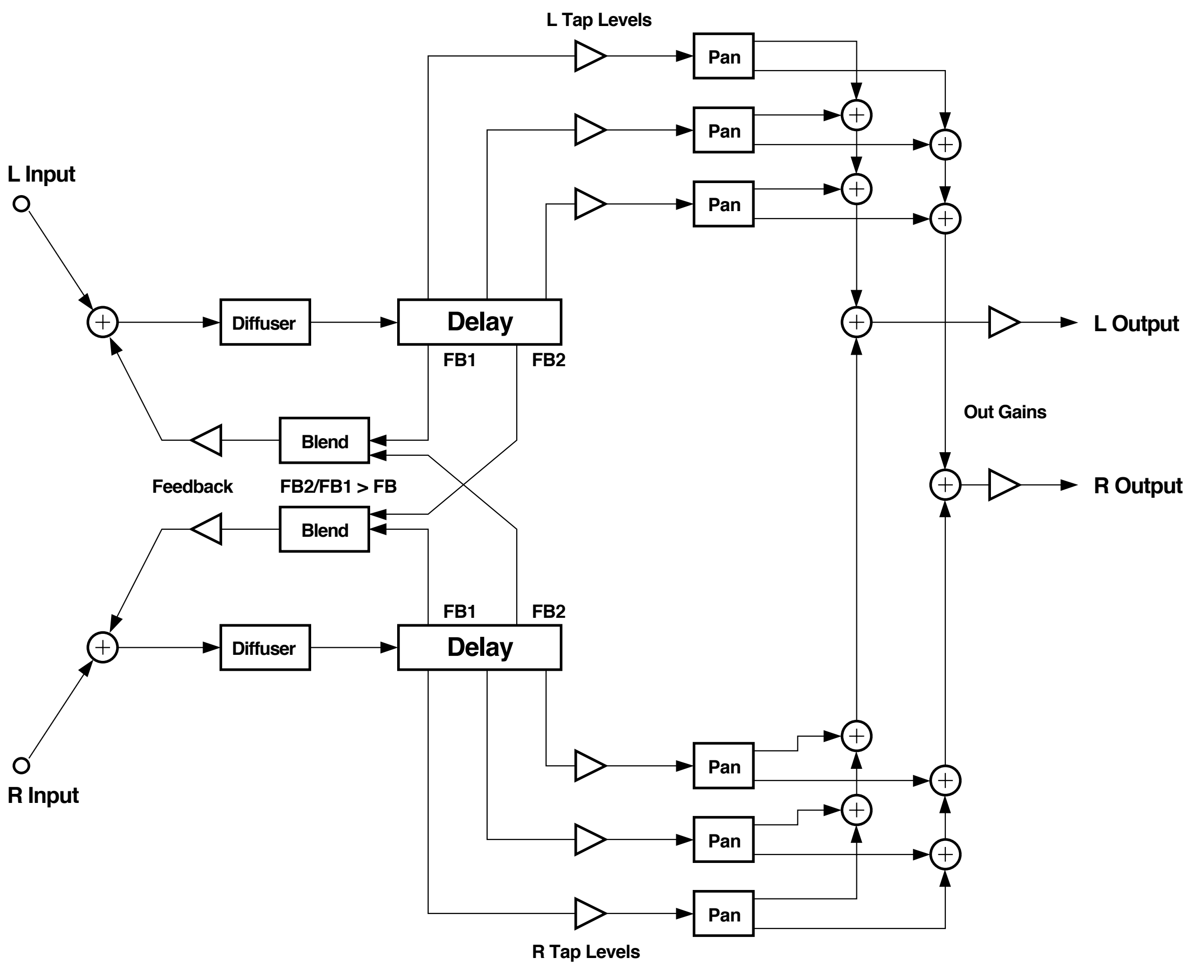
### Multitap delay line effect, consisting of 6 independent output taps and 4 independent feedback taps

*Allocation Units:* 1

Complex Echo is an elaborate delay line with three independent output taps per channel, two independent feedback taps per channel, equal-power output tap panning, feedback diffuser, and high frequency damping. Each channel has three output taps which can each be delayed up to 2600ms (2.6 sec) then panned at the output. Feedback taps can also be delayed up to 2600ms, but both feedback channels do slightly different things. Feedback line 1 feeds the signal back to the delay input of the same channel, while feedback line 2 feeds the signal back to the opposite channel—it can be considered "ping-pong" feedback. Relative levels for each feedback line can be set with the "FB2/FB1>FB" control where 0% only allows FB1 to be used, and 100% only allows FB2 to be used.

The diffuser sits at the beginning of the delay line, and consists of three controls. Separate left and right Diff Dly parameters control the length that a signal is smeared from 0 to 100ms as it passes through these diffusers. Diff Amt adjusts the smearing intensity. Short diffuser delays can diffuse the sound while large delays can drastically alter the spectral flavor. Setting all three diffuser parameters to 0 will disable the diffuser.

Also at the input to the delays are 1-pole (6dB/oct) lowpass filters controlled by the HF Damping parameter.




**Signal flow of Complex Echo**

## Parameters:

### PAGE 1

| | | | |
|---|---|---|---|
| Wet/Dry | 0 to 100%wet | Out Gain | Off, -79.0 to 24.0 dB |
| Feedback | 0 to 100% | L Diff Dly | 0 to 100 ms |
| FB2/FB1>FB | 0 to 100% | R Diff Dly | 0 to 100 ms |
| HF Damping | 16 to 25088 Hz | Diff Amt | 0 to 100% |

### PAGE 2

| | | | |
|---|---|---|---|
| L Fdbk1 Dly | 0 to 2600 ms | R Fdbk1 Dly | 0 to 2600 ms |
| L Fdbk2 Dly | 0 to 2600 ms | R Fdbk2 Dly | 0 to 2600 ms |
| L Tap1 Dly | 0 to 2600 ms | R Tap1 Dly | 0 to 2600 ms |
| L Tap2 Dly | 0 to 2600 ms | R Tap2 Dly | 0 to 2600 ms |
| L Tap3 Dly | 0 to 2600 ms | R Tap3 Dly | 0 to 2600 ms |

### PAGE 3

| | | | |
|---|---|---|---|
| L Tap1 Lvl | 0 to 100% | R Tap1 Lvl | 0 to 100% |
| L Tap2 Lvl | 0 to 100% | R Tap2 Lvl | 0 to 100% |
| L Tap3 Lvl | 0 to 100% | R Tap3 Lvl | 0 to 100% |

### PAGE 4

| | | | |
|---|---|---|---|
| L Tap1 Pan | -100 to 100% | R Tap1 Pan | -100 to 100% |
| L Tap2 Pan | -100 to 100% | R Tap2 Pan | -100 to 100% |
| L Tap3 Pan | -100 to 100% | R Tap3 Pan | -100 to 100% |

Wet/Dry — The relative amount of input signal and effected signal that is to appear in the final effect output mix. When set to 0%, the output is taken only from the input (dry). When set to 100%, the output is all wet.

Out Gain — The overall gain or amplitude at the output of the effect.

Feedback — The amplitude of the feedback tap(s) fed back to the beginning of the delay.

FB2/FB1>FB — Balance control between feedback line 1 and line 2. Setting this to 0% turns off feedback line 2, only allowing use of feedback line 1. A setting of 50% is an even mix of both lines, and 100% turns off line 1.

HF Damping — The amount of high frequency content of the signal to the input of the delay. This control determines the cutoff frequency of the one-pole (-6dB/octave) lowpass filters.

L Diff Dly, R Diff Dly — Adjusts delay length of the diffusers.

Diff Amt — Adjusts the diffuser intensity.

L Fdbk1 Dly — Adjusts the delay length of the left channel's feedback tap, fed back to the left channel's delay input.

L Fdbk2 Dly — Adjusts the delay length of the left channel's feedback tap, fed back to the right channel's delay input.

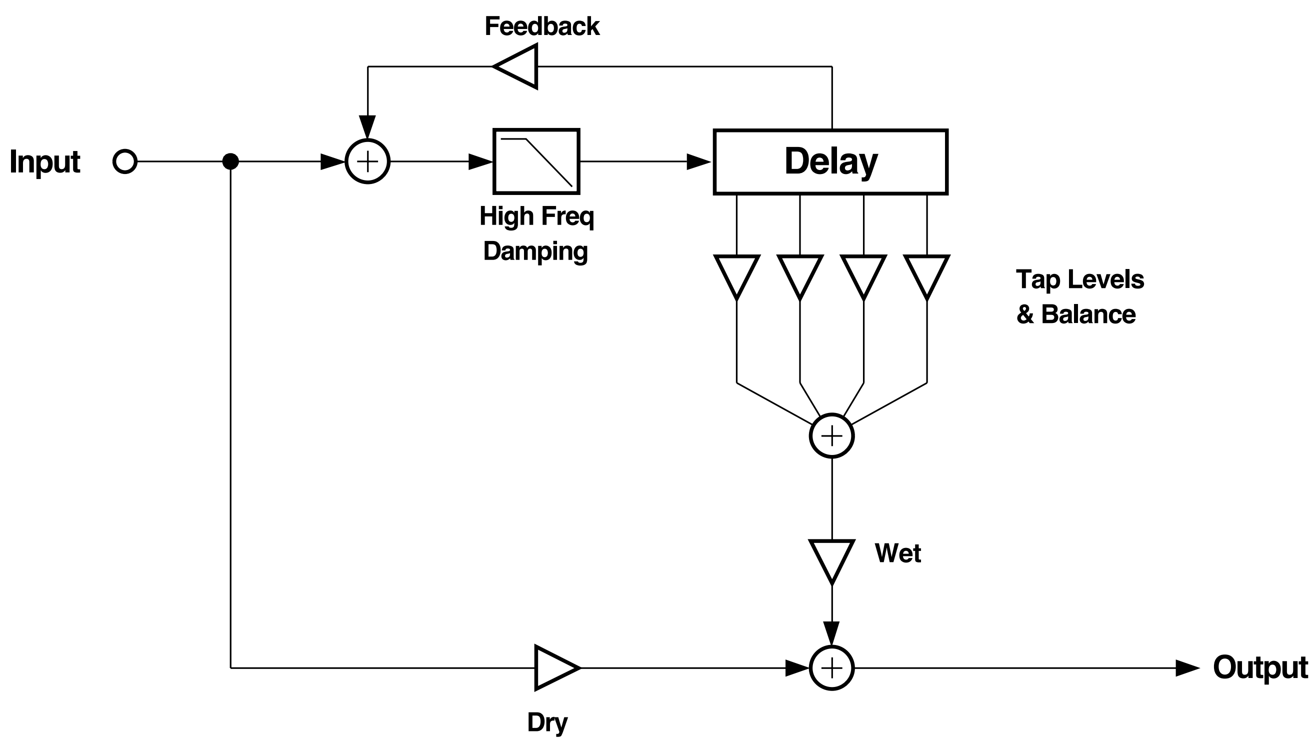
R Fdbk1 Dly — Adjusts the delay length of the right channel's feedback tap, fed back to the right channel's delay input.

| | |
|---|---|
| R Fdbk2 Dly | Adjusts the delay length of the right channel’s feedback tap, fed back to the left channel’s delay input. |
| L Tap*n* Dly, R Tap*n* Dly | Adjusts the delay length of the left and right channel’s three output taps. |
| L Tap*n* Lvl, L Tap*n* Lvl | Adjusts the listening level of the left and right channel’s three output taps. |
| L Tap*n* Pan, L Tap*n* Pan | Adjusts the equal power pan position of the left and right channel’s three output taps. 0% is center pan, negative values pan to left, and positive values pan to the right. |

# FXAlg #131: 4-Tap Delay • FXAlg #132: 4-Tap Delay BPM

**A stereo four-tap delay with feedback**
*Allocation Units:* 1

This is a simple stereo 4-tap delay algorithm with delay lengths defined either in milliseconds (ms) (#131), or in tempos and beats (#132). The left and right channels are fully symmetric (all controls affect both channels). The duration of each stereo delay tap (length of the delay) and the signal level from each stereo tap may be set. Prior to output each delay tap passes through a level and left/right balance control. The taps are summed and added to the dry input signal through a Wet/Dry control. The delayed signal from the "Loop" tap may be fed back to the delay input.




**Left Channel of 4-Tap Delay**

The delay length for any given tap is the sum of the coarse and fine parameters for the tap, multiplied by the DelayScale parameter which is common to all taps. The DelayScale parameter allows you to change the lengths of all the taps together.

A repetitive loop delay is created by turning up the Fdbk Level parameter. Only the Loop tap is fed back to the input of the delay, so this is the tap which controls the loop rate. Usually you will want the Loop delay length to be longer than the other tap lengths. Set the Loop delay length to the desired length, then set the other taps to fill in the measure with interesting rhythmical patterns. Setting tap levels allows some "beats" to receive more or less emphasis than others.

The delay lengths for 4-Tap Delay are in units of milliseconds (ms). If you want to base delay lengths on tempo, then the 4-Tap Delay BPM algorithm may be more convenient.

The feedback (Fdbk Level) controls how long a sound in the delay line takes to die out. At 100% feedback, the sound will be repeated indefinitely. HF Damping selectively removes high-frequency content from the delayed signal and will also cause the sound to eventually disappear.

The Hold parameter is a switch which controls signal routing. When turned on, Hold will play whatever signal is in the delay line indefinitely. Hold overrides the feedback parameter and prevents any incoming signal from entering the delay. You may have to practice using the Hold parameter. Each time the sound goes through the delay, it is reduced by the feedback amount. If feedback is fairly low and you turn on Hold at the wrong moment, you can get a disconcerting jump in level at some point in the loop. The Hold parameter has no effect on the Wet/Dry or HF Damping parameters, which continue to work as usual, so if there is some HF Damping, the delay will eventually die out.

## Parameters:

**PAGE 1**

| | | | |
|---|---|---|---|
| Wet/Dry | 0 to 100%wet | Out Gain | Off, -79.0 to 24.0 dB |
| Fdbk Level | 0 to 100% | | |
| | | Dry Bal | -100 to 100% |
| HF Damping | 16 Hz to 25088 Hz | Hold | On or Off |

**PAGE 2**

| | | | |
|---|---|---|---|
| Loop Crs | 0 to 2540 ms | DelayScale | 0.00x to 10.00x |
| Loop Fine | -20 to 20 ms | | |
| Tap1 Crs | 0 to 2540 ms | Tap3 Crs | 0 to 2540 ms |
| Tap1 Fine | -20 to 20 ms | Tap3 Fine | -20 to 20 ms |
| Tap2 Crs | 0 to 2540 ms | Tap4 Crs | 0 to 2540 ms |
| Tap2 Fine | -20 to 20 ms | Tap4 Fine | -20 to 20 ms |

**PAGE 3**

| | | | |
|---|---|---|---|
| Loop Level | 0 to 100% | Loop Bal | -100 to 100% |
| Tap2 Level | 0 to 100% | Tap2 Bal | -100 to 100% |
| Tap3 Level | 0 to 100% | Tap3 Bal | -100 to 100% |
| Tap4 Level | 0 to 100% | Tap4 Bal | -100 to 100% |

Wet/Dry — The relative amount of input signal and delay signal that are to appear in the final effect output mix. When set to 0%, the output is taken only from the input (dry). When set to 100%, the output is all wet.

Out Gain — The overall gain or amplitude at the output of the effect.

Fdbk Level — The percentage of the delayed signal to feed back or return to the delay input. Turning up the feedback will cause the effect to repeatedly echo or act as a crude reverb.

HF Damping — The -3 dB frequency in Hz of a one-pole lowpass filter (-6 dB/octave) placed in front of the delay line. The filter is specified for a signal passing through the filter once. Multiple passes through the feedback will cause the signal to become more and more dull.

| | |
|---|---|
| Dry Bal | The left-right balance of the dry signal. A setting of -100% allows only the left dry signal to pass to the left output, while a setting of 100% lets only the right dry signal pass to the right output. At 0%, equal amounts of the left and right dry signals pass to their respective outputs. |
| Hold | A switch which when turned on, locks any signal currently in the delay to play until Hold is turned off. When Hold is on, no signal can enter the delay and Feedback is set to 100% behind the scenes. Hold does not affect the HF Damping or Wet/Dry mix. A good candidate for a KDFX FXMod mapped to a momentary foot switch. |
| Tap*n* Level | The amount of signal from each of the taps (*n* = 1...4) which get sent to the output. With the Loop Lvl control, you can give different amounts of emphasis to the various taps in the loop. |
| Tap*n* Bal | The left-right balance of each of the stereo taps (*n* = 1...4). A setting of -100% allows only the left tap to pass to the left output, while a setting of 100% lets only the right tap pass to the right output. At 0%, equal amounts of the left and right taps pass to their respective outputs. |

**The following parameters are in #131 4-Tap Delay only:**

| | |
|---|---|
| Loop Crs | The coarse delay length of the Loop tap. If the feedback is turned up, this parameter sets the repeating delay loop length. The resolution of the coarse adjust is 20 milliseconds, but finer resolution can be obtained using the Loop Fine parameter. The maximum delay length is 2.55 seconds (2550ms). |
| Loop Fine | A fine adjustment to the Loop tap delay length. The delay resolution is 0.2 milliseconds (ms). Loop Fine is added to Loop Crs (coarse) to get the actual delay length. |
| Delay Scale | Multiplies all tap times by a common factor. |
| Tap*n* Crs | The coarse delay lengths of the output taps (*n* = 1...4). The resolution of the coarse adjust is 20 milliseconds, but finer resolution can be obtained using the Tap*n* Fine parameters. The maximum delay length is 2.55 seconds (2550ms). |
| Tap*n* Fine | A fine adjustment to the output tap delay lengths (*n* = 1...4). The delay resolution is 0.2 milliseconds (ms). Tap*n* Fine is added to Tap*n* Crs (coarse) to get actual delay lengths. |

# FXAlg #132: 4-Tap Delay BPM

In this Algorithm, the delay length for any given tap is determined by the tempo, expressed in beats per minute (BPM), and multiples or fractions of beats (bts). The tempo alters all tap lengths together.

The “bts” parameter is adjustable in increments of 1/24th of a beat, which is a useful fraction because it can divide beats into 2, 3, 4, 6, 8, or 12 parts. The length of a delay in seconds can be calculated as T = (beats/tempo) * 60.

**IMPORTANT NOTE**: KDFX has a limited amount of delay memory available (over 2.5 seconds for 4-Tap BPM). When slow tempos and/or long lengths are specified, you may run out of delay memory, at which point the delay length will be cut in half. When you slow down the tempo, you may find the delays suddenly getting shorter.

A repetitive loop delay is created by turning up the feedback parameter (Fdbk Level). Only the Loop tap is fed back to the input of the delay, so this is the tap which controls the loop rate. Usually you will want the Loop tap (LoopLength parameter) to be longer than the other tap lengths. To repeat a pattern on a 4/4 measure (4 beats per measure) simply set LoopLength to 4 bts. The output taps can then be used to fill in the measure with interesting rhythmical patterns. Setting tap levels allows some “beats” to receive more or less emphasis than others.

## Parameters:

**PAGE 1**

| | | | |
|---|---|---|---|
| Wet/Dry | 0 to 100%wet | Out Gain | Off, -79.0 to 24.0 dB |
| Fdbk Level | 0 to 100% | Tempo | System, 1 to 255 BPM |
| | | Dry Bal | -100 to 100% |
| HF Damping | 16 Hz to 25088 Hz | Hold | On or Off |

**PAGE 2**

| | | | |
|---|---|---|---|
| LoopLength | 0 to 32 bts | | |
| Tap1 Delay | 0 to 32 bts | | |
| Tap2 Delay | 0 to 32 bts | | |
| Tap3 Delay | 0 to 32 bts | | |
| Tap4 Delay | 0 to 32 bts | | |

**PAGE 3**

| | | | |
|---|---|---|---|
| Tap1 Level | 0 to 100% | Tap1 Bal | -100 to 100% |
| Tap2 Level | 0 to 100% | Tap2 Bal | -100 to 100% |
| Tap3 Level | 0 to 100% | Tap3 Bal | -100 to 100% |
| Tap4 Level | 0 to 100% | Tap4 Bal | -100 to 100% |

Tempo — Basis for the delay lengths, as referenced to a musical tempo in bpm (beats per minute). When this parameter is set to “System”, the tempo is locked to the internal sequencer tempo or to incoming MIDI clocks. When it is set to “System”, sources (FUNs, LFOs, ASRs etc.) will have no effect on the Tempo parameter.

LoopLength — The delay length of the Loop tap. If the feedback is turned up, this parameter sets the repeating delay loop length. The loop length is specified as a fraction or multiple of the tempo, in “beats”. The length of a delay loop in seconds can be calculated as T = (beats/tempo) * 60.

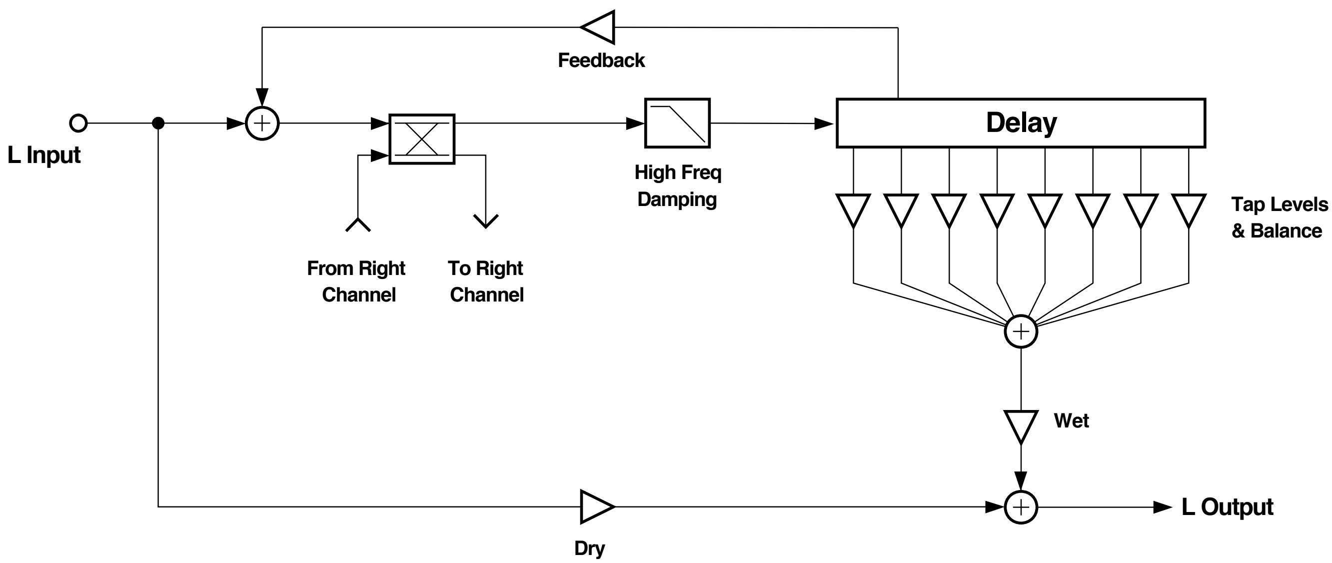
Tap*n* Delay — The delay lengths of the taps (*n* = 1...4). as tempo beat durations. The delay length is specified as a fraction or multiple of the tempo, in “beats”. The length of a delay in seconds can be calculated as T = (beats/tempo) * 60. Use the output taps to create interesting rhythmic patterns within the repeating loop.

# FXAlg #133: 8-Tap Delay • FXAlg #134: 8-Tap Delay BPM

### A stereo eight-tap delay with cross-coupled feedback

*Allocation Units:* 2

This is a simple stereo 8-tap delay algorithm with delay lengths defined in milliseconds (ms) (#133), or in tempos and beats (#134). The left and right channels are fully symmetric (all controls affect both channels). The duration of each stereo delay tap (length of the delay) and the signal level from each stereo tap may be set. Prior to output each delay tap passes through a level and left/right balance control. Pairs of stereo taps are tied together with balance controls acting with opposite left/right sense. The taps are summed and added to the dry input signal through a Wet/Dry control. The delayed signal from the "Loop" tap may be fed back to the delay input. The sum of the input signal and the feedback signal may be mixed or swapped with the input/feedback signal from the other channel (cross-coupling). When used with feedback, cross-coupling can achieve a ping-pong effect between the left and right channels.




**Left Channel of 8-Tap Delay**

The delay length for any given tap is the sum of the coarse and fine parameters for the tap multiplied by the DelayScale parameter which is common to all taps. The DelayScale parameter allows you to change the lengths of all the taps together.

