Rumour and Mangler

Algorithm Reference

Part Number: 910394

CAUTION

RISK OF ELECTRIC SHOCK
DO NOT OPEN

CAUTION: TO REDUCE THE RISK OF ELECTRIC SHOCK, DO NOT REMOVE THE COVER. NO USER-SERVICEABLE PARTS INSIDE. REFER SERVICING TO QUALIFIED SERVICE PERSONNEL.

THIS PRODUCT IS INTENDED FOR INDOOR USE ONLY.

The exclamation point within an equilateral triangle is intended to alert the user to the presence of important operating and maintenance (servicing) instructions in the literature accompanying the product.

The symbol of a house with an arrow pointing inside is intended to alert the user that the product is to be used indoors only.

The lightning flash/arrowhead symbol within an equilateral triangle is intended to alert the user to the presence of uninsulated "dangerous voltage" within the product's enclosure, which may be of sufficient magnitude to constitute a risk of electric shock to persons.

IMPORTANT SAFETY & INSTALLATION INSTRUCTIONS

INSTRUCTIONS PERTAINING TO THE RISK OF FIRE, ELECTRIC SHOCK, OR INJURY TO PERSONS

WARNING: When using electric products, basic precautions should always be followed, including the following:

1. Read all of the Safety and Installation Instructions and Explanation of Graphic Symbols before using the product.
2. Do not use this product near water—for example, near a bathtub, washbowl, kitchen sink, in a wet basement, or near a swimming pool, or the like.
3. This product should be used only with a stand or cart that is recommended by the manufacturer.
4. This product, either alone or in combination with an amplifier and speakers or headphones, may be capable of producing sound levels that could cause permanent hearing loss. Do not operate for a long period of time at a high volume level or at a level that is uncomfortable. If you experience any hearing loss or ringing in the ears, you should consult an audiologist.
5. The product should be located so that its location or position does not interfere with its proper ventilation.
6. The product should be located away from heat sources such as radiators, heat registers, or other products that produce heat.
7. The product should be connected to a power supply only of the type described in the operating instructions or as marked on the product.
8. This product may be equipped with a polarized line plug (one blade wider than the other). This is a safety feature. If you are unable to insert the plug into the outlet, contact an electrician to replace your obsolete outlet. Do not defeat the safety purpose of the plug.
9. The power supply cord of the product should be unplugged from the outlet when left unused for a long period of time. When unplugging the power supply cord, do not pull on the cord, but grasp it by the plug.
10. Care should be taken so that objects do not fall and liquids are not spilled into the enclosure through openings.
11. The product should be serviced by qualified service personnel when:
    A. The power supply cord or the plug has been damaged;
    B. Objects have fallen onto, or liquid has been spilled into the product;
    C. The product has been exposed to rain;
    D. The product does not appear to be operating normally or exhibits a marked change in performance;
    E. The product has been dropped, or the enclosure damaged.
12. Do not attempt to service the product beyond that described in the user maintenance instructions. All other servicing should be referred to qualified service personnel.
13. **WARNING:** Do not place objects on the product's power supply cord, or place the product in a position where anyone could trip over, walk on, or roll anything over cords of any type. Do not allow the product to rest on or be installed over cords of any type. Improper installations of this type create the possibility of a fire hazard and/or personal injury.

RADIO AND TELEVISION INTERFERENCE

WARNING: Changes or modifications to this instrument not expressly approved by Young Chang could void your authority to operate the instrument.

IMPORTANT: When connecting this product to accessories and/or other equipment use only high quality shielded cables.

NOTE: This instrument has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the instrument is used in a commercial environment. This instrument generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this instrument in a residential area is likely to cause harmful interference, in which case the user will be required to correct the interference at his or her own expense.

Changes and modifications not expressly approved by the manufacturer or registrant of this instrument can void the user's authority to operate this instrument under Federal Communications Commission rules.

In order to maintain compliance with FCC regulations, shielded cables must be used with this instrument. Operation with unapproved equipment or unshielded cables is likely to result in harmful interference to radio and television reception.

NOTICE

This apparatus does not exceed the Class A limits for radio noise emissions from digital apparatus set out in the Radio Interference Regulations of the Canadian Department of Communications.

AVIS

Le present appareil numerique n'emet pas de bruits radioelectriques depassant les limites applicables aux appareils numeriques de la class A prescrites dans le Reglement sur le brouillage radioelectrique edicte par le ministere des Communications du Canada.

SAVE THESE INSTRUCTIONS

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4-Tap Delay BPM	150	1	64
5 Band EQ	351	3	222
8-Tap Delay	153	2	68
8-Tap Delay BPM	152	2	68
Allpass Phaser 3	256	3	117
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AutoPanner BPM	279	1	126
Band Compress	335	3	201
Barberpole Comb	258	4	120
Big KB3 Effect	298	8	139
Cabinet	284	3	138
Cabinet+Dly+Rvrb	285	3	173
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Chor+Dly+Rvrb ms	423	2	299
Chor<>LasrDly ms	425	2	299
Chorus+4Tap	401	1	279
Chorus+4Tap ms	421	1	299
Chorus+Delay	400	1	279
Chorus+Delay ms	420	1	299
Chorus<>4Tap	402	2	289
Chorus<>4Tap ms	422	2	299
Chorus<>LasrDly	405	2	289
Chorus<>Reverb	404	2	289
ClascVrb<>Comprs	52	3	43
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Flan<>LasrDly ms	475	2	299
Flange+4Tap	451	1	279
Flange+4Tap ms	471	1	299
Flange+Delay	450	1	279
Flange+Delay ms	470	1	299
Flange<>4Tap	452	2	289
Flange<>4Tap ms	472	2	299
Flange<>LasrDly	455	2	289
Flange<>Pitcher	384	2	265
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Gate Synth	393	3	275
Gate w/SC EQ LFX	342	2	212
Gate+Cmp<>EQ+Rvb	54	4	47
Gate+Cmp[EQ]+Rvb	53	4	47
Gate+Tube+Reverb	311	4	173
Gate+TubeAmp	310	3	173
Gated Delay	174	2	90
Gated LaserVerb	104	3	59
Gated MiniVerb	3	2	13
Grand Plate	14	3	35
Graphic EQ	352	3	225
Gt+Cmp+Dst+EQ+Ch	315	4	173
Gt+Cmp+Dst+EQ+Fl	316	4	173
Gt+Tube<>2MD	314	4	173

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Gt+Tube<>MD+Chor	312	4	173
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HardKneeCompress	330	1	191
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LFO Phaser	250	1	111
LFOPhaserTwinLFX	251	1	111
LFO Sweep Filter	362	2	233
Manual Phaser	255	1	111
MiniVerb	1	1	9
Mono -> Stereo	281	1	132
Mono Distortion	300	1	153
Mono EQ Morpher	366	2	238
Mono LaserVerb	102	1	53
MonoDistort + EQ	302	2	153
MonoDistort+Cab	301	2	153
MonoPitch+StChor	412	2	299
MonoPitcher+Chor	411	2	285
MonoPitcher+Flan	461	2	285
Moving Delay	190	1	93
MutualFreqOffset	386	2	266
OmniPlace	10	3	16
OmniVerb	11	3	16
Opto Compress	333	2	197
Opto Comprs SCEQ	334	3	197
Panaural Room	12	3	29
Pitch+StChor+Dly	410	2	299
Pitch+StFlan+Dly	460	2	299
Pitcher	381	1	256
Pitcher+Chor+Dly	409	2	279
Pitcher+Flan+Dly	459	2	279
Pitcher+MiniVerb	383	2	262
Poly Pitcher	382	2	260
PolyAmp<>MDBP>Ch	325	3	184
PolyAmp<>MDBP>Fl	326	3	184
PolyDistort + EQ	303	2	158
Ptch+Chor+Dly ms	429	2	299
Ptch+Flan+Dly ms	479	2	299
Ptch+StCh+Dly ms	430	2	299
Ptch+StFl+Dly ms	480	2	299
Quantize+Alias	308	1	165
Quantize+Flange	309	1	169
Resonant Filter	363	1	236
Reverb+Compress	50	2	39
Reverb<>Compress	51	3	39
Revrse LaserVerb	103	4	56

Name	ID	PAUs	Page
Ring Modulator	380	1	252
Rotor 1	295	1	139
Shaper<>Reverb	322	2	190
SingleLFO Phaser	253	1	111
SoftKneeCompress	331	1	191
Spectral 4-Tap	154	2	72
Spectral 6-Tap	155	3	72
St Chor+4Tap ms	427	1	299
St Chor+Delay ms	426	1	299
StChor+Dly+RvrbL	408	2	299
St Chorus+4Tap	407	1	299
St Chorus+Delay	406	1	299
St Flan+4Tap ms	477	1	299
St Flan+Delay ms	476	1	299
StFlan+Dly+RvrbL	458	2	299
St Flange+4Tap	457	1	299
St Flange+Delay	456	1	299
StCh+Dly+Rvrb ms	428	2	299
Stereo Analyze	499	1	302
Stereo Hall	13	3	32
Stereo Image	280	1	130
StereoDistort+EQ	304	3	153
StFl+Dly+Rvrb ms	478	2	299
Subtle Distort	305	1	162
Super Shaper	306	1	163
Switch Loops	172	2	85
Tone Suppressor	375	2	247
TQ Place	6	3	16
TQ Verb	7	3	16
Tremolo	271	1	123
Tremolo BPM	270	1	123
TrigEnvelopeFilt	361	2	230
Tube+Reverb	327	3	173
TubeAmp<>MDBP>Ch	323	3	184
TubeAmp<>MDBP>Fl	324	3	184
VC+Dist+1Rotor 2	293	2	139
VC+Dist+HiLoRot2	294	2	139
VC+Dist+HiLoRotr	292	2	139
VC+Dist+Rotor 4	296	4	139
VC+Tube+Rotor 4	297	4	139
VibChor+Rotor 2	290	2	139
VibratoPhaser	254	1	111
WackedPitchLFO	387	3	270

Algorithm Specifications

MiniVerbs

1 MiniVerb
2 Dual MiniVerb
600 Mn MiniVerb

Versatile, small stereo and dual mono reverbs

PAUs: 1 for **MiniVerb**
2 for **Dual MiniVerb**

MiniVerb is a versatile stereo reverb 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.

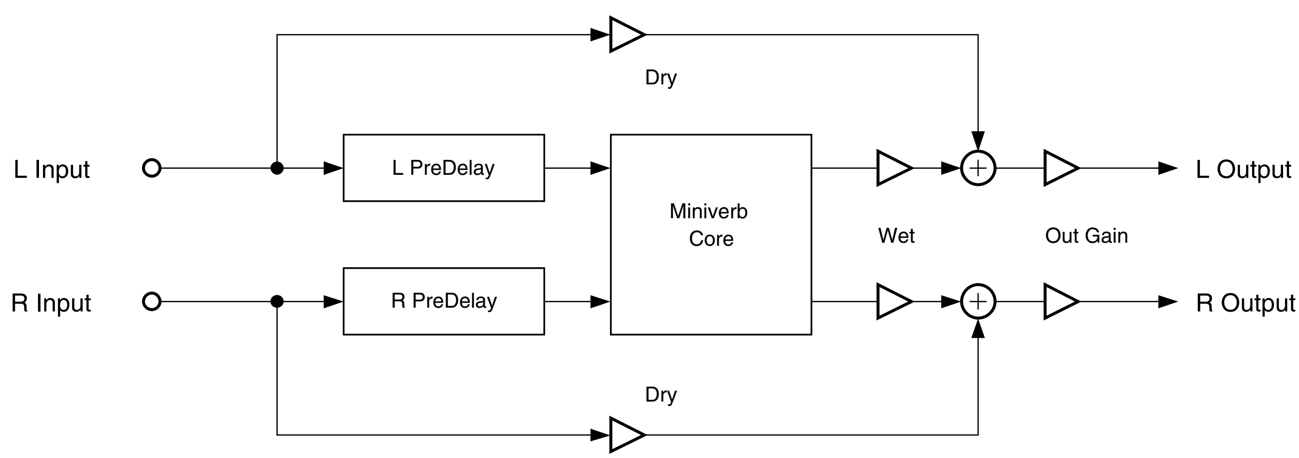

Figure 1 Simplified Block Diagram of MiniVerb

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.

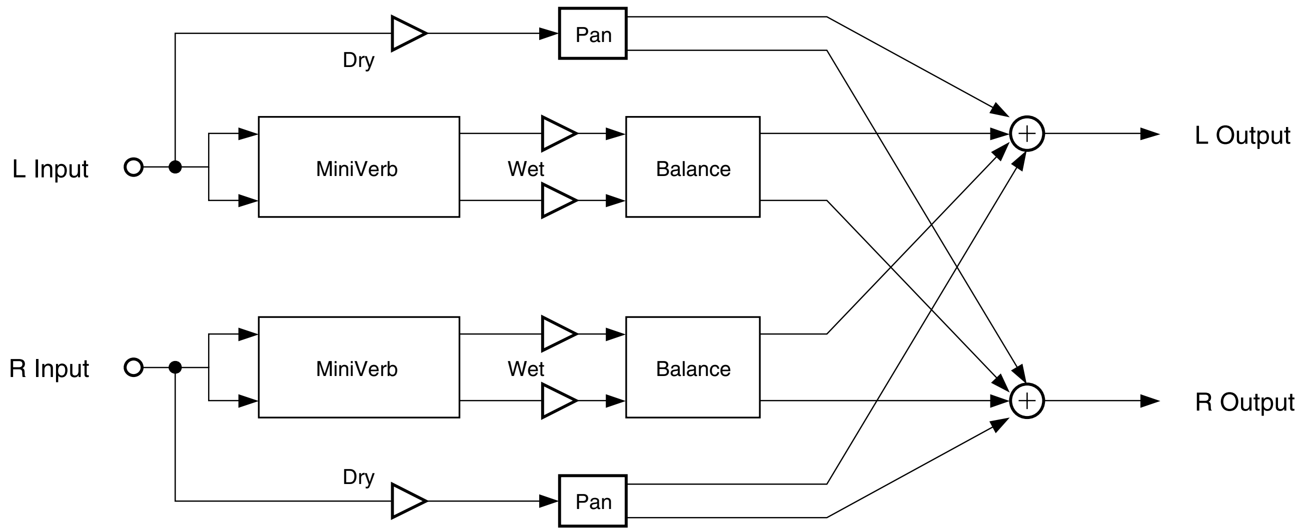

Figure 2 **Simplified Block Diagram of Dual MiniVerb**

Dual MiniVerb has a full **MiniVerb**, including Wet/Dry, Pre Delay and Out Gain controls, dedicated to both the left and right channels. In Figure 2, the two blocks labeled **MiniVerb** contain a complete copy of the contents of **Figure 1**. **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 reverberation. 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**.

MiniVerb Parameters:

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	8 to 25088 Hz
L Pre Dly	0 to 620 ms	R Pre Dly	0 to 620 ms

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

Dual MiniVerb Parameters

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	8 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	8 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 predelays 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 modulate it.)

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.

3 Gated MiniVerb

A reverb and gate in series

PAUs: 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.

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 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 reflection 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.

The gate turns the output of the reverb on and off based on the amplitude of the input signal.

A gate behaves like an on off switch for a signal. One or both input channels is used to control whether the switch is on (gate is open) or off (gate is closed). The 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 with 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.

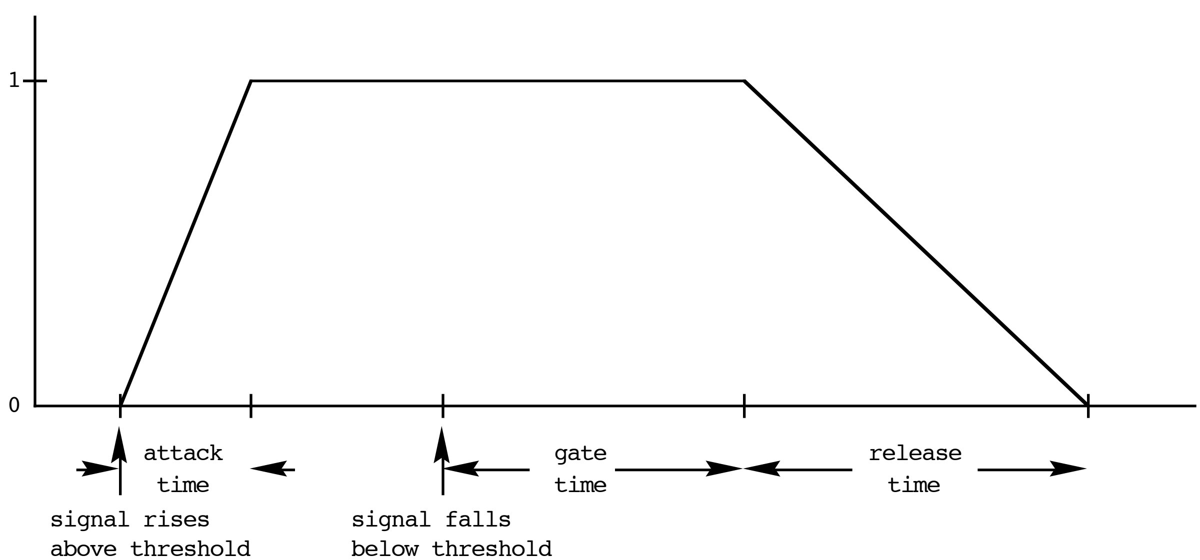

Figure 3 Gate Behavior

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 use 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	8 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
	-dB 60 40 ✻ 16	✻ 8 4 0	

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

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

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 predelays 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 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 may not modulate it.)

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.

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

Gate Duck 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.

Gate Time 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.

Gate Atk The attack time for the gate to ramp from closed to open (reverse if Gate Duck is **On**) after the signal rises above threshold.

Gate Rel The release time for the gate to ramp from open to closed (reverse if Gate Duck is **On**) after the gate timer has elapsed.

Signal Dly The delay in milliseconds (ms) of the reverb 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.

Reverbs

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

Reverb algorithms

PAUs: 2 (Classic) or 3 (others)

This set of 2- and 3-PAU 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 reverb algorithm consists of a several components: a diffuser, an injector, predelay, an ambience generator with feedback, and various filters. These components provide sonic building blocks for both the body of the reverb and the early reflection portions.

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 consists 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 flutteriness and ringiness 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 and mimics high frequency energy that is 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 change the amount and the length of time that the sound is smeared. The Diffuse reverbs, however, implement diffusion a little differently. See the sections on **Diffuse Verb and Diffuse Place** on **page 22** 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 time intervals that the copies are made. 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 that 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 use Early Reflection circuits employ 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.

Your starting point when creating a new reverb preset should be the Room Type parameter. This parameter selects the basic type of reverb being. 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 you select a room type, 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 value. When set to **1.00x**, each of these elements is equivalent to its preset value as 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. See Room Type for more detailed information.

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.

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, lopass 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.

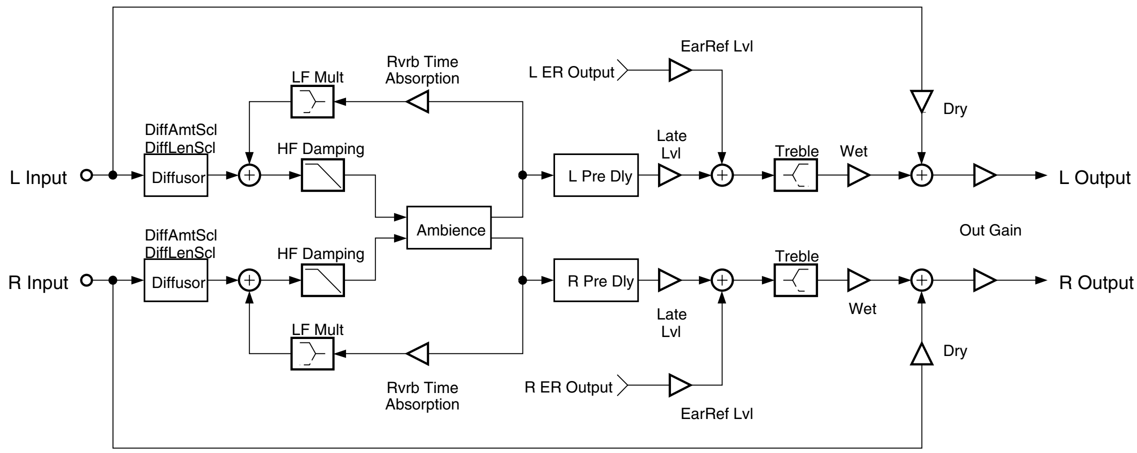

Figure 4 **Signal flow of Classic Verb and Classic Place**

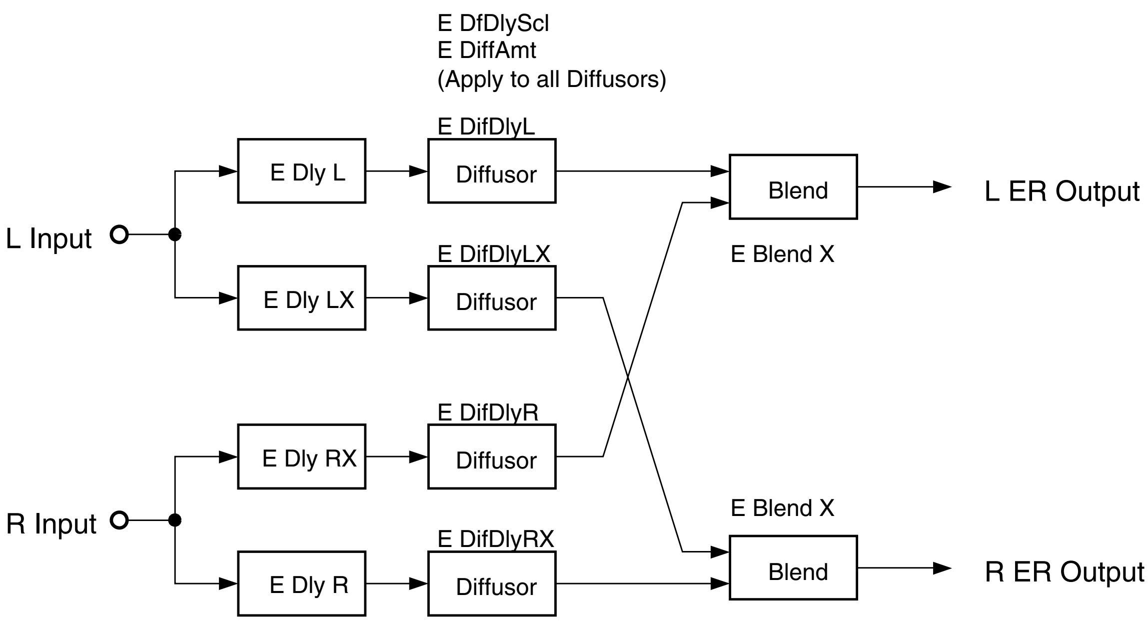

Figure 5 **Early reflection portion of Classic Verb and Classic Place**

Parameters for 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	8 to 25088 Hz	LF Split	8 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	Hall1, ...	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	8 to 25088 Hz	LF Split	8 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 lopass filter, low shelving filter, and LFO moving delays, and predelay.

The early reflection portion combines a combination of delays, diffusers, and feedback outlined by **Figure 7**. 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 amount 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 amount 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.

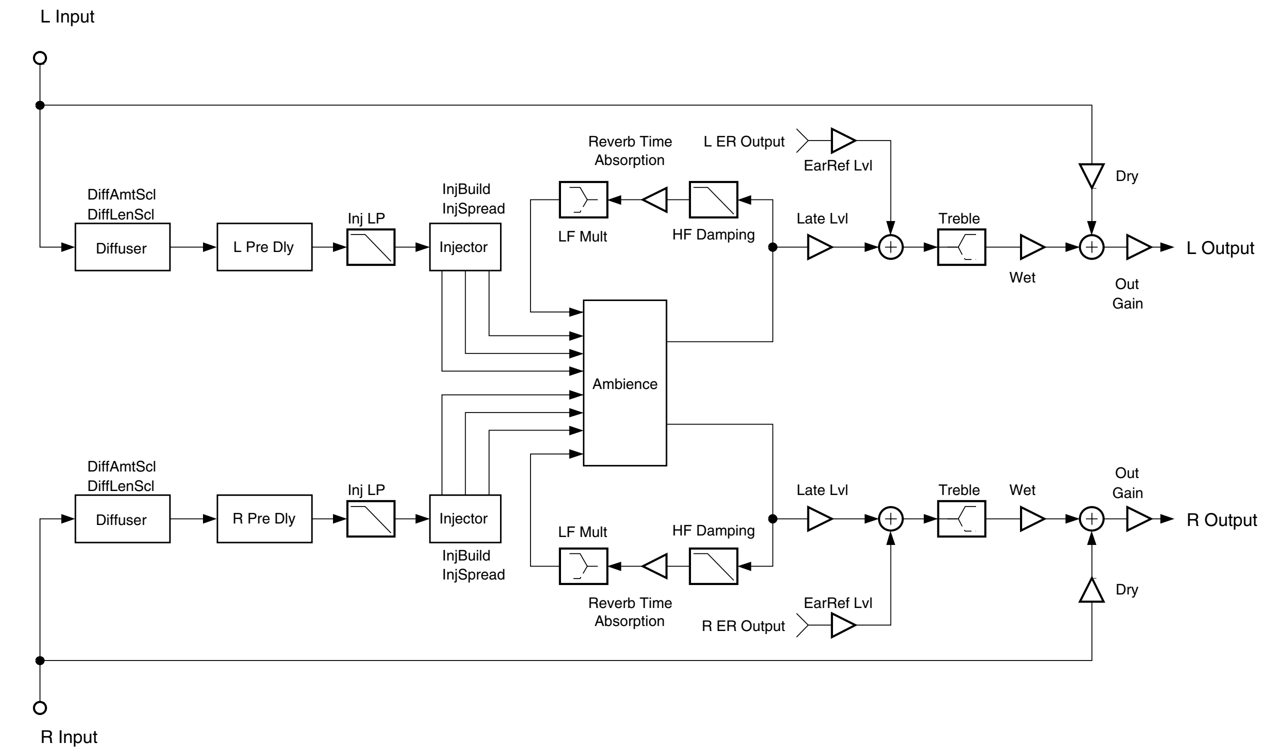

Figure 6 **Signal flow of TQ Verb and TQ Place**

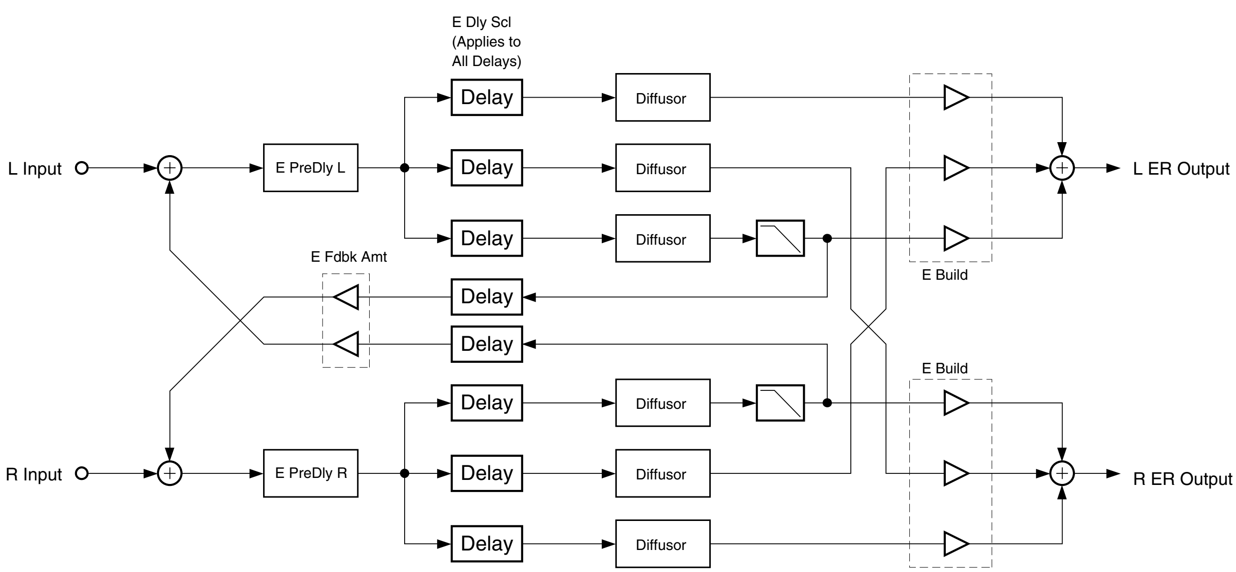

Figure 7 **Early reflection portion of TQ Verb and TQ Place**

Parameters for 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	8 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	8 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	Hall1, ...	TrebShlf F	8 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	8 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	8 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	8 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. The diffusion consists of an input diffuser, ambience generator with a lopass 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 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.

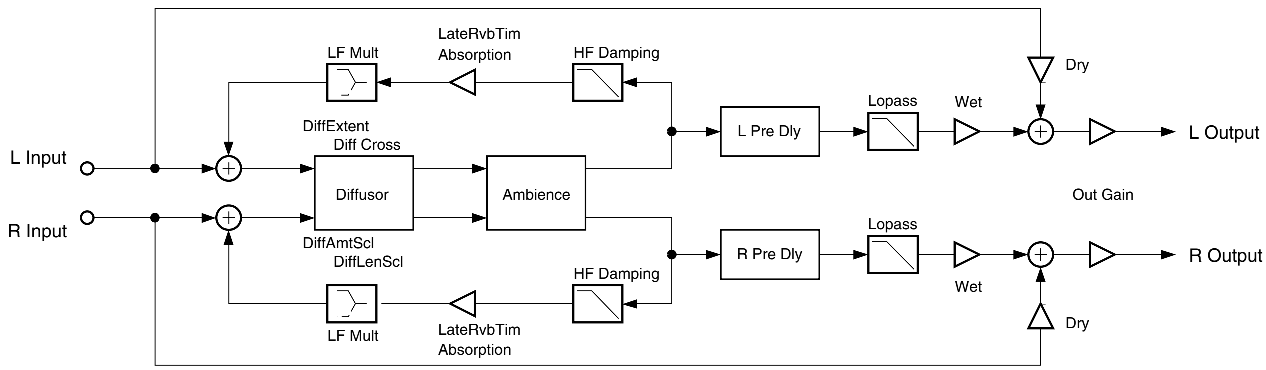

Figure 8 Signal flow of Diffuse Verb and Diffuse Place

Parameters for 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	8 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	8 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	8 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	Hall1, ...	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	8 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 consists of an input diffuser, injector, ambience generator with a lopass 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**% concentrates energy in the center of the image, while non-zero values spread it out. Positive and negative values impose different characteristics on the reverb image.

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

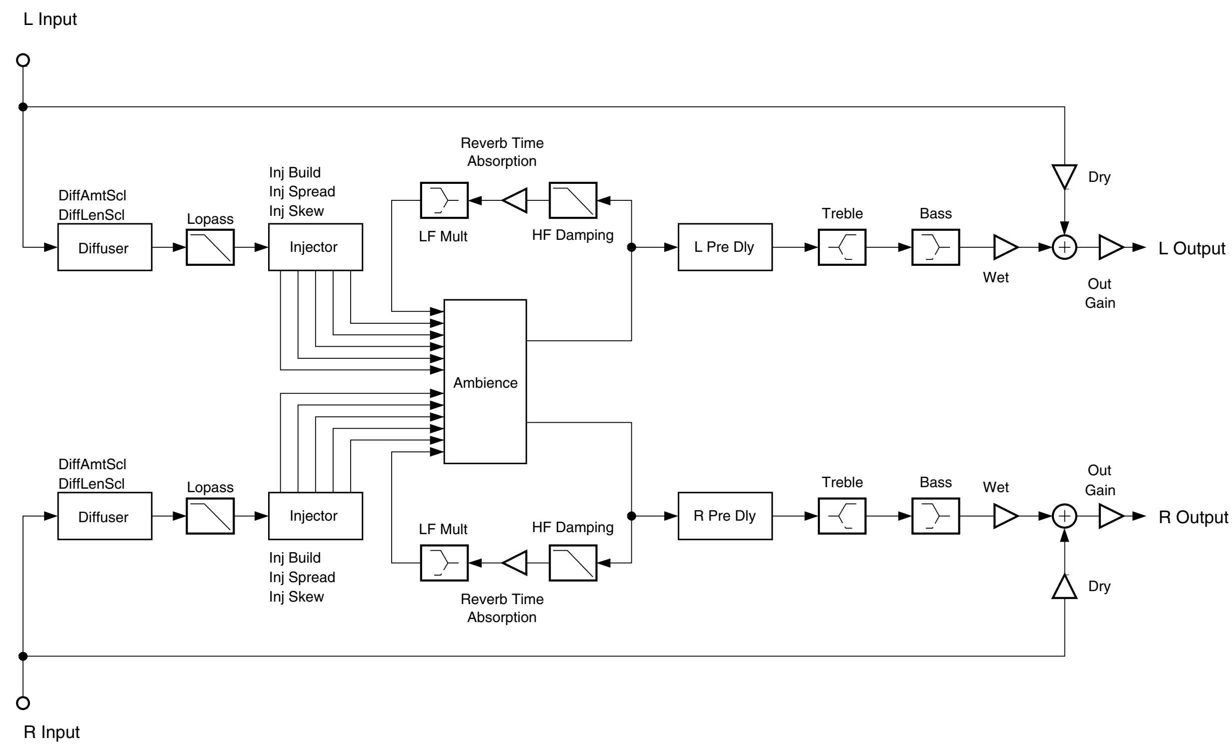

Figure 9 Signal flow of OmniVerb and OmniPlace

Parameters for 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	8 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	8 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	8 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	Hall1, ...	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	8 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	8 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	8 to 25088 Hz
Inj Skew	-200 to 200 ms	BassShlf G	-79.0 to 24.0 dB

Parameters

Absorption

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.

Rvrb Time

Adjusts the basic decay time of the late portion of the reverb.

LateRvbTim

Adjusts the basic decay time of the late portion of the reverb after diffusion.

HF Damping

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.

L Pre Dly, R Pre Dly

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.

Lopass

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.

EarRef Lvl
The mix level of the early reflection portion of algorithms offering early reflections.

Late Lvl
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 your 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 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**.

Size Scale
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. See Room Type for more detailed information.

InfinDecay
Found in “Verb” algorithms. When turned **On**, the reverb tail will decay indefinitely. When turned **Off**, the decay time is determined by the Rvrb Time or LateRvbTim parameters.

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
The frequency of a high shelving filter at the output of the late reverb.

TrebShlf G
The gain of a high shelving filter at the output of the late reverb.

BassShlf F
The frequency of a low shelving filter at the output of the late reverb.

BassShlf G
The gain of a low shelving filter at the output of the late reverb.

DiffAmtScl
The amount of diffusion at the onset of the reverb. For true representation of the selected Room Type diffusion amount, set to **1.00x**.

DiffLenScl
The length of the diffusion at the onset of the reverb. For true representation of the selected Room Type diffusion length, set to **1.00x**.

DiffExtent	The onset diffusion duration. Higher values create longer diffuse bursts at the onset of the reverb.
Diff Cross	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 affect.
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 Rate	The rate at which certain reverb delay lines move. See LFO Depth for more information.
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 unsuitable for real acoustic spaces, but can nonetheless create interesting effects. Instruments that have little if 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).
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	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	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	The amount of diffusion applied to the early reflection network.
E DfLenScl	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	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	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 predelays. Overall, it lengthens the decay rate of the early reflection network. Negative values polarity invert the feedback signal.

E HF Damp	The cutoff frequency of a 1 pole (6dB/oct) lowpass filter applied to the early reflection feedback signal.
E PreDlyL, E PreDlyR	The amount of delay in early reflections 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	The left and right early reflection delays fed to the same output channels.
E Dly LX, E Dly RX	The left and right early reflection delays fed to the opposite output channels.
E DifDlyL, E DifDlyR	The diffusion delays of the diffusers on delay taps fed to the same output channels.
E DifDlyLX, E DifDlyRX	The diffusion delays of the diffusers on delay taps fed to the opposite output channels.
E X Blend	The balance between early reflection delay tap signals with diffusers fed to their same output channel, and those fed to opposite channels. **0**% will only allow 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.

12 Panaural Room

Room reverberation algorithm

PAUs: 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.

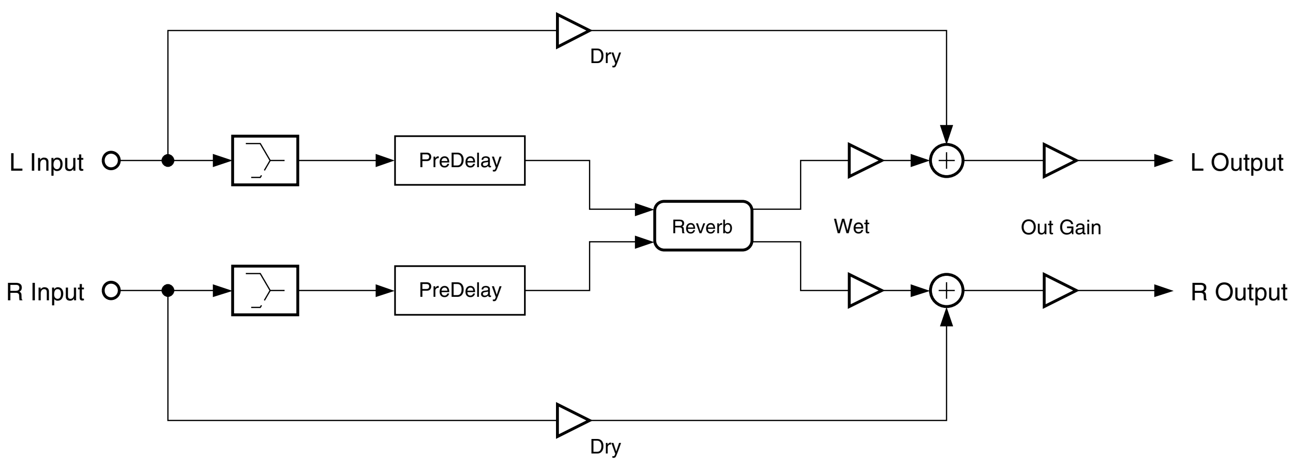

Figure 10 Simplified block diagram of Panaural Room

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	8 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 "RT60"), 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 between 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 **0m**s 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 repetitively 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.

13 Stereo Hall

A stereo hall reverberation algorithm.

PAUs: 3

The **Stereo Hall** reverberation is implemented using a special arrangement of allpass 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.

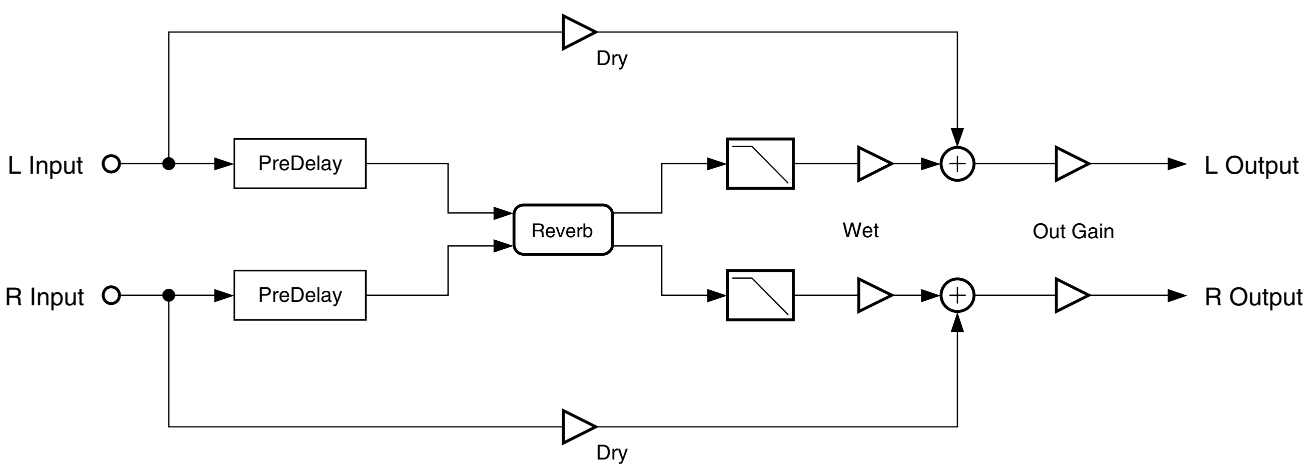

Figure 11 Simplified block diagram of Stereo Hall

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 allpass 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 (1**00**%). 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	8 to 25088 Hz		

Page 2

Bass Gain	-15 to 0 dB	Build Time	0 to 500 ms
Lowpass	8 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 HF Roll control. The wet signal is affected by the HF Roll 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 “RT60”), 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 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.

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.

Pre Dly

Introducing predelay creates a gap of silence between 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 **0%** 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 repetitively 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 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 some of the allpass 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."

14 Grand Plate

A plate reverberation algorithm.

PAUs: 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.

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 out 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, pre delay 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 presenting 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 highpass 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	8 to 25088 Hz	LF Damping	1 to 294 Hz

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Lowpass	8 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 between the dry sound and the reverberation, allowing the dry signal to stand out with greater clarity and intelligibility against the reverberant background. Especially helpful with vocals or classical music.

Decay Time The reverberation decay time (mid-band “RT60”), 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 in the range of **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, a typical value is **5920 Hz**.

LF Damping Adjusts highpass 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 **3951 Hz**.

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 **-12 dB**.

15 Finite Verb

Reverse reverberation algorithm.

PAUs: 3

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.

Dly Length and Rvb Length range from **300** to **3000** milliseconds. With the R1 Rvb Env variants, Rvb Length corresponds to a decay time (RT60).

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	8 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	8 to 25088 Hz
Mid Bass	-15 to 15 dB	Mid Damp	8 to 25088 Hz
Late Bass	-15 to 15 dB	Late Damp	8 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 the sound source.

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 RT60 for the Rvb Env cases ..R1.. where there is only one repetition.

Bass Early, Mid, and Late. 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.

Damp Early, Mid, and Late. 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.

Combination Reverbs

50 Reverb+Compress
51 Reverb<>Compress

A reverb and compressor in series.

PAUs: 3 for **Reverb<>Compress**; 2 for **Reverb+Compress**

Reverb<>Compress is configurable with the **A->B cfg** parameter as a reverb followed by a compressor **Rvb->Cmp**, or as a compressor followed by a reverb **Cmp->Rvb**. **Reverb+Compress** is configured only as a reverb followed by a compressor. The reverbs used in **Reverb<>Compress** and **Reverb+Compress** are the same as Algorithm **1 MiniVerb**. The compressor is a soft-knee compressor and can be configured as a feed-forward or feedback compressor.

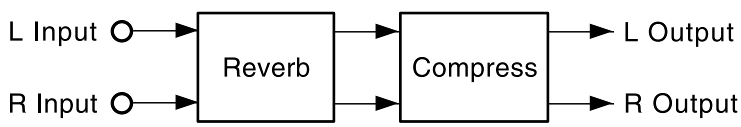

Figure 12 **Simplified block diagrams of Reverb<>Compress when set to**
(i) Rvb->Cmp
(ii) Cmp->Rvb

The main control for the reverbs is the Rv Type parameter. Rv Type changes the structure of the algorithms to simulate many carefully crafted room types and sizes. Spaces characterized as booths, small rooms, chambers, halls and large spaces can be selected. For a complete discussion on the reverbs see the sections on Algorithm **1 MiniVerb**.

The compressor reduces the signal level when the signal exceeds a threshold. The amount of compression is expressed as a ratio. The compression ratio is the inverse of the slope of the compressor input/output characteristic. The amount of compression is based on the sum of the magnitudes of the left and right channels. A compression ratio of **1:1** will have no effect on the signal. An infinite ratio, will compress all signal levels above the threshold level to the threshold level (zero slope). For ratios in between infinite and **1:1**, increasing the input will increase the output, but by less than it would if there was no compression. The threshold is expressed as a decibel level relative to digital full-scale (dBFS) where **0 dBFS** is digital full-scale and all other available values are negative.

In the soft-knee compressor there is a gradual transition from compressed to unity gain.

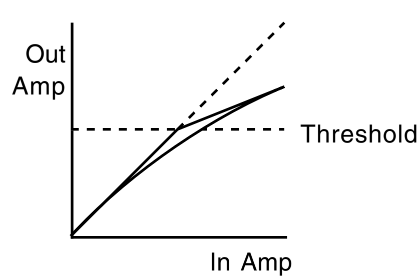

Figure 13 **Soft-Knee compression characteristics**

To determine how much to compress the signal, the compressor must measure the signal level. Since musical signal levels will change over time, the compression amounts must change as well. You can control the rate at which compression changes in response to changing signal levels with the attack and release time controls. With the attack time, you set how fast the compressor responds to increased levels. At long attack times, the signal may overshoot the threshold level for some time before it becomes fully compressed, while at short attack times, the compressor will rapidly clamp down on the level. The release time controls how long it takes the compressor to respond to a reduction in signal levels. At long release times, the signal may stay compressed well after the signal falls below threshold. At short release times, the compressor will open up almost as soon as the signal drops.

For typical compressor behavior, the attack time is considerably shorter than the release time. At very short attack and release times, the compressor is almost able to keep up with the instantaneous signal levels and the algorithm will behave more like distortion than compression. In addition to the attack and release times, there is another time parameter: CompSmooth. The smoothing parameter will increase both the attack and release times, although the effect is significant only when its time is longer than the attack or release time. Generally the smoothing time should be kept at or shorter than the attack time.

You have the choice of using the compressors configured as feed-forward or feedback compressors. For feed-forward, set the FdbkComprs parameter to **Out**; for feedback compression, set it to **In**. The feed-forward configuration uses the input signal as the side-chain source. The feedback compressor on the other hand uses the compressor output as the side-chain source. Feedback compression tends to be more subtle, but you cannot get an instant attack.

In the feedback configuration, the signal being compressed may be delayed relative to the side chain compression processing. The delay allows the signal to start being compressed just before an attack transient arrives. Since the side chain processing "knows" what the input signal is going to be before the main signal path does, it can tame down an attack transient by compressing the attack before it actually happens. In the feed-forward configuration, the delay affects both the main signal and the side chain, and so is of limited usefulness. In compressors which use more than 1 PAU, the delay affects the main signal only, regardless of the side chain configuration.

The **Reverb<>Compress** algorithm also provides side chain equalization. Using side chain equalization allows you to compress your signal based on the spectral (frequency) content of your signal. For example, by boosting the treble shelf filter, you can compress the signal only when there is a lot of high frequencies present.

A meter displays the amount of gain reduction applied to the signal as a result of compression.

Parameters:

Page 1

In/Out	In or Out	ReverbGain	Off, -79.0 to 24.0 dB
Reverb W/D	0 to 100 %wet	Rv Time	0.5 to 30.0 s, Inf
Rv PreDlyL	0 to 620 ms	Rv PreDlyR	0 to 620 ms
Rv HFDamp	8 to 25088 Hz	CompIn/Out	In or Out

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A->B cfg	Rvb->Cmp, Cmp->Rvb		
Rv Type	Hall1, etc.	Rv DiffScl	0.00 to 2.00 x
		Rv SizeScl	0.00 to 4.00 x
		Rv Density	0.00 to 4.00 x

Page 3

<table>
<tr><td>Comp Atk</td><td>0.0 to 228.0 ms</td><td>Comp Ratio</td><td>1.0:1 to 100.0:1, Inf:1</td></tr>
<tr><td>Comp Rel</td><td>0 to 3000 ms</td><td>Comp Thres</td><td>-79.0 to 0.0 dB</td></tr>
<tr><td>CompSmooth</td><td>0.0 to 228.0 ms</td><td>CompMakeUp</td><td>Off, -79.0 to 24.0 dB</td></tr>
<tr><td>CompSigDly</td><td>0.0 to 25.0 ms</td><td>FdbkComprs</td><td>In or Out</td></tr>
<tr><td></td><td></td><td>||||||||||||||||||||||||||||||</td><td>Reduction</td></tr>
<tr><td></td><td>-dB 40 20 12 8</td><td>6 4 2</td><td>0</td></tr>
</table>

Page 4 (Reverb<>Compress only)

SCBassGain	-79.0 to 24.0 dB	SCTrebGain	-79.0 to 24.0 dB
SCBassFreq	8 to 25088 Hz	SCTrebFreq	8 to 25088 Hz
SCMidGain	-79.0 to 24.0 dB	SCEQIn/Out	In or Out
SCMidFreq	8 to 25088 Hz		
SCMidWidth	0.010 to 5.000 oct		

In/Out When set to **In** the overall algorithm is active; when set to **Out** the algorithm is bypassed.

ReverbW/D This is a simple mix of the reverb input (dry) with the reverb output (wet) to produce the final reverb output.

ReverbGain An overall level control of the reverb's output (applied after the reverb Wet/Dry mix).

Rv HFDamp Reduces high frequency components of the reverb above the displayed cutoff frequency. Removing higher reverb frequencies can often make rooms sound more natural.

Rv PreDlyL/R The delay between the start of a sound and the output of the first reverb reflections from that sound. Longer predelays 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.

CompIn/Out When set to **In** the compressor is active; when set to **Out** the compressor is bypassed.

A->B cfg For **Reverb<>Compress** only, a switch to configure the algorithm as reverb followed by compressor **Rvb->Cmp** or as compressor followed by reverb **Cmp->Rvb**. **Reverb+Compress** is always configured as **Rvb->Cmp**.

Rv 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 Rv Types will sound best with shorter Rv Times, and vice versa. (Note that since this parameter changes the structure of the reverb algorithm, you may not modulate it.)

Comp Atk The time for the compressor to start to cut in when there is an increase in signal level (attack) above the threshold.

Comp Rel The time for the compressor to stop compressing when there is a reduction in signal level (release) from a signal level above the threshold.

CompSmooth	A lowpass filter in the control signal path. It is intended to smooth the output of the expander's envelope detector. Smoothing will affect the attack or release times when the smoothing time is longer than one of the other times.
CompSigDly	The time in ms by which the input signal should be delayed with respect to compressor side chain processing (i.e. side chain predelay). This allows the compression to appear to take effect just before the signal actually rises.
Comp Ratio	The compression ratio. High ratios are highly compressed; low ratios are moderately compressed.
Comp Thres	The threshold level in dBFS (decibels relative to full scale) above which the signal begins to be compressed.
CompMakeUp	Provides an additional control of the output gain. The Out Gain and MakeUpGain controls are additive (in decibels) and together may provide a maximum of 24 dB boost to offset gain reduction due to compression.
FdbkComprs	A switch to set whether the compressor side chain is configured for feed-forward (**Out**) or feedback (**In**).

The following apply only to the **Reverb<>Compress** algorithm:

SCEQIn/Out	A switch to bypass the compressor side chain equalization.
SCBassGain	The amount of boost or cut that the side chain bass shelving filter should apply to the low frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the bass signal below the specified frequency. Negative values cut the bass signal below the specified frequency.
SCBassFreq	The center frequency of the side chain bass shelving filter in intervals of one semitone.
SCTrebGain	The amount of boost or cut that the side chain treble shelving filter should apply to the high frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the treble signal above the specified frequency. Negative values cut the treble signal above the specified frequency.
SCTrebFreq	The center frequency of the side chain treble shelving filters in intervals of one semitone.
SCMidGain	The amount of boost or cut that the side chain parametric mid filter should apply in dB to the specified frequency band. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the signal at the specified frequency. Negative values cut the signal at the specified frequency.
SCMidFreq	The center frequency of the side chain parametric mid filter in intervals of one semitone. The boost or cut will be at a maximum at this frequency.
SCMidWidth	The bandwidth of the side chain parametric mid filter may be adjusted. You specify the bandwidth in octaves. Small values result in a very narrow filter response. Large values result in a very broad response.

52 ClascVrb<>Comprs

A reverb and compressor in series.

PAUs: 3

ClascVrb<>Comprs is configurable with the “A->B cfg” parameter as a reverb followed by a compressor “Rvb->Cmp”, or as a compressor followed by a reverb “Cmp->Rvb”. It uses the same reverb as **5 Classic Verb**.

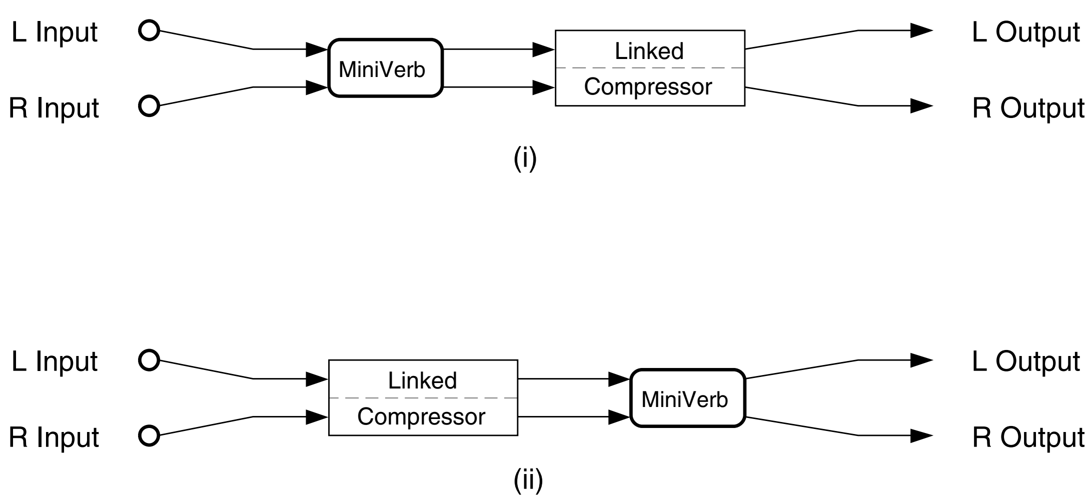

Figure 14 Simplified block diagrams of ClascVrb<>Comprs when set to
(i) Rvb->Cmp
(ii) Cmp->Rvb

The main control for the reverbs is the Rv Type parameter. Rv Type changes the structure of the algorithms to simulate many carefully crafted room types and sizes. Spaces characterized as booths, small rooms, chambers, halls and large spaces can be selected. For a complete discussion on the reverbs see the section on **5 Classic Verb**.

The compressor reduces the signal level when the signal exceeds a threshold. The amount of compression is expressed as a ratio. The compression ratio is the inverse of the slope of the compressor input/output characteristic. The amount of compression is based on the sum of the magnitudes of the left and right channels. A compression ratio of **1:1** will have no effect on the signal. An infinite ratio, will compress all signal levels above the threshold level to the threshold level (zero slope). For ratios in between infinite and **1:1**, increasing the input will increase the output, but by less than it would if there was no compression. The threshold is expressed as a decibel level relative to digital full-scale (dBFS) where **0 dBFS** is digital full-scale and all other available values are negative.

In the soft-knee compressor there is a gradual transition from compressed to unity gain.

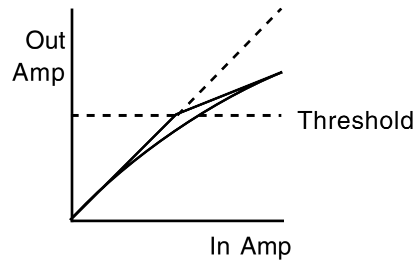

Figure 15 Soft-Knee compression characteristics

To determine how much to compress the signal, the compressor must measure the signal level. Since musical signal levels will change over time, the compression amounts must change as well. You can control the rate at which compression changes in response to changing signal levels with the attack and release time controls. With the attack time, you set how fast the compressor responds to increased levels. At long attack times, the signal may over-shoot the threshold level for some time before it becomes fully compressed, while at short attack times, the compressor will rapidly clamp down on the level. The release time controls how long it takes the compressor to respond to a reduction in signal levels. At long release times, the signal may stay compressed well after the signal falls below threshold. At short release times, the compressor will open up almost as soon as the signal drops.

For typical compressor behavior, the attack time is considerably shorter than the release time. At very short attack and release times, the compressor is almost able to keep up with the instantaneous signal levels and the algorithm will behave more like distortion than compression. In addition to the attack and release times, there is another time parameter: CompSmooth. The smoothing parameter will increase both the attack and release times, although the effect is significant only when its time is longer than the attack or release time. Generally the smoothing time should be kept at or shorter than the attack time.

A meter is provided to display the amount of gain reduction that is applied to the signal as a result of compression.

Parameters:

Page 1

Reverb W/D	0 to 100%wet	ReverbGain	Off, -79.0 to 24.0 dB
Rv Time	0.5 to 30.0s, Inf	Rv HF Damp	8 to 25088 Hz
Rv PreDlyL	0 to 620ms	Rv PreDlyR	0 to 620ms
A->B cfg	Rvb->Cmp	Rv ErefLvl	-100 to 100%

Page 2

Rv Type	Hall1, ...	RvDfAmtScl	0.00 to 2.00x
Rv SizeScl	0.01 to 2.00x	RvDfLenScl	0.00 to 2.00x
Rv LateLvl	-100 to 100%	RvLFORate	0.01 to 10.00 Hz
RvInfDecay	On or Off	RvLFODepth	0.0 to 100.0 ct
RvTrbShlfF	8 to 25088 Hz	RvLFSplit	8 to 25088 Hz
RvTrbShlfG	-79.0 to 24.0 dB	RvLFTime	0.50 to 1.50x

Page 3

RvEDfDlySc	0.00 to 2.00x	RvE X Blend	0 to 100 %
RvEDiffAmt	-100 to 100%		
RvEDly L	0.0 to 720.0 ms	RvEDlyR	0.0 to 720.0 ms
RvEDlyLX	0.0 to 720.0 ms	RvEDlyRX	0.0 to 720.0 ms
RvEDfDlyL	0.0 to 160.0 ms	RvEDfDlyR	0.0 to 160.0 ms
RvEDfDlyLX	0.0 to 230.0 ms	RvEDfDlyRX	0.0 to 230.0 ms

Page 4

CompIn/Out	In or Out	Comp Ratio	1.0:1 to 100:1, Inf:1
Comp Atk	0.0 to 228.0 ms	Comp Thres	-79.0 to 0.0dB
Comp Rel	0 to 3000 ms	CompMakeUp	Off, -79.0 to 24.0 dB
CompSmooth	0.0 to 228.0 ms	CompSigDly	0.0 to 25.0ms
		||||||||||||||||||||||||||||	Reduction
	-dB 40 20 12 8	6 4 2	0

In/Out When set to **In** the overall algorithm is active; when set to **Out** the algorithm is bypassed.

ReverbW/D This is a simple mix of the reverb input (dry) with the reverb output (wet) to produce the final reverb output.

ReverbGain An overall level control of the reverb's output (applied after the reverb Wet/Dry mix).

Rv HFDamp Reduces high frequency components of the reverb above the displayed cutoff frequency. Removing higher reverb frequencies can often make rooms sound more natural.

Rv PreDlyL/R The delay between the start of a sound and the output of the first reverb reflections from that sound. Longer predelays 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.

CompIn/Out When set to **In** the compressor is active; when set to **Out** the compressor is bypassed.

A->B cfg A switch to configure the algorithm as reverb followed by compressor **Rvb->Cmp** or as compressor followed by reverb **Cmp->Rvb**.

Rv 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 Rv Types will sound best with shorter Rv Times, and vice versa. (Note that since this parameter changes the structure of the reverb algorithm, you may not modulate it.)

Comp Atk The time for the compressor to start to cut in when there is an increase in signal level (attack) above the threshold.

Comp Rel The time for the compressor to stop compressing when there is a reduction in signal level (release) from a signal level above the threshold.

CompSmooth A lowpass filter in the control signal path. It is intended to smooth the output of the expander's envelope detector. Smoothing will affect the attack or release times when the smoothing time is longer than one of the other times.

CompSigDly The time in ms by which the input signal should be delayed with respect to compressor side chain processing (i.e. side chain predelay). This allows the compression to appear to take effect just before the signal actually rises.

Comp Ratio The compression ratio. High ratios are highly compressed; low ratios are moderately compressed.

Comp Thres The threshold level in dBFS (decibels relative to full scale) above which the signal begins to be compressed.

CompMakeUp Provides an additional control of the output gain. The Out Gain and MakeUpGain controls are additive (in decibels) and together may provide a maximum of 24 dB boost to offset gain reduction due to compression.

Vocal Combination Algorithms

53 Gate+Cmp[EQ]+Rvb
54 Gate+Cmp<>EQ+Rvb
608 MnGt+Cmp[EQ]+Rvb
609 MnGt+Cmp<>EQ+Rvb

Combination algorithms designed for vocal processing.

PAUs: 4 each

Two combination algorithms are provided with vocal processing in mind. Both include a gate followed by a compressor and a reverb. In **Gate+Cmp[EQ]+Rvb**, equalization is included as part of the compressor's side-chain processing. Side-chain equalization allows some interesting processing possibilities including "de-essing" (by boosting the treble in the side-chain). In **Gate+Cmp<>EQ+Rvb**, the equalization can be configured before or after the compressor. For each configuration of compressor and EQ, the EQ includes bass, treble and mid controls (gain and frequency for each plus width for the mid EQ).

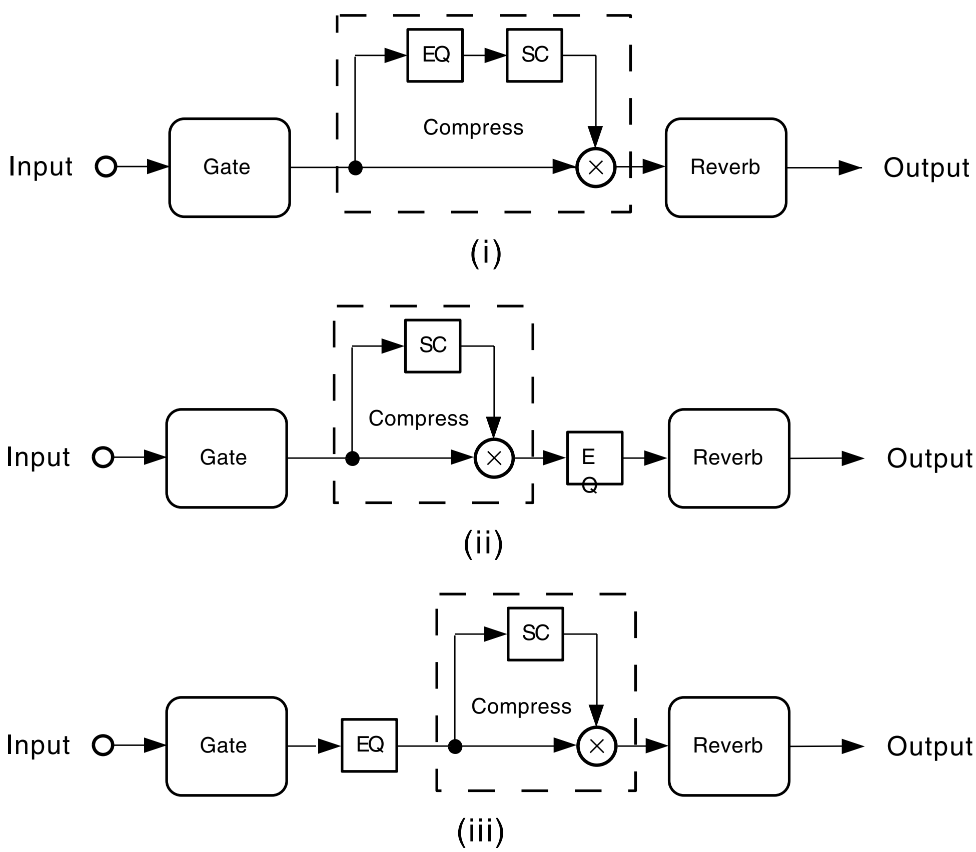

Figure 16 **Simplified compressor and EQ configurations**
(i) Gate+Cmp[EQ]+Rvb
(ii) Gate+Cmp<>EQ+Rvb set to Cmp->EQ
(iii) set to EQ->Cmp

The gate (same gate as Algorithm **343 Gate**) allows you to cut out noise during vocal silence. You must decide whether to gate based on left or right channels or to gate based on both channels (average magnitude). Both the gate and compressor have their own side-chain processing paths. For both the gate and compressor, side-chain input may be taken from either the left or right channels, or the average signal magnitude of the left and right channels may be selected using the GateSCInp or CompSCInp parameters.

The reverb is the same as used in Algorithm **1 MiniVerb**. You will find all the same controls and room settings. In the FXPreset editor, you will have to scroll with the **more>** soft button to find the **PARAM5** soft button containing the reverb parameters.

Parameters:

Page 1   (for Gate+Cmp[EQ]+Rvb)

GateIn/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
GateSCInp	L, R, (L+R)/2	CompIn/Out	In or Out
		CompSCInp	L, R, (L+R)/2
		FdbkComprs	In or Out

Page 1   (for Gate+Cmp<>EQ+Rvb)

GateIn/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
GateSCInp	L, R, (L+R)/2	CompIn/Out	In or Out
		CompSCInp	L, R, (L+R)/2
		A->B cfg	Cmp->EQ

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Gate Thres	-79.0 to 0.0 dB	Gate Time	25 to 3000 ms
Gate Duck	On or Off	Gate Atk	0.0 to 228.0 ms
		Gate Rel	0 to 3000 ms
		GateSigDly	0.0 to 25.0 ms
		Reduction	-dB 60 40 ✻ 16 ✻ 8 4 0

Page 3

Comp Atk	0.0 to 228.0 ms	Comp Ratio	1.0:1 to 100:1, Inf:1
Comp Rel	0 to 3000 ms	Comp Thres	-79.0 to 0.0dB
CompSmooth	0.0 to 228.0 ms	CompMakeUp	Off, -79.0 to 24.0 dB
CompSigDly	0.0 to 25.0ms		

Page 4 (for Gate+Cmp[EQ]+Rvb)

CmpSCBassG	-79.0 to 24.0 dB	CmpSCTrebG	-79.0 to 24.0 dB
CmpSCBassF	8 to 25088 Hz	CmpSCTrebF	8 to 25088 Hz
CmpSCMidG	-79.0 to 24.0 dB	Comp SC EQ	In or Out
CmpSCMidF	8 to 25088 Hz		
CmpSCMidW	0.010 to 5.000 oct		

Page 4 (Gate+Cmp<>EQ+Rvb)

Bass Gain	-79.0 to 24.0 dB	Treb Gain	-79.0 to 24.0 dB
Bass Freq	8 to 25088 Hz	Treb Freq	8 to 25088 Hz
Mid Gain	-79.0 to 24.0 dB		
Mid Freq	8 to 25088 Hz		
Mid Width	0.010 to 5.000 oct		

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Reverb W/D	0 to 100 %wet		
Rv Type	Hall1, etc.		
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	8 to 25088 Hz
Rv PreDlyL	0 to 620 ms	Rv PreDlyR	0 to 620 ms

Out Gain The overall gain or amplitude at the output of the entire algorithm.

GateIn/Out When set to **In** the gate is active; when set to **Out** the gate is bypassed.

GateSCInp Select the input source channel for gate side-chain processing—left, right or both. For both (L+R)/2 the averaged magnitude is used.

CompIn/Out When set to **In** the compressor is active; when set to **Out** the compressor is bypassed.

CompSCInp Select the input source channel for compressor side-chain processing—**Left**, **Right** or **Both**. For both (L+R)/2 the averaged magnitude is used.

FdbkComprs A switch to set whether the compressor side-chain is configured for feed-forward (**Out**) or feedback (**In**). Feedback compression is not available in the **Gate+Cmp<>EQ+Rvb** algorithm.

A->B cfg Controls the routing order of the compressor and EQ in **Gate+Cmp<>EQ+Rvb**. When set to **Cmp->EQ**, the output of the compressor feeds into the EQ. When set to **EQ->Cmp**, the EQ feeds into the compressor. A compressor is a non-linear, time-variant effect, so the relative order can make a difference, particularly when the compression is extreme enough to behave as distortion.

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

Gate Duck 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.

Gate Time 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. If Retrigger is **On**, the gate timer is continually reset while the side chain signal is above the threshold.

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

Gate Rel The time for the gate to ramp from open to closed (reverse if Gate Duck is **On**) after the gate timer has elapsed.

GateSigDly The delay in milliseconds (ms) of the signal to be gated relative to the side chain signal. By delaying the main signal, the gate can be opened before the main signal rises above the gating threshold.

Comp Atk The time for the compressor to start to cut in when there is an increase in signal level (attack) above the threshold.

Comp Rel The time for the compressor to stop compressing when there is a reduction in signal level (release) from a signal level above the threshold.

CompSmooth A lowpass filter in the compressor side-chain signal path. It is intended to smooth the output of the compressor's envelope detector. Smoothing will affect the attack or release times when the smoothing time is longer than one of the other times.

CompSigDly The time in ms by which the input signal should be delayed with respect to compressor side-chain processing (i.e. side-chain predelay). This allows the compression to appear to take effect just before the signal actually rises.

Comp Ratio The compression ratio. High ratios are highly compressed; low ratios are moderately compressed.

Comp Thres The compressor threshold level in dBFS (decibels relative to full scale) above which the signal begins to be compressed.

CompMakeUp A gain or amplitude control provided to offset gain reduction due to compression.

The EQ parameters with names starting with **CmpSC** refer to EQ filters in the side-chain processing path of **Gate+Cmp[EQ]+Rvb**. The prefix is not used in **Gate+Cmp<>EQ+Rvb** where the EQ is in the main signal path.

CmpSCBassG, Bass Gain — The amount of boost or cut that the bass shelving filter should apply to the low frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the bass signal below the specified frequency. Negative values cut the bass signal below the specified frequency.

CmpSCBassF, Bass Freq — The center frequency of the bass shelving filter in intervals of one semitone.

CmpSCTrebG, Treb Gain — The amount of boost or cut that the treble shelving filter should apply to the high frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the treble signal above the specified frequency. Negative values cut the treble signal above the specified frequency.

CmpSCTrebF, Treb Freq — The center frequency of the treble shelving filters in intervals of one semitone.

CmpSCMidG, Mid Gain — The amount of boost or cut that the parametric mid filter should apply in dB to the specified frequency band. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the signal at the specified frequency. Negative values cut the signal at the specified frequency.

CmpSCMidF, Mid Freq — The center frequency of the parametric mid filter in intervals of one semitone. The boost or cut will be at a maximum at this frequency.

CmpSCMidW, Mid Width — The bandwidth of the side chain parametric mid filter may be adjusted. You specify the bandwidth in octaves. Small values result in a very narrow filter response. Large values result in a very broad response.

Reverb W/D — A simple mix of the reverb sound with the dry (compressed) sound.

Rv PreDlyL/R — The delay between the start of a sound and the output of the first reverb reflections from that sound. Longer predelays 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.

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

Rv 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 Rv Types will sound best with shorter Rv Times, and vice versa. (Note that since this parameter changes the structure of the reverb algorithm, you may not modulate it.)

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

Rv DiffScl — A multiplier which affects the diffusion of the reverb. At **1.00x**, the diffusion will be the normal, carefully adjusted amount for the current

Rv Type. Altering this parameter will change the diffusion from the preset amount.

Rv SizeScl	A multiplier which changes the reverb size of the current room. At **1.00x**, the room will be the normal, carefully tweaked size of the current Rv 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).
Rv 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 Rv Type. Altering this parameter will change the density of the reverb, which may color the room slightly.

More Reverbs

100 LaserVerb
101 LaserVerb Lite
102 Mono LaserVerb
605 Mn LaserVerb

A bizarre reverb with a falling buzz

PAUs: 1 for **Mono LaserVerb**
2 for **LaserVerb Lite**
3 for **LaserVerb**

LaserVerb has to be heard to be believed! Feed it an impulsive sound such as a snare drum, and **LaserVerb** plays the impulse back as a delayed train of closely spaced impulses, and as time passes, the spacing between the impulses gets wider. The close spacing of the impulses produces a discernible buzzy pitch which gets lower as the impulse spacing increases. The following figure is a simplified representation of the **LaserVerb** impulse response. (An impulse response of a system is what you would see if you had an oscilloscope on the system output and you gave the system an impulse or a spike for an input.)

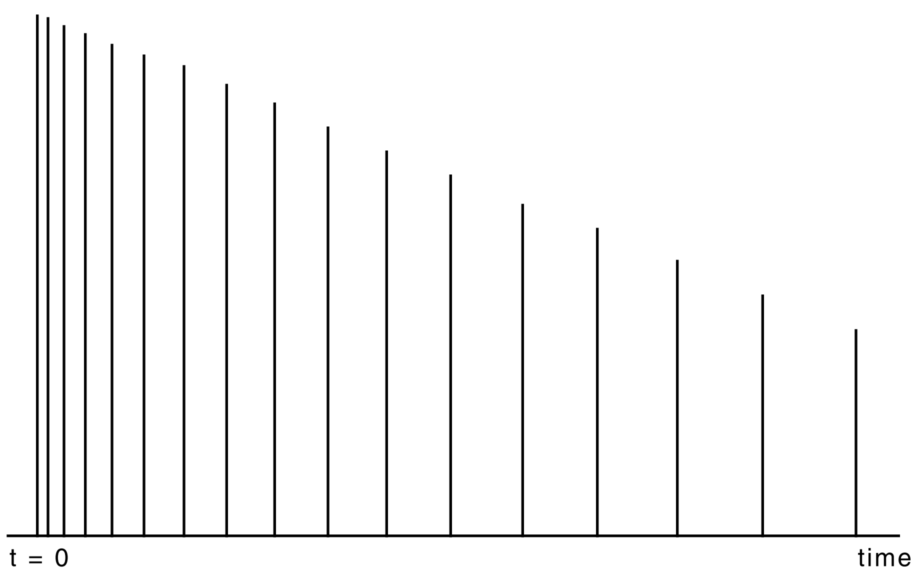


Figure 17 Simplified impulse response of LaserVerb

With appropriate parameter settings this effect produces a descending buzz or whine somewhat like a diving airplane or a siren being turned off. The descending buzz is most prominent when given an impulsive input such as a drum hit. When used as a reverb, it tends to be highly metallic and has high pitched tones at certain parameter settings. To get the descending buzz, start with about half a second of delay, set the Contour parameter to a high value (near **1**), and set the HF Damping to a low value (at or near **0**). The Contour parameter controls the overall shape of the LaserVerb impulse response. At high values the response builds up very quickly decays slowly. As the Contour value is reduced, the decay becomes shorter and the sound takes longer to build up. At a setting of **0**, the response degenerates to a simple delay.

The Spacing parameter controls the initial separation of impulses in the impulse response and the rate of their subsequent separation. Low values result in a high initial pitch (impulses are more closely spaced) and takes longer for the pitch to lower.

The output from LaserVerb can be fed back to the input. By turning up the feedback, the duration of the LaserVerb sound can be greatly extended. Cross-coupling may also be used to move the signal between left and right channels, producing a left/right ping-pong effect at the most extreme settings.

The two processing allocation unit (PAU) version is a sparser version than the three-PAU version. Its buzzing is somewhat coarser. The one-PAU version is like the two-PAU version except the two input channels are summed and run through a single mono **LaserVerb**. The one-PAU version does not have the cross-coupling control but does have output panning.

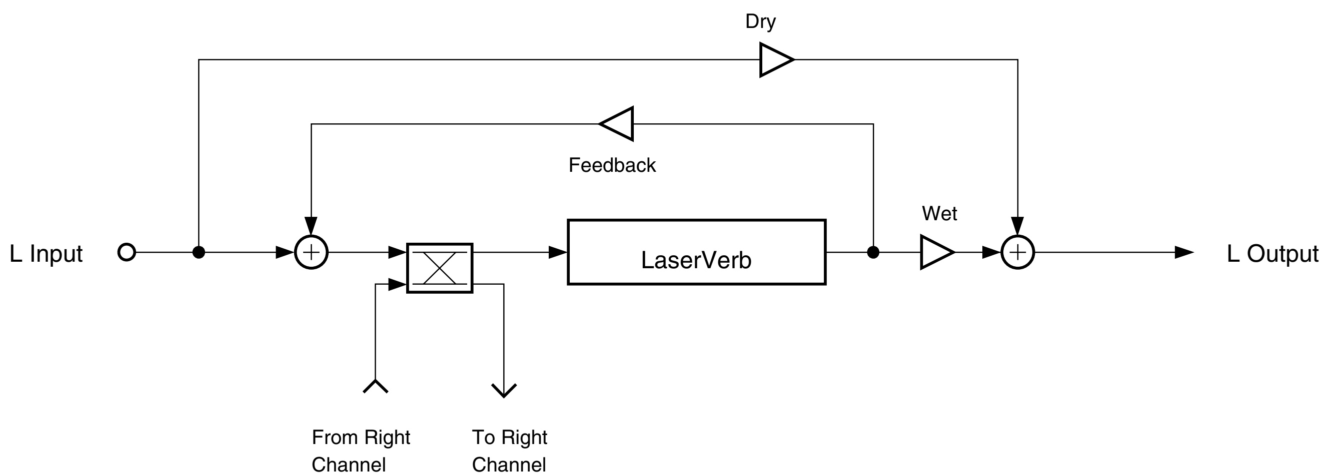


Figure 18 LaserVerb

Parameters for LaserVerb and LaserVerb Lite

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0dB
Fdbk Lvl	0 to 100%		
Xcouple	0 to 100%		
HF Damping	8 to 25088Hz		

Parameters for Mono LaserVerb

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0dB
Fdbk Lvl	0 to 100%	Pan	-100 to 100%
HF Damping	8 to 25088Hz		

Page 2

Dly Coarse	0 to 5000ms	Contour	0.0 to 100.0%
Dly Fine	-20.0 to 20.0ms		
Spacing	0.0 to 40.0samp		

Wet/Dry The amount of reverbed (wet) signal relative to unaffected (dry) signal.

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

Fdbk Lvl The percentage of the reverb output to feed back or return to the reverb input. Turning up the feedback is a way to stretch out the duration of the reverb, or, if the reverb is set to behave as a delay, to repeat the delay. The higher feedback is set, the longer the decay or echo will last.

Xcouple **LaserVerb** and **LaserVerb Lite** are stereo effects. The cross-coupling control lets you send the sum of the input and feedback from one channel to its own LaserVerb effect (0% cross coupling) or to the other channel's effect (100% cross coupling) or somewhere in between. This control is not available in Mono LaserVerb.

HF Damping The damping of high frequencies relative to low frequencies. When set to the highest frequency (**25088 Hz**), there is no damping and all frequencies decay at the same rate. At lower frequency settings, high frequency signal components will decay faster than low frequency components. If set too low, everything will decay almost immediately.

Pan The Pan control is available in the **Mono LaserVerb**. The left and right inputs get summed to mono, the mono signal passes through the LaserVerb, and the final mono output is panned to the left and right outputs. Panning ranges from **-100**% (fully left), through **0**% (centered), through to **100**% (fully right).

Dly Coarse You can set the overall delay length from **0** to **2** seconds (three-PAU) or **0** to **1.3** seconds (two-PAU). Lengthening the delay will increase the duration or decay time of the reverb. To reduce **LaserVerb** to a simple delay, set the Contour and Feedback controls to **0**. Use a delay of about half a second as a starting point.

Dly Fine The delay fine adjust is added to the delay coarse adjust to provide a delay resolution down to **0.1 ms**.

Spacing Determines the starting pitch of the descending buzz and how fast it descends. The Spacing parameter sets the initial separation of impulses in the impulse response and subsequent rate of increasing impulse separation. The spacing between impulses is given in samples and may be a fraction of a sample. (A sample is the time between successive digital words which is 20.8 μs or 1/48000 seconds.) For low values, the buzz starts at high frequencies and drops slowly. At high values the buzz starts at a lower pitch and drops rapidly.

Contour Controls the overall envelope shape of the reverb. When set to a high value, sounds passed through the reverb 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. At a value of around **34**, the reverb is behaving like a reverse reverb, building up to a hit. When the Contour is set to **0**, **LaserVerb** is reduced to a simple delay.

103 Revrse LaserVerb

A bizarre reverb which runs backwards in time (uh, yeah).

PAUs: 4

Revrse LaserVerb is a mono effect that simulates the effect of running the **LaserVerb** (Algorithms **100–102**) in reverse. When you play a sound through the algorithm, it starts out relatively diffuse then builds to the final "hit." Since KDFX cannot break the universal rules of causality (sorry, KDFX doesn't know what you are about to play!), there can be a significant delay between what you play and when you hear it. In addition to the normal Wet/Dry control, with the Rvrs W/D, the dry signal is considered to be the delayed "hit" signal.

Revrse LaserVerb is **LaserVerb** in reverse, so when it is fed an impulsive sound such as a snare drum, it plays the impulse back as a delayed train of closely spaced impulses, and as time passes, the spacing between the impulses gets closer until they coalesce at the "hit." The close spacing of the impulses produces a discernible buzzy pitch which gets higher as the impulse spacing decreases. The following figure is a simplified representation of the **Revrse LaserVerb** impulse response. (An impulse response of a system is what you would see if you had an oscilloscope on the system output and you gave the system an impulse or a spike for an input.)

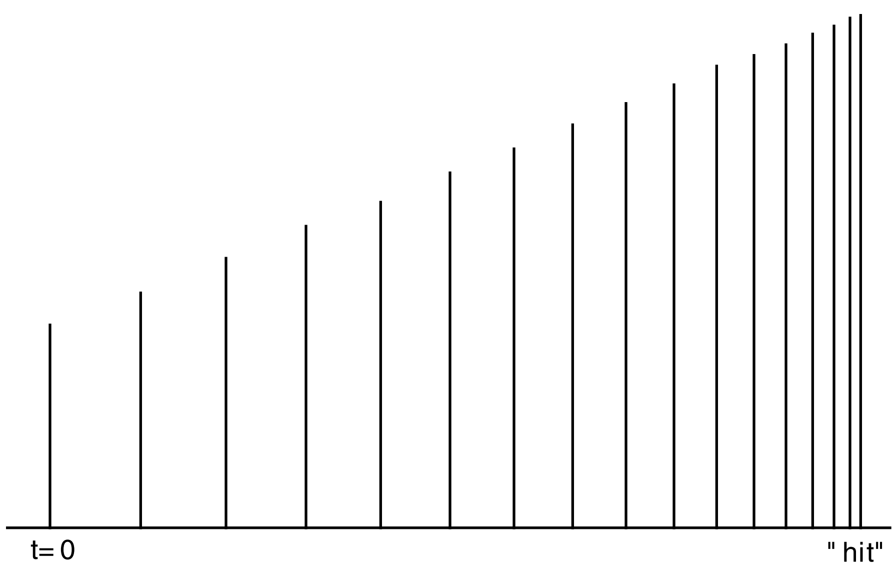


Figure 19 Simplified impulse response of Revrse LaserVerb

With appropriate parameter settings this effect produces an ascending buzz or whine. The ascending buzz is most prominent when given an impulsive input such as a drum hit. To get the ascending buzz, start with about half a second of delay and set the Contour parameter to a high value (near **100**%). The Contour parameter controls the overall shape of the **LaserVerb** impulse response. At high values the response builds up slowly to the "hit." As the Contour value is reduced, the response starts out lower and rises more rapidly to the "hit."

The Spacing parameter controls the initial separation of impulses in the impulse response and the rate of their subsequent separation. Low values result in a high initial pitch (impulses are more closely spaced) and takes longer for the pitch to lower.

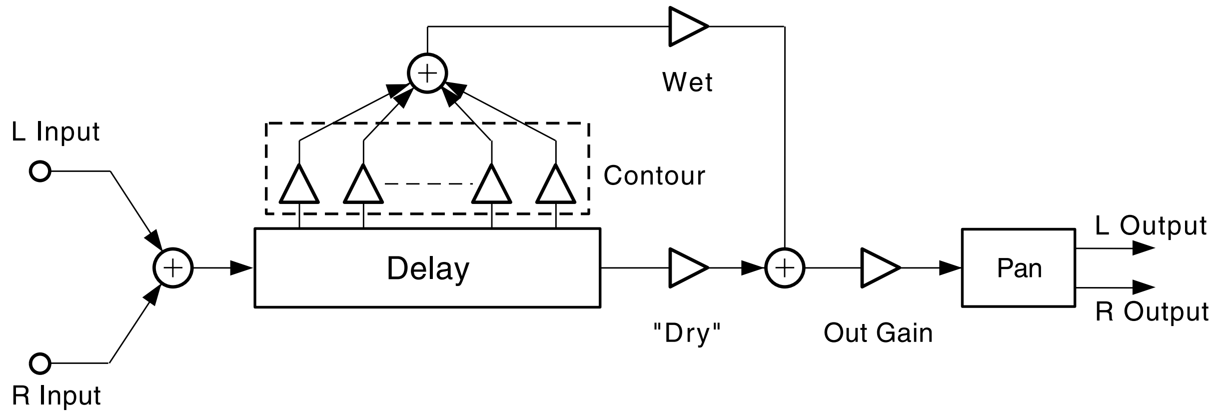


Figure 20 Revrse LaserVerb

Parameters:

Page 1

Wet/Dry	0 to 100 %wet	Out Gain	Off, -79.0 to 24.0 dB
Rvrs W/D	0 to 100 %wet	Pan	-100 to 100 %

Page 2

Dly Coarse	0 to 5000 ms	Contour	0.0 to 100.0 %
Dly Fine	-20.0 to 20.0 ms		
Spacing	0 to 200 samp		

Wet/Dry The amount of reverbed (wet) signal relative to unaffected (dry) signal.

Rvrs W/D A special wet/dry control in which the "dry" signal is in fact delayed so that it is the last sound to be sent to the output, as if the LaserVerb is being played in reverse.

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

Pan The left and right inputs get summed to mono, the mono signal passes through the **Revrse LaserVerb**, and the final mono output is panned to the left and right outputs. Panning ranges from **-100**% (fully left), through **0**% (centered), through to **100**% (fully right).

Dly Coarse You can set the overall delay length from **0** to **5** seconds. Lengthening the delay will increase the duration or decay time of the reverb.

Dly Fine The delay fine adjust is added to the delay coarse adjust to provide a delay resolution down to **0.2** ms.

Spacing Determines the starting pitch of the ascending buzz and how fast it ascends. The Spacing parameter sets the initial separation of impulses in the impulse response and subsequent rate of decreasing impulse separation. The spacing between impulses is given in samples and may be a fraction of a sample. (A sample is the time between successive digital words which is 20.8 μs or 1/48000 seconds.) For low values, the buzz builds to a higher frequency than for higher Spacing settings.

Contour Controls the overall envelope shape of the reverb. When set to a high value, sounds start at a high level and build slowly to the final "hit." As the control value is reduced, sounds start lower and build rapidly to the final "hit."

104 Gated LaserVerb

The LaserVerb algorithm with a gate on the output.

PAUs: 3

Gated LaserVerb is Algorithm **101 LaserVerb Lite** with a gate on the output. For a detailed explanation of **LaserVerb** see the section for Algorithm **101 LaserVerb Lite**. The gate controls are covered under Algorithm **343 Gate**. Signal routings between the inputs, the **LaserVerb**, the gate, and the outputs are described here.

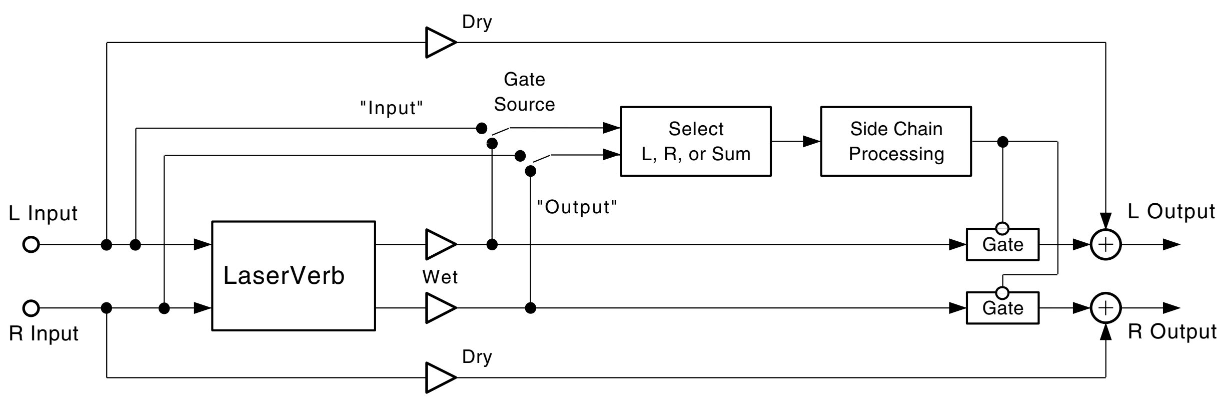


Figure 21 Signal flow of Gated LaserVerb

LaserVerb is a stereo algorithm that produces interesting sounds in the reverb decay. However, the decay often lasts longer than desired. The gate may be used to cut the output signal after the input signal drops below a threshold. You may select whether to gate the **LaserVerb** output based on the input signal level or the signal level at the output of the **LaserVerb**. In most cases the gate would be based on the input signal. When you gate on the output signal, you must wait for the **LaserVerb** tail to drop below the threshold before the gate will close. Whether you gate based on the input or the output signal strength, you can select which input or output channel to use as the gating side chain signal: left, right, or the average of the left and right magnitudes.

Parameters:

Page 1

Wet/Dry	0 to 100 %wet	Out Gain	Off, -79.0 to 24.0 dB
Fdbk Lvl	0 to 100 %	GateIn/Out	In or Out
Xcouple	0 to 100 %	GateSCInp	L, R, (L+R)/2
HF Damping	8 to 25088 Hz	GateSCSrc	Input or Output

Page 2

Delay Crs	0 to 5000 ms	Contour	0.0 to 100.0 %
Delay Fine	-20.0 to 20.0 ms		
Spacing	0.0 to 40.0 samp		

Page 3

Gate Thres	-79.0 to 0.0 dB	Gate Time	25 to 3000 ms
Gate Duck	On or Off	Gate Atk	0.0 to 228.0 ms
		Gate Rel	0 to 3000 ms
		GateSigDly	0.0 to 25.0 ms
		||||||||||||||||||||||||||||||	Reduction
	-dB 60 40 * 16	* 8 4	0

Wet/Dry The amount of reverbed and gated (wet) signal relative to unaffected (dry) signal. The gate is on the wet signal path.

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

Fdbk Lvl The percentage of the reverb output to feed back or return to the reverb input. Turning up the feedback is a way to stretch out the duration of the reverb, or, if the reverb is set to behave as a delay, to repeat the delay. The higher feedback is set, the longer the decay or echo will last.

Xcouple **LaserVerb Lite** is a stereo effect. The cross-coupling control lets you send the sum of the input and feedback from one channel to its own **LaserVerb** effect (0% cross coupling) or to the other channel's effect (100% cross coupling) or somewhere in between.

HF Damping The damping of high frequencies relative to low frequencies. When set to the highest frequency (**25088 Hz**), there is no damping and all frequencies decay at the same rate. At lower frequency settings, high frequency signal components will decay faster than low frequency components. If set too low, everything will decay almost immediately.

GateIn/Out Enables (**On**) or disables (**Off**) the gate. Not affected by Wet/Dry.

GateSCInp Select whether the gate side chain signal should use the left (**L**) channel, right (**R**) channel or the average magnitude of left and right channels (**(L+R)/2**) to control the gate.

GateSCSrc Select whether the gate side chain signal should be taken from the algorithm input or from the **LaserVerb** output.

Dly Coarse You can set the overall delay length from **0** to **5** seconds. Lengthening the delay will increase the duration or decay time of the reverb. To reduce **LaserVerb** to a simple delay, set the Contour and Feedback controls to **0**%. Use a delay of about half a second as a starting point.

Dly Fine The delay fine adjust is added to the delay coarse adjust to provide a delay resolution down to **0.1 ms**.

Spacing Determines the starting pitch of the descending buzz and how fast it descends. The Spacing parameter sets the initial separation of impulses in the impulse response and subsequent rate of increasing impulse separation. The spacing between impulses is given in samples and may be a fraction of a sample. (A sample is the time between successive digital words which is 20.8 μs or 1/48000 seconds.) For low values, the buzz starts at high frequencies and drops slowly. At high values the buzz starts at a lower pitch and drops rapidly.

Contour Controls the overall envelope shape of the reverb. When set to a high value, sounds passed through the reverb 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. At a value of around **34**, the reverb is behaving like a reverse reverb, building up to a hit. When the Contour is set to **0**, LaserVerb is reduced to a simple delay.

Gate Thresh The signal level in dB required to open the gate (or close the gate if Ducking is on).

Gate Duck 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.

Gate Time 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.

Gate Atk The time for the gate to ramp from closed to open (reverse if Ducking is on) after the signal rises above threshold.

Gate Rel The time for the gate to ramp from open to closed (reverse if Ducking is on) after the gate timer has elapsed.

GateSigDly The delay in milliseconds (ms) of the signal to be gated relative to the side chain signal. By delaying the main signal, the gate can be opened before the main signal rises above the gating threshold.

105 LasrDly<>Reverb

A configurable combination algorithm

PAUs: 2

This algorithm is one of a group of *configurable* combination algorithms—that is, there's more than one effect and you can change the sequence of those effects. With this algorithm, for example, you can have either a laser delay followed by a reverb, or vice versa.

The combination algorithms are organized in groups, with IDs predominantly in the 400s (there are a few exceptions, of course). For a description of Algorithm **105** and other combination algorithms, follow one of the links below:

Combination Algorithms on page 279

Configurable Combination Algorithms on page 289

More Combination Algorithms on page 299

106 LasrDly<>Rvrb ms

A configurable combination algorithm with some parameters expressed in absolute units

PAUs: 2

This algorithm is almost identical to **105 LasrDly<>Reverb**. The only difference is that Algorithm **106** uses absolute units for two features: milliseconds for delay line lengths, and Hz for LFO frequencies. Algorithm **105**, on the other hand, uses the values of the Tempo parameters to determine delay line lengths and LFO rates.

Delays

150 4-Tap Delay BPM
151 4-Tap Delay
610 Mn 6-TapDelayBPM
611 Mn 6-Tap Delay

A stereo four tap delay with feedback

PAUs: 1

These are simple stereo 4-tap delay algorithms with delay lengths defined in tempo beats (**150 4-Tap Delay BPM**) or in milliseconds (ms) (**151 4-Tap Delay**). 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.

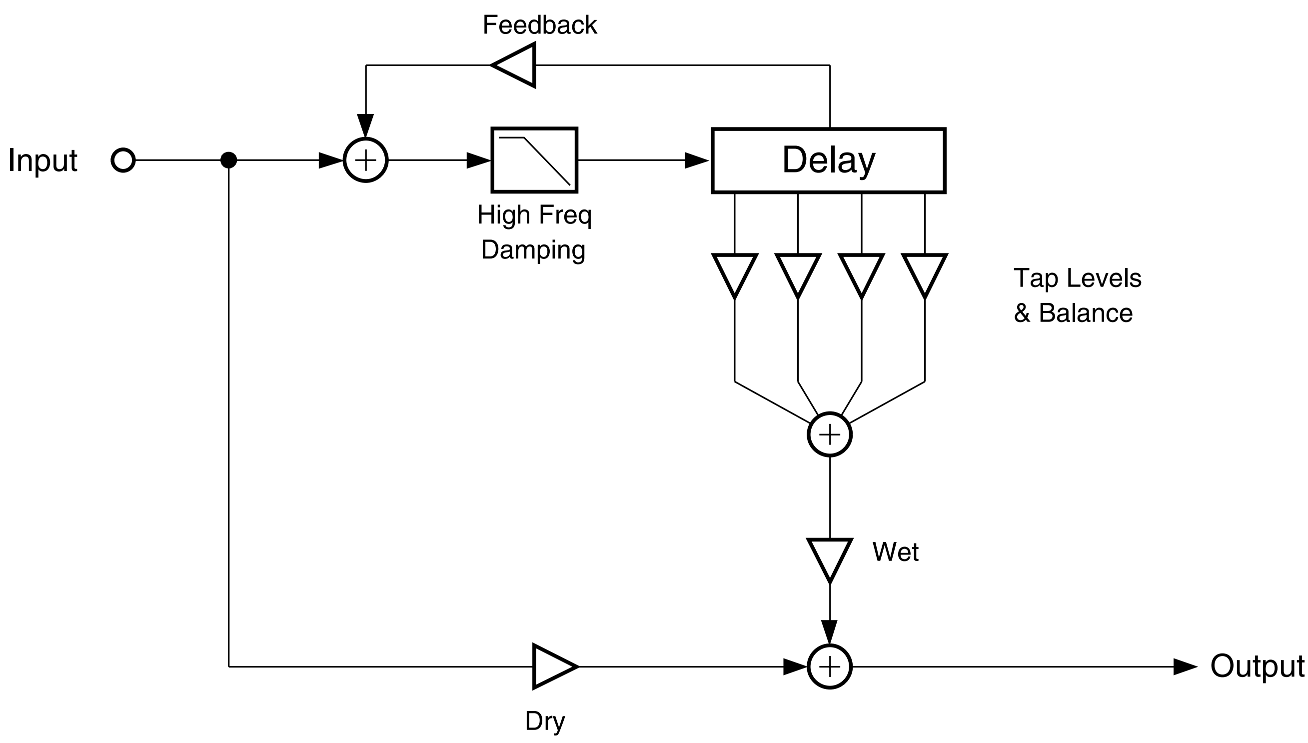


Figure 22 Left channel of 4-Tap Delay

The delay length for non-BPM tap delays is the sum of the coarse and fine parameters for the tap multiplied by the DelayScale parameter which is common to all non-BPM 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 different 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%**, your sound will be repeated indefinitely. HF Damping selectively removes high frequency content from your delayed signal and will also cause your 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 your 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.

See also the versions of these algorithms which specify delay lengths in terms of tempo and beats. They have similar names, followed by the letters **ms**; they have IDs in the **400s**.

Parameters for 4-Tap Delay

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 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.
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.
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) for the **4-Tap Delay**.
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.
Tapn 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 Tapn Fine parameters. The maximum delay length is 2.55 seconds (2550ms) for the **4-Tap Delay**.
Tap*n* Fine	A fine adjustment to the output tap delay lengths (n = 1...4). The delay resolution is 0.2 milliseconds (ms). Tapn Fine is added to Tapn Crs (coarse) to get actual delay lengths.
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 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.

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 the delay length of the tap expressed in beats (bts). The tempo alters all tap 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). IMPORTANT NOTE: KDFX has a limited amount of delay memory available (over 2.5 seconds for **4-Tap Delay 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 different 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. In this case, FXMods (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. LoopLength sets the loop delay length as a tempo beat duration. The tempo is specified with the Tempo parameter and the delay length is given in beats (bts). The delay length in seconds is calculated as beats/tempo $*$ 60 (sec/min).

Tap*n* Delay The delay lengths of the taps (n = 1...4) as tempo beat durations. The tempo is specified with the Tempo parameter and the delay length is given in beats (bts). The delay length in seconds is calculated as beats/tempo $*$ 60 (sec/min). Use the output taps to create interesting rhythmic patterns within the repeating loop.

152 8-Tap Delay BPM
153 8-Tap Delay

A stereo eight-tap delay with cross-coupled feedback

PAUs: 2

These are simple stereo 8-tap delay algorithms with delay lengths defined in tempo beats (**152 8-Tap Delay BPM**) or in milliseconds (ms) (**153 8-Tap Delay**). 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.

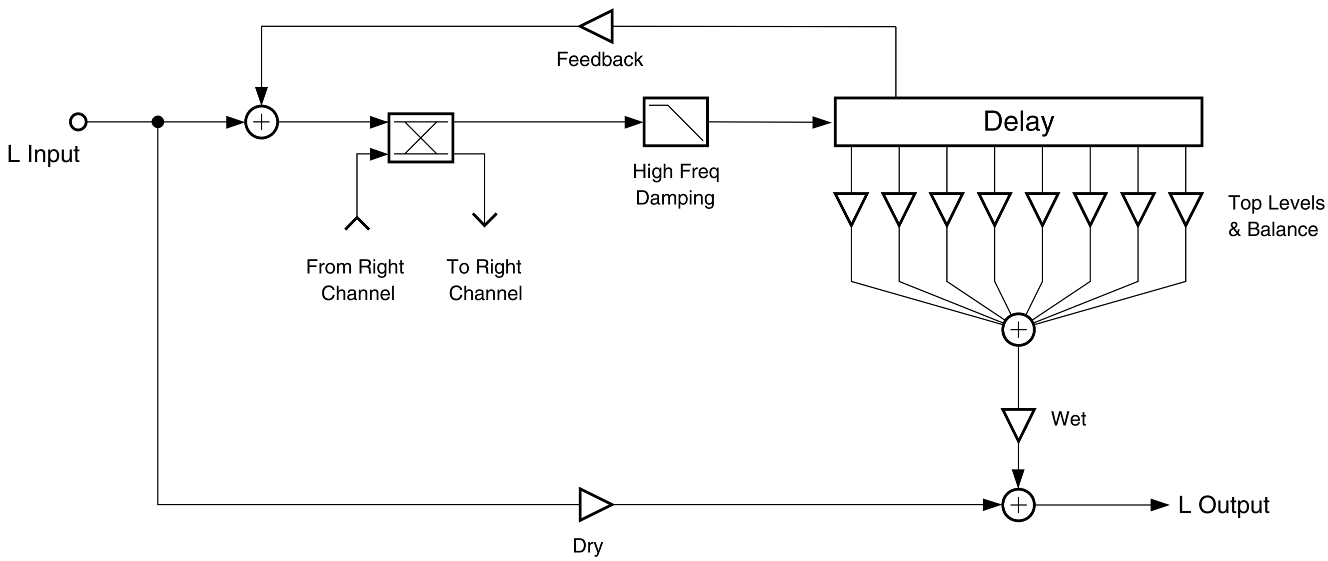


Figure 23 Left channel of 8-Tap Delay

The delay length for non-BPM tap delays is the sum of the coarse and fine parameters for the tap multiplied by the DelayScale parameter which is common to all non-BPM 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 different emphasis than others. The delay lengths for **8-Tap Delay** are in units of 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%**, your sound will be repeated indefinitely. HF Damping selectively removes high frequency content from your delayed signal and will also cause your 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 your 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.

See also the versions of these algorithms which specify delay lengths in terms of tempo and beats.

Parameters for 8-Tap Delay

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.
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) for the **8-Tap Delay**.
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.
Tapn **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) for the **8-Tap Delay**.
Tapn **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.
Tapn **Leve**l	The amount of signal from each of the taps (*n* = 1...8) which get 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 and 5, 2 and 6, 3 and 7, and 4 and 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.

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 the delay length of the tap expressed in beats (bts). The tempo alters all tap 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). IMPORTANT NOTE: KDFX has a limited amount of delay memory available (over 5 seconds for **8-Tap Delay 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 different 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. In this case, FXMods (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. LoopLength sets the loop delay length as a tempo beat duration. The tempo is specified with the Tempo parameter and the delay length is given in beats (bts). The delay length in seconds is calculated as beats/tempo $*$ 60 (sec/min).

Tap*n* Delay The delay lengths of the taps (n = 1...8) as tempo beat durations. The tempo is specified with the Tempo parameter and the delay length is given in beats (bts). The delay length in seconds is calculated as beats/tempo $*$ 60 (sec/min). Use the output taps to create interesting rhythmic patterns within the repeating loop.

154 Spectral 4-Tap
155 Spectral 6-Tap
612 Mn Spectral 4Tap

Tempo based 4 and 6 tap delays with added shapers and resonant comb filters on each tap

PAUs: 2 for **Spectral 4-Tap**
3 for **Spectral 6-Tap**

Spectral 4-Tap and **Spectral 6-Tap** are 2- and 3-PAU) tempo-based multi-tap delay effects. They are similar to a 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 are a diffuser, highpass 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 (Figure 1).

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 are also in the feedback path: highpass and lowpass. The highpass 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 appendices in the KSP8 *User’s Guide*. The spectral multi-tap shapers offer four shaping loops as opposed to eight found in the V.A.S.T. shapers, but can allow up to **6.00x** intensity (**Figure 25**). 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.

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 by adjusting Tempo in BPM. The tempo can be synced to the system clock by setting Tempo to **System**. Each tap’s delay is adjusted relative to one beat, in 1/24 beat increments. Notice that 24 is a musically useful beat division because it can divide a beat into halves, 3rds, 4ths, 6ths, 8ths, 12ths, and of course 24ths. For example, setting LoopLength to **1-12/24 bts** will put the feedback tap at 1-1/2 beats (dotted quarter note in 4/4 time) of delay making the feedback repetition occur every one and a half beats. This is equivalent to 3/4 of a second at 120 BPM. When Tempo is set to **60 BPM**, each 1/24th of a beat is equivalent to 1/24th of a second. When tempo is set to **250 BPM**, each 1/24th of a beat is equivalent to 10ms of delay.

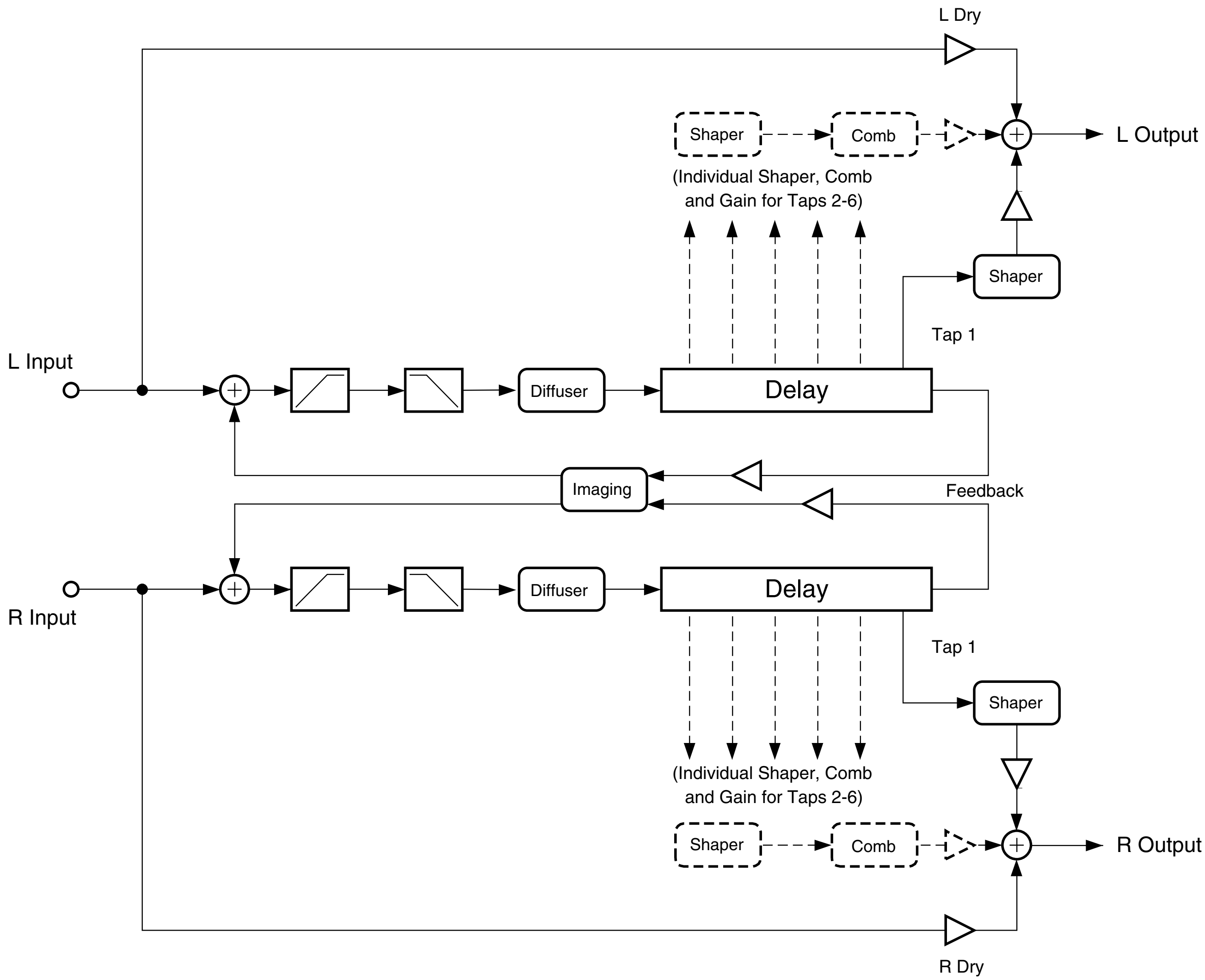


Figure 24 **Spectral 6-Tap**

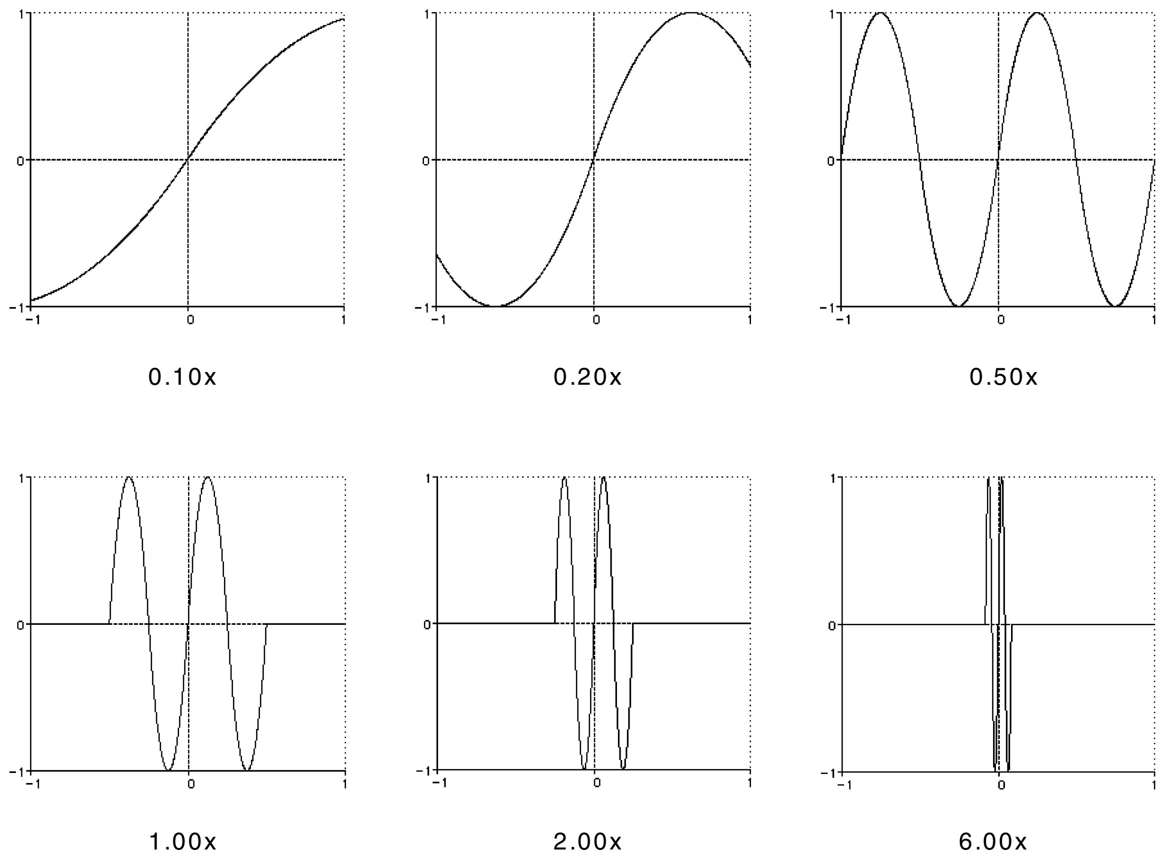


Figure 25 Various shaper curves used in the spectral multi-taps

Parameters for 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	8 to 25088 Hz	Diff Delay	0 to 20.0 ms
LF Damping	8 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 for 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	8 to 25088 Hz	Diff Delay	0 to 20.0 ms
LF Damping	8 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) lowpass 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. In this case, FXMods (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.
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 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 that each output tap is heard.
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.

156 Complex Echo
613 Mn Complex Echo

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

PAUs: 1

Complex Echo is an elaborate delay line with 3 independent output taps per channel, 2 independent feedback taps per channel, equal power output tap panning, feedback diffuser, and high frequency damping. Each channel has three output taps, each of which can 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. Feedback line 2 may also be referred to as a "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** disables the diffuser.

The delay inputs have one-pole (6dB/oct) lowpass filters controlled by the HF Damping parameter.

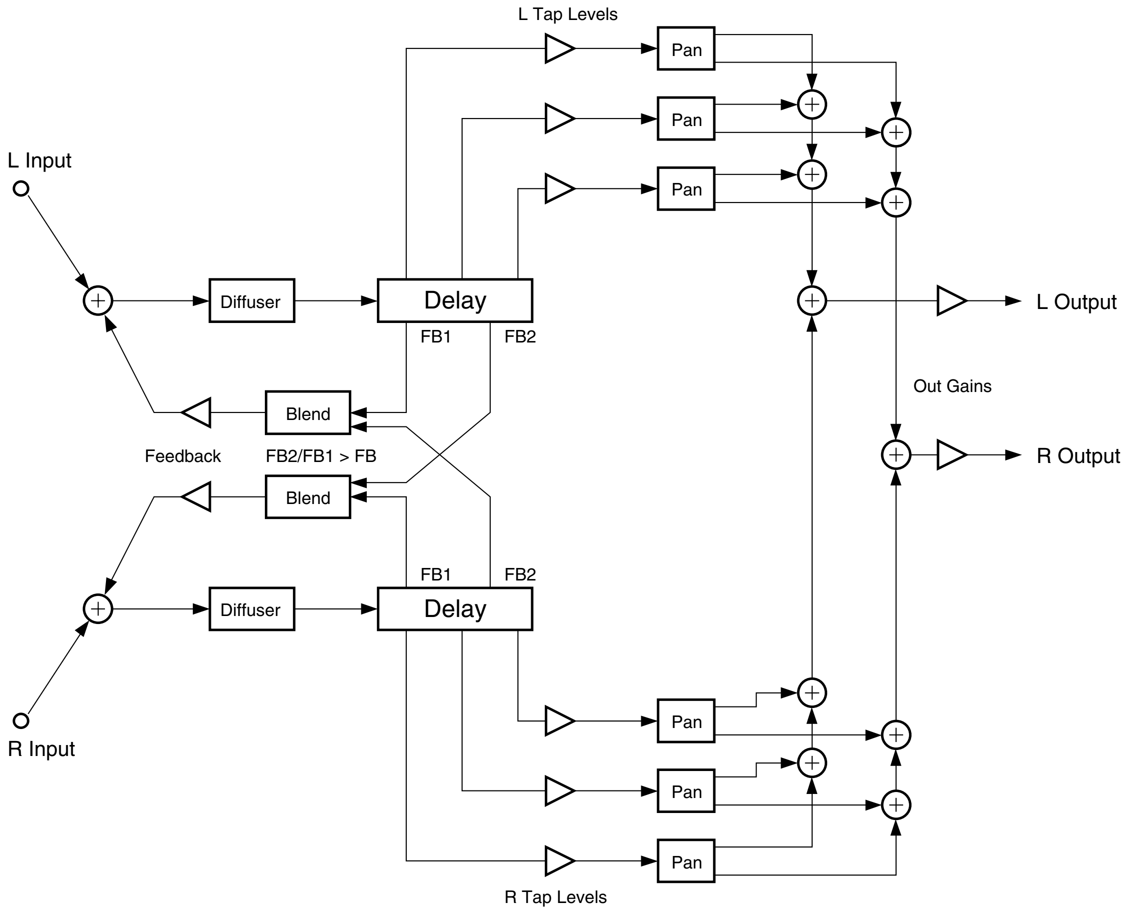


Figure 26 Signal flow of Complex Echo

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	8 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. 0% turns off feedback line 2 only allowing use of feedback line 1. **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.

Diff Dly Left and Right. 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.

Tap*n* Dly Left and Right. Adjusts the delay length of the left and right channel's three output taps.

Tap*n* Lvl Left and Right. Adjusts the listening level of the left and right channel's three output taps.

Tap*n* Pan Left and Right. 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.

168 Degen Regen LFX
169 DegenRegenBPMLF
614 Mn DegenRegenBPM
615 Mn Degen Regen

Long delay allowing loop instability

PAUs: 3 each

Degen Regen LFX starts as a simple mono delay line with feedback. However with the Fdbk Gain and Dist Drive parameters, the algorithm can be pushed hard into instability. When **Degen Regen LFX** is unstable, your sound gets a little louder on each pass through the delay line. Eventually the sound will hit digital clipping when the effects processor runs out of headroom bits. To keep this all under control, a soft-knee compressor has been included inside the delay line loop. With the compressor properly set, the sound never reaches digital clipping, but it does become more and more distorted as it gets pushed harder and harder into the compressor. To make things really nasty, there's also a distortion in the delay path. (The distortion parameters are on the PARAM4 page with the compressor parameters.)

Degen Regen LFX uses all 4 PAUs available for insert effects. With the resources of all 4 PAUs available, **Degen Regen LFX** lets you set the longest mono delay line available in KDFX which is just over 20 seconds. If you want a long delay, this is the algorithm to do it. (You don't have to over-drive the feedback or use the distortion.)

The delay has two output taps in addition to the feedback tap. Each tap may be moved along the delay line using an LFO (internal to the effects processor). The output taps have separate controls for level and panning (in the stereo configurations).

Throw a few filters into the delay line loop, and you get a pretty versatile delay line. The available filters are highpass (LF Damping), lowpass (HF Damping), bass shelf, and parametric EQ (Mid).

For details about the compressor see **331 SoftKneeCompress on page 191**. For the distortion see **300 Mono Distortion on page 153**.

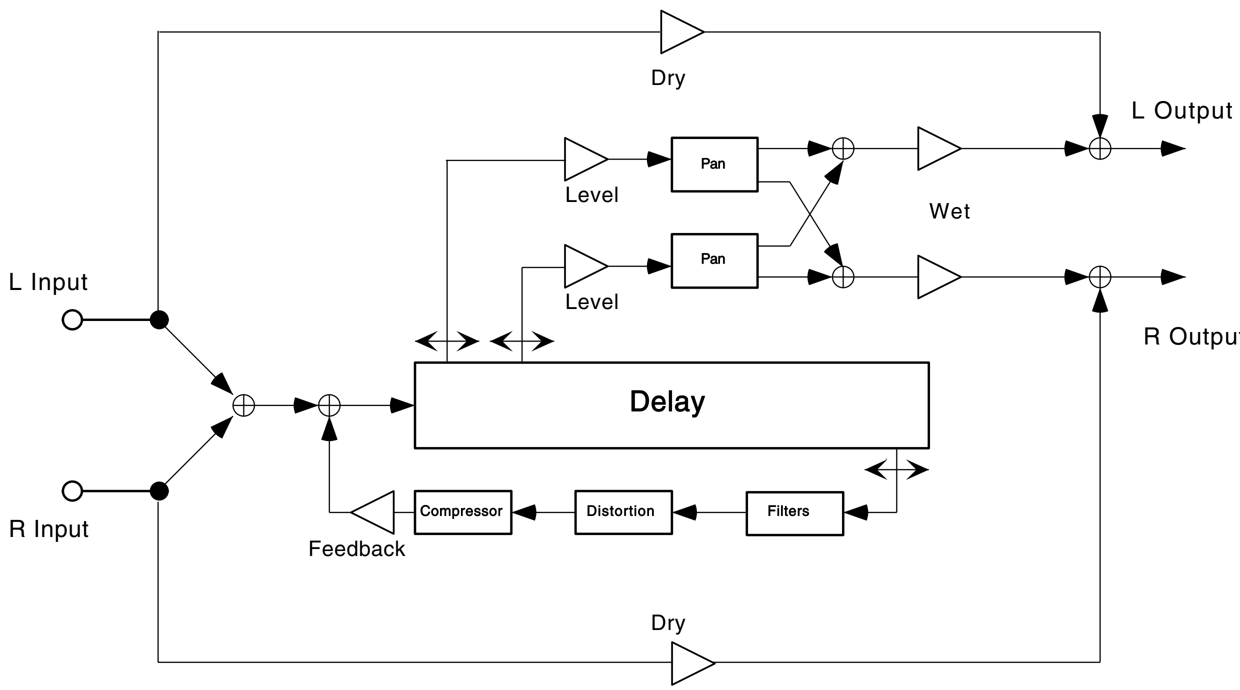


Figure 27 Degen Regen LFX

Parameters:

Page 1

Wet/Dry	-100 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
Loop Gain	Off, -79.0 to 24.0 dB	Tempo	System, 1 to 255 BPM
Loop Lvl	-100 to 100%	Send Gain	Off, -79.0 to 24.0 dB
HF Damping	8 to 25088 Hz	LF Damping	8 to 25088 Hz

Page 2 (Degen Regen LFX)

LoopLength	0.00 to 21.5 s	MidGain	-79.0 to 24.0 dB
LFO Rate	0.00 to 10.00 Hz	MidFreq	8 to 25088 Hz
Bass Gain	-79.0 to 24.0 dB	MidWidth	0.010 to 5.000 oct
Bass Freq	8 to 25088 Hz		

Page 2 (DegenRegenBPMLF)

LoopLength	0 to 32 bts	MidGain	-79.0 to 24.0 dB
LFO Period	1/24 to 32 bts	MidFreq	8 to 25088 Hz
Bass Gain	-79.0 to 24.0 dB	MidWidth	0.010 to 5.000 oct
Bass Freq	8 to 25088 Hz		

Page 3 (Degen Regen LFX)

LpLFODepth	0.0 to 230.0 ct	Tap1 Delay	0.00 to 21.5 s
LpLFOPhase	0.0 to 360.0 deg	Tap1 Level	0 to 100 %
T1LFODepth	0.0 to 230.0 ct	Tap1 Pan	-100 to 100%
T1LFOPhase	0.0 to 360.0 deg	Tap2 Delay	0.00 to 21.5 s
T2LFODepth	0.0 to 230.0 ct	Tap2 Level	0 to 100 %
T2LFOPhase	0.0 to 360.0 deg	Tap2 Pan	-100 to 100%

Page 3 (DegenRegenBPMLF)

LpLFODepth	0.0 to 230.0 ct	Tap1 Delay	0 to 32 bts
LpLFOPhase	0.0 to 360.0 deg	Tap1 Level	0 to 100 %
T1LFODepth	0.0 to 230.0 ct	Tap1 Pan	-100 to 100%
T1LFOPhase	0.0 to 360.0 deg	Tap2 Delay	0 to 32 bts
T2LFODepth	0.0 to 230.0 ct	Tap2 Level	0 to 100 %
T2LFOPhase	0.0 to 360.0 deg	Tap2 Pan	-100 to 100%

Page 4

Comp Atk	0.0 to 228.0 ms	Comp Ratio	1.0:1 to 100.0:1, Inf:1
Comp Rel	0 to 3000 ms	Comp Thres	-79.0 to 0.0 dB
CompSmooth	0.0 to 228.0 ms	Dist Drive	0 to 96 dB
		DistWarmth	8 to 25088 Hz
		||||||||||||||||||||||||||||	Reduction
	-dB 40 20 12	8 6 4 2 0	

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.

Send Gain The input gain or amplitude to the **Degen Regen LFX** delay loop.

Loop Gain Controls the signal level of the signal which is fed back to the input of the delay line. If other elements of **Degen Regen LFX** were removed (set flat), then Loop Gain would cause the algorithm to become unstable above 0 dB. However other parameters interact resulting in a more complex gain structure. See also Loop Lvl.

Loop Lvl A convenience parameter which may be used to reduce the Fdbk Gain feedback strength. It may be helpful if you are used to dealing with feedback as a linear (percent) control. At **100**%, the feedback strength is as you have it set with Loop Gain. Lower levels reduce the feedback signal, so at **50**% the feedback signal is reduced by -6 dB from the selected Loop Gain level. Negative values polarity invert the feedback loop signal.

Tempo In **DegenRegenBPMLF**, Tempo is the 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. In this case, FXMods (FUNs, LFOs, ASRs etc.) will have no effect on the Tempo parameter.

LF Damping The -3 dB frequency in Hz of a one-pole highpass filter (6 dB/octave) placed in the feedback path of the delay line. The signal does not go through the filter the first time through the delay line. Multiple passes through the feedback will cause the signal to become more and more bright (removing low frequencies).

HF Damping The -3 dB frequency in Hz of a one-pole lowpass filter (-6 dB/octave) placed in the feedback path of the delay line. The signal does not go through the filter the first time through the delay line. Multiple passes through the feedback will cause the signal to become more and more dull.

LoopLength The delay length of the feedback tap. If feedback is turned up from **0%**, this parameter sets the repeating delay loop length. For **Degen Regen LFX**, the loop length is specified in seconds. In **DegenRegenBPMLF**, 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 from beats as T = (beats/tempo) $*$ 60.

LFO Rate The feedback tap and the output taps lengths can be modulated with an LFO internal to the effects processor. The rate at which the tap positions move are tied to a common rate control which is expressed in Hz. The LFO Rate control is specific to **Degen Regen LFX**. The depth of modulation is specified by the LpLFODepth parameter.

LFO Period The feedback tap and the output taps lengths can be modulated with an LFO internal to the effects processor. The rate at which the tap positions move are tied to a common period control (time for one complete cycle) which is expressed in beats. The LFO Period control is specific to **DegenRegenBPMLF**. The depth of modulation is specified by the LpLFODepth parameter. Frequency in Hz can be calculated from the period in beats as F = tempo/(beats $*$ 60). Since this moving delay tap is part of the feedback path through the delay, subsequent passes of the signal through the delay may result in some strange pitch modulations. It is possible to set LFO Period with LoopLength so that alternate passes through the loop detune then retune the signal (for example, set the LFO period to double the LoopLength). The maximum pitch shift up is not identical to the maximum pitch shift down, so the alternating detune/retune effect is not perfect.

Bass Gain The amount of boost or cut in dB that the bass shelving filter should apply to the low frequency signal components. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the bass signal below the specified frequency. Negative values cut the bass signal below the specified frequency. Since the filters are in the delay feedback loop, the cut or boost is cumulative on each pass the sound makes through the loop.

Bass Freq The center frequency of the bass shelving filter in intervals of one semitone.

Mid Gain The amount of boost or cut in dB that the parametric filter should apply to the specified signal band. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the signal at the specified frequency. Negative values cut the signal at the specified frequency. Since the filters are in the delay feedback loop, the cut or boost is cumulative on each pass the sound makes through the loop.

Mid Freq The center frequency of the parametric EQ in intervals of one semitone. The boost or cut will be a maximum at this frequency.

Mid Width The bandwidth of the parametric EQ may be adjusted. You specify the bandwidth in octaves. Small values result in a very narrow (high-Q) filter response. Large values result in a very broad response.

LpLFODepth The feedback (loop) delay tap will have its position modulated by an LFO (internal to the FX processor) if the LpLFODepth parameter is non-zero. A moving tap on a delay line will result in a pitch shift, and LpLFODepth sets the maximum pitch shift (up and down) in cents.

LpLFOPhase Specifies the phase angle of the feedback (loop) LFO relative to the output tap LFOs and the system (or MIDI) 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 (or MIDI) tempo clock is turned on (**DegenRegenBPMLF** only), the LFOs are synchronized to the clock with absolute phase.

TnLFODepth The output delay taps (1 and 2) will have their positions modulated by an LFO (internal to the FX processor) if the TnLFODepth parameter is non-zero. A moving tap on a delay line will result in a pitch shift, and TnLFODepth sets the maximum pitch shift (up and down) in cents.

TnLFOPhase Specifies the phase angle of the output LFO tap (1 or 2) relative to the other output LFO tap, the feedback (loop) LFO tap, and the system (or MIDI) 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 (or MIDI) tempo clock is turned on (**DegenRegenBPMLF** only), the LFOs are synchronized to the clock with absolute phase.

Tapn Delay The delay length of the output tap 1 or 2. For **Degen Regen LFX**, the tap length is specified in seconds. In **DegenRegenBPMLF**, the tap length is specified as a fraction or multiple of the tempo, in “beats.” The length of a delay tap in seconds can be calculated from beats as $T = (\text{beats}/\text{tempo}) * 60$.

Tapn Level The level of the output tap 1 or 2 expressed as a percent.

Tapn Pan The output taps 1 and 2 are mono sources that can be panned to the left or right output channels. A pan setting of **-100%** is fully left while **100%** is fully right.

Comp Atk The time for the compressor to start to cut in when there is an increase in signal level (attack) above the threshold.

Comp Rel The time for the compressor to stop compressing when there is a reduction in signal level (release) from a signal level above the threshold.

CompSmooth A lowpass filter in the compressor control signal path. It is intended to smooth the output of the expander’s envelope detector. Smoothing will affect the attack or release times when the smoothing time is longer than one of the other times.

Comp Ratio The compression ratio. High ratios are highly compressed; low ratios are moderately compressed.

Comp Thres The threshold level in dBFS (decibels relative to full scale) above which the signal begins to be compressed.

Dist Drive Applies a boost to the feedback signal to overdrive the distortion algorithm. When overdriven, the distortion algorithm will soft-clip the signal. Since distortion drive will make your signal very loud, you may have to reduce the feedback amount or turn on the compressor as the drive is increased.

DistWarmth A lowpass filter in the distortion control path. This filter may be used to reduce some of the harshness of some distortion settings without reducing the bandwidth of the signal.

172 Switch Loops

Looped delay lines with input switching

PAUs: 2

Switch Loops allows you to run up to four parallel recirculating delay lines of different lengths, switching which delay line(s) are receiving the input signal at a given moment. The stereo input is summed to mono and sent to any of the four delay lines. You can select which delay lines are receiving input with the DlySelect parameters.

The gain in decibels of each of the four delays can be set individually. The amount of feedback to apply to each delay is set with a DecayRate parameter. The DecayRate controls how many decibels the signal will be reduced for every second the signal is recirculating in the delay.

The length of the delays are set based on tempo (system tempo or set locally) and duration in beats. Assuming a 4/4 time signature with tempo beats on the quarter note, 8/24 bts is an eighth triplet (8/24 equals 1/3 of a quarter note), 12/24 bts is an eighth, 16/24 bts is a quarter triplet, and 1 bts is a quarter note duration. Dividing the quarter note into 24ths, allows delay lengths based on the most common note lengths. To determine a delay length in seconds, divide the length of the delay (in beats) by the tempo and multiply by 60 seconds/minute (beats/tempo * 60).

Switch Loops has a few more specialized parameters. HF Damping controls a one pole lowpass filter on each of the delay lines. Max Fdbk overrides all of the DecayRate parameters and prevents the signals in the delay lines from decaying at all. Fdbk Kill will override the DecayRate parameters and the Max Fdbk parameter by completely turning of the feedback for all the delays. Fdbk Kill stops all the delay line recirculation.

The outputs of all the delay lines are summed, and the output gain is applied to the mono result which can be panned between the two output channels.

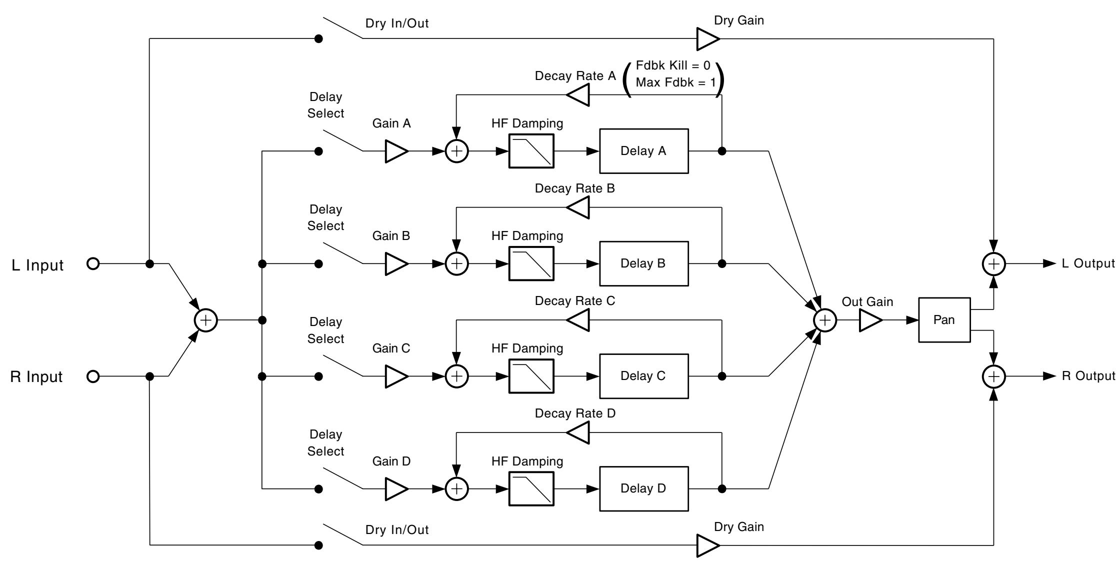


Figure 28 **Switch Loops**

Parameters:

Page 1

Dry In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
Dry Gain	Off, -79.0 to 24.0 dB	Tempo	System, 1 to 255 BPM
Fdbk Kill	On or Off	Pan	-100 to 100 %
Max Fdbk	On or Off	HF Damping	8 to 25088 Hz

Page 2

DlySelect1	Off, A, B, C, D		
DlySelect2	Off, A, B, C, D		
DlySelect3	Off, A, B, C, D		
DlySelect4	Off, A, B, C, D		

Page 3

Dly Len A	0 to 32 bts	Dly Len C	0 to 32 bts
DecayRateA	0.0 to 230.0 dB/s	DecayRateC	0.0 to 230.0 dB/s
Gain A	Off, -79.0 to 24.0 dB	Gain C	Off, -79.0 to 24.0 dB
Dly Len B	0 to 32 bts	Dly Len D	0 to 32 bts
DecayRateB	0.0 to 230.0 dB/s	DecayRateD	0.0 to 230.0 dB/s
Gain B	Off, -79.0 to 24.0 dB	Gain D	Off, -79.0 to 24.0 dB

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

Dry In/Out If set to **In**, Dry In/Out allows the dry input signal to be added to the final algorithm output.

Dry Gain If Dry In/Out is **In**, then Dry Gain controls the level of the dry input signal that is summed to the final algorithm output.

Fdbk Kill Forces the delay recirculation of all delay lines to stop by turning off the delay line feedback. Fdbk Kill provides a quick way to silence the algorithm to start over with new sounds in the delays. Fdbk Kill overrides the Max Fdbk and DecayRate parameters.

Max Fdbk Prevents the recirculating delay lines from decaying by turning the delay line feedback fully on. Max Fdbk overrides the DecayRate parameters, but does not function when Fdbk Kill is **On**.

Tempo Tempo is the 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. In this case, FXMods (FUNs, LFOs, ASRs etc.) will have no effect on the Tempo parameter.

Pan The summed mono signal from the delay lines may be panned between left and right output channels. **-100**% is panned fully left, **0**% is centered, and **100**% is fully right.

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

DlySelect*n*	You select which delay lines (A, B, C, or D) receive the mono input signal with the DlySelect (1, 2, 3, or 4) parameters. Since there are four delay lines, you can turn on none, 1, 2, 3, or 4 of the delay lines. All four of the DlySelect parameters are equivalent—it doesn't matter which you use. If you turn on a particular delay line in more than one DlySelect parameter, it's the same as turning it on in just one DlySelect parameter.
Dly Len *n*	The delay length of the delay line n (n = A, B, C, or D). If the DecayRate for the delay is low or Max Fdbk is **On**, this parameter sets the repeating delay loop length for this delay. The delay length is specified as a fraction or multiple of the tempo, in "beats." The length of a delay loop in seconds can be calculated from beats as T = (beats/tempo) ∗ 60.
DecayRate*n*	The rate at which the delay line n (n = A, B, C, or D) will decay or reduce in level. DecayRate controls a feedback level which is calculated based on DecayRate and Dly Len. By basing the feedback gain on DecayRate, all four of the delay lines can decay at the same rate in spite of differing delay lengths. DecayRate is expressed as decibels of signal reduction per second.
Gain *n*	The level of the delay n (n = A, B, C, or D) output tap expressed in decibels.

173 3 Band Delay
616 Mn 3 Band Delay

Three delays operating on selectable frequency bands

PAUs: 2 for **3 Band Delay** and 1 for **Mn 3 Band Delay**

3 Band Delay uses a band splitting filter to divide the input signal into 3 frequency bands. The filtered bands of the signal are then passed through 3 parallel delay lines. You can select the frequencies at which the bands are split. You can select which frequency band (Low, Mid, or High) gets passed through a particular delay line. You can choose to pass the same band through all 3 delay lines, or you can send each band through its own delay line.

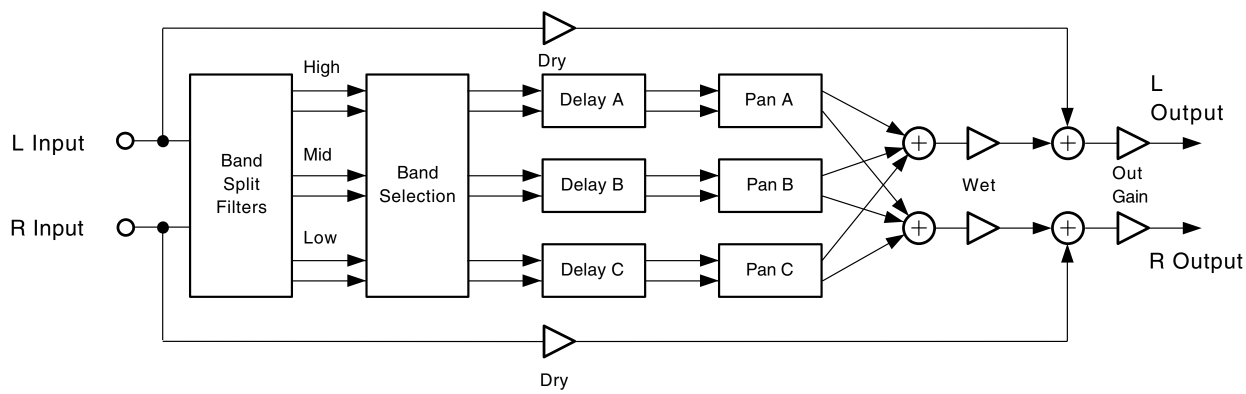


Figure 29 Stereo 3 Band Delay

Delay line lengths are tempo based. Tempo is expressed in beats per minute (BPM) and the delay lengths are expressed as the number of beats (bts) at the tempo. The delay length beats are 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.

For the stereo version of **3 Band Delay**, the outputs of each stereo delay line can be panned to the final stereo output. The full stereo field is moved with this panner, and the width of the stereo field can be reduced with the Width parameter.

Parameters:

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
		Tempo	System, 1 to 255 BPM
		Crossover1	8 to 25088 Hz
		Crossover2	8 to 25088 Hz

Page 2

BandSelctA	Low, Mid, or High	BandSelctB	Low, Mid, or High
DelayLenA	0 to 6 bts	DelayLenB	0 to 6 bts
DelayLvlA	0 to 100%	DelayLvlB	0 to 100%
PanA	-100 to 100%	PanB	-100 to 100%
WidthA	-100 to 100%	WidthB	-100 to 100%

Page 3

BandSelctC	Low, Mid, or High		
DelayLenC	0 to 6 bts		
DelayLvlC	0 to 100%		
PanC	-100 to 100%		
WidthC	-100 to 100%		

Wet/Dry The relative amount of input signal and delay signal that is to appear at the final effect output mix. At **0**% only the dry input is heard; at **100**% only the delayed (wet) signal is heard.

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

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 system tempo or to incoming MIDI clocks. In this case, FXMods (FUNs, LFOs, ASRs etc.) will have no effect on the Tempo parameter.

CrossoverN The Crossover parameters (1 and 2) set the frequencies which divide the three frequency bands. The two Crossover parameters are interchangeable, so either may contain the higher frequency value.

BandSelect Selects which of the three frequency bands (Low, Mid, or High) is to pass through the particular delay (A, B, or C).

DelayLen The delay lengths (for delays A, B, and C) 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.

DelayLvl The amount of signal from the delays (A, B, and C) which gets sent to the final wet/dry mix.

Pan Each stereo delay (A, B, and C) has a stereo panner. The stereo image is maintained but is "tilted" to the right or left. When Pan is set to **0**% there is no change to the signal, while at **100**% both input signals are sent to the right channel. At **50**%, what had been hard left at the input will now be in the center, and what had been in the center at the input will now be halfway between center and right. Negative values tilt the signal to the left.

Width The stereo image width of each panner can be controlled with the Width parameter. **100**% is full stereo width, so the left input is sent to the left output channel while the right input is sent to the right output channel. At **0**% Width, the stereo width is narrowed to mono (left and right are summed), and the panner behaves like a mono-to-stereo panner. Negative Width settings swap the left and right channels.

174 Gated Delay
617 Mn Gated Delay

Delay with gating and ducking

PAUs: 2 each

Gated Delay is a delay with feedback which has its output and feedback controlled by a gate. The gate side-chain is the same as in Algorithm **345 Gate w/SC EQ LFX**, except this algorithm does not include side-chain EQ filtering. Gating a delay is not particularly interesting until the sense of the gate is reversed by turning on the Ducking parameter. With ducking, the gate passes signal only when the side-chain input signal is below the gate threshold.

The with ducking turned on, **Gated Delay** could also be called the "Monster Truck Effect." Set Wet/Dry to about **50%**. What happens is that as long as a signal is coming in that is above the gate threshold, all you will hear is the dry signal. When the input signal stops, then the gate opens up, and suddenly the delay takes over. For example, if you sent the speech phrase "Welcome to the monster truck rally" through the effect, what you would hear is "Welcome to the monster truck rally, rally, rally…"

Of course to really get the desired effect, you may need to adjust the gate, the delay and the feedback. See Algorithm **345 Gate w/SC EQ LFX** for details on controlling the gate. The loop delay length (for feedback) is the same for both left and right channels to keep timing constant. The output delay lengths may be different for the two channels to give a syncopated or "ping-pong" feel. Of course **Mn Gated Delay** is a pure mono effect, so it has no left/right control. The Feedback parameter controls how long it will take for the looping delay sound to decay.

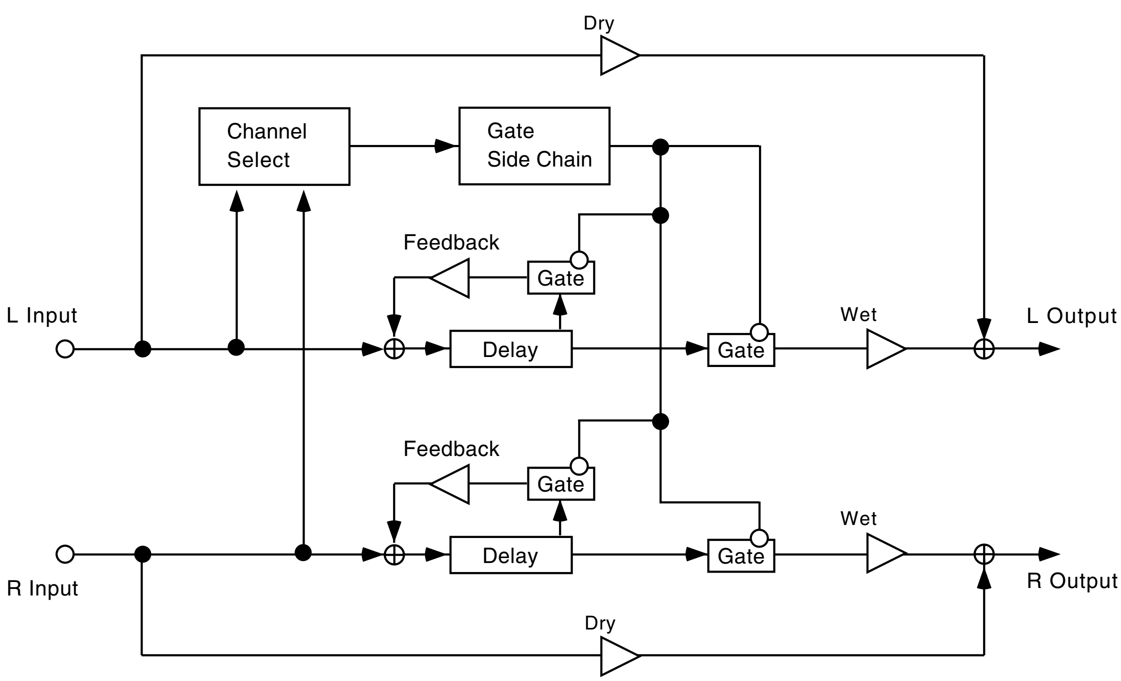


Figure 30 Block diagram of Gated Delay

Parameters:

Page 1

Wet/Dry	0 to 100%	Out Gain	Off, -79.0 to 24.0 dB
Feedback	0 to 100%		

Page 2

Loop Crs	0 to 5100 ms		
Loop Fine	-20.0 to 20.0 ms		
L Dly Crs	0 to 5100 ms	R Dly Crs	0 to 5100 ms
L Dly Fine	-20.0 to 20.0 ms	R Dly Fine	-20.0 to 20.0 ms

Page 3

<table>
<tr><td>Threshold</td><td>-79.0 to 24.0 dB</td><td>Gate Time</td><td>0 to 3000 ms</td></tr>
<tr><td>Ducking</td><td>On or Off</td><td>Atk Time</td><td>0.0 to 228.0 ms</td></tr>
<tr><td>Retrigger</td><td>On or Off</td><td>Rel Time</td><td>0 to 3000 ms</td></tr>
<tr><td>Env Time</td><td>0 to 3000 ms</td><td></td><td></td></tr>
<tr><td></td><td></td><td>||||||||||||||||||||||||||||</td><td>Reduction</td></tr>
<tr><td></td><td>-dB 60 40 ✻ 16</td><td>✻ 8 4</td><td>0</td></tr>
</table>

Wet/Dry — The amount of gated delay signal (wet) relative to the input dry signal to send to the output.