A repetitive loop delay is created by turning up the Fdbk Level parameter. Only the Loop tap is fed back to the input of the delay, so this is the tap which controls the loop rate. Usually you will want the Loop delay length to be longer than the other tap lengths. Set the Loop delay length to the desired length, then set the other taps to fill in the measure with interesting rhythmical patterns. Setting tap levels allows some "beats" to receive more or less emphasis than others.

The delay lengths for 8-Tap Delay are in milliseconds (ms). If you want to base delay lengths on tempo, then the 8-Tap Delay BPM algorithm may be more convenient.

The feedback (Fdbk Level) controls how long a sound in the delay line takes to die out. At 100% feedback, the sound will be repeated indefinitely. HF Damping selectively removes high-frequency content from the delayed signal and will also cause the sound to eventually disappear.

The Hold parameter is a switch which controls signal routing. When turned on, Hold will play whatever signal is in the delay line indefinitely. Hold overrides the feedback parameter and prevents any incoming signal from entering the delay. You may have to practice using the Hold parameter. Each time the sound goes through the delay, it is reduced by the feedback amount. If feedback is fairly low and you turn on Hold at the wrong moment, you can get a disconcerting jump in level at some point in the loop. The Hold parameter has no effect on the Wet/Dry or HF Damping parameters, which continue to work as usual, so if there is some HF Damping, the delay will eventually die out. It is an excellent candidate for a KDFX Modulation routing using a momentary foot switch as a source.

## Parameters:

### PAGE 1

| | | | |
|---|---|---|---|
| Wet/Dry | 0 to 100%wet | Out Gain | Off, -79.0 to 24.0 dB |
| Fdbk Level | 0 to 100% | | |
| Xcouple | 0 to 100% | Dry Bal | -100 to 100% |
| HF Damping | 16 Hz to 25088 Hz | Hold | On or Off |

### PAGE 2

| | | | |
|---|---|---|---|
| Loop Crs | 0 to 5100 ms | DelayScale | 0.00x to 10.00x |
| Loop Fine | -20 to 20 ms | | |
| Tap1 Crs | 0 to 5100 ms | Tap3 Crs | 0 to 5100 ms |
| Tap1 Fine | -20 to 20 ms | Tap3 Fine | -20 to 20 ms |
| Tap2 Crs | 0 to 5100 ms | Tap4 Crs | 0 to 5100 ms |
| Tap2 Fine | -20 to 20 ms | Tap4 Fine | -20 to 20 ms |

### PAGE 3

| | | | |
|---|---|---|---|
| Tap5 Crs | 0 to 5100 ms | Tap7 Crs | 0 to 5100 ms |
| Tap5 Fine | -20 to 20 ms | Tap7 Fine | -20 to 20 ms |
| Tap6 Crs | 0 to 5100 ms | Tap8 Crs | 0 to 5100 ms |
| Tap6 Fine | -20 to 20 ms | Tap8 Fine | -20 to 20 ms |

### PAGE 4

| | | | |
|---|---|---|---|
| Tap1 Level | 0 to 100% | Tap5 Level | 0 to 100% |
| Tap2 Level | 0 to 100% | Tap6 Level | 0 to 100% |
| Tap3 Level | 0 to 100% | Tap7 Level | 0 to 100% |
| Tap4 Level | 0 to 100% | Tap8 Level | 0 to 100% |
| Tap1/-5Bal | -100 to 100% | Tap3/-7Bal | -100 to 100% |
| Tap2/-6Bal | -100 to 100% | Tap4/-8Bal | -100 to 100% |

Wet/Dry — The relative amount of input signal and delay signal that is to appear in the final effect output mix. When set to 0%, the output is taken only from the input (dry). When set to 100%, the output is all wet.

Out Gain — The overall gain or amplitude at the output of the effect.

Fdbk Level — The percentage of the delayed signal to feed back or return to the delay input. Turning up the feedback will cause the effect to repeatedly echo or act as a crude reverb.

Xcouple — 8-Tap Delay is a stereo effect. The cross-coupling control lets you send the feedback from a channel to its own input (0% cross-coupling) or to the other channel's input (100% cross-coupling) or somewhere in between. This control has no effect if the Fdbk Level control is set to 0%.

HF Damping — The -3 dB frequency in Hz of a one-pole lowpass filter (-6 dB/octave) placed in front of the delay line. The filter is specified for a signal passing through the filter once. Multiple passes through the feedback will cause the signal to become more and more dull.

Dry Bal — The left/right balance of the dry signal. A setting of -100% allows only the left dry signal to pass to the left output, while a setting of 100% lets only the right dry signal pass to the right output. At 0%, equal amounts of the left and right dry signals pass to their respective outputs.

Hold — A switch which when turned on, locks any signal currently in the delay to play until Hold is turned off. When Hold is on, no signal can enter the delay and Feedback is set to 100% behind the scenes. Hold does not affect the HF Damping or Wet/Dry mix. It is an excellent candidate for a KDFX Modulation routing using a momentary foot switch as a source.

Tap*n* Level — The amount of signal from each of the taps (*n* = 1...8) which gets sent to the output.

Tap*m*/-*n* Bal — The left/right balance of each of the stereo taps. The balances are controlled in pairs of taps: 1 & 5, 2 & 6, 3 & 7, and 4 & 8. The balance controls work in opposite directions for the two taps in the pair. When the balance is set to -100%, the first tap of the pair is fully right while the second is fully left. At 0%, equal amounts of the left and right taps pass to their respective outputs.

**The following parameters are in #133 8-Tap Delay only:**

Loop Crs — The coarse delay length of the Loop tap. If the feedback is turned up, this parameter sets the repeating delay loop length. The resolution of the coarse adjust is 20 milliseconds, but finer resolution can be obtained using the Loop Fine parameter. The maximum delay length is 5.10 seconds (5100ms.

Loop Fine — A fine adjustment to the Loop tap delay length. The delay resolution is 0.2 milliseconds (ms). Loop Fine is added to Loop Crs (coarse) to get the actual delay length.

Delay Scale — Multiplies all tap times by a common factor.

Tap*n* Crs — The coarse delay lengths of the output taps (*n* = 1...8). The resolution of the coarse adjust is 20 milliseconds, but finer resolution can be obtained using the Tap*n* Fine parameters. The maximum delay length is 5.1 seconds (5100ms).

Tap*n* Fine — A fine adjustment to the output tap delay lengths (*n* = 1...8). The delay resolution is 0.2 milliseconds (ms). Tap*n* Fine is added to Tap*n* Crs (coarse) to get actual delay lengths.

# FXAlg #134: 8-Tap Delay BPM

In this Algorithm, the delay length for any given tap is determined by the tempo, expressed in beats per minute (BPM), and multiples or fractions of beats (bts). The tempo alters all tap lengths together.

The “bts” parameter is adjustable in increments of 1/24th of a beat, which is a useful fraction because it can divide beats into 2, 3, 4, 6, 8, or 12 parts. The length of a delay in seconds can be calculated as T = (beats/tempo) * 60.

**IMPORTANT NOTE**: KDFX has a limited amount of delay memory available (over 5 seconds for 8-Tap BPM). When slow tempos and/or long lengths are specified, you may run out of delay memory, at which point the delay length will be cut in half. When you slow down the tempo, you may find the delays suddenly getting shorter.

A repetitive loop delay is created by turning up the feedback parameter (Fdbk Level). Only the Loop tap is fed back to the input of the delay, so this is the tap which controls the loop rate. Usually you will want the Loop tap (LoopLength parameter) to be longer than the other tap lengths. To repeat a pattern on a 4/4 measure (4 beats per measure) simply set LoopLength to 4 bts. The output taps can then be used to fill in the measure with interesting rhythmical patterns. Setting tap levels allows some “beats” to receive more or less emphasis than others.

## Parameters:

**PAGE 1**

| | | | |
|---|---|---|---|
| Wet/Dry | 0 to 100%wet | Out Gain | Off, -79.0 to 24.0 dB |
| Fdbk Level | 0 to 100% | Tempo | System, 1 to 255 BPM |
| Xcouple | 0 to 100% | Dry Bal | -100 to 100% |
| HF Damping | 16 Hz to 25088 Hz | Hold | On or Off |

**PAGE 2**

| | | | |
|---|---|---|---|
| LoopLength | 0 to 32 bts | | |
| Tap1 Delay | 0 to 32 bts | Tap5 Delay | 0 to 32 bts |
| Tap2 Delay | 0 to 32 bts | Tap6 Delay | 0 to 32 bts |
| Tap3 Delay | 0 to 32 bts | Tap7 Delay | 0 to 32 bts |
| Tap4 Delay | 0 to 32 bts | Tap8 Delay | 0 to 32 bts |

**PAGE 3**

| | | | |
|---|---|---|---|
| Tap1 Level | 0 to 100% | Tap5 Level | 0 to 100% |
| Tap2 Level | 0 to 100% | Tap6 Level | 0 to 100% |
| Tap3 Level | 0 to 100% | Tap7 Level | 0 to 100% |
| Tap4 Level | 0 to 100% | Tap8 Level | 0 to 100% |

**PAGE 4**

| | | | |
|---|---|---|---|
| Tap1 Bal | -100 to 100% | Tap5 Bal | -100 to 100% |
| Tap2 Bal | -100 to 100% | Tap6 Bal | -100 to 100% |
| Tap3 Bal | -100 to 100% | Tap7 Bal | -100 to 100% |
| Tap4 Bal | -100 to 100% | Tap8 Bal | -100 to 100% |

Tempo — Basis for the delay lengths, as referenced to a musical tempo in bpm (beats per minute). When this parameter is set to “System”, the tempo is locked to the internal sequencer tempo or to incoming MIDI clocks. When it is set to “System”, sources (FUNs, LFOs, ASRs etc.) will have no effect on the Tempo parameter.

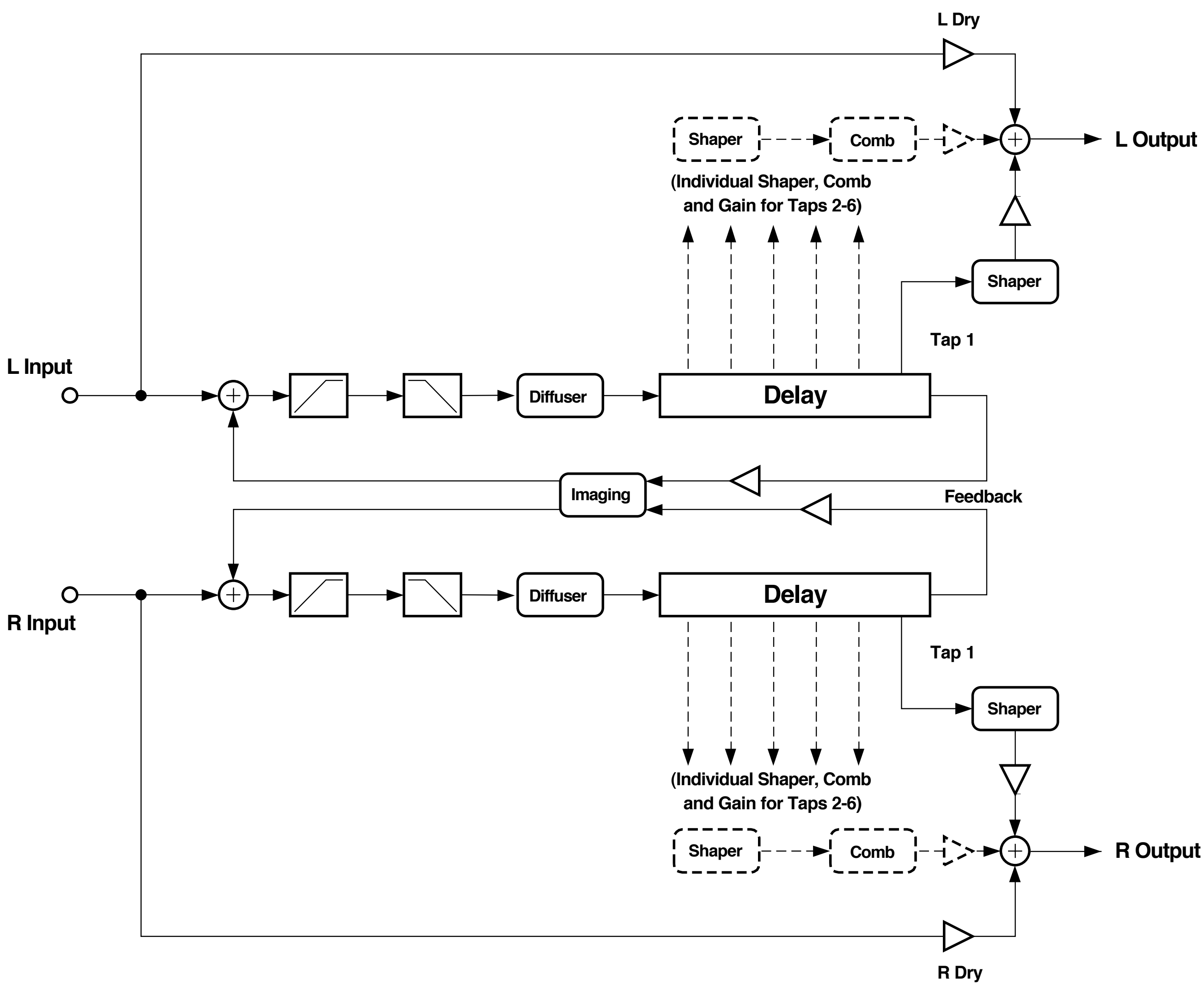
LoopLength — The delay length of the Loop tap. If the feedback is turned up, this parameter sets the repeating delay loop length. The loop length is specified as a fraction or multiple of the tempo, in “beats”. The length of a delay loop in seconds can be calculated as T = (beats/tempo) * 60.

Tap*n* Delay — The delay lengths of the taps (*n* = 1...4). as tempo beat durations. The delay length is specified as a fraction or multiple of the tempo, in “beats”. The length of a delay in seconds can be calculated as T = (beats/tempo) * 60. Use the output taps to create interesting rhythmic patterns within the repeating loop.

# FXAlg #135: Spectral 4-Tap • FXAlg #136: Spectral 6-Tap

**Tempo based 4- and 6-tap delays with**
**added shapers and resonant comb filters on each tap**
*Allocation Units:* 2 for Spectral 4-Tap; 3 for Spectral 6-Tap

Spectral 4 Tap and Spectral 6 Tap are respectively 2- and 3-PAU tempo-based multi-tap delay effects. They are similar to simple 4- and 6-tap delays with feedback, but have their feedback and output taps modified with shapers and filters. In the feedback path of each channel are a diffuser, hipass filter, lowpass filter, and imager. Each delay tap has a shaper, comb filter, balance and level controls with the exception of Tap 1, which does not have a comb filter.



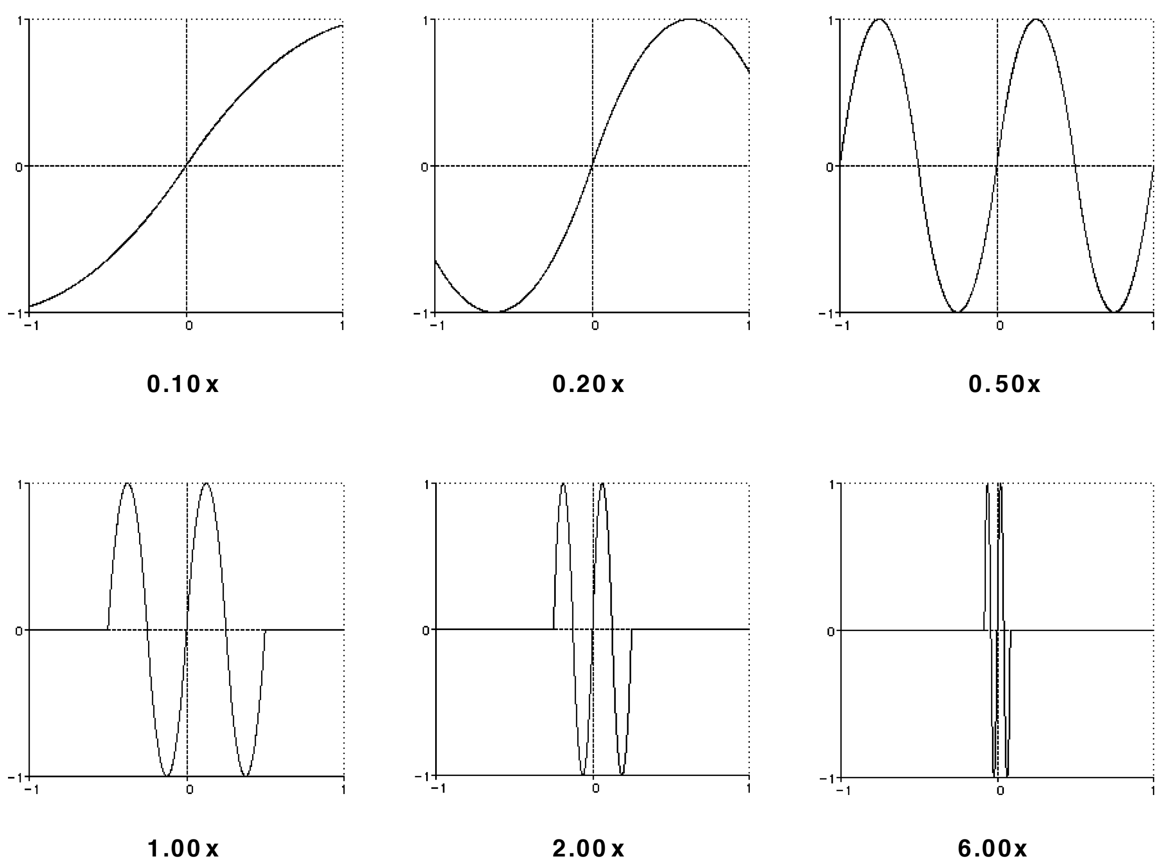

Spectral 6-Tap

Diffusers add a quality that can be described as “smearing” the feedback signal. The more a signal has been regenerated through feedback and consequently fed through the diffuser, the more it is smeared. It requires two parameters, one for the duration a signal is smeared, labeled Diff Delay, and the other for the amount it is smeared, labeled Diff Amt. Positive diffusion settings will add diffusion while maintaining image integrity. Negative diffusion amounts will cause the feedback image to lose image integrity and become wide. Short Diff Delay settings have subtle smearing effects. Increasing Diff Delay will be more noticeable, and long delay settings will take on a ringy resonant quality. To disable the diffuser, both Diff Delay and Diff Amt should be set to zero.

Two 1-pole 6dB/oct filters—hipass and lowpass—are also in the feedback path. The hipass filter roll-off frequency is controlled with LF Damping, and the lowpass filter roll-off frequency is controlled by HF Damping.

The imager (found on PARAM2) shifts the stereo input image when fed through feedback. Small positive or negative values shift the image to the right or left respectively. Larger values shift the image so much that the image gets scrambled through each feedback generation.

On each output tap is a shaper. For an overview of shaper functionality, refer to the section on shapers in the *K2500 Performance Guide*. The Spectral Multi-Tap shapers offer 4 shaping loops as opposed to 8 found in the VAST shapers, but can allow up to 6.00x intensity (Figure 2). Immediately following the shapers on taps 2 and above are resonant comb filters tuned in semitones. These comb filters make the taps become pitched. When a comb filter is in use, the shaper before it can be used to intensify these pitched qualities.




**Various shaper curves used in the Spectral Multi-Taps**

Each tap also has separate balance and level controls.

Since these are tempo based effects, tap delay values and feedback delay (labeled LoopLength on PARAM2) values are set relative to a beat. The beat duration is set be adjusting Tempo in BPM. The tempo can be synced to the system clock by setting Tempo to System.

The "bts" parameter is adjustable in increments of 1/24th of a beat, which is a useful fraction because it can divide beats into 2, 3, 4, 6, 8, or 12 parts. The length of a delay in seconds can be calculated as T = (beats/tempo) * 60.

## Parameters (Spectral 4-Tap):

### PAGE 1

| | | | |
|---|---|---|---|
| Wet/Dry | 0 to 100% | Out Gain | Off, -79.0 to 24.0 dB |
| Fdbk Level | 0 to 100% | Tempo | System, 0 to 255 BPM |
| HF Damping | 16 to 25088 Hz | Diff Delay | 0 to 20.0 ms |
| LF Damping | 16 to 25088 Hz | Diff Amt | -100 to 100% |

### PAGE 2

| | | | |
|---|---|---|---|
| LoopLength | On or Off | Tap2 Delay | 0 to 32 bts |
| Fdbk Image | -100 to 100% | Tap2 Shapr | 0.10 to 6.00 x |
| Tap1 Delay | 0 to 32 bts | Tap2 Pitch | C-1 to C8 |
| Tap1 Shapr | 0.10 to 6.00 x | Tap2 PtAmt | 0 to 100% |
| Tap1 Level | 0 to 100% | Tap2 Level | 0 to 100% |
| Tap1 Bal | -100 to 100% | Tap2 Bal | -100 to 100% |

### PAGE 3

| | | | |
|---|---|---|---|
| Tap3 Delay | 0 to 32 bts | Tap4 Delay | 0 to 32 bts |
| Tap3 Shapr | 0.10 to 6.00 x | Tap4 Shapr | 0.10 to 6.00 x |
| Tap3 Pitch | C-1 to C8 | Tap4 Pitch | C-1 to C8 |
| Tap3 PtAmt | 0 to 100% | Tap4 PtAmt | 0 to 100% |
| Tap3 Level | 0 to 100% | Tap4 Level | 0 to 100% |
| Tap3 Bal | -100 to 100% | Tap4 Bal | -100 to 100% |

## Parameters (Spectral 6-Tap):

### PAGE 1

| | | | |
|---|---|---|---|
| Wet/Dry | 0 to 100% | Out Gain | Off, -79.0 to 24.0 dB |
| Fdbk Level | 0 to 100% | Tempo | System, 0 to 255 BPM |
| HF Damping | 16 to 25088 Hz | Diff Delay | 0 to 20.0 ms |
| LF Damping | 16 to 25088 Hz | Diff Amt | -100 to 100% |

**PAGE 2**

| | | | |
|---|---|---|---|
| LoopLength | On or Off | Tap2 Delay | 0 to 32 bts |
| Fdbk Image | -100 to 100% | Tap2 Shapr | 0.10 to 6.00 x |
| Tap1 Delay | 0 to 32 bts | Tap2 Pitch | C-1 to C8 |
| Tap1 Shapr | 0.10 to 6.00 x | Tap2 PtAmt | 0 to 100% |
| Tap1 Level | 0 to 100% | Tap2 Level | 0 to 100% |
| Tap1 Bal | -100 to 100% | Tap2 Bal | -100 to 100% |

**PAGE 3**

| | | | |
|---|---|---|---|
| Tap3 Delay | 0 to 32 bts | Tap4 Delay | 0 to 32 bts |
| Tap3 Shapr | 0.10 to 6.00 x | Tap4 Shapr | 0.10 to 6.00 x |
| Tap3 Pitch | C-1 to C8 | Tap4 Pitch | C-1 to C8 |
| Tap3 PtAmt | 0 to 100% | Tap4 PtAmt | 0 to 100% |
| Tap3 Level | 0 to 100% | Tap4 Level | 0 to 100% |
| Tap3 Bal | -100 to 100% | Tap4 Bal | -100 to 100% |

**PAGE 4**

| | | | |
|---|---|---|---|
| Tap5 Delay | 0 to 32 bts | Tap6 Delay | 0 to 32 bts |
| Tap5 Shapr | 0.10 to 6.00 x | Tap6 Shapr | 0.10 to 6.00 x |
| Tap5 Pitch | C-1 to C8 | Tap6 Pitch | C-1 to C8 |
| Tap5 PtAmt | 0 to 100% | Tap6 PtAmt | 0 to 100% |
| Tap5 Level | 0 to 100% | Tap6 Level | 0 to 100% |
| Tap5 Bal | -100 to 100% | Tap6 Bal | -100 to 100% |

Wet/Dry — The relative amount of input signal and effected signal that is to appear in the final effect output mix. When set to 0%, the output is taken only from the input (dry). When set to 100%, the output is all wet. Negative values polarity-invert the wet signal.

Out Gain — The overall gain or amplitude at the output of the effect.

Fdbk Level — The amount that the feedback tap is fed to the input of the delay.

HF Damping — The amount of high-frequency content of the signal to the input of the delay. This control determines the cutoff frequency of the one-pole (-6dB/octave) lowpass filters.

LF Damping — The amount of low-frequency content of the signal to the input of the delay. This control determines the cutoff frequency of the one-pole (-6dB/octave) hipass filters.

Tempo — Basis for the rates of the delay times, as referenced to a musical tempo in BPM (beats per minute). When this parameter is set to “System”, the tempo is locked to the internal sequencer tempo or to incoming MIDI clocks. When it is set to “System”, sources (FUNs, LFOs, ASRs etc.) will have no effect on the Tempo parameter.

Diff Dly — The length that the diffuser smears the signal sent to the input of the delay.

Diff Amt — The intensity that the diffuser smears the signal sent to the input of the delay. Negative values decorrelate the stereo signal.

LoopLength — The delay length of the feedback tap in 24ths of a beat. The length of a delay in seconds can be calculated as T = (beats/tempo) * 60.

| | |
|---|---|
| Fdbk Image | Sets the amount the stereo image is shifted each time it passes through the feedback line. |
| Tap *n* Delay | Adjusts the length of time, in 1/24ths of a beat, each output tap is delayed. |
| Tap *n* Shapr | Adjusts the intensity of the shaper at each output tap. |
| Tap *n* Pitch | Adjusts the frequency in semitones of the comb filter at each output tap. |
| Tap *n* PtAmt | Adjusts the intensity of the comb filter at each output tap. |
| Tap *n* Level | Adjusts the relative amplitude of each output tap. |
| Tap *n* Bal | Adjusts the left/right balance of each output tap. Negative values bring down the right channel, and positive values bring down the left channel. |

# FXAlgs #150–153: Choruses

## FXAlg #150: Chorus 1
## FXAlg #151: Chorus 2
## FXAlg #152: Dual Chorus 1
## FXAlg #153: Dual Chorus 2

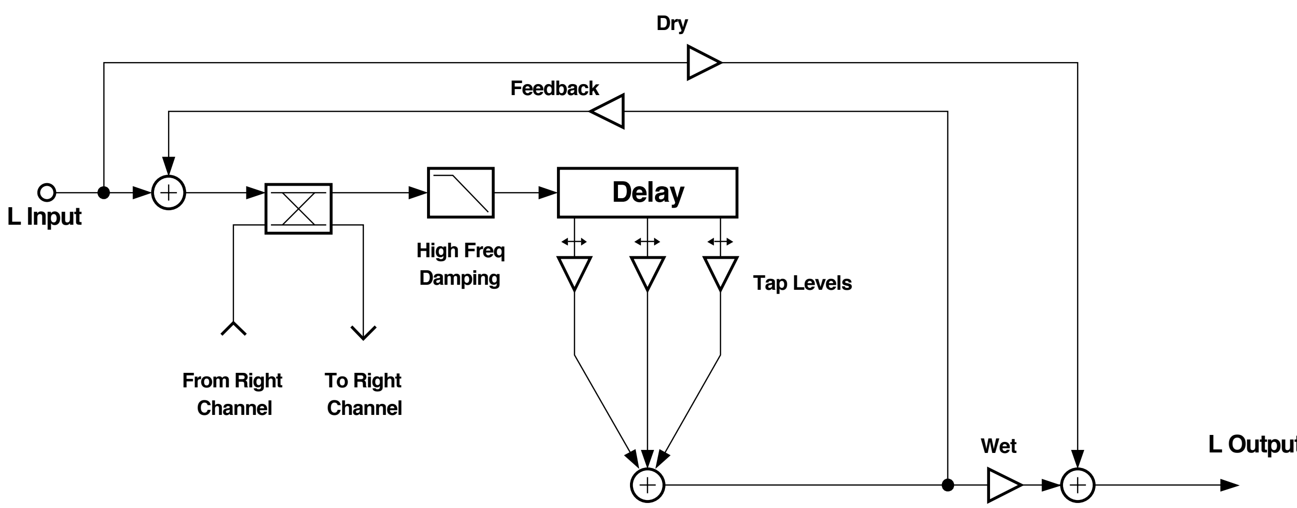
**One- and three-tap stereo and dual-mono choruses**

*Allocation Units:* 1 for Chorus 1 and Dual Chorus 1; 2 for Chorus 2 and Dual Chorus 2

Chorus is an effect which gives the illusion of multiple voices playing in unison. The effect is achieved by detuning copies of the original signal and summing the detuned copies back with the original. Low-frequency oscillators (LFOs) are used modulate the positions of output taps from a delay line. The delay line tap modulation causes the pitch of the signal to shift up and down, producing the required detuning.

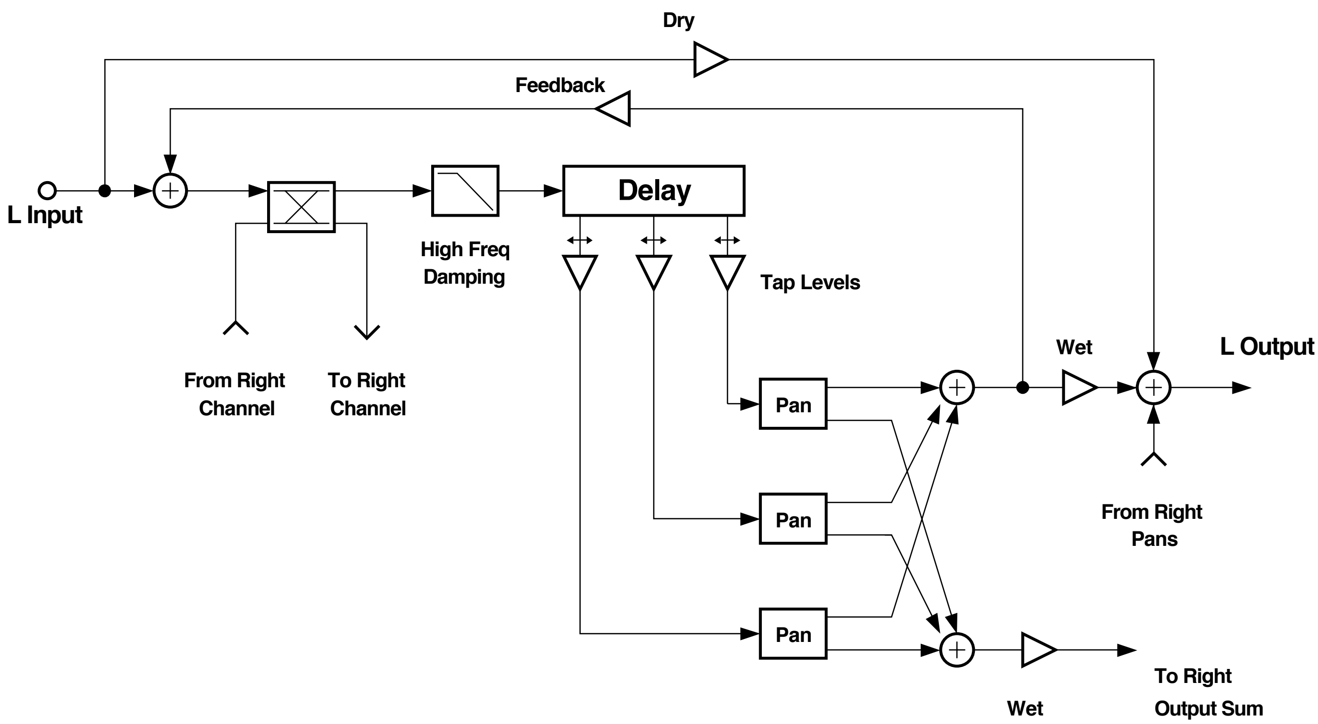
The choruses are available as stereo or dual mono. The stereo choruses have the parameters for the left and right channels ganged.

Chorus 2 is a 2-PAU multi-tapped delay (3 taps) based chorus effect with cross-coupling and individual output tap panning. Figure 1 is a simplified block diagram of the left channel of Chorus 2.



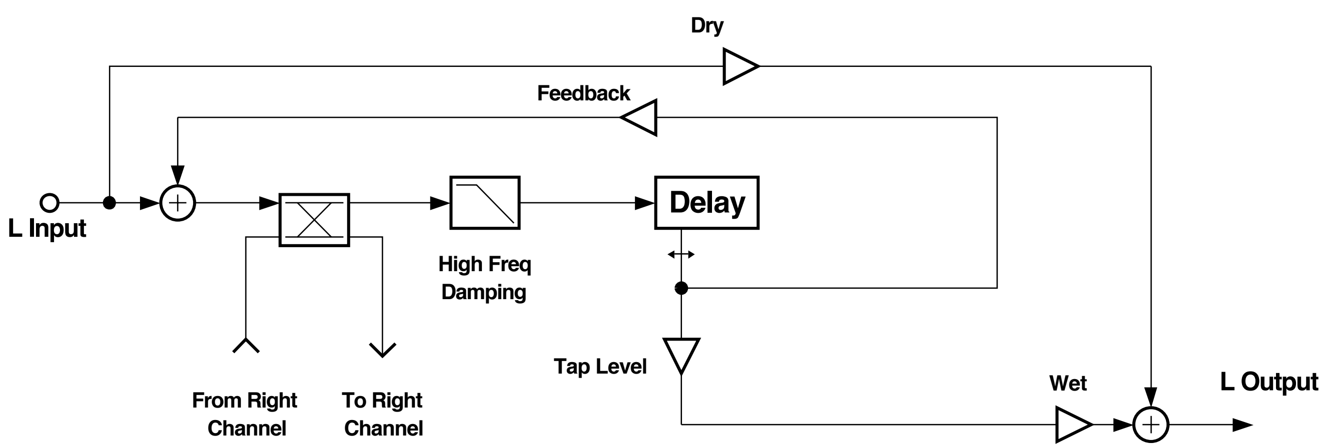

**Block diagram of left channel of Chorus 2. Right channel is the same.**

The dual mono choruses are like the stereo choruses but have separate left and right controls. Dual mono choruses also allow you to pan the delay taps between left or right outputs.

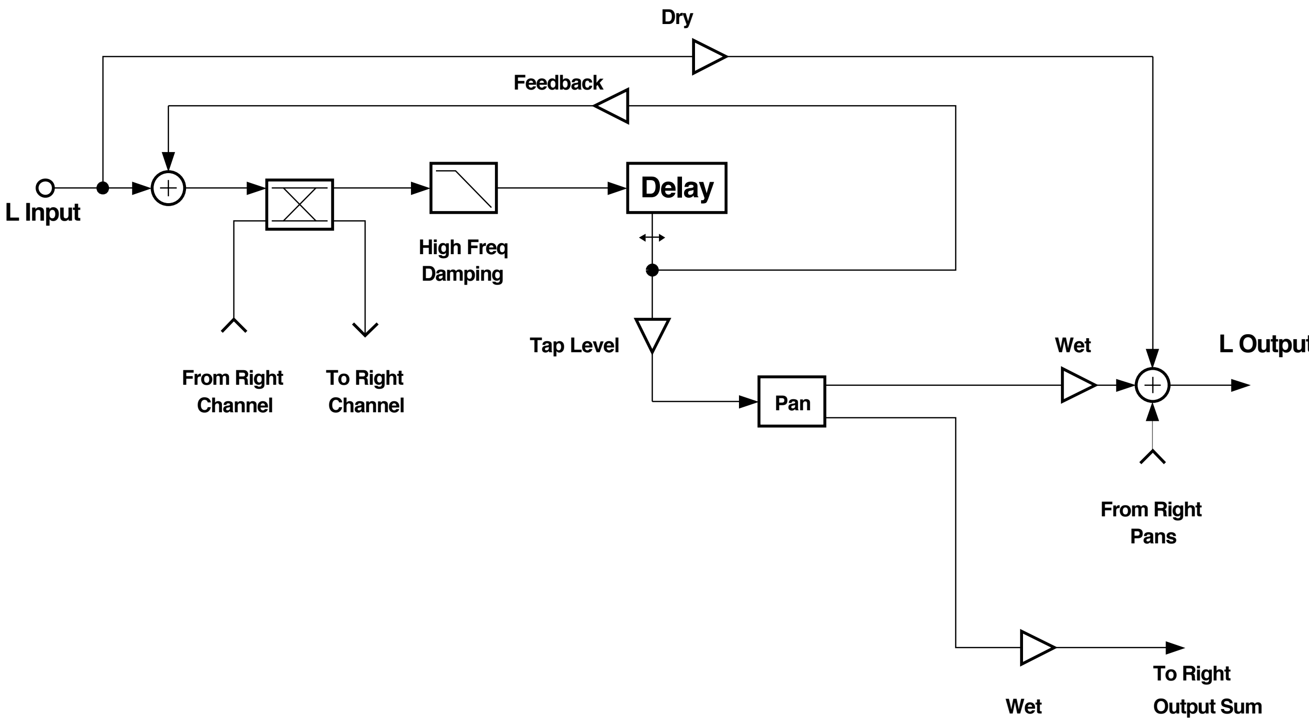



**Block diagram of left channel of Dual Chorus 2. Right channel is similar.**

Chorus 1 uses just 1 PAU and has a single delay tap.




**Block diagram of left channel of Chorus 1. Right channel is the same.**



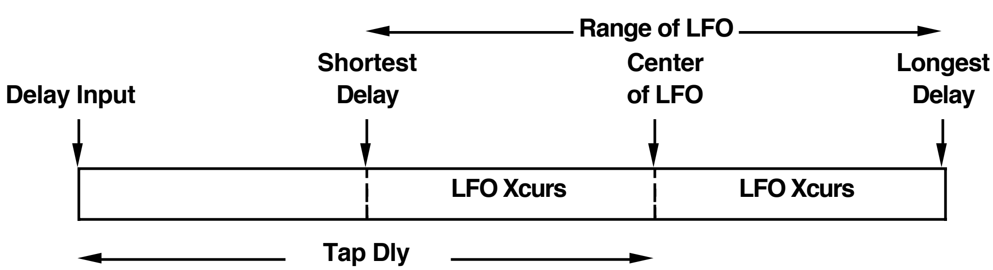

**Block diagram of left channel of Dual Chorus 1. Right channel is similar.**

The left and right channels pass through their own chorus blocks. There may be cross-coupling between the channels. For Chorus 2 and Dual Chorus 2, each channel has three moving taps which are summed, while Chorus 1 and Dual Chorus 1 have one moving tap for both channels. In the dual mono choruses you can pan the taps to left or right. The summed taps (or the single tap of Chorus 1) is used for the wet output signal. The summed tap outputs, weighted by their level controls, are used for feedback to the delay line input. The input and feedback signals go through a one-pole lowpass filter (HF Damping) before going entering the delay line.

The Wet/Dry control is an equal power cross-fade. The Output Gain parameters affect both wet and dry signals.

For each of the LFO tapped delay lines, you may set the tap levels, the left/right pan position, delays of the modulating delay lines, the rates of the LFO cycles, and the maximum depths of the pitch detuning. The LFOs detune the pitch of signal copies above <u>and</u> below the original pitch. The depth units are in cents, or 1/100ths of a semitone.

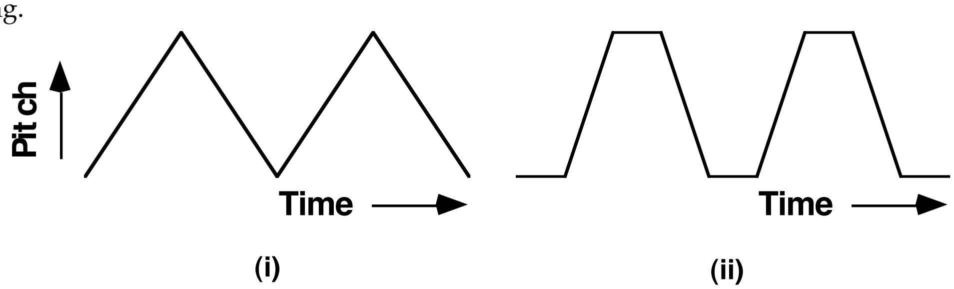
In the stereo Chorus 1 and Chorus 2, the relative phases of the LFOs modulating the left and right channels may be adjusted.




**Delay for a Single LFO**

The settings of the LFO rates and the LFO depths determine how far the LFOs will sweep across their delay lines from the shortest delays to the longest delays (the LFO excursions). The Tap Delays specify the average amount of delay of the LFO-modulated delay lines—in other words, the timing of the center of the LFO excursion. The center of LFO excursion can not move smoothly, and changing that parameter creates discontinuities in the tapped signal, which is heard as zipper noise. It is therefore a good idea to adjust the Tap Dly parameter to a reasonable setting (one which gives enough delay for the maximum LFO excursion), then leave it. If you increase the LFO modulation depth or reduce the LFO rate to a point where the LFO excursion exceeds the specified Tap Dly, the center of LFO excursion will be moved up, and again cause signal discontinuities. However, if enough Tap Dly is specified, Depth and Rate will be modulated smoothly.

As the LFOs sweep across the delay lines, the signal will change pitch. The pitch will change with a triangular envelope (rise-fall-rise-fall) or with a trapzoidal envelope (rise-hold-fall-hold). You can choose the pitch envelope with the Pitch Env parameter. Unfortunately rate and depth cannot be smoothly modulated when set to the “trapzoid” setting.




**Pitch Envelope—(i) “Triangle''; (ii) “trapzoid”**

## Parameters (Chorus1):

**PAGE 1**

| | | | |
|---|---|---|---|
| Wet/Dry | -100 to 100%wet | Out Gain | Off, -79.0 to 24.0 dB |
| Fdbk Level | -100 to 100% | | |
| Xcouple | 0 to 100% | | |
| HF Damping | 16 Hz to 25088 Hz | Pitch Env | Triangle or trapzoid |

**PAGE 2**

| | | | |
|---|---|---|---|
| Tap Lvl | -100 to 100% | LFO Rate | 0.01 to 10.00 Hz |
| Tap Dly | 0.0 to 1000.0 ms | LFO Depth | 0.0 to 50.0 ct |
| | | L/R Phase | 0.0 to 360.0 deg |

## Parameters (Chorus 2):

**PAGE 1**

| | | | |
|---|---|---|---|
| Wet/Dry | -100 to 100%wet | Out Gain | Off, -79.0 to 24.0 dB |
| Fdbk Level | -100 to 100% | | |
| Xcouple | 0 to 100% | | |
| HF Damping | 16 Hz to 25088 Hz | Pitch Env | Triangle or trapzoid |

PAGE 2

| | | | |
|---|---|---|---|
| Tap1 Lvl | -100 to 100% | Tap1 Dly | 4.0 to 1000.0 ms |
| Tap2 Lvl | -100 to 100% | Tap2 Dly | 4.0 to 1000.0 ms |
| Tap3 Lvl | -100 to 100% | Tap3 Dly | 4.0 to 1000.0 ms |

PAGE 3

| | | | |
|---|---|---|---|
| LFO1 Rate | 0.01 to 10.00 Hz | LFO1 LRPhs | 0.0 to 360.0 deg |
| LFO2 Rate | 0.01 to 10.00 Hz | LFO2 LRPhs | 0.0 to 360.0 deg |
| LFO3 Rate | 0.01 to 10.00 Hz | LFO3 LRPhs | 0.0 to 360.0 deg |
| LFO1 Dpth | 0.0 to 50.0 ct | | |
| LFO2 Dpth | 0.0 to 50.0 ct | | |
| LFO3 Dpth | 0.0 to 50.0 ct | | |

## Parameters (Dual Chorus1):

PAGE 1

| | | | |
|---|---|---|---|
| L Wet/Dry | -100 to 100%wet | R Wet/Dry | -100 to 100%wet |
| L Out Gain | Off, -79.0 to 24.0 dB | R Out Gain | Off, -79.0 to 24.0 dB |
| L Fdbk Lvl | -100 to 100% | R Fdbk Lvl | -100 to 100% |
| Xcouple | 0 to 100% | | |

PAGE 2

| | | | |
|---|---|---|---|
| L Tap Lvl | -100 to 100% | R Tap Lvl | -100 to 100% |
| L Tap Pan | -100 to 100% | R Tap Pan | -100 to 100% |
| L LFO Rate | 0.01 to 10.00 Hz | R LFO Rate | 0.01 to 10.00 Hz |
| L LFODepth | 0.0 to 50.0 ct | R LFO Depth | 0.0 to 50.0 ct |
| L Tap Dly | 0.0 to 1000.0 ms | R Tap Dly | 0.0 to 1000.0 ms |
| L HF Damp | 16 Hz to 25088 Hz | R HF Damp | 16 Hz to 25088 Hz |

PAGE 3

| | | | |
|---|---|---|---|
| L PitchEnv | Triangle or trapzoid | R PitchEnv | Triangle or trapzoid |

## Parameters (Dual Chorus 2):

PAGE 1

| | | | |
|---|---|---|---|
| L Wet/Dry | -100 to 100%wet | R Wet/Dry | -100 to 100%wet |
| L Out Gain | Off, -79.0 to 24.0 dB | R Out Gain | Off, -79.0 to 24.0 dB |
| L Fdbk Lvl | -100 to 100% | R Fdbk Lvl | -100 to 100% |
| Xcouple | 0 to 100% | | |

PAGE 2

| | | | |
|---|---|---|---|
| L Tap1 Lvl | -100 to 100% | R Tap1 Lvl | -100 to 100% |
| L Tap2 Lvl | -100 to 100% | R Tap2 Lvl | -100 to 100% |
| L Tap3 Lvl | -100 to 100% | R Tap3 Lvl | -100 to 100% |
| L Tap1 Pan | -100 to 100% | R Tap1 Pan | -100 to 100% |
| L Tap2 Pan | -100 to 100% | R Tap2 Pan | -100 to 100% |
| L Tap3 Pan | -100 to 100% | R Tap3 Pan | -100 to 100% |

PAGE 3

| | | | |
|---|---|---|---|
| L LFO1Rate | 0.01 to 10.00 Hz | R LFO1Rate | 0.01 to 10.00 Hz |
| L LFO2Rate | 0.01 to 10.00 Hz | R LFO2Rate | 0.01 to 10.00 Hz |
| L LFO3Rate | 0.01 to 10.00 Hz | R LFO3Rate | 0.01 to 10.00 Hz |
| L LFO1Dpth | 0.0 to 50.0 ct | R LFO1Dpth | 0.0 to 50.0 ct |
| L LFO2Dpth | 0.0 to 50.0 ct | R LFO2Dpth | 0.0 to 50.0 ct |
| L LFO3Dpth | 0.0 to 50.0 ct | R LFO3Dpth | 0.0 to 50.0 ct |

PAGE 4

| | | | |
|---|---|---|---|
| L Tap1 Dly | 0.0 to 1000.0 ms | R Tap1 Dly | 0.0 to 1000.0 ms |
| L Tap2 Dly | 0.0 to 1000.0 ms | R Tap2 Dly | 0.0 to 1000.0 ms |
| L Tap3 Dly | 0.0 to 1000.0 ms | R Tap3 Dly | 0.0 to 1000.0 ms |
| L HF Damp | 16 Hz to 25088 Hz | R HF Damp | 16 Hz to 25088 Hz |
| L PitchEnv | Triangle or trapzoid | R PitchEnv | Triangle or trapzoid |

Wet/Dry — The relative amount of input (dry) signal and chorus (wet) signal that appears in the final effect output mix. When set to 0%, the output is taken only from the input. When set to 100%, the output is all wet. Negative values polarity-invert the wet signal.

Out Gain — The overall gain or amplitude at the output of the effect.

Fdbk Level — The level of the feedback signal into the delay line. The feedback signal is taken from the LFO1 delay tap. Negative values polarity-invert the feedback signal.

Xcouple — Controls how much of the left channel input and feedback signals are sent to the right channel delay line and vice versa. At 50%, equal amounts from both channels are sent to both delay lines. At 100%, the left feeds the right delay and vice versa.

HF Damping — The amount of high-frequency content of the signal that is sent into the delay lines. This control determines the cutoff frequency of the one-pole (-6dB/octave) lowpass filter.

Pitch Env — The pitch of the chorus modulation can be made to follow a triangular “Triangle” envelope (rise-fall-rise-fall) or a trapzoidal “trapzoid” envelope (rise-hold-fall-hold).

Tap Lvl — Levels of the LFO-modulated delay taps. Negative values polarity-invert the signal. Setting any tap level to 0% effectively turns off the delay tap. Since these controls allow the full input level to pass through all the delay taps, a 100% setting on all the summed taps will significantly boost the wet signal relative to dry. A 50% setting may be more reasonable.