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

Feedback — The amount of the loop delay signal to add to the input of the delay. Feedback controls how long the looped delay takes to decay.

Loop Crs/Fine — The length of the delay loop in milliseconds (ms). The loop time controls the duration of the repeated “snippet” of sound.

Dly Crs/Fin — The length of the delay for the final output taps in milliseconds (ms). The stereo version of **Gated Delay** has separate lengths for the left and right channels.

Threshold — The signal level in dB required to open the gate (or close the gate if Ducking is on).

Ducking — 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. This effect is most interesting when Ducking is on.

Retrigger — If Retrigger is **On**, the gate timer is constantly restarted (retriggered) as long as the side chain signal is above the threshold. The gate then remains open (assuming Ducking is **Off**) until the signal falls below the threshold and the gate timer has elapsed. If Retrigger is **Off**, then the gate timer starts at the moment the signal rises above the threshold and the gate closes after the timer elapses, whether or not the signal is still above threshold. With Retrigger off, use the Env Time to control how fast the side chain signal envelope drops below the threshold. With Retrigger set to **Off**, the side chain envelope must fall below threshold before the gate can open again.

Env Time	Envelope time is for use when Retrigger is set to **Off**. The envelope time controls the time for the side chain signal envelope to drop below the threshold. At short times, the gate can reopen rapidly after it has closed, and you may find the gate opening unexpectedly due to an amplitude modulation of the side chain signal. For long times, the gate will remain closed until the envelope has a chance to fall, and you may miss gating events.
Gate Time	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. If Retrigger is **On**, the gate timer is continually reset while the side chain signal is above the threshold.
Atk Time	The time for the gate to ramp from closed to open (reverse if Ducking is **On**) after the signal rises above threshold.
Rel Time	The time for the gate to ramp from open to closed (reverse if Ducking is **On**) after the gate timer has elapsed.

190 Moving Delay

Generic stereo moving delay lines

PAUs: 1

Moving Delay is identical to Algorithm **191 Dual MovDelay** except that the algorithm now has stereo controls rather than dual mono. This means all the controls except L Pan and R Pan are no longer dual left and right but are ganged into single controls controlling both left and right channels.

Parameters:

Page 1

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

Page 2

Delay	0.0 to 1000.0 ms		
LFO Mode	ChorTri, ChorTrap, Delay, Flange		
LFO Rate	0.00 to 10.00 Hz		
LFO Depth	0.0 to 200.0 %		
Feedback	-100 to 100 %		
HF Damping	8 to 25088 Hz		

Wet/Dry 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 input (dry) signal. When set to **100%**, the output is all wet.

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

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

Delay Adjusts the delay time for the moving delay circuits, which is the center of LFO excursion.

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 trapezoidal 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.

LFO Rate Adjusts the LFO speed for the moving delay circuits.

LFO Depth 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.

Feedback Adjusts the level of the moving delay circuits' output signal fed back into their own inputs. Negative values polarity invert the feedback signal.

HF Damping Adjusts the cutoff frequency of a 1-pole (6dB/oct) lowpass filter in the moving delay circuits.

191 Dual MovDelay
192 Dual MvDly+MvDly

Generic dual mono moving delay lines

PAUs: 1 for **Dual MovDelay**
2 for **Dual MvDly+MvDly**

Each of these algorithms offers generic moving delay lines in a dual mono configuration. 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 trapezoidal pitch shifting. Refer to **Choruses on page 98** for more information on these envelope shapes.

The value functions much like a wet/dry mix where **0%** means that only the algorithm input dry signal is fed into effect B (putting the effects in parallel), and **100%** means only the output of effect A is fed into effect B (putting the effects in series). See **Figure 140** (**page 279**) for signal flow of **Chorus+4Tap** as an example.

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.

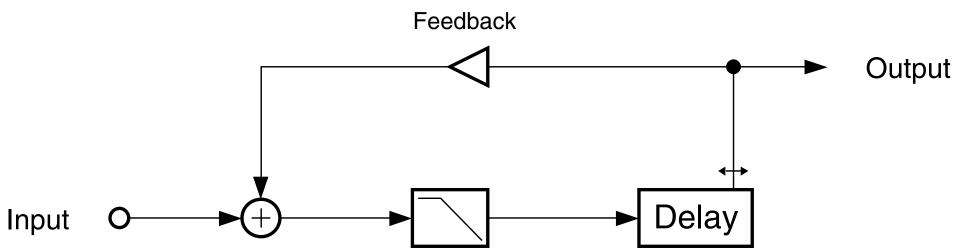


Figure 31 Generic monaural moving delay line

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.

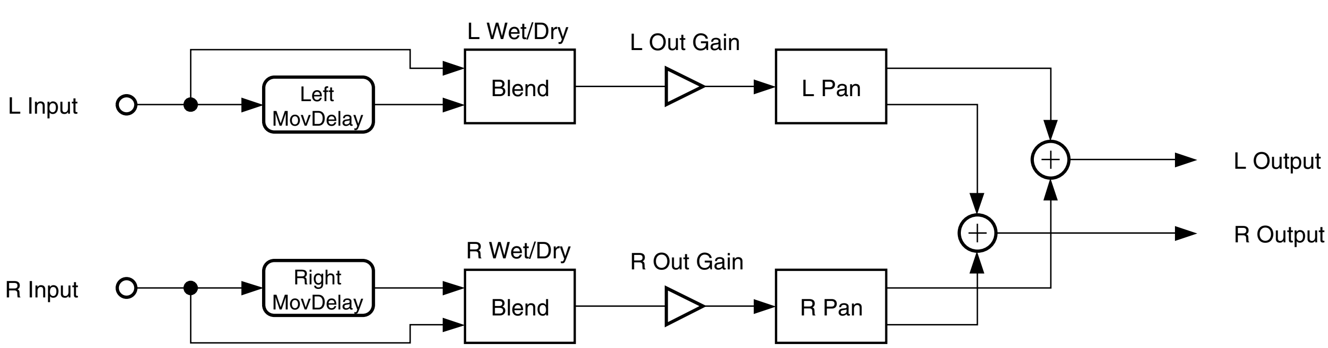


Figure 32 Signal flow of Dual MovDelay

In **Dual MvDly+MvDly**, 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.

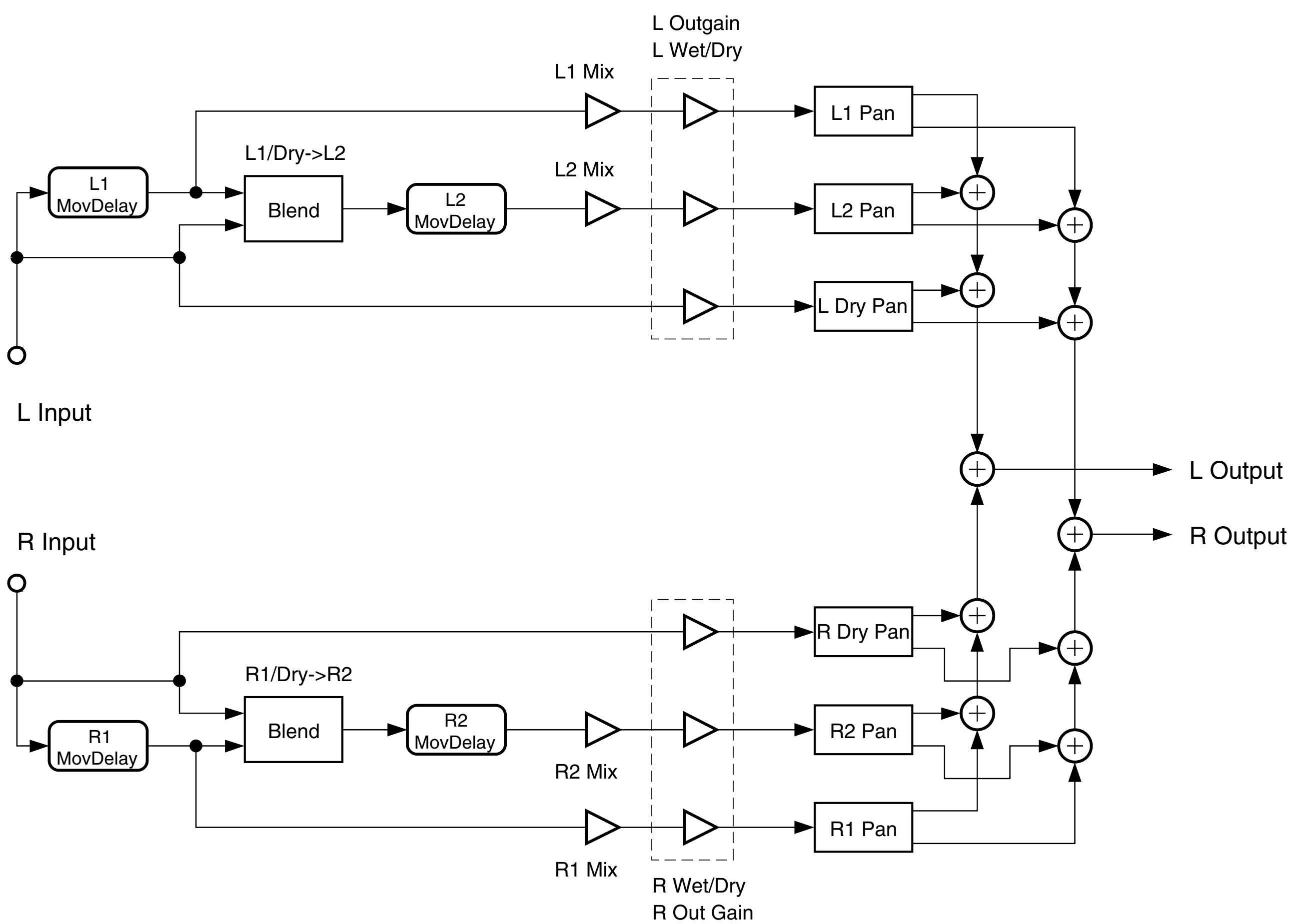


Figure 33 Signal flow of Dual MvDly+MvDly

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	8 to 25088 Hz	R HF Damp	8 to 25088 Hz

Parameters (Dual MvDly+MvDly):

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	8 to 25088 Hz	L2 HF Damp	8 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	8 to 25088 Hz	R2 HF Damp	8 to 25088 Hz

L Wet/Dry **R Wet/Dry**	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.
L Out Gain **R Out Gain**	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 **Dual MvDly+MvDly**. 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 trapezoidal 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.

Choruses

204 Dual Chorus 1 LFX
205 Dual Chorus 2 LFX

One- and three-tap stereo and dual mono choruses

PAUs: 1 for **Dual Chorus 1 LFX**
2 for **Dual Chorus 2 LFX**

Chorusing is an effect that 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.

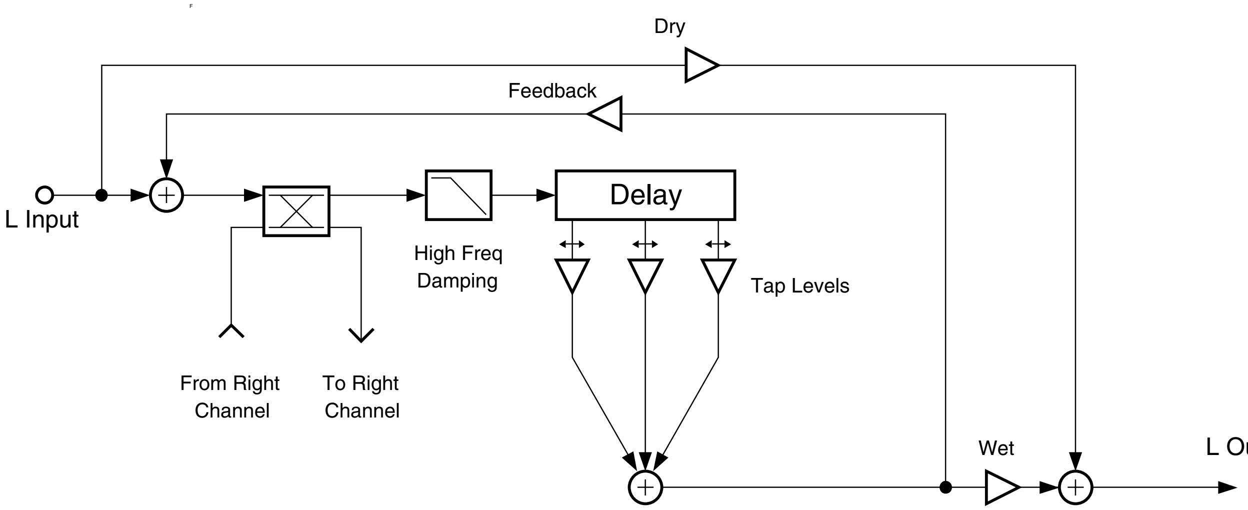


Figure 34 **Block diagram of left channel of Chorus 2 (right channel is the same)**

Chorus 2 is a two-PAU multi-tapped delay (3 taps) based chorus effect with cross-coupling and individual output tap panning. **Figure 34** is a simplified block diagram of the left channel of **Chorus 2**.

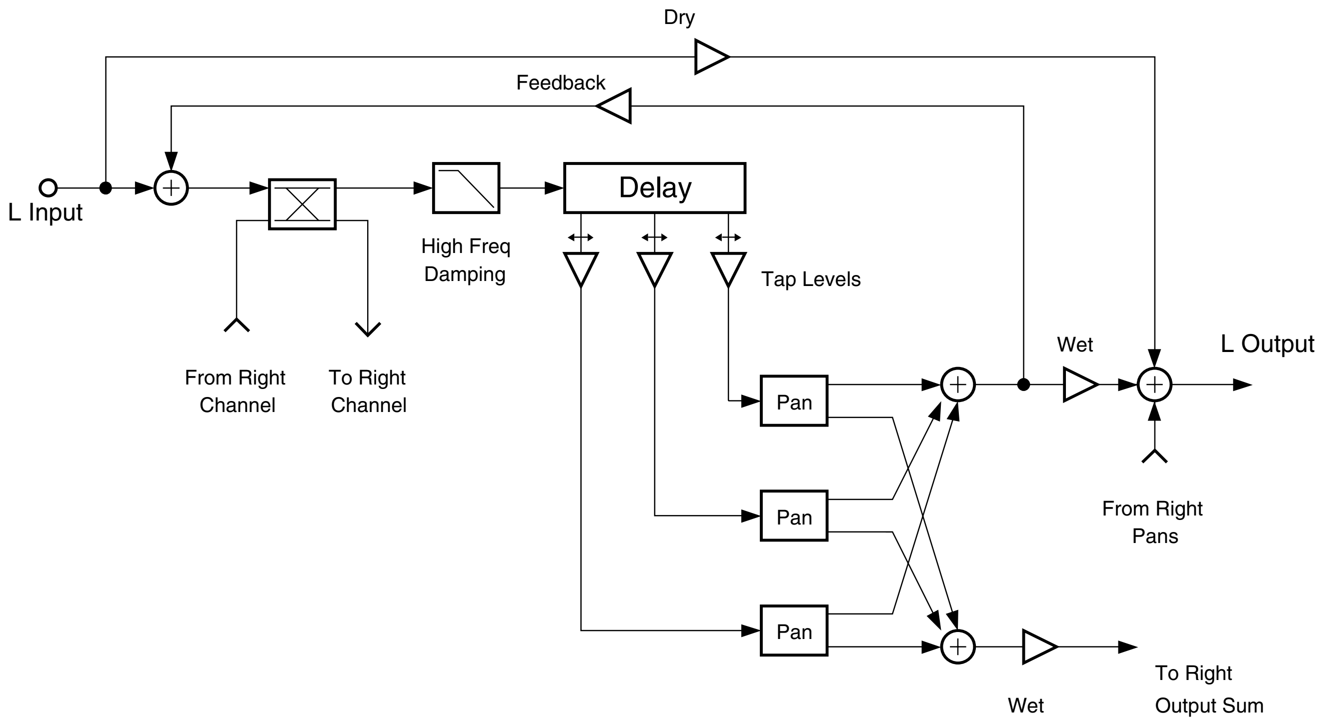


Figure 35 Block diagram of left channel of Dual Chorus 2 LFX (right channel is similar)

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

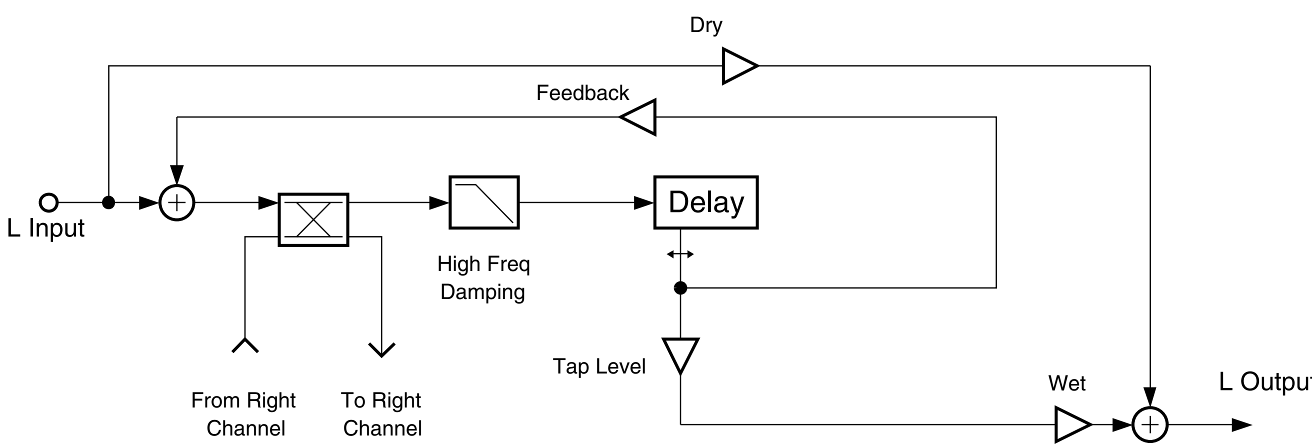


Figure 36 Block diagram of left channel of Chorus 1 (right channel is the same)

Chorus 1 uses one PAU and has one delay tap. Figure 36 is a simplified block diagram of its left channel.

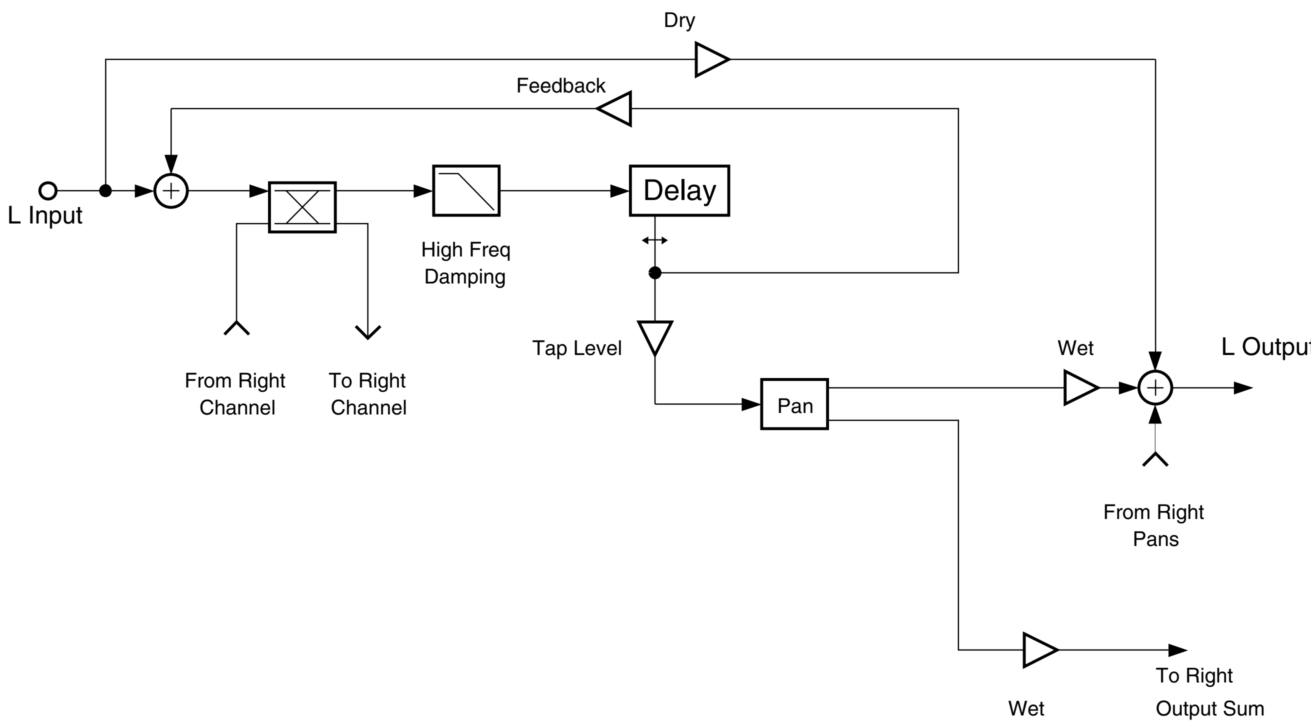


Figure 37 Block diagram of left channel of Dual Chorus 1 LFX (right channel is similar)

The left and right channels pass through their own chorus blocks and there may be cross-coupling between the channels. For **Chorus 2** and **Dual Chorus 2 LFX**, each channel has three moving taps which are summed, while **Chorus 1** and **Dual Chorus 1 LFX** have one moving tap for both channels. For 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 back 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 crossfade. Note that the Output Gain parameters affects 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 *and* below the original pitch. The depth units are in cents, and there are 100 cents in 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.

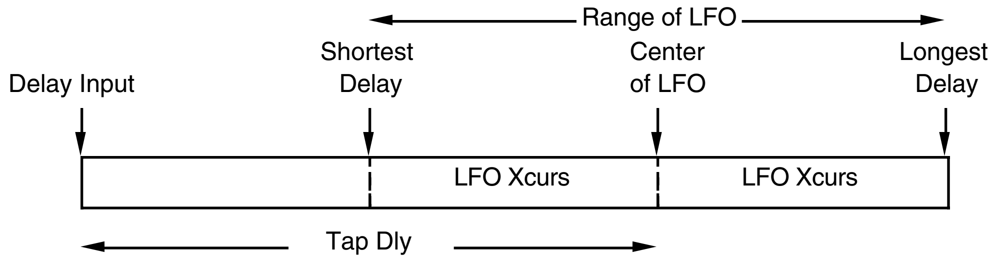


Figure 38 Delay for a single LFO

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, or in other words the delay to the center of the LFO excursion. The center of LFO excursion can not move smoothly. Changing the center of LFO excursion creates discontinuities in the tapped signal. 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. Modulating Tap Dly will produce unwanted zipper noise. 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 trapezoidal 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 Pitch Env is set to **Trapzoid**.

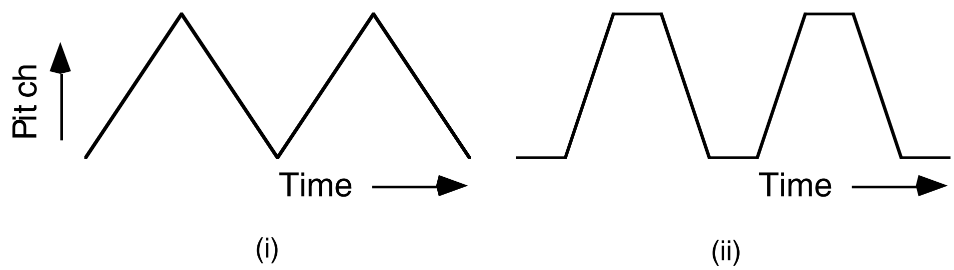


Figure 39 Pitch envelopes (i) Triangle and (ii) Trapzoid

Parameters for Dual Chorus 1 LFX

Page 1

Wet/Dry	-100 to 100 %wet	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 for Dual Chorus 2 LFX

Page 1

Wet/Dry	-100 to 100 %wet	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 is to appear 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 trapezoidal (**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 LFX** and **Dual Chorus 2 LFX** 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 depths of detuning of the LFO modulated delay lines. The depth controls range from **0** to **50 cents**. (There are 100 cents in 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 (with signal 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 **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 **LFOn LRPhs**) In the stereo **Chorus 1** and **Chorus 2**, the relative phases of the LFOs for the left and right channels may be adjusted.

Flangers

225 Flanger 1
226 Flanger 2
625 Mn Flanger 1

Multi-tap flangers

PAUs: 1 for **Flanger 1**
2 for **Flanger 2**

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

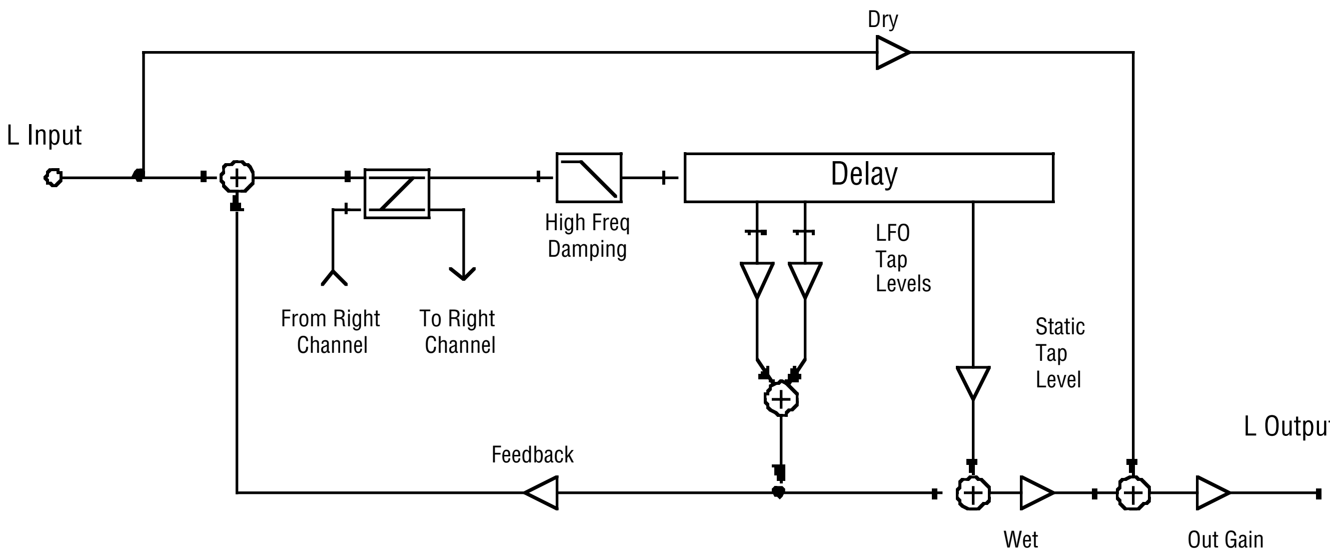


Figure 40 Simplified block diagram of the left channel of Flanger 1 (right channel is similar)

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

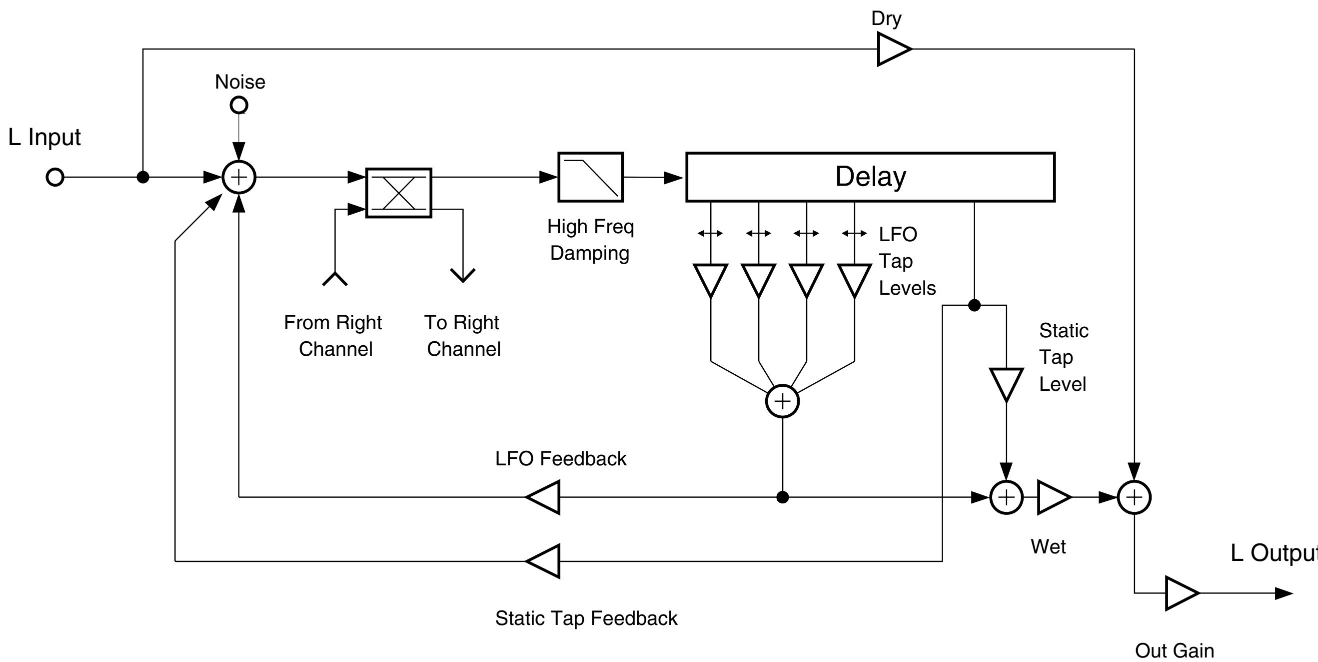


Figure 41 Simplified block diagram of the left channel of Flanger 2 (right channel is similar)

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

Adding or subtracting a signal with a time-displaced replica of itself results in 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). See Figure 41 above. 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.

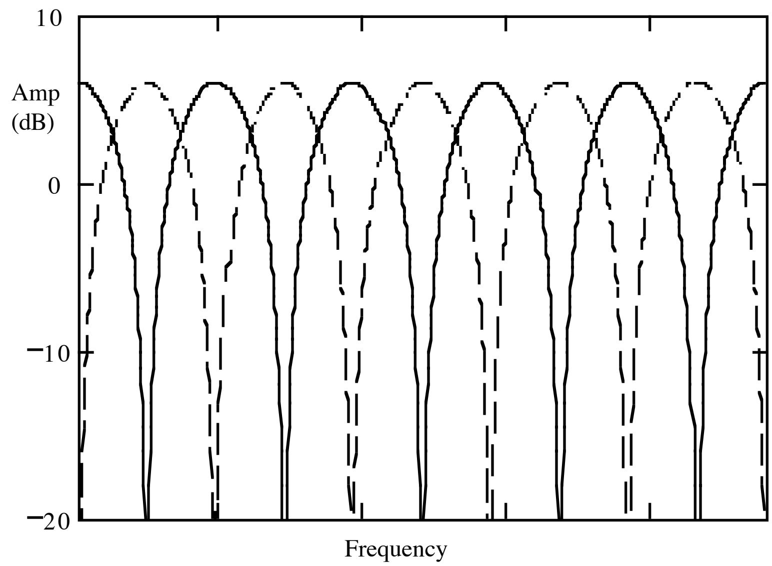


Figure 42 Comb filters: solid line for addition; dashed line for subtraction

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 inverting). 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. Where is the LFO in its sweep: going away from the input tap or coming 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, and the total range of an LFO is twice the excursion. 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 a continuous controller and use the controller to do manual flanging. Figure 43 below shows the delay line for a single LFO.

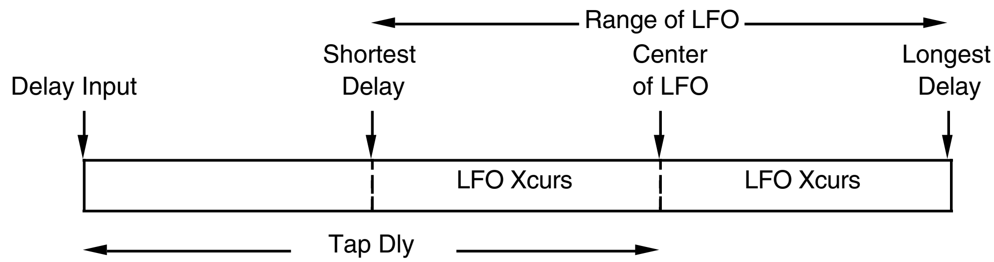


Figure 43 Delay for a single LFO

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 the LFOs 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 which is fed back is from the first LFO delay tap (LFO1), but with 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 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 previously, 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 in a proportionally related way 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." With a low-noise instrument (like the K2600), 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). White noise has a lot of high frequency content and may sound too bright. The noise may be tamed with a first order lowpass filter.

Parameters for 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	8 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 for 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	8 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. In this case, FXMods (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 **Flanger 2** only. 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 with low-noise instruments, if there is no input signal, there is no noise floor unless it is explicitly added.

Noise LP **Flanger 2** only. The cut-off frequency of a one pole lowpass filter acting on the noise injection signal. The lowpass removes high frequencies from an otherwise pure white noise signal.

StatDlyCrs The length of the static delay tap. 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.

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 any tap level to **0%** turns off the delay tap.

Xcurs *n* Crs The 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 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). 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.

Phasers

250 LFO Phaser
251 LFOPhaserTwinLFX
253 SingleLFO Phaser
254 VibratoPhaser
255 Manual Phaser
630 Mn LFO Phaser
631 Mn LFOPhaserTwin
632 Mn SingleLFOPhsr
633 Mn VibratoPhaser
634 Mn Manual Phaser

A variety of single notch/bandpass phasers

PAUs: 1 each

A simple phaser is an algorithm that 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 you.

A phaser uses a special filter called an allpass filter to modify the phase response of a signal's spectrum without changing the amplitude of the spectrum. Okay, that was a bit of a mouthful—so what does it mean? As the term "allpass filter" suggests, the filter by itself does not change the amplitude response of a signal passing through it. An allpass filter does not cut or boost any frequencies. An allpass filter 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 produced 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.

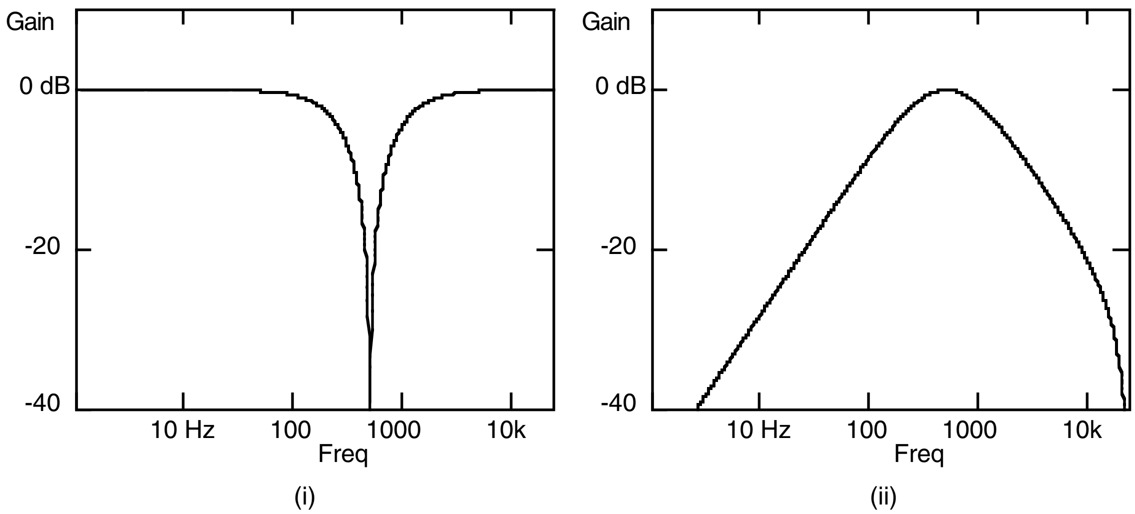


Figure 44 Response of typical phaser with (i) Wet/Dry = 50% and (ii) WetDry = -50%

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 are 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 which 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 LFO. The rates of the LFOs 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 a 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.

LFOPhaserTwinLFX 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. **LFOPhaserTwinLFX** does not have Out Gain or feedback parameters.

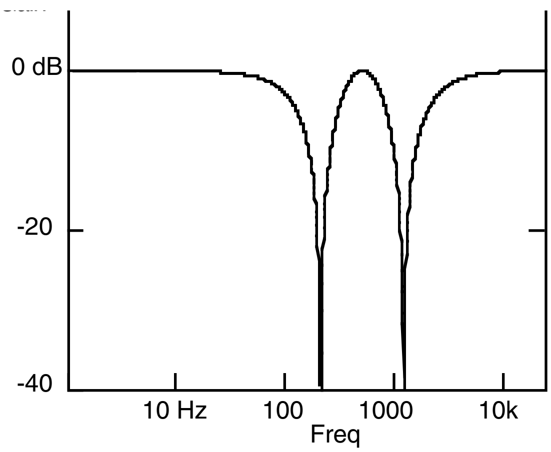


Figure 45 Response of LFOPhaserTwinLFX with Wet/Dry set to 100%.

The **VibratoPhaser** algorithm has a couple of interesting twists. 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. Setting Wet/Dry to **50%** will produce a full notch. At **-50%** Wet/Dry, you get a bandpass.

Parameters for 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	8 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 for 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	8 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 being at a 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 with 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 for 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	8 to 25088 Hz	R Ctr Freq	8 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 for LFOPhaserTwinLFX

Page 1

Notch/Dry	0 to 200 %	Out Gain	Off, -79.0 to 24.0 dB
CenterFreq	8 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 allpass 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 Hz.

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 being at the minimum frequency while the other channel is at the maximum.

Parameters for VibratoPhaser

Page 1

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

Page 2

CenterFreq	8 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 Hz. 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.

256 Allpass Phaser 3
257 Allpass Phaser 4
635 Mn AP Phaser 3
636 Mn AP Phaser 4

Allpass filter phasers

PAUs: 3 for **Allpass Phaser 3**, 4 for **Allpass Phaser 4**, and 2 for the mono allpass phasers

The allpass phasers are algorithms that use allpass filters to achieve a phaser effect. These algorithms do not have built in LFOs, so like Algorithm **255 Manual Phaser**, any motion must be supplied with an FXMod. Unlike the other phasers, the allpass phasers use high order allpass filters. The order of the allpass filters sets the number of notches that will appear in the frequency response when the dry and filtered signals are mixed. The number of notches in the frequency response ranges from 3 to 6 for **Allpass Phaser 3** and 7 to 10 for **Allpass Phaser 4**. **Allpass Phaser 3** and **Allpass Phaser 4** are identical except for the number of notches and PAU usage. The allpass phaser algorithms use a typical signal routing with wet/dry and cross-coupled feedback. A different number of notches may be chosen for the feedback path than for the direct output.

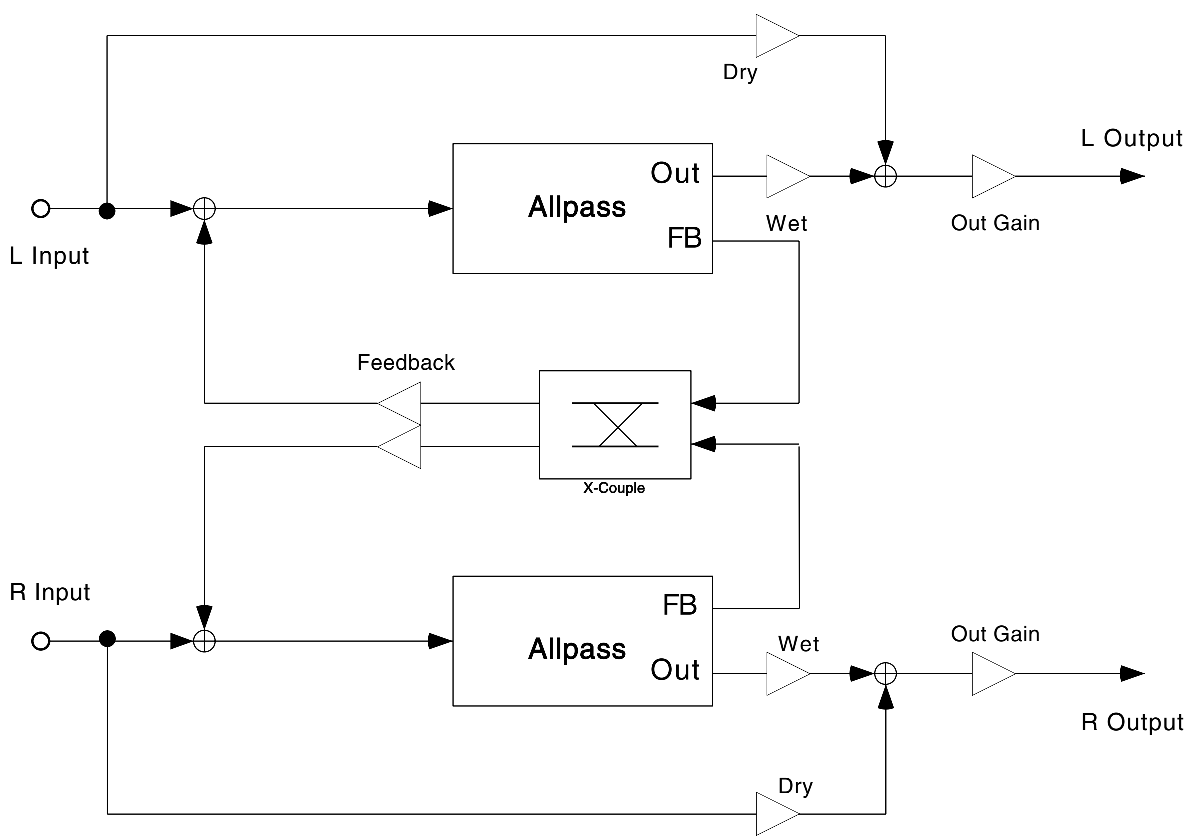


Figure 46 **An allpass phaser**

A phaser uses a special filter called an allpass filter to modify the phase response of a signal's spectrum without changing the amplitude of the spectrum. Okay, that was a bit of a mouthful—so what does it mean? As the term "allpass filter" suggests, the filter by itself does not change the amplitude response of a signal passing through it. An allpass filter does not cut or boost any frequencies. An allpass filter 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 produced 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.

As mentioned earlier, allpass phasers leave the phaser motion up to you, so they have no built in LFOs. 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.

When feedback is used, it can greatly exaggerate the peaks and notches, producing a much more resonant sound. Cross-coupling (XCouple) the feedback between the left an right channels increases the complexity of the frequency response.

In the figure above, you'll notice that the spacing of the notches and peaks are not harmonically related. When a lot of feedback is used, the non-harmonic structure produces very bell-like tones, particularly with XCouple set to **100%.** (Don't modulate the frequencies to get this effect.) Try experiments using different allpass orders for the feedback, different frequency arrangements, changing the sign (+/-) of the feedback (Fdbk Level) parameter, and different input sources (drums are a good starting point).

Parameters:

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%
LCenterFreq	8 to 25088 Hz	RCenterFreq	8 to 25088 Hz
FB APNotch	3 to 6 or 7 to 10	OutAPNotch	3 to 6 or 7 to 10

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.

XCouple Determines how much of the right feedback signal to feed into the left input channel and how much left feedback to feed into the right input channel. When increasing cross-coupling, the amount of feedback from one channel into its own input is reduced, so that at **100%** the left feeds back entirely to the right channel and vice versa. [Stereo versions only]

CenterFreq The nominal center frequency of the phaser filter. The frequency LFO modulates the phaser filter centered at this frequency. There are separate left and right controls in the stereo version.

FB APNotch The number of notches the allpass filter can produce when summed with a dry signal. Used in the feedback loop. Higher values produce more resonant peaks, for a more complex resonant structure.

OutAPNotch The number of notches the allpass filter can produce when summed with a dry signal. Used on the algorithm output. Higher values produce a steeper, longer phase response resulting in more peaks and notches when combined with the dry signal.

Comb Filters

258 Barberpole Comb
637 Mn Barberpole

Comb filter with constantly rising or falling frequency

PAUs: 4 for **Barberpole Comb** and 2 for **Mn Barberpole**

The **Barberpole Comb** is a comb filter with a constantly rising or falling frequency. A comb filter produces a series of evenly spaced notches or resonant peaks in the frequency response. The comb filter gets its name from the comb-like appearance of the frequency response. A comb filter producing notches is created by adding the input signal to a delayed (and possibly attenuated) version of the input signal. To produce peaks, the output signal is passed through a short delay and attenuation (level reduction) and added to the input signal to produce a delay feedback.

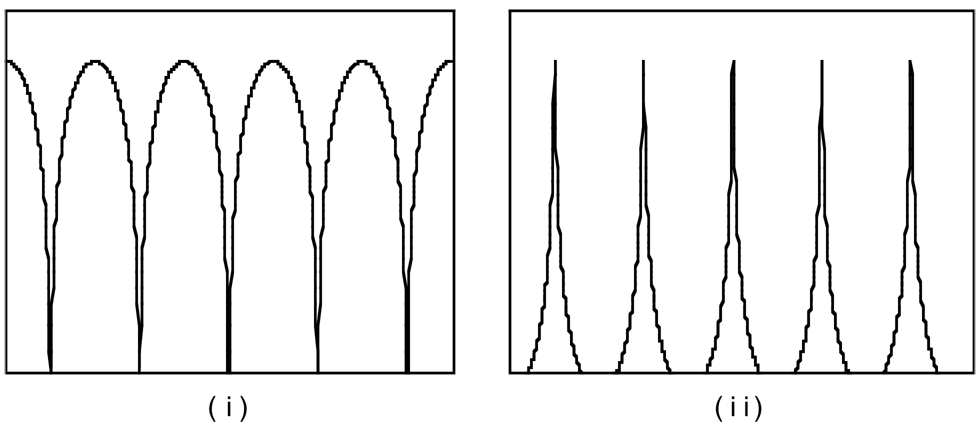


Figure 47 Frequency response of comb filter with (i) notches and (ii) peaks

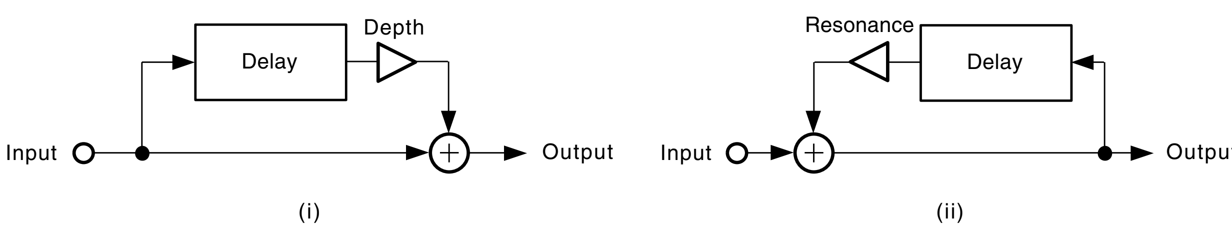


Figure 48 Simple comb filter designs for (i) notches and (ii) peaks

The **Barberpole Comb** can be configured to produce either notches or peaks. There is a twist to the **Barberpole Comb** algorithm in that the notches or peaks can be made to shift up (or down) in frequency. As the notches or peaks shift up, the highest notches or peaks go away while new notches or peaks appear at the lowest frequencies.

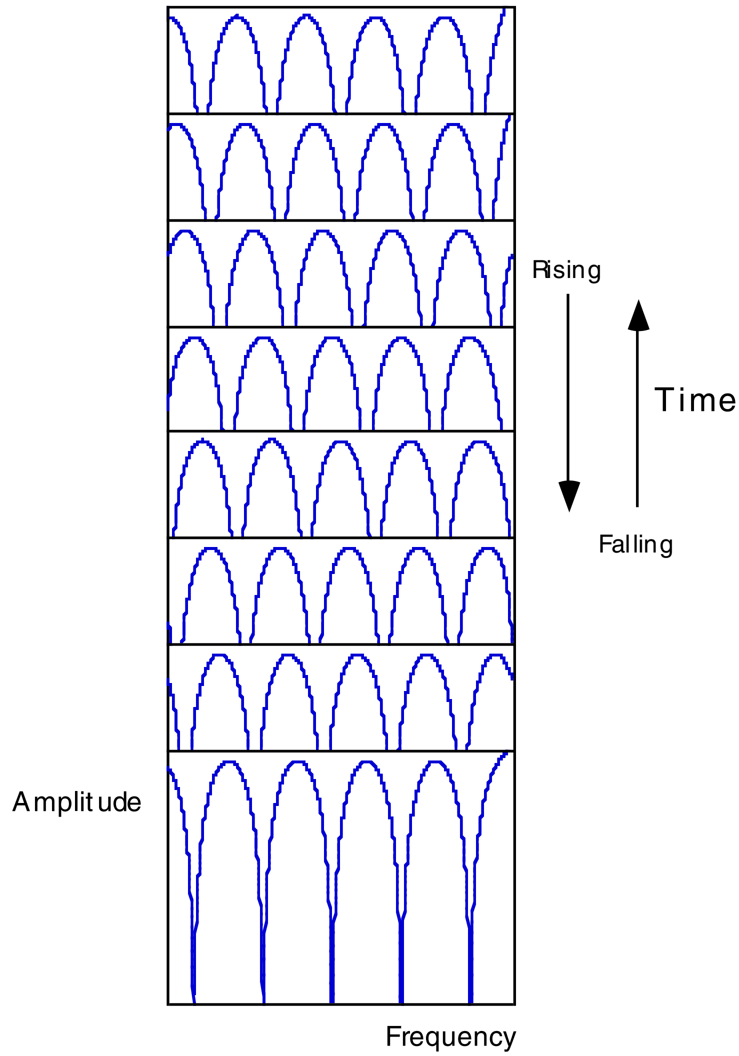


Figure 49 **Barberpole comb filter responses at different instants of time**

Parameters:

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
Rise/Fall	Rise or Fall	Notch/Peak	Notch or Peak
Rate	0.00 to 10.00 Hz	Depth/Res	0 to 100%
L/R Phase	-180.0 to 178.5 deg	Comb Freq	8 to 25088 Hz

In/Out When set to **In** the algorithm is active; when set to **Out** the algorithm is bypassed.

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

Rise/Fall When set to **Rise**, the comb filter frequencies rotate to higher frequencies. When set to **Fall**, the comb filter frequencies rotate to lower frequencies.

Rate	The LFO rate at which the comb filters rise or fall through a complete cycle of frequencies.
Comb Freq	The frequency separation of the notches or peaks in the comb filter.
Notch/Peak	The comb filter can be constructed to produce a series of notches in the frequency response or a series on resonant peaks.
Depth/Res	The depth of the notches (set Notch/Peak to **Notch**) or the height of the resonant peaks (set Notch/Peak to **Peak**) can be set with Depth/Res. A setting of **100%** produces the deepest notches or the highest peaks.
L/R Phase	The left and right channels may place the comb filter notches or peaks at different frequencies, controlled by the relative phase angles of the LFOs controlling the comb frequency rotations for the left and right channels.

Tremolo Effects

270 Tremolo BPM
271 Tremolo
640 Mn Tremolo BPM
641 Mn Tremolo

A stereo tremolo or auto-balance effect

PAUs: 1

Tremolo and **Tremolo BPM** are one-PAU stereo tremolo effects. In the classical sense, a tremolo is the rapid repetition of a single note created by an instrument. Early music synthesists imitated this by using an LFO to modulate the amplitude of a tone. This is the same concept as amplitude modulation, except that a tremolo usually implies that the modulation rate is much slower.

Tremolo and Tremolo BPM provide six different LFO shapes (Figure 2), an additional shape modifier called “50% Weight,” “L/R Phase” for auto-balancing, and LFO metering. L/R Phase flips the LFO phase of the left channel for auto-balancing applications. The 50% Weight parameter bends the LFO shape up or down relative to its -6dB point (Figure 1). At 0dB, there is no change to the LFO shape. Positive values will bend the LFO up towards unity, while negative values will bend it down towards full attenuation. Additionally, LFO metering can be viewed on the bottom of PARAM2 page.

Tremolo also includes an LFO rate scale for AM synthesis, and **Tremolo BPM** provides tempo based LFO syncing including system syncing.

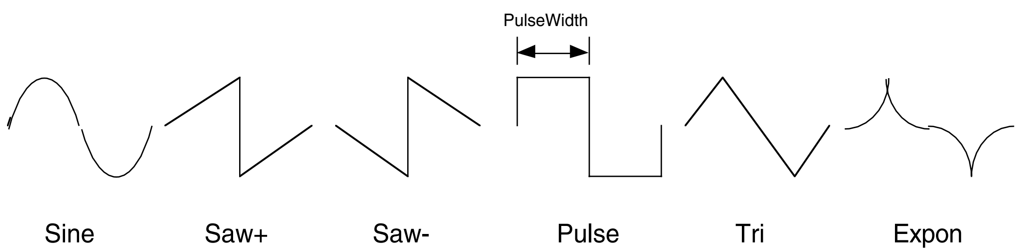


Figure 50 **LFO Shapes available for Tremolo and Tremolo BPM**

Parameters for Tremolo

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB

Page 2

LFO Rate	0 to 10.00 Hz	LFO Shape	Tri
Rate Scale	1 to 25088 x	PulseWidth	0 to 100 %
Depth	0 to 100 %	50% Weight	-6 to 3 dB
		L/R Phase	In or Out
A			
0% 50% 100%			

Parameters for Tremolo BPM

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
		Tempo	System, 0 to 255 BPM

Page 2

LFO Rate	0 to 12.00 x	LFO Shape	Tri
LFO Phase	0.0 to 360.0 deg	PulseWidth	0 to 100 %
Depth	0 to 100 %	50% Weight	-6 to 3 dB
		L/R Phase	In or Out
A			
0% 50% 100%			

In/Out When set to **In** the effect is active; when set to **Out** the effect is bypassed.

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

Tempo For Tremolo BPM. Basis for the rate of the LFO, 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. In this case, FXMods (FUNs, LFOs, ASRs etc.) will have no effect on the Tempo parameter.

LFO Rate For Tremolo. The speed of the tremolo LFO in cycles per second.

LFO Rate For Tremolo BPM. The number of LFO cycles in one beat relative to the selected Tempo. For example, **1.00x** means the LFO repeats once per beat; **2.00x** twice per beat, and so on.

Rate Scale For Tremolo. This multiplies the speed of the LFO rate into the audio range. When above **19x**, the values increment in semitone steps. These steps are accurate when LFO Rate is set to 1.00 Hz.

LFO Phase For Tremolo BPM. This parameter shifts the phase of the tremolo LFO relative to an internal beat reference. It is most useful when Tempo is set to **System** and LFO Phase controls the phase of the LFO relative to MIDI clock.

Depth This controls the amount of attenuation applied when the LFO is at its deepest excursion point.

LFO Shape The waveform type for the LFO. Choices are **Sine**, **Saw+**, **Saw–**, **Pulse**, **Tri**, and **Expon**.

PulseWidth When the LFO Shape is set to **Pulse**, this parameter sets the pulse width as a percentage of the waveform period. The pulse is a square wave when the width is set to **50%**. This parameter is active only when the Pulse waveform is selected.

50% Weight The relative amount of attenuation added when the LFO is at the -6dB point. This causes the LFO shape to bow up or down depending on whether this parameter is set positive or negative (see **Figure 51**).

L/R Phase LFO phase relationship of the left channel. Flipping the left channel's LFO out of phase causes the effect to become an auto-balancer.

Panners and Stereo-Image Effects

279 AutoPanner BPM

A stereo auto-panner

PAUs: 1

AutoPanner BPM is a one-PAU stereo auto pan effect. The process of panning a stereo image consists of shrinking the image width of the input program then cyclically moving this smaller image from side to side while maintaining relative distances between program point sources (Figure 51). This effect provides six different LFO shapes (Figure 52), variable center attenuation, and a rate scaler that scales LFO rate into the audible range for a new flavor of amplitude modulation effects.

Final image placement can be monitored on the lower right of the PARAM2 page. The top meter labeled "L" shows the left edge of the image while the second meter labeled "R" shows the right edge. The entire image will fall between these two marks.

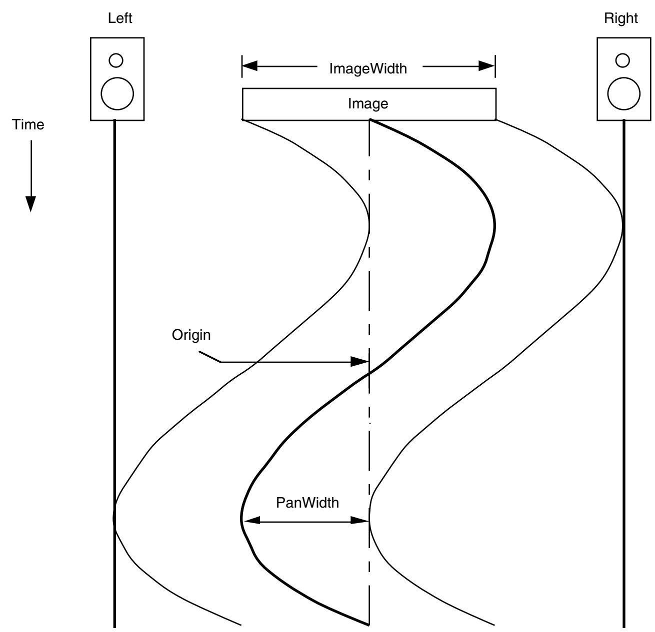


Figure 51 Stereo autopanning

In Figure 52, ImageWidth is set to **50**%, LFO Shape is set to **Sine**, Origin is set to **0**%, and PanWidth is set to **100**%.

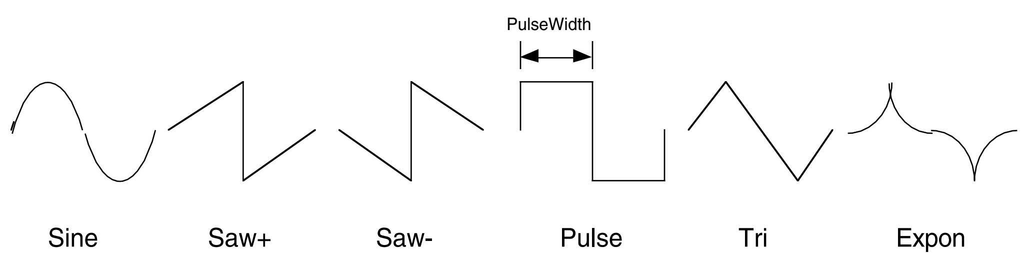


Figure 52 LFO Shapes available for AutoPanner BPM

Parameters

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB

Page 2

LFO Tempo	System, 1 to 255 BPS	LFO Shape	Tri
LFO Period	1/24 to 32 bts	PulseWidth	0 to 100%
Origin	-100 to 100 %		
PanWidth	0 to 100 %	L	
ImageWidth	0 to 100 %	R	
CentrAtten	-12 to 0 dB	L C R	

In/Out When set to **In** the auto-panner is active; when set to **Out** auto-panner is bypassed.

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

LFO Tempo Basis for the rates of the LFO, 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. In this case, FXMods (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).

Rate Scale Multiplies the speed of the LFO rate into the audio range. When above **19x**, the values increment in semitone steps. These steps are accurate when LFO Rate is set to **1.00 Hz**.

Origin The axis for the panning motion. At **0**%, panning excursion is centered between the listening speakers. Positive values shift the axis to the right, while negative values shift it to the left. At **-100**% or **+100**%, there is no room for panning excursion.

Pan Width The amount of auto pan excursion. This value represents the percentage of total panning motion available after Origin and ImageWidth are set.

ImageWidth The width of the original input program material before it is auto panned. At **0**%, the input image is shrunk to a single point source allowing maximum panning excursion. At **100**%, the original width is maintained leaving no room for panning excursion.

CentrAtten Amount the signal level is dropped as it is panned through the center of the listening stereo speaker array. For the smoothest tracking, a widely accepted subjective reference is **-3dB**. Values above **-3dB** will cause somewhat of a bump in level as an image passes through the center. Values below **-3dB** will cause a dip in level at the center.

LFO Shape The waveform type for the LFO. Choices are **Sine**, **Saw+**, **Saw–**, **Pulse**, **Tri**, and **Expon**.

PulseWidth When the LFO Shape is set to **Pulse**, this parameter sets the pulse width as a percentage of the waveform period. The pulse is a square wave when the width is set to **50**%. This parameter is active only when the Pulse waveform is selected.

276 Dual AutoPanner

A dual mono auto-panner

PAUs: 2

Dual AutoPanner is a two-PAU dual mono auto pan effect. Left and right inputs are treated as two mono signals, which can each be independently auto-panned. Parameters beginning with “L” control the left input channel, and parameters beginning with “R” control the right input channel. Autopanning a mono signal consists of choosing an axis offset, or Origin, as the center of LFO excursion, then adjusting the desired excursion amount, or PanWidth. Note that the PanWidth parameter is a percentage of the available excursion space after Origin is adjusted. If Origin is set to full left (**-100%**) or full right (**100%**) then there will be no room for LFO excursion. Control of six different LFO shapes (**Figure 54**), variable center attenuation, and a rate scaler that scales LFO rate into the audible range for a new flavor of amplitude modulation effects are also provided for each channel.

Final image placement can be seen on the bottom right of the PARAM2 and PARAM3 pages respectively for left and right input channels. The moving mark represents the location of each channel within the stereo field.

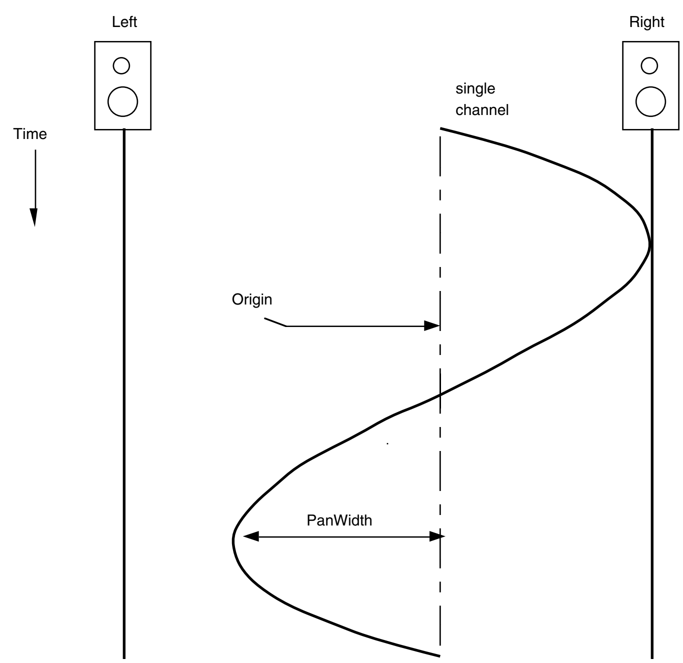


Figure 53 Mono autopanning

In **Figure 53**, LFO Shape is set to **Sine**, Origin is set to **15%**, and PanWidth is set to **100%**.

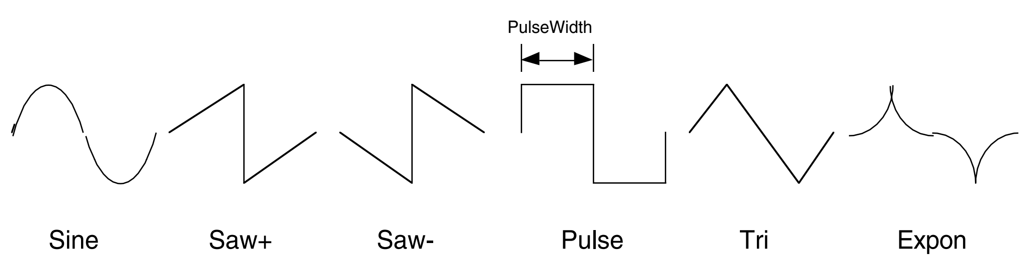


Figure 54 LFO Shapes available for Dual AutoPanner

Parameters

Page 1

L In/Out	In or Out	R In/Out	In or Out
L Out Gain	Off, -79.0 to 24.0 dB	R Out Gain	Off, -79.0 to 24.0 dB

Page 2

L LFO Rate	0 to 10.00 Hz	L LFO Shape	Tri
L RateScal	1 to 25088 x	L PlseWdth	0 to 100 %
L Origin	-100 to 100 %		
L PanWidth	0 to 100 %		
L CentrAtt	0 to 100 %	L	
		L C R	

Page 3

R LFO Rate	0 to 10.00 Hz	R LFO Shape	Tri
R RateScal	1 to 25088 x	R PlseWdth	0 to 100 %
R Origin	-100 to 100 %		
R PanWidth	0 to 100 %		
R CentrAtt	0 to 100 %	R	
		L C	

In/Out When set to **In** the algorithm is active; when set to **Out** the algorithm is bypassed.

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

LFO Rate The speed of the panning motion.

Origin The axis for the panning motion. At **0%**, panning excursion will be centered at the center of the listening speakers. Positive values shift the axis to the right, while negative values shift it to the left. At **-100%** or **+100%**, there is no room for panning excursion.

Pan Width The amount of auto pan excursion. This value represents the percentage of total panning motion available after Origin is set.

CentrAtten Amount the signal level is dropped as it is panned through the center of the listening stereo speaker array. For the smoothest tracking, a widely accepted subjective reference is **-3dB**. Values above **-3dB** will cause somewhat of a bump in level as an image passes through the center. Values below **-3dB** will cause a dip in level at the center.

LFO Shape The waveform type for the LFO. Choices are **Sine**, **Saw+**, **Saw–**, **Pulse**, **Tri**, and **Expon**.

PulseWidth When the LFO Shape is set to **Pulse**, this parameter sets the pulse width as a percentage of the waveform period. The pulse is a square wave when the width is set to **50%**. This parameter is active only when the Pulse waveform is selected.

280 Stereo Image

Stereo enhancement with stereo channel correlation metering

PAUs: 1

Stereo Image is a stereo enhancement algorithm with metering for stereo channel correlation. The stereo enhancement performs simple manipulations of the sum and difference of the left and right input channels to allow widening of the stereo field and increased sound field envelopment. After manipulating sum and difference signals, the signals are recombined (a sum and difference of the sum and difference) to produce final left and right output.

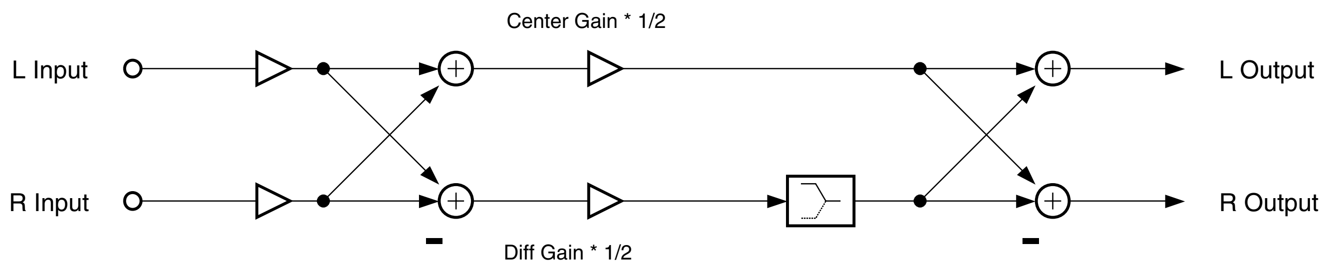


Figure 55 Block diagram of Stereo Image algorithm

The sum of left and right channels represents the mono or center mix of your stereo signal. The difference of left and right channels contains the part of the signal that contains stereo spatial information. The **Stereo Image** algorithm has controls to change the relative amounts of sum (or center) versus difference signals. By increasing the difference signal, you can broaden the stereo image. Be warned, though, that too much difference signal will make your stereo image sound "phasey." With phasey stereo, acoustic images become difficult to localize and can sound like they are coming from all around or from within your head.

A bass shelf filter on the difference signal is also provided. By boosting only the low frequencies of the difference signal, you can greatly improve your sense of stereo envelopment without destroying your stereo sound field. Envelopment is the feeling of being surrounded by your acoustic environment. Localized stereo images still come from between your stereo loudspeakers, but there is an increased sense of being wrapped in the sound field.

The **Stereo Image** algorithm contains a stereo correlation meter. The stereo correlation meter tells you how alike or how different your output stereo channels are from each other. When the meter is at 100% correlation, then your signal is essentially mono. At 0% correlation, your left and right channels are the same, but polarity inverted (there is only difference signal). The correlation meter can give you an indication of how well a recording will mix to mono. The meter follows RMS signal levels (root-mean-square) and the RMS Settle parameter controls how responsive the meter is to changing signals. The 'M' part of RMS is "mean" or average of the squared signal. Since a mean over all time is neither practical or useful, we must calculate the mean over shorter periods of time. If the time is too short we are simply following the signal wave form, which is not helpful either, since the meter would constantly bounce around. The RMS Settle parameter provides a range of useful time scales.

See also the **Stereo Analyze** algorithm (**page 302**), which allows you to experiment directly with sum and difference signals.

Parameters

Page 1

L In Gain	Off, -79.0 to 24.0 dB	R In Gain	Off, -79.0 to 24.0 dB
CenterGain	Off, -79.0 to 24.0 dB	Diff Gain	Off, -79.0 to 24.0 dB
L/R Delay	-500.0 to 500.0 samp	RMS Settle	0.0 to 300.0 dB/s

Page 2

DiffBassG	-79.0 to 24.0 dB		
DiffBassF	8 to 25088 Hz		
	Stereo Correlation		
	100 75 50 25 0%		

L In Gain The input gain of the left channel in decibels (dB).

R In Gain The input gain of the right channel in decibels (dB).

CenterGain The level of the sum of left and right channels in decibels (dB). The summed stereo signal represents the mono or center mix.

Diff Gain The level of the difference of left and right channels in decibels (dB). The difference signal contains the spatial component of the stereo signal.

L/R Delay If this parameter is positive, the left signal is delayed by the indicated amount. If it is negative, the right channel is delayed. You can use this parameter to try to improve cancellation of the difference signal if you suspect one channel is delayed with respect to the other.

RMS Settle Controls how fast the RMS meters can rise or fall with changing signal levels.

DiffBassG By boosting the low frequency components of the difference signal you can increase the sense of acoustic envelopment, the sense of being surrounded by an acoustic space. DiffBassG is the gain parameter of a bass shelf filter on the difference signal. DiffBassG sets how many decibels (dB) to boost or cut the low frequencies.

DiffBassF The transition frequency in Hertz (Hz) of the difference signal bass shelf filter is set by DiffBassF.

281 Mono -> Stereo

Stereo simulation from a mono input signal

PAUs: 1

Mono -> Stereo creates a stereo signal from a mono input signal. The algorithm works by combining a number of band-splitting, panning and delay tricks. The In Select parameter lets you choose the left or right channel for you mono input, or you may choose to sum the left and right inputs.

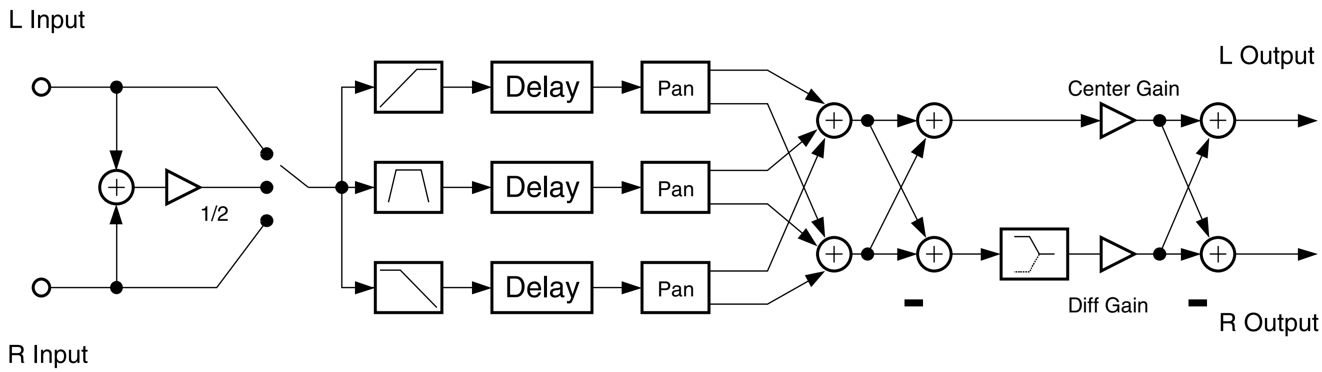


Figure 56 Block diagram of Mono -> Stereo

The mono input signal is split into three frequency bands (Low, Mid, and High). The frequencies at which the bands get split are set with the Crossover parameters. Each band can then be delayed and panned to some position within your stereo field.

The final step manipulates the sum and difference signals of the pseudo-stereo signal created by recombining the split frequency bands. The sum of left and right channels represents the mono or center mix of your stereo signal. The difference of left and right channels contains the part of the signal that contains stereo spatial information. The **Mono -> Stereo** algorithm has controls to change the relative amounts of sum (or center) versus difference signals. By increasing the difference signal, you can broaden the stereo image. Be warned, though, that too much difference signal will make your stereo image sound "phasey." With phasey stereo, acoustic images become difficult to localize and can sound like they are coming from all around you or from within your head.

A bass shelf filter on the difference signal is also provided. By boosting only the low frequencies of the difference signal, you can greatly improve your sense of stereo envelopment without destroying your stereo sound field. Envelopment is the feeling of being surrounded by your acoustic environment. Localized stereo images still come from between your stereo loudspeakers, but there is an increased sense of being wrapped in the sound field.

Parameters

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
CenterGain	Off, -79.0 to 24.0 dB	Diff Gain	Off, -79.0 to 24.0 dB
In Select	L, R, or (L+R)/2	DiffBassG	-79.0 to 24.0 dB
		DiffBassF	8 to 25088 Hz

Page 2

Crossover1	8 to 25088 Hz		
Crossover2	8 to 25088 Hz		
Pan High	-100 to 100 %	Delay High	0.0 to 1000.0 ms
Pan Mid	-100 to 100 %	Delay Mid	0.0 to 1000.0 ms
Pan Low	-100 to 100 %	Delay Low	0.0 to 1000.0 ms

In/Out The algorithm is functioning when In/Out is set to **In**. If set to **Out**, whatever is on the input channels gets passed to the output unaltered.

Out Gain The output gain of the pseudo-stereo signal in decibels (dB).

CenterGain The level of the sum of the intermediate left and right stereo channels in decibels (dB). The summed stereo signal represents the mono or center mix.

Diff Gain The level of the difference of the intermediate left and right stereo channels in decibels (dB). The difference signal contains the spatial component of the stereo signal.

In Select The input signal may come from the left L or right R input channel, or the left and right channels may be summed to obtain the mono signal (L+R)/2. You should set this parameter to match your Studio configuration.

DiffBassG By boosting the low frequency components of the difference signal of the intermediate stereo result, you can increase the sense of acoustic envelopment, the sense of being surrounded by an acoustic space. DiffBassG is the gain parameter of a bass shelf filter on the difference signal. DiffBassG sets how many decibels (dB) to boost or cut the low frequencies.

DiffBassF The transition frequency in Hertz (Hz) of the difference signal bass shelf filter is set by DiffBassF.

Crossover*N* The two Crossover parameters set the frequencies at which the band-split filters split the mono signal into three bands. The two parameters are interchangeable: either may have a higher frequency than the other.

Pan Low, Mid, and High. The panning of each band is separately controllable. **-100%** is fully left and **100%** is fully right.

Delay Low, Mid, and High. The delays are set in milliseconds (ms).

282 DynamicStereoize

Stereo widening based on dynamic signal levels

PAUs: 2

DynamicStereoize is a stereo enhancement (or reduction) algorithm. By increasing the level of the difference signal between left and right input channels relative to the summed left and right channels (mono), you get an increased sense of stereo separation. Likewise, reducing the difference signal relative to the summed signal makes the sound more mono or centered. So far this description differs little from Algorithm **280 Stereo Image**.

Now if we place dynamic range controls (compressor and/or expander) on either the summed or difference signal paths, some interesting things happen. A compressor reduces output signal level when the input signal level gets louder. An expander reduces output signal level when the input signal gets softer. With a compressor or expander on one of the sum or difference signal paths, your sound can be made very spacious at low signal levels but centered at higher levels. You can also achieve the opposite effect with low level signal centered and high signal levels wide.

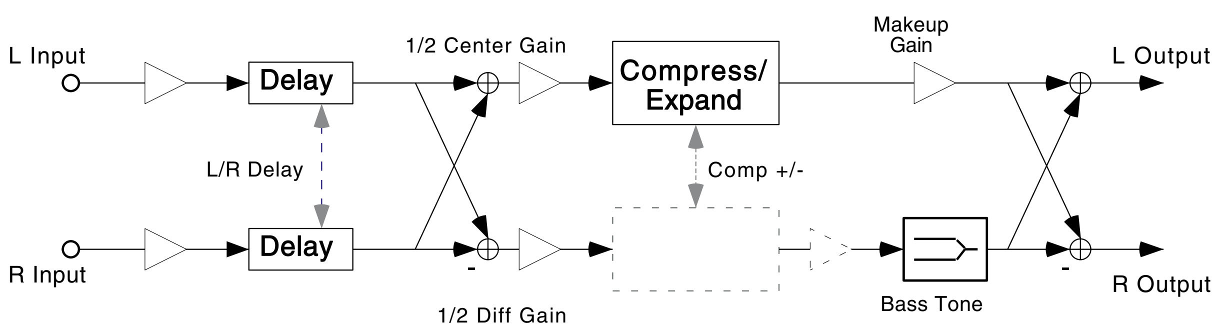


Figure 57 **DynamicStereoize**

The compressor/expander switching in the figure above looks a little complicated, but conceptually it is very simple. Using the Comp +/- parameter you select whether to compress or expand the summed left and right signal ((L+R)/2) or the difference signal ((L-R)/2). The final sum and difference calculation reconstructs the original left and right signals (assuming you turned off the intermediate processing). You can prove this to yourself by solving the equations (L+R)/2 + (L-R)/2 and (L+R)/2 - (L-R)/2.

Let's look at some examples using compression and expansion on the sum and difference signals. We want to make the sound mono when it is loud and spacious when it is soft. There are two approaches: you can expand the summed signal, or compress the difference signal. By expanding the summed signal, the mono component gets reduced as the sound gets quieter, producing a more spacious sound. By compressing the difference signal, the out of phase components are reduced as the signal gets louder for a more mono signal at higher levels. You will have to work with the CenterGain and Diff Gain parameters to achieve the balance of spaciousness and mono you are looking for.

For details on how the compressor works, see Algorithm **341 Compress/Expand**. The difference between **Compress/Expand** and the compressor/expander used here is that this c/e is mono, working on a single (sum or difference) channel.

A bass shelf filter on the difference signal is also provided. By boosting only the low frequencies of the difference signal, you can greatly improve your sense of stereo envelopment without destroying your stereo sound field. Envelopment is the feeling of being surrounded by your acoustic environment.

Localized stereo images still come from between your stereo loudspeakers, but there is an increased sense of being wrapped in the sound field.

The **DynamicStereoize** algorithm contains a stereo correlation meter. The stereo correlation meter tells you how alike or how different your output stereo channels are from each other. When the meter is at 100% correlation, then your signal is essentially mono. At 0% correlation, your left and right channels are the same, but polarity inverted (there is only difference signal). The correlation meter can give you an indication of how well a recording will mix to mono. The meter follows RMS signal levels (root-mean-square) and the RMS Settle parameter controls how responsive the meter is to changing signals. The 'M' part of RMS is "mean" or average of the squared signal. Since a mean over all time is neither practical or useful, we must calculate the mean over shorter periods of time. If the time is too short we are simply following the signal wave form, which is not helpful either, since the meter would constantly bounce around. The RMS Settle parameter provides a range of useful time scales.

The left input can be delayed with respect to the right input channel (or the other way around). You can use the L/R Delay to realign your inputs in time or to experiment with the precedence effect. With the precedence effect, when we hear a sound first with the left ear then with the right ear, it sounds like the sound is coming from the left side.

Parameters:

Page 1

L In Gain	Off, -79.0 to 24.0 dB	R In Gain	Off, -79.0 to 24.0 dB
CenterGain	Off, -79.0 to 24.0 dB	Diff Gain	Off, -79.0 to 24.0 dB
L/R Delay	-500.0 to 500.0 samp	RMS Settle	0.0 to 300.0 dB/s

Page 2

DiffBassG	-79.0 to 24.0 dB		
DiffBassF	8 to 25088 Hz		
	Stereo Correlation		
	●		
	100 75 50 25 0%		

Page 3

Comp+/-	(L+R)/2 or (L-R)/2		
Comp Atk	0.0 to 228.0 ms	Exp Atk	0.0 to 228.0 ms
Comp Rel	0 to 3000 ms	Exp Rel	0 to 3000ms
SmoothTime	0.0 to 228.0 ms		
Signal Dly	0.0 to 25.0 ms		

Page 4

Comp Ratio	1.0:1 to 100.0:1, Inf:1	Exp Ratio	1:1.0 to 1:17.0
Comp Thres	-79.0 to 0.0 dB	Exp Thres	-79.0 to 0.0 dB
		MakeUpGain	Off, -79.0 to 24.0 dB
		|||||||||||||||||||||||||||	Reduction
	-dB 40 20 12 8	6 4 2	0

L In Gain The input gain of the left channel in decibels (dB).

R In Gain The input gain of the right channel in decibels (dB).

CenterGain The level of the sum of left and right channels in decibels (dB). The summed stereo signal represents the mono or center mix.

Diff Gain The level of the difference of left and right channels in decibels (dB). The difference signal contains the spatial component of the stereo signal.

L/R Delay If this parameter is positive, the left signal is delayed by the indicated amount. If it is negative, the right channel is delayed. You can use this parameter to try to improve cancellation of the difference signal if you suspect one channel is delayed with respect to the other.

RMS Settle Controls how fast the RMS meters can rise or fall with changing signal levels.

DiffBassG By boosting the low frequency components of the difference signal you can increase the sense of acoustic envelopment, the sense of being surrounded by an acoustic space. DiffBassG is the gain parameter of a bass shelf filter on the difference signal. DiffBassG sets how many decibels (dB) to boost or cut the low frequencies.

DiffBassF The transition frequency in Hertz (Hz) of the difference signal bass shelf filter is set by DiffBassF.

Comp +/- A selector switch to choose the signal path which gets dynamics processing (compress and/or expand). The choice of signal paths is the sum of left and right ((L+R)/R) and the difference ((L-R)/2).

Comp Atk The time for the compressor to start to cut in when there is an increase in signal level (attack) above the threshold.

Comp Rel The time for the compressor to stop compressing when there is a reduction in signal level (release) from a signal level above the threshold.

Exp Atk The time for the expander to increase the gain of the signal (turns off the expander) after the signal rises above threshold.

Exp Rel The time for the expander to reduce the signal level when the signal drops below the threshold (turning on expansion).

SmoothTime A lowpass filter in the control signal path. It is intended to smooth the output of the compressor/expander's envelope detector. Smoothing will affect the attack or release times when the smoothing time is longer than one of the other times.

Signal Dly The time in ms by which the input signal should be delayed with respect to compressor/expander side chain processing (i.e. side chain predelay). This allows the compression/expansion to appear to take effect just before the signal actually rises.

Comp Ratio The compression ratio in effect above compression threshold (Comp Thres). High ratios are highly compressed; low ratios are moderately compressed.

Comp Thres Threshold is expressed in dBFS (decibels relative to full scale) above which the signal begins to be compressed.

Exp Ratio The expansion ratio. High values (1:17 max) are highly expanded, low values (**1:1** min) are moderately expanded.

Exp Thres The expansion threshold level in dBFS (decibels relative to full scale) below which the signal begins to be expanded.

MakeUpGain Provides an additional control of the compressor/expander output gain. The Out Gain and MakeUpGain controls are additive (in decibels) and together may provide a maximum of 24 dB boost to offset gain reduction due to compression or expansion.

Guitar Cabinet Simulators

284 Cabinet
645 Mn Cabinet

Guitar cabinet simulation filters

PAUs: 3 for **Cabinet** and 2 for **Mn Cabinet**

The cabinet simulator models the responses of various types of mic'd guitar cabinets. The preset can be selected using the Cab Preset parameter.

Parameters:

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
Cab Preset	Open 12		

In/Out When set to **In** the cabinet filter is active; when set to **Out** the cabinet filter is bypassed.

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

Cab Preset Eight preset cabinets have been created based on measurements of real guitar amplifier cabinets. They're described below.

Basic Flat response from 100 Hz to 4 khz with 24dB/oct rolloffs on each end
Lead 12 Open back hard American type with one 12″ driver
2x12 Closed back classic American type with two 12″ drivers
Open 12 Open back classic American type with one 12″ driver
Open 10 Open back classic American type with one 10″ driver
4x12 Closed back British type with four 12″ drivers
Hot 2x12 Closed back hot rod type with two 12″ drivers
Hot 12 Open back hot rod type with one 12″ driver

Rotary Effects

290 VibChor+Rotor 2
291 Distort + Rotary
292 VC+Dist+HiLoRotr
293 VC+Dist+1Rotor 2
294 VC+Dist+HiLoRot2
295 Rotor 1
296 VC+Dist+Rotor 4
297 VC+Tube+Rotor 4
298 Big KB3 Effect
646 Mn VC+Dist+Rotor

Rotating speaker algorithms

PAUs: 1 for **Rotor 1**
2 each for **VibChor+Rotor 2**, **Distort + Rotary**, **VC+Dist+1Rotor 2**, **VC+Dist+HiLoRotr**, **Mn VC+Dist+Rotor** and **VC+Dist+HiLoRot2**
4 each for **VC+Dist+Rotor 4** and **VC+Tube+Rotor 4**
8 for **Big KB3 Effect**

The rotary algorithms contain multiple effects designed for the Hammond B3® emulation (KB3 mode). These effects may include the Hammond® vibrato/chorus, amplifier distortion, cabinet emulation and rotating speaker (Leslie®). A variety of rotating speaker algorithms have been designed to deal with different circumstances Some of the algorithms are designed to trade off features or model quality to allow the rotating speaker model to work in fewer PAUs.

The first effect in the chain is often the Hammond® vibrato/chorus algorithm. The vibrato/chorus has six settings which are the same as those used in the Hammond B3®: three vibrato (V1, V2, V3) and three chorus (C1, C2, C3) settings. In **Big KB3 Effect**, **VC+Dist+Rotor 4**, and **VC+Tube+Rotor 4**, the vibrato chorus has been carefully modeled after the electromechanical vibrato/chorus in the B3. The vibrato/chorus in the other smaller algorithms use a conventional design, which has been set to match the B3 sound as closely as possible, but does not quite have the same character as the fully modeled vibrato/chorus.

The final section of each of the rotary algorithms is the rotating speaker routine. The various algorithms may trade off some features of the rotating speaker routine and the tradeoffs will be discussed for each algorithm separately. However as an introduction, let’s discuss a full featured rotating speaker.

The rotating speaker has separately controllable tweeter and woofer drivers. The signal is split into high and low frequency bands and the two bands are run through separate rotors. The upper and lower rotors each have one or two virtual microphones which can be positioned at varying positions (angles) around the rotors. An angle of 0° is loosely defined as the front of the speaker. You can also control the levels and left-right panning of each virtual microphone. The signal is then passed through a final cabinet filter to simulate the band-limiting effect of the speaker cabinet. The cabinet filter is often a simple lowpass filter.

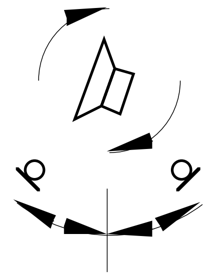


Figure 58 Rotating speaker with virtual microphones

For the rotating speakers, you can control the cross-over frequency of the high and low frequency bands (the frequency where the high and low frequencies get separated). The rotating speakers for the high and low frequencies have their own controls. For both, the rotation speed, the effective driver size and tremolo can be set. The effective driver size is the radius of the path followed by the speaker relative to its center of rotation. This parameter is used to calculate the resulting Doppler shift of the moving speaker. Doppler shift is the pitch shift that occurs when a sound source moves toward or away from you the listener. In a rotating speaker, the Doppler shift will sound like vibrato. As well as Doppler shift, there will be some acoustic shadowing as the speaker is alternately pointed away from you and toward you. The shadowing is simulated with a tremolo over which you can control the tremolo depth and "width." The high frequency driver (rotating horn) will have a narrower acoustic beam width (dispersion) than the low frequency driver, and the widths of both may be adjusted. Note that it can take up to one full speaker rotation before you hear changes to tremolo when parameter values are changed. Negative microphone angles take a longer time to respond to tremolo changes than positive microphone angles.

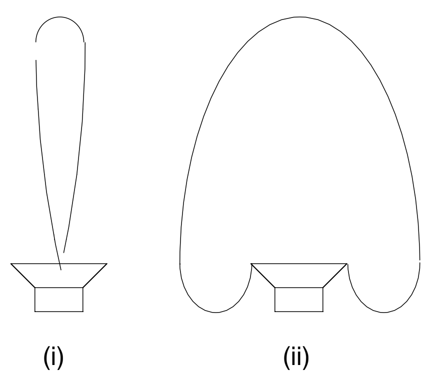


Figure 59 Acoustic beams for (i) low frequency driver and (ii) high frequency driver

You can control resonant modes within the rotating speaker cabinet with the Lo and Hi Resonate parameters. For a realistic rotating speaker, the resonance level and delay excursion should be set quite low. High levels will give wild pitch shifting.

The rotating speaker algorithms give you a great deal of control over the rotation speeds. The direction of rotation (clockwise or counter-clockwise) can be set. A rotating speaker generally has two rotation speeds: fast or slow. There is also a Brake parameter to stop the speakers from rotating. You can set the fast and slow rotation rates in Hz for both the high and low frequency speakers. When you switch between the fast speed and slow speed, the rotating speaker takes time to ramp the speed up or down, just like a real rotating speaker. The time to ramp from slow to fast can be set and a different time can be set for fast to slow.

The low frequency speaker can work in three modes: Normal, NoAccel or Stopped. Finally, the shape of the acceleration itself can be controlled. The acceleration curve parameter produces a constant acceleration (linear change of speed) when set to **0%**. For positive settings, acceleration is slow at slow speed and speeds up for fast speeds. For negative settings, acceleration is fast at slow speeds, then slows down at fast speeds. At the most negative settings, the speed will overshoot the fast rate before settling down (acceleration goes negative when approaching the fast speed).

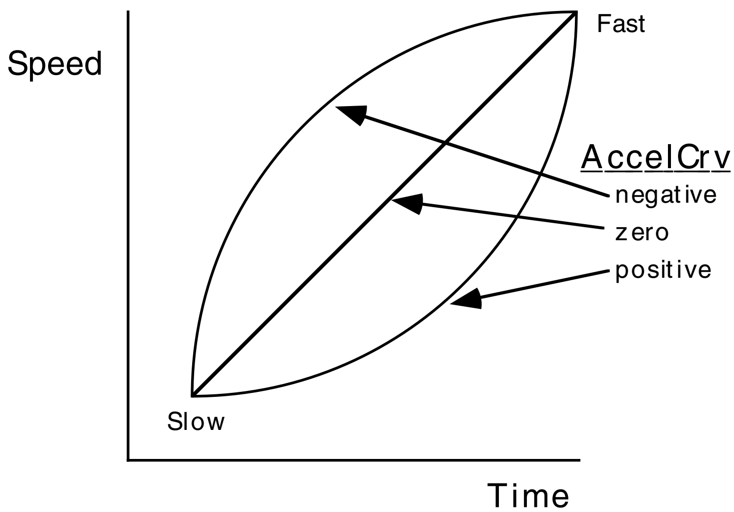


Figure 60 Effect of acceleration curve on speed

Algorithm **298 Big KB3 Effect** is the most full featured of the various rotary algorithms. The algorithm begins with the full vibrato/chorus model and is followed by an amplifier distortion. For details on the distortion, see the description of Algorithm **300 Mono Distortion**. The distorted signal is next passed to a cabinet emulation filter and a pair of crossover filters for band splitting. The measurements of a real Leslie speaker was used in the design of these filters. Default parameter values reflect these measurements, but you may alter them if you like. The Lo HP parameter controls a highpass filter that defines the lowest frequency to pass through the speaker. Likewise the Hi LP parameter is a lowpass filter controlling the highest frequency. The crossover filters for the lower and upper drivers may be set independently. A small amount of overlap seems to work well. The gains of the high and low band signals may also be separately controlled. Finally the signal enters the rotating speaker routine described above.

Parameters (Big KB3 Effect):

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
VibChInOut	In or Out	Dist Drive	0 to 96 dB
Vib/Chor	V1	DistWarmth	8 to 25088 Hz
Roto InOut	In or Out		

Page 2

Speed	Slow or Fast		
Brake	On or Off	Lo Mode	Normal
Lo Gain	Off, -79.0 to 24.0 dB	Hi Gain	Off, -79.0 to 24.0 dB
Lo Xover	8 to 25088 Hz	Hi Xover	8 to 25088 Hz
Lo HP	8 to 25088 Hz	Hi LP	8 to 25088 Hz

Page 3

Lo Size	0 to 250 mm	Hi Size	0 to 250 mm
Lo Trem	0 to 100%	Hi Trem	0 to 100%
Lo Beam W	45.0 to 360.0 deg	Hi Beam W	45.0 to 360.0 deg

Page 4

Lo Slow	0.00 to 2.00 Hz	Hi Slow	0.00 to 2.00 Hz
Lo Fast	3.00 to 10.00 Hz	Hi Fast	3.00 to 10.00 Hz
LoSlow>Fst	0.10 to 10.00 s	HiSlow>Fst	0.10 to 10.00 s
LoFst>Slow	0.10 to 10.00 s	HiFst>Slow	0.10 to 10.00 s
LoAccelCrv	-100 to 100%	HiAccelCrv	-100 to 100%
LoSpinDir	CW or CCW	HiSpinDir	CW or CCW

Page 5

LoResonate	0 to 100%	HiResonate	0 to 100%
Lo Res Dly	10 to 2550 samp	Hi Res Dly	10 to 2550 samp
LoResXcurs	0 to 510 samp	HiResXcurs	0 to 510 samp
ResH/LPhs	0.0 to 360.0 deg		

Page 6

LoMicA Pos	-180.0 to 180.0 deg	LoMicB Pos	-180.0 to 180.0 deg
LoMicA Lvl	0 to 100%	LoMicB Lvl	0 to 100%
LoMicA Pan	-100 to 100%	LoMicB Pan	-100 to 100%
HiMicA Pos	-180.0 to 180.0 deg	HiMicB Pos	-180.0 to 180.0 deg
HiMicA Lvl	0 to 100%	HiMicB Lvl	0 to 100%
HiMicA Pan	-100 to 100%	HiMicB Pan	-100 to 100%

Algorithm **296 VC+Dist+Rotor 4** begins with the full vibrato/chorus model and is followed by amplifier distortion (see Algorithm **300 Mono Distortion on page 153** for more details). The distortion is followed by the rotating speaker model and a cabinet lowpass filter.

Algorithm **290 VibChor+Rotor 2** is similar in design to Algorithm **296 VC+Dist+Rotor 4** but lacks the distortion section and uses a simplified vibrato/chorus model.

Algorithm **297 VC+Tube+Rotor 4** is similar to Algorithm **296 VC+Dist+Rotor 4**, except a different distortion is used—one which faithfully models the response and smooth distortion caused by overloading a vacuum tube circuit. The Tube Drive parameter replaces Dist Drive and DistWarmth and the acoustic beam width for the low driver is unavailable.

Algorithm **646 Mn VC+Dist+Rotor** is a pure mono effect. There are no microphone parameters since the microphones are used for stereo placement. The simple vibrato/chorus model is used as is the amplifier distortion from Algorithm **300 Mono Distortion**. The algorithm lacks the cabinet lowpass filter and the acoustic beam width control for the low driver.

Parameters (VibChor+Rotor 2, VC+Dist+Rotor 4, VC+Tube+Rotor 4, Mn VC+Dist+Rotor):

Page 1 (VibChor+Rotor 2)

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
VibChInOut	In or Out		
Vib/Chor	V1		
Roto InOut	In or Out	Cabinet LP	8 to 25088 Hz

Page 1 (VC+Dist+Rotor 4 and Mn VC+Dist+Rotor)

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
VibChInOut	In or Out	Dist Drive	0.0 to 96.0 dB
Vib/Chor	V1	Dist Warmth	8 to 25088 Hz
Roto InOut	In or Out	Cabinet LP	8 to 25088 Hz

Page 1 (VC+Tube+Rotor 4)

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
VibChInOut	In or Out	Tube Drive	Off, -79.0 to 60.0 dB
Vib/Chor	V1		
Roto InOut	In or Out	Cabinet LP	8 to 25088 Hz

Page 2

Speed	Slow or Fast	Xover	8 to 25088 Hz
Brake	On or Off	Lo Mode	Normal
Lo Gain	Off, -79.0 to 24.0 dB	Hi Gain	Off, -79.0 to 24.0 dB
Lo Size	0 to 250 mm	Hi Size	0 to 250 mm
Lo Trem	0 to 100%	Hi Trem	0 to 100%
Lo Beam W	45.0 to 360.0 deg	Hi Beam W	45.0 to 360.0 deg

Page 3

Lo Slow	0.00 to 2.00 Hz	Hi Slow	0.00 to 2.00 Hz
Lo Fast	3.00 to 10.00 Hz	Hi Fast	3.00 to 10.00 Hz
LoSlow>Fst	0.10 to 10.00 s	HiSlow>Fst	0.10 to 10.00 s
LoFst>Slow	0.10 to 10.00 s	HiFst>Slow	0.10 to 10.00 s
LoAccelCrv	-100 to 100%	HiAccelCrv	-100 to 100%
LoSpinDir	CW or CCW	HiSpinDir	CW or CCW

Page 4

LoResonate	0 to 100%	HiResonate	0 to 100%
Lo Res Dly	10 to 2550 samp	Hi Res Dly	10 to 2550 samp
LoResXcurs	0 to 510 samp	HiResXcurs	0 to 510 samp
ResH/LPhase	0.0 to 360.0 deg		

Page 5

LoMicA Pos	-180.0 to 180.0 deg	LoMicB Pos	-180.0 to 180.0 deg
LoMicA Lvl	0 to 100%	LoMicB Lvl	0 to 100%
LoMicA Pan	-100 to 100%	LoMicB Pan	-100 to 100%
HiMicA Pos	-180.0 to 180.0 deg	HiMicB Pos	-180.0 to 180.0 deg
HiMicA Lvl	0 to 100%	HiMicB Lvl	0 to 100%
HiMicA Pan	-100 to 100%	HiMicB Pan	-100 to 100%

Algorithm **291 Distort + Rotary** models an amplifier distortion followed by a rotating speaker. The rotating speaker has separately controllable tweeter and woofer drivers. The algorithm has three main sections. First, the input stereo signal is summed to mono and may be distorted by a tube amplifier simulation (see Algorithm 300 Mono Distortion for details). The signal is then passed into the rotator section where it is split into high and low frequency bands and the two bands are run through separate rotators. The two bands are recombined and measured at two positions, spaced by a controllable relative angle (microphone simulation) to obtain a stereo signal again. Finally the signal is passed through a speaker cabinet simulation.

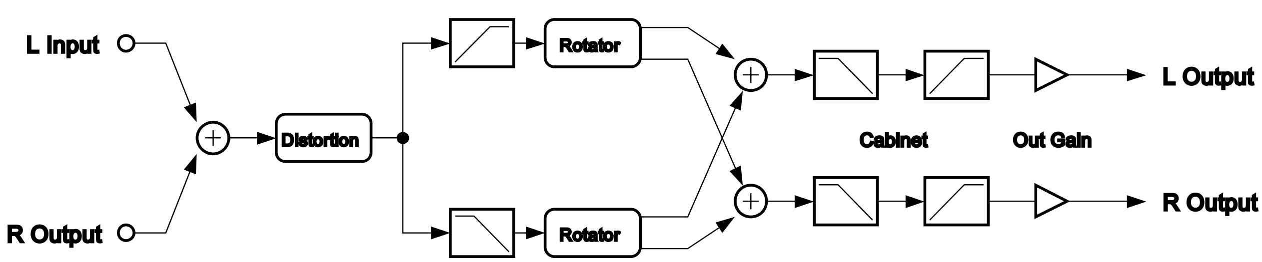


Figure 61 Distort + Rotary

Parameters (Distort + Rotary):

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
Cabinet HP	8 to 25088 Hz	Dist Drive	0 to 96 dB
Cabinet LP	8 to 25088 Hz	DistWarmth	8 to 25088 Hz

Page 2

Speed	Slow or Fast	Xover	8 to 25088 Hz
Brake	On or Off	Lo Mode	Normal
Lo Gain	Off, -79.0 to 24.0 dB	Hi Gain	Off, -79.0 to 24.0 dB
Lo Size	0 to 250 mm	Hi Size	0 to 250 mm
Lo Trem	0 to 100%	Hi Trem	0 to 100%

Page 3

Lo Slow	0.00 to 2.00 Hz	Hi Slow	0.00 to 2.00 Hz
Lo Fast	3.00 to 10.00 Hz	Hi Fast	3.00 to 10.00 Hz
LoSlow>Fst	0.10 to 10.00 s	HiSlow>Fst	0.10 to 10.00 s
LoFst>Slow	0.10 to 10.00 s	HiFst>Slow	0.10 to 10.00 s
LoAccelCrv	-100 to 100%	HiAccelCrv	-100 to 100%
LoSpinDir	CW or CCW	HiSpinDir	CW or CCW

Page 4

LoResonate	0 to 100%	HiResonate	0 to 100%
Lo Res Dly	10 to 2550 samp	Hi Res Dly	10 to 2550 samp
LoResXcurs	0 to 510 samp	HiResXcurs	0 to 510 samp
ResH/LPhs	0.0 to 360.0 deg	Mic Angle	0.0 to 360.0 deg

Algorithm **293 VC+Dist+1Rotor 2** models a single rotating speaker in a two-PAU algorithm. In other respects the algorithm is quite full featured and includes the Hammond vibrato/chorus model, distortion, full control of the rotating speaker model (speed, size for Doppler shift, tremolo, acoustic beam width, cabinet resonance) and microphone positions and panning. You get all the features, but only for one driver. The signal does not get split into a high band and low band and passed through separate drivers.

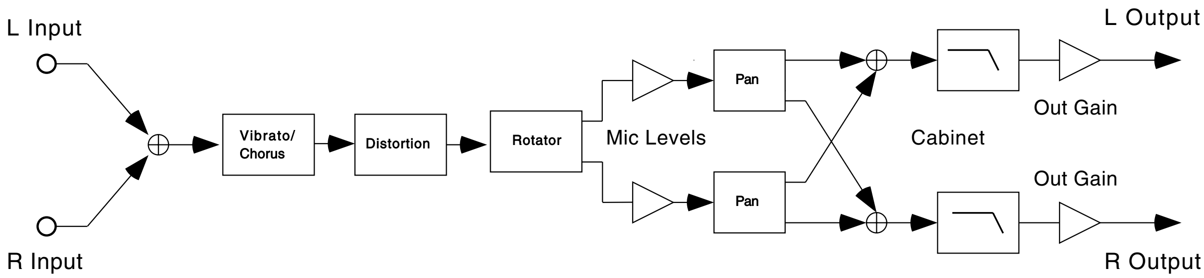


Figure 62 VC+Dist+1Rotor 2

Parameters (VC+Dist+1Rotor 2):

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
VibChInOut	In or Out	Dist Drive	0.0 to 96.0 dB
Vib/Chor	V1	DistWarmth	8 to 25088 Hz
Roto InOut	In or Out	Cabinet LP	8 to 25088 Hz

Page 2

Gain	Off, -79.0 to 24.0 dB		
Rate	-10.00 to 10.00 Hz		
Size	0 to 250 mm		
Trem	0 to 100%		
Beam W	45.0 to 360.0 deg		

Page 3

Mic A Pos	-180.0 to 180.0 deg	Mic B Pos	-180.0 to 180.0 deg
Mic A Lvl	0 to 100%	Mic B Lvl	0 to 100%
Mic A Pan	-100 to 100%	Mic B Pan	-100 to 100%

Page 4

Resonate	0 to 100%		
Res Dly	10 to 2550 samp		
Res Xcurs	0 to 510 samp		
Res Phs	0.0 to 360.0 deg		

Algorithm **292 VC+Dist+HiLoRotr** gives you a model of the Hammond vibrato/chorus, distortion and the band splitting for high and low frequency drivers. To pack all this into a two-PAU algorithm, a few sacrifices had to be made to the list of parameters for the rotating speaker model. So what's missing? The resonance controls for the low frequency driver are gone. There is no control of the acoustic beam width for the low driver. The microphone panning is gone and there is a single microphone level control for the A and B microphones. The distortion used is a smaller version of Algorithm **303 PolyDistort + EQ**. Even with fewer features, this algorithm gives a convincing Leslie effect while allowing space for more algorithms on other buses.

Algorithm **294 VC+Dist+HiLoRot2** makes different tradeoffs than Algorithm **292 VC+Dist+HiLoRotr**. The distortion is the same as used in Algorithm **300 Mono Distortion**. This distortion uses more processor resources than the **PolyDistort + EQ**, so **VC+Dist+HiLoRot2** does not include the acoustic beam width control for either the high or low frequency drivers. The signal flow is the same as for Algorithm **292 VC+Dist+HiLoRotr**.

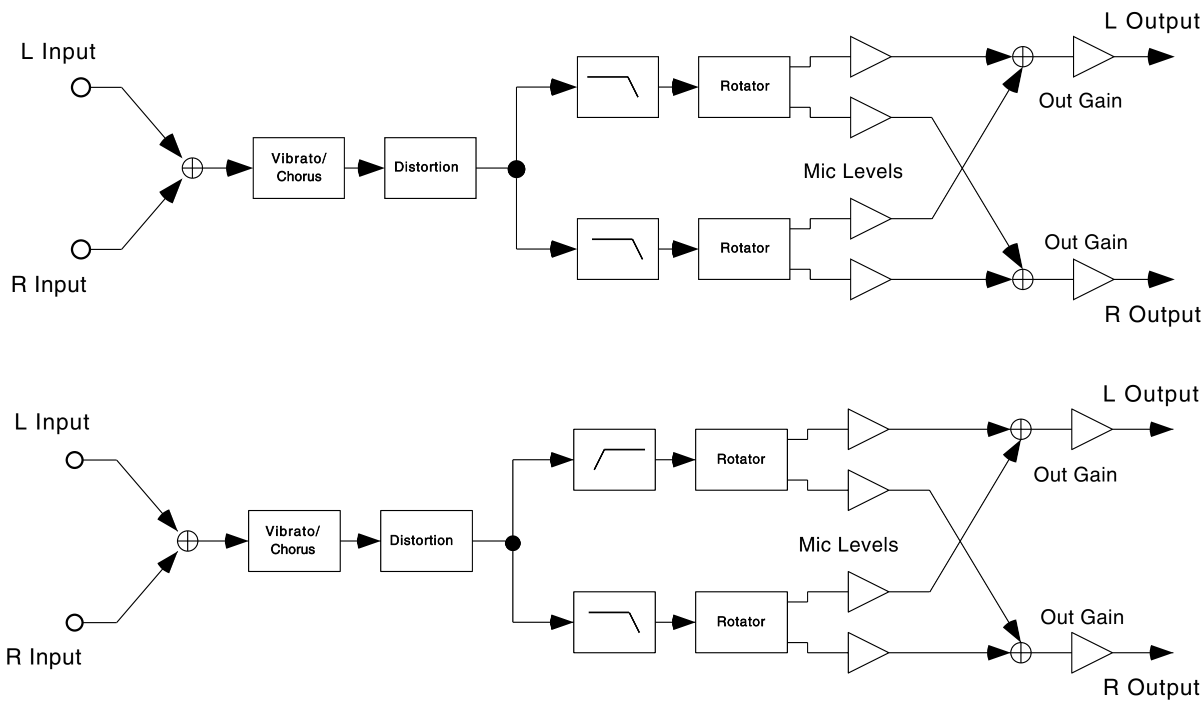


Figure 63 VC+Dist+HiLoRotr and VC+Dist+HiLoRot2

Parameters (VC+Dist+HiLoRotr and VC+Dist+HiLoRot2):

Page 1 (VC+Dist+HiLoRotr)

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
VibChInOut	In or Out	Dist Drive	Off, -79.0 to 48.0 dB
Vib/Chor	V1	Dist Curve	0 to 127%
Roto InOut	In or Out	DistLPFreq	8 to 25088 Hz

Page 1   (VC+Dist+HiLoRot2)

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
VibChInOut	In or Out	Dist Drive	0.0 to 96.0 dB
Vib/Chor	V1	Dist Warmth	8 to 25088 Hz
Roto InOut	In or Out		

Page 2

Speed	Slow or Fast	Xover	8 to 25088 Hz
Brake	On or Off	Lo Mode	Normal
Lo Size	0 to 250 mm	Hi Size	0 to 250 mm
Lo Trem	0 to 100%	Hi Trem	0 to 100%
		Hi Beam W	45.0 to 360.0 deg

Page 3

Lo Slow	0.00 to 2.00 Hz	Hi Slow	0.00 to 2.00 Hz
Lo Fast	3.00 to 10.00 Hz	Hi Fast	3.00 to 10.00 Hz
LoSlow>Fst	0.10 to 10.00 s	HiSlow>Fst	0.10 to 10.00 s
LoFst>Slow	0.10 to 10.00 s	HiFst>Slow	0.10 to 10.00 s
LoAccelCrv	-100 to 100%	HiAccelCrv	-100 to 100%
LoSpinDir	CW or CCW	HiSpinDir	CW or CCW

Page 4

		LoMic Lvls	0 to 100%
HiResonate	0 to 100%	LoMicA Pos	-180.0 to 180.0 deg
Hi Res Dly	10 to 2550 samp	LoMicB Pos	-180.0 to 180.0 deg
HiResXcurs	0 to 510 samp	HiMic Lvls	0 to 100%
		HiMicA Pos	-180.0 to 180.0 deg
Res HiPhs	0.0 to 360.0 deg	HiMicB Pos	-180.0 to 180.0 deg

Algorithm **295 Rotor 1** is a rotating speaker model on a budget. Its most attractive feature is its small size (one PAU). Obviously a few things had to be scaled back. There is no vibrato/chorus model and no distortion control. There is only a single rotating driver rather than a pair for high and low frequency bands. Aside from these omissions, the rotating speaker model is quite full featured. It includes full control of the rotating speaker including speed, size for Doppler shift, tremolo, acoustic beam width, cabinet lowpass filter and resonance and full microphone control for two microphone positions.

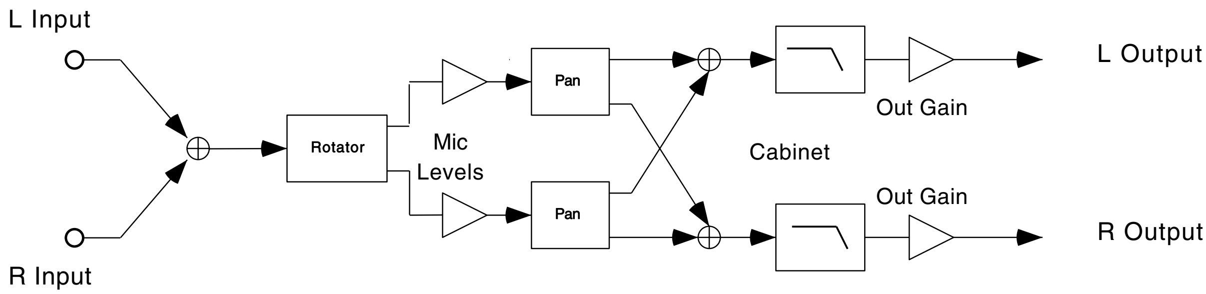


Figure 64 Rotor 1

Parameters (Rotor 1):

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
		Cabinet LP	8 to 25088 Hz

Page 2

Speed	Slow or Fast	Slow Rate	0.00 to 2.00 Hz
Brake	On or Off	Fast Rate	3.00 to 10.00 Hz
Gain	Off, -79.0 to 24.0 dB	Slow>Fast	0.10 to 10.00 s
Size	0 to 250 mm	Fast>Slow	0.10 to 10.00 s
Trem	0 to 100%	AccelCurve	-100 to 100 %
Beam W	45.0 to 360.0 deg	SpinDirec	CW or CCW

Page 3

		Mic A Pos	-180.0 to 180.0 deg
Resonate	0 to 100%	Mic A Lvl	0 to 100%
Res Dly	10 to 2550 samp	Mic A Pan	-100 to 100%
Res Xcurs	0 to 510 samp	Mic B Pos	-180.0 to 180.0 deg
		Mic B Lvl	0 to 100%
Res Phs	0.0 to 360.0 deg	Mic B Pan	-100 to 100%

In/Out When set to **In**, the algorithm is active; when set to **Out** the algorithm is bypassed.

Out Gain The overall gain or amplitude at the output of the effect. For distortion, it is often necessary to turn the output gain down as the distortion drive is turned up.

VibChInOut When set to **In** the vibrato/chorus is active; when set to **Out** the vibrato/chorus is bypassed.

Vib/Chor	This control sets the Hammond B3 vibrato/chorus. There are six settings for this effect: three vibratos **V1**, **V2**, and **V3**, and three choruses **C1**, **C2**, and **C3**.
Roto InOut	When set to **In** the rotary speaker is active; when set to **Out** the rotary speaker is bypassed.
Dist Drive or Tube Drive	Applies a boost to the input signal to overdrive the distortion algorithm. When overdriven, the distortion algorithm will soft-clip the signal. Since distortion drive will make your signal very loud, you may have to reduce the Out Gain as the drive is increased.
Dist Curve	Controls the curvature of the distortion. **0**% is no curvature (no distortion at all). At **100**%, the curve bends over smoothly and becomes perfectly flat right before it goes into clipping.
DistWarmth	A lowpass filter in the distortion control path. This filter may be used to reduce some of the harshness of some distortion settings without reducing the bandwidth of the signal.
DistLPFreq	Controls one-pole lowpass filters in the **PolyDistort + EQ** (in **VC+Dist+HiLoRotr**). Without the lowpass filters, the sound tends to be too bright and raspy. With less distortion drive, less filtering is needed. If you turn off the distortion curve (set to **0**%), you should turn off the lowpass filter by setting it to the highest frequency.
Cabinet LP	A lowpass filter to simulate the band-limiting of a speaker cabinet. The filter controls the upper frequency limit of the output.
Xover	The frequency at which high and low frequency bands are split and sent to separate rotating drivers.
Lo Xover	The Big KB3 Effect has separate high and low crossover frequency controls. Lo Xover controls a lowpass filter.
Hi Xover	The Big KB3 Effect has separate high and low crossover frequency controls. Hi Xover controls a highpass filter.
Speed	Sets the rotating speakers to run at either the slow rate or the fast rate.
Brake	When set to **On**, the rotating speakers will slow to a halt.
Lo Mode	In the **Normal** setting, you will have full control of the low frequency speaker with the Speed parameter. The **NoAccel** setting will hold the low frequency speaker at the slow speed, and the Speed parameter will have no effect on its speed, though Brake will still work. In the **Stopped** position, the low frequency speaker will not spin at all.
Lo Slow **Hi Slow**	The rotation rate in hertz (Hz) of the speaker when Speed is set to **Slow**.
Lo Fast **Hi Fast**	The rotation rate in hertz (Hz) of the speaker when Speed is set to **Fast**.
LoSlow>Fst **HiSlow>Fst**	The time for the speaker to accelerate from slow speed to fast speed.
LoFst>Slow **HiFst>Slow**	The time for the speaker to decelerate from fast speed to slow speed.

LoAccelCrv **HiAccelCrv**	The shape of the acceleration curve for the speaker. **0%** is a constant acceleration. Positive values cause the speaker to speed up slowly at first then quickly reach the fast rate. Negative values cause a quick initial speed-up then slowly settle in to the fast speed—overshoot is possible. See the discussion in the main text above.
LoSpinDir **HiSpinDir**	The direction of rotation of the speaker. The choice is clockwise (**CW**) or counter-clockwise (**CCW**).

Lo Gain The gain or amplitude of the signal passing through the rotating woofer (low frequency) driver.

Lo Size The effective size (radius of rotation) of the rotating woofer in millimeters. Affects the amount of Doppler shift or vibrato of the low frequency signal.

Lo Trem Controls the depth of tremolo of the low frequency signal. Expressed as a percentage of full scale tremolo.

Lo Beam W The rotating speaker effect attempts to model a rotating woofer for the low frequency driver. The acoustic radiation pattern of a woofer tends to range from omnidirectional (radiates in directions in equal amounts) to a wide beam. You may adjust the beam width from **45°** to **360°**. If you imagine looking down on the rotating speaker, the beam angle is the angle between the -6 dB levels of the beam. At **360°**, the woofer is omnidirectional.

Hi Gain The gain or amplitude of the signal passing through the rotating tweeter (high frequency) driver.

Hi Size The effective size (radius of rotation) of the rotating tweeter in millimeters. Affects the amount of Doppler shift or vibrato of the high frequency signal.

Hi Trem Controls the depth of tremolo of the high frequency signal. Expressed as a percentage of full scale tremolo.

Hi Beam W The rotating speaker effect attempts to model a rotating horn for the high frequency driver. The acoustic radiation pattern of a horn tends to be a narrow beam. You may adjust the beam width from **45°** to **360°**. If you imagine looking down on the rotating speaker, the beam angle is the angle between the -6 dB levels of the beam. At **360°**, the horn is omnidirectional (radiates in all directions equally).

Lo HP A highpass filter in **Big KB3 Effect** to simulate the band-limiting of a speaker cabinet. The filter controls the lower frequency limit of the output.

Hi LP A lowpass filter in **Big KB3 Effect** to simulate the band-limiting of a speaker cabinet. The filter controls the upper frequency limit of the output.

Mic Pos The angle of the virtual microphones in degrees from the “front” of the rotating speaker. This parameter is not well suited to modulation because adjustments to it result in large sample skips (audible as clicks when signal is passing through the effect). There are four of these parameters to include two pairs (A and B) for high and low frequency drivers.

Mic Lvl The level of the virtual microphone signal being sent to the output. There are four of these parameters to include two pairs (A and B) for high and low frequency drivers.

Mic Pan Left-right panning of the virtual microphone signals. A setting of **-100%** is panned fully left, and **100%** is panned fully right. There are four of these parameters to include two pairs (A and B) for high and low frequency drivers.

LoResonate	A simulation of cabinet resonant modes express as a percentage. For realism, you should use very low settings. This is for the low frequency signal path.
Lo Res Dly	The number of samples of delay in the resonator circuit in addition to the rotation excursion delay. This is for the low frequency signal path.
LoResXcurs	The number of samples of delay to sweep through the resonator at the rotation rate of the rotating speaker. This is for the low frequency signal path.
HiResonate	A simulation of cabinet resonant modes express as a percentage. For realism, you should use very low settings. This is for the high frequency signal path.
Hi Res Dly	The number of samples of delay in the resonator circuit in addition to the rotation excursion delay. This is for the high frequency signal path.
HiResXcurs	The number of samples of delay to sweep through the resonator at the rotation rate of the rotating speaker. This is for the high frequency signal path.
ResH/LPhs	This parameter sets the relative phases of the high and low resonators. The angle value in degrees is somewhat arbitrary and you can expect the effect of this parameter to be rather subtle.

Distortion

300 Mono Distortion
301 MonoDistort+Cab
302 MonoDistort + EQ
304 StereoDistort+EQ
650 Mn Distortion
651 Mn Distort+Cab
652 Mn Distort + EQ

Small distortion algorithms

PAUs: 1 for **Mono Distortion**
2 for **MonoDistort+Cab**
2 for **MonoDistort + EQ**
3 for **StereoDistort+EQ**

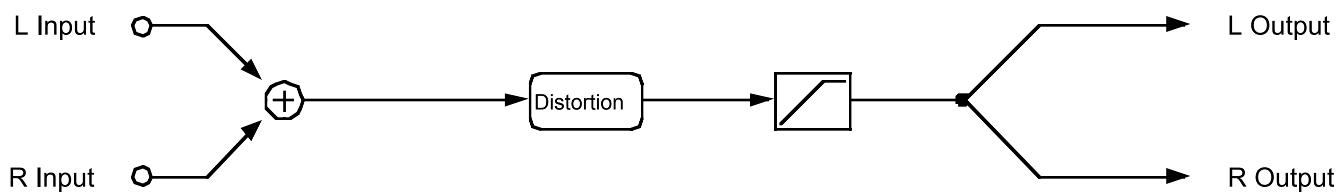


Figure 65 **Block diagram of Mono Distortion**

Mono Distortion sums its stereo input to mono, performs distortion followed by a highpass filter and sends the result as centered stereo.

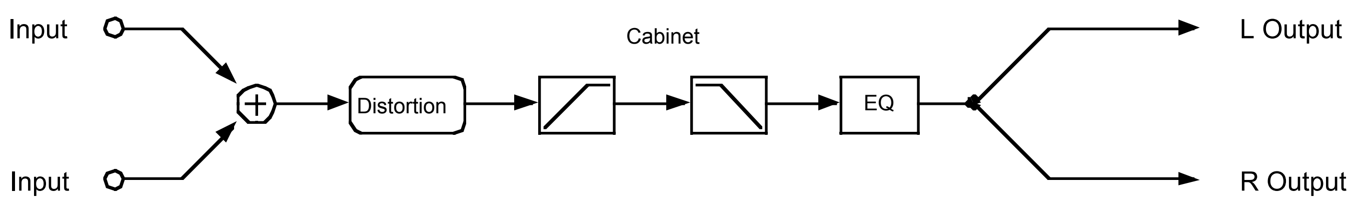


Figure 66 **Block diagram of MonoDistort + EQ**

MonoDistort + EQ is similar to **Mono Distortion** except the single highpass filter is replaced with a pair of second-order highpass/lowpass filters to provide rudimentary speaker cabinet modeling. The highpass and lowpass filters are then followed by an EQ section with bass and treble shelf filters and two parametric mid filters.

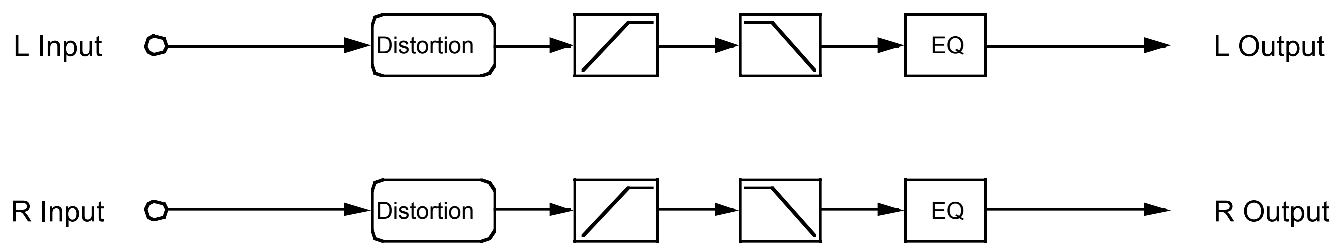


Figure 67 Block diagram of StereoDistort+EQ

StereoDistort+EQ processes the left and right channels separately, though there is only one set of parameters for both channels. The stereo distortion has only 1 parametric mid filter.

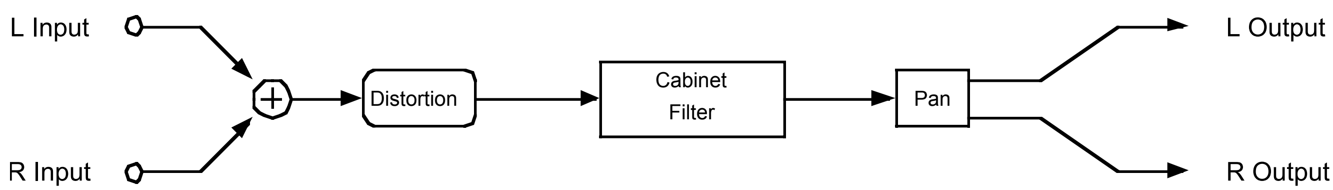


Figure 68 Block diagram of MonoDistort+Cab

The distortion segment of **MonoDistort+Cab** is similar to **Mono Distortion** except the highpass is replaced by a full speaker cabinet model (the same as in Algorithm **284 Cabinet**). There is also a panner to route the mono signal between left and right outputs. In **MonoDistort+Cab**, the distortion is followed by a model of a guitar amplifier cabinet. The model can be bypassed, or there are 8 presets which were derived from measurements of real cabinets.

The distortion algorithm will soft clip the input signal. The amount of soft clipping depends on how high the distortion drive parameter is set. Soft clipping means that there is a smooth transition from linear gain to saturated overdrive. Higher distortion drive settings cause the transition to become progressively sharper or "harder." The distortion never produces hard or digital clipping, but it does approach it at high drive settings. When you increase the distortion drive parameter you are increasing the gain of the algorithm until the signal reaches saturation. You will have to compensate for increases in drive gain by reducing the output gain. These algorithm will not digitally clip unless the output gain is over-driven.

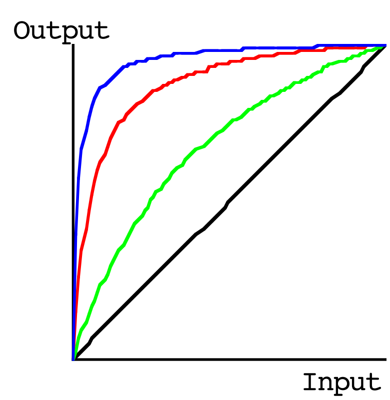


Figure 69 Input/Output transfer characteristic of soft clipping at various drive settings

Signals that are symmetric in amplitude (they have the same shape if they are inverted, positive for negative) will usually produce odd harmonic distortion. For example, a pure sine wave will produce smaller copies of itself at 3, 5, 7, etc. times the original frequency of the sine wave. In **MonoDistort + EQ**, a DC offset may be added to the signal to break the amplitude symmetry and will cause the distortion to produce even harmonics. This can add a "brassy" character to the distorted sound. The DC offset added prior to distortion gets removed at a later point in the algorithm.

Parameters for Mono Distortion

Wet/Dry	0 to 100 %wet	Out Gain	Off, -79.0 to 24.0 dB
Dist Drive	0 to 96 dB		
Warmth	8 to 25088 Hz		
Highpass	8 to 25088 Hz		

Parameters for MonoDistort+Cab

Wet/Dry	0 to 100 %wet	Out Gain	Off, -79.0 to 24.0 dB
Dist Drive	0 to 96 dB		
Warmth	8 to 25088 Hz	Cab Bypass	In or Out
		Cab Preset	Plain

Parameters for MonoDistort + EQ

Page 1

Wet/Dry	0 to 100 %wet	Out Gain	Off, -79.0 to 24.0 dB
Dist Drive	0 to 96 dB		
Warmth	8 to 25088 Hz	dc Offset	-100 to 100 %
Cabinet HP	8 to 25088 Hz	Cabinet LP	8 to 25088 Hz

Page 2

Bass Gain	-79.0 to 24.0 dB	Treb Gain	-79.0 to 24.0 dB
Bass Freq	8 to 25088 Hz	Treb Freq	8 to 25088 Hz
Mid1 Gain	-79.0 to 24.0 dB	Mid2 Gain	-79.0 to 24.0 dB
Mid1 Freq	8 to 25088 Hz	Mid2 Freq	8 to 25088 Hz
Mid1 Width	0.010 to 5.000 oct	Mid2 Width	0.010 to 5.000 oct

Parameters for StereoDistort+EQ

Page 1

Wet/Dry	0 to 100 %wet	Out Gain	Off, -79.0 to 24.0 dB
Dist Drive	0 to 96 dB	Dist Atten	Off, -79.0 to 0.0 dB
Warmth	8 to 25088 Hz	Dist Invrt	In or Out
Cabinet HP	8 to 25088 Hz	Cabinet LP	8 to 25088 Hz

Page 2

Bass Gain	-79.0 to 24.0 dB	Treb Gain	-79.0 to 24.0 dB
Bass Freq	8 to 25088 Hz	Treb Freq	8 to 25088 Hz
Mid Gain	-79.0 to 24.0 dB		
Mid Freq	8 to 25088 Hz		
Mid Width	0.010 to 5.000 oct		

Wet/Dry The amount of distorted (wet) signal relative to unaffected (dry) signal.

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

Dist Drive Applies a boost to the input signal to overdrive the distortion algorithm. When overdriven, the distortion algorithm will soft-clip the signal. Since distortion drive will make your signal very loud, you may have to reduce the Out Gain as the drive is increased.

Dist Atten The amount to reduce the distorted (wet) signal without affecting the dry signal. For distortion, it is often necessary to turn the distortion attenuation down as the distortion drive is turned up.

Dist Invrt When set to "In", the distorted signal is polarity inverted.

Warmth A lowpass filter in the distortion control path. This filter may be used to reduce some of the harshness of some distortion settings without reducing the bandwidth of the signal.

Cab Bypass For **MonoDistort+Cab**. The guitar amplifier cabinet simulation may be bypassed. When set to **In**, the cabinet simulation is active; when set to **Out**, there is no cabinet filtering.

Cab Preset For **MonoDistort+Cab**. Eight preset cabinets have been created based on measurements of real guitar amplifier cabinets. The presets are Plain, Lead 12, 2x12, Open 12, Open 10, 4x12, Hot 2x12, and Hot 12.

Highpass For **Mono Distortion**. Allows you to reduce the bass content of the distortion content. If you need more filtering to better simulate a speaker cabinet, you will have to choose a larger distortion algorithm.

Cabinet HP For **MonoDistort + EQ** and **StereoDistort+EQ**. A highpass filter that controls the low frequency limit of a simulated loudspeaker cabinet.

Cabinet LP For **MonoDistort + EQ** and **StereoDistort+EQ**. A lowpass filter that controls the high frequency limit of a simulated loudspeaker cabinet.

Bass Gain For **MonoDistort + EQ** and **StereoDistort+EQ**. The amount of boost or cut that the bass shelving filter should apply to the low frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the bass signal below the specified frequency. Negative values cut the bass signal below the specified frequency.

Bass Freq For **MonoDistort + EQ** and **StereoDistort+EQ**. The center frequency of the bass shelving filter in intervals of one semitone.

Treb Gain For **MonoDistort + EQ** and **StereoDistort+EQ**. The amount of boost or cut that the treble shelving filter should apply to the high frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the treble signal above the specified frequency. Negative values cut the treble signal above the specified frequency.

Treb Freq For **MonoDistort + EQ** and **StereoDistort+EQ**. The center frequency of the treble shelving filter in intervals of one semitone.

Mid Gain For **MonoDistort + EQ** and **StereoDistort+EQ**. The amount of boost or cut that the mid parametric filter should apply in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the signal at the specified frequency. Negative values cut the signal at the specified frequency.

Mid Freq For **MonoDistort + EQ** and **StereoDistort+EQ**. The center frequency of the mid parametric filter in intervals of one semitone. The boost or cut will be at a maximum at this frequency.

Mid Wid For **MonoDistort + EQ** and **StereoDistort+EQ**. The bandwidth of the mid parametric filter may be adjusted. You specify the bandwidth in octaves. Small values result in a very narrow filter response. Large values result in a very broad response.

303 PolyDistort + EQ

Eight-stage distortion followed by equalization

PAUs: 2

PolyDistort + EQ is a distortion algorithm followed by equalization. The algorithm consists of an input gain stage, and then eight cascaded distortion stages. Each stage is followed by a one pole lowpass filter. There is also a one-pole lowpass filter in front of the first stage. After the distortion there is a four-band EQ section: Bass, Treble, and two Parametric Mids.

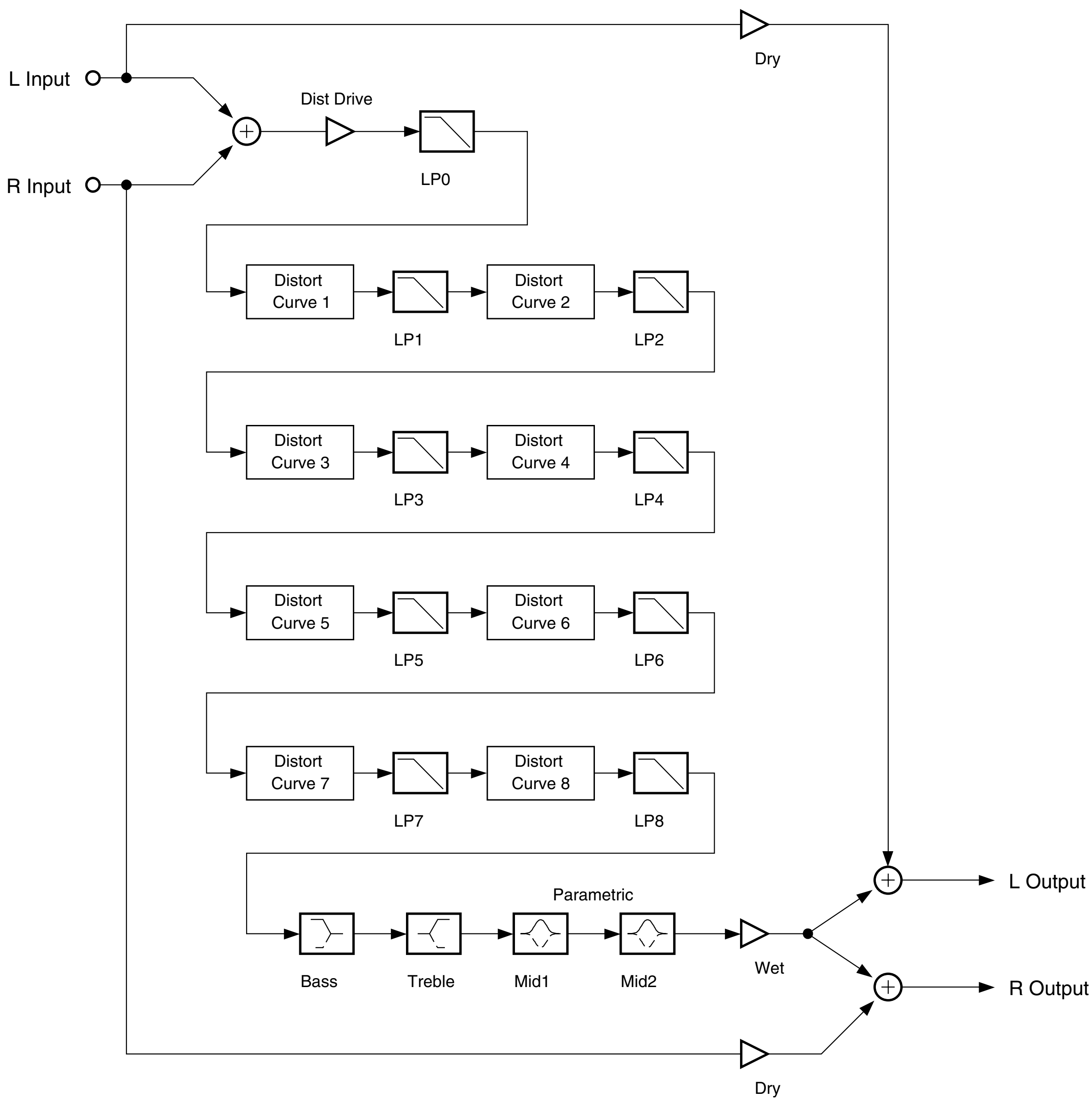


Figure 70 **Block diagram of PolyDistort + EQ**

PolyDistort + EQ is an unusual distortion algorithm that provides a great number of parameters to build a distortion sound from the ground up. The eight distortion stages each add a small amount of distortion to your sound. Taken together, you can get a very harsh heavy metal sound. Between each distortion stage is a lowpass filter. The lowpass filters work with the distortion stages to help mellow out the sound. Without any lowpass filters the distortion will get very harsh and raspy.

Stages of distortion can be removed by setting the Curve parameter to **0**. You can then do a 6-, 4-, or 2-stage distortion algorithm. The corresponding lowpasses should be turned off if there is no distortion in a section. More than four stages seem necessary for lead guitar sounds. For a cleaner sound, you may want to limit yourself to only four stages.

Once you have set up a distorted sound you are satisfied with, the Dist Drive parameter controls the input gain to the distortion, providing a single parameter for controlling distortion amount. You will probably find that you will have to cut back on the output gain as you drive the distortion louder.

Post-distortion EQ is definitely needed for make things sound right. This should be something like a guitar speaker cabinet simulator, although not exactly, since we are already doing a lot of lowpass filtering inside the distortion itself. Possible EQ settings you can try are Treble **-20 dB** at **5 kHz**, Bass **-6 dB** at **100 Hz**, Mid1, wide, **+6 dB** at **2 kHz**, Mid2, wide, **+3 dB** at **200 Hz**, but of course you should certainly experiment to get your sound. The Treble is helping to remove raspiness, the Bass is removing the extreme low end like an open-back guitar cabinet (not that guitar speaker have that much low end anyway), Mid1 adds enough highs so that things can sound bright even in the presence of all the HF roll-off, and Mid2 adds some warmth. Your favorite settings will probably be different. Boosting the Treble may not be a good idea.

Pre-distortion EQ, available on the Studio INPUT page, is also useful for shaping the sound. EQ done in front of the distortion will not be heard as simple EQ, because the distortion section makes an adjustment in one frequency range felt over a much wider range due to action of the distortion. Simple post EQ is a bit too obvious for the ear, and it can get tired of it after a while.

Parameters for PolyDistort + EQ

Parameters

Page 1

Wet/Dry	0 to 100 %wet	Out Gain	Off, -79.0 to 24.0 dB
Dist Drive	Off, -79.0 to 48.0 dB	Wet Gain	Off, -79.0 to 0.00 dB
		Invert	In or Out

Page 2

Curve 1	0 to 127 %	Curve 5	0 to 127 %
Curve 2	0 to 127 %	Curve 6	0 to 127 %
Curve 3	0 to 127 %	Curve 7	0 to 127 %
Curve 4	0 to 127 %	Curve 8	0 to 127 %

Page 3

LP0 Freq	8 to 25088 Hz		
LP1 Freq	8 to 25088 Hz	LP5 Freq	8 to 25088 Hz
LP2 Freq	8 to 25088 Hz	LP6 Freq	8 to 25088 Hz
LP3 Freq	8 to 25088 Hz	LP7 Freq	8 to 25088 Hz
LP4 Freq	8 to 25088 Hz	LP8 Freq	8 to 25088 Hz

Page 4

Bass Gain	-79.0 to 24.0 dB	Treb Gain	-79.0 to 24.0 dB
Bass Freq	8 to 25088 Hz	Treb Freq	8 to 25088 Hz
Mid1 Gain	-79.0 to 24.0 dB	Mid2 Gain	-79.0 to 24.0 dB
Mid1 Freq	8 to 25088 Hz	Mid2 Freq	8 to 25088 Hz
Mid1 Width	0.010 to 5.000 oct	Mid2 Width	0.010 to 5.000 oct

Wet/Dry This is a simple mix of the distorted signal relative to the dry undistorted input signal.

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

Dist Drive Applies gain to the input prior to distortion. It is the basic “distortion drive” control. Anything over 0 dB could clip. Normally clipping would be bad, but the distortion algorithm tends to smooth things out. Still, considering that for some settings of the other parameters you would have to back off the gain to **-48 dB** in order to get a not very distorted sound for full scale input, you should go easy on this amount.

Dist Atten The amount to reduce the distorted (wet) signal without affecting the dry signal. For distortion, it is often necessary to turn the distortion attenuation down as the distortion drive is turned up.

Dist Invrt When set to “In”, the distorted signal is polarity inverted.

Curve *n* The curvature of the individual distortion stages. **0**% is no curvature (no distortion at all). At **100**%, the curve bends over smoothly and becomes perfectly flat right before it goes into clipping.

LP *n* Freq These are the one pole lowpass controls. LP0 Freq handles the initial lowpass prior to the first distortion stage. The other lowpass controls follow their respective distortion stages. With all lowpasses out of the circuit (set to the highest frequency), the sound tends to be too bright and raspy. With less distortion drive, less filtering is needed. If you turn off a distortion stage (set to **0**%), you should turn off the lowpass filter by setting it to the highest frequency.

Bass Gain The amount of boost or cut that the bass shelving filter should apply to the low frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the bass signal below the specified frequency. Negative values cut the bass signal below the specified frequency.

Bass Freq The center frequency of the bass shelving filter in intervals of one semitone.

Treb Gain The amount of boost or cut that the treble shelving filter should apply to the high frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the treble signal above the specified frequency. Negative values cut the treble signal above the specified frequency.

Treb Freq The center frequency of the treble shelving filter in intervals of one semitone.

Mid Gain The amount of boost or cut that the mid parametric filter should apply in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the signal at the specified frequency. Negative values cut the signal at the specified frequency.

Mid Freq The center frequency of the mid parametric filter in intervals of one semitone. The boost or cut will be at a maximum at this frequency.

Mid Wid The bandwidth of the mid parametric filter may be adjusted. You specify the bandwidth in octaves. Small values result in a very narrow filter response. Large values result in a very broad response.

305 Subtle Distort

Adds small amount of distortion to signal.

PAUs: 1

Use **Subtle Distort** to apply small amounts of distortion to a signal. The distortion characteristic is set with the Curvature and EvenOrders parameters. Increasing Curvature increases the distortion amount while EvenOrders increases the asymmetry of the distortion, adding even distortion harmonics. The distorted signal then is sent through two one-pole lowpass filters and added to the dry input signal. The lowpass filters can reduce any harshness from the raw distortion operation. The Dry In/Out is provided as a utility to audition the distortion signal in the absence of dry signal. Out Gain and Dist Gain can be adjusted together to match the level of the bypassed (dry only) signal. Adding distortion to the dry signal will increase the output level unless Out Gain is reduced.

Parameters:

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
Dry In/Out	In or Out	Dist Gain	Off, -79.0 to 0.0 dB
Curvature	0 to 100 %	Dist LP A	8 to 25088 Hz
EvenOrders	0 to 100 %	Dist LP B	8 too25088 Hz

In/Out When set to **In** the distortion is active; when set to **Out** the distortion is bypassed.

Dry In/Out Utility parameter to listen to distortion without the dry signal.

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

Dist Gain The gain or amplitude of the distorted signal path prior to passing through the Out Gain adjustment.

Curvature The amount of distortion; none at **0**% and maximum at **100**%.

Even Orders The asymmetry of the distortion (number of even harmonics); none at **0**% and maximum at **100**%.

Dist LP A Frequency of Lowpass Filter A.

Dist LP B Frequency of Lowpass Filter B.

306 Super Shaper
653 Mn Super Shaper

Ridiculous shaper

PAUs: 1

The **Super Shaper**algorithm packs 2-1/2 times the number of shaping loops, and 8 times the gain of the V.A.S.T. shaper. Refer to the appendices in the KSP8 *User's Guide* for an overview of V.A.S.T. shaper.

Setting **Super Shaper** amount under **1.00x** produces the same nonlinear curve found in the V.A.S.T. shaper. At values above **1.00x** where the V.A.S.T. shaper will pin at zero, the **Super Shaper** provides 6 more sine intervals before starting to zero-pin at **2.50x**. The maximum shaper amount for **Super Shaper** is **32.00x**.