Tap Pan — The left or right output panning of the delay taps. The range is -100% for fully left to 100% for fully right. Setting the pan to 0% sends equal amounts to both left and right channels for center or mono panning. [Dual Chorus 1 & 2 only]

LFO Rate — Used to set the speeds of modulation of the delay lines. Low rates increase LFO excursion (see LFO Dpth below). If Pitch Env is set to “trapzoid”, you will be unable to put the rate on an FXMod or otherwise change the rate without introducing discontinuities (glitches or zippering) to your output signal. The triangular “Triangle” Pitch Env setting does allow smooth rate modulation, provided you’ve specified enough delay.

LFO Depth — The maximum depth of detuning of the LFO-modulated delay lines. The range is 0 to 50 cents, with a cent equal to 1/100th of a semitone. If you do not have enough delay specified with Tap Dly to get the depth you’ve dialed up, then Tap Dly will be forced to increase, which can cause discontinuities if signal is present. The LFOs move a tap back and forth across the delay lines to shift the pitch of the tapped signal. The maximum distance the taps get moved from the center position of the LFO is called the LFO excursion. Excursion is calculated from both the LFO depth and rate settings. Large depths and low rates produce large excursions. If Pitch Env is set to “trapzoid”, you will be unable to put the depth on an FXMod or otherwise change the depth without introducing discontinuities (glitches or zippering) to your output signal. The triangular “Triangle” Pitch Env setting does allow smooth depth modulation, provided you’ve specified enough delay.

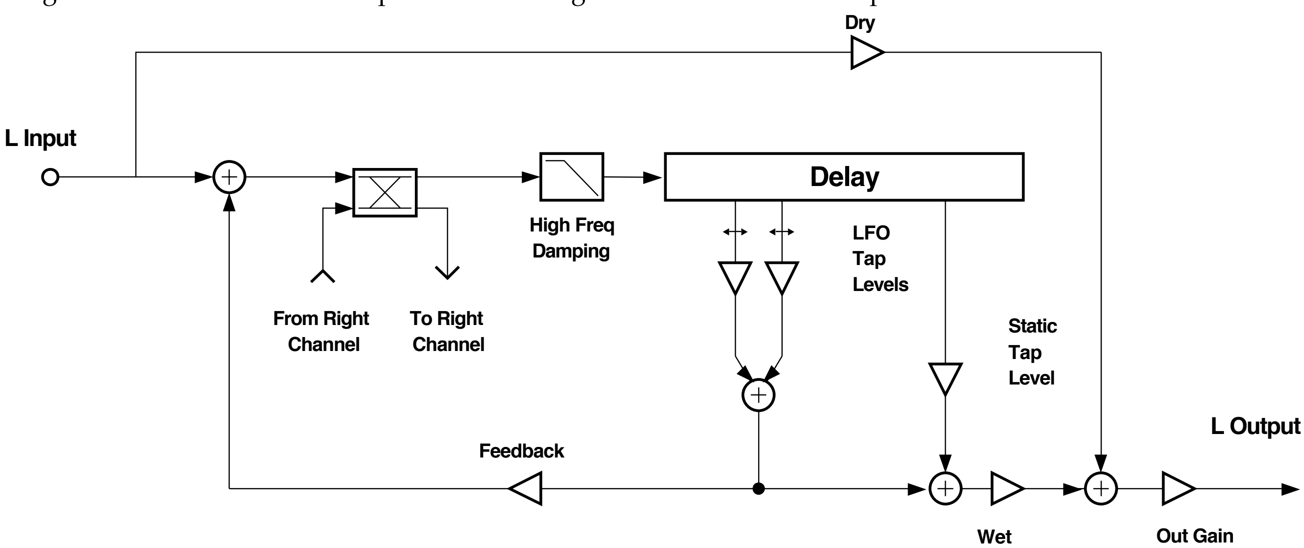
Tap Dly — The average delay length, or the delay to the center of the LFO sweep. If the delay is shorter than the LFO excursion, then the Tap Dly will be forced to a longer length, equal to the amount of required excursion (the parameter display will not change though). Changing this parameter while signal is present will cause signal discontinuities. It’s best to set and forget this one. Set it long enough so that there are no discontinuities with the largest Depth and lowest Rates that you will be using.

L/R Phase or LFO*n* LRPhs — In Chorus 1 and Chorus 2, the relative phases of the LFOs for the left and right channels may be adjusted.

# FXAlg #154: Flanger 1 • FXAlg #155: Flanger 2

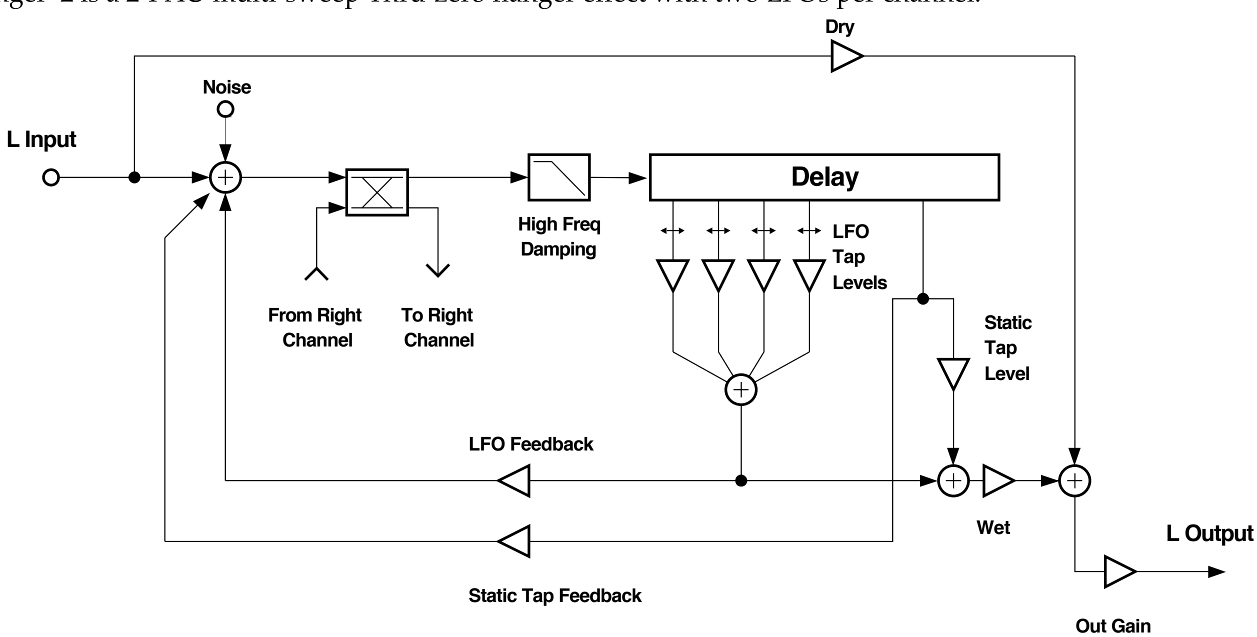
**Multi-tap flangers**
*Allocation Units:* 1 for Flanger 1; 2 for Flanger 2

Flanger 1 is a 1-PAU multi-sweep Thru-zero flanger effect with two LFOs per channel.




**Simplified block diagram of the left channel of Flanger 1; the Right channel is similar.**

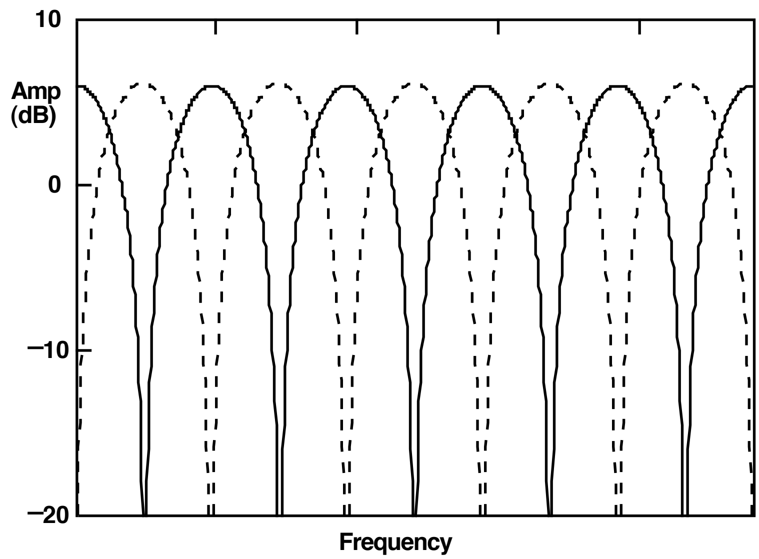
Flanger 2 is a 2-PAU multi-sweep Thru-zero flanger effect with two LFOs per channel.




**Simplified block diagram of the left channel of Flanger 2; the Right channel is similar.**

Flanging was originally created by summing the outputs of two un-locked tape machines while varying their sync by pressing a hand on the outside edge of one reel—thus the name “reel-flanging”. The key to achieving the flanging effect is the summing of a signal with a time-displaced replica of itself.

The result is a series of notches in the frequency spectrum. These notches are equally spaced in (linear) frequency at multiples whose wavelengths are equal to the time delay. The result is generally referred to as a comb filter (the name arising from the resemblance of the spectrum to a comb). If the levels of the signals being added or subtracted are the same, the notches will be of infinite depth (in dB) and the peaks will be up 6 dB. Flanging is achieved by time-varying the delay length, thus changing the frequencies of the notches. The shorter the delay time, the greater the notch separation. This delay time variation imparts a sense of motion to the sound. Typically the delay times are on the order of 0-5 ms. Longer times begin to get into the realm of chorusing, where the ear begins to perceive the audio output as nearly two distinct signals, but with a variable time displacement.

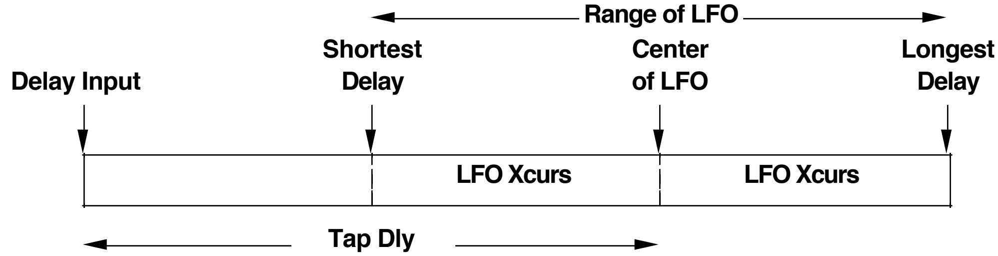



**Comb Filters - Solid Line for Addition, Dashed Line for Subtraction**

The heart of the flanger implemented here is a multi-tap delay line. You can set the level of each tap as a percentage of the input level, and the level may be negative (phase-inverted). One tap is a simple static delay over which you can control the length of delay (from the input tap). Four of the taps can have their lengths modulated up and down by a low frequency oscillator (LFO). You are given control of the rate of the LFOs, how far each LFO can sweep through the delay line, and the relative phases of the LFOs (i.e., whether the LFO is taking the taps from the input tap or bringing them toward it).

The flanger uses tempo units (based on the sequencer tempo or MIDI clock if you like), together with the number of tempo beats per LFO cycle. Thus if the tempo is 120 bpm (beats per minute) and the LFO Period is set to 1 beat, the LFOs will pass through 120 complete cycles in a minute or 2 cycles per second (2 Hz). Increasing the LFO Period increases the period of the LFOs (slows them down). An LFO Period setting of 16 beats will take 4 measures (in 4/4 time) for a complete LFO oscillation.

You can set how far each LFO can sweep through the delay line with the excursion controls (Xcurs). The excursion is the maximum distance an LFO will move from the center of its sweep. The total range of an LFO is twice the excursion. You set the delay to the center of LFO excursion with the Dly parameters. The excursion and delay controls both have coarse and fine adjustments. By setting the excursion to zero length, the LFO delay tap becomes a simple static tap. Note that modifying the delay to the center of LFO excursion will result in a sudden change of delay length and consequently, a discontinuity in the signal being read from the delay line. This can produce a characteristic zippering effect. The Dly parameters should be as long as the Xcurs parameters or longer, or else changing (or modulating) the excursion will force the center of LFO excursion to move, with the resulting signal discontinuities. The static delay tap does not suffer the zippering problem, and changes to its length will occur smoothly. You can assign the static delay tap to an FX Mod, and use the source controller to do manual flanging.




**Delay for a Single LFO**

Consider a simple example where you have an LFO tap signal being subtracted from the static delay tap signal. If the delays are set such that at certain times both taps are the same length, then both taps have the same signal and the subtraction produces a null or zero output. The effect is most pronounced when the static tap is set at one of the ends of the LFO excursion where the LFO tap motion is the slowest. This is the classic Thru-Zero flanger effect. Adding other LFO taps to the mix increases the complexity of the final sound, and obtaining a true Thru-Zero effect may take some careful setting of delays and LFO phases.

The flanger has a Wet/Dry control as well, which can further add complexity to the output as the dry signal is added to various delayed wet components for more comb filtering.

When using more than one LFO, you can set up the phase relationships between each of the LFOs. The LFOs of the left channel and those of the right channel will be set up in the same phase relationship except that you may offset the phases of the right channel as a group relative to the left channel (L/R Phase). L/R Phase is the only control which treats left and right channels differently and has a significant effect on the stereo image. If you have tempo set to the system tempo, the phases will maintain their synchronization with the tempo clock. At the beat of the tempo clock, a phase set to 0° will be at the center of the LFO excursion and moving away from the delay input.

Regenerative feedback has been incorporated in order to produce a more intense resonant effect. The signal is fed back is from the first LFO delay tap (LFO1), and has its own level control (Fdbk Level). In-phase spectral components arriving at the summer add together, introducing a series of resonant peaks in the frequency spectrum between the notches. The amplitude of these peaks depends on the degree of feedback, and they can be made very resonant.

Cross-coupling (Xcouple) allows the signals of the right and left channels to be mixed or swapped. The cross-coupling is placed after the summation of the feedback to the input signal. When feedback and cross-coupling are turned up, you will get a ping-pong effect between right and left channels.

A lowpass filter (HF Damping) right before the input to the delay line is effective in emulating the classic sounds of older analog flangers with their limited bandwidths (typically 5-6kHz).

As stated earlier, it is the movement of the notches created in the frequency spectrum that give the flanger its unique sound. It should be obvious that sounds with a richer harmonic structure will be effected in a much more dramatic way than harmonically starved sounds. Having more notches, i.e. a greater 'notch-density', should produce an even more intense effect. This increase in notch density may be achieved by having a number of modulating delay lines, all set at the same rate, but different depths. Setting the depths proportionately results in a more pleasing effect.

An often characteristic effect of flanging is the sound of system noise being flanged. Various pieces of analog gear add noise to the signal, and when this noise passes through a flanger, you can hear the noise "whooshing". In the K2500, the noise level is very low, and in fact if no sound is being played, there is no noise at all at this point in the signal chain. To recreate the effect of system noise flanging, white noise may be added to the input of the flanger signal (Flanger 2 only). Since white noise has a lot of high frequency content and may sound too bright, it may be tamed with a first-order lowpass filter.

## Parameters (Flanger 1):

### PAGE 1

| | | | |
|---|---|---|---|
| **Wet/Dry** | **-100 to 100% wet** | **Out Gain** | **Off, -79.0 to 24.0 dB** |
| **Fdbk Level** | **-100 to 100%** | **LFO Tempo** | **System, 1 to 255 BPM** |
| **Xcouple** | **0 to 100%** | **LFO Period** | **1/24 to 32 bts** |
| **HF Damping** | **16 to 25088 Hz** | | |

### PAGE 2

| | | | |
|---|---|---|---|
| **StatDlyLvl** | **-100 to 100%** | **L/R Phase** | **0.0 to 360.0 deg** |
| **LFO1 Level** | **-100 to 100%** | **LFO1 Phase** | **0.0 to 360.0 deg** |
| **LFO2 Level** | **-100 to 100%** | **LFO2 Phase** | **0.0 to 360.0 deg** |

### PAGE 3

| | | | |
|---|---|---|---|
| **StatDlyCrs** | **0.0 to 228.0 ms** | | |
| **StatDlyFin** | **-127 to 127 samp** | | |
| **Xcurs1 Crs** | **0.0 to 228.0 ms** | **Dly1 Crs** | **0.0 to 228.0 ms** |
| **Xcurs1 Fin** | **-127 to 127 samp** | **Dly1 Fin** | **-127 to 127 samp** |
| **Xcurs2 Crs** | **0.0 to 228.0 ms** | **Dly2 Crs** | **0.0 to 228.0 ms** |
| **Xcurs2 Fin** | **-127 to 127 samp** | **Dly2 Fin** | **-127 to 127 samp** |

## Parameters (Flanger 2):

### PAGE 1

| | | | |
|---|---|---|---|
| **Wet/Dry** | **-100 to 100%wet** | **Out Gain** | **Off, -79.0 to 24.0 dB** |
| **LFO Fdbk** | **-100 to 100%** | **Stat Fdbk** | **-100 to 100%** |
| **Xcouple** | **0 to 100%** | **LFO Tempo** | **System, 1 to 255 BPM** |
| **HF Damping** | **16 Hz to 25088 Hz** | **LFO Period** | **1/24 to 32 bts** |

**PAGE 2**

| | | | |
|---|---|---|---|
| Noise Gain | Off, -79.0 to -30.0 dB | Noise LP | 16 to 25088 Hz |
| StatDlyLvl | -100 to 100% | L/R Phase | 0.0 to 360.0 deg |
| LFO1 Level | -100 to 100% | LFO1 Phase | 0.0 to 360.0 deg |
| LFO2 Level | -100 to 100% | LFO2 Phase | 0.0 to 360.0 deg |
| LFO3 Level | -100 to 100% | LFO3 Phase | 0.0 to 360.0 deg |
| LFO4 Level | -100 to 100% | LFO4 Phase | 0.0 to 360.0 deg |

**PAGE 3**

| | | | |
|---|---|---|---|
| StatDlyCrs | 0.0 to 228.0 ms | | |
| StatDlyFin | -127 to 127 samp | | |
| Xcurs1 Crs | 0.0 to 228.0 ms | Xcurs3 Crs | 0.0 to 228.0 ms |
| Xcurs1 Fin | -127 to 127 samp | Xcurs3 Fin | -127 to 127 samp |
| Xcurs2 Crs | 0.0 to 228.0 ms | Xcurs4 Crs | 0.0 to 228.0 ms |
| Xcurs2 Fin | -127 to 127 samp | Xcurs4 Fin | -127 to 127 samp |

**PAGE 4**

| | | | |
|---|---|---|---|
| Dly1 Crs | 0.0 to 228.0 ms | Dly3 Crs | 0.0 to 228.0 ms |
| Dly1 Fin | -127 to 127 samp | Dly3 Fin | -127 to 127 samp |
| Dly2 Crs | 0.0 to 228.0 ms | Dly4 Crs | 0.0 to 228.0 ms |
| Dly2 Fin | -127 to 127 samp | Dly4 Fin | -127 to 127 samp |

Wet/Dry — The relative amount of input signal and flanger signal that is to appear in the final effect output mix. When set to 0%, the output is taken only from the input (dry). When set to 100%, the output is all wet. Negative values polarity-invert the wet signal.

Out Gain — The overall gain or amplitude at the output of the effect.

Fdbk Level — The level of the feedback signal into the delay line. The feedback signal is taken from the LFO1 delay tap. Negative values polarity-invert the feedback signal.

Xcouple — How much of the left channel input and feedback signals are sent to the right channel delay line and vice versa. At 50%, equal amounts from both channels are sent to both delay lines. At 100%, the left feeds the right delay and vice versa. Xcouple has no effect if Fdbk Level is set to 0%.

HF Damping — The amount of high frequency content of the signal sent into the delay lines. This control determines the cutoff frequency of the one-pole (-6dB/octave) lowpass filters.

LFO Tempo — Basis for the rates of the LFOs, as referenced to a musical tempo in bpm (beats per minute). When this parameter is set to “System”, the tempo is locked to the internal sequencer tempo or to incoming MIDI clocks. When it is set to “System”, sources (FUNs, LFOs, ASRs etc.) will have no effect on the Tempo parameter.

LFO Period — Sets the LFO rate based on the Tempo determined above: the number of beats corresponding to one period of the LFO cycle. For example, if the LFO Period is set to “4”, the LFOs will take four beats to pass through one oscillation, so the LFO rate will be 1/4th of the Tempo setting. If it is set to “6/24” (=1/4), the LFO will oscillate four times as fast as the Tempo. At “0”, the LFOs stop oscillating and their phase is undetermined (wherever they stopped).

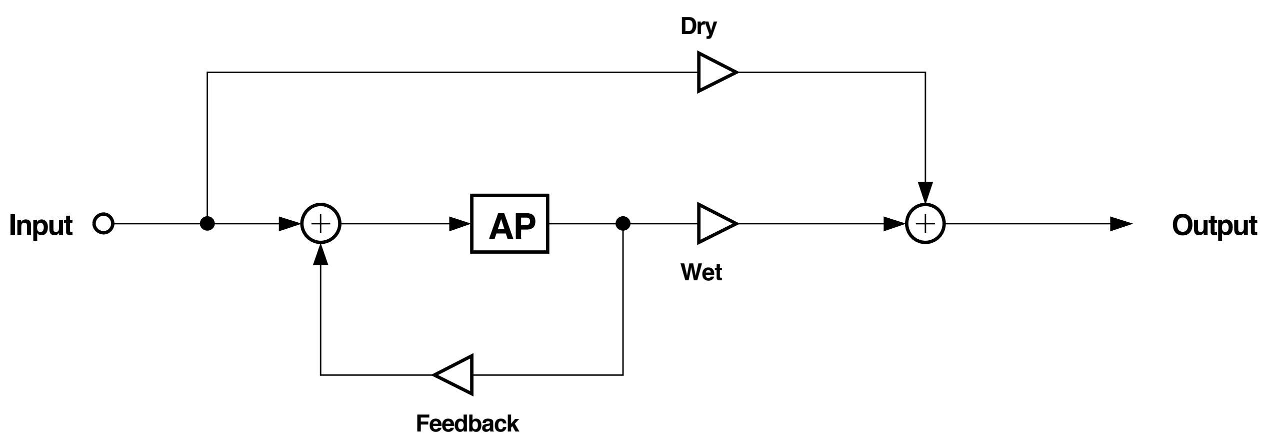
| | |
|---|---|
| Noise Gain | The amount of noise (dB relative to full scale) to add to the input signal. In many flangers, you can hear the noise floor of the signal being flanged, but in the K2500, if there is no input signal, there is no noise floor unless it is explicitly added. [Flanger 2 only] |
| Noise LP | The cut-off frequency of a one-pole lowpass filter acting on the injected noise. The lowpass removes high frequencies from an otherwise pure white noise signal. [Flanger 2 only] |
| StatDlyCrs | The coarse adjustment to the static delay tap length. The name suggests the tap is stationary, but it can be connected through an FX Mod to a control source to smoothly vary the delay length. The range for all delays and excursions is 0 to 230 ms, but for flanging the range 0 to 5 ms is most effective. |
| StatDlyFin | A fine adjustment to the static delay tap length. The resolution is one sample. |
| StatDlyLvl | The level of the static delay tap. Negative values polarity-invert the signal. Setting the tap level to 0% turns off the delay tap. |
| Xcurs *n* Crs | These set how far the LFO-modulated delay taps can move from the center of their ranges. The total range of the LFO sweep is twice the excursion. If the excursion is set to 0, the LFO does not move and the tap behaves like a simple delay line set to the minimum delay. The excursion cannot be made longer than the delay to the center of excursion (see Dly Crs and Dly Fin below) because delays cannot be made shorter than 0. If you attempt longer excursions, the length of the Dly Crs/Fin will be forced to increase (though you will not see the increased length displayed in the Dly Crs/Fin parameters), and you will hear discontinuities in the signal, usually in the form of zipper noise. The range for all delays and excursions is 0 to 230 ms, but for flanging the range 0 to 5 ms is most effective. This parameter is a coarse adjustment for the excursion. |
| Xcurs *n* Fin | A fine adjustment for the LFO excursions. The resolution is one sample. |
| Dly *n* Crs | The delay to the center of LFO tap range. The maximum delay will be this delay plus the LFO excursion delay. The minimum delay will be this delay minus the LFO excursion delay. Since delays cannot be less than 0 ms in length, the this delay length will be increased if LFO excursion is larger than this delay length. The range for all delays and excursions is 0 to 230 ms, but for flanging the range 0 to 5 ms is most effective. This parameter is a coarse adjustment for the delay. |
| Dly *n* Fin | A fine adjustment to the minimum delay tap lengths. The resolution is one sample. |
| LFO*n* Level | The levels of the LFO modulated delay taps. Negative values polarity-invert the signal. Setting any tap level to 0% turns off the delay tap. |
| LFO*n* Phase | The phase angles of the LFOs relative to each other and to the system tempo clock, if turned on (see Tempo). For example, if one LFO is set to 0° and another is set to 180°, then when one LFO delay tap is at its shortest, the other will be at its longest. If the system tempo clock is on, the LFOs are synchronized to the clock with absolute phase. A phase of 0° will put an LFO tap at the center of its range and its lengthening. |
| L/R Phase | Adds the specified phase angle to the right channel LFOs. In all other respects the right and left channels are symmetric. By moving this control away from 0°, the stereo sound field is broken up and a stereo image becomes difficult to spatially locate. The effect is usually described as “phasey”. It tends to impart a greater sense of motion. |

# FX Algs #156-160: Phasers

## FXAlg #156: LFO Phaser
## FXAlg #157: LFO Phaser Twin
## FXAlg #158: Manual Phaser
## FXAlg #159: Vibrato Phaser
## FXAlg #160: SingleLFO Phaser

**A variety of single notch/bandpass Phasers**
*Allocation Units:* 1 (each)

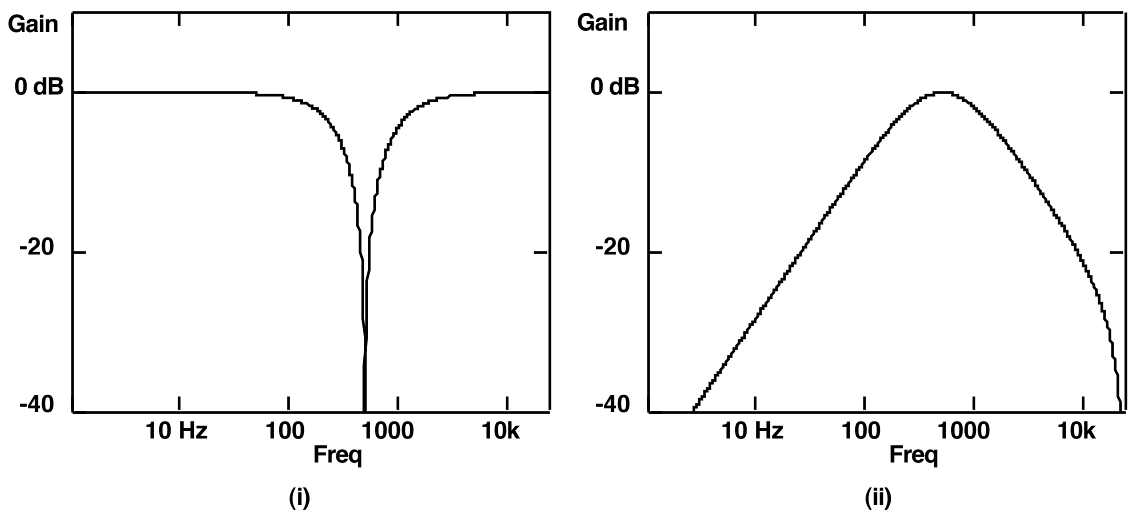
A simple phaser is an algorithm which produces a vague swishing or phasey effect. When the phaser signal is combined with the dry input signal or the phaser is fed back on itself, peaks and/or notches can be produced in the filter response making the effect much more pronounced. Most of the phaser algorithms presented here have built in low frequency oscillators (LFOs) to generate the motion of the phasers. In the case of Manual Phaser, the phaser motion is left to the user.




**One channel of a typical phaser**

A phaser uses a special filter called an all-pass filter to modify the phase response of a signal's spectrum without changing the amplitude of the spectrum. As the name suggests, an all-pass filter by itself does not change the amplitude of the frequency response of a signal passing through it—it does not cut or boost any frequencies. It does cause some frequencies to be delayed a little in time, and this small time shift is also known as a phase change. The frequency where the phase change has its greatest effect is a parameter that you can control. By modulating the frequency of the phaser, you get the swishy phaser sound. With a modulation rate of around 6 Hz, an effect similar to vibrato may be obtained, but only in a limited range of filter frequencies.

By adding the phaser output to the dry input using, for example, a Wet/Dry parameter, you can produce peaks and notches in the frequency response. At frequencies where the phaser is "in phase" with the dry signal, the signal level doubles (or there is a 6 dB level increase approximately). At frequencies where the phaser and dry signals are "out of phase", the two signals cancel each other out and there is a notch in the frequency response. You can get a complete notch when Wet/Dry is set to 50%. If subtraction is used instead of addition by setting Wet/Dry to -50%, then the notches become peaks and the peaks become notches.




**Response of typical phaser: (i) Wet/Dry = 50% and (ii) Wet/Dry = -50%.**

Some of the phaser algorithms have feedback. When feedback is used, it can greatly exaggerate the peaks and notches, producing a much more resonant sound.

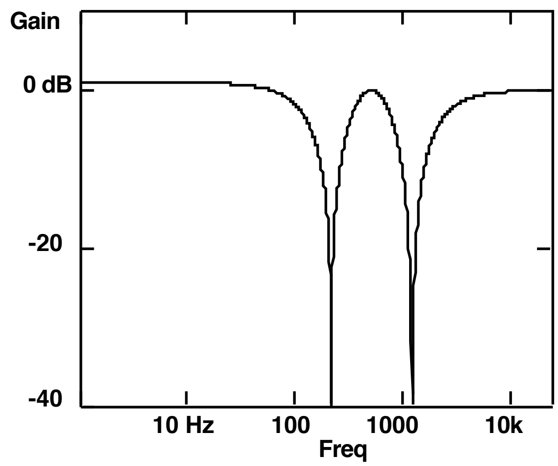
LFO Phaser is a simple phaser algorithm with Wet/Dry and Fdbk Level parameters. Two LFOs used to control the filter frequency and the depth of the resulting notch. You can control the depths, rates, and phases of both the LFOs. The algorithm is stereo so the relative phases of the LFOs for the left and right channels can be set. When setting the LFO that controls the filter frequency, you specify the center frequency around which the LFO will modulate and the depth of the LFO. The depth specifies how many cents (hundredths of a semitone) to move the filter frequency up and down.

The NotchDepth parameter provides an alternative way of combining wet and dry phaser signals to produce a notch. In this case the parameter specifies the depth of the notch in decibels (dB). The depth of the notch can be modulated with the notch LFO. The notch LFO is completely independent of the frequency, and their rates may be different. The relative phases of the notch and frequency LFOs (N/F Phase) only has meaning when the LFOs are running at the same rate. As with all KDFX LFO phases, it is not recommended to directly modulate the phase settings with an FXMod.

SingleLFO Phaser is identical to LFO Phaser except that the notch and frequency LFOs always run at the same rate.

Manual Phaser leaves the phaser motion up to you, so it has no built in LFOs. Manual Phaser has a Notch/BP parameter which produces a complete notch at the center frequency when Wet/Dry is set to -100% and a resonant bandpass when set to 100%. At 0% the signal is dry. To get phaser motion, you have to change the filter center frequencies (left and right channels) yourself. The best way to do this is with an FXMod. There are also feedback parameters for the left and right channels.

LFO Phaser Twin produces a pair of notches separated by a spectral peak. The center frequency parameter sets the frequency of the center peak. Like LFO Phaser, the filter frequency can be modulated with a built in LFO. The Notch/Dry parameter produces a pair of notches when set to 100%. The output signal is dry when set to 0% and at 200%, the signal is a pure (wet) allpass response. LFO Phaser Twin does not have Out Gain or feedback parameters.




**Response of LFO Phaser Twin with Wet/Dry set to 100%.**

In the Vibrato Phaser algorithm, the bandwidth of the phaser filter can be adjusted exactly like a parametric EQ filter. The In Width controls how the stereo input signal is routed through the effect. At 100% In Width, left input is processed to the left output, and right to right. Lower In Width values narrow the input stereo field until at 0%, the processing is mono. Negative values reverse left and right channels. The dry signal is not affected by In Width. As described earlier, setting Wet/Dry to 50% will produce a full notch. At -50% Wet/Dry, you get a bandpass.

## Parameters (LFO Phaser):

### PAGE 1

| | | | |
|---|---|---|---|
| Wet/Dry | 0 to 100%wet | Out Gain | Off, -79.0 to 24.0 dB |
| Fdbk Level | -100 to 100% | | |

### PAGE 2

| | | | |
|---|---|---|---|
| CenterFreq | 16 to 25088 Hz | NotchDepth | -79.0 to 6.0 dB |
| FLFO Depth | 0 to 5400 ct | NLFO Depth | 0 to 100% |
| FLFO Rate | 0.00 to 10.00 Hz | NLFO Rate | 0.00 to 10.00 Hz |
| FLFO LRPhs | 0.0 to 360.0 deg | NLFO LRPhs | 0.0 to 360.0 deg |
| | | N/F Phase | 0.0 to 360.0 deg |

## Parameters (SingleLFO Phaser):

**PAGE 1**

| | | | |
|---|---|---|---|
| Wet/Dry | 0 to 100%wet | Out Gain | Off, -79.0 to 24.0 dB |
| Fdbk Level | -100 to 100% | | |

**PAGE 2**

| | | | |
|---|---|---|---|
| LFO Rate | 0.00 to 10.00 Hz | N/F Phase | |
| CenterFreq | 16 to 25088 Hz | NotchDepth | -79.0 to 6.0 dB |
| FLFO Depth | 0 to 5400 ct | NLFO Depth | 0 to 100% |
| FLFO LRPhs | 0.0 to 360.0 deg | NLFO LRPhs | 0.0 to 360.0 deg |

Wet/Dry — The amount of phaser (wet) signal relative to unaffected (dry) signal as a percent.

Out Gain — The output gain in decibels (dB) to be applied to the combined wet and dry signals.

Fdbk Level — The phaser output can be added back to its input to increase the phaser resonance. Negative values polarity-invert the feedback signal.

LFO Rate — The rate of both the center frequency LFO and the notch depth LFO for the SingleLFO Phaser algorithm.

CenterFreq — The nominal center frequency of the phaser filter. The frequency LFO modulates the phaser filter centered at this frequency.

FLFO Depth — The depth in cents that the frequency LFO sweeps the phaser filter above and below the center frequency.

FLFO Rate — The rate of the center frequency LFO for the LFO Phaser algorithm.

FLFO LRPhs — Sets the phase difference between the left and right channels of the center frequency LFO. A setting of 180 degrees results in one channel being at the minimum frequency while the other channel is at the maximum.

NotchDepth — The nominal depth of the notch. The notch depth LFO modulates the depth of the notch. For maximum LFO depth, set NotchDepth to 0 dB and NLFO Depth to 100%.

NLFO Depth — The excursion of the notch depth LFO in units of percentage of the total range. The depth of the LFO is limited to the range of the NotchDepth parameter such that a full 100% modulation is only possible when the NotchDepth is at the center of its range (0 dB).

NLFO Rate — The rate of the notch depth LFO for the LFO Phaser algorithm.

NLFO LRPhs — The phase difference between the left and right channels of the notch depth LFO. A setting of 180 degrees results in one channel being at highest amplitude while the other channel is at lowest amplitude.

N/F Phase — The phase difference between the notch depth and center frequency LFOs. For LFO Phaser, this parameter is largely meaningless unless the FMod Rate and NMod Rate are set identically.

## Parameters (Manual Phaser):

**PAGE 1**

| | | | |
|---|---|---|---|
| Notch/BP | -100 to 100% | Out Gain | Off, -79.0 to 24.0 dB |
| L Feedback | -100 to 100% | R Feedback | -100 to 100% |
| L Ctr Freq | 16 to 25088 Hz | R Ctr Freq | 16 to 25088 Hz |

Notch/BP — The amount of notch depth or bandpass. At -100% there is a complete notch at the center frequency. At 100% the filter response is a peak at the center frequency. 0% is the dry unaffected signal.

Out Gain — The output gain in decibels (dB) to be applied to the final output.

Feedback — The phaser output can be added back to its input to increase the phaser resonance (left and right). Negative values polarity-invert the feedback signal.

Ctr Freq — The nominal center frequency of the phaser filter (left and right). For a true phaser effect you may want to modulate these parameters by setting up FX Mods.

## Parameters (LFO Phaser Twin):

**PAGE 1**

| | | | |
|---|---|---|---|
| Notch/Dry | 0 to 200% | | |
| CenterFreq | 16 to 25088 Hz | LFO Rate | 0.00 to 10.00 Hz |
| LFO Depth | 0 to 5400 ct | L/R Phase | 0.0 to 360.0 deg |

Notch/Dry — The amount of phaser (wet) signal relative to unaffected (dry) signal as a percent. At 100% the phaser produces a pair of full notches above and below the center frequency. At 200% the output is a pure all-pass response (no amplitude changes, but phase changes centered about the center frequency).

CenterFreq — The nominal center frequency of the phaser filter. When configured for a maximum notch (Notch/Dry is 100%), the CenterFreq specifies the frequency of the peak between two notches. The LFO modulates the phaser filter centered at this frequency.

LFO Rate — The rate of the phaser frequency modulating LFO in Hertz.

LFO Depth — The depth in cents that the frequency LFO sweeps the phaser filter above and below the center frequency.

L/R Phase — The phase difference between the left and right channels of the LFO. A setting of 180 degrees results in one channel being at the minimum frequency while the other channel is at the maximum.

## Parameters (Vibrato Phaser):

**PAGE 1**

| | | | |
|---|---|---|---|
| Wet/Dry | -100 to 100%wet | Out Gain | Off, -79.0 to 24.0 dB |

**PAGE 2**

| | | | |
|---|---|---|---|
| CenterFreq | 16 to 25088 Hz | Bandwidth | 0.010 to 5.000 oct |
| LFO Depth | 0 to 100% | L/R Phase | 0.0 to 360.0 deg |
| LFO Rate | 0.00 to 10.00 Hz | | |
| | | In Width | -100 to 100% |

Wet/Dry — The amount of phaser (wet) signal relative to unaffected (dry) signal as a percent. When set to 50% you get a complete notch. When set to -50%, the response is a bandpass filter. 100% is a pure allpass filter (no amplitude changes, but a strong phase response).

Out Gain — The output gain in decibels (dB) to be applied to the combined wet and dry signals.

CenterFreq — The nominal center frequency of the phaser filter. The frequency LFO modulates the phaser filter centered at this frequency.

Bandwidth — If the phaser is set to behave as a sweeping notch or bandpass, the bandwidth of the notch or bandpass is set with Bandwidth. This parameter works the same as for parametric EQ filter bandwidths.

LFO Depth — The depth that the frequency LFO sweeps the phaser filter above and below the center frequency as a percent.

LFO Rate — The rate of the LFO in Hertz. The LFO Rate may be scaled up by the Rate Scale parameter.

L/R Phase — Sets the phase difference between the left and right channels of the center frequency LFO. A setting of 180 degrees results in one being at a at the minimum frequency while the other channel is at the maximum.

In Width — The width of the stereo field that passes through the stereo phaser filtering. This parameter does not affect the dry signal. When set to 100%, the left and right channels are processed to their respective outputs. Smaller values narrow the stereo image until at 0% the input channels are summed to mono and set to left and right outputs. Negative values interchange the left and right channels.

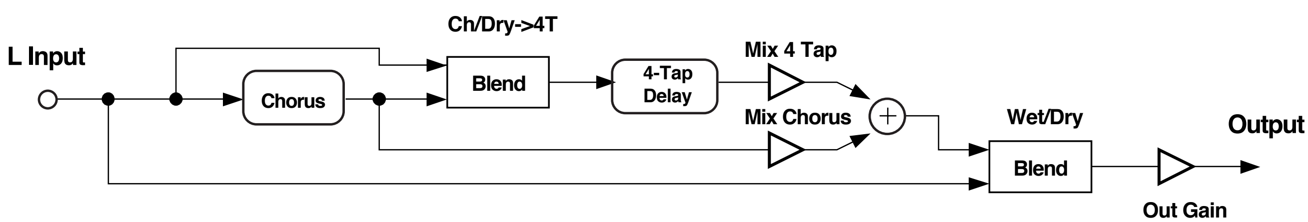
# Combination Algorithms [“+”]

## FXAlg #700 — Chorus+Delay
## FXAlg #701 — Chorus+4Tap
## FXAlg #703 — Chor+Dly+Reverb
## FXAlg #706 — Flange+Delay
## FXAlg #707 — Flange+4Tap
## FXAlg #709 — Flan+Dly+Reverb
## FXAlg #722 — Pitcher+Chor+Dly
## FXAlg #723 — Pitcher+Flan+Dly

**A family of combination effect algorithms**
*Allocation Units:* 1 or 2

## Signal Routing (2 effects)

The algorithms listed above with two effects can be arranged in series or parallel. The output of the first effect in the algorithm name—A—is wired to the input of the second effect—B—and the input into effect B is a mix of effect A and the algorithm input dry signal. The effect B input mix is controlled by a parameter *A*/Dry>*B*. For example, in Chorus+Delay, the parameter name is “Ch/Dry>Dly”. The value functions much like a wet/dry mix where 0% means that only the algorithm input dry signal is fed into effect B, thus putting the effects in parallel; and 100% means only the output of effect A is fed into effect B, thus putting the effects in series.




**An example of routing using Chorus+4Tap**

Both effect A and B outputs are mixed at the algorithm output to become the wet signal. These mix levels are controlled with the two parameters that begin with “Mix”. These allow only one or both effect outputs to be heard. Negative mix amounts polarity-invert the signal, which can change the character of each effect when mixed together or with the dry signal. The Wet/Dry parameter adjusts the balance between the sum of both effects determined by the Mix parameters, and the input dry signal. Negative Wet/Dry values polarity-invert the summed wet signal relative to dry.

### Two-Effect Routing:

| | | | |
|---|---|---|---|
| Wet/Dry | -100 to 100% | Out Gain | Off; -79.0 to 24.0 dB |
| Mix Effect | -100 to 100% | | |
| Mix Effect | -100 to 100% | | |
| | | A/Dry->B | 0 to 100% |

| | |
|---|---|
| Mix *Effect* | Adjusts the amount of each effect that is mixed together as the algorithm wet signal. Negative values polarity-invert that particular signal. |
| *A*/Dry->*B* | This parameter controls how much of the A effect is mixed with dry and fed into the B effect. A and B are designated in the algorithm name. This control functions like a wet/dry mix, where 0% is completely dry and 100% is effect A only. |

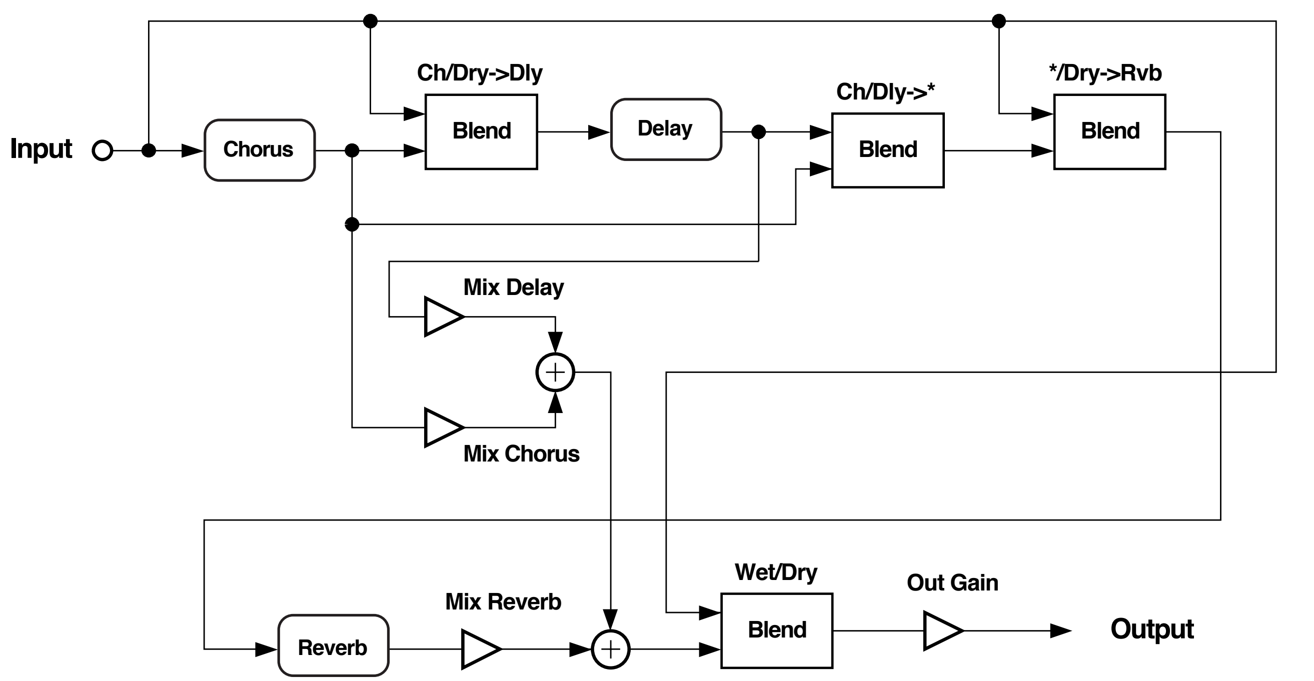
# Signal Routing (3 effects)

The algorithms listed above with three effects allow serial or parallel routing between any three effects. Effect A is wired to the input of effect B and C, and effect B is wired into effect C. The input of effect B is a mix between effect A and the algorithm dry input. The input into effect C is a three-way mix between effect A, effect B, and the dry signal.

As in the two-effect routing, the input of effect B is controlled by a parameter *A*/Dry>*B*. For example, in Chor+Dly+Rvb, the parameter name is “Ch/Dry>Dly”.

The input into effect C is controlled by two parameters named A/B ->* and */Dry->C where A, B, and C correspond to the names of effects A, B, and C. The first parameter mixes effect A and B into a temporary buffer represented by the symbol “*”. The second parameter mixes this temporary buffer “*” with the dry signal to be fed into effect C. These mixing controls function similarly to Wet/Dry parameters. A setting of 0% only mixes the effect to the right of the “/” in the parameter name, while 100% only mixes the effect to the left of the “/”. Negative values polarity-invert the numerator’s signal.

Effects A, B, and C outputs are mixed at the algorithm output to become the wet signal. Separate mixing levels are provided for left and right channels, and are named “L Mix” or “R Mix”. Negative mix amounts polarity-invert the signal which can change the character of each effect when mixed together or with the dry signal. The Wet/Dry parameter adjusts the balance between the sum of all effects determined by the Mix parameters, and the input dry signal. Negative Wet/Dry values polarity-invert the summed wet signal relative to dry.




**An example of routing using Chorus+Delay+Reverb**

**Three-Effect Routing:**

| | | | |
|---|---|---|---|
| Wet/Dry | -100 to 100% | Out Gain | Off; -79.0 to 24.0 dB |
| L Mix Effect A | -100 to 100% | R Mix Effect A | -100 to 100% |
| L Mix Effect B | -100 to 100% | R Mix Effect B | -100 to 100% |
| L Mix Effect C | -100 to 100% | R Mix Effect C | -100 to 100% |

| | |
|---|---|
| A/Dry>B | -100 to 100% |
| A/B ->* | -100 to 100% |
| */Dry->C | -100 to 100% |

L Mix *Effect*, R Mix *Effect* Adjusts the amount of each effect that is mixed together as the algorithm wet signal. Separate left and right controls are provided. Negative values polarity-invert that particular signal.

*A*/Dry>*B* This parameter controls how much of the A effect is mixed with dry and fed into the B effect. A and B are designated in the algorithm name. This control functions like a wet/dry mix, where 0% is completely dry and 100% is effect A only.

*A*/*B* ->* This parameter is first of two parameters that control whet is fed into effect C. This adjusts how much of the effect A is mixed with effect B, the result of which is represented as the symbol “*”. 0% is completely B effect, and 100% is completely A effect. negative values polarity-invert the A effect.

*/Dry->*C* This parameter is the second of two parameters that control whet is fed into effect C. This adjusts how much of the “*” signal (sum of effects A and B determined by *A*/*B* ->*) is mixed with the dry signal and fed into effect C. 0% is completely dry signal, and 100% is completely “*” signal.