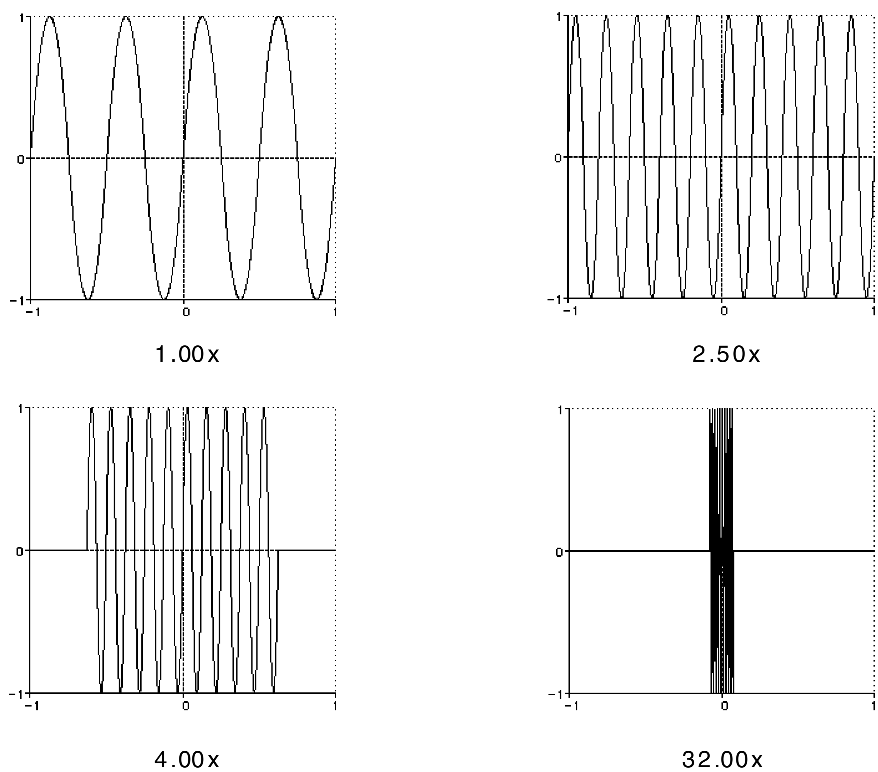


Figure 71 Super Shaper: four values of the Amount parameter

Parameters

Wet/Dry	-100 to 100%	Out Gain	Off, -79.0 to 24.0 dB
Amount	0.10 to 32.00 x		

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.

Amount Adjusts the shaper intensity.

307 3 Band Shaper
654 Mn 3 Band Shaper

A three-band shaper

PAUs: 2

The **3 Band Shaper** non-destructively splits the input signal into 3 separate bands using 1 pole (6dB/oct) filters, and applies a V.A.S.T.-type shaper to each band separately. Refer to the appendices in the KSP8 *User's Guide* for an overview of V.A.S.T. shaping. The cutoff frequencies for these filters are controlled with the CrossOver1 and CrossOver2 parameters. The low band contains frequencies from 0 Hz (DC) to the lower of the two CrossOver settings. The mid band contains frequencies between the two selected frequencies, and the hi band contains those from the higher of the two CrossOver settings, up to 24kHz.

Each frequency band has an enable switch for instantly bypassing any processing for that band, and a Mix control for adjusting the level of each band that is mixed at the output. negative Mix values polarity invert that band. The shaper Amt controls provide the same type of shaping as V.A.S.T. shapers, but can go to **6.00x**.

Parameters

Page 1

Wet/Dry	-100 to 100%	Out Gain	Off, -79.0 to 24.0 dB
CrossOver1	17 to 25088 Hz		
CrossOver2	17 to 25088 Hz		

Page 2

Lo Enable	On or Off	Lo Enable	On or Off
Lo Amt	0.10 to 6.00x	Lo Amt	0.10 to 6.00x
Lo Mix	-100 to 100%	Lo Mix	-100 to 100%
Mid Enable	On or Off		
Mid Amt	0.10 to 6.00x		
Mid Mix	-100 to 100%		

In/Out When set to **In** the effect is active; when set to **Out** the effect is bypassed.

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

CrossOver1 Adjusts one of the -6dB crossover points at which the input signal will be divided into the high, mid and low bands.

CrossOver2 Adjusts the other -6dB crossover points at which the input signal will be divided into the high, mid and low bands.

Enable Low, Mid, and High. Turns processing for each band on or off. Turning each of the three bands. **Off** results in a dry output signal.

Amt Low, Mid, and High. Adjusts the shaper intensity for each band.

Mix Low, Mid, and High. Adjusts the level that each band is summed together as the wet signal. Negative values polarity invert the particular bands signal.

308 Quantize+Alias
655 MnQuantize+Alias

Digital quantization followed by simulated aliasing.

PAUs: 1

The **Quantize+Alias** algorithm offers some of the worst artifacts that digital has to offer! Digital audio engineers will go to great lengths to remove, or at least hide the effects of digital quantization distortion and sampling aliasing. In **Quantize+Alias** we do quite the opposite, making both quantization and aliasing in-your-face effects. The quantizer will give your sound a dirty, grundgy, perhaps industrial sound. The aliasing component simulates the effect of having sampled a sound without adequately band limiting the signal (anti-alias filtering).

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. An 18-bit digital-to-analog converter (DAC) like the one in the K2600 can interpret 262,144 different amplitude levels (2^{18}).

Let’s take a look at how finite precision of digital words affects audio signals. The figures below are plots of a decaying sine wave with varying word lengths.

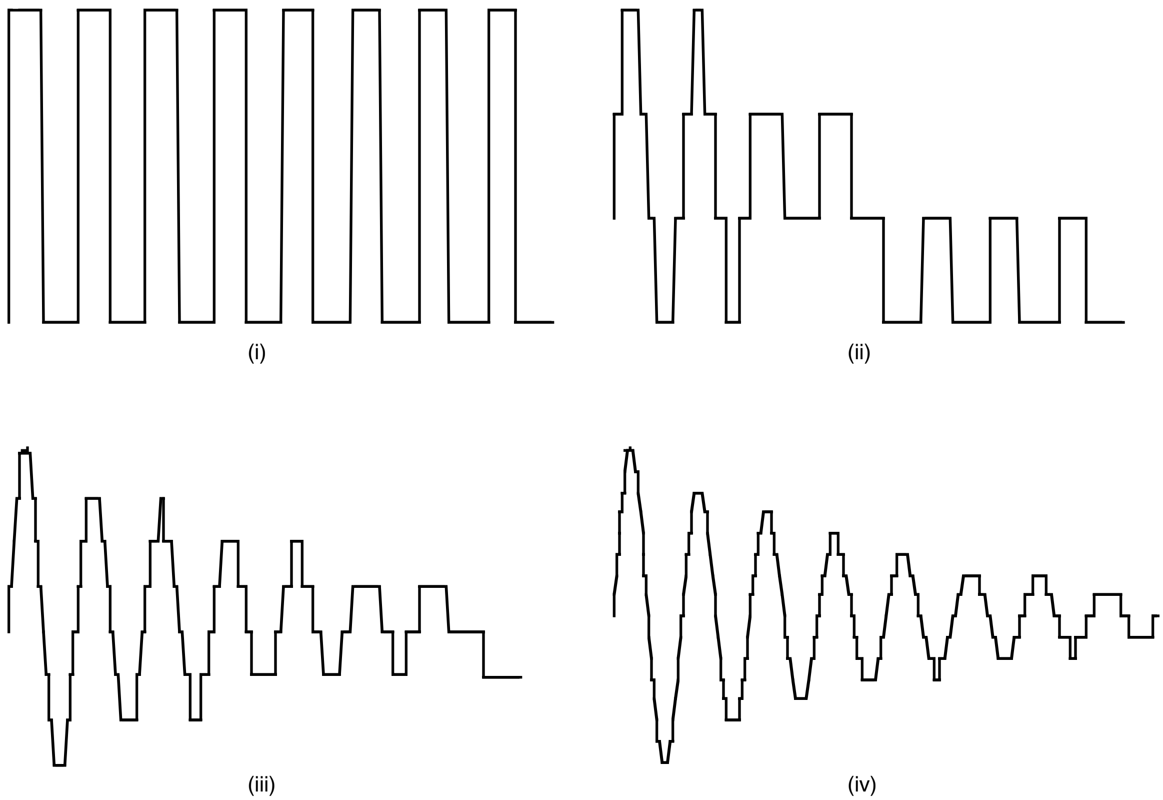


Figure 72 **A decaying sine wave represented with different word lengths: (i) 1-bit, (ii) 2-bit, (iii) 3-bit, (iv) 4-bit.**

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 parameter (dynamic range). 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 counter-act 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.

Aliasing is an unwanted artifact (usually!) of digital sampling. It's an established rule in digital sampling that all signal frequency components above half the sampling frequency (the Nyquist rate) must be removed with a lowpass filter (anti-aliasing filter). If frequencies above the Nyquist rate are not removed, you will hear aliasing. A digital sampler cannot represent frequencies above the Nyquist rate, but rather than remove the high frequencies, the sampler folds the high frequencies back down into the lower frequencies where they are added to the original low frequencies. If you were to play a rising pure tone through a sampler without an anti-alias filter, you would hear the tone start to fall when it past the Nyquist rate. The pitch will continue to drop as the input tone's frequency increases until the input tone reaches the sampling rate. The sampled tone would then have reached dc (frequency is 0) and will start to rise again. Usually a lowpass anti-aliasing filter is placed before the sampler to prevent this from happening.

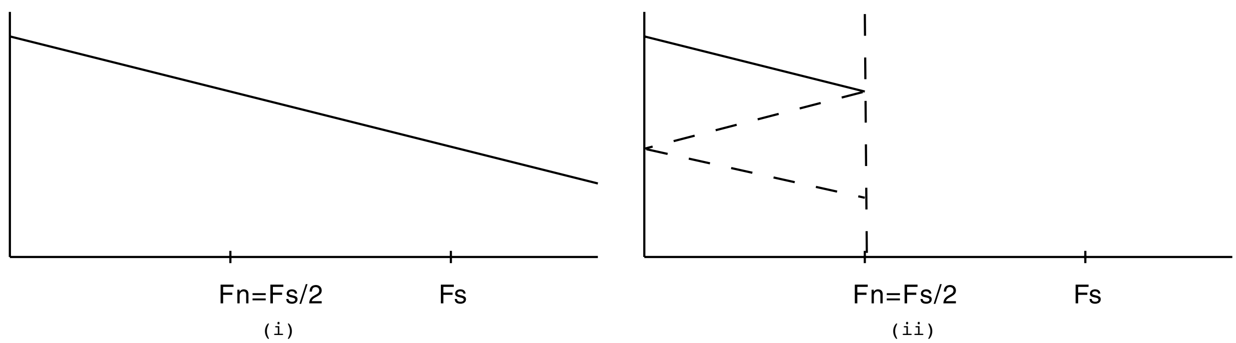


Figure 73 **Spectra of (i) an analog signal and the (ii) same signal after sampling without filtering**

In the **Quantize+Alias** algorithms, we do not actually sample the incoming signal at a lower rate. Instead we use a special modulation algorithm to simulate the effect of pitches falling when they should be rising. The Pitch (coarse and fine) parameters roughly correspond to setting the Nyquist frequency. Higher pitches result in modulating your input signal with higher frequencies. The LFO Depth parameter changes the strength of the modulation. Larger values of LFO Depth produce a deeper modulation which may be considered analogous to inputting a insufficiently band-limited signal for sampling.

Parameters:

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
Quant W/D	0 to 100 %wet	DynamRange	0 to 144 dB
Alias W/D	0 to 100 %wet	dc Offset	-79.0 to 0.0 dB
Lowpass	8 to 25088 Hz	Headroom	0 to 144 dB

Page 2

Pitch Crs	8 to 25088 Hz		
Pitch Fine	-100 to 100 ct		
LFO Depth	1 to 49 samp		

In/Out	When set to **In**, the quantizer and aliaser are active; when set to **Out**, the quantizer and aliaser are bypassed.
Out Gain	The overall gain or amplitude at the output of the effect.
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 you signal. At low DynamRange settings, adding DC offset can may the output sputter. dc Offset is expressed in decibels (dB) relative to full-scale digital.
Alias W/D	Amount of aliaser output signal (wet) relative to aliaser input signal (dry) to send to the final output. The dry signal here is taken to mean the output of the quantizer.
Pitch C **Pitch F**	Pitch sets the frequency (coarse and fine) at which the input signal is modulated. Higher pitches produce a high frequency modulation.
LFO Depth	The depth of the modulation, controlling how strong the modulation sounds. Larger values produce a more extreme modulation effect.

309 Quantize+Flange

Digital quantization followed by flanger

PAUs: 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, grundgy, 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. An 18-bit digital-to-analog converter (DAC) like the one in the K2600 can interpret 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.

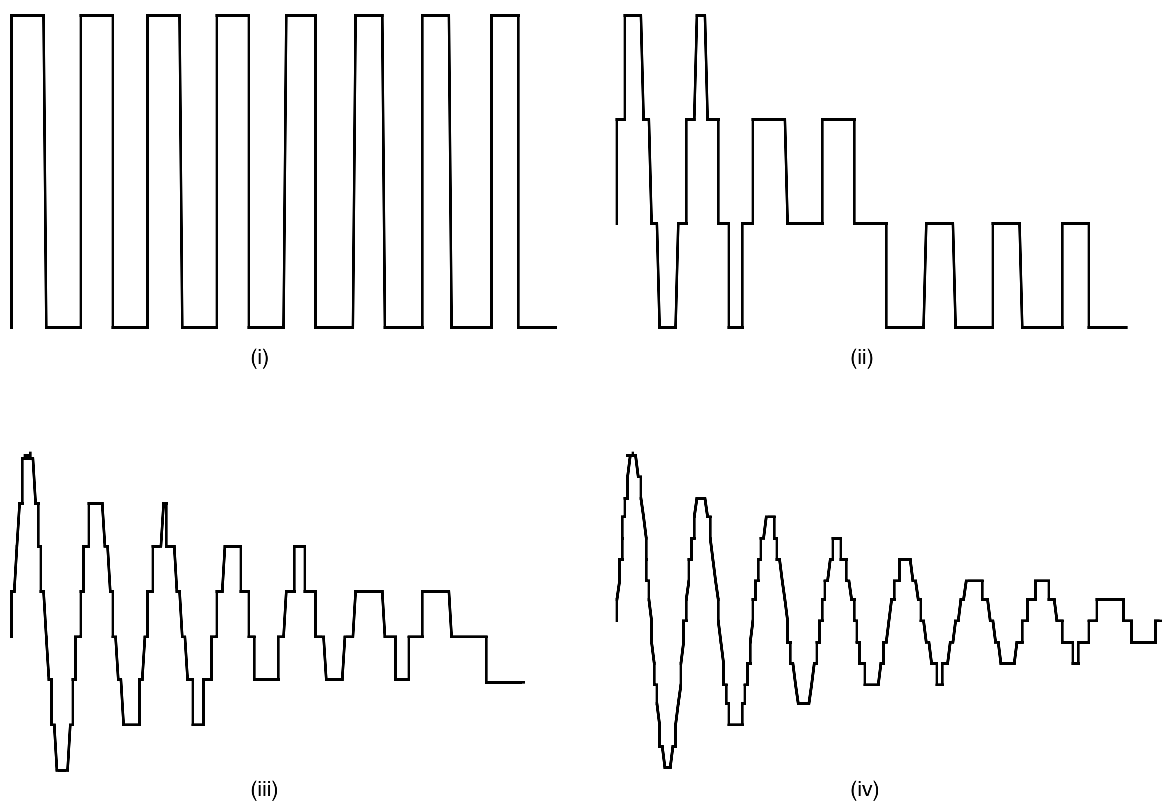


Figure 74 **A decaying sine wave represented with different word lengths: (i) 1-bit, (ii) 2-bit, (iii) 3-bit, (iv) 4-bit.**

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 parameter (dynamic range). 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 on multi-tap flangers (**Flanger 1** and **Flanger 2 on page 104**) for a detailed explanation of how the flanger works.

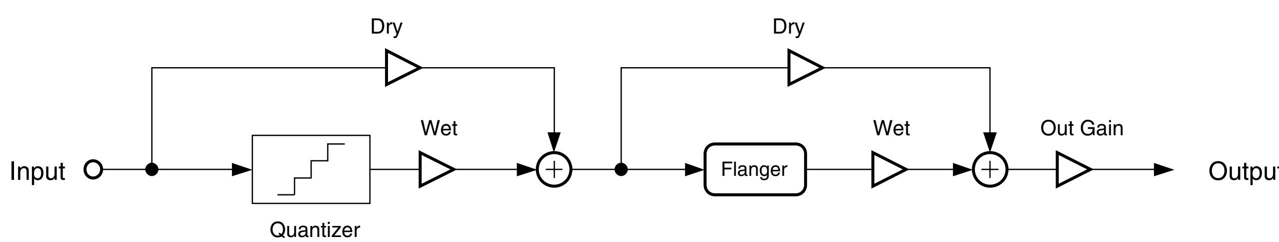


Figure 75 Block diagram of one channel of Quantize+Flange

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 you signal. At low DynamRange settings, adding DC offset can may 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. In this case, FXMods (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.

Guitar Combination Algorithms

285 Cabinet+Dly+Rvrb
310 Gate+TubeAmp
311 Gate+Tube+Reverb
312 Gt+Tube<>MD+Chor
313 Gt+Tube<>MD+Flan
314 Gt+Tube<>2MD
315 Gt+Cmp+Dst+EQ+Ch
316 Gt+Cmp+Dst+EQ+Fl
327 Tube+Reverb
656 Mn Gate+TubeAmp

Combination algorithms designed for guitar processing.

PAUs: 3 for **Gate+TubeAmp**, **Cabinet+Dly+Rvrb**, **Tube+Reverb**, 4 for the others

These combination algorithms are provided with guitar processing in mind. Each of the algorithms sends the signal through a gate, tone controls, tube distortion and cabinet simulation or EQ section. Also depending on the algorithm selected, the signal may pass through one or more of compressor, equalization, chorus, flange, moving delay or reverb. The algorithms are mono, though the chorus or flange can provide stereo spreading at the output. **Gate+TubeAmp** is the simplest of these algorithms:

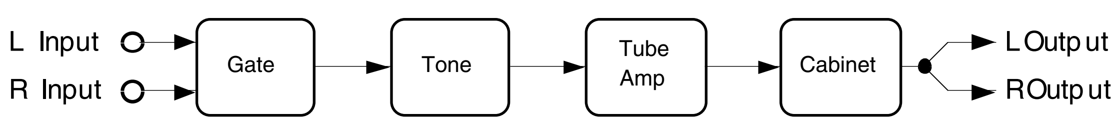


Figure 76 **Gate+TubeAmp**

The gate (same gate as Algorithm **343 Gate**) allows you to cut out noise during silence. Both the gate and compressor have their own side-chain processing paths, and a number of signal routing options for side-chain processing are provided. The gate side-chain input may be taken from either the left or right channels, or the average signal magnitude of the left and right channels may be selected with the GateSCInp parameter. Also you may choose to gate the sum of left and right channels or just one of the channels with the Gate Chan parameter. Since the effect is mono, if you gate only one channel (left or right), then that channel will be sent to the next stage of the effect, and the channel that is not selected will be discarded. If you choose both (L+R)/2, the sum (mix) of both channels will be used for further processing.

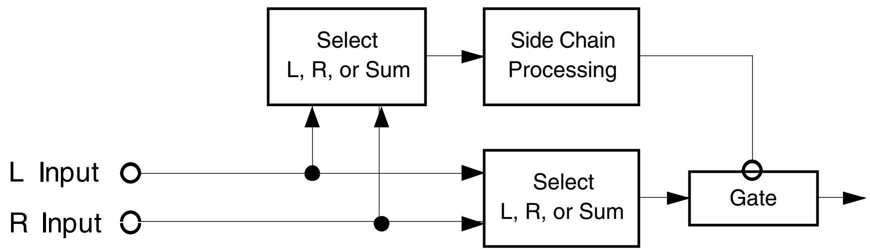


Figure 77 **Gate routings**

Each of the guitar combi algorithms contains a tone control and tube distortion model. The 3-band tone control authentically recreates the response in many guitar preamps based on real measurements collected by Kurzweil engineers. It is adjusted with the Bass Tone, Mid Tone, and Treb Tone controls with values ranging from **0** to **10** commonly found on many guitar amps. The flattest frequency response is obtained by setting Mid Tone to **10**, and both Bass and Treb Tone controls to **0**. The Tube Drive parameter faithfully model the response and smooth distortion caused by overloading a vacuum tube circuit.

Following the tube distortion is cabinet simulation or an EQ section with parametric bass, treble and mid-range equalization filters. The cabinet simulator models the responses of various types of mic'd guitar cabinets. The preset can be selected using the Cab Preset parameter. The presets are described below.

Basic	Flat response from 100 Hz to 4 khz with 24dB/oct rolloffs on each end
Lead 12	Open back hard American type with one 12" driver
2x12	Closed back classic American type with two 12" drivers
Open 12	Open back classic American type with one 12" driver
Open 10	Open back classic American type with one 10" driver
4x12	Closed back British type with four 12" drivers
Hot 2x12	Closed back hot rod type with two 12" drivers
Hot 12	Open back hot rod type with one 12" driver

Gate+Tube+Reverb is a guitar combi effect which includes a reverb. The reverb is the same as Algorithm **1 MiniVerb**:

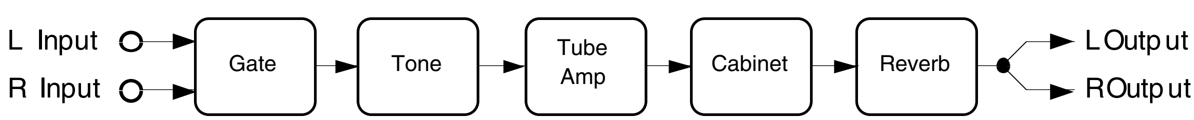


Figure 78 **Gate+Tube+Reverb**

Tube+Reverb is similar to **Gate+Tube+Reverb**, but does not have the gate and tone controls. As you would expect, **Cabinet+Dly+Rvrb** is the cabinet filter followed by a simple delay and the reverb.

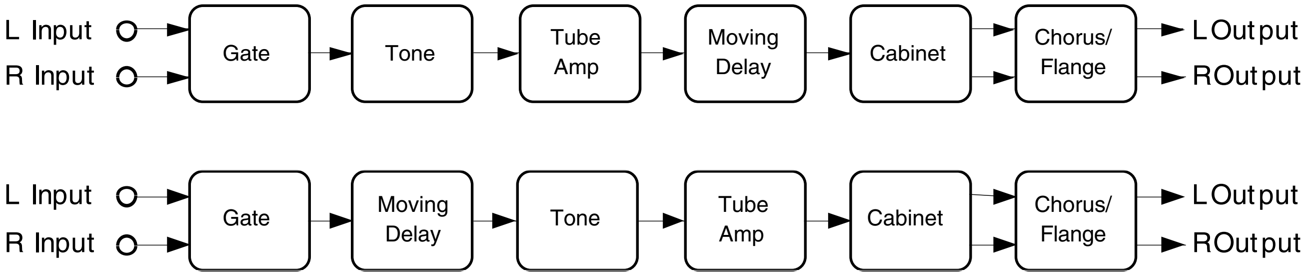


Figure 79 Gt+Tube<>MD+Chor and Gt+Tube<>MD+Flan; MD Insert set to PostDist and PreDist

Gt+Tube<>MD+Chorand **Gt+Tube<>MD+Flan** in the figures above include a monaural moving delay segment. Its parameters begin with the letters "MD." The moving delay is flexible enough that it can serve as a chorus, flange, or straight delay. With the MD Insert parameter, the moving delay may be placed before or after the tube distortion in the signal path. For more detailed information, refer to the section describing Algorithm **191 Dual MovDelay**. The cabinet output is provided with a panner to send its filtered output into the stereo chorus or flanger. For a description of the chorus see Algorithm **200 Chorus 1**, and for the flanger see Algorithm **225 Flanger 1**.

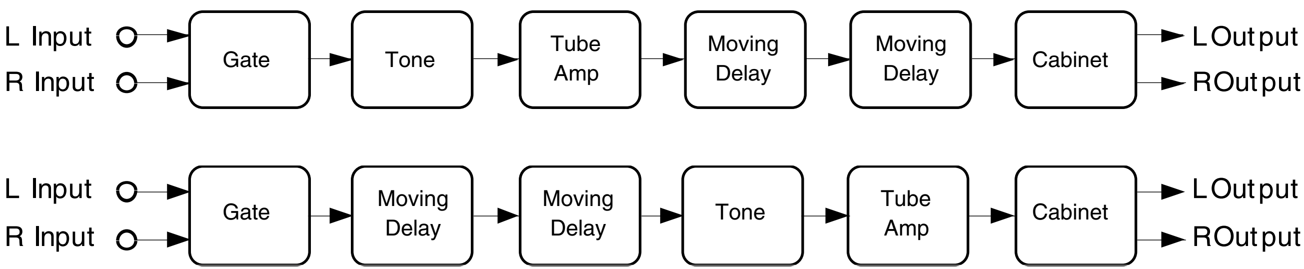


Figure 80 Gt+Tube<>MD+Chor with MD Insert set to PostDist and PreDist

Gt+Tube<>2MD in the figure above has a pair of moving delay segments. The dual moving delays behave like Algorithm **192 Dual MvDly+MvDly**. With the MD Insert parameter, the moving delay may be placed before or after the tube distortion in the signal path.

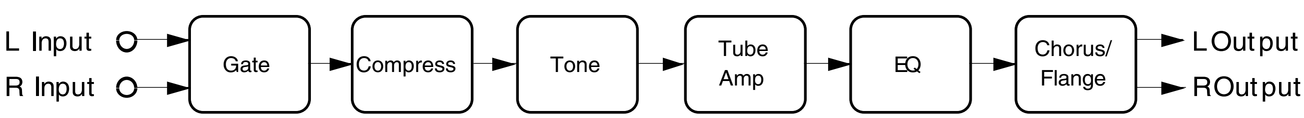


Figure 81 Gt+Cmp+Dst+EQ+Ch and Gt+Cmp+Dst+EQ+Fl

Gt+Cmp+Dst+EQ+Ch and **Gt+Cmp+Dst+EQ+Fl** are illustrated in the figure above. The compressor is the soft knee compressor described in Algorithm **331 SoftKneeCompress**. Instead of the cabinet simulation, there is an EQ section with bass and treble shelf filters and a parametric EQ. The next effect after EQ is either chorus or flanger. The chorus and flange have mono inputs and stereo outputs. Each is a standard single tap chorus or flange (see Algorithm 225).The Chorus W/D of Flange W/D control determines the mix of equalized distortion to chorus or flange signal to send to the output.

Parameters (Cabinet+Dly+Rvrb):

Page 1

Cab In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
Cab Mix	0 to 100%	Delay Mix	0 to 100%
Cab Pan	-100 to 100%	Delay Pan	-100 to 100%
Cab Preset	Open 12, ...	Rvrb W/D	0 to 100% wet

Page 2

Delay Crs	0 to 2540 ms
Delay Fine	-20 to 20 ms
Delay Fdbk	0 to 100%
Dly LF Damp	8 to 25088 Hz
Dly HF Damp	8 to 25088 Hz

Page 3

Rv Type	Hall 1, ...		
Rv Time	0.5 to 30.0s, Inf		
Rv DiffScl	0.00 to 2.00x	Rv Density	0.00 to 4.00x
Rv Size Scl	0.00 to 4.00x	Rv HF Damp	8 to 25088 Hz
Rv PreDlyL	0 to 620ms	Rv PreDly R	0 to 620ms

Parameters (Gate+TubeAmp):

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
GateIn/Out	In or Out		
GateSCInp	L, R, (L+R)/2		
Gate Chan	L, R, (L+R)/2		

Page 2

Gate Thres	-79.0 to 0.0 dB	Gate Time	25 to 3000 ms
Gate Duck	On or Off	Gate Atk	0.0 to 228.0 ms
		Gate Rel	0 to 3000 ms
		GateSigDly	0.0 to 25.0 ms
		||||||||||||||||||||||||||||	Reduction
-dB 60 40 *	16 * 8 4		0

Page 3

Bass Tone	0.0 to 10.0	Tube Drive	Off, -79.0 to 60.0 dB
Mid Tone	0.0 to 10.0	Warmth	8 to 25088 Hz
Treb Tone	0.0 to 10.0	Cab Preset	Open 12, ...

Parameters (Gate+Tube+Reverb):

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
GateIn/Out	In or Out	Rv Wet/Dry	0 to 100 %wet
GateSCInp	L, R, (L+R)/2		
Gate Chan	L, R, (L+R)/2		

Page 2

Gate Thres	-79.0 to 0.0 dB	Gate Time	25 to 3000 ms
Gate Duck	On or Off	Gate Atk	0.0 to 228.0 ms
		Gate Rel	0 to 3000 ms
		GateSigDly	0.0 to 25.0 ms
		|||||||||||||||||||||||||||||	Reduction
	-dB 60 40 *	16 * 8 4	0

Page 3

		Tube Drive	Off, -79.0 to 60.0 dB
		Warmth	8 to 25088 Hz
Bass Tone	0.0 to 10.0		
Mid Tone	0.0 to 10.0		
Treb Tone	0.0 to 10.0	Cab Preset	Open 12, ...

Page 4

Rv Type	Hall 1, ...		
Rv Time	0.5 to 30.0 s, Inf		
Rv DiffScl	0.00 to 2.00 x	Rv Density	0.00 to 4.00 x
Rv SizeScl	0.00 to 4.00 x	Rv HF Damp	8 to 25088 Hz
Rv PreDly L	0 to 620 ms	Rv PreDlyR	0 to 620 ms

Parameters (Gt+Tube<>MD+Chor and Gt+Tube<>MD+Flan):

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
GateIn/Out	In or Out		
GateSCInp	L, R, (L+R)/2		
Gate Chan	L, R, (L+R)/2		

Page 2

Gate Thres	-79.0 to 0.0 dB	Gate Time	25 to 3000 ms
Gate Duck	On or Off	Gate Atk	0.0 to 228.0 ms
		Gate Rel	0 to 3000 ms
		GateSigDly	0.0 to 25.0 ms
		||||||||||||||||||||||||||||	Reduction
-dB 60 40 ✻ 16 ✻ 8 4 0			

Page 3

		Tube Drive	Off, -79.0 to 60.0 dB
		Warmth	8 to 25088 Hz
Bass Tone	0.0 to 10.0		
Mid Tone	0.0 to 10.0	Cab In/Out	In or Out
Treb Tone	0.0 to 10.0	Cab Preset	Open 12, ...
		Cab Pan	-100 to 100%

Page 4

MD Insert	PostDist, ...	MD Delay	0.0 to 1000.0 ms
MD Wet/Dry	0 to 100%	MD LFOMode	8 to 25088 Hz
		MD LFORate	0.00 to 10.00 Hz
		MD LFODpth	0.0 to 200.0%
		MD Fdbk	-100 to 100%

Page 5 (Gt+Tube<>MD+Chor)

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 PtchEnv	Triangle or Trapzoid		
Ch Wet/Dry	0 to 100%	Ch Out Bal	-100 to 100%

Page 5 (Gt+Tube<>MD+Flan)

Fl Rate	0 to 32 bts	Fl Tempo	System, 1 to 255 BPM
Fl Xcurs L	0.0 to 230.0 ms	Fl Xcurs R	0.0 to 230.0 ms
Fl Delay L	0.0 to 230.0 ms	Fl Delay R	0.0 to 230.0 ms
Fl Fdbk L	-100 to 100%	Fl Fdbk R	-100 to 100%
Fl Phase L	0.0 to 360.0 deg	Fl Phase R	0.0 to 360.0 deg
Fl Wet/Dry	0 to 100%	FL Out Bal	-100 to 100%

Parameters (Gt+Tube<>2MD):

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
GateIn/Out	In or Out	MD Insert	PostDist, ...
GateSCInp	L, R, (L+R)/2	MD Wet/Dry	0 to 100%
Gate Chan	L, R, (L+R)/2		

Page 2

Gate Thres	-79.0 to 0.0 dB	Gate Time	25 to 3000 ms
Gate Duck	On or Off	Gate Atk	0.0 to 228.0 ms
		Gate Rel	0 to 3000 ms
		GateSigDly	0.0 to 25.0 ms
		|||||||||||||||||||||||||||||	Reduction
	-dB 60 40 ✻	16 ✻ 8 4	0

Page 3

		Tube Drive	Off, -79.0 to 60.0 dB
		Warmth	8 to 25088 Hz
Bass Tone	0.0 to 10.0		
Mid Tone	0.0 to 10.0	Cab In/Out	In or Out
Treb Tone	0.0 to 10.0	Cab Preset	Open 12, ...
		Cab Pan	-100 to 100%

Page 4

MD1Delay	0.0 to 1000.0 ms	MD2Delay	0.0 to 1000.0 ms
MD1LFOMode	8 to 25088 Hz	MD2LFOMode	8 to 25088 Hz
MD1LFORate	0.00 to 10.00 Hz	MD2LFORate	0.00 to 10.00 Hz
MD1LFODpth	0.0 to 200.0%	MD2LFODpth	0.0 to 200.0%
MD1Fdbk	-100 to 100%	MD2Fdbk	-100 to 100%

Parameters (Gt+Cmp+Dst+EQ+Ch and Gt+Cmp+Dst+EQ+Fl):

Page 1

GateIn/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
GateSCInp	L, R, (L+R)/2	CompIn/Out	In or Out
Gate Chan	L, R, (L+R)/2	FdbkComprs	In or Out

Page 2

Gate Thres	-79.0 to 0.0 dB	Gate Time	25 to 3000 ms
Gate Duck	On or Off	Gate Atk	0.0 to 228.0 ms
		Gate Rel	0 to 3000 ms
		GateSigDly	0.0 to 25.0 ms
		||||||||||||||||||||||||||||	Reduction
	-dB 60 40 * 16	* 8 4	0

Page 3

Comp Atk	0.0 to 228.0 ms	Comp Ratio	1.0:1 to 100:1, Inf:1
Comp Rel	0 to 3000 ms	Comp Thres	-79.0 to 0.0dB
CompSmooth	0.0 to 228.0 ms	CompMakeUp	Off, -79.0 to 24.0 dB
CompSigDly	0.0 to 25.0ms		
		||||||||||||||||||||||||||||	Reduction
	-dB 40 20 12 8	6 4 2	0

Page 4

Bass Tone	0.0 to 10.0	Tube Drive	Off, -79.0 to 60.0 dB
Mid Tone	0.0 to 10.0		
Treb Tone	0.0 to 10.0		

Page 5

Bass Gain	-79.0 to 24.0 dB	Treb Gain	-79.0 to 24.0 dB
Bass Freq	8 to 25088 Hz	Treb Freq	8 to 25088 Hz
Mid Gain	-79.0 to 24.0 dB		
Mid Freq	8 to 25088 Hz		
Mid Width	0.010 to 5.000 oct		

Page 6 (Gt+Cmp+Dst+EQ+Ch)

Chorus W/D	-100 to 100 %		
Ch Rate	0.01 to 10.00 Hz		
Ch Depth	0.0 to 50.0 ct		
Ch Delay	4.0 to 1000.0 ms		
Ch Fdbk	-100 to 100 %	Ch L/RPhas	0.0 to 360.0 deg
Ch Xcouple	0 to 100 %	Ch HF Damp	8 to 25088 Hz

Page 6 (Gt+Cmp+Dst+EQ+Fl)

Flange W/D	-100 to 100 %	Fl Tempo	System, 1 to 255 BPM
Fl Period	1/24 to 32 bts		
Fl LFO Lvl	-100 to 100 %		
Fl StatLvl	-100 to 100 %		
Fl Fdbk	-100 to 100 %		
Fl L Phase	0.0 to 360.0 deg	Fl R Phase	0.0 to 360.0 deg

Page 7 (Gt+Cmp+Dst+EQ+Fl)

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

Parameters (Tube+Reverb):

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
Tube Drive	Off, -79.0 to 60.0 dB	Cab In/Out	In or Out
Warmth	8 to 25088 Hz	Cab Preset	Open 12, ...

Page 2

Rv Wet/Dry	0 to 100% wet		
Rv Type	Hall 1, ...		
Rv Time	0.5 to 30.0s, 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	8 to 25088 Hz
Rv PreDlyL	0 to 620ms	Rv PreDly R	0 to 620ms

In/Out When set to **In** the effect is active; when set to **Out** the effect is bypassed.

Out Gain The overall gain or amplitude at the output of the entire algorithm.

GateIn/Out	When set to **In** the gate is active; when set to **Out** the gate is bypassed.
GateSCInp	Select the input source channel for gate side-chain processing—left, right or both. For both (L+R)/2 the averaged magnitude is used.
Gate Chan	Select which input channel will receive gate processing—left, right or mix. This selects the mono input for the algorithm.
CompIn/Out	When set to **In** the compressor is active; when set to **Out** the compressor is bypassed.
FdbkComprs	A switch to set whether the compressor side-chain is configured for feed-forward (**Out**) or feedback (**In**).
Gate Thres	The signal level in dB required to open the gate (or close the gate if Ducking is **On**).
Gate Duck	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.
Gate Time	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.
Gate Atk	The time for the gate to ramp from closed to open (reverse if Gate Duck is **On**) after the signal rises above threshold.
Gate Rel	The time for the gate to ramp from open to closed (reverse if Gate Duck is **On**) after the gate timer has elapsed.
GateSigDly	The delay in milliseconds (ms) of the signal to be gated relative to the side chain signal. By delaying the main signal, the gate can be opened before the main signal rises above the gating threshold.
Comp Atk	The time for the compressor to start to cut in when there is an increase in signal level (attack) above the threshold.
Comp Rel	The time for the compressor to stop compressing when there is a reduction in signal level (release) from a signal level above the threshold.
CompSmooth	A lowpass filter in the compressor side-chain signal path. It is intended to smooth the output of the compressor's envelope detector. Smoothing will affect the attack or release times when the smoothing time is longer than one of the other times.
CompSigDly	The time in ms by which the input signal should be delayed with respect to compressor side-chain processing (i.e. side-chain predelay). This allows the compression to appear to take effect just before the signal actually rises.
Comp Ratio	The compression ratio. High ratios are highly compressed; low ratios are moderately compressed.
Comp Thres	The compressor threshold level in dBFS (decibels relative to full scale) above which the signal begins to be compressed.
CompMakeUp	A gain or amplitude control provided to offset gain reduction due to compression.
Bass Tone **Mid Tone** **Treb Tone**	Adjusts the three bands of tone control integrated with the distortion drive circuit. Flattest response is obtained by setting Mid Tone to **10.0** and both Bass Tone and Treb Tone to 0.0.
Tube Drive	Adjusts the gain into the distortion circuit. Higher values produce more distortion.

Cab In/Out	When set to **In** the cabinet filter is active; when set to **Out** the cabinet filter is bypassed.
Cab Preset	Eight preset cabinets have been created based on measurements of real guitar amplifier cabinets. The presets are **Basic**, **Lead 12**, **2x12**, **Open 12**, **Open 10**, **4x12**, **Hot 2x12**, and **Hot 12**.
Warmth	Adjusts a 1 pole (6dB/oct) lowpass filter applied after distortion.
Bass Gain	The amount of boost or cut that the bass shelving filter should apply to the low frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the bass signal below the specified frequency. Negative values cut the bass signal below the specified frequency.
Bass Freq	The center frequency of the bass shelving filter in intervals of one semitone.
Treb Gain	The amount of boost or cut that the treble shelving filter should apply to the high frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the treble signal above the specified frequency. Negative values cut the treble signal above the specified frequency.
Treb Freq	The center frequency of the treble shelving filters in intervals of one semitone.
Mid Gain	The amount of boost or cut that the parametric mid filter should apply in dB to the specified frequency band. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the signal at the specified frequency. Negative values cut the signal at the specified frequency.
Mid Freq	The center frequency of the parametric mid filter in intervals of one semitone. The boost or cut will be at a maximum at this frequency.
Mid Width	The bandwidth of the side chain parametric mid filter may be adjusted. Specify the bandwidth in octaves. Small values result in a very narrow filter response. Large values result in a very broad response.
MD Wet/Dry **Chorus W/D** **Flange W/D**	Adjusts the ratio of moving delay, chorus or flange signal and the equalized distortion signal fed to the gate. **0%** feeds only the input to the next processing stage or the output, bypassing the moving delay, chorus or flange. **100%** feeds only the moving delay, chorus or flange to the next processing stage or the output.
MD Delay	Adjusts the delay time for each moving delay circuit, which is the center of LFO excursion.
MD LFOMode	Adjusts the LFO excursion type. In Flange mode, the LFO is optimized for flange effects and MD LFODpth adjusts the excursion amount. In ChorTri and ChorTrap modes, the LFO is optimized for triangle and trapezoidal pitch envelopes respectively, and MD LFODpth adjusts the amount of chorus detuning. In Delay mode, the LFO is turned off leaving a basic delay. MD LFORate and MD LFODpth in Delay mode are disabled.
MD LFORate	Adjusts the LFO speed for each moving delay circuit.
MD LFODpth	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.
MD Fdbk	Adjusts the level of each moving delay circuits output signal fed back into their own inputs. Negative values polarity invert the feedback signal.

323 TubeAmp<>MDBP>Ch
324 TubeAmp<>MDBP>Fl
325 PolyAmp<>MDBP>Ch
326 PolyAmp<>MDBP>Fl

Mono distortion circuits in combination with moving delays and a stereo chorus or flange

PAUs: 3 each

Each of these algorithms offers a flexible chain of effects designed primarily for guitar processing. Each chain offers a different combination of a 3-band tone control, tube-amp distortion drive, poly-amp distortion drive, cabinet simulation, chorus, flange, and a generic moving delay. The entire algorithm is monaural with the exception of the final chorus or flange at the end of each chain, which have one input and a stereo output.

At the beginning of each chain is a 3-band tone control authentically recreating the response in many guitar preamps based on real measurements collected by Kurzweil engineers. It is adjusted with the Bass Tone, Mid Tone, and Treb Tone controls with values ranging from **0** to **10** commonly found on many guitar amps. The flattest frequency response is obtained by setting Mid Tone to **10.0**, and both Bass and Treb Tone controls to **0.0**.

The tone controls are integrated with one of two types of preamp drive circuits: TubeAmp and PolyAmp. The TubeAmp faithfully models the response and smooth distortion caused by overloading a vacuum tube circuit. PolyAmp is closely related to the **PolyDistort + EQ** algorithm offering a brighter sound quality with more sustain. The amount of distortion is controlled by adjusting the Tube Drive or Poly Drive parameter. High frequency energy caused by distortion can be rolled off by using the Warmth parameter.

Following the distortion drive element is a cabinet simulator. The cabinet simulator models the responses of various types of mic'd guitar cabinets. The preset can be selected using the Cab Preset parameter. The following is the list of cabinet presets and their descriptions:

Basic	Flat response from 100 Hz to 4 kHz with 24dB/oct rolloffs on each end
Lead 12	Open back hard American type with one 12" driver
2x12	Closed back classic American type with two 12" drivers
Open 12	Open back classic American type with one 12" driver
Open 10	Open back classic American type with one 10" driver
4x12	Closed back British type with four 12" drivers
Hot 2x12	Closed back hot rod type with two 12" drivers
Hot 12	Open back hot rod type with one 12" driver

The cabinet can by switched on or off with the Cab In/Out parameter. The Cab Pan parameter adjusts the final pan position of the cabinet at the output of the algorithm, but this does not affect the cabinet signal fed into the final stereo flange or chorus. If Ch Wet/Dry or Fl Wet/Dry is set to **100**%, this pan control will not have any audible affect since the entire output of the cabinet is fed into the flange or chorus instead of the algorithm output.

At the end of the chain is either a chorus or a flange controlled by parameters beginning with "Ch" or "Fl." The chorus and flange have mono inputs and stereo outputs. Each is a standard KDFX single tap dual channel chorus or flange (see Algorithm **225**) with independent controls for left and right channels found in many other 1-PAU combination algorithms. The Ch Wet/Dry or Fl Wet/Dry control determines the final output mix of the algorithm. When set at **0**%, only the cabinet simulator output is fed to the output of the algorithm. At **100%**, only the output of the chorus or flange is heard. Left/right balance specifically for the chorus or flange can be adjusted with the Out Bal control.

In addition, there is a generic monaural moving delay segment. Its parameters begin with the letters "MD." The moving delay is flexible enough that it can serve as a chorus, flange, or straight delay. For more detailed information, refer to the section describing the **Dual MovDelay** and **Dual MvDly+MvDly** algorithms (Algorithms **191** and **192**). As implemented in these four algorithms, it can be inserted either before the tone controls (PreDist), or after the distortion drive (PostDist), or bypassed altogether. This is selected with the MD Insert parameter. Also provided is the MD Wet/Dry parameter that mixes the output of the moving delay circuit with its own input to be fed into the next effect in the chain.

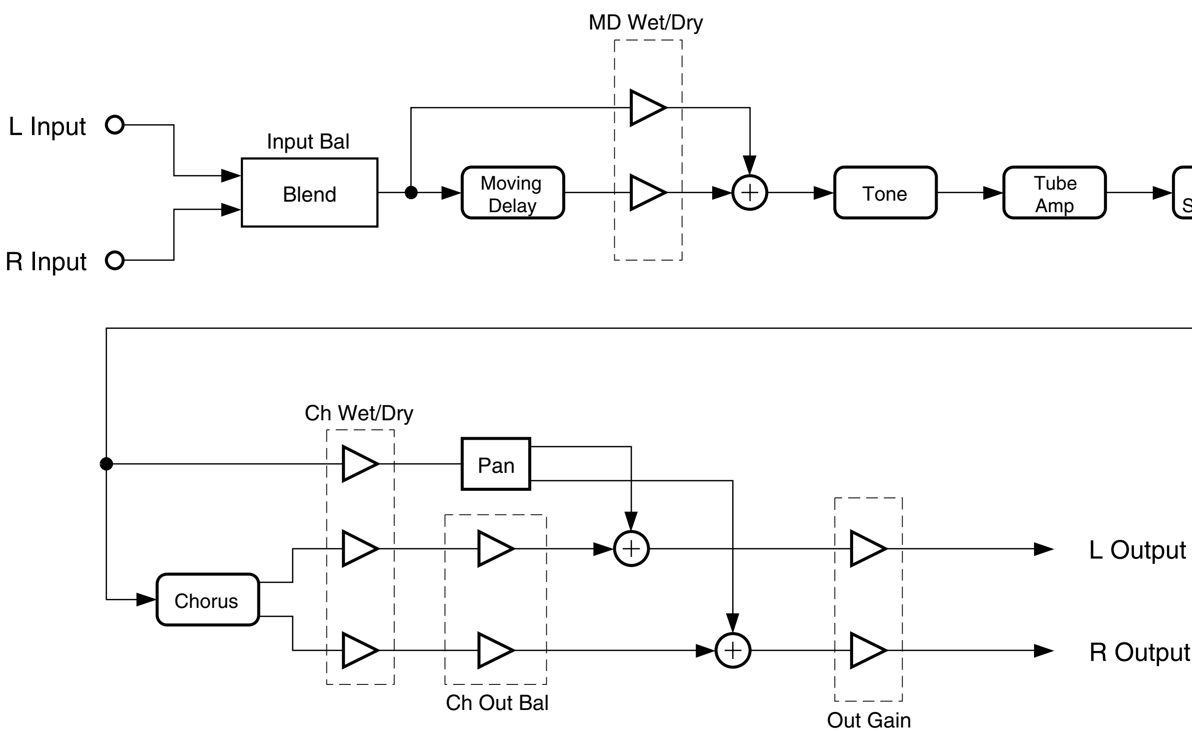


Figure 82 TubeAmp<>MDBP>Ch with moving delay inserted pre-distortion

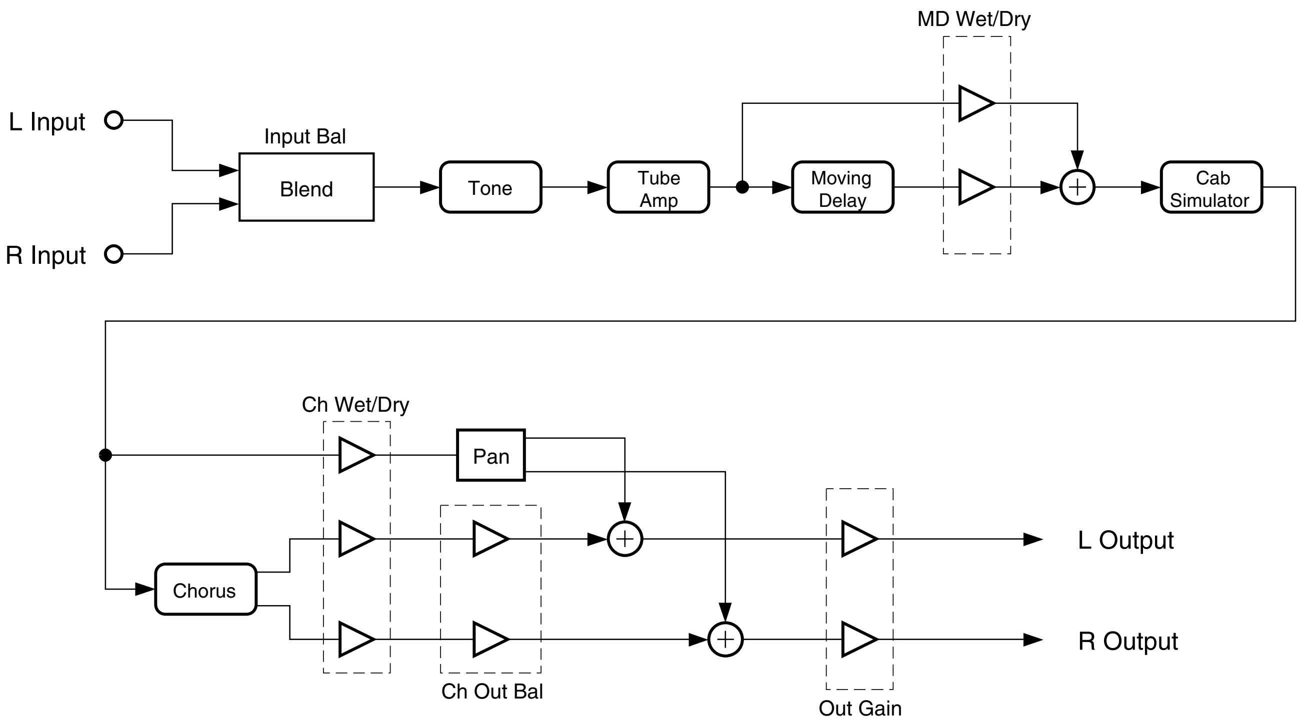


Figure 83 TubeAmp<>MDBP>Ch with moving delay inserted post-distortion

Parameters:

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
Input Bal	-100 to 100 %		

Page 2 (TubeAmp Algorithms)

		Tube Drive	Off, -79.0 to 60.0 dB
		Warmth	8 to 25088 Hz
Bass Tone	0.0 to 10.0		
Mid Tone	0.0 to 10.0	Cab In/Out	In or Out
Treb Tone	0.0 to 10.0	Cab Preset	Open 12, etc.
		Cab Pan	-100 to 100 %

Page 2 (PolyAmp Algorithms

		Poly Drive	0.0 to 60.0 dB
		Warmth	8 to 25088 Hz
Bass Tone	0.0 to 10.0		
Mid Tone	0.0 to 10.0	Cab In/Out	In or Out
Treb Tone	0.0 to 10.0	Cab Preset	Open 12, etc.
		Cab Pan	-100 to 100 %

Page 3

MD Insert	PreDist, PostDist, Bypass	MD Dly bts	0 to 32 bts
MD Wet/Dry	0 to 100 %	MD Dly ms	0.0 to 1000.0 ms
		MD LFOMode	ChorTri, ChorTrap, Delay, Flange
MD Tempo	System, 1 to 255 BPM	MD LFORate	0.00 to 10.00 Hz
		MD LFODpth	0.0 to 200.0 %
		MD Fdbk	-100 to 100 %

Page 4   (Chorus Algorithms)

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 PtchEnv	Triangle or Trapzoid		
Ch Wet/Dry	0 to 100 %	Ch Out Bal	-100 to 100 %

Page 4   (Flange Algorithms)

Fl Rate	0 to 32 bts	Fl Tempo	System, 1 to 255 BPM
Fl Xcurs L	0.0 to 230.0 ms	Fl Xcurs R	0.0 to 230.0 ms
Fl Delay L	0.0 to 230.0 ms	Fl Delay R	0.0 to 230.0 ms
Fl Fdbk L	-100 to 100 %	Fl Fdbk R	-100 to 100 %
Fl Phase L	0.0 to 360.0 deg	Fl Phase R	0.0 to 360.0 deg
Fl Wet/Dry	0 to 100 %	Fl Out Bal	-100 to 100 %

In/Out Toggles the entire effect on or off. When **Off**, the input signal is passed.

Input Bal Adjusts the ratio of left and right algorithm inputs to be summed into the monaural signal that is processed by the effect. **0**% blends equal amount of left and right. Negative values blend increasing amounts of left, while positive values blend increasing amounts of right.

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

Bass, Tone
Mid Tone
Treb Tone Adjusts the three bands of the tone control integrated with the distortion drive circuit. Flattest response is obtained by setting Mid Tone to **10.0**, and both Bass Tone and Treb Tone to **0.0**.

Tube Drive,
Poly Drive Adjusts the gain into each distortion circuit. Higher values produce more distortion.

Warmth Adjusts a 1-pole (6dB/oct) lowpass filter applied after distortion.

Cab In/Out Turns the cabinet simulator on or off.

Cab Preset Selects the preset cabinet type.

Cab Pan	Adjusts the output pan position of the cabinet simulator signal that is mixed at the output of the algorithm. Note that when Ch Wet/Dry or Fl Wet/Dry is set to **100%**, no signal from the cabinet is mixed directly to the output, so this parameter has no affect.
MD Insert	Selects where in the signal chain the moving delay is to be. PreDist places it before the distortion and tone circuit. PostDist places it between the distortion circuit and cabinet simulator, and Bypass takes it completely out of the path.
MD Wet/Dry	Adjusts the ratio of the moving delay output mixed with its own input to be fed to the next effect in the chain
MD Tempo	The basis for the moving delay lines 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. In this case, FX Mods (FUNs, LFOs, ASRs, etc.) will have no effect on the Tempo parameter.
MD Dly bts	Adjusts the delay time for the moving delay circuit in tempo beats, which is the center of LFO excursion.
MD Dly ms	Adjusts the delay time for the moving delay circuit in milliseconds, which is the center of LFO excursion.
MD LFOMode	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 trapezoidal 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.
MD LFORate	Adjusts the LFO speed for the moving delay circuit.
MD LFODpth	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.
MD Fdbk	Adjusts the level of the moving delay circuit output signal fed back into its own input. Negative values polarity-invert the feedback signal.
Ch Wet/Dry **Fl Wet/Dry**	Adjusts the ratio of flange or chorus signal and the cabinet simulator signal fed to the output of the algorithm. **0%** feeds only the cabinet simulator to the output bypassing the final chorus or flange. **100%** feeds only the flange or chorus to the output.
Ch Out Bal **Fl Out Bal**	Adjusts the left/right output balance of the chorus or flange signal. Negative values balance toward the left while positive values balance toward the right.

321 Flange<>Shaper

This algorithm is one of a group of *configurable* combination algorithms—that is, there's more than one effect in the algorithm, and you can change the sequence of those effects. With this algorithm, for example, you can have either a flanger followed by a shaper, or vice versa.

The combination algorithms are organized in groups, with IDs predominantly in the 400s (there are a few exceptions, of course). For a description of Algorithm **321** and other combination algorithms, follow one of the links below:

321 Flange<>Shaper on page 189

Combination Algorithms on page 279

Configurable Combination Algorithms on page 289

More Combination Algorithms on page 299

322 Shaper<>Reverb

This algorithm is one of a group of *configurable* combination algorithms—that is, there's more than one effect in the algorithm and you can change the sequence of those effects. With this algorithm, for example, you can have either a shaper followed by a reverb, or vice versa.

The combination algorithms are organized in groups, with IDs predominantly in the **400**s (there are a few exceptions, of course). For a description of Algorithm **322** and other combination algorithms, follow one of the links below:

Combination Algorithms on page 279

Configurable Combination Algorithms on page 289

More Combination Algorithms on page 299

Compressors and Expanders

330 HardKneeCompress
331 SoftKneeCompress
347 Dual SKCompress
660 Mn HK Compress
661 Mn SK Compress

Stereo hard- and soft-knee signal compression algorithms

PAUs: 2 for **Dual SKCompress**; 1 for the others

The stereo hard- and soft-knee compressors are very similar algorithms and provide identical parameters and user interface. Both algorithms compress (reduce) the signal level when the signal exceeds a threshold. The amount of compression is expressed as a ratio. The compression ratio is the inverse of the slope of the compressor input/output characteristic. A compression ratio of **1:1** will have no effect on the signal. An infinite ratio, will compress all signal levels above the threshold level to the threshold level (zero slope). For ratios in between infinite and **1:1**, increasing the input will increase the output, but by less than it would if there was no compression. The threshold is expressed as a decibel level relative to digital full-scale (dBFS) where **0 dBFS** is digital full-scale and all other available values are negative.

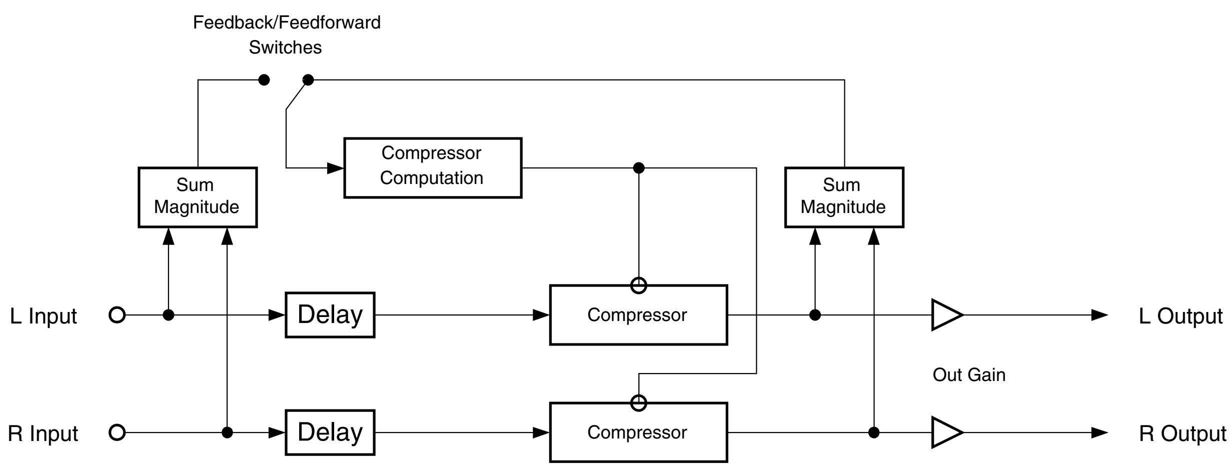


Figure 84 A stereo compressor

The above diagram shows a stereo compressor. Other configurations include mono and dual mono. The dual mono configuration has separate compression controls for the left and right channels, though you can still select which input channel(s) to use for side chain processing.

The hard-knee compressor makes a sudden transition from uncompressed to compressed at the compression threshold. The soft-knee compressor gives a more gradual transition from compressed to unity gain.

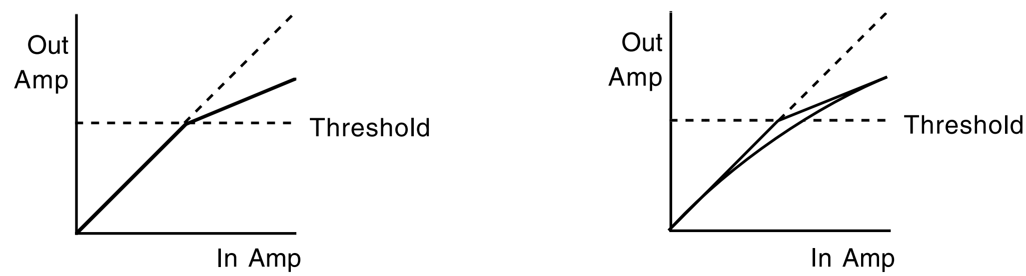


Figure 85 Hard- and Soft-Knee compression characteristics

To determine how much to compress the signal, the compressor must measure the signal level. Since musical signal levels will change over time, the compression amounts must change as well. You can control the rate at which compression changes in response to changing signal levels with the attack and release time controls. With the attack time, you set how fast the compressor responds to increased levels. At long attack times, the signal may overshoot the threshold level for some time before it becomes fully compressed, while at short attack times, the compressor will rapidly clamp down on the level. The release time controls how long it takes the compressor to respond to a reduction in signal levels. At long release times, the signal may stay compressed well after the signal falls below threshold. At short release times, the compressor will open up almost as soon as the signal drops.

For typical compressor behavior, the attack time is considerably shorter than the release time. At very short attack and release times, the compressor is almost able to keep up with the instantaneous signal levels and the algorithm will behave more like distortion than compression. In addition to the attack and release times, there is another time parameter: SmoothTime. The smoothing parameter will increase both the attack and release times, although the effect is significant only when its time is longer than the attack or release time. Generally the smoothing time should be kept at or shorter than the attack time.

The stereo and dual mono compressors allow you to select which input(s) to use for side chain processing. The choices are left (L), right (R) or the average of both left and right (L+R)/2. The stereo compressors let you select whether compression should affect both channels or only one channel.

You have the choice of using the compressors configured as feed-forward or feedback compressors. For feed-forward, set the FdbkComprs parameter to **Out**; for feedback compression, set it to **In**. The feed-forward configuration uses the input signal as the side-chain source. The feedback compressor on the other hand uses the compressor output as the side-chain source. Feedback compression tends to be more subtle, but you cannot get an instant attack.

In the feedback configuration, the signal being compressed may be delayed relative to the side chain compression processing. The delay allows the signal to start being compressed just before an attack transient arrives. Since the side chain processing "knows" what the input signal is going to be before the main signal path does, it can tame down an attack transient by compressing the attack before it actually happens. In the feed-forward configuration, the delay affects both the main signal and the side chain, and so is of limited usefulness. (The larger compressors do no have this limitation.)

A meter displays the amount of gain reduction that is applied to the signal as a result of compression.

Parameters:

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
FdbkComprs	In or Out	SC Input	L, R, (L+R)/2
		ComprsChan	L, R, L & R

Page 2

Atk Time	0.0 to 228.0 ms	Ratio	1.0:1 to 100:1, Inf:1
Rel Time	0 to 3000 ms	Threshold	-79.0 to 0.0dB
SmoothTime	0.0 to 228.0 ms	MakeUpGain	Off, -79.0 to 24.0 dB
Signal Dly	0.0 to 25.0ms		
		||||||||||||||||||||||||||||	Reduction
	-dB 40 20 12 8	6 4 2	0

In/Out When set to **In** the compressor is active; when set to **Out** the compressor is bypassed.

Out Gain Compressing the signal causes a reduction in signal level. To compensate, the output gain parameter may be used to increase the gain by as much as 24 dB. Note that the Out Gain parameter does not control the signal level when the algorithm is set to **Out**.

FdbkComprs Sets the compressor side chain for feed-forward (Out) or feedback (In).

SC Input The input source channel for side-chain processing—left, right or the average of both.

ComprsChan Select which input channel will receive dynamics processing—left, right or both. If you select left or right, the opposite channel will pass through unaffected.

Atk Time The time for the compressor to start to cut in when there is an increase in signal level (attack) above the threshold.

Rel Time The time for the compressor to stop compressing when there is a reduction in signal level (release) from a signal level above the threshold.

SmoothTime A lowpass filter in the control signal path. It is intended to smooth the output of the compressor's envelope detector. Smoothing will affect the attack or release times when the smoothing time is longer than one of the other times.

Signal Dly When doing feed-forward compression, Signal Dly is the time in ms by which the input signal should be delayed with respect to compressor side chain processing (i.e. side chain predelay). This allows the compression to appear to take effect just before the signal actually rises. When doing feedback compression, this parameter causes both the side-chain and main signal path to be delayed together for limited benefit.

Ratio The compression ratio. The higher the ratio, the greater the amount of compression.

Threshold The threshold level in dBFS (decibels relative to full scale) above which the signal begins to be compressed.

MakeUpGain Provides an additional control of the output gain. The Out Gain and MakeUpGain controls are additive (in decibels) and together may provide a maximum of 24 dB boost to offset gain reduction due to compression.

332 Compress w/SC EQ
348 Dual Comprs SCEQ

Soft-knee compression algorithm with filtering in the side chain.

PAUs: 2 for **Compress w/SC EQ** and 3 for **Dual Comprs SCEQ**

The **Compress w/SC EQ** algorithm is the same as the **SoftKneeCompress** algorithm (**331**, **page 191**)except that equalization has been added to the side chain signal path. The equalization to the side chain includes bass and treble shelf filters and a parametric mid-range filter.

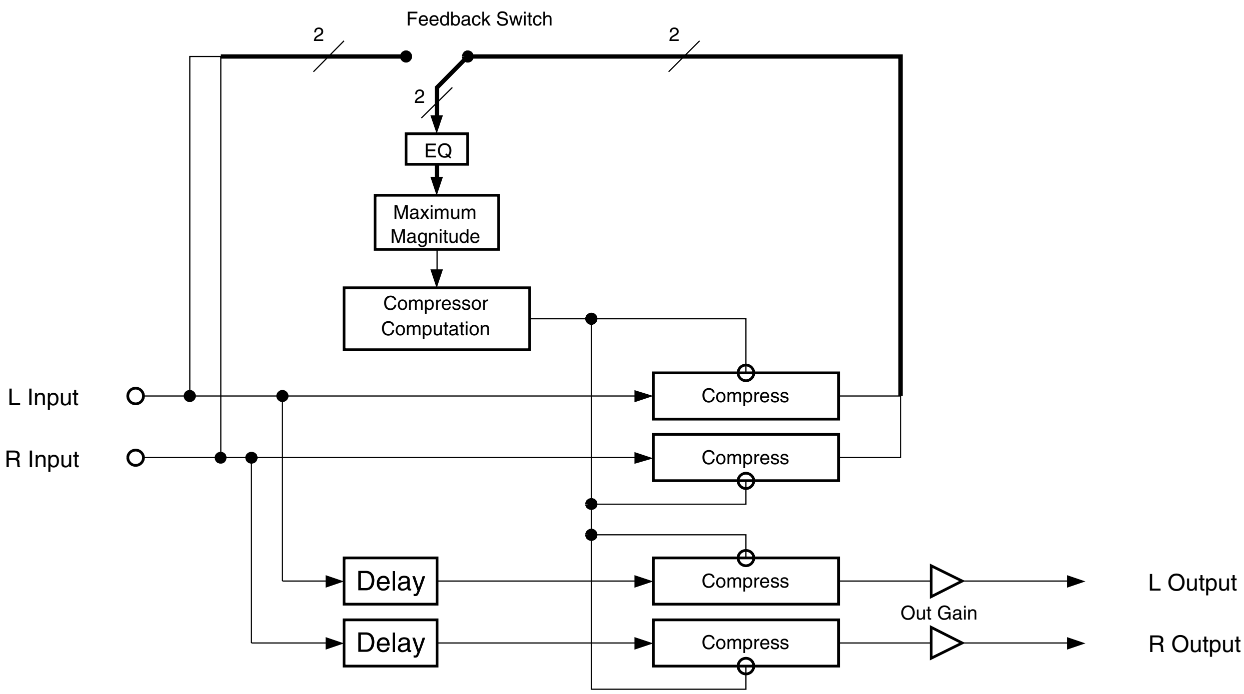


Figure 86 Compressor with side chain equalization

Using side chain equalization allows you to compress your signal based on the spectral (frequency) content of your signal. For example, by boosting the treble shelf filter, you can compress the signal only when there is a lot of high frequencies present.

Parameters:

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
FdbkComprs	In or Out	SC Input	L, R, (L+R)/2
		ComprsChan	L, R, L & R

Page 2

Atk Time	0.0 to 228.0 ms	Ratio	1.0:1 to 100.0:1, Inf:1
Rel Time	0 to 3000 ms	Threshold	-79.0 to 24.0 dB
SmoothTime	0.0 to 228.0 ms	MakeUpGain	Off, -79.0 to 24.0 dB
Signal Dly	0.0 to 25.0 ms		
		|||||||||||||||||||||||||||||	Reduction
	-dB 40 20 12 8	6 4 2	0

Page 3

SCBassGain	-79.0 to 24.0 dB	SCTrebGain	-79.0 to 24.0 dB
SCBassFreq	8 to 25088 Hz	SCTrebFreq	8 to 25088 Hz
SCMidGain	-79.0 to 24.0 dB		
SCMidFreq	8 to 25088 Hz		
SCMidWidth	0.010 to 5.000 oct		

In/Out When set to **In** the compressor is active; when set to **Out** the compressor is bypassed.

Out Gain Compressing the signal causes a reduction in signal level. To compensate, the output gain parameter may be used to increase the gain by as much as 24 dB. Note that the Out Gain parameter does not control the signal level when the algorithm is set to **Out**.

FdbkComprs A switch to set whether the compressor side chain is configured for feed-forward (**Out**) or feedback (**In**).

SC Input Select the input source channel for side-chain processing—left, right or the average of both.

ComprsChan Select which input channel will receive dynamics processing—left, right or both. If you select left or right, the opposite channel will pass through unaffected.

Atk Time The time for the compressor to start to cut in when there is an increase in signal level (attack) above the threshold.

Rel Time The time for the compressor to stop compressing when there is a reduction in signal level (release) from a signal level above the threshold.

SmoothTime A lowpass filter in the control signal path. It is intended to smooth the output of the compressor’s envelope detector. Smoothing will affect the attack or release times when the smoothing time is longer than one of the other times.

Signal Dly The time in ms by which the input signal should be delayed with respect to compressor side chain processing (i.e. side chain predelay). This allows the compression to appear to take effect just before the signal actually rises.

Ratio The compression ratio. High ratios are highly compressed; low ratios are moderately compressed.

Threshold The threshold level in dBFS (decibels relative to full scale) above which the signal begins to be compressed.

MakeUpGain Provides an additional control of the output gain. The Out Gain and MakeUpGain controls are additive (in decibels) and together may provide a maximum of 24 dB boost to offset gain reduction due to compression.

SCBassGain The amount of boost or cut that the side chain bass shelving filter should apply to the low frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the bass signal below the specified frequency. Negative values cut the bass signal below the specified frequency.

SCBassFreq The center frequency of the side chain bass shelving filter in intervals of one semitone.

SCTrebGain The amount of boost or cut that the side chain treble shelving filter should apply to the high frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the treble signal above the specified frequency. Negative values cut the treble signal above the specified frequency.

SCTrebFreq The center frequency of the side chain treble shelving filters in intervals of one semitone.

SCMidGain The amount of boost or cut that the side chain parametric mid filter should apply in dB to the specified frequency band. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the signal at the specified frequency. Negative values cut the signal at the specified frequency.

SCMidFreq The center frequency of the side chain parametric mid filter in intervals of one semitone. The boost or cut will be at a maximum at this frequency.

SCMidWidth The bandwidth of the side chain parametric mid filter may be adjusted. You specify the bandwidth in octaves. Small values result in a very narrow filter response. Large values result in a very broad response.

333 Opto Compress
334 Opto Comprs SCEQ

Compression with 2 release time constants

PAUs: 2 for **Opto Compress** and 3 for **Opto Comprs SCEQ**

Opto Compress is a basic compressor with two different release rates, which change from one rate to another as the compression gain reduction crosses a threshold set by the Rel Thres (release threshold) parameter. Except for the additional release rate and release threshold parameters, this compressor is like Algorithm **331 SoftKneeCompress on page 191**.

Like a normal compressor, this compressor reduces the gain of a signal as the signal increases above the compression threshold set with CompThres parameter. The amount of compression is expressed as a ratio. The compression ratio is the inverse of the slope of the compressor input/output characteristic. The amount of compression is based on the side chain input signal (left, right or larger of left and right magnitudes). A compression ratio of **1:1** will have no effect on the signal. An infinite ratio, will compress all signal levels above the threshold level to the threshold level (zero slope). For ratios in between infinite and **1:1**, increasing the input will increase the output, but by less than it would if there was no compression. The threshold is expressed as a decibel level relative to digital full-scale (dBFS) where **0 dBFS** is digital full-scale and all other available values are negative.