# Individual Effect Components

## Chorus:

The choruses are basic 1 tap dual choruses. Separate LFO controls are provided for each channel. Slight variations between algorithms exist. Some algorithms offer separate left and right feedback controls, while some offer only one for both channels. Also, cross-coupling and high frequency damping may be offered in some and not in others. Parameters associated with chorus control begin with “Ch” in the parameter name. A general description of chorus functionality can be found in the Chorus section of this book, FXAlgs #150-153.

| | | | |
|---|---|---|---|
| Ch PtchEnv | Triangle or trapzoid | | |
| Ch Rate L | 0.01 to 10.00 Hz | Ch Rate R | 0.01 to 10.00 Hz |
| Ch Depth L | 0.0 to 100 ct | Ch Depth R | 0.0 to 100 ct |
| Ch Delay L | 0 to 1000 ms | Ch Delay R | 0 to 1000 ms |
| Ch Fdbk | -100 to 100% | | |
| Ch Xcouple | 0 to 100% | Ch HF Damp | 16 to 25088 Hz |

## Flange:

The flangers are basic 1-tap dual flangers. Separate LFO controls are provided for each channel. Slight variations between algorithms exist. Some algorithms offer separate left and right feedback controls, while some offer only one for both channels. Also, cross-coupling and high-frequency damping may be offered in some and not in others. Parameters associated with flange control begin with “Fl” in the parameter name. A general description of flanger functionality can be found in the Flanger section or this book, FXAlgs #154-155.

In addition to the LFO delay taps, some flangers may offer a static delay tap for creating through-zero flange effects. The maximum delay time for this tap is 230ms and is controlled by the Fl StatDly parameter. Its level is controlled by the Fl StatLvl parameter.

**PAGE 1**

| | | | |
|---|---|---|---|
| Fl Tempo | System; 1 to 255 BPM | Fl HF Damp | 16 to 25088 Hz |
| Fl Rate | 0.01 to 10.00 Hz | | |
| Fl Xcurs L | 0 to 230 ms | Fl Xcurs R | 0 to 230 ms |
| Fl Delay L | 0 to 230 ms | Fl Delay R | 0 to 230 ms |
| Fl Fdbk L | -100 to 100% | Fl Fdbk R | -100 to 100% |
| Fl Phase L | 0 to 360 deg | Fl Phase R | 0 to 360 deg |

**PAGE 2**

| | |
|---|---|
| Fl HF Damp | 16 to 25088 Hz |
| Fl Xcouple | 0 to 100% |
| Fl StatDly | 0 to 230 ms |
| Fl StatLvl | -100 to 100% |

## Delay:

The Delay is a basic tempo based dual channel delay with added functionality, including image shifting and high-frequency damping. Separate left and right controls are generally provided for delay time and feedback. Parameters associated with a delay effect in a combination algorithm begin with “Dly”.

The delay length for each channel is determined by Dly Tempo, expressed in beats per minute (BPM), and the delay length (Dly Time L and Dly Time R) of each channel is expressed in beats (bts). The tempo alters both channel delay lengths together. With the tempo in beats per minute and delay lengths in beats, you can calculate the length of a delay in seconds as beats/tempo * 60 (sec/min). Since KDFX has a limited amount of delay memory available (usually 1.5 seconds for these delays), selecting slow tempos and/or long delay lengths may cause you to run out of delay memory. At this point, each delay will pin at its maximum possible time. Because of this, when you slow down the tempo, you may find the delays lose their sync.

Delay regeneration is controlled by Dly Fdbk. In some combinations, separate left and right feedback controls are provided, while in others there is a single control for both channels.

Dly Image and Dly HFDamp are just like the HFDamp and Image parameters found in other algorithms. Not all delays in combination algorithms will have both of these parameters.

| | | | |
|---|---|---|---|
| **Dly Time L** | **0 to 32 bts** | **Dly Time R** | **0 to 32 bts** |
| **Dly Fdbk L** | **-100 to 100%** | **Dly Fdbk R** | **-100 to 100%** |
| **Dly HFDamp** | **0 to 32 bts** | **Dly Image** | **-100 to 100%** |

Dly Time L, Dly Time R — The delay lengths of each channel in beats. The duration of a beat is specified with the Tempo parameter. The delay length in seconds is calculated as beats/tempo * 60 (sec/min).

Dly Fdbk L, Dly Fdbk R — The amount of the output of the effect that is fed back to the input.

Dly HFDamp — Controls the cutoff frequency of a 1 pole (6dB/oct slope) lowpass filter in the feedback path. The filter is heard when either Dly Fdbk or LsrCntour is used.

Dly Image — This parameter controls the amount of image shifting during each feedback regeneration, and is heard only when Dly Fdbk is used. Small positive values shift the image to the right, while small negative values shift to the left. Larger values tend to shift the image so far that the image gets scrambled, and in some cases create ambience.

## Combination 4-Tap:

Combination 4-Tap is a tempo based 4 tap delay with feedback used in combination algorithms. Parameters associated with the 4 tap effect start with “4T”. The control over the feedback tap and individual output taps is essentially the same as the 4-Tap Delay BPM algorithm, with the exception that the delay times will pin at the maximum delay time instead of automatically cutting their times in half. For more information, see the section on 4-Tap Delay BPM of this book, FXAlg #132.

**PAGE 1**

| | |
|---|---|
| **4T Tempo** | **System; 1 to 255 BPM** |
| **4T LoopLen** | **0 to 8 bts** |
| **4T FB Lvl** | **-100 to 100%** |

**PAGE 2:**

| | | | |
|---|---|---|---|
| **Tap1 Delay** | **0 to 8 bts** | **Tap3 Delay** | **0 to 8 bts** |
| **Tap1 Level** | **-100 to 100%** | **Tap3 Level** | **-100 to 100%** |
| **Tap1 Bal** | **-100 to 100%** | **Tap3 Bal** | **-100 to 100%** |
| **Tap2 Delay** | **0 to 8 bts** | **Tap4 Delay** | **0 to 8 bts** |
| **Tap2 Level** | **-100 to 100%** | **Tap4 Level** | **-100 to 100%** |
| **Tap2 Bal** | **-100 to 100%** | **Tap4 Bal** | **-100 to 100%** |

## Reverb:

The reverbs offered in these combination effects is MiniVerb. See FXAlg #1 in this book for information about the parameters. Parameters associated with this reverb begin with “Rv”.

| | | | |
|---|---|---|---|
| | | Rv Type | Hall1 |
| | | Rv Time | 0.5 to 30.0 s; Inf |
| Rv DiffScl | 0.00 to 2.00x | Rv Density | 0.00 to 4.00x |
| Rv SizeScl | 0.00 to 4.00x | Rv HF Damp | 16 to 25088 Hz |
| Rv PreDlyL | 0 to 620 ms | Rv PreDlyR | 0 to 620 ms |

## Pitcher:

The pitchers offered in these effects are the same as that found in its standalone version. Review the Pitcher section of this book, FXAlg #908, for more information. Parameters associated with this effect begin with “Pt”.

| | |
|---|---|
| Pt Pitch | C-1 to G9 |
| Pt Offset | -12.0 to 12.0 ST |
| Pt Odd Wts | -100 to 100% |
| Pt PairWts | -100 to 100% |
| Pt 1/4 Wts | -100 to 100% |
| Pt 1/2 Wts | -100 to 100% |

# Configurable Combination Algorithms [“<>”]

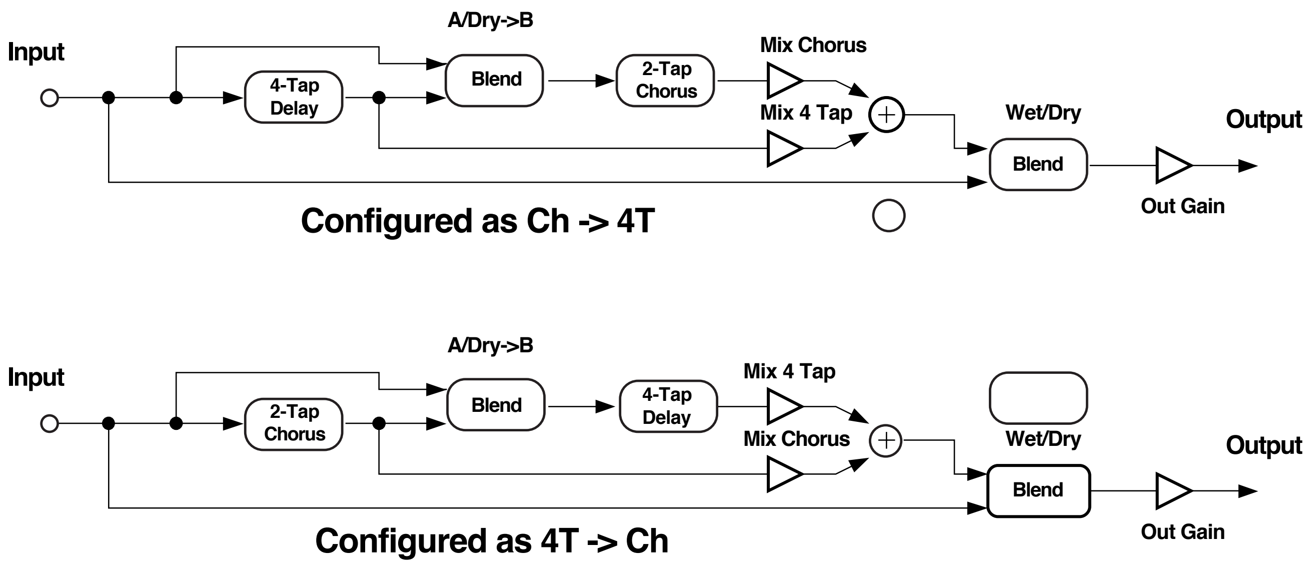
## FXAlg #702 — Chorus<>4Tap
## FXAlg #704 — Chorus<>Reverb
## FXAlg #705 — Chorus<>LasrDly
## FXAlg #708 — Flange<>4Tap
## FXAlg #710 — Flange<>Reverb
## FXAlg #711 — Flange<>LasrDly
## FXAlg #712 — Flange<>Pitcher
## FXAlg #713 — Flange<>Shaper
## FXAlg #717 — LasrDly<>Reverb
## FXAlg #718 — Shaper<>Reverb

**A family of re-configurable combination effect algorithms**
*Allocation Units:* 2

## Signal Routing

Each of these combination algorithms offer two separate effects combined with a flexible signal-routing mechanism. This mechanism allows the two effects to either be in series bi-directionally or in parallel. This is done by first designating one effect “A”, and the other “B” where the output of effect A is always wired to effect B. A and B are assigned with the A->B cfg parameter. For example, when A->B cfg is set to “Ch->Dly”, then effect A is the chorus, and effect B is the delay, and the output of the chorus is wired to the input of the delay. The amount of effect A fed into effect B is controlled by the A/Dry->B parameter. This controls the balance between effect A output, and the algorithm dry input signal fed into effect B behaving much like a wet/dry mix. When set to 0%, only the dry signal is fed into B allowing parallel effect routing. At 100%, only the A output is fed into B, and at 50%, there is an equal mix of both. For an example of signal flow in the Chor<>4Tap algorithm, see the figure on the next page.

Both effect A and B outputs are mixed at the algorithm output to become the wet signal. These mix levels are controlled with the two parameters that begin with “Mix”. These allow only one or both effect outputs to be heard. Negative mix amounts polarity-invert the signal which can change the character of each effect when mixed together or with the dry signal. The Wet/Dry parameter adjusts the balance between the sum of both effects determined by the Mix parameters, and the input dry signal. Negative Wet/Dry values polarity-invert the summed wet signal relative to dry.




**Algorithm 702, Chor<>4Tap, when *A->B cfg* is set to (top) “Ch->4T” and (bottom) “4T->Ch”.**

**Bi-directional Routing:**

| | | | |
|---|---|---|---|
| Wet/Dry | -100 to 100% | Out Gain | Off; -79.0 to 24.0 dB |
| Mix Effect | -100 to 100% | | |
| Mix Effect | -100 to 100% | | |
| A->B cfg | EffectA>EffectB | A/Dry->B | 0 to 100% |

<u>Mix *Effect*</u> Adjusts the amount of each effect that is mixed together as the algorithm wet signal. Negative values polarity-invert that particular signal.

<u>A->B cfg</u> This parameter controls the order of the effects routing. The output of effect A is wired into the input of effect B. So, when set to Ch->4T for example, effect A is chorus, and effect B is 4-tap. This is used in conjunction with the A/Dry->B parameter.

<u>A/Dry->B</u> This parameter controls how much of the A effect is mixed with dry and fed into the B effect. A and B are determined by the A->B cfg parameter. This works like a wet/dry mix, where 0% is completely dry and 100% is effect A only.

# Individual Effect Components

## Configurable Chorus and Flange:

The configurable chorus and flange have 2 moving delay taps per channel. Parameters associated with chorus control begin with “Ch” in the parameter name, and those associated with flange begin with “Fl”. General descriptions of chorus and flange functionality can be found in the Chorus (FXAlg #150) or Flange (FXAlg #154) sections of this book.

Since these effects have 2 taps per channel, control over four LFOs is necessary with a minimum number of user parameters (Fig. 1). This is accomplished by offering two sets of LFO controls with three user interface modes: Dual1Tap, Link1Tap, or Link2Tap. These are selectable with the LFO cfg parameter and affect the functionality of the two sets of rate, depth and delay controls (and also phase and feedback controls for the flange). Each parameter is labeled with a 1 or a 2 in the parameter name to indicate to which control set it belongs. Control set 1 consists of controls whose name ends with a 1, and control set 2 consists of controls whose name ends with a 2.

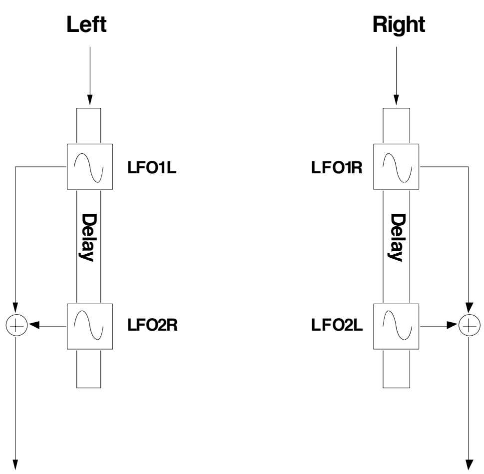



**Fig. 1 — LFO delay taps in the configurable chorus and flange**

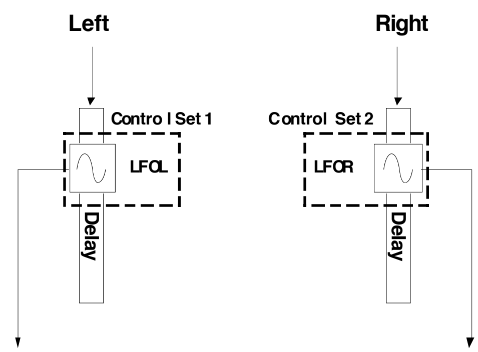



**Fig. 2 — LFO control in Dual1Tap mode**

In Dual1Tap mode (Fig. 2), each control set independently controls one tap in each channel. This is useful for dual mono applications where separate control over left and right channels is desired. Control set 1 controls the left channel, and control set 2 controls the right channel. The second pair of moving delay taps are disabled in this mode. LRPhase is unpredictable unless both rates are set to the same speed. Then, the phase value is accurate only after the LFOs are reset. LFOs can be reset by either changing the LFO cfg parameter, or loading in the algorithm by selecting a preset or studio that uses it. For user-friendly LRPhase control, use either the Link1Tap or Link2Tap modes.

In Link1Tap mode (Fig. 3), control set 1 controls 1 tap in both the left and right channels. Control set 2 has no effect, and the second pair of LFO delay taps are disabled. This mode is optimized for an accurate LRPhase relationship between the left and right LFOs.

In Link2Tap mode (Fig. 4), control set 1 controls the first left and right pair of LFOs, while control set 2 controls the second pair. This mode uses all 4 LFOs for a richer sound, and is optimized for LRPhase relationships. Each of the 2 taps per channel are summed together at the output, and the Fdbk parameters control the sum of both LFO taps on each channel fed back to the input.

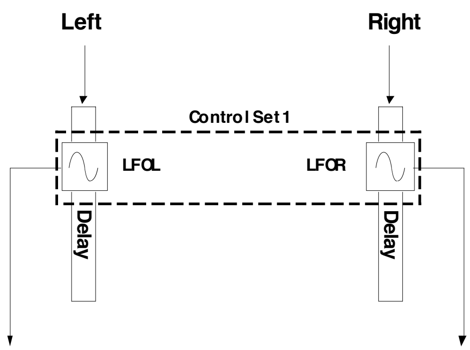



Fig. 3 — LFO control in Link1Tap mode

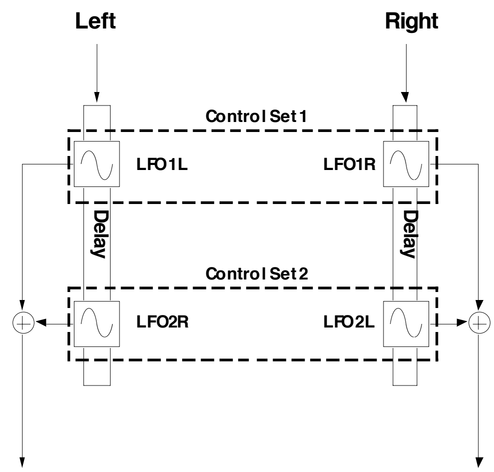



Fig. 4 — LFO control in Link2Tap mode

In addition to the LFO delay taps, the flange offers a static delay tap for creating through-zero flange effects. The maximum delay time for this tap is 230ms and is controlled by the Fl StatDly parameter. Its feedback amount is controlled by the Fl StatFB. Separate mix levels for the LFO taps and the static tap are then controlled by the Fl StatLvl and Fl LFO Lvl controls. The feedback and level controls can polarity-invert each signal by setting them to negative values.

**Chorus:**

| | | | |
|---|---|---|---|
| Ch LFO cfg | Dual1Tap... | Ch LRPhase | 0 to 360 deg |
| Ch Rate 1 | 0.01 to 10.00 Hz | Ch Rate 2 | 0.01 to 10.00 Hz |
| Ch Depth 1 | 0.0 to 100 ct | Ch Depth 2 | 0.0 to 100 ct |
| Ch Delay 1 | 0 to 1000 ms | Ch Delay 2 | 0 to 1000 ms |
| Ch Fdbk L | -100 to 100% | Ch Fdbk R | -100 to 100% |
| Ch Xcouple | 0 to 100% | Ch HF Damp | 16 to 25088 Hz |

**Flange (PAGE 1):**

| | | | |
|---|---|---|---|
| Fl LFO cfg | Dual1Tap... | Fl LRPhase | 0 to 360 deg |
| Fl Rate 1 | 0.01 to 10.00 Hz | Fl Rate 2 | 0.01 to 10.00 Hz |
| Fl Xcurs 1 | 0 to 230 ms | Fl Xcurs 2 | 0 to 230 ms |
| Fl Delay 1 | 0 to 1000 ms | Fl Delay 2 | 0 to 1000 ms |
| Fl Fdbk 1 | -100 to 100% | Fl Fdbk 2 | -100 to 100% |
| Fl Phase 1 | 0 to 360 deg | Fl Phase 2 | 0 to 360 deg |

**Flange (PAGE 2):**

| | |
|---|---|
| **Fl HF Damp** | **16 to 25088 Hz** |
| **Fl Xcouple** | **0 to 100%** |
| **Fl StatDly** | **0 to 230 ms** |
| **Fl StatFB** | **-100 to 100%** |
| **Fl StatLvl** | **-100 to 100%** |
| **Fl LFO Lvl** | **-100 to 100%** |

Ch LFO cfg — Sets the user interface mode for controlling each of the 4 chorus LFOs.

Ch LRPhase — Controls the relative phase between left channel LFOs and right channel LFOs. In Dual1Tap mode, however, this parameter is accurate only when Ch Rate 1 and Ch Rate 2 are set to the same speed, and only after the Ch LFO cfg parameter is moved, or the algorithm is called up.

Ch Fdbk L, Ch Fdbk R — These control the amount that the output of the chorus is fed back into the input.

Fl LFO cfg — Sets the user interface mode for controlling each of the 4 flange LFOs.

Fl LRPhase — Controls the relative phase between left channel LFOs and right channel LFOs. In Dual1Tap mode, however, this parameter is accurate only when Fl Rate 1 and Fl Rate 2 are set to the same speed, and only after the Fl LFO cfg parameter is moved, or the algorithm is called up.

Fl Phase 1, Fl Phase 2 — These adjust the phase relationship between the corresponding LFO and the internal beat clock.

## Laser Delay:

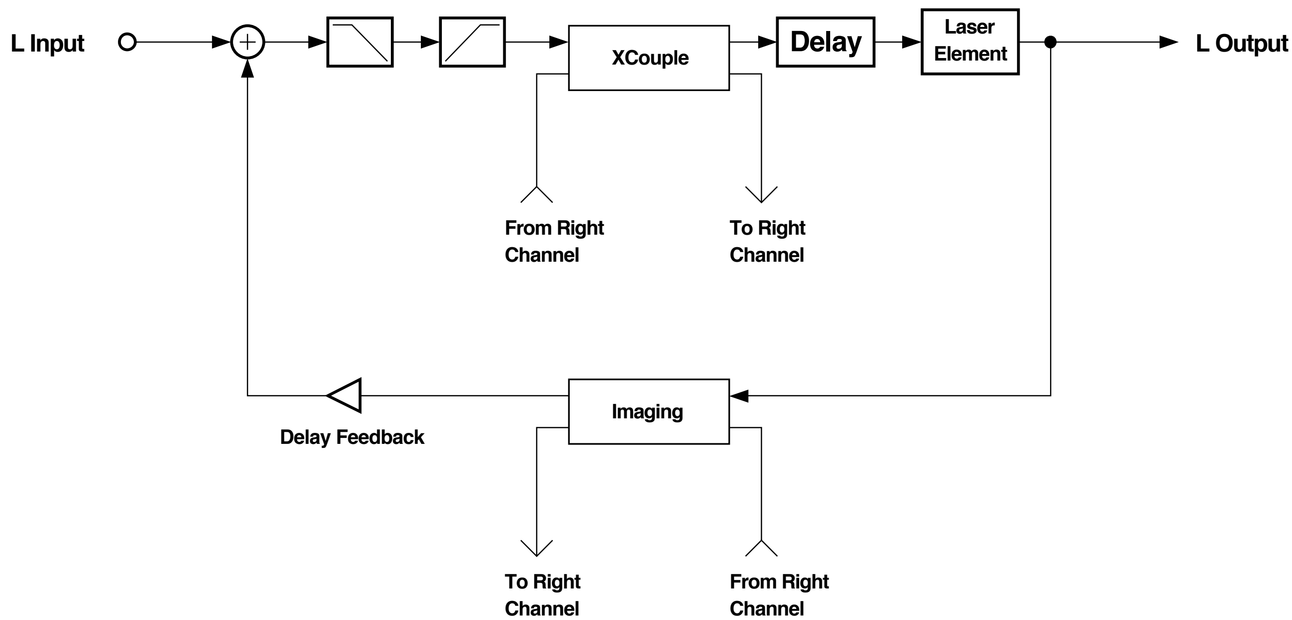
Laser Delay is a tempo based delay with added functionality, including image shifting, cross-coupling, high frequency damping, low frequency damping, and a LaserVerb element. Separate left and right controls are provided for delay time, feedback, and laser controls. Parameters associated with Laser Verb in a combination algorithm begin with “Dly” or “Lsr”.

The delay length for each channel is determined by Dly Tempo, expressed in beats per minute (BPM), and the delay length (Dly Time L and Dly Time R) of each channel is expressed in beats (bts). The tempo alters both channel delay lengths together. With the tempo in beats per minute and delay lengths in beats, you can calculate the length of a delay in seconds as beats/tempo * 60 (sec/min). Since KDFX has a limited amount of delay memory available (usually 1.5 seconds for Laser Delay), selecting slow tempos and/or long delay lengths may cause you to run out of delay memory. At this point, each delay will pin at its maximum possible time. When you slow down the tempo, you may find the delays lose their sync.

The laser controls perform similarly to those found in LaserVerb (see FXAlg# 911 in this book), and affect the laser element of the effect. The LsrCntour changes the laser regeneration envelope shape. Higher values increase the regeneration amount, and setting it to 0% will disable the Laser Delay portion completely turning the effect into a basic delay. LsrSpace controls the impulse spacing of each regeneration. Low values create a strong initial pitched quality with slow descending resonances, while higher values cause the resonance to descend faster through each regeneration. See the LaserVerb section for more detailed information.

Delay regeneration is controlled collectively by the Dly Fdbk and LsrCntour parameters since the laser element contains feedback within itself. Setting both to 0% defeats all regeneration, including the laser element entirely. Increasing either one will increase regeneration overall, but with different qualities. Dly Fdbk is a feedback control in the classic sense, feeding the entire output of the effect back into the input, with negative values polarityinverting the signal. The LsrCntour parameter adds only the Laser Delay portion of the effect, including it’s own regeneration. For the most intense laser-ness, keep Dly Fdbk at 0% while LsrCntour is enabled.

Dly FBImag, Dly Xcouple, Dly HFDamp, and Dly LFDamp are just like those found in other algorithms. Not all Laser Delays in combination algorithms will have all four of these parameters.




**Laser Delay (left channel)**

| | | | |
|---|---|---|---|
| Dly Time L | 0 to 6 bts | Dly Time R | 0 to 6 bts |
| Dly Fdbk L | -100 to 100% | Dly Fdbk R | -100 to 100% |
| Dly HFDamp | 0 to 32 bts | Dly FBImag | -100 to 100% |
| Dly LFDamp | 0.10 to 6.00 x | Dly Xcple | 0 to 100% |
| LsrCntourL | 0 to 100% | LsrCntourR | 0 to 100% |
| LsrSpace L | 0 to 100 samp | LsrSpace R | 0 to 100 samp |

Dly Time L, Dly Time R — The delay lengths of each channel in beats. The duration of a beat is specified with the Tempo parameter. The delay length in seconds is calculated as beats/tempo * 60 (sec/min).

Dly Fdbk L, Dly Fdbk R — The amount of the output of the effect that is fed back to the input.

Dly HFDamp — Controls the cutoff frequency of a 1 pole (6dB/oct slope) lowpass filter in the feedback path. The filter is heard when either Dly Fdbk or LsrCntour is used.

Dly LFDamp — Controls the cutoff frequency of a 1 pole (6dB/oct slope) hipass filter in the feedback path. The filter is heard when either Dly Fdbk or LsrCntour is used.

Dly FBImag — This parameter controls the amount of image shifting during each feedback regeneration, and is heard only when Dly Fdbk is used. Small positive values shift the image to the right, while small negative values shift to the left. Larger values tend to shift the image so far that the image gets scrambled, and in some cases create ambience.

Dly Xcple — This parameter controls the amount of signal that is swapped between the left and right channels through each feedback generation when Dly Fdbk is used. A setting of 0% has no affect. 50% causes equal amounts of signal to be present in both channels causing the image to collapse into a center point source. A setting of 100% causes the left and right channels to swap each regeneration, which is also referred to as “ping-ponging”. The regeneration affects of cross-coupling are not heard when LsrCntour is used by itself.

<u>LsrCntour L, LsrCntour R</u> Controls the overall envelope shape of the laser regeneration. When set to a high value, sounds passing through will start at a high level and slowly decay. As the control value is reduced, it takes some time for the effect to build up before decaying. When the Contour is set to zero, the laser portion is turned off turning regeneration into straight feedback.

<u>LsrSpace L, LsrSpace R</u> Determines the starting pitch of the descending resonance and how fast it descends. See the section on Laser Reverb for more detailed information.

## Combination 4-Tap:

Combination 4-Tap is a tempo-based 4-tap delay with feedback used in combination algorithms. Parameters associated with the 4 tap effect start with “4T”. The control over the feedback tap and individual output taps is essentially the same as the 4-Tap Delay BPM algorithm (see FXAlg #132 in this book), with the exception that the delay times will pin at the maximum delay time instead of automatically cutting their times in half. Additionally, the feedback path may also offer cross-coupling, an imager, a hipass filter, and/or a lowpass filter.

**PAGE 1**

| | |
|---|---|
| **4T LoopLen** | **0 to 32 bts** |
| **4T FB Lvl** | **-100 to 100%** |
| **4T FB Imag** | **-100 to 100%** |
| **4T FB XCpl** | **0 to 100%** |
| **4T HF Damp** | **16 to 25088 Hz** |
| **4T LF Damp** | **16 to 25088 Hz** |

**PAGE 2**

| | | | |
|---|---|---|---|
| **Tap1 Delay** | **0 to 32 bts** | **Tap3 Delay** | **0 to 32 bts** |
| **Tap1 Level** | **-100 to 100%** | **Tap3 Level** | **-100 to 100%** |
| **Tap1 Bal** | **-100 to 100%** | **Tap3 Bal** | **-100 to 100%** |
| **Tap2 Delay** | **0 to 32 bts** | **Tap4 Delay** | **0 to 32 bts** |
| **Tap2 Level** | **-100 to 100%** | **Tap4 Level** | **-100 to 100%** |
| **Tap2 Bal** | **-100 to 100%** | **Tap4 Bal** | **-100 to 100%** |

<u>4T FB Imag</u> This parameter controls the amount of image shifting during each feedback regeneration. Small positive values shift the image to the right, while small negative values shift to the left. Larger values tend to shift the image so far that the image gets scrambled, and in some cases create ambience.

<u>4T FB Xcpl</u> This parameter controls the amount of signal that is swapped between the left and right channels through each feedback regeneration. A setting of 0% has no affect. 50% causes equal amounts of signal to be present in both channels causing the image to collapse into a center point source. A setting of 100% causes the left and right channels to swap each regeneration, which is also referred to as “ping-ponging”.

<u>4T LoopLen</u>, <u>4T Tempo</u>, <u>4T FB Lvl</u>, <u>Tap*n* Delay</u>, <u>Tap*n* Level</u>, <u>Tap*n* Bal</u> Refer to 4-Tap Delay BPM documentation.

## Reverb:

The reverbs offered in these combination effects is MiniVerb. Information about it can be found in the MiniVerb documentation (FXAlg#1 in this book). Parameters associated with this reverb begin with “Rv”.

| | | | |
|---|---|---|---|
| **Rv Type** | **Hall1** | | |
| **Rv Time** | **0.5 to 30.0 s; Inf** | | |
| **Rv DiffScl** | **0.00 to 2.00x** | **Rv Density** | **0.00 to 4.00x** |
| **Rv SizeScl** | **0.00 to 4.00x** | **Rv HF Damp** | **16 to 25088 Hz** |
| **Rv PreDlyL** | **0 to 620 ms** | **Rv PreDlyR** | **0 to 620 ms** |

## Pitcher:

The pitchers offered in these effects are the same as that found in its standalone version. Review the Pitcher section of this book, FXAlg #908, for more information. Parameters associated with this effect begin with “Pt”.

| | |
|---|---|
| **Pt Pitch** | **C-1 to G9** |
| **Pt Offset** | **-12.0 to 12.0 ST** |
| **Pt Odd Wts** | **-100 to 100%** |
| **Pt PairWts** | **-100 to 100%** |
| **Pt 1/4 Wts** | **-100 to 100%** |
| **Pt 1/2 Wts** | **-100 to 100%** |

## Shaper:

The shaper offered in these combination effects have the same qualities as those found in VAST. Refer to the section on shapers in the *K2500 Performance Guide* for an overview. Parameters associated with this effect begin with “Shp”.

This KDFX shaper also offers input and output 1 pole (6dB/oct) lowpass filters controlled by the Shp Inp LP and Shp Out LP respectively. There is an additional output gain labeled Shp OutPad to compensate for the added gain caused by shaping a signal.

| | |
|---|---|
| **Shp Inp LP** | **16 to 25088 Hz** |
| **Shp Amt** | **0.10 to 6.00 x** |
| **Shp Out LP** | **16 to 25088 Hz** |
| **Shp OutPad** | **Off; -79.0 to 0.0 dB** |

Shp Inp LP — Adjusts the cutoff frequency of the 1 pole (6dB/oct) lowpass filter at the input of the shaper.

Shp Out LP — Adjusts the cutoff frequency of the 1 pole (6dB/oct) lowpass filter at the output of the shaper.

Shp Amount — Adjusts the shaper intensity. This is exactly like the one in VAST.

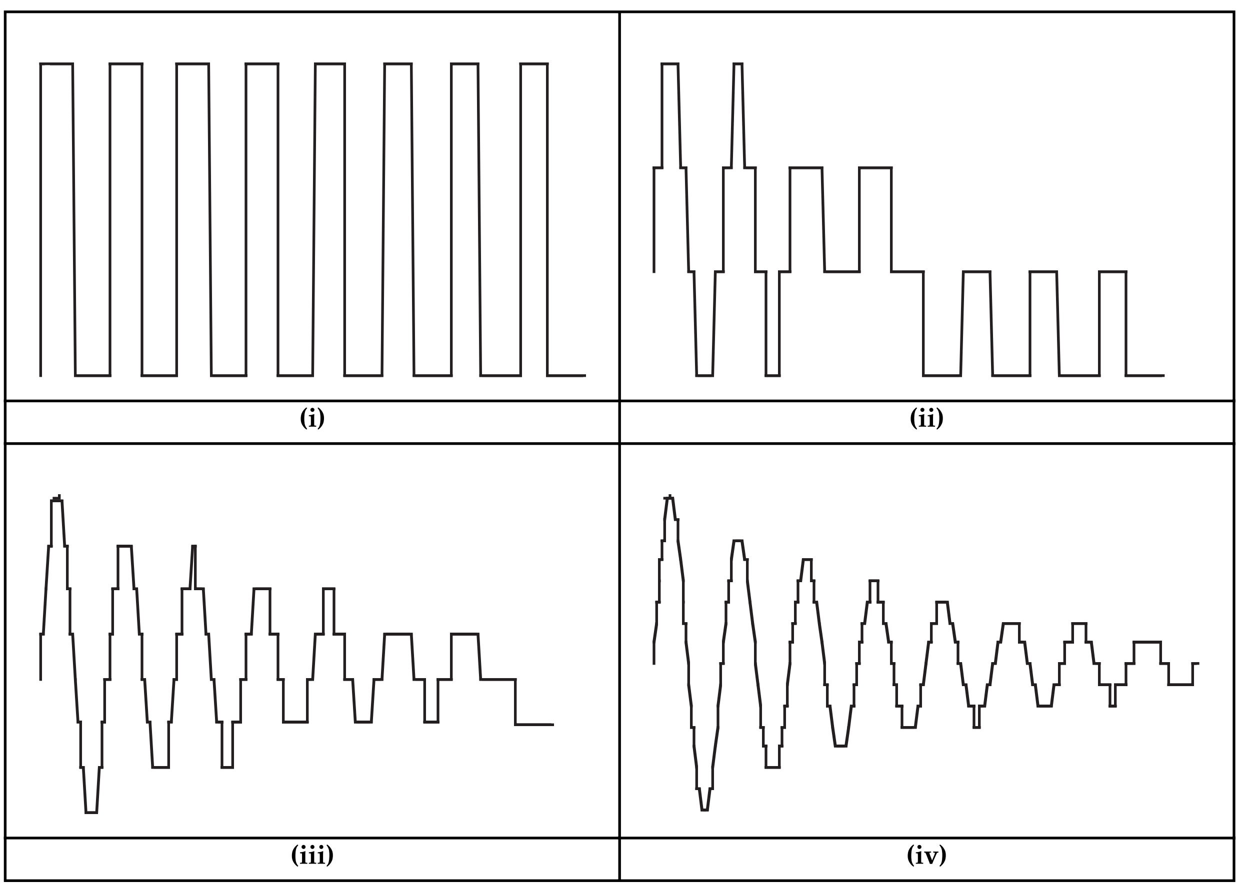
Shp OutPad — Adjusts the output gain at the output of the shaper to compensate for added gain caused by the shaper.

# FXAlg #714: Quantize+Flange

**Digital quantization followed by flanger.**
*Allocation Units:* 1

Digital audio engineers will go to great lengths to remove, or at least hide the effects of digital quantization distortion. In Quantize+Flange we do quite the opposite, making quantization an in-your-face effect. The quantizer will give your sound a dirty, grungy, perhaps industrial sound. As you've already gathered from the name, the quantization is followed by a flanger. Quantize+Flange is a stereo effect.

Quantization distortion is a digital phenomenon caused by having only a limited number of bits with which to represent signal amplitudes (finite precision). You are probably aware that a bit is a number which can have only one of two values: 0 or 1. When we construct a data or signal word out of more than one bit, each additional bit will double the number of possible values. For example a two-bit number can have one of four different values: 00, 01, 10 or 11. A three-bit number can take one of eight different values, a four-bit number can take one of sixteen values, etc.  The 18 bits of the K2500's digital-to-analog converter (DAC) represents 262,144 different amplitude levels ($2^{18}$). Let's take a look at how finite precision of digital words affects audio signals. The figures following are plots of a decaying sine wave with varying word lengths.



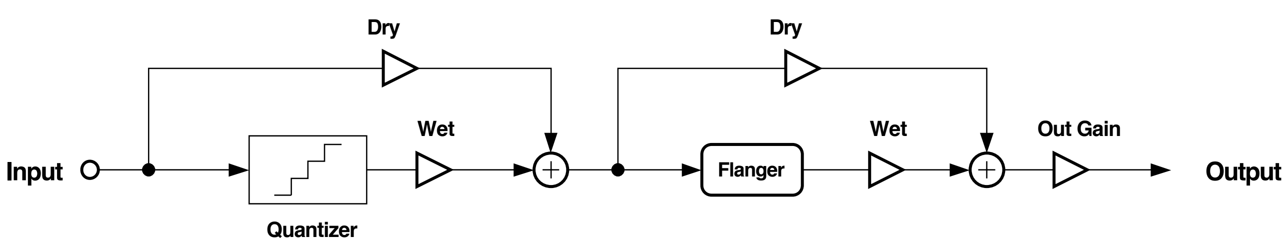
**A decaying sine wave represented with different word lengths: (i) 1-bit, (ii) 2-bit, (iii) 3-bit, (iv) 4-bit.**

Clearly a one-bit word gives a very crude approximation to the original signal while four bits is beginning to do a good job of reproducing the original decaying sine wave. When a good strong signal is being quantized (its word length is being shortened), quantization usually sounds like additive noise. But notice that as the signal decays in the above figures, fewer and fewer quantization levels are being exercised until, like the one bit example, there are only two levels being toggled. With just two levels, your signal has become a square wave.

Controlling the bit level of the quantizer is done with the DynamRange (dynamic range) parameter. At 0 dB we are at a one-bit word length. Every 6 dB adds approximately one bit, so at 144 dB, the word length is 24 bits. The quantizer works by cutting the gain of the input signal, making the lowest bits fall off the end of the word. The signal is then boosted back up so we can hear it. At very low DynamRange settings, the step from one bit level to the next can become larger than the input signal. The signal can still make the quantizer toggle between bit level whenever the signal crosses the zero signal level, but with the larger bit levels, the output will get louder and louder. The Headroom parameter prevents this from happening. When the DynamRange parameter is lower than the Headroom parameter, no more signal boost is added to counteract the cut used to quantize the signal. Find the DynamRange level at which the output starts to get too loud, then set Headroom to that level. You can then change the DynamRange value without worrying about changing the signal level. Headroom is a parameter that you set to match your signal level, then leave it alone.

At very low DynamRange values, the quantization becomes very sensitive to dc offset. It affects where your signal crosses the digital zero level. A dc offset adds a constant positive or negative level to the signal. By adding positive dc offset, the signal will tend to quantize more often to a higher bit level than to a lower bit level. In extreme cases (which is what we're looking for, after all), the quantized signal will sputter, as it is stuck at one level most of the time, but occasionally toggles to another level.

A flanger with one LFO delay tap and one static delay tap follows the quantizer. See the section in this book on multi-tap flangers (FXAlgs #154-155) for a detailed explanation of how the flanger works.




**Block diagram of one channel of Quantize+Flange.**

Quant W/D is a wet/dry control setting the relative amount of quantized (wet) and not quantized (dry) signals being passed to the flanger. The Flange W/D parameter similarly controls the wet/dry mix of the flanger. The dry signal for the flanger is the wet/dry mix output from the quantizer.

## Parameters:

**PAGE 1**

| | | | |
|---|---|---|---|
| In/Out | In or Out | Out Gain | Off, -79.0 to 24.0 dB |
| Quant W/D | 0 to 100% | DynamRange | 0 to 144 dB |
| Flange W/D | -100 to 100% | dc Offset | -79.0 to 0.0 dB |
| | | Headroom | 0 to 144 dB |

**PAGE 2**

| | | | |
|---|---|---|---|
| **Fl Tempo** | **System, 1 to 255 BPM** | **Fl Fdbk** | **-100 to 100%** |
| **Fl Period** | **0 to 32 bts** | | |
| **Fl L Phase** | **0.0 to 360.0 deg** | **Fl R Phase** | **0.0 to 360.0 deg** |
| **Fl StatLvl** | **-100 to 100%** | **Fl LFO Lvl** | **-100 to 100%** |

**PAGE 3**

| | | | |
|---|---|---|---|
| **FlStatDlyC** | **0.0 to 230.0 ms** | **Fl Xcurs C** | **0.0 to 230.0 ms** |
| **FlStatDlyF** | **-127 to 127 samp** | **Fl Xcurs F** | **-127 to 127 samp** |
| | | **Fl Delay C** | **0.0 to 230.0 ms** |
| | | **Fl Delay F** | **-127 to 127 samp** |

In/Out — When set to "In", the quantizer and flanger are active; when set to "Out", the quantizer and flanger are bypassed.

Out Gain — The overall gain or amplitude at the output of the effect.

Quant W/D — The relative amount of quantized (wet) to unaffected (dry) signal passed to the flanger. At 100%, you hear only quantized signal pass to the flanger.

Flange W/D — The relative amount of input signal (from the quantizer) and flanger signal that is to appear in the final effect output mix. When set to 0%, the output is taken only from the quantizer (dry). When set to 100%, the output is all wet. Negative values polarity-invert the wet signal.

DynamRange — The digital dynamic range controls signal quantization, or how many bits to remove from the signal data words. At 0 dB the hottest of signals will toggle between only two bit (or quantization) levels. Every 6 dB added doubles the number of quantization levels. If the signal has a lot of headroom (available signal level before digital clipping), then not all quantization levels will be reached.

Headroom — When the signal has a lot of headroom (available signal level before digital clipping), turning down DynamRange can cause the amplitude of adjacent quantization levels to exceed the input signal level. This causes the output to get very loud. Set Headroom to match the amount of digital signal level still available (headroom). This is easily done by finding the DynamRange level at which the signal starts getting louder and matching Headroom to that value.

dc Offset — Adds a positive dc Offset to the input signal. By adding dc Offset, you can alter the position where digital zero is with respect to your signal. At low DynamRange settings, adding dc Offset can make the output "sputter". dc Offset is expressed in decibels (dB) relative to full scale digital.

Fl Tempo — Basis for the rates of the LFOs, as referenced to a musical tempo in bpm (beats per minute). When this parameter is set to "System", the tempo is locked to the internal sequencer tempo or to incoming MIDI clocks. When it is set to "System", sources (FUNs, LFOs, ASRs etc.) will have no effect on the Tempo parameter.

Fl Period — Sets the LFO rate based on the Tempo determined above: the number of beats corresponding to one period of the LFO cycle. For example, if the Fl Period is set to "4", the LFOs will take four beats to pass through one oscillation, so the LFO rate will be 1/4th of the Tempo setting. If it is set to "6/24" (=1/4), the LFO will oscillate four times as fast as the Tempo. At "0", the LFOs stop oscillating and their phase is undetermined (wherever they stopped).

Fl Fdbk — The level of the flanger feedback signal into the flanger delay line. The feedback signal is taken from the LFO delay tap. Negative values polarity-invert the feedback signal.

| | |
|---|---|
| Fl L/R Phase | The phase angles of the left and right LFOs relative to each other and to the system tempo clock, if turned on (see Fl Tempo). In all other respects the right and left channels are symmetric. For example, if one LFO is set to 0° and another is set to 180°, then when one LFO delay tap is at its shortest, the other will be at its longest. If the system tempo clock is on, the LFOs are synchronized to the clock with absolute phase. A phase of 0° will put an LFO tap at the center of its range and its lengthening. Using different phase angles for left and right, the stereo sound field is broken up and a stereo image becomes difficult to spatially locate. The effect is usually described as "phasey". It tends to impart a greater sense of motion. |
| Fl StatLvl | The level of the flanger static delay tap. Negative values polarity-invert the signal. Setting the tap level to 0% turns off the delay tap. |
| Fl LFO Lvl | The level of the flanger LFO modulated delay tap. Negative values polarity-invert the signal. Setting the tap level to 0% turns off the delay tap. |
| FlStatDlyC | The nominal length of the flanger static delay tap from the delay input. The name suggests the tap is stationary, but it can be connected to a control source such as a data slider, a ribbon, or a V.A.S.T. function to smoothly vary the delay length. The range for all delays and excursions is 0 to 230 ms, but for flanging the range 0 to 5 ms is most effective. |
| FlStatDlyF | A fine adjustment to the flanger static delay tap length. The resolution is one sample. |
| Fl Xcurs C | The flanger LFO excursion controls set how far the LFO modulated delay taps can move from the center of their ranges. The total range of the LFO sweep is twice the excursion. If the excursion is set to 0, the LFO does not move and the tap behaves like a simple delay line set to the minimum delay. The range for all delays and excursions is 0 to 230 ms, but for flanging the range 0 to 5 ms is most effective. This parameter is a coarse adjustment for the excursion. |
| Fl Xcurs F | A fine adjustment for the flanger LFO excursions. The resolution is one sample. |
| Fl Delay C | The minimum delay for the flanger LFO modulated delay taps. The maximum delay will be the minimum plus twice the excursion. The range for all delays and excursions is 0 to 230 ms, but for flanging the range 0 to 5 ms is most effective. This parameter is a coarse adjustment for the delay. |
| Fl Delay F | A fine adjustment to the minimum flanger delay tap lengths. The resolution is one sample. |

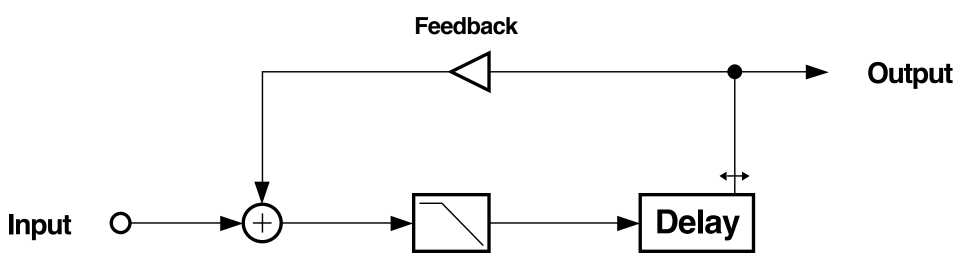
# FXAlg #715: Dual MovDelay • FXAlg #716: Quad MovDelay

### Generic dual-mono moving delay lines

*Allocation Units:* Dual MovDelay 1; Quad MovDelay 2

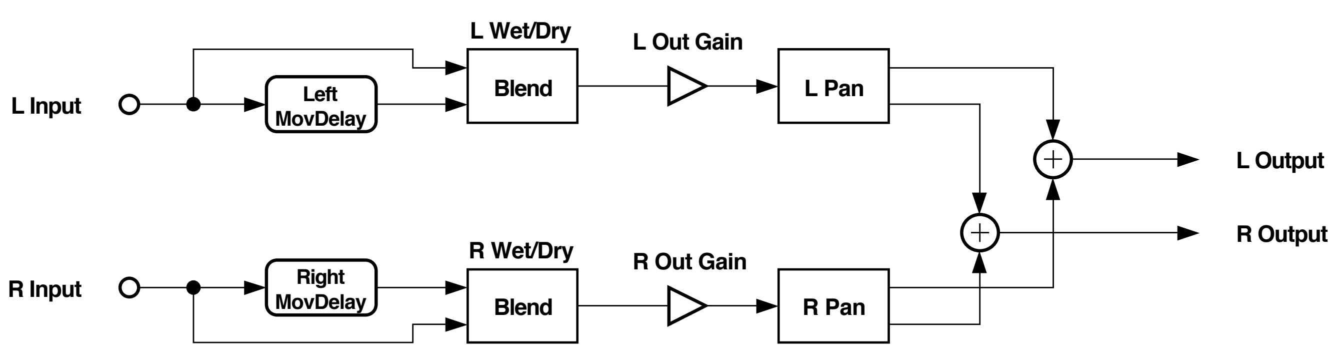
Each of these two algorithms offers generic monaural moving delay lines in a dual mono algorithm. Each separate moving delay can be used as a flanger, chorus, or static delay line selectable by the LFO Mode parameter. Both flavors of chorus pitch envelopes are offered: ChorTri for triangle, and ChorTrap for trapzoidal pitch shifting. Refer to the Chorus section of this book (FXAlgs #150-153) for more information on these envelope shapes.