To determine how much to compress the signal, the compressor must measure the signal level. Since musical signal levels will change over time, the compression amounts must change as well. You can control the rate at which compression changes in response to changing signal levels with the attack and release time controls. With the attack time, you set how fast the compressor responds to increased levels. At long attack times, the signal may overshoot the threshold level for some time before it becomes fully compressed, while at short attack times, the compressor will rapidly clamp down on the level. The release time controls how long it takes the compressor to respond to a reduction in signal levels. At long release times, the signal may stay compressed well after the signal falls below threshold. At short release times, the compressor will open up almost as soon as the signal drops.

Opto Compress has a special way of handling the release time. There are in fact two release times: Rel Time A and Rel Time B. There is also a release threshold (Rel Thres) parameter. While the amount of compression gain reduction exceeds the release threshold, the release time will be determined by Rel Time A. As soon as the amount of reduction is less than the threshold, the release time changes to Rel Time B.

You can set the release times so that the release is initially very fast, and then let it slow down when it crosses the release threshold. Likewise, you can have a very slow release, which suddenly speeds up as reduction reaches the release threshold. Be sure to watch the reduction meter to follow what is happening as you change the parameters.

You can select which channel, left (**L**), right (**R**) or the maximum amplitude of the two (**L & R**) is used to control the compression (side chain processing) with the SC Input parameter. You can also select which channel is actually compressed, again left (**L**), right (**R**) or both (**L & R**) using the ComprsChan parameter.

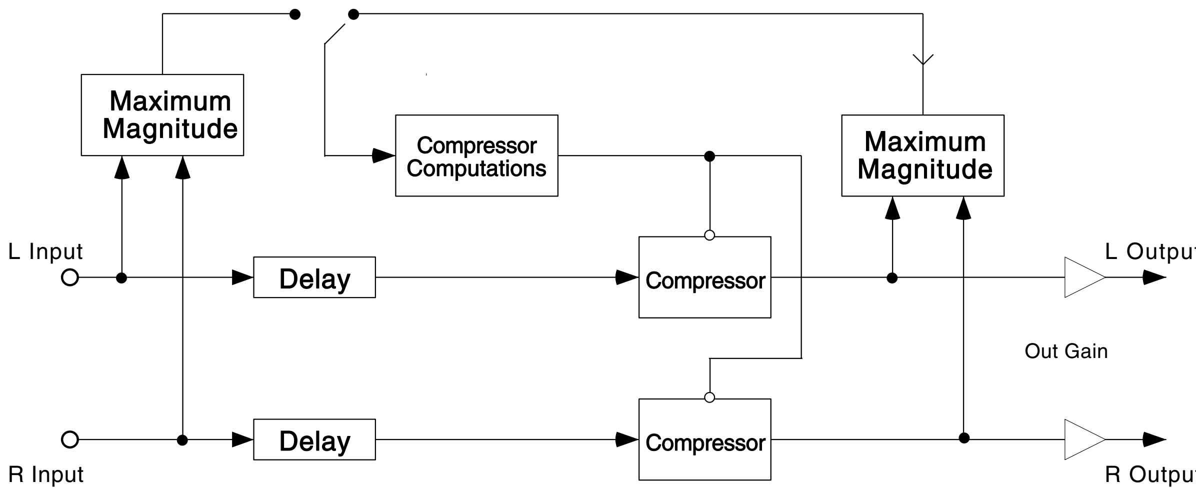


Figure 87 Opto Compress

The soft-knee compressor is used which has a more gradual transition from compressed to unity gain.

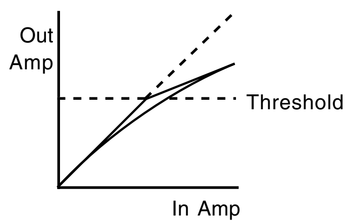


Figure 88 Soft-Knee compression characteristic

For typical compressor behavior, the attack time is considerably shorter than the release time. At very short attack and release times, the compressor is almost able to keep up with the instantaneous signal levels and the algorithm will behave more like distortion than compression. In addition to the attack and release times, there is another time parameter: SmoothTime. The smoothing parameter will increase both the attack and release times, although the effect is significant only when its time is longer than the attack or release time. Generally the smoothing time should be kept at or shorter than the attack time.

Opto Comprs SCEQ is very similar to **Opto Compress** except equalization of the side chain is provided. Using side chain equalization allows you to compress your signal based on the spectral (frequency) content of your signal. For example, by boosting the treble shelf filter, you can compress the signal only when there is a lot of high frequencies present.

You have the choice of using the compressors configured as feed-forward or feedback compressors. For feed-forward, set the FdbkComprs parameter to **Out**; for feedback compression, set it to **In**. The feed-forward configuration uses the input signal as the side-chain source. The feedback compressor on the other hand uses the compressor output as the side-chain source. Feedback compression tends to be more subtle, but you cannot get an instant attack.

In the feedback configuration, the signal being compressed may be delayed relative to the side chain compression processing. The delay allows the signal to start being compressed just before an attack transient arrives. Since the side chain processing “knows” what the input signal is going to be before the main signal path does, it can tame down an attack transient by compressing the attack before it actually happens.

A meter is provided to display the amount of gain reduction that is applied to the signal as a result of compression.

Parameters:

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
FdbkComprs	In or Out	SC Input	L, R, L & R
Signal Dly	0.0 to 25.0 ms	ComprsChan	L, R, L & R

Page 2

Atk Time	0.0 to 228.0 ms	Ratio	1:1.0 to 1:17.0
Rel Time A	0 to 3000 ms	Comp Thres	-79.0 to 0.0 dB
Rel Time B	0 to 3000 ms	Rel Thres	-79.0 to 0.0 dB
SmthTime	0.0 to 228.0 ms	MakeUpGain	Off, -79.0 to 24.0 dB
		||||||||||||||||||||||||||||	Reduction
	-dB 40 20 12	8 6 4 2	0

Page 3 (Opto Comprs SCEQ)

SCBassGain	-79.0 to 24.0 dB	SCTrebGain	-79.0 to 24.0 dB
SCBassFreq	8 to 25088 Hz	SCTrebFreq	8 to 25088 Hz
SCMidGain	-79.0 to 24.0 dB		
SCMidFreq	8 to 25088 Hz		
SCMidWidth	0.010 to 5.000 oct		

In/Out When set to **In** the compressor is active; when set to **Out** the compressor is bypassed.

Out Gain The output gain parameter may be used to increase the gain by as much as 24 dB, or reduce the gain to nothing. Note that the Out Gain parameter does not control the signal level when the algorithm is set to **Out**.

SC Input Select the input source channel for side-chain processing—left (**L**), right (**R**) or both (**L & R**). When set to **L & R**, the maximum amplitude is used.

ComprsChan Select which input channel will receive compression processing—left, right or both. If you select left or right, the opposite channel will pass through unaffected.

FdbkComprs A switch to set whether the compressor side chain is configured for feed-forward (**Out**) or feedback (**In**).

Atk Time The time for the compressor to start to cut in when there is an increase in signal level (attack) above the threshold.

Rel Time A The time for the compressor to stop compressing when there is a reduction in signal level (release) from a signal level above the threshold. This release time is active while the signal is reduced by more than the release threshold setting.

Rel Time B The time for the compressor to stop compressing when there is a reduction in signal level (release) from a signal level above the threshold. This release time is active while the signal is reduced by less than the release threshold setting.

Rel Thres When the signal is reduced by more than this release threshold, the release time is set by Rel Time A. Otherwise the release time is set by Rel Time B.

SmthTime A lowpass filter in the control signal path. It is intended to smooth the output of the compressor's envelope detector. Smoothing will affect the attack or release times when the smoothing time is longer than one of the other times.

Signal Dly The time in ms by which the input signal should be delayed with respect to compressor side chain processing (i.e. side chain predelay). This allows the compression to appear to take effect just before the signal actually rises.

Ratio The compression ratio in effect above the compression threshold. High ratios are highly compressed; low ratios are moderately compressed.

Comp Thres The compression threshold level in dBFS (decibels relative to full scale) above which the signal begins to be compressed.

MakeUpGain Provides an additional control of the output gain. The Out Gain and MakeUpGain controls are additive (in decibels) and together may provide a maximum of 24 dB boost to offset gain reduction due to compression.

SCBassGain For **Opto Comprs SCEQ**. The amount of boost or cut that the side chain bass shelving filter should apply to the low frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the bass signal below the specified frequency. Negative values cut the bass signal below the specified frequency.

SCBassFreq For **Opto Comprs SCEQ**. The center frequency of the side chain bass shelving filter in intervals of one semitone.

SCTrebGain For **Opto Comprs SCEQ**. The amount of boost or cut that the side chain treble shelving filter should apply to the high frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the treble signal above the specified frequency. Negative values cut the treble signal above the specified frequency.

SCTrebFreq For **Opto Comprs SCEQ**. The center frequency of the side chain treble shelving filters in intervals of one semitone.

SCMidGain For **Opto Comprs SCEQ**. The amount of boost or cut that the side chain parametric mid filter should apply in dB to the specified frequency band. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the signal at the specified frequency. Negative values cut the signal at the specified frequency.

SCMidFreq For **Opto Comprs SCEQ**. The center frequency of the side chain parametric mid filter in intervals of one semitone. The boost or cut will be at a maximum at this frequency.

SCMidWidth For **Opto Comprs SCEQ**. The bandwidth of the side chain parametric mid filter may be adjusted. You specify the bandwidth in octaves. Small values result in a very narrow filter response. Large values result in a very broad response.

335 Band Compress

Stereo algorithm to compress a single frequency band

PAUs: 3

Band Compress is in most respects identical to Algorithm **331 SoftKneeCompress on page 191**. However, **Band Compress** compresses only on a single band of frequencies. Frequency band selection is based on a parametric filter. You control the filter center frequency and bandwidth. The compressor controls the filter gain.

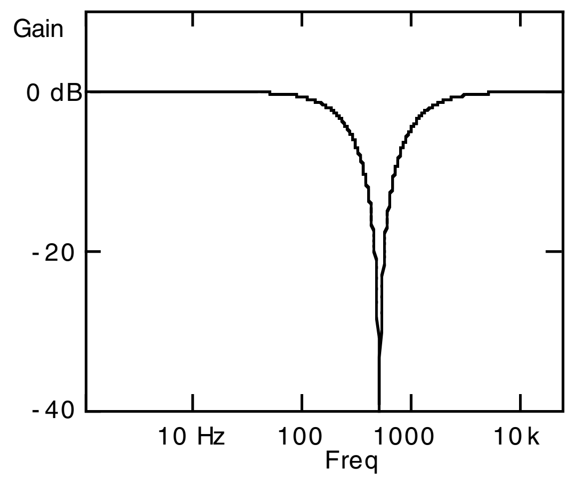


Figure 89 Band Compress filtering at full compression

The compressor reduces the signal level when the signal level exceeds a threshold. The amount of compression is expressed as a ratio. The compression ratio is the inverse of the slope of the compressor input/output characteristic. The amount of compression is based on the sum of the magnitudes of the left and right channels. A compression ratio of **1:1** will have no effect on the signal. An infinite ratio, will compress all signal levels above the threshold level to the threshold level (zero slope). For ratios in between infinite and **1:1**, increasing the input will increase the output, but by less than it would if there was no compression. The threshold is expressed as a decibel level relative to digital full-scale (dBFS) where **0 dBFS** is digital full-scale and all other available values are negative.

With **Band Compress**, the side chain processing acts only on a specified band of frequencies. At the input of the side chain is a bandpass filter which passes only the frequency band of interest. The side chain output then controls the gain of a band cut filter acting on the main signal path. The bandpass and band cut filters both are set to the same center frequency and bandwidth which you control. The depth of the band cut filter is of course set by the compressor side chain processing.

You can select which channel, left (**L**), right (**R**) or the maximum amplitude of the two (**L & R**) is used to control the compression (side chain processing) with the SC Input parameter. You can also select which channel is actually compressed, again left (**L**), right (**R**) or both (**L & R**) using the ComprsChan parameter.

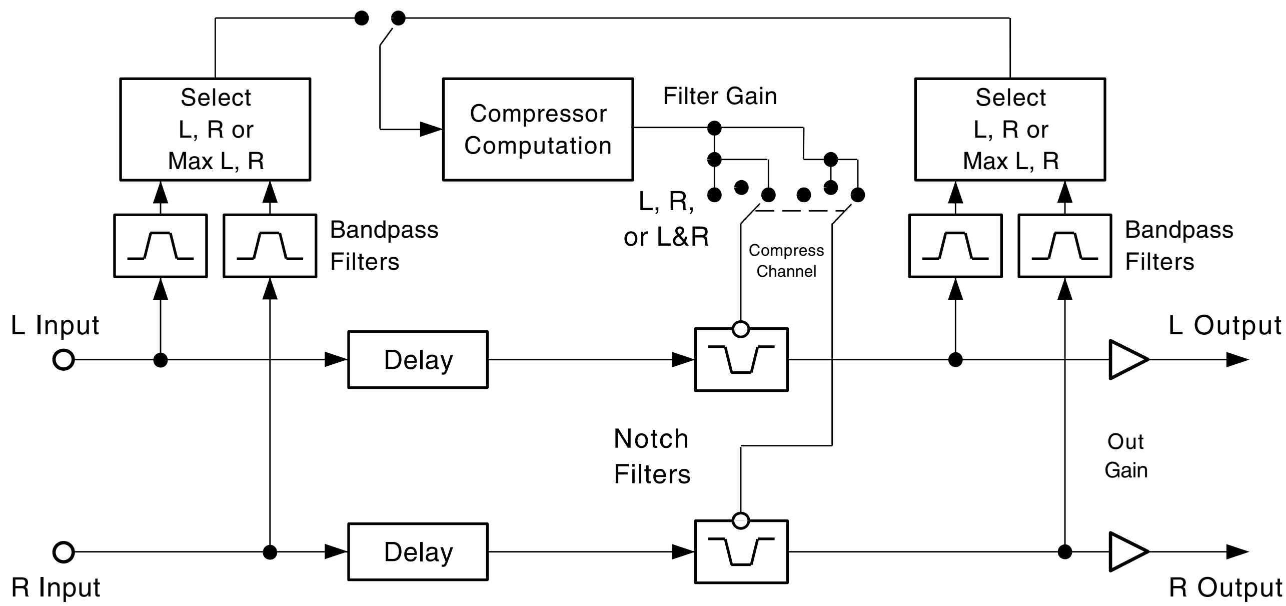


Figure 90 Band Compress block diagram

The soft-knee compressor is used which has a more gradual transition from compressed to unity gain.

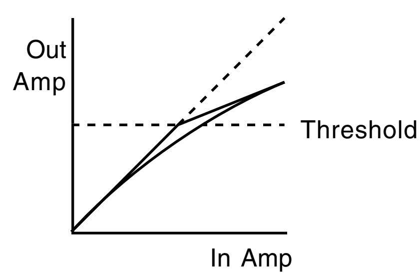


Figure 91 Soft-Knee compression characteristic

To determine how much to compress the signal, the compressor must measure the signal level. Since musical signal levels will change over time, the compression amounts must change as well. You can control the rate at which compression changes in response to changing signal levels with the attack and release time controls. With the attack time, you set how fast the compressor responds to increased levels. At long attack times, the signal may overshoot the threshold level for some time before it becomes fully compressed, while at short attack times, the compressor will rapidly clamp down on the level. The release time controls how long it takes the compressor to respond to a reduction in signal levels. At long release times, the signal may stay compressed well after the signal falls below threshold. At short release times, the compressor will open up almost as soon as the signal drops.

For typical compressor behavior, the attack time is considerably shorter than the release time. At very short attack and release times, the compressor is almost able to keep up with the instantaneous signal levels and the algorithm will behave more like distortion than compression. In addition to the attack and release times, there is another time parameter: SmoothTime. The smoothing parameter will increase both the attack and release times, although the effect is significant only when its time is longer than the attack or release time. Generally the smoothing time should be kept at or shorter than the attack time.

You have the choice of using the compressors configured as feed-forward or feedback compressors. For feed-forward, set the FdbkComprs parameter to **Out**; for feedback compression, set it to **In**. The feed-forward configuration uses the input signal as the side-chain source. The feedback compressor on the other hand uses the compressor output as the side-chain source. Feedback compression tends to be more subtle, but you cannot get an instant attack.

In the feedback configuration, the signal being compressed may be delayed relative to the side chain compression processing. The delay allows the signal to start being compressed just before an attack transient arrives. Since the side chain processing "knows" what the input signal is going to be before the main signal path does, it can tame down an attack transient by compressing the attack before it actually happens.

A meter is provided to display the amount of gain reduction that is applied to the signal as a result of compression.

Parameters:

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
Band FreqC	8 to 25088 Hz	FdbkCmprs	In or Out
Band FreqF	-100 to 100 ct	SC Input	L, R, L & R
Band Width	0.010 to 5.000 oct	ComprsChan	L, R, L & R

Page 2

Atk Time	0.0 to 228.0 ms	Ratio	1.0:1 to 100.0:1, Inf:1
Rel Time	0 to 3000 ms	Threshold	-79.0 to 0.0 dB
SmoothTime	0.0 to 228.0 ms	MakeUpGain	Off, -79.0 to 24.0 dB
Signal Dly	0.0 to 25.0 ms		
		||||||||||||||||||||||||||||	Reduction
	-dB 40 20 12 8	6 4 2	0

In/Out When set to **In** the compressor is active; when set to **Out** the compressor is bypassed.

Out Gain The output gain parameter may be used to increase the gain by as much as 24 dB, or reduce the gain to nothing. Note that the Out Gain parameter does not control the signal level when the algorithm is set to **Out**.

Band FreqC The coarse control for the center frequency of the filter band to be compressed. Only signal components centered in the band will be compressed.

Band FreqF The fine control for the center frequency of the filter band to be compressed. Only signal components centered in the band will be compressed.

Band Width The width of the frequency band to be compressed in octaves. Small values compress a very narrow range of frequencies. Large values compress a broad range of frequencies.

SC Input Select the input source channel for side-chain processing—left (**L**), right (**R**) or both (**L & R**). When set to **L & R**, the maximum amplitude is used.

ComprsChan Select which input channel will receive compression processing—left, right or both. If you select left or right, the opposite channel will pass through unaffected.

FdbkComprs A switch to set whether the compressor side chain is configured for feed-forward (**Out**) or feedback (**In**).

Atk Time The time for the compressor to start to cut in when there is an increase in signal level (attack) above the threshold.

Rel Time The time for the compressor to stop compressing when there is a reduction in signal level (release) from a signal level above the threshold.

SmoothTime A lowpass filter in the control signal path. It is intended to smooth the output of the compressor's envelope detector. Smoothing will affect the attack or release times when the smoothing time is longer than one of the other times.

Signal Dly The time in ms by which the input signal should be delayed with respect to compressor side chain processing (i.e. side chain predelay). This allows the compression to appear to take effect just before the signal actually rises.

Ratio The compression ratio in effect above the compression threshold. High ratios are highly compressed; low ratios are moderately compressed.

Threshold The compression threshold level in dBFS (decibels relative to full scale) above which the signal begins to be compressed.

MakeUpGain Provides an additional control of the output gain. The Out Gain and MakeUpGain controls are additive (in decibels) and together may provide a maximum of 24 dB boost to offset gain reduction due to compression.

336 3 Band Compress
349 Dual 3 Band Comp
665 Mn 3 Band Comprs

Soft-knee 3 frequency band compression algorithm

PAUs: 3 for **Mn 3 Band Comprs**
4 for **3 Band Compress**
8 for **Dual 3 Band Comp**

The three-band compressor divides the input stereo signal into 3 frequency bands and runs each band through its own soft-knee compressor. After compression, the bands are summed back together to produce the output. You may set the frequencies at which the bands are split.

Three configurations of the three-band compressor are available. **3 Band Compress** is a stereo algorithm with the controls affecting both left and right channels equally. **Dual 3 Band Comp** is a pair of independent mono three-band compressors sitting on a stereo bus (one compressor on the left channel, the other on the right). **Mn 3 Band Comprs** is a pure mono three-band compressor.

The compressors reduce the signal level when the signal exceeds a threshold. The amount of compression is expressed as a ratio. The compression ratio is the inverse of the slope of the compressor input/output characteristic. The amount of compression is based on the sum of the magnitudes of the left and right channels. A compression ratio of **1:1** will have no effect on the signal. An infinite ratio, will compress all signal levels above the threshold level to the threshold level (zero slope). For ratios in between infinite and **1:1**, increasing the input will increase the output, but by less than it would if there was no compression. The threshold is expressed as a decibel level relative to digital full-scale (dBFS) where **0 dBFS** is digital full-scale and all other available values are negative.

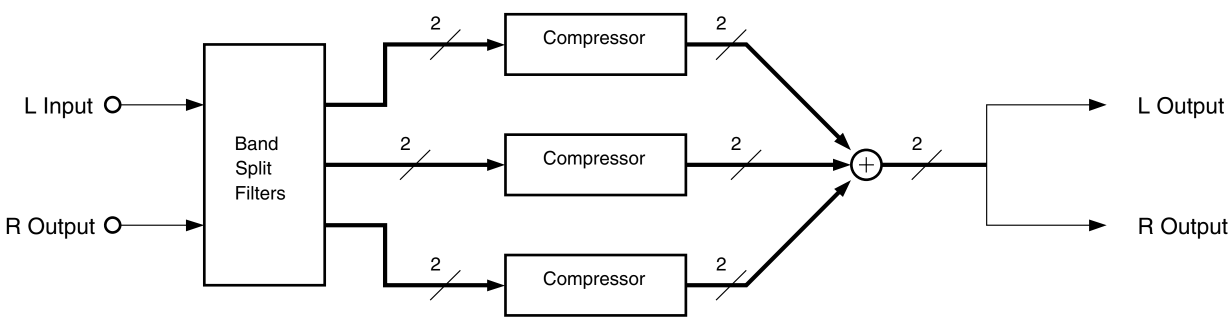


Figure 92 A three-band compressor

In the soft-knee compressor there is a gradual transition from compressed to unity gain.

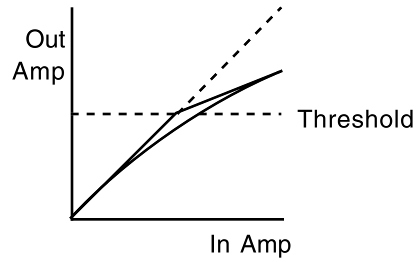


Figure 93 Soft-Knee compression characteristics

To determine how much to compress the signal, the compressor must measure the signal level. Since musical signal levels will change over time, the compression amounts must change as well. You can control the rate at which compression changes in response to changing signal levels with the attack and release time controls. With the attack time, you set how fast the compressor responds to increased levels. For long attack times, the signal may overshoot the threshold level for some time before it becomes fully compressed, while for short attack times, the compressor will rapidly clamp down on the level. The release time controls how long it takes the compressor to respond to a reduction in signal levels. At long release times, the signal may stay compressed well after the signal falls below threshold. At short release times, the compressor will open up almost as soon as the signal drops.

For typical compressor behavior, the attack time is considerably shorter than the release time. At very short attack and release times, the compressor is almost able to keep up with the instantaneous signal levels and the algorithm will behave more like distortion than compression. In addition to the attack and release times, there is another time parameter: “Smth *Band*.” The smoothing parameter will increase both the attack and release times, although the effect is significant only when its time is longer than the attack or release time. Generally the smoothing time should be kept at or shorter than the attack time.

The stereo compressor allows you to select which input(s) to use for side chain processing. The choices are left (**L**), right (**R**) or the average of both left and right**—(L+R)/2**. The stereo compressors let you select whether compression should affect both channels or only one channel.

You have the choice of using the compressors configured as feed-forward or feedback compressors. For feed-forward, set the FdbkComprs parameter to “Out”; for feedback compression, set it to “In”. The feed-forward configuration uses the input signal as the side-chain source. The feedback compressor on the other hand uses the compressor output as the side-chain source. Feedback compression tends to be more subtle, but you cannot get an instant attack.

The signal being compressed may be delayed relative to the side chain compression processing. The delay allows the signal to start being compressed just before an attack transient arrives. Since the side chain processing “knows” what the input signal is going to be before the main signal path does, it can tame down an attack transient by compressing the attack before it actually happens. This feature works whether the side chain is configured for feed-forward or feedback.

A meter is provided for each compression band to display the amount of gain reduction that is applied to the signal as a result of compression.

Parameters:

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
SC Input	L, R, (L+R)/2	ComprsChan	L, R, L&R
FdbkComprs	In or Out	Crossover1	8 to 25088 Hz
Signal Dly	0.0 to 25.0 ms	Crossover2	8 to 25088 Hz

Page 2

Atk Low	0.0 to 228.0 ms	Ratio Low	1.0:1 to 100.0:1, Inf:1
Rel Low	0 to 3000 ms	Thres Low	-79.0 to 24.0 dB
Smth Low	0.0 to 228.0 ms	MakeUp Low	Off, -79.0 to 24.0 dB
		Low Bans	
			Reduction
	-dB 40 20 12	8 6 4 2	0

Page 3

Atk Mid	0.0 to 228.0 ms	Ratio Mid	1.0:1 to 100.0:1, Inf:1
Rel Mid	0 to 3000 ms	Thres Mid	-79.0 to 24.0 dB
Smth Mid	0.0 to 228.0 ms	MakeUp Mid	Off, -79.0 to 24.0 dB
		Mid Band	
			Reduction
	-dB 40 20 12	8 6 4 2	0

Page 4

Atk High	0.0 to 228.0 ms	Ratio High	1.0:1 to 100.0:1, Inf:1
Rel High	0 to 3000 ms	Thres High	-79.0 to 24.0 dB
Smth High	0.0 to 228.0 ms	MakeUpHigh	Off, -79.0 to 24.0 dB
		High Band	
			Reduction
	-dB 40 20 12	8 6 4 2	0

In/Out When set to **In** the compressor is active; when set to **Out** the compressor is bypassed.

Out Gain Compressing the signal causes a reduction in signal level. To compensate, the output gain parameter may be used to increase the gain by as much as 24 dB. Note that the Out Gain parameter does not control the signal level when the algorithm is set to **Out**.

SC Input Select the input source channel for side-chain processing—left, right or the average of both.

ComprsChan Select which input channel will receive dynamics processing—left, right or both. If you select left or right, the opposite channel will pass through unaffected.

FdbkComprs A switch to set whether the compressor side chain is configured for feed-forward (**Out**) or feedback (**In**).

Signal Dly The time in ms by which the input signal should be delayed with respect to compressor side chain processing (i.e. side chain predelay). This allows the compression to appear to take effect just before the signal actually rises.

CrossoverN The Crossover parameters (1 and 2) set the frequencies which divide the three compression frequency bands. The two parameters are interchangeable, so either may contain the higher frequency value.

Atk Band (Band is Low, Mid or High) The time for the compressor to start to cut in when there is an increase in signal level (attack) above the threshold.

Rel Band (Band is Low, Mid or High) The time for the compressor to stop compressing when there is a reduction in signal level (release) from a signal level above the threshold.

Smth Band (Band is Low, Mid or High) A lowpass filter in the control signal path. It is intended to smooth the output of the compressor's envelope detector. Smoothing will affect the attack or release times when the smoothing time is longer than one of the other times.

RatioBand (Band is Low, Mid or High) The compression ratio. High ratios are highly compressed; low ratios are moderately compressed.

ThresBand (Band is Low, Mid or High) The threshold level in dBFS (decibels relative to full scale) above which the signal begins to be compressed.

340 Expander
662 Mn Expander

A stereo expansion algorithm

PAUs: 1

This is a stereo expander algorithm. The algorithms expands the signal (reduced the signal's gain) when the signal falls below the expansion threshold. The amount of expansion is based on the larger magnitude of the left and right channels. The amount of expansion is expressed as an expansion ratio. Expanding a signal reduces its level below the threshold. The expansion ratio is the inverse of the slope of the expander input/output characteristic. An expansion ratio of **1:1** will have no effect on the signal. A zero ratio (**1:∞**), will expand all signal levels below the threshold level to the null or zero level. (This expander expands to **1:17** at most.) Thresholds are expressed as a decibel level relative to digital full-scale (dBFS) where **0 dBFS** is digital full-scale and all other available values are negative.

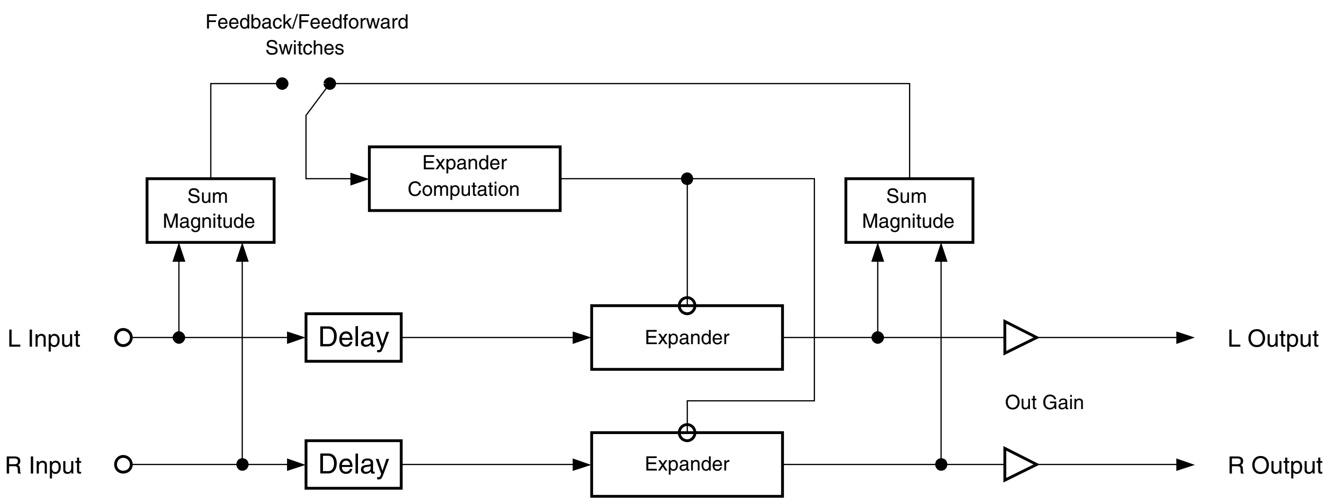


Figure 94 Expander

To determine how much to expand the signal, the expander must measure the signal level. Since musical signal levels will change over time, the expansion amounts must change as well. You can control how fast the expansion changes in response to changing signal levels with the attack and release time controls.

The attack time is defined as the time for the expansion to turn off when the signal rises above the threshold. This time should be very short for most applications. The expander release time is the time for the signal to expand down after the signal drops below threshold. The expander release time may be set quite long. An expander may be used to suppress background noise in the absence of signal, thus typical expander settings use a fast attack (to avoid losing real signal), slow release (to gradually fade out the noise), and the threshold set just above the noise level. You can set just how far to drop the noise with the expansion ratio.

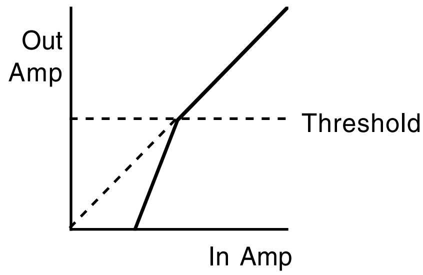


Figure 95 Expansion transfer characteristic

The signal being expanded may be delayed relative to the side chain processing. The delay allows the signal to stop being expanded just before an attack transient arrives. Since the side chain processing "knows" what the input signal is going to be before the main signal path does, it can tame down an attack transient by releasing the expander before the attack actually happens.

A meter is provided to display the amount of gain reduction that is applied to the signal as a result of expansion.

Parameters

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB

Page 2

Atk Time	0.0 to 228.0 ms	Ratio	1:1.0 to 1:17.0
Rel Time	0 to 3000 ms	Threshold	-79.0 to 0.0 dB
SmoothTime	0.0 to 228.0 ms	MakeUpGain	Off, -79.0 to 24.0 dB
Signal Dly	0.0 to 25.0 ms		

In/Out When set to **In** the expander is active; when set to **Out** the expander is bypassed.

Out Gain The output gain parameter may be used to increase the gain by as much as 24 dB, or reduce the gain to nothing. Note that the Out Gain parameter does not control the signal level when the algorithm is set to **Out**.

Atk Time The time for the expander to increase the gain of the signal (turns off the expander) after the signal rises above threshold.

Rel Time The time for the expander to reduce the signal level when the signal drops below the threshold (turning on expansion).

SmoothTime A lowpass filter in the control signal path. It is intended to smooth the output of the expander's envelope detector. Smoothing will affect the attack or release times when the smoothing time is longer than one of the other times.

Signal Dly The time in ms by which the input signal should be delayed with respect to expander side chain processing (i.e. side chain predelay). This allows the expansion to appear to turn off just before the signal actually rises.

Ratio	The expansion ratio. High values (**1:17** max) are highly expanded, low values (**1:1** min) are moderately expanded.
Threshold	The expansion threshold level in dBFS (decibels relative to full scale) below which the signal begins to be expanded.
MakeUpGain	Provides an additional control of the output gain. The Out Gain and MakeUpGain controls are additive (in decibels) and together may provide a maximum of 24 dB boost to offset gain reduction due to expansion.

341 Compress/Expand
342 Comp/Exp + EQ
664 Mn Comprs/Expand

A stereo soft-knee compression and expansion algorithm with and without equalization

PAUs: 2 for **Compress/Expand**
3 for **Comp/Exp + EQ**

These are stereo compressor and expander algorithms. One version is followed by equalization and the other is not. The algorithms compress the signal level when the signal exceeds a compression threshold and expands the signal when the signal falls below the expansion threshold. The amount of compression and/or expansion is based on the larger magnitude of the left and right channels.

Compression is expressed as a ratio: the inverse of the slope of the compressor input/output characteristic. A compression ratio of **1:1** has no effect on the signal. An infinite ratio compresses all signal levels above the threshold level to the threshold level (zero slope). For ratios between infinite and **1:1**, increasing the input will increase the output, but by less than it would without compression. The compressor is a soft-knee compressor, so the transition from compressed to linear is gradual.

The amount of expansion is expressed as an expansion ratio. Expanding a signal reduces its level below the threshold. The expansion ratio is the inverse of the slope of the expander input/output characteristic. An expansion ratio of **1:1** will have no effect on the signal. A zero ratio (**1:∞**), will expand all signal levels below the threshold level to the null or zero level. (This expander expands to **1:17** at most.) Thresholds are expressed as a decibel level relative to digital full-scale (dBFS) where **0 dBFS** is digital full-scale and all other available values are negative.

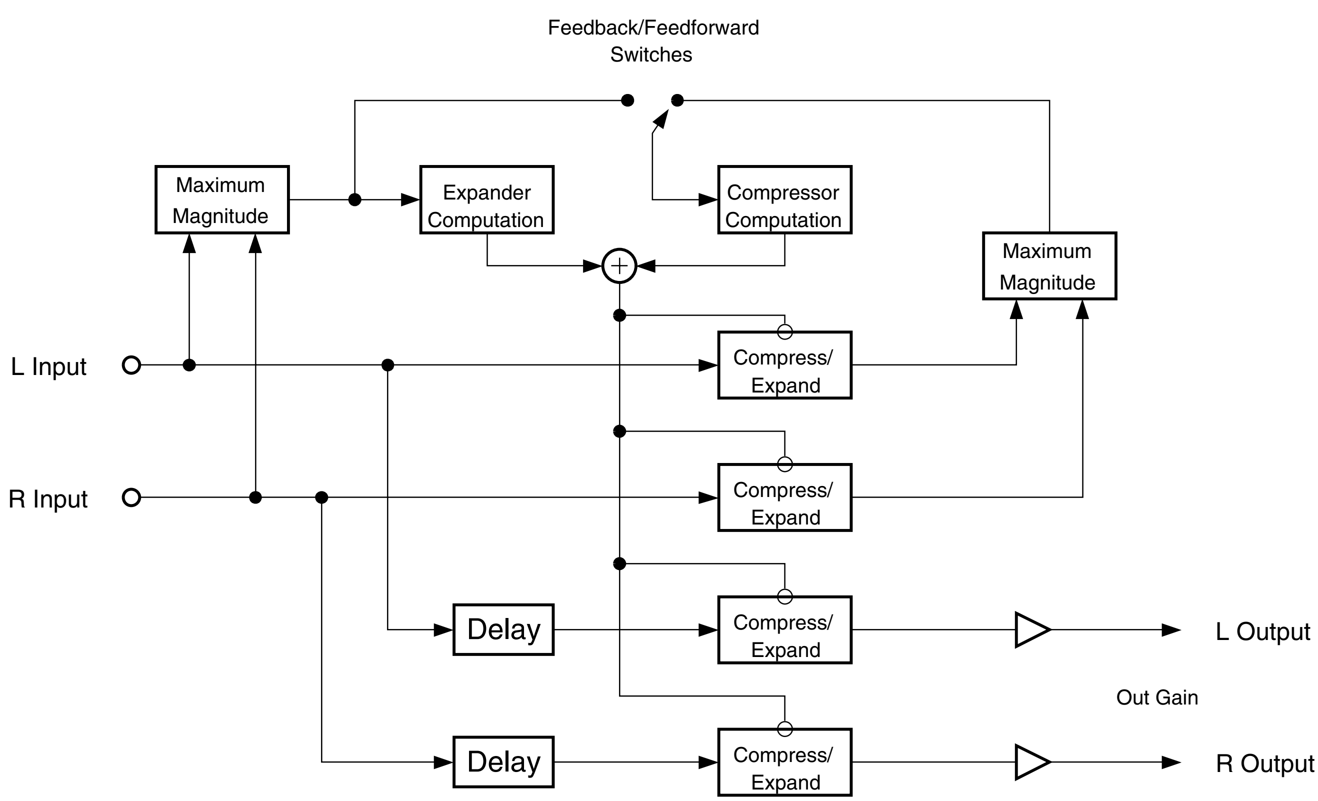


Figure 96 **Compress/Expand (optional EQ not shown)**

To determine how much to compress or expand the signal, the compressor/expander must measure the signal level. Since musical signal levels will change over time, the compression and expansion amounts must change as well. You can control how fast the compression or expansion changes in response to changing signal levels with the attack and release time controls. Compression and expansion have separate controls.

First consider the compressor. With the attack time, you set how fast the compressor responds to increased levels. At long attack times, the signal may overshoot the threshold level for some time interval before it becomes fully compressed, while at short attack times, the compressor will rapidly clamp down on the level. The release time controls how long it takes the compressor to respond to a reduction in signal levels. At long release times, the signal may stay compressed well after the signal falls below threshold. At short release times, the compressor will open up almost as soon as the signal drops.

For typical compressor behavior, the attack time is considerably shorter than the release time. At very short attack and release times, the compressor is almost able to keep up with the instantaneous signal levels and the algorithm will behave more like distortion than compression. In addition to the attack and release times, there is another time parameter: SmoothTime. The smoothing parameter will increase both the attack and release times, although the effect is significant only when its time is longer than the attack or release times. Generally the smoothing time should be kept at or shorter than the attack time.

This compressor provides two compressed segments. The signal below the lower threshold is not compressed. The compression ratio corresponding to the lower threshold sets the amount of compression for the lower compression segment. Above the upper threshold, the signal is compressed even further by the ratio corresponding to the upper threshold. You may use the upper segment as a limiter (infinite compression), or you may use the two compression segments to produce compression with a softer knee than you would get otherwise. For example, to make the algorithm a compressor and limiter, first choose the two thresholds. The limiter will of course have the higher threshold. Set the compression ratio for the higher threshold to **Inf:1**. This gives you your limiter. Finally set the compression ratio for the lower threshold to the amount of compression that you want. Either pair of threshold and ratio parameters may be used for the upper compression segment—they are interchangeable. Above the upper threshold, the two compression ratios become additive. If both ratios are set to **3.0:1**, then the compression of the upper segment will be **6.0:1**. Another way to think of it is as two compressors wired in series (one after the other).

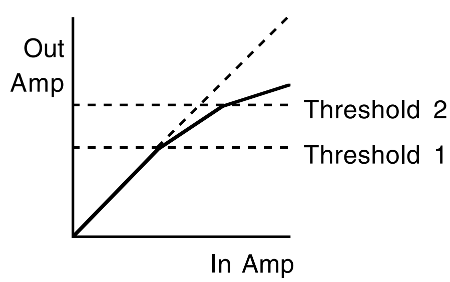


Figure 97 Two-segment compression characteristic

You have the choice of using the compressor configured as feed-forward or feedback. For feed-forward, set the FdbkComprs parameter to **Out**; for feedback compression, set it to **In**. The feed-forward configuration uses the input signal as the side-chain source. The feedback compressor on the other hand uses the compressor output as the side-chain source. Feedback compression tends to be more subtle, but you cannot get an instant attack.

The expander attack/release times are similar, though there is only one expand segment. The expander works independently of the compressor. The expander cannot be configured for feedback (if it could, it would always shut itself off permanently). The signal delay path does affect the expander. The attack time is defined as the time for the expansion to turn off when the signal rises above the threshold. This time should be very short for most applications. The expander release time is the time for the signal to expands down after the signal drops below threshold. The expander release time may be set quite long. An expander may be used to suppress background noise in the absence of signal, thus typical expander settings use a fast attack (to avoid losing real signal), slow release (to gradually fade out the noise), and the threshold set just above the noise level. You can set just how far to drop the noise with the expansion ratio.

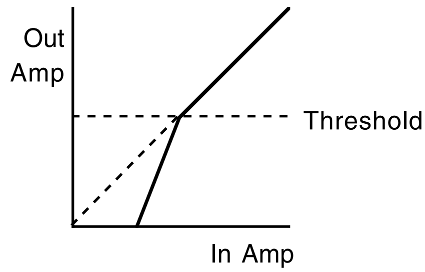


Figure 98 Expansion transfer characteristic

The signal being compressed/expanded may be delayed relative to the side chain processing. The delay allows the signal to start being compressed (or stop being expanded) just before an attack transient arrives. Since the side chain processing “knows” what the input signal is going to be before the main signal path does, it can tame down an attack transient by compressing the attack before it actually happens (or releasing the expander before the attack happens). This feature works whether the side chain is configured for feed-forward or feedback.

A meter is provided to display the amount of gain reduction that is applied to the signal as a result of compression and expansion.

The algorithm **Comp/Exp + EQ** differs from **Compress/Expand** in that the compressor and expander sections are followed by equalization filters. The output signal may be filtered with bass and treble shelving filters and a mid-range parametric filter.

Parameters

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
FdbkComprs	In or Out		

Page 2

Comp Atk	0.0 to 228.0 ms	Exp Atk	0.0 to 228.0 ms
Comp Rel	0 to 3000 ms	Exp Rel	0 to 3000 ms
SmoothTime	0.0 to 228.0 ms		
Signal Dly	0.0 to 25.0 ms		

Page 3

Comp1Ratio	1.0:1 to 100.0:1, Inf:1	Exp Ratio	1:1.0 to 1:17.0
Comp1Thres	-79.0 to 0.0 dB	Exp Thres	-79.0 to 0.0 dB
Comp2Ratio	1.0:1 to 100.0:1, Inf:1	MakeUpGain	Off, -79.0 to 24.0 dB
Comp2Thres	-79.0 to 0.0 dB		

Page 4

Bass Gain	-79.0 to 24.0 dB	Treb Gain	-79.0 to 24.0 dB
Bass Freq	8 to 25088 Hz	Treb Freq	8 to 25088 Hz
Mid Gain	-79.0 to 24.0 dB		
Mid Freq	8 to 25088 Hz		
Mid Wid	0.010 to 5.000 oct		

In/Out When set to **In** the compressor/expander is active; when set to **Out** the compressor/expander is bypassed.

Out Gain Compressing the signal causes a reduction in signal level. To compensate, the output gain parameter may be used to increase the gain by as much as 24 dB. Note that the Out Gain parameter does not control the signal level when the algorithm is set to **Out**.

FdbkComprs A switch to set whether the compressor side chain is configured for feed-forward (**Out**) or feedback (**In**). The expander is unaffected.

Comp Atk The time for the compressor to start to cut in when there is an increase in signal level (attack) above the threshold.

Comp Rel The time for the compressor to stop compressing when there is a reduction in signal level (release) from a signal level above the threshold.

Exp Atk The time for the expander to increase the gain of the signal (turns off the expander) after the signal rises above threshold.

Exp Rel The time for the expander to reduce the signal level when the signal drops below the threshold (turning on expansion).

SmoothTime A lowpass filter in the control signal path. It is intended to smooth the output of the expander's envelope detector. Smoothing will affect the attack or release times when the smoothing time is longer than one of the other times.

Signal Dly The time in ms by which the input signal should be delayed with respect to compressor side chain processing (i.e. side chain predelay). This allows the compression to appear to take effect just before the signal actually rises.

Comp1Ratio The compression ratio in effect above compression Threshold 1 (Comp1Thres). High ratios are highly compressed; low ratios are moderately compressed.

Comp1Thres One of two compression threshold levels. Threshold is expressed in dBFS (decibels relative to full scale) above which the signal begins to be compressed.

Comp2Ratio The compression ratio in effect above compression Threshold 2 (Comp2Thres). High ratios are highly compressed; low ratios are moderately compressed.

Comp2Thres One of two compression threshold levels. Threshold is expressed in dBFS (decibels relative to full scale) above which the signal begins to be compressed.

Exp Ratio The expansion ratio. High values (**1:17** max) are highly expanded, low values (**1:1** min) are moderately expanded.

Exp Thres The expansion threshold level in dBFS (decibels relative to full scale) below which the signal begins to be expanded.

MakeUpGain Provides an additional control of the output gain. The Out Gain and MakeUpGain controls are additive (in decibels) and together may provide a maximum of 24 dB boost to offset gain reduction due to compression or expansion.

Bass Gain For **Comp/Exp + EQ**. The amount of boost or cut that the bass shelving filter should apply to the low frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the bass signal below the specified frequency. Negative values cut the bass signal below the specified frequency.

Bass Freq For **Comp/Exp + EQ**. The center frequency of the bass shelving filter in intervals of one semitone.

Treb Gain For **Comp/Exp + EQ**. The amount of boost or cut that the treble shelving filter should apply to the high frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the treble signal above the specified frequency. Negative values cut the treble signal above the specified frequency. [Comp/Exp + EQ only]

Treb Freq For **Comp/Exp + EQ**. The center frequency of the treble shelving filter in intervals of one semitone.

Mid Gain For **Comp/Exp + EQ**. The amount of boost or cut that the mid parametric filter should apply in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the signal at the specified frequency. Negative values cut the signal at the specified frequency.

Mid Freq For **Comp/Exp + EQ**. The center frequency of the mid parametric filter in intervals of one semitone. The boost or cut will be at a maximum at this frequency.

Mid Wid For **Comp/Exp + EQ**. The bandwidth of the mid parametric filter may be adjusted. You specify the bandwidth in octaves. Small values result in a very narrow filter response. Large values result in a very broad response.

Gates

343 Gate
345 Gate w/SC EQ LFX
663 Mn Gate

Signal gate algorithms

PAUs: 1 for **Gate**
2 for **Gate w/SC EQ LFX**

Gate and **Gate w/SC EQ LFX** perform stand-alone gate processing and can be configured as a stereo or mono effects. As a stereo effect, the stereo signal gates itself based on its amplitude. As a mono effect, you can use one mono input signal to gate a second mono input signal (or one channel can gate itself). Separate output gain and panning for both channels is provided for improved mono processing flexibility.

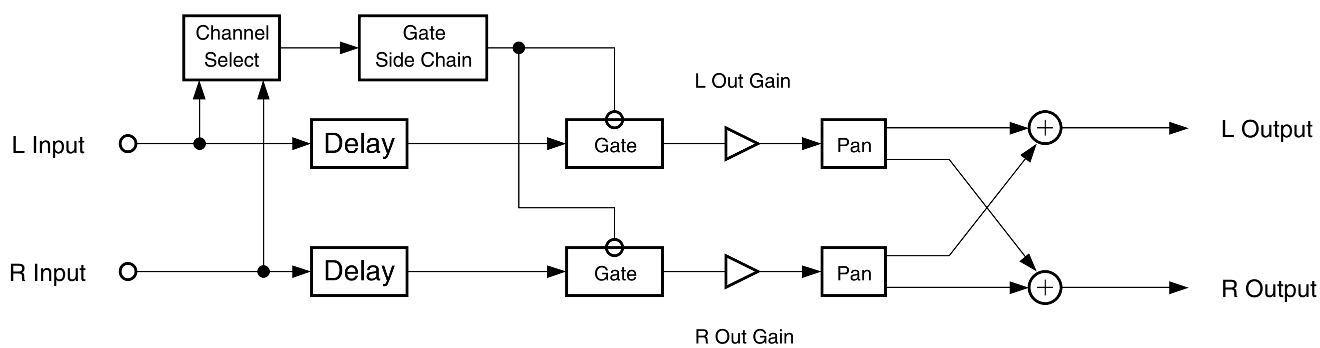


Figure 99 Gate

A gate behaves like an on off switch for a signal. One or both input channels is used to control whether the switch is on (gate is open) or off (gate is closed). The 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.

Gate w/SC EQ LFX will behave differently depending on whether the Retrigger parameter is set to **Off** or **On**. For the simpler **Gate**, there is no Retrigger parameter, and it is as if Retrigger is always **On**. If Retrigger is **On**, 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 with 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. You will typically use the gate with Retrigger set to on for percussive sounds.

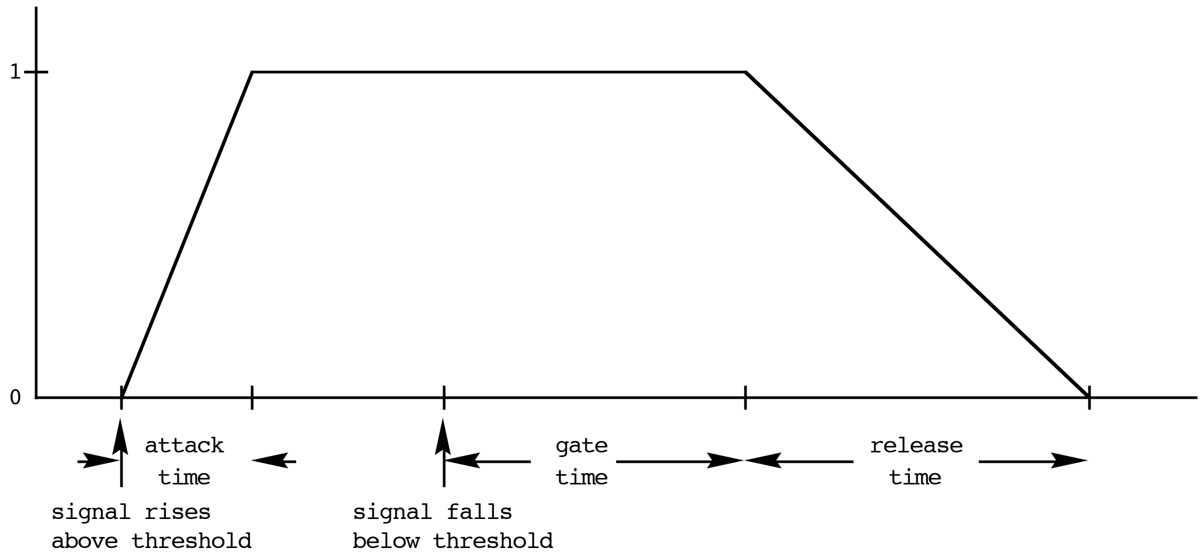


Figure 100 Signal envelope for Gate and Gate w/SC EQ LFX when Retrigger is On

If Retrigger is **Off** (**Gate w/SC EQ LFX** only), then the gate will open when the side chain signal rises above threshold as before. The gate will then close as soon as the gate time has elapsed, whether or not the signal is still above threshold. The gate will not open again until the envelope of the side chain signal falls below the threshold and rises above threshold again. Since an envelope follower is used, you can control how fast the envelope follows the signal with the Env Time parameter. Retrigger set to **Off** is useful for gating sustained sounds or where you need precise control of how long the gate should remain open.

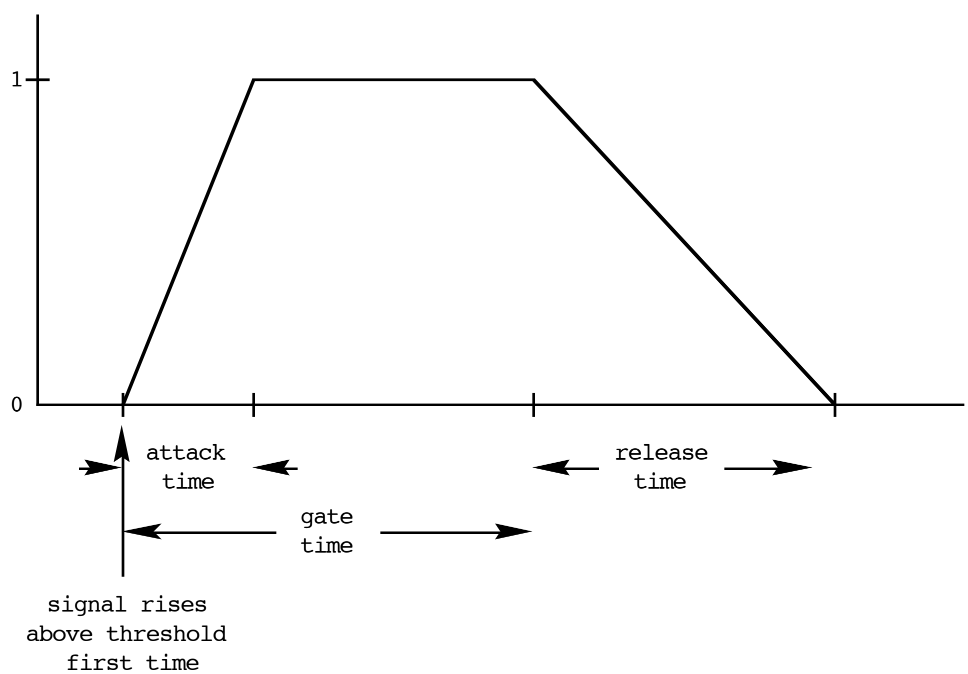


Figure 101 Gate w/SC EQ LFX signal envelope when Retrigger is Off

If Ducking 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 Atk Time (attack) and Rel Time (release) parameters are use to set the times for the gate to open and close. More precisely, depending on whether Ducking is off or on, Atk Time sets how fast the gate opens or closes when the side chain signal rises above the threshold. The Rel Time 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!

For **Gate w/SC EQ LFX** (not the simpler **Gate**), filtering can be done on the side chain signal. There are controls for a bass shelf filter, a treble shelf filter and a parametric (mid) filter. By filtering the side chain, you can control the sensitivity of the gate to different frequencies. For example, you can have the gate open only if high frequencies are present—or only if low frequencies are present.

Parameters for Gate

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
L Pan	-100 to 100 %	R Pan	-100 to 100 %
SC Input	(L+R)/2		

Page 2

Threshold	-79.0 to 0.0 dB	Gate Time	0 to 3000 ms
Ducking	On or Off	Atk Time	0.0 to 228.0 ms
Retrigger [Super]	On or Off	Rel Time	0 to 3000 ms
Env Time [Super]	0 to 3000 ms	Signal Dly	0.0 to 25.0 ms

Additional Parameters for Gate w/SC EQ LFX

Page 3

SCBassGain	-79.0 to 24.0 dB	SCTrebGain	-79.0 to 24.0 dB
SCBassFreq	8 to 25088 Hz	SCTrebFreq	8 to 25088 Hz
SCMidGain	-79.0 to 24.0 dB		
SCMidFreq	8 to 25088 Hz		
SCMidWidth	0.010 to 5.000 oct		

In/Out When set to **In** the gate is active; when set to **Out** the gate is bypassed.

Out Gain The output signal level in dB. The output gains are calculated before the final output panning.

L/R Pan Both of the gated signal channels can be panned between left and right prior to final output. This can be useful when the gate is used as a mono effect, and you don't want to hear one of the input channels, but you want your mono output panned to stereo. **-100%** is panned to the left, and **100%** is panned to the right.

SC Input The side chain input may be the amplitude of the left L input channel, the right R input channel, or the sum of the amplitudes of left and right (L+R)/2. You can gate a stereo signal with itself by using the sum, a mono signal with itself, or you can gate a mono signal using a second mono signal as the side chain.

Threshold The signal level in dB required to open the gate (or close the gate if Ducking is **On**).

Ducking 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.

Env Time Envelope time is for use when Retrigger is set to **Off**. The envelope time controls the time for the side chain signal envelope to drop below the threshold. At short times, the gate can reopen rapidly after it has closed, and you may find the gate opening unexpectedly due to an amplitude modulation of the side chain signal. For long times, the gate will remain closed until the envelope has a chance to fall, and you may miss gating events.

Gate Time 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. If Retrigger is **On**, the gate timer is continually reset while the side chain signal is above the threshold.

Atk Time The time for the gate to ramp from closed to open (reverse if Ducking is **On**) after the signal rises above threshold.

Rel Time The time for the gate to ramp from open to closed (reverse if Ducking is **On**) after the gate timer has elapsed.

Signal Dly The delay in milliseconds (ms) of the signal to be gated relative to the side chain signal. By delaying the main signal, the gate can be opened before the main signal rises above the gating threshold.

Gate w/SC EQ LFX Parameters

Retrigger If Retrigger is **On**, the gate timer is constantly restarted (retriggered) as long as the side chain signal is above the threshold. The gate then remains open (assuming Ducking is **Off**) until the signal falls below the threshold and the gate timer has elapsed. If Retrigger is **Off**, then the gate timer starts at the moment the signal rises above the threshold and the gate closes after the timer elapses, whether or not the signal is still above threshold. With Retrigger off, use the Env Time to control how fast the side chain signal envelope drops below the threshold. With Retrigger set to off, the side chain envelope must fall below threshold before the gate can open again.

SCBassGain The amount of boost or cut that the side chain bass shelving filter should apply to the low frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the bass signal below the specified frequency. Negative values cut the bass signal below the specified frequency.

SCBassFreq The center frequency of the side chain bass shelving filters in intervals of one semitone.

SCTrebGain The amount of boost or cut that the side chain treble shelving filter should apply to the high frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the treble signal above the specified frequency. Negative values cut the treble signal above the specified frequency.

SCTrebFreq The center frequency of the side chain treble shelving filters in intervals of one semitone.

SCMidGain The amount of boost or cut that the side chain parametric mid filter should apply in dB to the specified frequency band. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the signal at the specified frequency. Negative values cut the signal at the specified frequency.

SCMidFreq The center frequency of the side chain parametric mid filter in intervals of one semitone. The boost or cut will be at a maximum at this frequency.

SCMidWidth The bandwidth of the side chain parametric mid filter may be adjusted. You specify the bandwidth in octaves. Small values result in a very narrow filter response. Large values result in a very broad response.

EQs

350 3 Band EQ
351 5 Band EQ
354 Dual 5 Band EQ
671 Mn 6 Band EQ

Bass and treble shelving filters and parametric EQs

PAUs: 1 for **3 Band EQ**
2 for **Mn 6 Band EQ**
3 for **5 Band EQ** and **Dual 5 Band EQ**

These algorithms are multi-band equalizers with 1–4 bands of parametric EQ and with bass and treble tone controls. You can control the gain, frequency and bandwidth of each band of parametric EQ and control of the gain and frequencies of the bass and treble tone controls. The small **3 Band EQ** does not provide control of the bandwidth for the parametric Mid filter.

The algorithms **5 Band EQ** and **3 Band EQ** are stereo, meaning the parameters for the left and right channels are ganged—the parameters have the same effect on both channels. **Dual 5 Band EQ** provides separate control for left and right channels. **Mn 6 Band EQ** is a pure mono EQ.

Parameters:

Page 1 (3 Band EQ)

In/Out	In or Out	Mid Gain	-79.0 to 24.0 dB
		Mid Freq	8 to 25088 Hz
Bass Gain	-79.0 to 24.0 dB	Treb Gain	-79.0 to 24.0 dB
Bass Freq	8 to 25088 Hz	Treb Freq	8 to 25088 Hz

Page 1 (5 Band EQ)

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
Bass Gain	-79.0 to 24.0 dB	Treb Gain	-79.0 to 24.0 dB
Bass Freq	8 to 25088 Hz	Treb Freq	8 to 25088 Hz

Page 2

Mid1 Gain	-79.0 to 24.0 dB	Mid2 Gain	-79.0 to 24.0 dB
Mid1 Freq	8 to 25088 Hz	Mid2 Freq	8 to 25088 Hz
Mid1 Width	0.010 to 5.000 oct	Mid2 Width	0.010 to 5.000 oct

Page 3

Mid3 Gain	-79.0 to 24.0 dB		
Mid3 Freq	8 to 25088 Hz		
Mid3 Width	0.010 to 5.000 oct		

Parameters for Mn 6 Band EQ

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
Bass Gain	-79.0 to 24.0 dB	Treb Gain	-79.0 to 24.0 dB
Bass Freq	8 to 25088 Hz	Treb Freq	8 to 25088 Hz

Page 2

Mid1 Gain	-79.0 to 24.0 dB	Mid2 Gain	-79.0 to 24.0 dB
Mid1 Freq	8 to 25088 Hz	Mid2 Freq	8 to 25088 Hz
Mid1 Width	0.010 to 5.000 oct	Mid2 Width	0.010 to 5.000 oct

Page 3

Mid3 Gain	-79.0 to 24.0 dB	Mid4 Gain	-79.0 to 24.0 dB
Mid3 Freq	8 to 25088 Hz	Mid4 Freq	8 to 25088 Hz
Mid3 Width	0.010 to 5.000 oct	Mid4 Width	0.010 to 5.000 oct

Parameters for Dual 5 Band EQ

Page 1

L In/Out	In or Out	R In/Out	In or Out
L Out Gain	Off, -79.0 to 24.0 dB	R Out Gain	Off, -79.0 to 24.0 dB
L BassGain	-79.0 to 24.0 dB	R BassGain	-79.0 to 24.0 dB
L BassFreq	8 to 25088 Hz	R BassFreq	8 to 25088 Hz

Page 2

L TrebGain	-79.0 to 24.0 dB	R TrebGain	-79.0 to 24.0 dB
L TrebFreq	8 to 25088 Hz	R TrebFreq	8 to 25088 Hz
LMid1Gain	-79.0 to 24.0 dB	RMid1Gain	-79.0 to 24.0 dB
LMid1Freq	8 to 25088 Hz	RMid1Freq	8 to 25088 Hz
lMid1Width	0.010 to 5.000 oct	RMid1Width	0.010 to 5.000 oct

Page 3

LMid2Gain	-79.0 to 24.0 dB	RMid2Gain	-79.0 to 24.0 dB
LMid2Freq	8 to 25088 Hz	RMid2Freq	8 to 25088 Hz
LMid2Width	0.010 to 5.000 oct	RMid2Width	0.010 to 5.000 oct
LMid3Gain	-79.0 to 24.0 dB	RMid3Gain	-79.0 to 24.0 dB
LMid3Freq	8 to 25088 Hz	RMid3Freq	8 to 25088 Hz
LMid3Width	0.010 to 5.000 oct	RMid3Width	0.010 to 5.000 oct

In/Out When set to **In** the tone controls are active; when set to **Out** the tone controls are bypassed.

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

Bass Gain The amount of boost or cut that the filter should apply to the low frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the bass signal below the specified frequency. Negative values cut the bass signal below the specified frequency.

Bass Freq The center frequency of the bass shelving filter in intervals of one semitone.

Treb Gain The amount of boost or cut that the filter should apply to the high frequency signals in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the treble signal above the specified frequency. Negative values cut the treble signal above the specified frequency.

Treb Freq The center frequency of the treble shelving filter in intervals of one semitone.

Midn Gain The amount of boost or cut that the filter should apply in dB. Every increase of 6 dB approximately doubles the amplitude of the signal. Positive values boost the signal at the specified frequency. Negative values cut the signal at the specified frequency.

Midn Freq The center frequency of the EQ in intervals of one semitone. The boost or cut will be at a maximum at this frequency.

Midn Width The bandwidth of the EQ may be adjusted. You specify the bandwidth in octaves. Small values result in a very narrow filter response. Large values result in a very broad response.

352 Graphic EQ
353 Dual Graphic EQ
670 Mn Graphic EQ

Dual mono 10 band graphic equalizer

PAUs: 3

The graphic equalizer is available as stereo (linked parameters for left and right) or dual mono (independent controls for left and right). The graphic equalizer has ten bandpass filters per channel. For each band the gain may be adjusted from **-12 dB** to **+24 dB**. The frequency response of all the bands is shown in Figure 102 below. The dual graphic equalizer has a separate set of controls for the two mono channels.

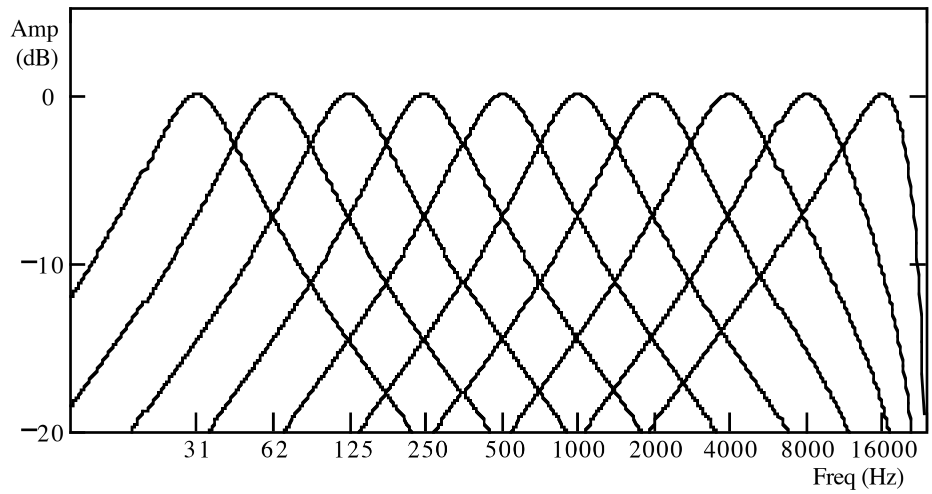


Figure 102 Filter Response of each Bandpass Filter

Like all graphic equalizers, the filter response is not perfectly flat when all gains are set to the same level (except at **0 dB**), but rather has ripple from band to band (see **Figure 103**). To minimize the EQ ripple, you should attempt to center the overall settings around **0 dB**.

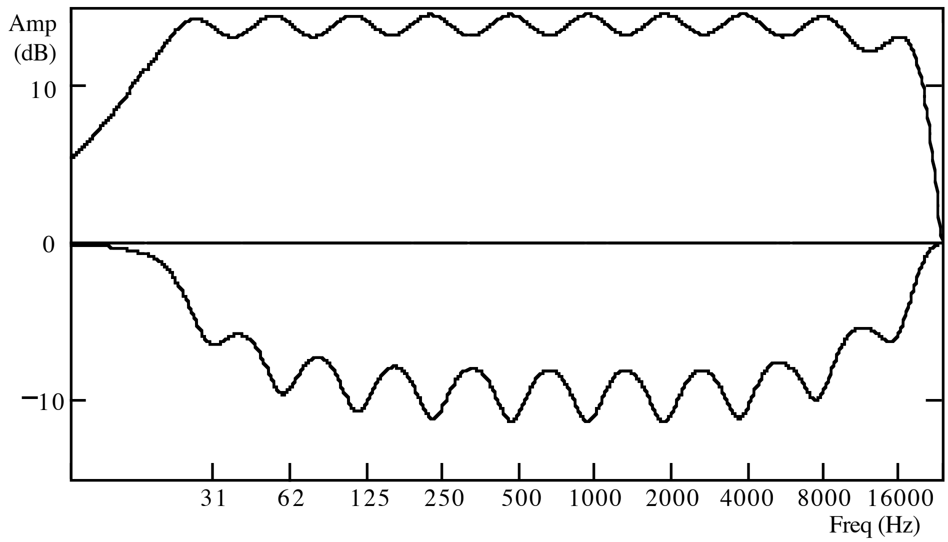


Figure 103 **Overall response with all gains set to +12 dB, 0 dB and -6 dB**

Parameters for Graphic EQ

Page 1

In/Out	In or Out		

Page 2

31Hz G	-12.0 to 24.0 dB	1000Hz G	-12.0 to 24.0 dB
62Hz G	-12.0 to 24.0 dB	2000Hz G	-12.0 to 24.0 dB
125Hz G	-12.0 to 24.0 dB	4000Hz G	-12.0 to 24.0 dB
250Hz G	-12.0 to 24.0 dB	8000Hz G	-12.0 to 24.0 dB
500Hz G	-12.0 to 24.0 dB	16000Hz G	-12.0 to 24.0 dB

Parameters for Dual Graphic EQ

Page 1

L In/Out	In or Out	R In/Out	In or Out

Page 2

L 31Hz G	-12.0 to 24.0 dB	L 1000Hz G	-12.0 to 24.0 dB
L 62Hz G	-12.0 to 24.0 dB	L 2000Hz G	-12.0 to 24.0 dB
L 125Hz G	-12.0 to 24.0 dB	L 4000Hz G	-12.0 to 24.0 dB
L 250Hz G	-12.0 to 24.0 dB	L 8000Hz G	-12.0 to 24.0 dB
L 500Hz G	-12.0 to 24.0 dB	L16000Hz G	-12.0 to 24.0 dB

Page 3

R 31Hz G	-12.0 to 24.0 dB	R 1000Hz G	-12.0 to 24.0 dB
R 62Hz G	-12.0 to 24.0 dB	R 2000Hz G	-12.0 to 24.0 dB
R 125Hz G	-12.0 to 24.0 dB	R 4000Hz G	-12.0 to 24.0 dB
R 250Hz G	-12.0 to 24.0 dB	R 8000Hz G	-12.0 to 24.0 dB
R 500Hz G	-12.0 to 24.0 dB	R16000Hz G	-12.0 to 24.0 dB

In/Out When set to **In** the left channel equalizer is active; when set to **Out** the left channel equalizer is bypassed.

31Hz G Gain of the left 31 Hz band in dB.

62Hz G Gain of the left 62 Hz band in dB.

125Hz G Gain of the left 125 Hz band in dB.

250Hz G Gain of the left 250 Hz band in dB.

500Hz G Gain of the left 500 Hz band in dB.

1000Hz G Gain of the left 1000 Hz band in dB.

2000Hz G Gain of the left 2000 Hz band in dB.

4000Hz G Gain of the left 4000 Hz band in dB.

8000Hz G Gain of the left 8000 Hz band in dB.

16000Hz G Gain of the left 16000 Hz band in dB.

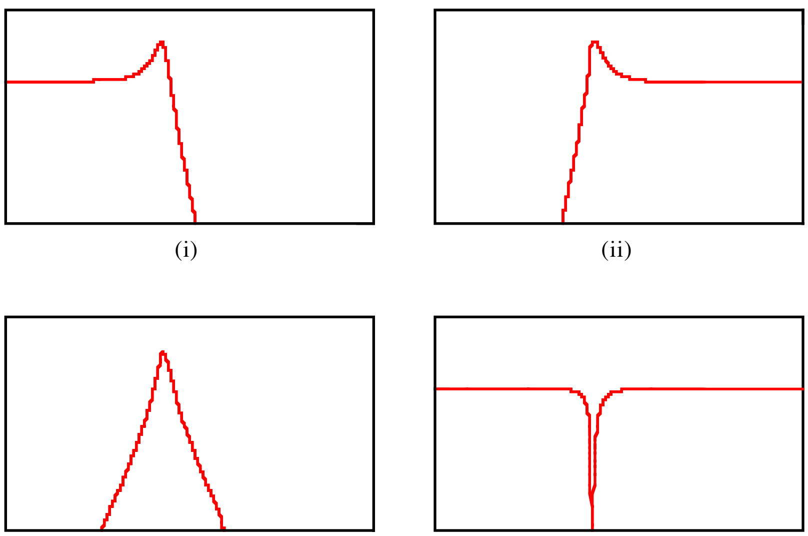
Miscellaneous Filters

360 Env Follow Filt
675 Mn Env Filter

Envelope following stereo two-pole resonant filter

PAUs: 2

The envelope following filter is a stereo resonant filter with the resonant frequency controlled by the envelope of the input signal (the maximum of left or right). The filter type is selectable and may be **lowpass**, **highpass**, **bandpass**, or **notch**.