Each moving delay offers control over center delay length, LFO excursion, LFO rate, feedback, and high frequency damping. The delay length, in milliseconds, is the center of LFO excursion. LFO excursion is controlled by the LFO Dpth parameter in percent. LFO Depth is an arbitrary value, and is the percentage of available excursion. When using LFO Mode Flange, this adjusts the range that the LFO will move the delay tap. When in LFO Mode ChorTri or ChorTrap, this controls the maximum pitch depth caused by the moving delay tap, and is constant regardless of LFO Rate.



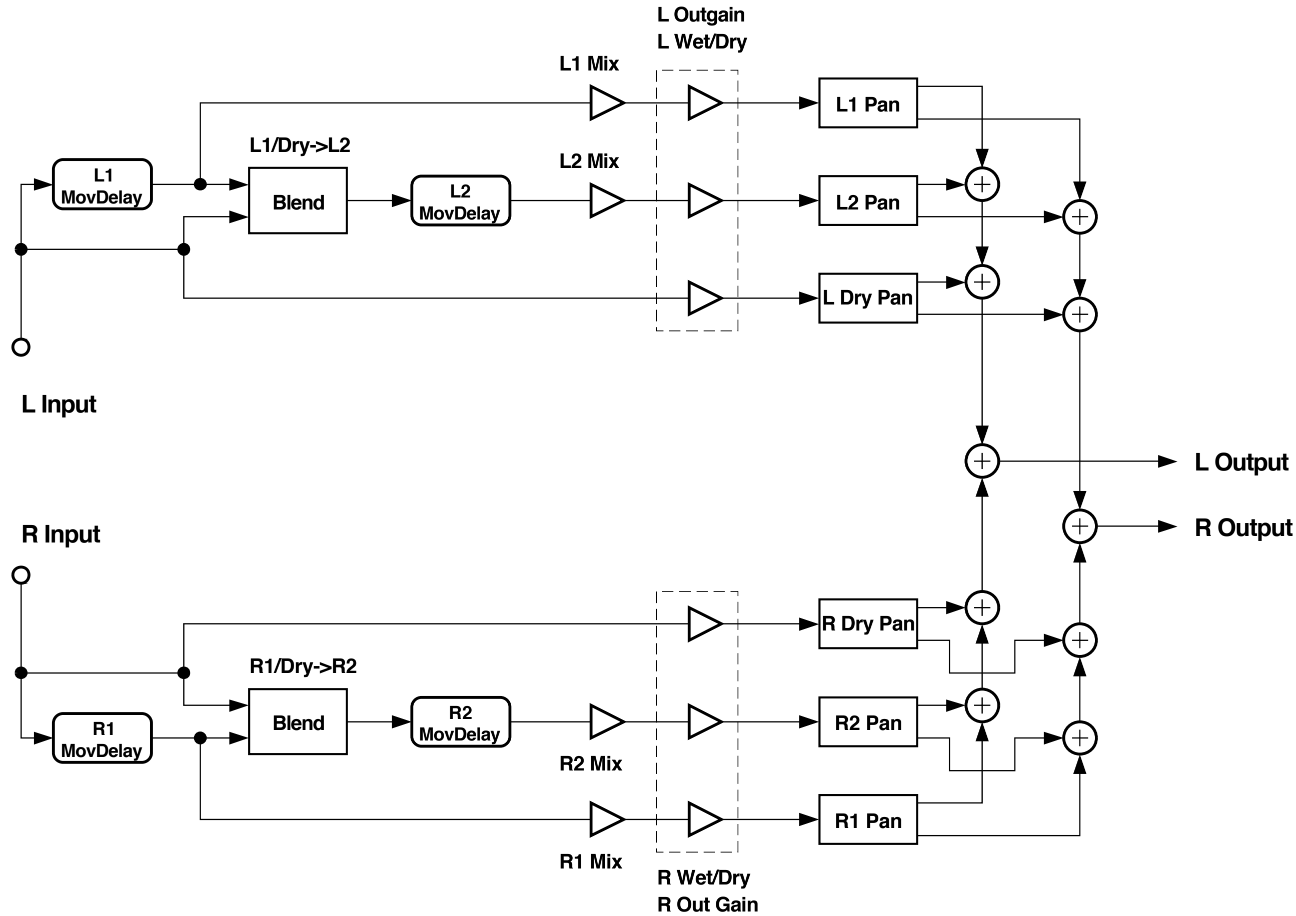

**Generic monaural moving delay line**

Both of these algorithms are configured with dual mono control meaning the left and right channels are set up to be completely independent of each other. In Dual MovDelay, each channel has a single moving delay segment. Parameters beginning with "L" and "R" control the left and right input channels respectively.




**Signal flow of Dual MovDelay**

In Quad MovDelay, there are 2 moving delay elements per channel distinguishable by parameters beginning with "L1", "L2", "R1", and "R2". The second moving delay on each channel is fed with a mix of the first delays and the input dry signal for that particular channel. These mixes are controlled by L1/Dry->L2 and R1/Dry->R2. Each of the four moving delays have separate Mix and Pan levels. The input dry signal for each channel can also be panned. The Wet/Dry parameter controls the ratio between the sum of both moving delay elements on that channel regardless of pan position, and the input dry signal. Out Gain, like Wet/Dry, adjusts the output level for each channel regardless of pan position.




**Signal flow of Quad MovDelay**

## Parameters (Dual MovDelay):

### PAGE 1

| | | | |
|---|---|---|---|
| L Wet/Dry | 0 to 100%wet | R Wet/Dry | 0 to 100%wet |
| L Out Gain | Off; -79.0 to 24.0 dB | R Out Gain | Off; -79.0 to 24.0 dB |
| L Pan | -100 to 100% | R Pan | -100 to 100% |

**PAGE 2**

| | | | |
|---|---|---|---|
| L Delay | 0.0 to 1000.0 ms | R Delay | 0.0 to 1000.0 ms |
| L LFO Mode | Flange, ... | R LFO Mode | Flange, ... |
| L LFO Rate | 0.00 to 10.00 Hz | R LFO Rate | 0.00 to 10.00 Hz |
| L LFO Dpth | 0.0 to 200.0% | R LFO Dpth | 0.0 to 200.0% |
| L Feedback | -100 to 100% | R Feedback | -100 to 100% |
| L HF Damp | 16 to 25088 Hz | R HF Damp | 16 to 25088 Hz |

## Parameters (Quad MovDelay):

**PAGE 1**

| | | | |
|---|---|---|---|
| L Wet/Dry | -100 to 100%wet | R Wet/Dry | -100 to 100%wet |
| L Out Gain | Off; -79.0 to 24.0 dB | R Out Gain | Off; -79.0 to 24.0 dB |
| L1 Mix | -100 to 100% | R1 Mix | -100 to 100% |
| L2 Mix | -100 to 100% | R2 Mix | -100 to 100% |

**PAGE 2**

| | | | |
|---|---|---|---|
| L1 Pan | -100 to 100% | R1 Pan | -100 to 100% |
| L2 Pan | -100 to 100% | R2 Pan | -100 to 100% |
| L Dry Pan | -100 to 100% | L Dry Pan | -100 to 100% |
| L1/Dry->L2 | 0 to 100% | L1/Dry->L2 | 0 to 100% |

**PAGE 3**

| | | | |
|---|---|---|---|
| L1 Delay | 0.0 to 1000.0 ms | L2 Delay | 0.0 to 1000.0 ms |
| L1 LFO Mode | Flange, ... | L2 LFO Mode | Flange, ... |
| L1 LFO Rate | 0.00 to 10.00 Hz | L2 LFO Rate | 0.00 to 10.00 Hz |
| L1 LFO Dpth | 0.0 to 200.0% | L2 LFO Dpth | 0.0 to 200.0% |
| L1 Feedback | -100 to 100% | L2 Feedback | -100 to 100% |
| L1 HF Damp | 16 to 25088 Hz | L2 HF Damp | 16 to 25088 Hz |

**PAGE 4**

| | | | |
|---|---|---|---|
| R1 Delay | 0.0 to 1000.0 ms | R2 Delay | 0.0 to 1000.0 ms |
| R1 LFO Mode | Flange, ... | R2 LFO Mode | Flange, ... |
| R1 LFO Rate | 0.00 to 10.00 Hz | R2 LFO Rate | 0.00 to 10.00 Hz |
| R1 LFO Dpth | 0.0 to 200.0% | R2 LFO Dpth | 0.0 to 200.0% |
| R1 Feedback | -100 to 100% | R2 Feedback | -100 to 100% |
| R1 HF Damp | 16 to 25088 Hz | R2 HF Damp | 16 to 25088 Hz |

<u>L Wet/Dry, R Wet/Dry</u> The relative amount of input signal and effected signal that is to appear in the final effect output mix for each input channel. When set to 0%, the output is taken only from the corresponding input (dry) signal. When set to 100%, the output is all wet. Negative values polarity-invert the wet signal.

<u>L Out Gain, R Out Gain</u> The overall gain or amplitude at the output of the effect for each input channel.

Ln Mix, Rn Mix — Adjusts the mix levels for each moving delay circuit. The resulting sum makes up the wet signal. Negative values polarity-invert the signal.

L Pan, R Pan, Ln Pan, Rn Pan — The output panning position of each moving delay circuit. 0% is center; Negative values pan left, while positive values pan right.

L Dry Pan, R Dry Pan — Adjusts the output pan position of the input dry signals. The dry level is controlled with Wet/Dry. 0% pans to center; Negative values pan left while positive values pan right.

L1/Dry->L2, R1/Dry->R2 — Adjusts the input mix into the second pair of moving delay circuits in Quad MovDelay. The value represents a ratio of the output of the first moving delay circuit and the input dry signal. A value of 0% allows only the input dry signal to be fed into the second delay, while a value of 100% only allows the first delay to be fed into the second.

L Delay, R Delay, Ln Delay, Rn Delay — Adjusts the delay time for each moving delay circuit, which is the center of LFO excursion.

L LFO Mode, R LFO Mode, Ln LFO Mode, Rn LFO Mode — Adjusts the LFO excursion type. In Flange mode, the LFO is optimized for flange effects and LFO Dpth adjusts the excursion amount. In ChorTri and ChorTrap modes, the LFO is optimized for triangle and trapzoidal pitch envelopes respectively, and LFO Dpth adjusts the amount of chorus detuning. In Delay mode, the LFO is turned off leaving a basic delay. LFO Rate and LFO Dpth in Delay mode are disabled.

L LFO Rate, R LFO Rate, Ln LFO Rate, Rn LFO Rate — Adjusts the LFO speed for each moving delay circuit.

L LFO Dpth, R LFO Dpth, Ln LFO Dpth, Rn LFO Dpth — In Flange LFO mode, this adjusts an arbitrary LFO excursion amount. In ChorTri and ChorTrap modes, this controls the chorus detune amount. In delay mode, this is disabled.

L Feedback, R Feedback, Ln Fdbk, Rn Fdbk — Adjusts the level of each moving delay circuits output signal fed back into their own inputs. Negative values polarity-invert the feedback signal.

L HF Damp, R HF Damp, Ln HF Damp, Rn HF Damp — Adjusts the cutoff frequency of a 1-pole (6dB/oct) lowpass filter in each moving delay circuit.

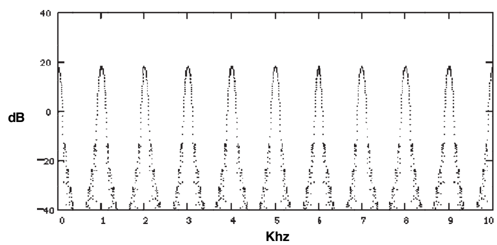
# FXAlg #720: MonoPitcher+Chor • FXAlg #721: MonoPitcher+Flan

### Mono pitcher (filter with harmonically related resonant peaks) algorithm with a chorus or flanger

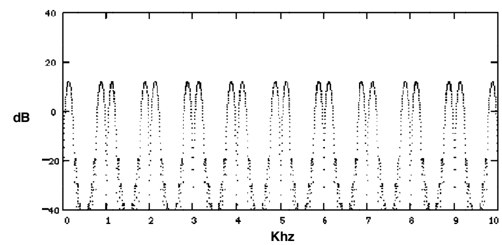
*Allocation Units:* 2 (each)

The mono pitcher algorithm applies a filter which has a series of peaks in the frequency response to the input signal. The peaks may be adjusted so that their frequencies are all multiples of a selectable frequency, all the way up to 24 kHz. When applied to a sound with a noise-like spectrum (white noise, with a flat spectrum, or cymbals, with a very dense spectrum of many individual components), an output is produced which sounds very pitched, since most of its spectral energy ends up concentrated around multiples of a fundamental frequency.

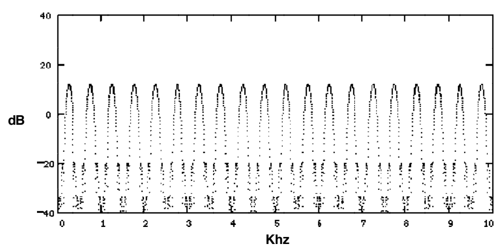
The following graphs show Pt PkSplit going from 0% to 100%, for a Pt Pitch of 1 kHz (approx. C6), and Pt PkShape set to 0.



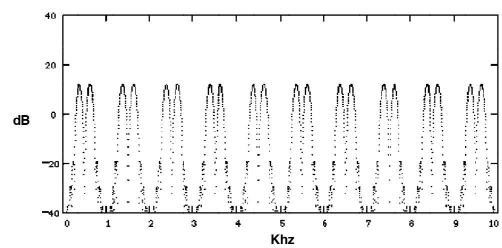

**PeakShape = 0**
**PeakSplit = 0%**



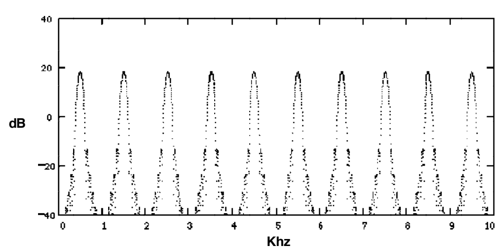

**PeakShape = 0**
**PeakSplit = 25%**




**PeakShape = 0**
**PeakSplit = 50%**




**PeakShape = 0**
**peakSplit = 75%**

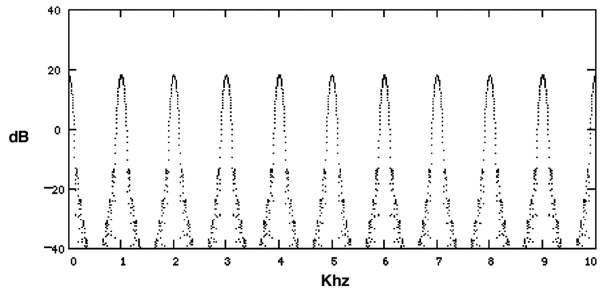



**PeakShape = 0**
**PeakSplit = 100%**

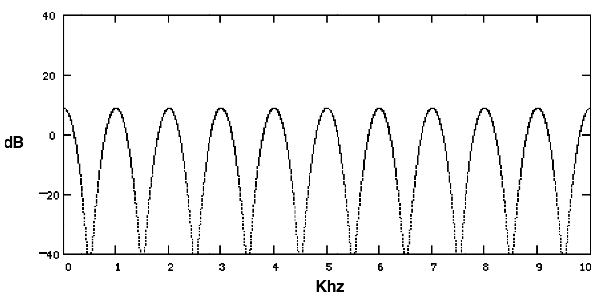
**Response of Pitcher with different PkSplit settings.**

Note that a Pt PkSplit of 100% gives only odd multiples of a fundamental that is one octave down from no splitting. The presence of only odd multiples will produce a hollow sort of sound, like a square wave (which also only has odd harmonics). Curiously enough, at a Pt PkSplit of 50% we also get odd multiples of a frequency that is now two octaves below the original Pitch parameter. In general, most values of PkSplit will give peak positions that are not harmonically related.

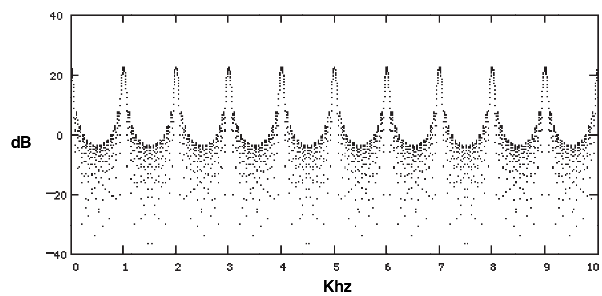
The figures below show Pt PkShape of -1.0, 0.0, and 1.0, for a Pitch of C6 and a PkSplit of 0%.




**PeakShape = 0**
**PeakSplit = 0%**




**PeakShape = 1.0**
**PeakSplit = 0%**




**PeakShape = -1.0**
**PeakSplit = 0%**

**Response of Pitcher with PkShape settings at 0, -1.0, and 1.0.**

Applying Pitcher to sounds such as a single sawtooth wave will tend to not produce much output, unless the sawtooth frequency and the Pitcher frequency match or are harmonically related, because otherwise the peaks in the input spectrum won't line up with the peaks in the Pitcher filter. If there are enough peaks in the input spectrum (obtained by using sounds with noise components, or combining lots of different simple sounds, especially low pitched ones, or severely distorting a simple sound) then Pitcher can do a good job of imposing its pitch on the sound.

Multiple Pitcher algorithms can be run (yes, it takes all of KDFX to get three) to produce chordal output.

At extremely low Pitch settings, the effect begins to sound more like a multi-tap delay, but this can be pretty cool, too.

A vocoder-like effect can be produced, although in some sense it works in exactly an opposite way to a real vocoder. A real vocoder will superimpose the spectrum of one signal (typically speech) onto a musical signal (which has only a small number of harmonically related spectral peaks). Pitcher takes an input such as speech, and then picks out only the components that match a harmonic series, as though they were from a musical note.

## Chorus:

The chorus used in FXAlg #720 is a basic dual-channel chorus. Refer to Chorus documentation (FXAlgs #150-153) in this book for more information on the effect.

## Configurable Flange:

The flange in FXAlg #721 is a configurable flange. Refer to the section on Flanger (FXAlg #702 and FXAlgs #154-155) in this book for details about this effect.

## Parameters (MonoPitcher+Chor):

PAGE 1

| | | | |
|---|---|---|---|
| Wet/Dry | 100 to 100%wet | Out Gain | Off, -79.0 to 24.0 dB |
| Mix Pitchr | -100 to 100% | | |
| Mix Chorus | -100 to 100% | | |
| Pt/Dry->Ch | 0 to 100% | | |

PAGE 3

| | | | |
|---|---|---|---|
| Pt Inp Bal | -100 to 100% | Pt Out Pan | -100 to 100% |
| Pt Pitch | C-1 to G 9 | Pt Offset | -12.0 to 12.0 ST |
| Pt PkSplit | 0 to 100% | Pt PkShape | -1.0 to 1.0 |

| | | | |
|---|---|---|---|
| ChPtchEnvL | Triangle or trapzoid | ChPtchEnvL | Triangle or trapzoid |
| Ch Rate L | 0.01 to 10.00 Hz | Ch Rate R | 0.01 to 10.00 Hz |
| Ch Depth L | 0.0 to 100.0 ct | Ch Depth R | 0.0 to 100.0 ct |
| Ch Delay L | 0.0 to 720.0 ms | Ch Delay R | 0.0 to 720.0 ms |
| Ch Fdbk L | -100 to 100% | Ch Fdbk R | -100 to 100% |
| Ch Xcouple | 0 to 100% | Ch HF Damp | 16 to 25088 Hz |

## Parameters (MonoPitcher+Flan):

PAGE 1

| | | | |
|---|---|---|---|
| Wet/Dry | 100 to 100%wet | Out Gain | Off, -79.0 to 24.0 dB |
| Mix Pitchr | -100 to 100% | | |
| Mix Flange | -100 to 100% | Fl Tempo | System, 1 to 255 BPM |
| Pt/Dry->Fl | 0 to 100% | | |

PAGE 3

| | | | |
|---|---|---|---|
| Pt Inp Bal | -100 to 100% | Pt Out Pan | -100 to 100% |
| Pt Pitch | C-1 to G 9 | Pt Offset | -12.0 to 12.0 ST |
| Pt PkSplit | 0 to 100% | Pt PkShape | -1.0 to 1.0 |

| | | | |
|---|---|---|---|
| Fl LFO cfg | Dual1Tap | Fl LRPhase | 0.0 to 360.0 deg |
| Fl Rate 1 | 0 to 32 bts | Fl Rate 2 | 0 to 32 bts |

| | | | |
|---|---|---|---|
| **Fl Xcurs 1** | **0.0 to 230.0 bts** | **Fl Xcurs 2** | **0.0 to 230.0 bts** |
| **Fl Delay 1** | **0.0 to 230.0 ms** | **Fl Delay 2** | **0.0 to 230.0 ms** |
| **Fl Phase 1** | **0.0 to 360.0 deg** | **Fl Phase 2** | **0.0 to 360.0 deg** |
| **Fl Fdbk** | **-100 to 100%** | **Fl HF Damp** | **16 to 25088 Hz** |

<u>Wet/Dry</u> This is a simple mix of the pitched and chorused or flanged signal relative to the dry input signal.

<u>Out Gain</u> The overall gain or amplitude at the output of the effect.

<u>Mix Pitchr</u> The amount of the pitcher signal to be sent directly to the output as a percent. Any signal that this parameter sends to the output does not get sent to the chorus or flanger.

<u>Mix Chorus</u>, <u>Mix Flange</u> The amount of the flanger or chorus signal to send to the output as a percent.

<u>Pt/Dry->Ch</u>, <u>Pt/Dry->Fl</u> The relative amount of pitcher signal to dry signal to send to the chorus or flanger. At 0% the dry input signal is routed to the chorus or flanger. At 100%, the chorus or flanger receives its input entirely from the pitcher.

<u>Pt Inp Bal</u> Since this is a mono algorithm, an input balance control is provided to mix the left and right inputs to the pitcher. -100% is left only, 0% is left plus right, and 100% is right only.

<u>Pt Out Pan</u> Pans the mono pitcher output from left (-100%) to center (0%) to right (100%)

<u>Pt Pitch</u> The "fundamental" frequency of the Pitcher output. This sets the frequency of the lowest peak in terms of standard note names. All the other peaks will be at multiples of this pitch.

<u>Pt PkSplit</u> Splits the pitcher peaks into two peaks, which both move away from their original unsplit position, one going up and the other down in frequency. At 0% there is no splitting; all peaks are at multiples of the fundamental. At 100% the peak going up merges with the peak going down from the next higher position.

<u>Pt Offset</u> An offset in semitones from the frequency specified in Pitch.

<u>Pt PkShape</u> Controls the shape of the pitcher spectral peaks. 0.0 gives the most "pitchiness" to the output, in that the peaks are narrow, and there is not much energy between them. -1.0 makes the peaks wider. 1.0 brings up the level between the peaks.

<u>Ch PtchEnv</u>, <u>Ch Rate</u>, <u>Ch Depth</u>, <u>Ch Delay</u>, <u>Ch Fdbk</u>, <u>Ch Xcouple</u>, <u>Ch HF Damp</u> Refer to Chorus documentation.

<u>Fl LFO cfg</u> Sets the user interface mode for controlling each of the 4 flange LFOs.

<u>Fl LRPhase</u> Controls the relative phase between left channel LFOs and right channel LFOs. In Dual1Tap mode, however, this parameter is accurate only when Fl Rate 1 and Fl Rate 2 are set to the same speed, and only after the Fl LFO cfg parameter is moved, or the algorithm is called up.

<u>Fl Phase 1</u>, <u>Fl Phase 2</u> These adjust the corresponding LFO phase relationships between themselves and the internal beat clock.

<u>Fl Rate</u>, <u>Fl Depth</u>, <u>Fl Delay</u>, <u>Fl Fdbk</u>, <u>Fl Xcouple</u>, <u>Fl HF Damp</u> Refer to Flanger documentation. Parameters with a 1 or 2 correspond to LFO taps organized as described above.