(i) (ii)

(iii) (iv)

Figure 104 Resonant filters: (i) lowpass; (ii) highpass; (iii) bandpass; (iv) notch

The resonant frequency of the filter will remain at the minimum frequency (Min Freq) as long as the signal envelope is below the Threshold. The Freq Sweep parameter controls how much the frequency will change with changes in envelope amplitude. The frequency range is 0 to 8372 Hz, though the minimum setting for Min Freq is **8 Hz**. Note that the term minimum frequency is a reference to the resonant frequency at the minimum envelope level; with a negative Freq Sweep, the filter frequency will sweep below the Min Freq. A meter is provided to show the current resonance frequency of the filter.

The filter Resonance level may be adjusted. The resonance is expressed in decibels (dB) of gain at the resonant frequency. Since 50 dB of gain is available, you will have to be careful with your gain stages to avoid clipping.

The attack and release rates of the envelope follower are adjustable. The rates are expressed in decibels per second (dB/s). The envelope may be smoothed by a lowpass filter which can extend the attack and release times of the envelope follower. A level meter with a threshold marker is provided.

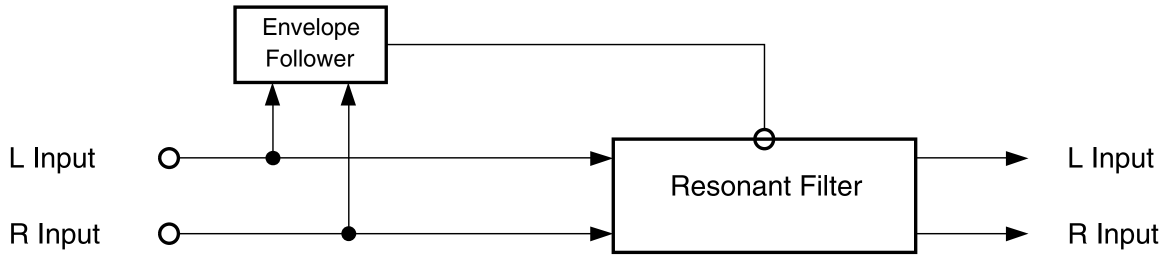


Figure 105 Block diagram of envelope following filter

Parameters

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
FilterType	Lowpass	Min Freq	8 to 8372 Hz
F		Freq Sweep	-100 to 100%
0Hz 2k 4k 6k		Resonance	0 to 50 dB

Page 2

Threshold	-79.0 to 0.0 dB	Atk Rate	0.0 to 300.0 dB/s
		Rel Rate	0.0 to 300.0 dB/s
		Smth Rate	0.0 to 300.0 dB/s

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

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

FilterType The type of resonant filter to be used. **Lowpass**, **Highpass**, **Bandpass**, or **Notch**.

Min Freq The base frequency of the resonant filter. The filter resonant frequency is set to the Min Freq while the signal envelope is at its minimum level or below the threshold.

Freq Sweep How far the filter frequency can change from the Min Freq setting as the envelope amplitude changes. Freq Sweep may be positive or negative so the filter frequency can rise above or fall below the Min Freq setting.

Resonance The resonance level of the resonant filter. Resonance sets the level of the resonant peak (or the amount of cut in the case of the notch filter).

Threshold The level above which signal envelope must rise before the filter begins to follow the envelope. Below the threshold, the filter resonant frequency remains at the Min frequency.

Atk Rate Adjusts the upward slew rate of the envelope detector.

Rel Rate Adjusts the downward slew rate of the envelope detector.

Smth Rate Smooths the output of the envelope follower. Smoothing slows down the envelope follower and can dominate the attack and release rates if set to a lower rate than either of these parameters.

361 TrigEnvelopeFilt
676 Mn Trig Env Filt

Triggered envelope following stereo two-pole resonant filter

PAUs: 2

The triggered envelope following filter is used to produce a filter sweep when the input rises above a trigger level. The triggered envelope following filter is a stereo resonant filter with the resonant frequency controlled by a triggered envelope follower. The filter type is selectable and may be **lowpass**, **highpass**, **bandpass**, or **notch**.

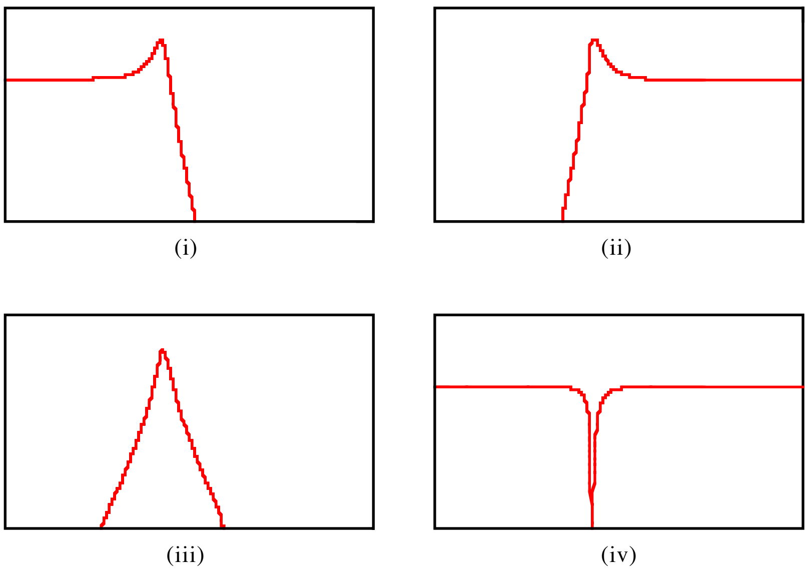


Figure 106 Resonant Filters: (i) lowpass; (ii) highpass; (iii) bandpass; (iv) notch

The resonant frequency of the filter will remain at the minimum frequency (Min Freq) prior to being triggered. On a trigger, the resonant frequency will sweep to the maximum frequency (Max Freq). The minimum and maximum frequencies may be set to any combination of frequencies between 58 and 8372 Hz. Note that the terms minimum and maximum frequency are a reference to the resonant frequencies at the minimum and maximum envelope levels; you may set either of the frequencies to be larger than the other. A meter is provided to show the current resonance frequency of the filter.

The filter Resonance level may be adjusted. The resonance is expressed in decibels (dB) of gain at the resonant frequency. Since 50 dB of gain is available, you will have to be careful with your gain stages to avoid clipping.

When the input signal envelope rises above the trigger level, an envelope generator is started which has an instant attack and exponential decay. The generated attack may be lengthened with the smoothing parameter. The smoothing parameter can also lengthen the generated decay if the smoothing rate is lower than the decay. The generated envelope is then used to control the resonant frequency of the filter.

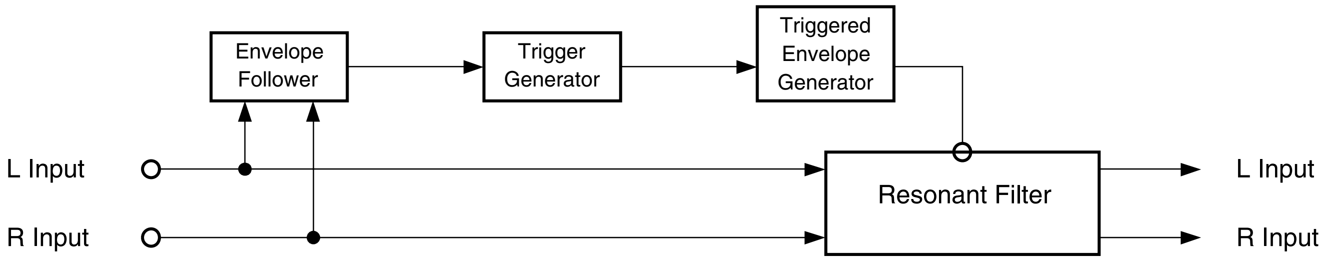


Figure 107 Block diagram of triggered envelope filter

The time constant of the envelope follower may be set (Env Rate) as well as the decay rate of the generated envelope (Rel Rate). After the detected envelope rises above the Trigger level, a trigger event cannot occur again until the signal drops below the Retrigger level. In general, Retrigger should be set lower than the Trigger level. A level meter with a trigger marker is provided.

Parameters

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
FilterType	Lowpass	Min Freq	8 to 8372 Hz
F		Max Freq	8 to 8372 Hz
0Hz 2k 4k 6k		Resonance	0 to 50 dB

Page 2

Trigger	-79.0 to 0.0 dB	Env Rate	0.0 to 300.0 dB/s
Retrigger	-79.0 to 0.0 dB	Rel Rate	0.0 to 300.0 dB/s
		Smth Rate	0.0 to 300.0 dB/s

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

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

FilterType The type of resonant filter to be used. May be one of **Lowpass**, **Highpass**, **Bandpass**, or **Notch**.

Min Freq The base frequency of the resonant filter. The filter resonant frequency is set to the base frequency while the signal envelope is below the threshold.

Max Freq The frequency of the resonant filter that can be reached when the envelope follower output reaches full-scale. The resonant frequency will sweep with the envelope from the base frequency, approaching the limit frequency with rising amplitudes.

Resonance The resonance level of the resonant filter. Resonance sets the level of the resonant peak (or the amount of cut in the case of the notch filter).

Trigger The threshold at which the envelope detector triggers in fractions of full scale where 0dB is full scale.

Retrigger The threshold at which the envelope detector resets such that it can trigger again in fractions of full scale where 0dB is full scale. This value is useful only when it is below the value of Trigger.

Env Rate The envelope detector decay rate which can be used to prevent false triggering. When the signal envelope falls below the retrigger level, the filter can be triggered again when the signal rises above the trigger level. Since the input signal can fluctuate rapidly, it is necessary to adjust the rate at which the signal envelope can fall to the retrigger level. The rate is provided in decibels per second (dB/s).

Rel Rate The downward slew rate of the triggered envelope generator. The rate is provided in decibels per second (dB/s).

Smth Rate Smooths the output of the envelope generator. Smoothing slows down the envelope follower and can dominate the release rate if set lower rate than this parameter. You can use the smoothing rate to lengthen the attack of the generated envelope which would otherwise have an instant attack. The rate is provided in decibels per second (dB/s).

362 LFO Sweep Filter
677 Mn LFOSweepFilt

LFO following stereo two-pole resonant filter

PAUs: 2

The LFO following filter is a stereo resonant filter with the resonant frequency controlled by an LFO (low-frequency oscillator). The filter type is selectable and may be **lowpass**, **highpass**, **bandpass**, or **notch**.

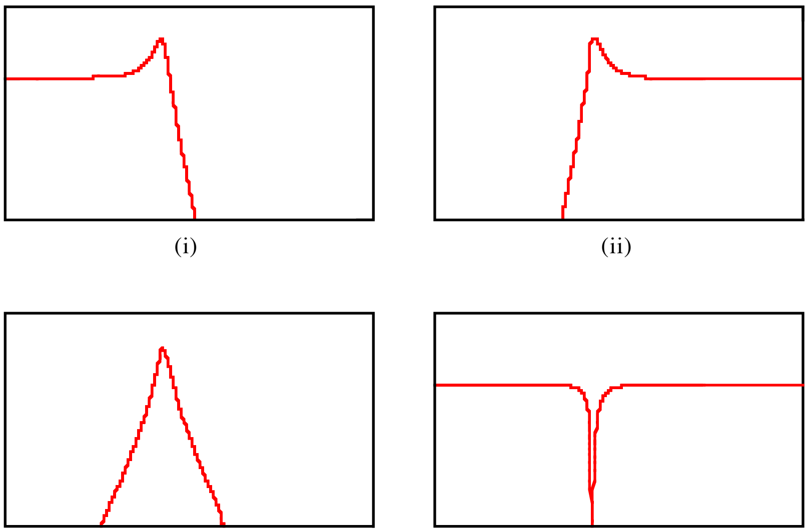


Figure 108 Resonant Filters: (i) lowpass; (ii) highpass; (iii) bandpass; (iv) notch

The resonant frequency of the filter will sweep between the minimum frequency (Min Freq) and the maximum frequency (Max Freq). The minimum and maximum frequencies may be set to any combination of frequencies between **8** and **8372 Hz**. Note that the terms minimum and maximum frequency are a reference to the resonant frequencies at the minimum and maximum envelope levels; you may set either of the frequencies to be larger than the other, though doing so will just invert the direction of the LFO. Meters are provided to show the current resonance frequencies of the left and right channel filters.

The filter Resonance level may be adjusted. The resonance is expressed in decibels (dB) of gain at the resonant frequency. Since 50 dB of gain is available, you will have to be careful with your gain stages to avoid clipping.

You can set the frequency of the LFO using the LFO Tempo and LFO Period controls. You can explicitly set the tempo or use the system tempo from the sequencer (or MIDI clock). The LFO Period control sets the period of the LFO (the time for one complete oscillation) in terms of the number of tempo beats per LFO period. The LFO may be configured to one of a variety of wave shapes. Available shapes are **Sine**, **Saw+**, **Saw–**, **Pulse** and **Tri** (Figure 109). **Sine** is simply a sinusoid waveform. **Tri** produces a triangular waveform, and **Pulse** produces a series of square pulses where the pulse width can be adjusted with the LFO PlsWid parameter. When pulse width is **50**%, the signal is a square wave. The LFO PlsWid parameter is active only when the **Pulse** waveform is selected. **Saw+** and **Saw–** produce rising and falling sawtooth waveforms. The **Pulse** and **Saw** waveforms have abrupt, discontinuous changes in amplitude which can be smoothed. The pulse wave is implemented as a hard clipped sine wave, and, at **50**% width, it turns into a sine wave when set to **100**% smoothing. The sudden change in amplitude of the sawtooth waves develops a more gradual slope with smoothing, ending up as triangle waves when set to **100**% smoothing.

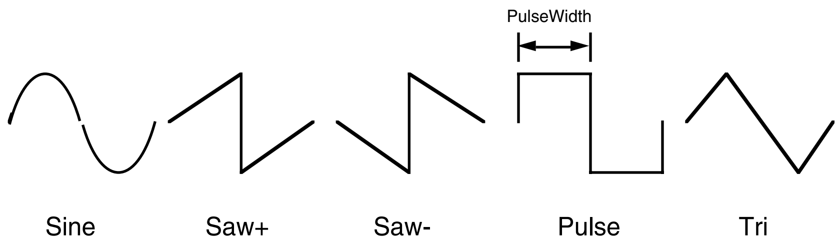


Figure 109 Configurable wave shapes

Parameters

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
LFO Tempo	System, 1 to 255 BPM	LFO Shape	Sine
LFO Period	1/24 to 32 bts	LFO PlsWid	0 to 100%
		LFO Smooth	0 to 100%

Page 2

FilterType	Lowpass	Min Freq	8 to 8372 Hz
		Max Freq	8 to 8372 Hz
		Resonance	0 to 50 dB
L Phase	0.0 to 360.0 deg	R Phase	0.0 to 360.0 deg
L		R	
0Hz 2k 4k 6k		0Hz 2k 4k 6k	

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

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

LFO Tempo Basis for the rates of the LFO, 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. In this case, FXMods (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).

LFO Shape The waveform type for the LFO. Choices are **Sine**, **Saw+**, **Saw–**, **Pulse**, and **Tri**.

LFO PlsWid When the LFO Shape is set to **Pulse**, the PlsWid parameter sets the pulse width as a percentage of the waveform period. The pulse is a square wave when the width is set to **50**%. This parameter is active only when the **Pulse** waveform is selected.

LFO Smooth Smooths the Saw+, Saw–, and Saw– waveforms. For the sawtooth waves, smoothing makes the waveform more like a triangle wave. For the Pulse wave, smoothing makes the waveform more like a sine wave.

FilterType The type of resonant filter to be used. May be one of **Lowpass**, **Highpass**, **Bandpass**, or **Notch**.

Min Freq The minimum frequency of the resonant filter. This is the resonant frequency at one of the extremes of the LFO sweep. The resonant filter frequency will sweep between the Min Freq and Max Freq.

Max Freq The maximum frequency of the resonant filter. This is resonant frequency at the other extreme of the LFO sweep. The resonant filter frequency will sweep between the Min Freq and Max Freq.

Resonance The resonance level of the resonant filter. Resonance sets the level of the resonant peak (or the amount of cut in the case of the notch filter).

L Phase The phase angle of the left channel LFO relative to the system tempo clock and the right channel phase.

R Phase The phase angle of the right channel LFO relative to the system tempo clock and the left channel phase.

363 Resonant Filter
364 Dual Res Filter
678 Mn Res Filter

Stereo and dual mono 2 pole resonant filters

PAUs: 1

The resonant filter is available as a stereo (linked parameters for left and right) or dual mono (independent controls for left and right). The filter type is selectable and may be **lowpass**, **highpass**, **bandpass**, or **notch**.

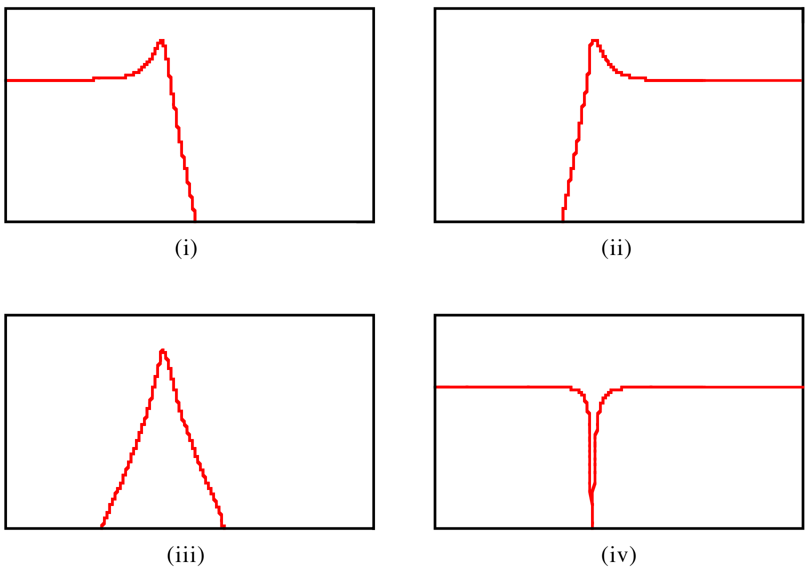


Figure 110 Resonant filters: (i) lowpass; (ii) highpass; (iii) bandpass; (iv) notch

You can adjust the resonant frequency of the filter and the filter resonance level.

Parameters for Resonant Filter

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
FilterType	Lowpass		
Frequency	58 to 8372 Hz		
Resonance	0 to 50 dB		

Parameters for Dual Res Filter

Page 1

L Wet/Dry	0 to 100%wet	R Wet/Dry	0 to 100%wet
L Output	Off, -79.0 to 24.0 dB	R Output	Off, -79.0 to 24.0 dB

Page 2

L FiltType	Lowpass	R FiltType	Highpass
L Freq	58 to 8372 Hz	R Freq	58 to 8372 Hz
LResonance	0 to 50 dB	RResonance	0 to 50 dB

Wet/Dry The amount of filtered (wet) signal relative to unaffected (dry) signal.

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

FilterType The type of resonant filter to be used. May be one of **Lowpass**, **Highpass**, **Bandpass**, or **Notch**.

Frequency The frequency of the resonant filter peak (or notch) in Hz. The frequencies correspond to semitone increments.

Resonance The resonance level of the resonant filter. Resonance sets the level of the resonant peak (or the amount of cut in the case of the notch filter).

365 EQ Morpher
366 Mono EQ Morpher
679 Mn EQ Morpher

Parallel resonant bandpass filters with parameter morphing

PAUs: 2 for **Mono EQ Morpher**
4 for **EQ Morpher**

The **EQ Morpher** algorithms have four parallel bandpass filters acting on the input signal and the filter results are summed for the final output. **EQ Morpher** is a stereo algorithm for which the left and right channels receive separate processing using the same linked controls. **Mono EQ Morpher** sums the input left and right channels into a mono signal, so there is only one channel of processing. Both algorithms have output panning. In **EQ Morpher**, a stereo panner like that on the INPUT page is used and includes a width parameter to control the width of the stereo field. **Mono EQ Morpher** uses a standard mono panner for positioning the mono signal between the left and right speakers.

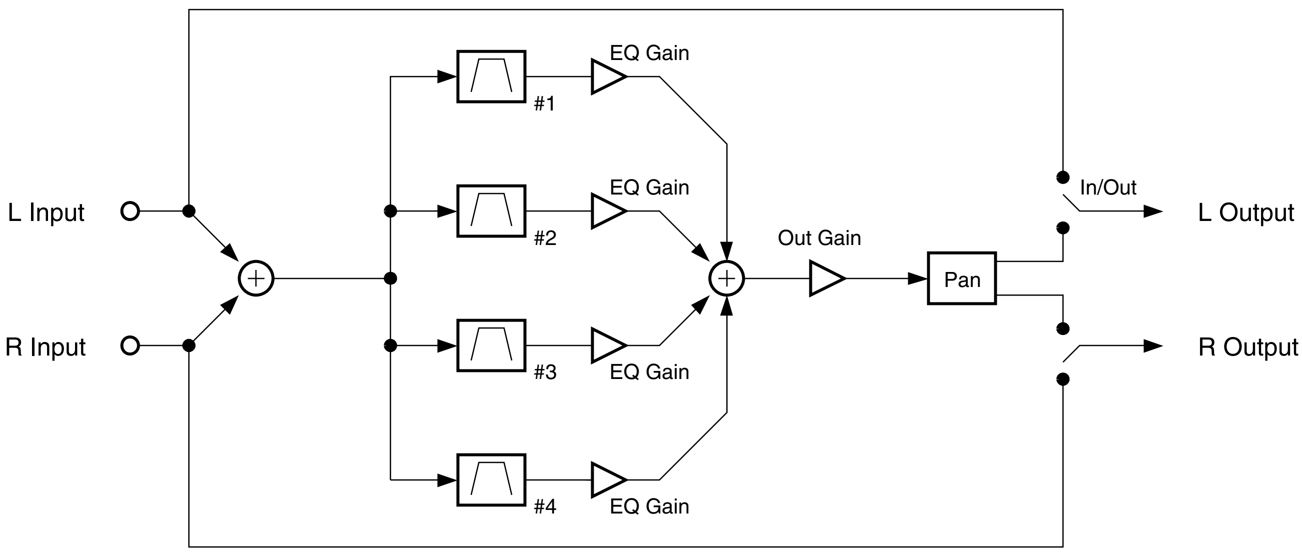


Figure 111 Mono EQ Morpher (EQ Morpher is similar)

For each filter, there are two sets of parameters, A and B. The parameter Morph A>B determines which parameter set is active. When Morph A>B is set to **0**%, you are hearing the A parameters; when set to **100**%, you are hearing the B parameters. The filters may be gradually moved from A to B and back again by moving the Morph A>B parameter between **0** and **100**%.

The four filters are parametric bandpass filters. These are not standard parametric filters, which boost or cut the signal at the frequency you specify relative to the signal at other frequencies. The bandpass filters used here pass only signals at the frequency you specify and cut all other frequencies. The gain controls for the filters set the levels of each filter's output. Like the normal parametric filters, you have control of the filters' frequencies and bandwidths. The Freq Scale parameters may be used to adjust the A or B filters' frequencies as a group. This allows you to maintain a constant spectral relationship between your filters while adjusting the frequencies up and down. The filters are arranged in parallel and their outputs summed, so the bandpass peaks are added together and the multiple resonances are audible.

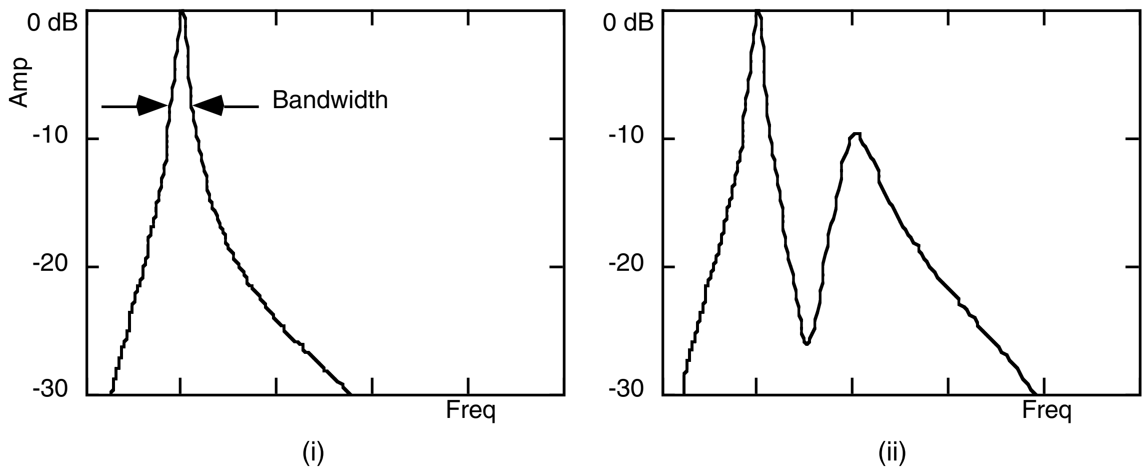


Figure 112 Frequency response of (i) single bandpass filter; (ii) sum of two bandpass filters

EQ Morpher can do an excellent job of simulating the resonances of the vocal tract. A buzz or sawtooth signal is a good choice of source material to experiment with the **EQ Morpher**. Set the Morph A>B parameter to **0%**, and find a combination of A filter settings which give an interesting vowel like sound. It may help to start from existing ROM presets. Next set Morph A>B to **100%** and set the B parameters to a different vowel-like sound. You can now set up some FXMods on Morph A>B to morph between the two sets of parameters, perhaps using Freq Scale to make it more expressive.

When morphing from the A parameters to the B parameters, A Filter 1 moves to B Filter 1, A Filter 2 moves to B Filter 2, and so on. For the most normal and predictable results, it's a good idea not to let the frequencies of the filters cross each other during the morphing. You can ensure this doesn't happen by making sure the four filters are arranged in ascending order of frequencies. Descending order is okay too, provided you choose an order and stick to it.

Parameters

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
Morph A>B	0 to 100%	Out Pan	-100 to 100%
		Out Width (**EQ Morpher** only)	-100 to 100%
AFreqScale	-8600 to 8600 ct	BFreqScale	-8600 to 8600 ct

Page 2

A Freq 1	8 to 25088 Hz	B Freq 1	8 to 25088 Hz
A Width 1	0.010 to 5.000 oct	B Width 1	0.010 to 5.000 oct
A Gain 1	-79.0 to 24.0 dB	B Gain 1	-79.0 to 24.0 dB
A Freq 2	8 to 25088 Hz	B Freq 2	8 to 25088 Hz
A Width 2	0.010 to 5.000 oct	B Width 2	0.010 to 5.000 oct
A Gain 2	-79.0 to 24.0 dB	B Gain 2	-79.0 to 24.0 dB

Page 3

A Freq 3	8 to 25088 Hz	B Freq 3	8 to 25088 Hz
A Width 3	0.010 to 5.000 oct	B Width 3	0.010 to 5.000 oct
A Gain 3	-79.0 to 24.0 dB	B Gain 3	-79.0 to 24.0 dB
A Freq 4	8 to 25088 Hz	B Freq 4	8 to 25088 Hz
A Width 4	0.010 to 5.000 oct	B Width 4	0.010 to 5.000 oct
A Gain 4	-79.0 to 24.0 dB	B Gain 4	-79.0 to 24.0 dB

In/Out When set to **In** the algorithm is active; when set to **Out** the algorithm is bypassed.

Out Gain An overall level control of the **EQ Morpher** output.

Out Pan Provides panning of the output signal between left and right output channels. A setting of **-100%** is panned left and **100%** is panned right. For **EQ Morpher**, this is a stereo panner which pans the entire stereo image as is done with the input sends on the INPUT page when set to the SP mode.

Out Width The width of the stereo field is controlled by this parameter. A setting of **100%** is the same full width as the input signal. At **0%** the left and right channels are narrowed to the point of being mono. Negative values reverse the left and right channels. This parameter is available in **EQ Morpher** but not in **Mono EQ Morpher**.

Morph A>B When set to **0%** the A parameters are controlling the filters, and when set to **100%**, the B parameters control the filters. Between **0** and **100%**, the filters are at interpolated positions. When morphing from the A to the B settings, A Filter 1 moves to B Filter 1, A Filter 2 moves to B Filter 2, and so on.

FreqScale The filter frequencies for the A and B parameter sets may be offset with the FreqScale parameters. After setting the filter parameters, the FreqScale parameters will move each of the four filter frequencies together by the same relative pitch.

Filter Parameters

For the two filter sets A and B, there are four filters 1, 2, 3 and 4:

Freq The center frequency of the bandpass filter peak in Hz. This frequency may be offset by the FreqScale parameter.

Width The bandwidth of the bandpass filter in octaves. Narrow bandwidths provide the most convincing vocal sounds.

Gain The level of the bandpass filter output. At **0 dB**, a sine wave at the same frequency as the filter will be neither boost not cut. At settings greater than **0 dB**, the (hypothetical) sine wave is boosted, and below **0 dB** the sine wave is cut. Signals at frequencies other than the filter frequency are always cut more than a signal at the filter frequency. The amount that other frequencies are cut depends on the bandwidth of the bandpass filter.

Enhancers, Suppressors, and Modulators

370 2 Band Enhancer

Two-band spectral modifier

PAUs: 1

The **2 Band Enhancer** modifies the spectral content of the input signal primarily by brightening signals with little or no high frequency content, and boosting pre-existing bass energy. First, the input is non-destructively split into 2 frequency bands using 6 dB/oct highpass and lowpass filters. The highpassed band is processed to add additional high frequency content by using a nonlinear transfer function in combination with a high shelving filter. Each band can then be separately delayed to sample accuracy and mixed back together in varying amounts. One sample of delay is approximately equivalent to 20 microseconds, or 180 degrees of phase shift at 24 khz. Using what we know about psychoacoustics, phase shifting, or delaying certain frequency bands relative to others can have useful affects without adding any gain. In this algorithm, delaying the lowpassed signal relative to the highpass signal brings out the high frequency transient of the input signal giving it more definition. Conversely, delaying the highpass signal relative to the lowpass signal brings out the low frequency transient information which can provide punch.

The transfer applied to the highpass signal can be used to generate additional high frequency content when set to a non-zero value. As the value is scrolled away from **0**, harmonic content is added in increasing amounts to brighten the signal. In addition to adding harmonics, positive values impose a dynamically compressed quality, while negative values sound dynamically expanded. This type of compression can bring out frequencies in a particular band even more. The expanding quality is particularly useful when trying to restore transient information.

Parameters

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
CrossOver	17 to 25088 Hz		

Page 2

Hi Drive	Off, -79.0 to 24.0 dB		
Hi Xfer	-100 to 100 %		
Hi Shelf F	8 to 25088 Hz		
Hi Shelf G	-96 to 24 dB		
Hi Delay	0 to 500 samp	Lo Delay	0 to 500 samp
Hi Mix	Off, -79.0 to 24.0 dB	Lo Mix	Off, -79.0 to 24.0 dB

In/Out When set to **In** the effect is active; when set to **Out** the effect is bypassed.

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

CrossOver Adjusts the -6dB crossover point at which the input signal will be divided into the highpass band and a lowpass bands.

Hi Drive Adjusts the gain into the transfer function. The affect of the transfer can be intensified or reduced by respectively increasing or decreasing this value.

Hi Xfer The intensity of the transfer function.

Hi Shelf F The frequency of where the high shelving filter starts to boost or attenuate.

Hi Shelf G The boost or cut of the high shelving filter.

Hi Delay Adjusts the number of samples the highpass signal is delayed.

Hi Mix Adjusts the output gain of the highpass signal.

Lo Delay Adjusts the number of samples the lowpass signal is delayed.

Lo Mix Adjusts the output gain of the lowpass signal.

371 3 Band Enhancer
672 Mn 3BandEnhancer

Three-band spectral modifier

PAUs: 2

The **3 Band Enhancer** modifies the spectral content of the input signal by boosting existing spectral content, or stimulating new ones. First, the input is non-destructively split into 3 frequency bands using 6 dB/oct highpass and lowpass filters (Figure 113). The high and mid bands are separately processed to add additional high frequency content by using two nonlinear transfer functions. The low band is processed by a single nonlinear transfer to enhance low frequency energy. Each band can also be separately delayed to sample accuracy and mixed back together in varying amounts. One sample of delay is approximately equivalent to 20 microseconds, or 180 degrees of phase shift with the KDFX 24 khz sampling rate. Using what we know about psychoacoustics, phase shifting, or delaying certain frequency bands relative to others can have useful affects without adding any gain. In this algorithm, delaying the lower bands relative to higher bands brings out the high frequency transient of the input signal giving it more definition. Conversely, delaying the higher bands relative to the lower bands brings out the low frequency transient information which can provide punch.

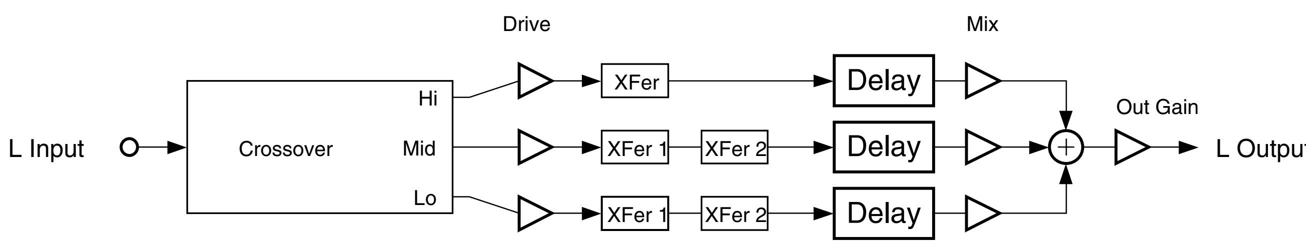


Figure 113 One channel of 3 Band Enhancer

The nonlinear transfers applied to the high and mid bands can be used to generate additional high and mid frequency content when Xfer1 and Xfer2 are set to non-zero values. As the value is scrolled away from **0**, harmonic content is added in increasing amounts. In addition, setting both positive or negative will respectively impose a dynamically compressed or expanded quality. This type of compression can bring out frequencies in a particular band even more. The expanding quality is useful when trying to restore transient information. More complex dynamic control can be obtained by setting these independent of each other. Setting one positive and the other negative can even reduce the noise floor in some applications.

The low band has a nonlinear transfer that requires only one parameter. Its affect is controlled similarly.

Parameters

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
CrossOver1	17 to 25088 Hz		
CrossOver2	17 to 25088 Hz		

Page 2

Lo Enable	On or Off	Mid Enable	On or Off
Lo Drive	Off, -79.0 to 24.0 dB	Mid Drive	Off, -79.0 to 24.0 dB
Lo Xfer	-100 to 100 %	Mid Xfer1	-100 to 100 %
		Mid Xfer2	-100 to 100 %
Lo Delay	0 to 1000 samp	Mid Delay	0 to 500 samp
Lo Mix	Off, -79.0 to 24.0 dB	Mid Mix	Off, -79.0 to 24.0 dB

Page 3

Hi Enable	On or Off		
Hi Drive	Off, -79.0 to 24.0 dB		
Hi Xfer1	-100 to 100 %		
Hi Xfer2	-100 to 100 %		
Hi Delay	0 to 500 samp		
Hi Mix	Off, -79.0 to 24.0 dB		

In/Out When set to **In** the effect is active; when set to **Out** the effect is bypassed.

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

CrossOver1 Adjusts one of the -6dB crossover points at which the input signal will be divided into the high, mid and low bands.

CrossOver2 Adjusts the other -6dB crossover points at which the input signal will be divided into the high, mid and low bands.

Enable Low, Mid, and High. Turns processing for each band on or off. Turning each of the 3 bands off results in a dry output signal.

Drive Low. Mid, and High. Adjusts the input into each transfer. Increasing the drive will increase the effects.

Xfer Low, Mid, and High; Xfer1 and Xfer2 for Mid and High. Adjusts the intensity of the transfer curves.

Delay Low, Mid, and High. Adjusts the number of samples the each signal is delayed.

Mix Low, Mid, and High. Adjusts the output gain of each band.

372 HF Stimulate 1
373 HF Stimulate 3
673 Mn HF Stimulate1
674 Mn HF Stimulate2

High-frequency stimulators

PAUs: 1 for **HF Stimulate 1** and **Mn HF Stimulate1**
2 for **Mn HF Stimulate2**
3 for **HF Stimulate 3**

The high-frequency stimulator algorithms are stereo and mono algorithms closely based on the V.A.S.T. High Frequency Stimulator DSP function, and the manual description is repeated here (edited for KDFX specifics).

The overall effect of a high-frequency stimulator is to boost the high frequency partials of the signal, and depending on the settings of the parameters, it can add high-frequency partials to the signal as well. It's useful for building sounds that cut through the mix, and have a bright crisp nature.

There's more to the high-frequency stimulator than meets the eye. It works like this: the signal is run through a highpass filter, then through a distortion function, then through a second highpass filter. Finally. it's mixed with the original signal after passing through the final Stim Gain level control of the algorithm.

The **HF Stimulate 1** algorithm is a close copy of the V.A.S.T. DSP function, giving control of the first highpass filter frequency, the distortion drive and the amplitude of the result (Stim Gain). As a bonus, the distortion curve can also be adjusted.

HF Stimulate 3 is also very similar, but uses a more sophisticated distortion routine (same as Algorithm **300 Mono Distortion** for example). You also have control of the second highpass filter.

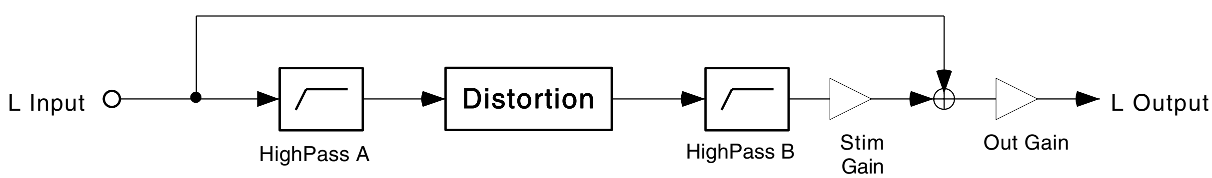


Figure 114 One channel of high-frequency stimulation

Parameters:

Page 1 (HF Stimulate 3 and Mn HF Stimulate2)

Stim Gain	Off, -79.0 to 24.0 dB	Out Gain	Off, -79.0 to 24.0 dB
Dist Drive	0 to 96 dB	Highpass A	8 to 25088 Hz
Warmth	8 to 25088 Hz	Highpass B	8 to 25088 Hz

Page 1 (HF Stimulate 1 and Mn HF Stimulate1)

Stim Gain	Off, -79.0 to 24.0 dB	Out Gain	Off, -79.0 to 24.0 dB
Dist Drive	-79.0 to 48.0 dB	Highpass	8 to 25088 Hz
Dist Curve	0 to 127%		

Stim Gain The gain of the high frequency stimulated signal applied prior to being added to the original input signal.

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

Dist Drive The amount to boost (or cut) the signal level to drive the distortion. Higher values will increase the distortion of high frequency signal components.

Warmth A lowpass filter in the distortion control path. This filter may be used to reduce some of the harshness of some distortion settings without reducing the bandwidth of the signal. (Used in **HF Stimulate 3** and **Mn HF Stimulate2**.)

Curve The curvature of the distortion. **0**% is no curvature (no distortion at all). At **100**%, the curve bends over smoothly and becomes perfectly flat right before it goes into digital clipping. (Used in **Mn HF Stimulate1** and **Mn HF Stimulate1**.)

Highpass A A first order highpass filter that removes low frequencies prior to being distorted.

Highpass B A first order highpass filter that removes low frequencies after being distorted. (Used in **HF Stimulate 3** and **Mn HF Stimulate2**.)

374 HarmonicSuppress
375 Tone Suppressor

Stereo algorithms to expand a single frequency band or harmonic bands.

PAUs: 2

Tone Suppressor and **HarmonicSuppress** are special expander algorithms. In most respects they are identical to Algorithm **340 Expander**. However, **Tone Suppressor** expands only on a single band of frequencies. **HarmonicSuppress** expands on only harmonically related bands of frequencies. Why would we do this? Imagine you are working with a sampled recording which contains an obnoxious 60 Hz hum. You can suppress just the 60 Hz component using **Tone Suppressor**. Set the threshold above the level of the 60 Hz tone so that real sounds in the 60 Hz range above threshold pass through the expander while 60 Hz hum below threshold get squashed. If the 60 Hz is more of a hum or buzz than a tone, then distortion harmonics are present. In this case, you might have more success using **HarmonicSuppress**.

Frequency band selection in **Tone Suppressor** is based on a parametric filter. You control the filter center frequency and bandwidth. The expander controls the filter gain. **HarmonicSuppress** is based on comb filtering—a simple filter which removes harmonically related frequency bands with a spectrum which looks like a comb. With the Harmonics parameter, you can choose to expand the odd harmonics (including the fundamental) or even harmonics (not including the fundamental) or all harmonics. (Choosing all harmonics is the same as choosing even harmonics at half the frequency.)

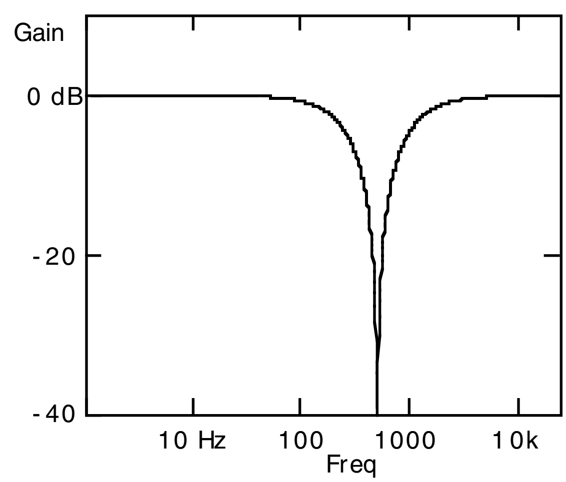


Figure 115 Tone Suppressor filtering at full expansion

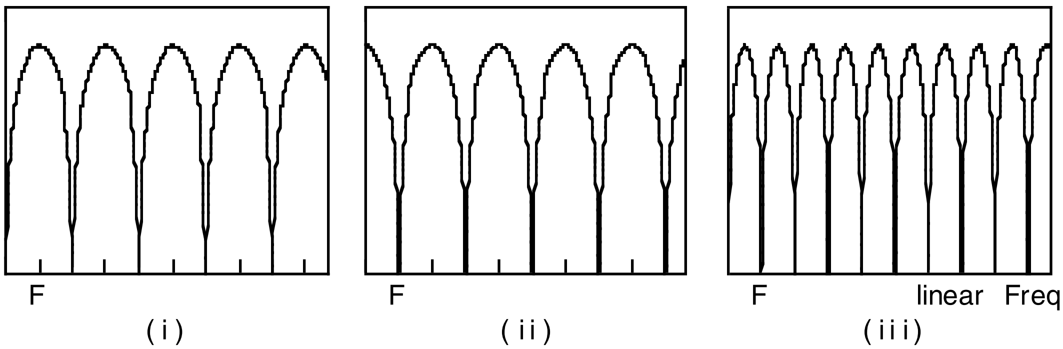


Figure 116 **HarmonicSuppress filtering at full expansion**
F marks fundamental
Harmonics are Even (i), Odd (ii), All (iii)

The algorithms expand the signal in the specified band(s) (reduce the signal's gain) when the signal falls below the expansion threshold in the specified band(s). You can select which channel, left (**L**), right (**R**) or the larger of the two (**L & R**) is used to control the expansion (side chain processing) with the SC Input parameter. You can also select which channel is actually expanded, again left (**L**), right (**R**) or both (**L & R**) using the ExpandChan parameter. The amount of expansion is expressed as an expansion ratio. Expanding a signal reduces its level below the threshold. The expansion ratio is the inverse of the slope of the expander input/output characteristic. An expansion ratio of **1:1** will have no effect on the signal. A zero ratio (**1:∞**), will expand all signal levels below the threshold level to the null or zero level. (This expander expands to **1:17** at most, but that's a lot.) Thresholds are expressed as a decibel level relative to digital full-scale (dBFS) where **0 dBFS** is digital full-scale and all other available values are negative.

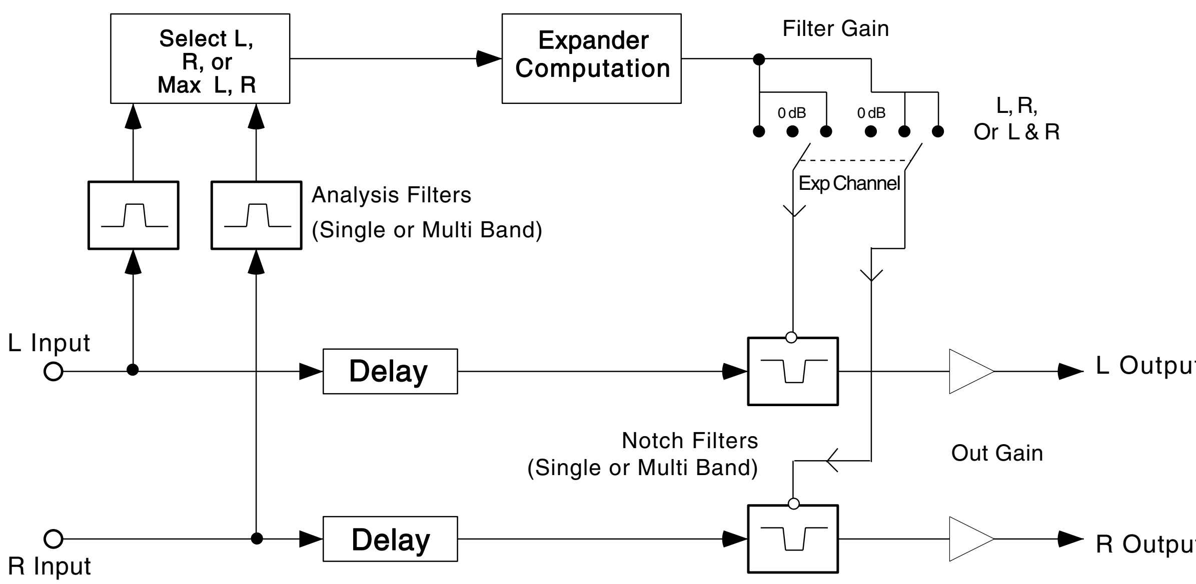


Figure 117 **Band suppression (Tone or Harmonic)**

To determine how much to expand the signal, the expander must measure the signal level. Since musical signal levels will change over time, the expansion amounts must change as well. You can control how fast the expansion changes in response to changing signal levels with the attack and release time controls.

The attack time is defined as the time for the expansion to turn off when the signal rises above the threshold. This time should be very short for most applications. The expander release time is the time for the signal to expand down after the signal drops below threshold. The expander release time may be set quite long. An expander may be used to suppress background noise in the absence of signal, thus typical expander settings use a fast attack (to avoid losing real signal), slow release (to gradually fade out the noise), and the threshold set just above the noise level. You can set just how far to drop the noise with the expansion ratio.

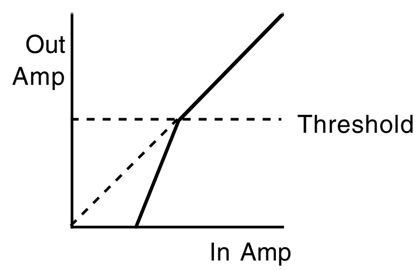


Figure 118 Expansion transfer characteristic

The signal being expanded may be delayed relative to the side chain processing. The delay allows the signal to stop being expanded just before an attack transient arrives. Since the side chain processing "knows" what the input signal is going to be before the main signal path does, it can tame down an attack transient by releasing the expander before the attack actually happens.

A meter is provided to display the amount of gain reduction that is applied to the signal as a result of expansion.

Parameters:

Page 1 (Tone Suppressor)

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
Band FreqC	8 to 25088 Hz		
Band FreqF	-100 to 100 ct	SC Input	L, R, L & R
Band Width	0.010 to 5.000 oct	ExpandChan	L, R, L & R

Page 1 (HarmonicSuppress)

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
Harmonics	Even, Odd, All		
Fund FreqC	8 to 25088 Hz	SC Input	L, R, L & R
Fund FreqF	-100 to 100 ct	ExpandChan	L, R, L & R

Page 2

<table>
<tr><td>Atk Time</td><td>0.0 to 228.0 ms</td><td>Ratio</td><td>1:1.0 to 1:17.0</td></tr>
<tr><td>Rel Time</td><td>0 to 3000 ms</td><td>Threshold</td><td>-79.0 to 0.0 dB</td></tr>
<tr><td>SmoothTime</td><td>0.0 to 228.0 ms</td><td>MakeUpGain</td><td>Off, -79.0 to 24.0 dB</td></tr>
<tr><td>Signal Dly</td><td>0.0 to 25.0 ms</td><td></td><td></td></tr>
<tr><td colspan="4">||||||||||||||||||||||||||||||| Reduction</td></tr>
<tr><td colspan="4">-dB 40 20 12 8 6 4 2 0</td></tr>
</table>

In/Out When set to **In** the expander is active; when set to **Out** the band suppressor is bypassed.

Out Gain The output gain parameter may be used to increase the gain by as much as 24 dB, or reduce the gain to nothing. Note that the Out Gain parameter does not control the signal level when the algorithm is set to **Out**.

Band FreqC For **Tone Suppressor**, the coarse control for the center frequency of the filter band to be expanded. Only signal components centered in the band will be expanded.

Band FreqF For **Tone Suppressor**, the fine control for the center frequency of the filter band to be expanded. Only signal components centered in the band will be expanded.

Band Width For **Tone Suppressor**, the width of the frequency band to be expanded expressed in octaves. Small values expand a very narrow range of frequencies. Large values expand a broad range of frequencies.

Fund FreqC For **HarmonicSuppress**, the coarse frequency control sets the fundamental frequency of the harmonic structure to be expanded. Since the filter is a comb filter, the separation between harmonically related expansion bands is also controlled.

Fund FreqF For **HarmonicSuppress**, the fine frequency control sets the fundamental frequency of the harmonic structure to be expanded. Since the filter is a comb filter, the separation between harmonically related expansion bands is also controlled.

Harmonics For **HarmonicSuppress**, sets the harmonic structure of the expansion comb filter. When set to **Even**, only the even harmonics of the specified fundamental frequency (including any dc signal level) are expanded, with no expansion of the fundamental. When set to **Odd**, the odd harmonics, including the fundamental, are expanded. The **All** setting expands all even and odd harmonics including any dc signal level. The **All** setting is the same as the **Even** when the **Even** frequency is set to half the value of **All**.

SC Input Select the input source channel for side-chain processing—left (**L**), right (**R**) or both (**L & R)**. When set to **L & R**, the maximum of left and right amplitudes is used.

ExpandChan Select which input channel will receive expander processing—left, right or both. If you select left or right, the opposite channel will pass through unaffected.

Atk Time The time for the expander to increase the gain of the signal (turns off the expander) after the signal rises above threshold.

Rel Time The time for the expander to reduce the signal level when the signal drops below the threshold (turning on expansion).

SmoothTime A lowpass filter in the control signal path. It is intended to smooth the output of the expander's envelope detector. Smoothing will affect the attack or release times when the smoothing time is longer than one of the other times.

Signal Dly	The time in ms by which the input signal should be delayed with respect to expander side chain processing (i.e. side chain predelay). This allows the expansion to appear to turn off just before the signal actually rises.
Ratio	The expansion ratio. High values (**1:17** max) are highly expanded, low values (**1:1** min) are moderately expanded.
Threshold	The expansion threshold level in dBFS (decibels relative to full scale) below which the signal begins to be expanded.
MakeUpGain	Provides an additional control of the output gain. The Out Gain and MakeUpGain controls are additive (in decibels) and together may provide a maximum of 24 dB boost to offset gain reduction due to expansion.

380 Ring Modulator
680 Mn Ring Modulate

A configurable ring modulator

PAUs: 1

Ring modulation is a simple effect in which two signals are multiplied together. Typically, an input signal is modulated with a simple carrier waveform such as a sine wave or a sawtooth. Since the modulation is symmetric ($a * b = b * a$), deciding which signal is the carrier and which is the modulation signal is a question of perspective. A simple, unchanging waveform is generally considered the carrier.

To see how the ring modulator works, we'll have to go through a little high school math and trigonometry. If you like, you can skip the how and why and go straight to the discussion of controlling the algorithm. Let's look at the simple case of two equal amplitude sine waves modulating each other. Real signals will be more complex, but they will be much more difficult to analyze. The two sine waves generally will be oscillating at different frequencies. A sine wave signal at any time t having a frequency f is represented as $\sin(ft + \phi)$ where ϕ is constant phase angle to correct for the sine wave not being 0 at $t = 0$. The sine wave could also be represented with a cosine function which is a sine function with a 90° phase shift. To simply matters, we will write $A = f_1 t + \phi_1$ for one of the sine waves and $B = f_2 t + \phi_2$ for the other sine wave. The ring modulator multiplies the two signals to produce sin A sin B. We can try to find a trigonometric identity for this, or we can just look it up in a trigonometry book:

$2 \sin A \sin B = \cos(A - B) - \cos(A + B)$

This equation tells us that multiplying two sine waves produces two new sine waves (or cosine waves) at the sum and difference of the original frequencies. The following figure shows the output frequencies (solid lines) for a given input signal pair (dashed lines):

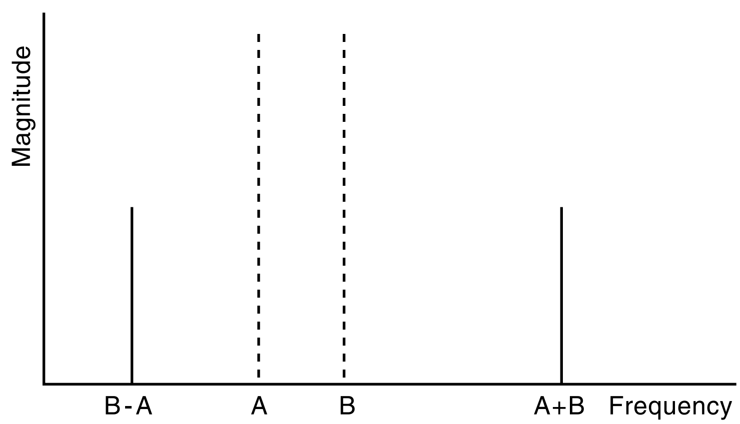


Figure 119 Result of modulating two sine waves A and B

This algorithm has two operating modes which is set with the Mod Mode parameter. In **L * R** mode, you supply the modulation and carrier signals as two mono signals on the left and right inputs. The output in **L * R** mode is also mono and you may use the **L * R** Pan parameter to pan the output. The oscillator parameters on parameter pages 2 and 3 will be inactive while in **L * R** mode. Figure 121 shows the signal flow when in **L * R** mode:

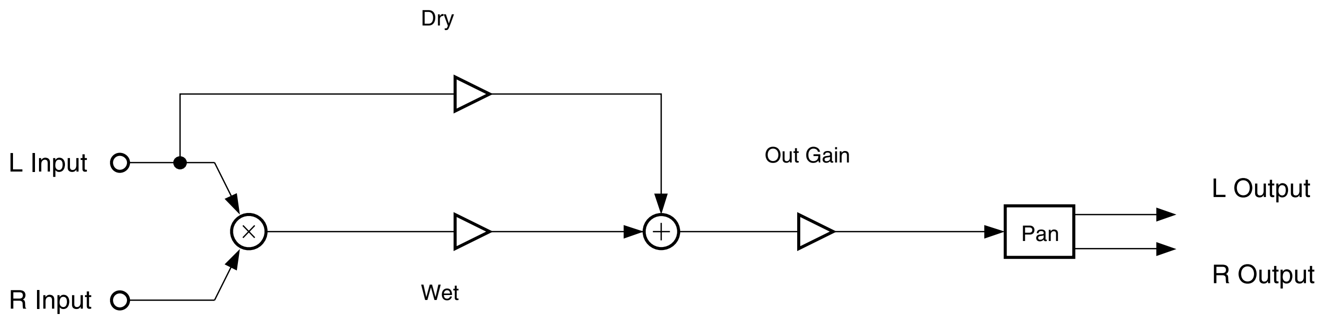


Figure 120 L ∗ R mode ring modulator

The other modulation mode is **Osc**. In **Osc** mode, the algorithm inputs and outputs are stereo, and the carrier signal for both channels is generated inside the algorithm. The carrier signal is the sum of five oscillators (see Figure 122).

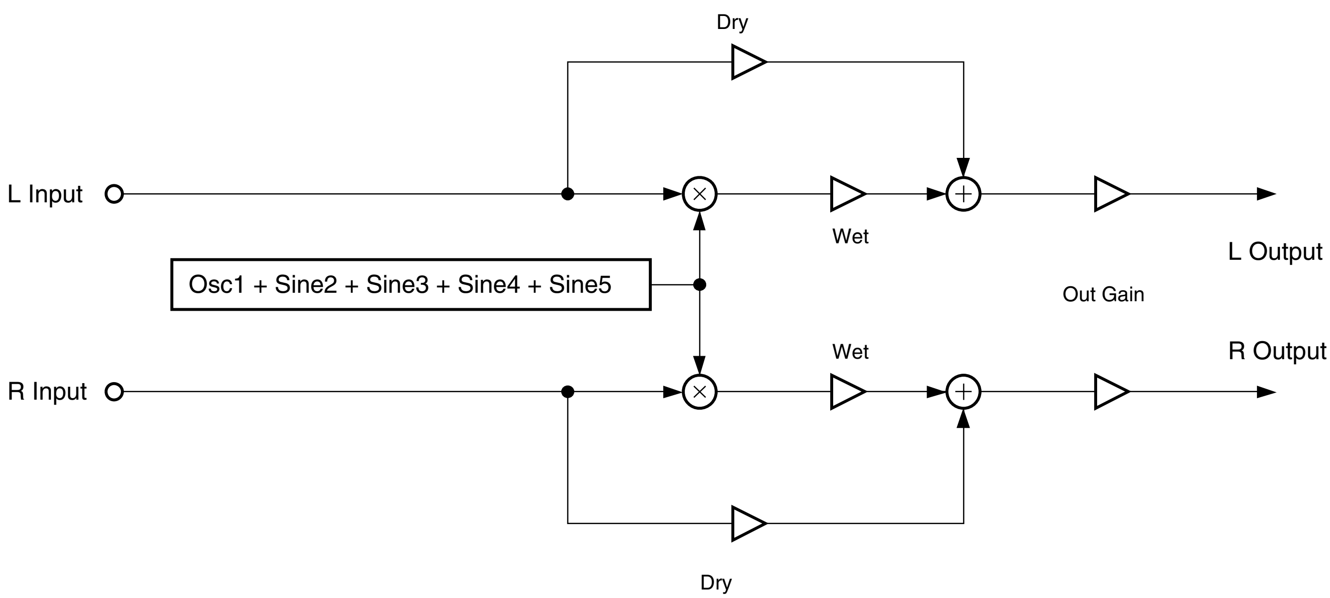


Figure 121 Osc mode ring modulator

Four of the oscillators are simple sine waves and a fifth may be configured to one of a variety of wave shapes. With all oscillators, you can set level and frequency. The configurable oscillator also lets you set the wave shape. Available shapes are **Sine**, **Saw+**, **Saw–**, **Pulse**, **Tri** and **Expon** (**Figure 122**). **Sine** is simply another sine waveform. **Tri** produces a triangular waveform, and **Expon** produces a waveform with narrow, sharp peaks which seems to rise exponentially from 0. **Pulse** produces a series of square pulses where the pulse width can be adjusted with the Osc1PlsWid parameter. When pulse width is **50**%, the signal is a square wave. The Osc1PlsWid parameter is active only when the **Pulse** waveform is selected. **Saw+** and **Saw–** produce rising and falling sawtooth waveforms. The **Pulse** and **Saw** waveforms have abrupt, discontinuous changes in amplitude which can be smoothed. The pulse wave is implemented as a hard clipped sine wave, and, at **50**% width, it turns into a sine wave when set to 100% smoothing. The sudden change in amplitude of the sawtooth waves develops a more gradual slope with smoothing, ending up as triangle waves when set to 100% smoothing.

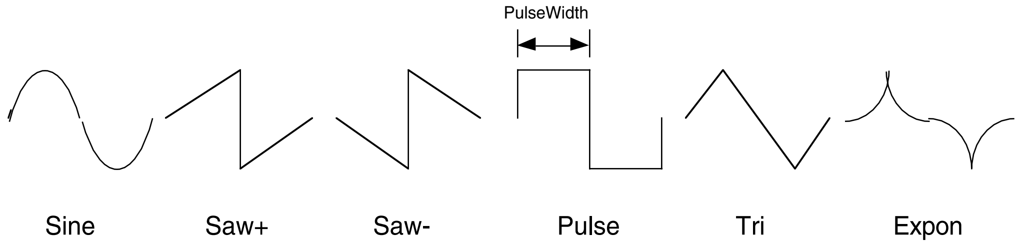


Figure 122 Configurable wave shapes

Parameters

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
Mod Mode	L ∗ R or Osc	L ∗ R Gain	Off, -79.0 to 48.0 dB
		L ∗ R Pan	-100 to 100%

Page 2

Osc1 Lvl	0 to 100%	Osc1 Freq	8 to 25088 Hz
Osc1 Shape	Sine		
Osc1PlsWid	0 to 100%		
Osc1Smooth	0 to 100%		

Page 3

Sine2 Lvl	0 to 100%	Sine2 Freq	8 to 25088 Hz
Sine3 Lvl	0 to 100%	Sine3 Freq	8 to 25088 Hz
Sine4 Lvl	0 to 100%	Sine4 Freq	8 to 25088 Hz
Sine5 Lvl	0 to 100%	Sine5 Freq	8 to 25088 Hz

Wet/Dry The amount of modulated (wet) signal relative to unaffected (dry) signal as a percent. When in **L ∗ R** mode, the left input will be used as the dry signal.

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

Mod Mode Switches between the two operating modes of the algorithm (**L ∗ R** and **Osc**). The **L ∗ R** mode treats the left and right inputs as the modulator and carrier signals. It does not matter which input is left and which is right except to note that only the left signal will be passed through as dry.

L ∗ R Pan The output panning of the both wet and dry signals. This control is active only in **L ∗ R** mode. **-100%** is panned fully left, **0%** is panned center and **100%** is panned right.

Osc1 Lvl The level of the configurable oscillator. **0%** is off and **100%** is maximum. This parameter is active only in **Osc** mode.

Osc1 Freq The fundamental frequency of the configurable oscillator. The oscillators can be set through the audible frequencies 8-25088 Hz with one-semitone resolution. This parameter is active only in **Osc** mode.

Osc1Shape	Shape selects the waveform type for the configurable oscillator. Choices are **Sine**, **Saw+**, **Saw–**, **Pulse**, **Tri**, and **Expon**. This parameter is active only in **Osc** mode.
Osc1PlsWid	When the configurable oscillator is set to **Pulse**, the PlsWid parameter sets the pulse width as a percentage of the waveform period. The pulse is a square wave when the width is set to **50**%. This parameter is active only in **Osc** mode and when the Pulse waveform is selected.
Osc1Smooth	Smooths the **Saw+**, **Saw–**, and **Pulse** waveforms. For the sawtooth waves, smoothing makes the waveform more like a triangle wave. For the **Pulse** wave, smoothing makes the waveform more like a sine wave.
Sine*n* Lvl	The four sine wave oscillators (*n* = 2...5) may have their levels set between **0**% (off) and **100**% (maximum). This parameter is active only in **Osc** mode.
Sine*n* Freq	The four sine wave oscillators (*n* = 2...5) may have their frequencies set with this parameter. The oscillators can be set through the audible frequencies 8–25088 Hz with one-semitone resolution. This parameter is active only in **Osc** mode.

381 Pitcher
681 Mn Pitcher

Creates pitch from pitched or non-pitched signal

PAUs: 1

This 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.

If the original signal has no significant components at the desired pitch or harmonics, the output level remains low. The left and right inputs are processed independently with common controls of pitch and weighting. 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.

The four weight parameters named Odd Wts, Pair Wts, Quartr Wts and Half Wts control the exact shape of the frequency response of **Pitcher**. An exact description of what each one does is, unfortunately, impossible, since there is a great deal of interaction between them. Here are some examples with a Pitch setting of **1 kHz**, which is close to a value of C6. Weight settings are listed in brackets following this format: [Odd, Pair, Quartr, Half].

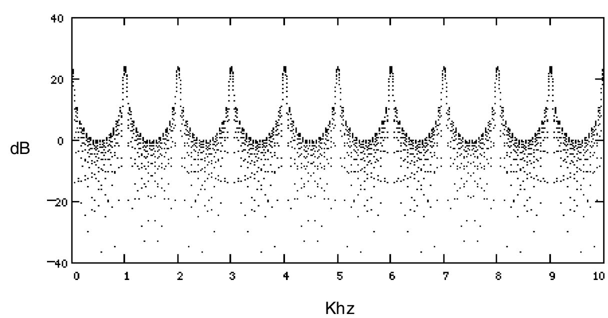


Figure 123 Pitcher at [100, 100, 100, 100]

In Figure 124, all peaks are exact multiples of the fundamental frequency set by the Pitch parameter. This setting gives the most "pitchiness" to the output.

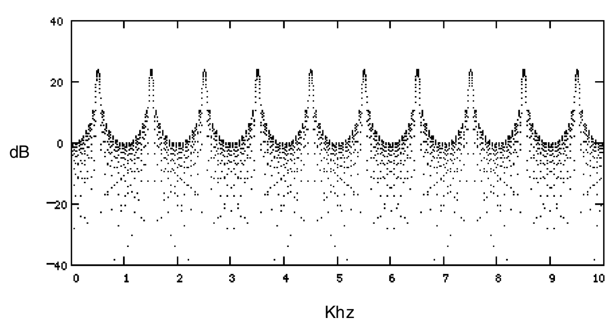


Figure 124 Pitcher at [-100, 100, 100, 100]

In Figure 125, peaks are odd multiples of a frequency one octave down from the Pitch setting. This gives a hollow, square-wave-like sound to the output.

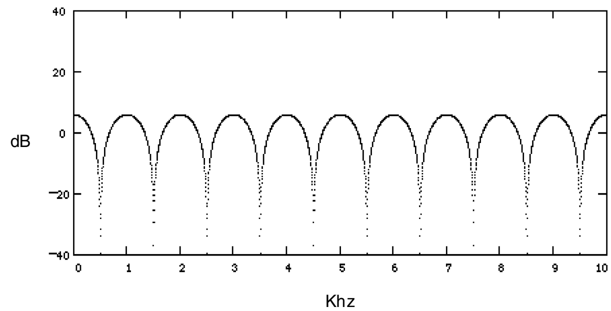


Figure 125 Pitcher at [100, 0, 0, 0]

In Figure 126, there are deeper notches between wider peaks

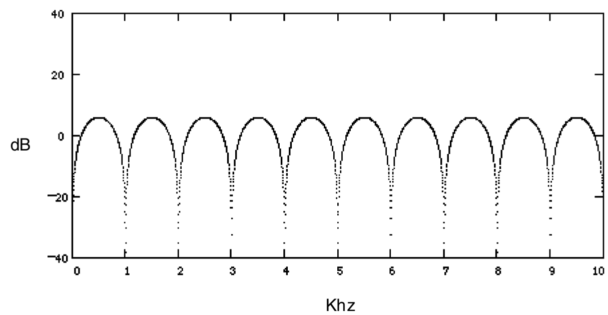


Figure 126 Pitcher at [-100, 0, 0, 0]

In Figure 127, there are peaks on odd harmonic multiples and notches on even harmonic multiples of a frequency one octave down from the Pitch setting.

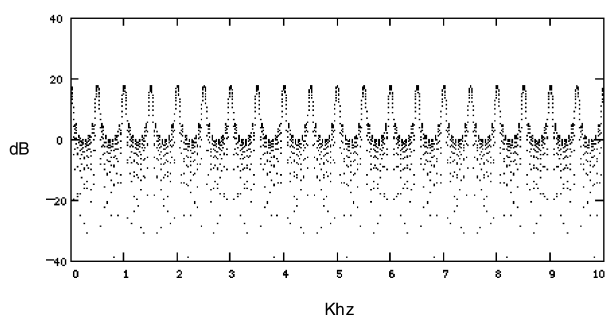


Figure 127 Pitcher at [0, 100, 100, 100]

Figure 128 is like [**100**,**100**,**100**,**100**], except that all the peaks are at (all) multiples of half the Pitch frequency.

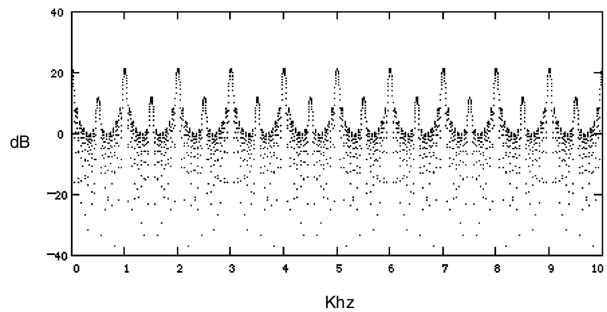


Figure 128 Pitcher at [50,100,100,100]

Figure 129 is halfway between [**0**,**100**,**100**,**100**] and [**100**,**100**,**100**,**100**].

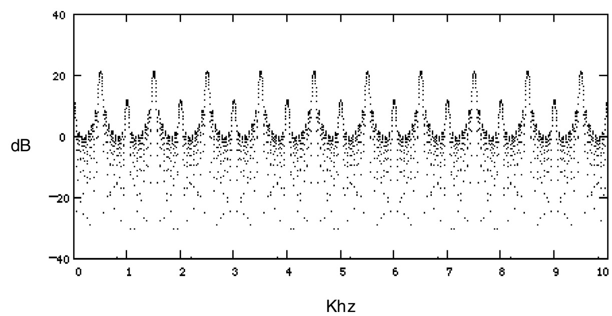


Figure 129 Pitcher at [-50,100,100,100]

Figure 130 is halfway between [**0**,**100**,**100**,**100**] and [**-100**,**100**,**100**,**100**]. If the Odd parameter is modulated with an FXMOD, then you can morph smoothly between the [**100**,**100**,**100**,**100**] and [**-100**,**100**,**100**,**100**] curves.

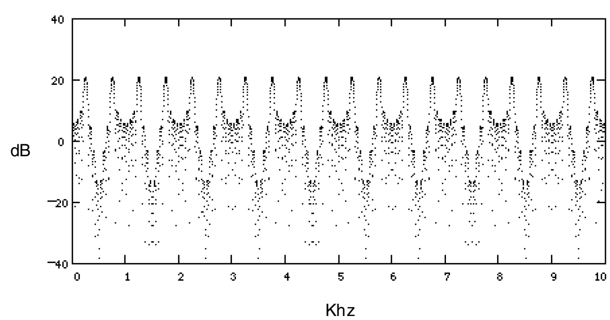


Figure 130 Pitcher at [100, -100, 100, 100]

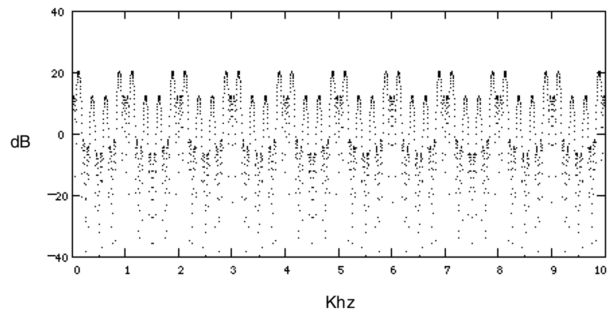


Figure 131 Pitcher at [100, 100, -100, 100]

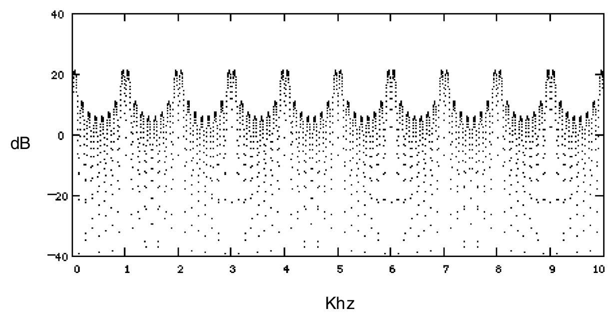


Figure 132 Pitcher at [100, 100, 100, -100]

To save space, we've left out the other 1,632,240,792 response curves.

Parameters

Wet/Dry	0 to 100 %wet	Out Gain	Off, -79.0 to 24.0 dB
Pitch	C-1 to G9	Ptch Offst	-12.0 to 12.0 ST
Odd Wts	-100 to 100 %	Quartr Wts	-100 to 100 %
Pair Wts	-100 to 100 %	Half Wts	-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.

Pitch The fundamental pitch imposed upon the input.

Ptch Offst An offset from the pitch frequency in semitones. This is also available for adding an additional continuous controller mod like pitch bend.

All other parameters These parameters control the exact shape of the frequency response of **Pitcher**. An exact description of what each one does is, unfortunately, impossible, since there is a great deal of interaction between them. For examples, examine the figures above.

382 Poly Pitcher

Creates pitch from pitched or non-pitched signal—twice.

PAUs: 2

Poly Pitcher is closely based on Algorithm **381 Pitcher**, and most of the features of **Poly Pitcher** are covered in the section on Algorithm **381 Pitcher**. **Poly Pitcher** is really just a pair of **Pitcher** algorithms (A and B) using the same inputs and summing to the same outputs. There is one set of weight parameters (Odd Wts, Pair Wts, Quartr Wts, and Half Wts), which are applied to both pitcher sections. However, the actual pitch settings for the two pitchers can be set independently. You can also set the relative level of the two pitchers with the A/B Mix parameter. One last difference from Algorithm **381 Pitcher** is that there are separate pitch offset parameters for left and right channels for both pitchers. With separate left/right controls for the pitch offset, you can produce a greater sense of stereo separation.

Parameters:

Page 1

Wet/Dry	0 to 100 %wet	Out Gain	Off, -79.0 to 24.0 dB
Odd Wts	-100 to 100 %	Quartr Wts	-100 to 100 %
Pair Wts	-100 to 100 %	Half Wts	-100 to 100 %

Page 2

A/B Mix	0 to 100 %		
Pitch A	C -1 to G 9		
PchOffs AL	-12.0 to 12.0 ST	PchOffs AR	-12.0 to 12.0 ST
Pitch B	C -1 to G 9		
PchOffs BL	-12.0 to 12.0 ST	PchOffs BR	-12.0 to 12.0 ST

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.

Odd Wts
Pair Wts
Quartr Wts
Half Wts These parameters control the exact shape of the frequency response of **Poly Pitcher**. An exact description of what each one does is, unfortunately, impossible, since there is a great deal of interaction between them. For examples, examine the figures in the section on Algorithm **381 Pitcher**.

A/B Mix The relative amount of pitcher A and pitcher B to mix to the final output. At **0%**, only pitcher A can be heard at the output, and at **100%**, you can hear only pitcher B. **50%** produces equal amounts of both.

Pitch A, B	The fundamental pitch imposed upon the input expressed in semitone scale intervals. Pitcher A and pitcher B may be set independently.
PchOff AL **PchOff AR** **PchOff BL** **PchOff BR**	An offset from the pitch frequency in semitones. Not only are the A and B pitchers treated separately, the left and right channels have their own controls for increased stereo separation. Pitch offset may be useful as a modifiable control resembling pitch bend.

383 Pitcher+MiniVerb

Combination algorithm of Pitcher followed by MiniVerb

PAUs: 2

Pitcher+MiniVerbis Algorithm **381 Pitcher** followed by Algorithm **1 MiniVerb**. **Pitcher** applies a filter to the signal, the filter having a regular series of peaks in its frequency response which generally imposes a pitch on the input signal. The **MiniVerb** reverb is then applied to the "pitched" signal. See the relevant sections for complete details on these algorithm components.

There are several parameters for controlling the routing and mixing of signals. As might be expected, Wet/Dry sets the level of the wet (**Pitcher+MiniVerb**) signal relative to the main dry input signal. The Mix Pitcher and Mix Reverb parameters set the amounts of the outputs of both of the component effects to send to the main algorithm outputs. The Pch/Dry>Rv parameter sets the amount of pitcher signal to feed to the reverb relative to dry input signal.

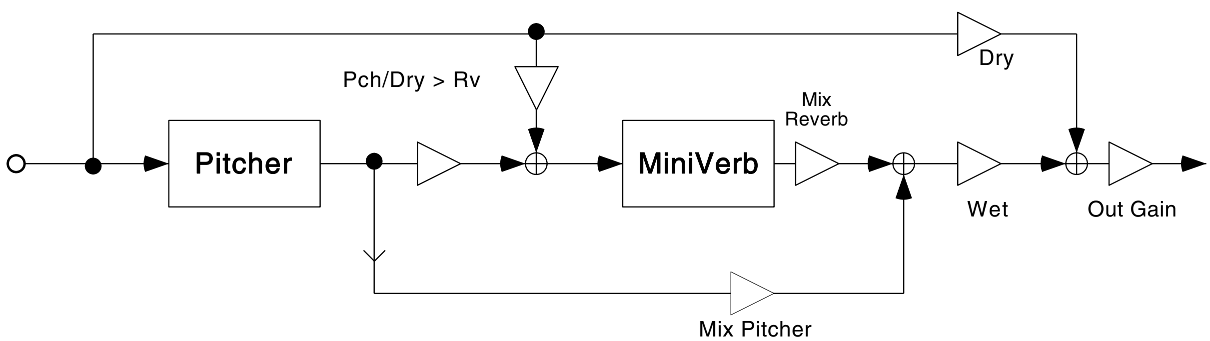


Figure 133 Signal flow of Pitcher+MiniVerb

Parameters:

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
Mix Pitcher	0 to 100%		
Mix Reverb	0 to 100%		

Page 2

Pt Pitch	C -1 to G 9		
Pt Offst	-12.0 to 12.0 ST		
Pt Odd Wts	-100 to 100%		
Pt Pair Wts	-100 to 100%		
Pt 1/4 Wts	-100 to 100%		
Pt 1/2 Wts	-100 to 100%		

Page 3

Pch/Dry>Rv	0 to 100 %		
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 HFDamp	8 to 25088 Hz
Rv PreDlyL	0 to 620 ms	Rv PreDlyR	0 to 620 ms

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.

Mix Pitcher Adjusts the amount of the pitcher effect that is mixed together as the algorithm wet signal. Negative values polarity invert that particular signal.

Mix Reverb Adjusts the amount of the reverb effect that is mixed together as the algorithm wet signal. Negative values polarity invert that particular signal.

Pt Pitch The fundamental pitch imposed upon the input. Values are in MIDI note numbers.

Pt Offst An offset from the pitch frequency in semitones. This is also available for adding an additional continuous controller mod like pitch bend.

Pt Odd Wts
Pt Pair Wts
Pt 1/4 Wts
Pt 1/2 Wts These parameters control the exact shape of the frequency response of **Poly Pitcher** An exact description of what each one does is, unfortunately, impossible, since there is a great deal of interaction between them. For examples, examine the figures in the section on Algorithm **381 Pitcher**.

Pch/Dry->Rv This parameter controls how much of the pitcher effect is mixed with dry and fed into the reverb effect. This control functions like a wet/dry mix, where **0%** is completely dry and **100%** is pitcher effect only.

Rv Time The reverb time displayed is accurate for normal settings of the other parameters (Rv HF Damp = **25088 kHz**, and Rv DiffScl, Rv SizeScl and Rv Density = **1.00x**). Changing Rv Time to **Inf** creates an infinitely sustaining reverb.

Rv Type 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 Rv Types will sound best with shorter Rv Times, and vice versa. (Note that since this parameter changes the structure of the reverb algorithm, you may not modulate it.)

Rv DiffScl A multiplier that affects the diffusion of the reverb. At **1.00x**, the diffusion will be the normal, carefully adjusted amount for the current Rv Type. Altering this parameter will change the diffusion from the preset amount.

Rv SizeScl A multiplier that changes the size of the current room. At **1.00x**, the room will be the normal, carefully tweaked size of the current Rv 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).

Rv Density A multiplier that affects the density of the reverb. At **1.00x**, the room density will be the normal, carefully set amount for the current Rv Type. Altering this parameter will change the density of the reverb, which may color the room slightly.

Rv HFDamp Reduces high frequency components of the reverb above the displayed cutoff frequency. Removing higher reverb frequencies can often make rooms sound more natural.

Rv PreDlyL/R The delay between the start of a sound and the output of the first reverb reflections from that sound. Longer predelays 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.

384 Flange<>Pitcher

This algorithm is one of a group of *configurable* combination algorithms—that is, there's more than one effect in the algorithm, and you can change the sequence of those effects. With this algorithm, for example, you can have either a flanger followed by a pitcher, or vice versa.

The combination algorithms are organized in groups, with IDs predominantly in the **400**s (there are a few exceptions, of course). For a description of Algorithm **384** and other combination algorithms, follow one of the links below:

Combination Algorithms on page 279

Configurable Combination Algorithms on page 289

More Combination Algorithms on page 299

385 Frequency Offset
386 MutualFreqOffset
682 Mn Freq Offset

Single Side Band Modulation

PAUs: 2

Frequency Offset and **MutualFreqOffset** perform single side band (SSB) modulation. Essentially what this means is that every frequency component of your input sound will be offset (in frequency) or modulated by the same amount. In the **Frequency Offset** algorithm, if you have the OffsetFreq and Offs Scale parameters set to a frequency of **100 Hz**, then all frequencies in your sound will be offset up (or down) by 100 Hz. Both algorithms produce modulation both up and down and you can control the relative amount of up and down modulation with separate level and pan controls. The **Frequency Offset** algorithms are very similar to Algorithm 380 Ring Modulator, which is a dual side band modulator. If you set the up and down level parameters to match, the output will be quite close to the **Ring Modulator** output. Unlike **Ring Modulator** however, you can choose to listen to just the up modulation or the down modulation, and not necessarily both. In addition, you can pan the up and down modulation outputs in different directions (left or right).

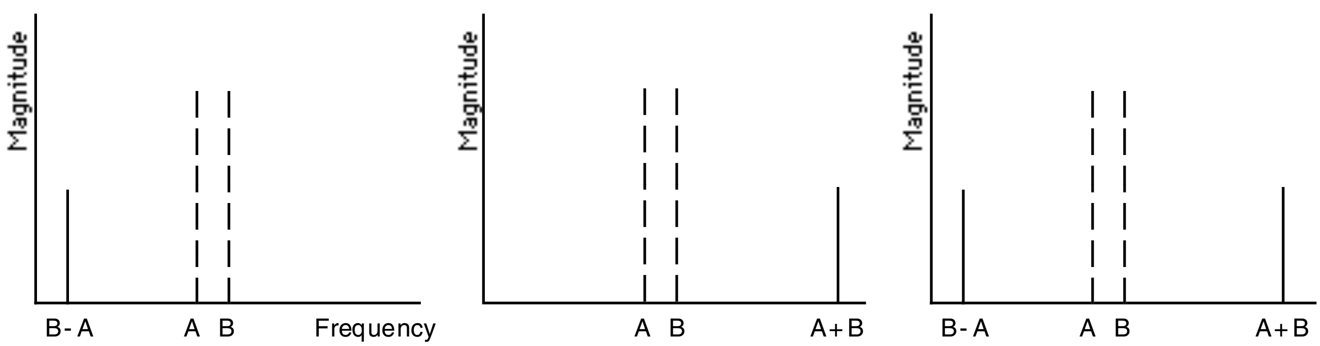


Figure 134 **Single side band modulation (frequency offset) (i) down and (ii) up. When combined (iii) we get dual side band modulation (ring modulation).**

Frequency Offset is a mono algorithm that modulates your input signal with a pure sine wave. A sine wave contains a single frequency, so your input signal will be offset in frequency by the frequency of the sine wave.

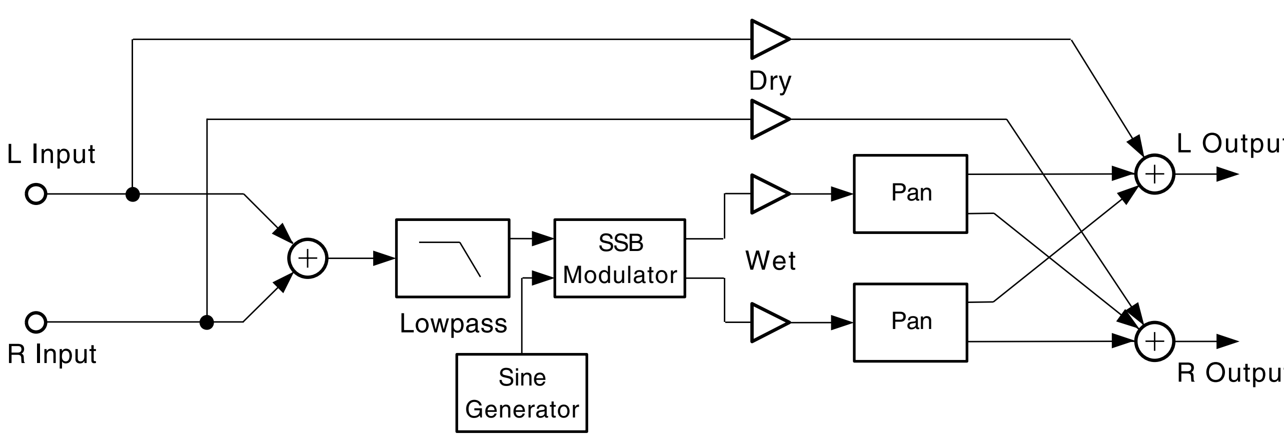


Figure 135 **Block diagram of Frequency Offset**

MutualFreqOffset modulates the two input signals (left and right) with each other. If one of the signals is a sine wave, the algorithm behaves like **Frequency Offset**. Now imagine that one of the input signals is the sum of two sine waves. Both of the two sine waves will modulate the signal on the other input. For example, if the two sine waves are at 100 Hz and 200 Hz, upward modulation of another signal at 1000 Hz will produce pitches at 1100 Hz and 1200 Hz. Obviously this is going to get very complicated to work out when the inputs are more than simple sine waves. **MutualFreqOffset** may require extra gain compensation so separate left, right input gain controls and a gain control for the final (wet) output are provided.

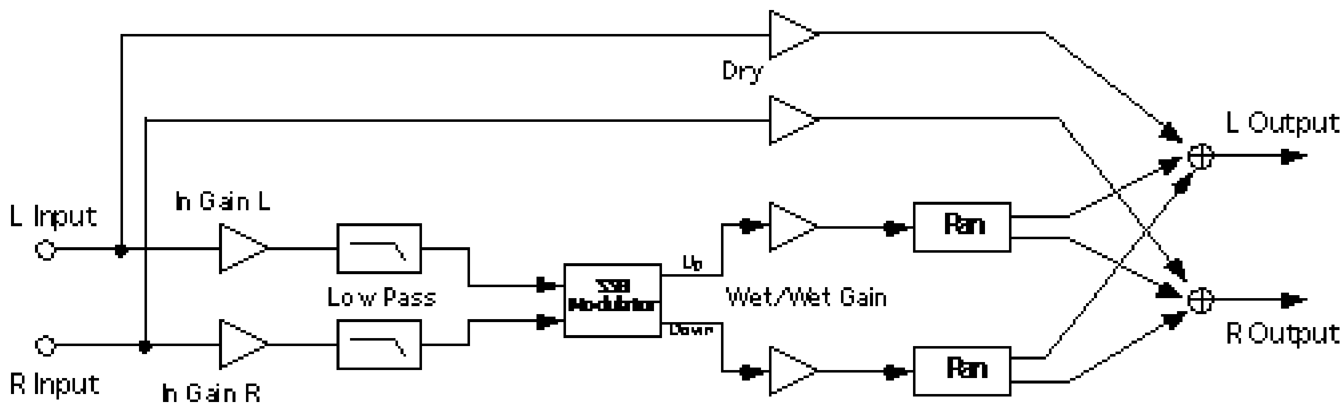


Figure 136 Block diagram of MutualFreqOffset

With downward modulation, you will hear the pitch drop as you increase the frequency of the input sound. The downward modulation is a difference (subtraction) in frequencies. If the difference drops to negative values, the frequency will start to rise again. It doesn't matter which frequency gets subtracted from the other, since the result will sound the same. For example 1000 Hz - 100 Hz = 900 Hz will produce the same pitch as 100 Hz - 1000 Hz = -900 Hz. Similarly, upward modulation is a sum of frequencies and pitch will rise as you increase the frequency of input sound. However in a digital sampled system, frequencies higher than half the sample rate (the Nyquist rate, 24 kHz in KDFX) cannot be represented. When the summed frequencies pass the Nyquist rate, the pitch starts coming back down.

Both **Frequency Offset** and **MutualFreqOffset** provide panning with width of the dry input signals directly to the output.

Parameters (Frequency Offset):

Page 1

Wet/Dry	0 to 100 %wet	Out Gain	Off, -79.0 to 24.0 dB
In Lowpass	8 to 25088 Hz		

Page 2

OffsetFreq	0.00 to 10.00 Hz		
Offs Scale	1 too25088x		
DwnOffsLvl	0 to 100 %	UpOffsLvl	0 to 100 %
DwnOffsPan	-100 to 100 %	UpOffsPan	-100 to 100 %

Parameters (MutualFreqOffset):

Page 1

Wet/Dry	0 to 100 %wet	Out Gain	Off, -79.0 to 24.0 dB
In Gain L	Off, -79.0 to 24.0 dB	Wet Gain	Off, -79.0 to 24.0 dB
In Gain R	Off, -79.0 to 24.0 dB		
InLowpassL	8 to 25088 Hz	InLowpassR	8 to 25088 Hz

Page 2

DwnOffsLvl	0 to 100 %	UpOffsLvl	0 to 100 %
DwnOffsPan	-100 to 100 %	UpOffsPan	-100 to 100 %

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

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

In Lowpass A first-order lowpass filter is provided to reduce the bandwidth of the input signal. Considering the many new frequency components that will be created, the lowpass filter may help tame the sound. **MutualFreqOffset** has separate controls for left and right input channels.

OffsetFreq **Frequency Offset** algorithm only. The frequency when multiplied with Offs Scale which is the modulation frequency. The offset or modulation frequency is the frequency in Hz which is added to and/or subtract from all the frequencies of the input signal.

Offs Scale **Frequency Offset** algorithm only. A scale factor which is multiplied with the OffsetFreq parameter to produce the offset or modulation frequency.

In Gain L/R **MutualFreqOffset** algorithm only. Two independent gain controls (left and right) to adjust the amplitude of the input signals. (See Wet Gain.)

Wet Gain The gain or amplitude of the modulated (wet) signal. The Wet Gain parameter and the In Gain L/R parameters are for the **MutualFreqOffset** algorithm which produces an output based on multiplying the left and right inputs. This is very different from adding signals, and controlling levels can be tricky. Ideally you would set the input gains and the wet gain so that the signal level remains flat when you adjust Wet/Dry while ensuring you hear no internal clipping. Use Out Gain for overall level control.

DwnOffsLvl The level of the down modulated signal. Negative values polarity invert the signal.

UpOffsLvl The level of the up modulated signal. Negative values polarity invert the signal.

DwnOffsPan The down modulated signal may be panned to the left or right algorithm outputs. **-100%** sends the signal to the left output and **100%** sends the signal to the right output.

UpOffsPan The up modulated signal may be panned to the left or right algorithm outputs. **-100%** sends the signal to the left output and **100%** sends the signal to the right output.

387 WackedPitchLFO

An LFO based pitch shifter.

PAUs: 3

Okay, it ain't pretty, but **WackedPitchLFO** uses LFO modulated delay lines with cross fades to produce a shift of signal pitch. You can set the amount of shift in coarse steps of semitones or fine steps of cents (hundredths of a semitone). This shifter works using the same concepts used to detune a sound in a chorus algorithm. In a chorus algorithm, an LFO is used to change the length of a delay line. By smoothly changing a delay line length from long to short to long, the signal is effectively resampled at a new rate causing the pitch to rise and fall. In the **WackedPitchLFO** algorithm, the signal level is made to rise and fall in time with the delay line movement so that we only hear signal from the delay line when the pitch is rising (or falling). By overlapping and adding several delay taps moved by several LFOs, we can then produce a relatively smooth pitch shifted signal.

Relatively.

It is possible for sounds coming out of the delay lines to be out of phase, which means that a certain amount of cancellation can occur. The result sounds like there is a certain amount of tremolo in the pitch shifted signal. The depth of the tremolo will depend on the pitch of the signal, the rate of the LFO and the amount of pitch shifting—it will be different for every pitch. The rate of the tremolo is the rate of the LFO. At higher rates the tremolo can be objectionable. At slow LFO rates, the pitch shifting is quite clean, though you will hear some flanging. However longer delay line lengths are needed at slower LFO rates for a given amount of pitch shift. The delays can get quite long, and it is possible to run out of available delay (in which case you will get less pitch shift than you request). The trade-off is tremolo for delay. Higher frequency signals will sound better when pitch shifted than lower frequency signals. Increasing the amount of pitch shift will increase both the amount of tremolo and the amount of delay.

You can introduce feedback in **WackedPitchLFO**. When you do, the signal can be made to continuously rise (or fall) as it repeatedly passes through the feedback loop.

The pitch shifter is based on delay lines. Changing the amount of pitch shift will produce large jumps in delay line lengths, and you will hear the jumps as clicks if you are playing a sound while changing the shift amount. For this reason, the shift amount parameters will not work well as modifiable parameters on an FXMOD page.

Parameters:

Page 1

Wet/Dry	-100 to 100 %wet	Out Gain	Off, -79.0 to 24.0 dB
Feedback	0 to 100 %	Highpass	8 to 25088 Hz
LFO Rate	0.01 to 10.00 Hz	Shift Crs	-24 to 24 ST
Lowpass	8 to 25088 Hz	Shift Fine	-100 to 100 ct

Wet/Dry The relative amount of input signal and pitch shifted 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 By introducing feedback, the pitch can be made to continually rise or fall as the signal makes successive passes through the pitch shifter.

LFO Rate	The frequency of the LFOs that drive the pitch shifter. The pitch shifter produces a certain amount of tremolo that will oscillate based on this rate. However reducing the rate will increase the delay lengths needed by the pitch shifter.
Shift Crs	A coarse adjust to the pitch shift amount from -24 to +24 semitones. The algorithm performs best when the amount of pitch shift is small.
Shift Fine	A fine adjust to the pitch shift amount from -100 to +100 cents (hundredths of a semitone).
Lowpass	A lowpass filter in the algorithm feedback loop. Use the lowpass to tame some of the higher frequency artifacts. This is especially important when using feedback.
Highpass	A highpass filter in the algorithm feedback loop. Use the highpass to tame the lower frequencies when using feedback.

390 Chaos!
690 Mn Chaos!

Fun with chaos and instability

PAUs: 2

The moment you scroll to the **Chaos!** algorithm, you will discover it is wildly unstable. **Chaos!** is a delay feedback algorithm which includes lots of gain with distortion plus plenty of filters tweaking the sound. Modifying the parameters will often cause the algorithm to jump from one chaotic instability state to another, often unpredictably. For the most part **Chaos!** howls and resonates on its own, and while an input signal can affect the output, the effect of the input signal on the output is usually small. When self-resonating, the sound you can get can be very strange. It is particularly interesting if you keep modifying the parameters. What do you use this effect for? Well, that's the creative challenge!

You should be very careful with the Out Gain or Drive Cut settings with **Chaos!** If you start the algorithm in a stable state (not self resonating) and start increasing gains (in the distortion drive or filters), the output level can build. The feedback can be every bit as unpleasant as putting a microphone next to a loudspeaker! (There's an application: simulating PA system feedback!)

Let's take a closer look at **Chaos!**

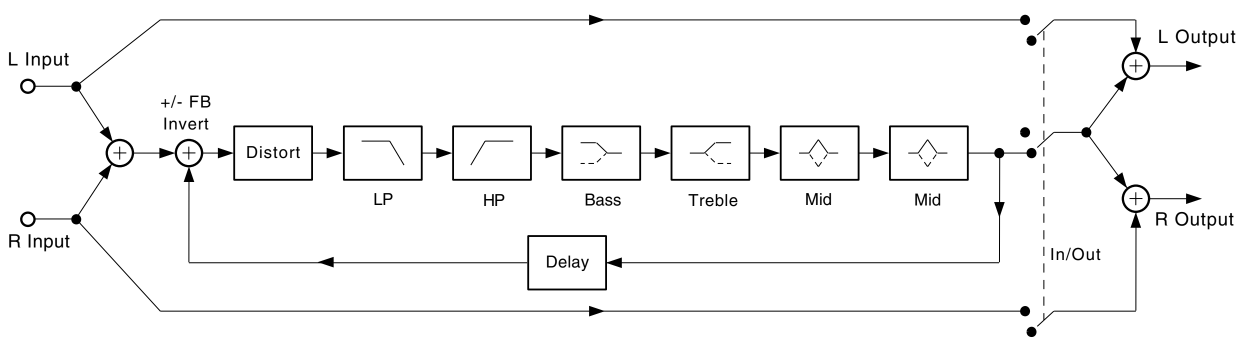


Figure 137 Chaos!

As advertised, **Chaos!** is a feedback loop with delay, distortion and lots of filters. Most of the effects in KDFX carefully manage levels on feedback loops to prevent instability. In a digital system, uncontrolled instability will usually rapidly enter digital clipping with full scale signal output. Very nasty. **Chaos!** also keeps a lid on levels, preventing digital clipping but allowing instability. You will still need to cut back on Out Gain (or Drive Cut) to bring the signal down to reasonable levels.

The distortion drive when turned up, will push **Chaos!** into instability unless Drive Cut is used to hold the level down. As the sound starts becoming unstable, your input signal will still have a strong effect on the output. As more and more drive is applied, the self-resonance dominates the output.

The delay length is expressed as a frequency where the length of the delay in seconds is 1/frequency. Why do this? A short delay line with a lot of feedback will resonate at a frequency of 1/length of the delay. It is the resonant behavior of **Chaos!** which is particularly interesting, which make the delay more naturally expressed as a frequency. Not only will the delay resonate at its natural frequency (1/length), but you may also hear many overtones (or harmonics). There is a switch to invert the feedback (FB Invert). When set to In, FB Invert will cause the natural frequency and its harmonics to be suppressed while frequencies between the harmonics now resonate. In this case the frequency one octave down and its odd harmonics are resonating.

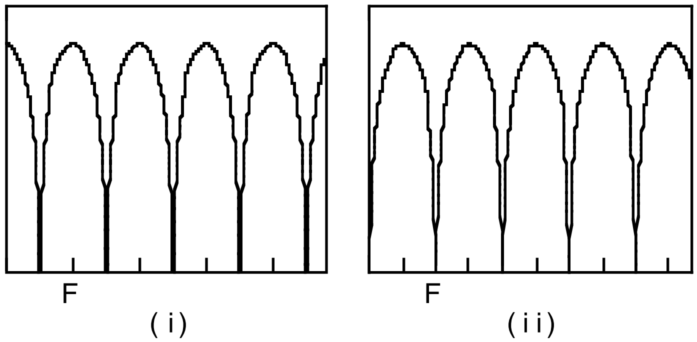


Figure 138 Resonating frequencies with FB Invert set to (i) Out and set to (ii) In.

In addition to the distortion warmth filter, there are six filters built into the delay line loop: a highpass, a lowpass, a treble and a bass shelf, and two parametric midrange filters. Boosting the shelves or mids increases the strength of instability at the boosted frequencies. Since overall level is controlled, the net effect is to reduce the level of the other frequencies. Using filters to cut frequencies is similar, but with cut it is possible to remove so much signal that the algorithm drops into stability and stops self-resonating.

The individual elements of **Chaos!** (filters and so forth) are fairly basic, and you may understand them well. When put together as the **Chaos!** algorithm, the interactions become very complex and many of the old rules don't seem to apply. Keep plugging at it.

Parameters:

Page 1

In/Out	In or Out	Out Gain	Off, -79.0 to 24.0 dB
Drive	0 to 96 dB	FB Invert	In or Out
Drive Cut	Off, -79.0 to 0.0 dB	Dly FreqC	8 to 25088 Hz
Warmth	8 to 25088 Hz	Dly FreqF	-100 to 100 ct

Page 2

Highpass	8 to 25088 Hz	Lowpass	8 to 25088 Hz
Bass Gain	-79.0 to 24.0 dB	Treb Gain	-79.0 to 24.0 dB
Bass Freq	8 to 25088 Hz	Treb Freq	8 to 25088 Hz
Mid1 Gain	-79.0 to 24.0 dB	Mid2 Gain	-79.0 to 24.0 dB
Mid1 Freq	8 to 25088 Hz	Mid2 Freq	8 to 25088 Hz
Mid1 Width	0.010 to 5.000 oct	Mid2 Width	0.010 to 5.000 oct

In/Out When set to **In**, the effect is active. When set to **Out**, the effect is bypassed.

Out Gain The overall gain or amplitude at the output of the effect. The output gain is outside and after the feedback loop.

Drive Sets how high the distortion is to be driven. The distortion and its drive gain are inside the feedback loop.

Drive Cut Reduces the signal level after the distortion. By reducing the signal level after the distortion, **Chaos!** can be returned to stability while still producing a lot of distortion. Drive Cut is also inside the feedback loop.

Warmth Warmth affects the character of the distortion. Warmth reduces (at low settings) the higher frequency distortion components without making the overall signal dull.

Dly FreqC The feedback signal path includes a short delay line which will tend to resonate at a frequency of 1/length of the delay. The delay length is therefore expressed as the resonant frequency. Note that all the filters in the feedback loop also add delay, so with more filtering, the resonance tuning will drift flat.

Dly FreqF The resonant frequency of the feedback delay line can be tuned sharp or flat in one cent (hundredths of a semitone) increments.

FB Invert The feedback signal can be inverted (subtracted instead of added) so that instead of resonance at the specified frequency and its harmonics, the resonance occurs between those frequencies. This is like setting resonance one octave lower, but using only the odd harmonics.

Highpass The highpass filter removes frequencies below the specified cut-off frequency. The filter is first order, cutting signal level at 6 dB per octave of frequency. When set to the lowest frequency, the filter is performing very little cut of the low frequencies. When **Chaos!** is self-resonating, turning up the highpass frequency will cause high frequencies to be emphasized.

Lowpass The lowpass filter removes frequencies above the specified cut-off frequency. The filter is first order, cutting signal level at 6 dB per octave of frequency. When set to the highest frequency, the filter is performing very little cut of the high frequencies. When **Chaos!** is self-resonating, turning down the lowpass frequency will cause low frequencies to be emphasized.

Bass Gain The amount of boost or cut in decibels to apply to the bass shelf filter inside the feedback loop. Boost will emphasize frequencies below the filter frequency, while cut will emphasize frequencies above the filter frequency.

Bass Freq The frequency in Hz below which the bass shelf filter performs boost or cut.

Treb Gain The amount of boost or cut in decibels to apply to the treble shelf filter inside the feedback loop. Boost will emphasize frequencies above the filter frequency, while cut will emphasize frequencies below the filter frequency.

Treb Freq The frequency in Hz above which the treble shelf filter performs boost or cut.

Mid*n* Gain The amount of boost or cut in decibels to apply to the midrange parametric filter n (1 or 2) inside the feedback loop. Boost will emphasize the specified filter frequency while cut will emphasize all other frequencies.

Mid*n* Freq The frequency in Hz at which the midrange parametric filter n (1 or 2) performs boost or cut.

Mid*n* Width The width of the frequency band in octaves of the midrange parametric filter n (1 or 2). When the filter is set for boost, a narrow band (low settings) will cause the resonating output to approach a pure tone more rapidly.

391 ADSR Synth
392 Env Synth
393 Gate Synth

Single voice waveform generator with envelope controls and envelope- following filter.

PAUs: *PAUs:* 4 for ADSR Synth and 3 for Env Synth and Gate Synth

The synthesizer algorithms perform a simple one-voice waveform generator. The generated waveform frequency is typically made to follow MIDI note events to get keyboard pitch tracking. A small number of simple waveform shapes are available. Input signals are used for envelope-following filtering and to control the synthesizer amplitude envelopes. The **ADSR Synth**, **Env Synth**, and **Gate Synth** algorithms differ only in the type of control that is available for amplitude envelopes. The input signals controlling envelope filtering and amplitude control are separately selectable between left, right, or the sum of the left and right input channels. The final synthesized waveform may be panned between left and right output channels.

The envelope-following filter is a stereo resonant filter with the resonant frequency controlled by the envelope of the input signal (the maximum of left or right). The filter type is selectable and may be one of low pass (i), high pass (ii), band pass (iii), or notch (iv).

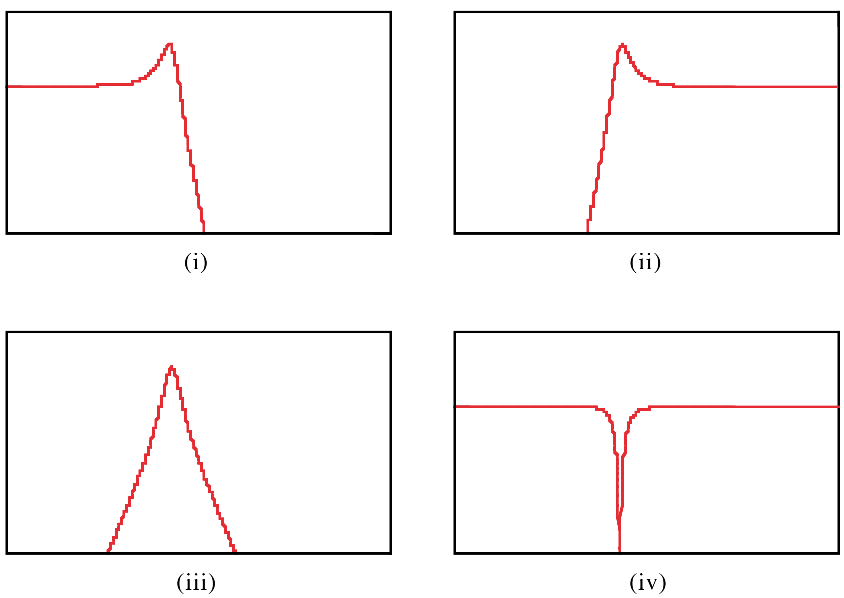


Figure 139 Resonant Filter Types: (i) lowpass, (ii) highpass, (iii) bandpass, and (iv) notch.

The resonant frequency of the filter will remain at the minimum frequency (Min Freq) as long as the signal envelope is below the Threshold. The Freq Sweep parameter controls how much the frequency will change with changes in envelope amplitude. The frequency range is 0 to 8372 Hz, though the minimum setting for Min Freq is 58 Hz. Note that the term minimum frequency is a reference to the resonant frequency at the minimum envelope level; with a negative Freq Sweep, the filter frequency will sweep below the Min Freq. A meter is provided to show the current resonance frequency of the filter.

The filter Resonance level may be adjusted. The resonance is expressed in decibels (dB) of gain at the resonant frequency. Since 50 dB of gain is available, you will have to be careful with your gain stages to avoid clipping.

The attack and release rates of the filter envelope follower are adjustable. The rates are expressed in decibels per second (dB/s). The envelope may be smoothed by a low pass filter, which can extend the attack and release times of the envelope follower. A level meter with a threshold marker is provided.

The selected input signal (left, right, or sum) may trigger an ADSR envelope (Attack, Decay, Sustain, Release), an ASR envelope, or create its own envelope for controlling the generated waveform amplitude. **ADSR Synth** has full ADSR control. The **Gate Synth** algorithm is slightly smaller and uses a gate circuit to control ASR. Finally **Env Synth** uses an input signal envelope to control the envelope of the output waveform.

Parameters:

Page 1

Wet/Dry	0 to 100%wet	Out Gain	Off, -79.0 to 24.0 dB
Pad Gain	Off, -79.0 to 24.0 dB	Filt SC In	L, R, (L+R)/2
		ADSR SC In	L, R, (L+R)/2
		Pan	-100 to 100%

Page 2

Wave Freq	8 to 25088 Hz	WaveXPose	-48 to 48 ST
Wave Shape	Sine, ...		
WavePlsWid	0 to 100%		
WaveSmooth	0 to 100%		

Page 3

FilterType	Lowpass	Filt Thres	-79.0 to 0.0 dB
FiltMinFrq	8 to 8372 Hz	FltAtkRate	0.0 to 1000.0 dB/s
Freq Sweep	-100 to 100%	FltRelRate	0.0 to 1000.0 dB/s
Filt Res	0 to 50 dB	FiltSmooth	0.0 to 1000.0 dB/s
E		F	
60 40 * 16 * 8 dB		0Hz 2k 4k 6k	

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

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

Pad Gain A gain or amplitude applied to the wet signal only. Pad Gain may be used to match wet and dry signal levels across the range of the Wet/Dry parameter.

Filt SC In The filter envelope follower may use the left input channel, the right input channel or the sum of both as the envelope-generating signal.

ADSR SC In, **Env SC In**, or **Gate SC In**

The amplitude envelope generator may use the left input channel, the right input channel or the sum of both as the envelope-generating signal.

Wave Freq The fundamental frequency of the waveform oscillator. The oscillator can be set through the audible frequencies 8-25088 Hz with one-semitone resolution.

Wave Shape Shape selects the waveform type for the waveform oscillator. Choices are **Sine**, **Saw+**, **Saw–**, **Pulse**, **Tri**, and **Expon**.

WavePlsWid When the configurable oscillator is set to **Pulse**, the PlsWid parameter sets the pulse width as a percentage of the waveform period. The pulse is a square wave when the width is set to **50%**. This parameter is active only when the Pulse waveform is selected.

WaveSmooth Smooths the **Saw+**, **Saw–**, and **Pulse** waveforms. For the sawtooth waves, smoothing makes the waveform more like a triangle wave. For the **Pulse** wave, smoothing makes the waveform more like a sine wave.

WaveXPose The specified wave frequency may be transposed in one-semitone increments.

FilterType The type of resonant filter to be used. May be one of "Lowpass", "Highpass", "Bandpass", or "Notch".

FiltMinFrq The base frequency of the resonant filter. The filter resonant frequency is set to the Min Freq while the signal envelope is at its minimum level or below the threshold.

Freq Sweep How far the filter frequency can change from the Min Freq setting as the envelope amplitude changes. Freq Sweep may be positive or negative, so the filter frequency can rise above or fall below the Min Freq setting.

Filt Res The resonance level of the resonant filter. Resonance sets the level of the resonant peak (or the amount of cut in the case of the notch filter).

Filt Thres Represents the level above which signal envelope must rise before the filter begins to follow the envelope. Below the threshold, the filter resonant frequency will remain at the Min frequency.

FiltAtkRate Adjusts the upward slew rate of the filter envelope detector.

FiltRelRate Adjusts the downward slew rate of the filter envelope detector.

FiltSmooth Smooths the output of the filter envelope follower. Smoothing slows down the envelope follower and can dominate the attack and release rates if set to a lower rate than either of these parameters.

ADSR Thres or **Gate Thres**

The input signal level, which triggers the ADSR circuit or the ASR (gate) circuit.

ADSR Atk or **Gate Atk**

The time for the ADSR or ASR (gate) envelope to rise from off (no signal) to its maximum level.

ADSR Decay The time for the ADSR envelope to decay from the maximum signal level to the sustain level.

ADSRSusLvl The sustain signal level as a percentage of the maximum attack level. (This is fixed at 100% in the **Gate Synth** algorithm.)

ADSR Sust or **Gate Time**

The sustain time for the ADSR and ASR (gate) envelopes.

ADSR Rel or **Gate Rel**

The time for the ADSR and ASR (gate) envelopes to release from the sustain level down to off (no signal).

EnvAtkRate For **Env Synth**, the rate in decibels per second for the envelope to rise in response to a rise in the input signal level.

EnvRelRate For **Env Synth**, the rate in decibels per second for the envelope to fall in response to a drop in the input signal level.

EnvSmooth For **Env Synth**, the rate in decibels per second for the envelope to rise or fall in response to a rise or fall in the input signal level. The slower of the EnvSmooth rate or the attack/release rate will dominate the envelope responsiveness.

Combination Algorithms

400 Chorus+Delay
401 Chorus+4Tap
408 StChor+Dly+RvrbL
409 Pitcher+Chor+Dly
450 Flange+Delay
451 Flange+4Tap
458 StFlan+Dly+RvrbL
459 Pitcher+Flan+Dly

A family of combination effect algorithms (combination indicated by “+”)

PAUs: 1 or 2

Signal Routing (algorithms containing 2 effects)

The algorithms listed above with 2 effects can be arranged in series or parallel. Effect A and B are respectively designated as the first and second listed effects in the algorithm name. The output of effect A is wired to the input of 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. where A is effect A, and B is effect 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 (putting the effects in parallel), and **100**% means only the output of effect A is fed into effect B (putting the effects in series). See Figure 140 below for the signal flow of **Chorus+4Tap** as an example.

Both effect A and B outputs are mixed at the algorithm output to become the wet signal. These mix levels are controlled with the 2 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.

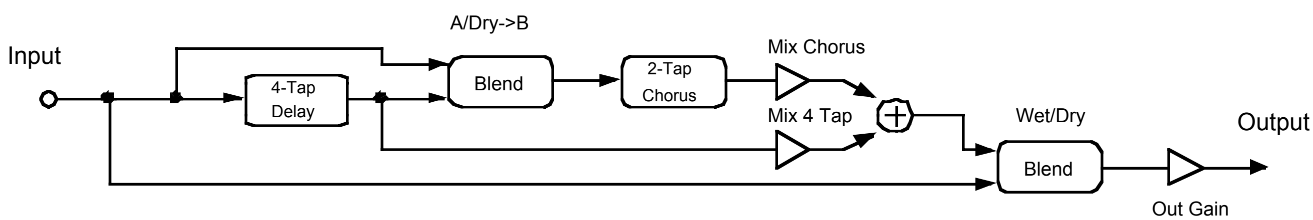


Figure 140 An example of routing using Chorus+4Tap

Parameters for Two-effect Routing

Page 1

Wet/Dry	-100 to 100 %	Out Gain	Off; -79.0 to 24.0 dB
Mix *Effect A*	-100 to 100 %		
Mix *Effect B*	-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 signal 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 (algorithms with 3 effects)

The algorithms listed above with 3 effects allow serial or parallel routing between any two effects. Effects A, B, and C are designated respectively by their order in the algorithm name. 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 with the two-effect routing, the input of effect B is controlled by a parameter A/Dry>B. where A is effect A, and B is effect B. For example, in **StChor+Dly+RvrbL**, 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 * (asterisk). 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 denominator, while **100%** mixes only the numerator. Negative values polarity invert the signal associated with the numerator.

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.

Parameters for Three-effect Routing

Page 1

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

Page 2

A/Dry>B	-100 to 100 %	A/Dry>B	-100 to 100 %
A/B -> ✻	-100 to 100 %	A/B -> ✻	-100 to 100 %

Mix Effect Left and Right. 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 what 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 ✻ (asterisk). **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 what 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

Choruses

The choruses are basic 1-tap dual choruses. Separate LFO controls are provided for each channel. Slight variations between algorithms may 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 **Choruses** section (**page 98**).

Parameters for Choruses

Page 1

Ch PtchEnv	Triangle or Trapzoid	Ch HF Damp	8 to 25088 Hz
Ch Rate	0.01 to 10.00 Hz	L/R Phase	0.0 to 360.0 deg
Ch Depth	0.0 to 100 ct	L/R DpthDif	-100 to 100%
Ch Delay	0.0 to 360.0 ms	L/R DlyDiff	-100 to 100%
Ch Fdbk	-100 to 100 %		
Ch Xcouple	0 to 100 %		

Ch Fdbk This controls the amount that the output of the chorus is fed back into the input.

All Other Parameters Refer to **Choruses** documentation.

Flangers

The flangers are basic 1-tap dual flangers. Separate LFO controls are provided for each channel. Slight variations between algorithms may 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.

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.

Parameters for Flangers

Page 1

Fl Tempo	System; 1 to 255 BPM	Fl HF Damp	8 to 25088 Hz
Fl Rate	0.01 to 10.00 Hz		
Fl Xcurs	0 to 230 ms	Fl Xcouple	0 to 100 %
Fl Delay	0 to 230 ms	Fl StatDly	0 to 230 ms
Fl Fdbk	-100 to 100 %	Fl StatLvl	-100 to 100 %
Fl Phase L	0 to 360 deg	Fl Phase R	0 to 360 deg

Fl Phase Left and Right. These adjust the corresponding LFO phase relationships between themselves and the internal beat clock.

Fl StatDly Sets the delay time for the non-moving delay tap for through-zero flange effects.

Fl StatLvl Adjusts the mix amount for the static tap. Negative values polarity invert the static tap signal.

All other parameters Refer to **Flangers** documentation (**page 104**). Parameters with a 1 or 2 correspond to LFO taps organized as described above.

Delays

Delay (sometimes **Dly**) 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, and laser controls. Parameters associated with delay 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. Separate left and right feedback control is generally provided, but due to resource allocation, some delays in combinations may have a single control for both channels.

Dly FBImag 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 due to resource allocation.

Parameters for Delays

Page 1

Dly Time L	0 to 32 bts	Dly Tempo	System, 1 to 255 bpm
Dly Fdbk L	-100 to 100 %	Dly Time R	0 to 32 bts
Dly HFDamp	0 to 32 bts	Dly Fdbk R	-100 to 100 %
		Dly Imag	-100 to 100 %

Dly Tempo The basis for the moving delay lines 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. In this case, FX Mods (FUNs, LFOs, ASRs, etc.) will have no effect on the Tempo parameter.

Dly Time Left and Right. 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 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 FBImag 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 (**150**, **page 64**), with the exception that the delay times will pin at the maximum delay time instead of automatically cutting their times in half.

Parameters for Combination 4-Tap

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 %

Reverbs

The reverbs offered in these combination effects is **MiniVerb**. Information about it can be found in the **MiniVerb** documentation. Parameters associated with this reverb begin with Rv.

MiniVerb

		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	8 to 25088 Hz
Rv PreDlyL	0 to 620 ms	Rv PreDlyR	0 to 620 ms

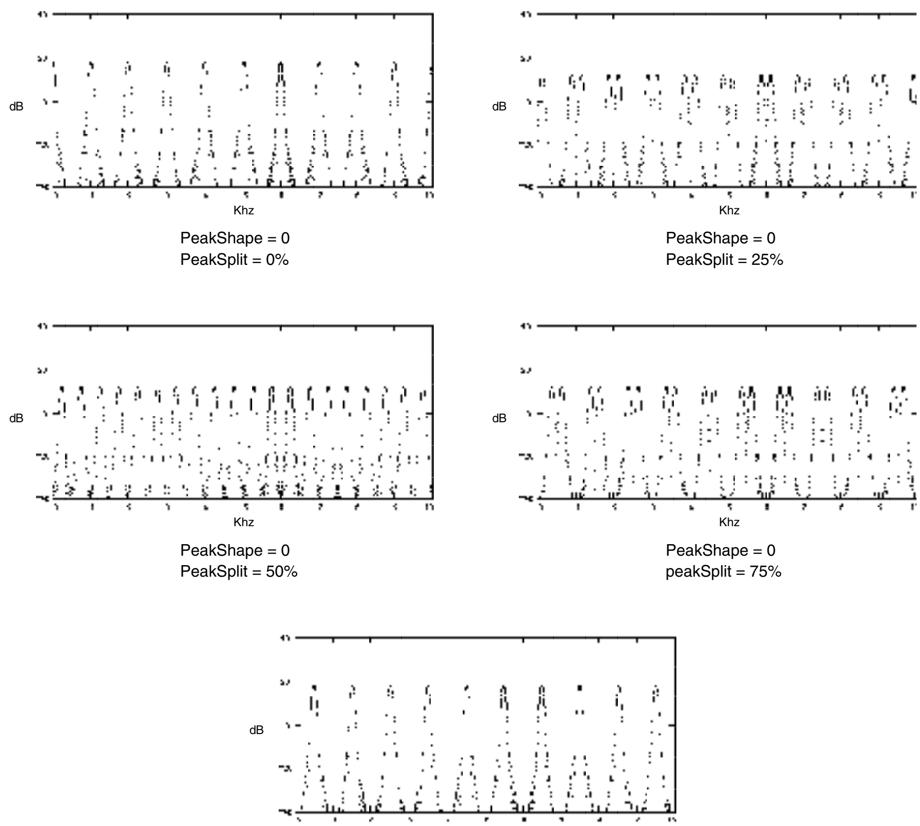
411 MonoPitcher+Chor
461 MonoPitcher+Flan

Monaural pitching algorithm (filter with harmonically related resonant peaks) with chorus or flanger

PAUs: 2 each

These algorithms each apply a filter that 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 graphs below show Pt PkSplit going from **0**% to **100**%, for a Pt Pitch of **1 kHz** (roughly. C6), and Pt PkShape set to **0**.



PeakShape = 0
PeakSplit = 0%

PeakShape = 0
PeakSplit = 25%

PeakShape = 0
PeakSplit = 50%

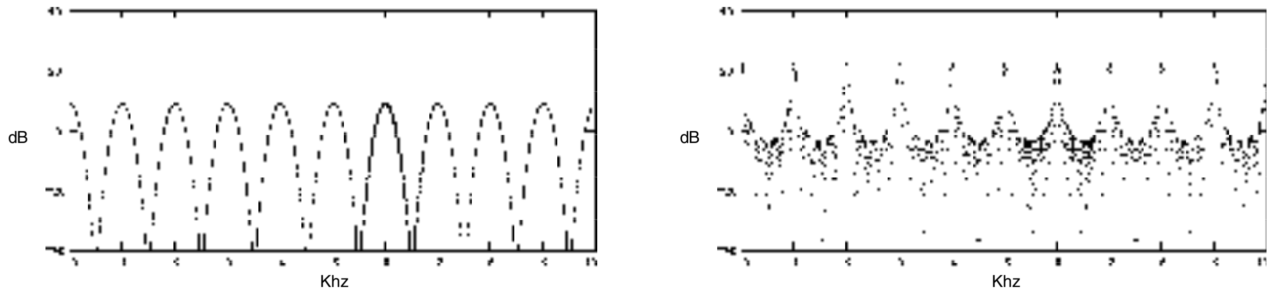
PeakShape = 0
peakSplit = 75%

PeakShape = 0
PeakSplit = 100%

Figure 141 Response of Pitcher with different PkSplit settings. Pitch is C6 and PkShape is 0.

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** and **1.0**, for a Pitch of **C6** and a PkSplit of **0**%.



PeakShape = 1.0
PeakSplit = 0%

PeakShape = -1.0
PeakSplit = 0%

Figure 142 Response of Pitcher with different PkShape settings

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.

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.

Configurable Flanger

The flange in Algorithm **461** is a configurable flange. See **Configurable Combination Algorithms on page 289** for details about this effect.

Chorus

The chorus used in Algorithm **411** is a basic dual channel chorus. See **Choruses on page 98** for more information on the effect.

Parameters for 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 2

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

Page 3

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	8 to 25088 Hz

Parameters for 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 2

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

Page 3

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	8 to 25088 Hz

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

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

Mix Pitchr 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.

Mix Chorus, Mix Flange	The amount of the flanger or chorus signal to send to the output as a percent.
Pt/Dry->Ch, Pt/Dry->Fl	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**.
Pt Inp Bal	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.
Pt Out Pan	Pans the **MonoPitcher+Chor** or **MonoPitcher+Flan** output from left (**-100**%) to center (**0**%) to right (**100**%).
Pt Pitch	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.
Pt PkSplit	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.
Pt Offset	An offset in semitones from the frequency specified in Pitch.
Pt PkShape	Controls the shape of the pitcher spectral peaks. **0.0** gives the most "pitchiness" to the output, in that the peaks are narrow, with not much energy between them. **-1.0** makes the peaks wider. **1.0** brings up the level between the peaks.
All other Chorus parameters	See **Choruses on page 98**.
Fl LFO cfg	Sets the user interface mode for controlling each of the four 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 corresponding LFO phase relationships between themselves and the internal beat clock.
All other Flange parameters	See **Flangers on page 104**. Parameters with a 1 or 2 correspond to LFO taps organized as described above.

Configurable Combination Algorithms

105 LasrDly<>Reverb
321 Flange<>Shaper
322 Shaper<>Reverb
384 Flange<>Pitcher
402 Chorus<>4Tap
404 Chorus<>Reverb
405 Chorus<>LasrDly
452 Flange<>4Tap
454 Flange<>Reverb
455 Flange<>LasrDly

A family of configurable combination effect algorithms (configurability indicated by "<>")

PAUs: 2

Signal Routing

Each of these combination algorithms offers two separate effects combined with flexible signal routing mechanism. This mechanism allows the two effects to be either in series bidirectionally 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 **Chorus<>4Tap** algorithm, see **Figure 143**.

Both effect A and B outputs are mixed at the algorithm output to become the wet signal. These mix levels are controlled with the 2 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.

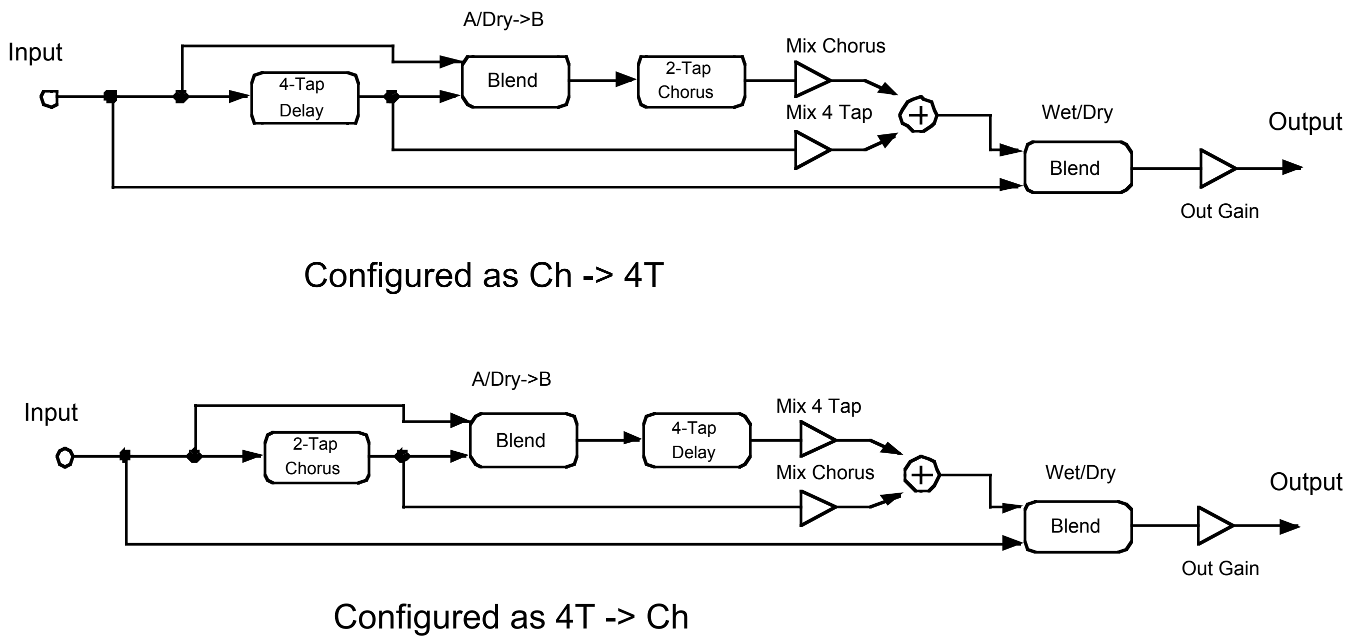


Figure 143 Chorus<>4Tap with A->B cfg set to Ch->4T and 4T->Ch

Bidirectional 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 %

Mix Effect Adjusts the amount of each effect is mixed together as the algorithm wet signal. Negative values polarity invert that particular signal.

A->B cfg 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.

A/Dry->B This parameter controls how much of the A effect is mixed with dry and fed into effect B. 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 two 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 **Choruses** or **Flangers** sections.

Since these effects have 2 taps per channel, control over 4 LFOs is necessary, but with a minimum number of user parameters (**Figure 144**). This is accomplished by offering 2 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.

In Dual1Tap mode (**Figure 145**), 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 (**Figure 146**), Control Set 1 controls 1 tap in both the left and right channels. Control Set 2 has no affect, 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 (**Figure 147**), Control Set 1 controls the first left and right pair of LFOs, while Control Set 2 controls the second pair. This mode uses all four LFOs for a richer sound, and is optimized for LRPhase relationships. Each of the two 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.

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 be setting them to negative values.

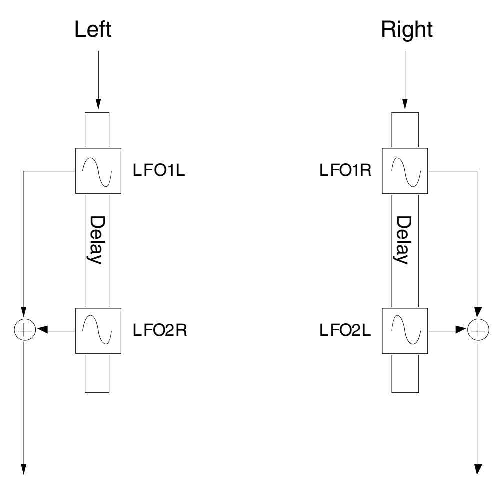


Figure 144 LFO delay taps in the configurable chorus and flange

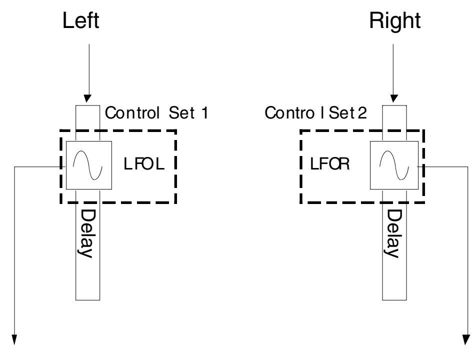


Figure 145 LFO control in Dual1Tap mode

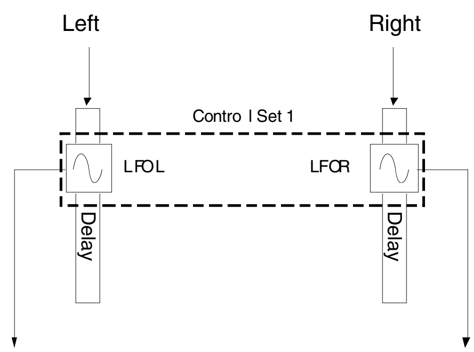


Figure 146 LFO control in Link1Tap mode

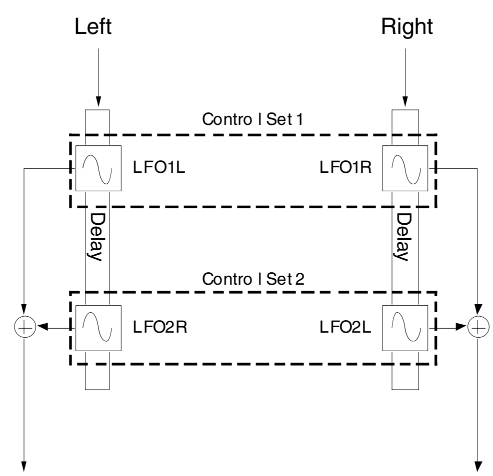


Figure 147 LFO control in Link2Tap mode

Parameters for Choruses

Page 1

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	8 to 25088 Hz

Parameters for Flangers

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

Page 2

Fl HF Damp	8 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 four 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.

All other Chorus parameters See **Choruses on page 98**.

Fl LFO cfg Sets the user interface mode for controlling each of the four 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 corresponding LFO phase relationships between themselves and the internal beat clock.

All other Flange parameters	See **Flangers on page 104**. Parameters with a 1 or 2 correspond to LFO taps organized as described above.

Laser Delay

Laser Delay (LasrDly) 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 **LaserVerb** 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**, 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 polarity inverting the signal. The LsrCntour parameter adds only the Laser Delay portion of the effect, including its 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. Due to resource allocation limits, not all Laser Delay**s** in combination algorithms will have all four of these parameters.

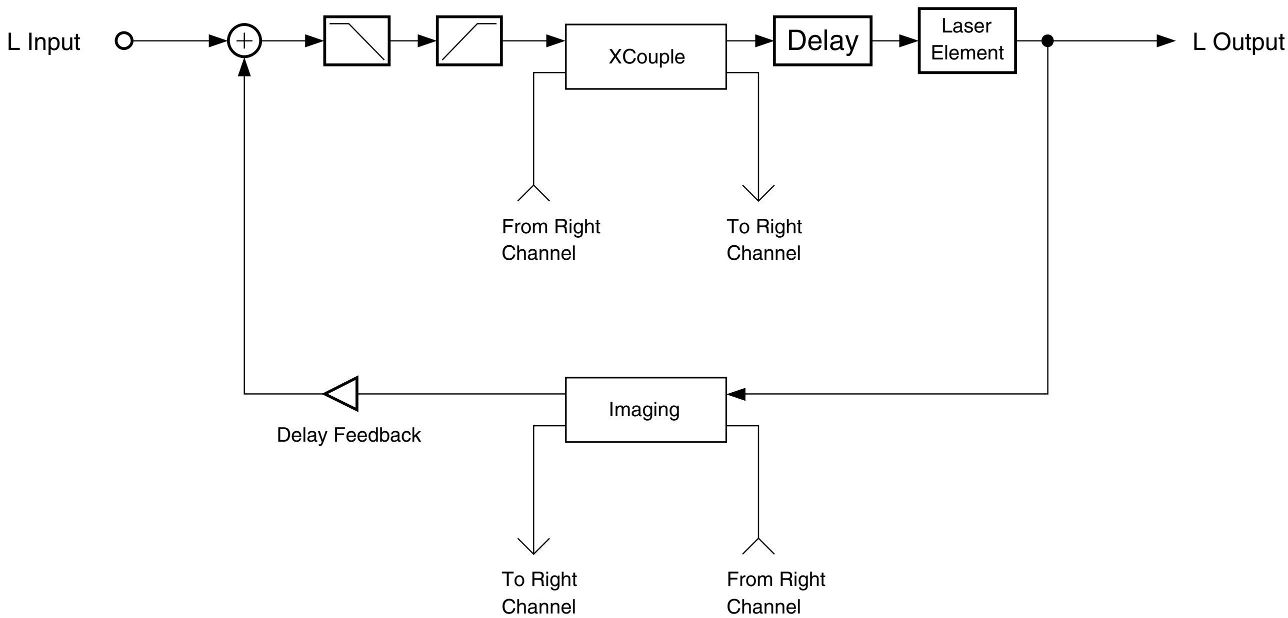


Figure 148 Laser Delay (left channel)

Parameters for Laser Delay

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 Left and Right. 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 Left and Right. 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) highpass 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.

LsrCntour Left and Right. 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 **0**, the laser portion is turned off turning regeneration into straight feedback.

LsrSpace Left and Right. Determines the starting pitch of the descending resonance and how fast it descends. See the **Laser Delay on page 294** 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 (**150**, **page 64**) 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 highpass filter, and/or a lowpass filter.

Parameters for Combination 4-Tap

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	8 to 25088 Hz
4T LF Damp	8 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 %

4T FB Imag 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.

4T FB Xcpl 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."

All other parameters Refer to documentation for **150 4-Tap Delay BPM**.

Reverbs

The reverbs offered in these combination effects are based on **MiniVerb**. Information about it can be found in the **MiniVerb** documentation (**page 9**). Parameters associated with this reverb begin with Rv.

MiniVerb

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	8 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 stand alone version. Review the **Pitcher** section (**page 256**) for more information. Parameters associated with this effect begin with Pt.

Parameters for Pitcher

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 sonic qualities as those found in V.A.S.T. Refer to the appendices in the KSP8 *User's 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.

Parameters for Shaper

Shp Inp LP	8 to 25088 Hz
Shp Amt	0.10 to 6.00 x
Shp Out LP	8 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 V.A.S.T.

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

More Combination Algorithms

406 St Chorus+Delay
407 St Chorus+4Tap
408 StChor+Dly+RvrbL
410 Pitch+StChor+Dly
412 MonoPitch+StChor
420 Chorus+Delay ms
421 Chorus+4Tap ms
422 Chorus<>4Tap ms
423 Chor+Dly+Rvrb ms
425 Chor<>LasrDly ms
426 St Chor+Delay ms
427 St Chor+4Tap ms
428 StCh+Dly+Rvrb ms
429 Ptch+Chor+Dly ms
430 Ptch+StCh+Dly ms
456 St Flange+Delay
457 St Flange+4Tap
458 StFlan+Dly+RvrbL
460 Pitch+StFlan+Dly
470 Flange+Delay ms
471 Flange+4Tap ms
472 Flange<>4Tap ms
473 Flan+Dly+Rvrb ms
475 Flan<>LasrDly ms
476 St Flan+Delay ms
477 St Flan+4Tap ms
478 StFl+Dly+Rvrb ms
479 Ptch+Flan+Dly ms
480 Ptch+StFl+Dly ms

Combination effect algorithms using time/frequency units instead of tempo

PAUs: 1 or 2

The algorithms listed here are identical in most respects to combination effects elsewhere documented. For example, Algorithm **420 Chorus+Delay ms** is closely based on Algorithm **400 Chorus+Delay**. The difference with algorithms with "ms" at the end of their names is that they do not use tempo features for setting delay line lengths or LFO rates. Instead, delay line lengths are set in units of milliseconds (ms) for time, and LFO rates are set in units of Hertz (Hz) for frequency. The difference for algorithms with "St" in the name is that they use stereo controls (ganged controls) rather than dual mono controls for the chorus and flange components of the algorithms.

For full details on using these algorithms use the following links to jump to the appropriate section.

Combination Algorithms on page 279

Configurable Combination Algorithms on page 289

More Combination Algorithms on page 299

498 FXMod Diagnostic

FXMod source metering utility algorithm

PAUs: 1

The FXMod diagnostic algorithm is used to obtain a metered display of FXMod sources. This algorithm allows you to view the current levels of any data sliders, MIDI controls, switches, or internally generated V.A.S.T. LFOs, ASRs, FUNs, etc. which are available as modulation sources. This algorithm has no effect on any signal being routed through it.

Up to eight modulation sources may be monitored simultaneously. Meters 1–4 can monitor bipolar sources, meaning sources which can have both positive and negative values. The range of the bipolar meters is -1 to +1. Four monopolar meters 5–8 provide better resolution, but the range is limited to 0 to +1. Use the monopolar meters for sources which you do not expect to go negative.

Eight parameters are provided to connect modulation sources to the meters. The parameter values are fixed at **NoDpth** and have no function except to connect sources to meters. You will not be able to modify the Adjust or Depth fields, but you can select any source you want. Finally press the **Edit** button to reenter the Studio and Preset editor where you can view the meters on page 2 of the parameters.

Parameters

Page 1

Bipole 1	NoDpth	Monopole 5	NoDpth
Bipole 2	NoDpth	Monopole 6	NoDpth
Bipole 3	NoDpth	Monopole 7	NoDpth
Bipole 4	NoDpth	Monopole 8	NoDpth

Page 2

1	5
2	6
-1 0 1	0 0.5 1
3	7
4	8

Bipole *n* Use the Bipole parameters to attach bipolar modulation sources (can go positive or negative) to the bipolar meters. The parameters are not adjustable.

Monopole *n* Use the Monopole parameters to attach monopolar modulation sources (can go positive only) to the monopolar meters. The parameters are not adjustable.

499 Stereo Analyze
699 Mn Analyze

Signal metering and channel summation utility algorithm

PAUs: 1

Stereo Analyze is a utility algorithm that provides metering of stereo signals as its primary function. In addition to metering, the gains of the two channels are separately controllable, either channel may be inverted, and sum and differences to the two channels may be metered and monitored. If you use this algorithm with Live Mode on a K2500 or K2600, you can obtain a significant amount of information not only about your own mix, but of any recording you have in your library.

There are separate meters for the left and right output channels. Two types of meters are provided: peak and RMS. Meter display units are decibels relative to digital full scale (dBFS). The peak meters display the levels of the maximum signal peak that occurred during the meter update period (every 40ms). The RMS meter displays the average power of the input signal. RMS is an abbreviation for root-mean-square, so the signal is squared, averaged and a square root is taken. For a real-time meter, we do not take an average over all time, but rather average past signals with a stronger weighting to signals in the recent past than the far past. The RMS Settle parameter controls how strong the weighting is for recent signals over much older signals. RMS Settle is expressed in units of dB/s (decibels per second), meaning how fast the RMS meter can rise or fall with changing signal levels.

You can choose to meter and monitor normal left (L) and right (R) stereo signals, or with the Out Mode parameters, you can select normalized sum and differences of the left and right channels. The Out Mode parameters control the signals being passed to the outputs and to the meters: what you see on the meters are the signals to which you are listening. The Invert parameters provide a quick polarity reversal to the input signals. This polarity reversal occurs before sum and differences. The Invert parameters are actually redundant since Out Mode provides signal inversions as well. The left and right Out Mode parameters may be set to any of the following:

L	left channel
R	right channel
(L+R)/2	normalized sum of left and right
(L-R)/2	normalized difference of left minus right
-L	polarity reversed left channel
-R	polarity reversed right channel
-(L+R)/2	polarity reversed and normalized sum of left and right
(R-L)/2	normalized difference of right minus left

You may well ask why you would want to meter or monitor reversals or sums or differences of your stereo channels. One important case is to determine if your final mix is mono compatible—very important if your mix is ever going to be broadcast on radio or television. Set both the left and right Out Mode parameters to **(L+R)/2** to listen to the mono signal. If you find that parts of your mix disappear or start to sound metallic (comb filtered), you may have to go back and do some work on your mix.

The difference signal (L-R)/2 provides a measure of the stereo content of your mix and can be very indicative of mixing style. Listening to the difference signal of someone else's recordings can often demonstrate interesting techniques (and mistakes!) in stereo production. The difference signal contains everything that doesn't make it into the mono mix. Out of phase signals will appear only in the difference signal. Panned signals will appear in both the sum and difference signals to varying degrees. A delay between left and right channels will sound metallic (comb filtered or flanged) in both the sum and difference channels. If the entire mix seems to have a relative left/right delay, you can use the L/R Delay parameter to attempt to correct the problem. Positive delays are delaying the left channel, while negative delays are delaying the right channel.

By inverting one channel with respect to the other, you can hear what is characterized as “phasey-ness.” Usually in stereo recordings, you can localize the phantom image of sound sources somewhere between the two loudspeakers. With a phasey signal, the localization cue get mixed up and you may hear the sound coming from everywhere or within your head. Polarity reversals are provided in this algorithm so you can test for mistakes, or simply for experimentation.

Parameters

Page 1

L In Gain	Off, -79.0 to 24.0 dB	R In Gain	Off, -79.0 to 24.0 dB
L Invert	In or Out	R Invert	In or Out
L Out Mode	L	R Out Mode	R
L/R Delay	-500.0 to 500.0 samp	RMS Settle	0.0 to 300.0 dB/s

Page 2

Peak (-dBFS)			
L		R	
55 40 * 16 8 4 0		55 40 * 16 8 4 0	
L		R	
RMS (-dBFS)			

L In Gain The input gain of the left channel in decibels (dB).

R In Gain The input gain of the right channel in decibels (dB).

L Invert When set to on, the polarity of the left channel is reversed.

R Invert When set to on, the polarity of the right channel is reversed.

L Out Mode Determines which signal is to be metered (left meter) and passed to the left output. Choices are **L** (left), **R** (right), **(L+R)/2** (normalized sum), **(L-R)/2** (normalized difference), and polarity inverted versions of these.

R Out Mode Determines which signal is to be metered (right meter) and passed to the right output. Choices are **L** (left), **R** (right), **(L+R)/2** (normalized sum), **(L-R)/2** (normalized difference), and polarity inverted versions of these.

L/R Delay If this parameter is positive, the left signal is delayed by the indicated amount. If it is negative, the right channel is delayed. You can use this parameter to try to improve cancellation of the difference signal if you suspect one channel is delayed with respect to the other.

RMS Settle RMS Settle controls how fast the RMS meters can rise or fall with changing signal levels. Units are decibels per second (dB/s).