# FXAlgs #724-6, 728: Distortion

## FXAlg #724 — Mono Distortion
## FXAlg #725 — MonoDistort + Cab
## FXAlg #726 — MonoDistort + EQ
## FXAlg #728 — StereoDistort+EQ

### Small distortion algorithms

*Allocation Units:* 1 for Mono Distortion; 2 for MonoDistort + Cab; 2 for MonoDistort + EQ; 3 for StereoDistort + EQ

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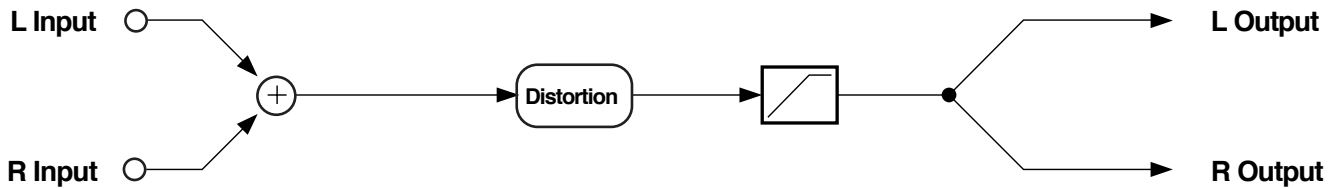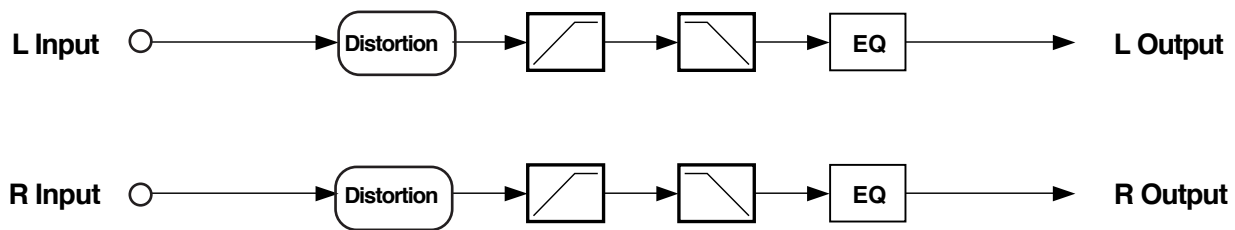


**Block diagram of Mono Distortion**

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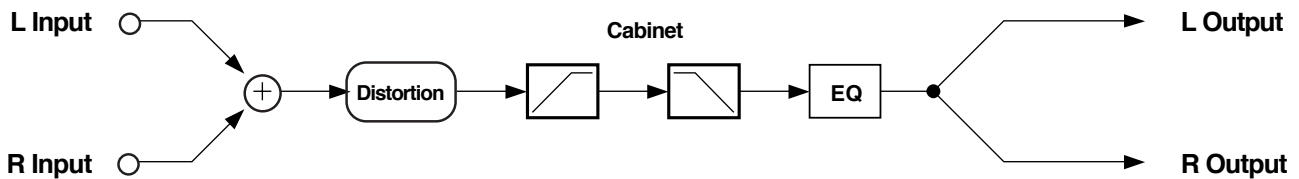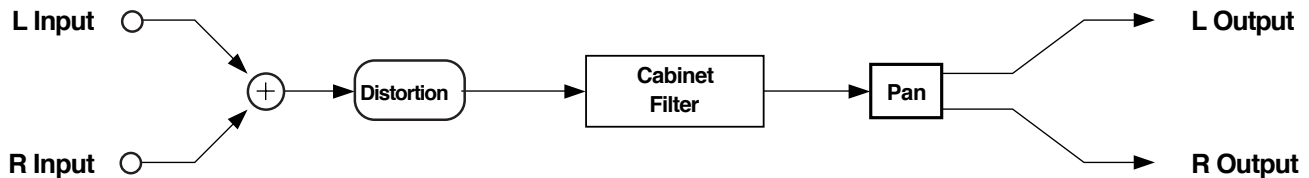


**Block diagram of MonoDistort + EQ**

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 one parametric mid filter.



**Block diagram of StereoDistort+EQ**

MonoDistort + Cab is also similar to Mono Distortion except the hipass is replaced by a full speaker cabinet model. 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 eight presets which were derived from measurements of real cabinets. (See descriptions of FXAlgs #729-732 in this book for more information.)



**Block diagram of MonoDistort + Cab**

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 algorithms will not digitally clip unless the output gain is over-driven.
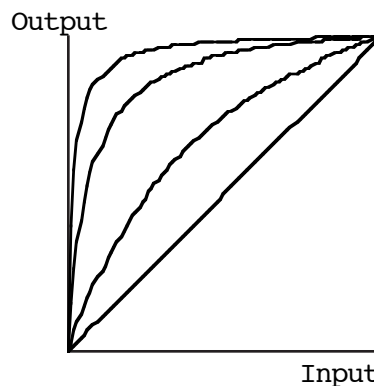


**Input/Output Transfer Characteristic of Soft Clipping at Various Drive Settings**

Signals which 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 the 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 - Mono Distortion:

PAGE 1

| Wet/Dry | 0 to 100%wet | Out Gain | Off, -79.0 to 24.0 dB |
|---------|--------------|----------|----------------------|
| Dist Drive | 0 to 96 dB | | |
| Warmth | 16 to 25088 Hz | | |
| Highpass | 16 to 25088 Hz | | |

## MonoDistort + Cab:

PAGE 1

| Wet/Dry | 0 to 100%wet | Out Gain | Off, -79.0 to 24.0 dB |
|---------|--------------|----------|----------------------|
| Dist Drive | 0 to 96 dB | | |
| Warmth | 16 to 25088 Hz | Cab Bypass | In or Out |
| | | Cab Preset | Basic |

## 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 | 16 to 25088 Hz | dc Offset | -100 to 100% |
| Cabinet HP | 16 to 25088 Hz | Cabinet LP | 16 to 25088 Hz |

PAGE 2

| Bass Gain | -79.0 to 24.0 dB | Treb Gain | -79.0 to 24.0 dB |
|-----------|------------------|-----------|------------------|
| Bass Freq | 16 to 25088 Hz | Treb Freq | 16 to 25088 Hz |
| Mid1 Gain | -79.0 to 24.0 dB | Mid2 Gain | -79.0 to 24.0 dB |
| Mid1 Freq | 16 to 25088 Hz | Mid2 Freq | 16 to 25088 Hz |
| Mid1 Width | 0.010 to 5.000 oct | Mid2 Width | 0.010 to 5.000 oct |

## 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 | | |
| Warmth | 16 to 25088 Hz | | |
| Cabinet HP | 16 to 25088 Hz | Cabinet LP | 16 to 25088 Hz |

**PAGE 2**

| Bass Gain | -79.0 to 24.0 dB | Treb Gain | -79.0 to 24.0 dB |
|-----------|------------------|-----------|------------------|
| Bass Freq | 16 to 25088 Hz | Treb Freq | 16 to 25088 Hz |
| Mid Gain | -79.0 to 24.0 dB | | |
| Mid Freq | 16 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. For distortion, it is often necessary to turn the output gain down as the distortion drive is turned up.

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.

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       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. [MonoDistort + Cab]

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. See description of FX Algs #729-732 for more information. [MonoDistort + Cab]

Highpass         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. [Mono Distortion]

## MonoDistort + EQ and StereoDistort+EQ

Cabinet HP       A hipass filter which controls the low-frequency limit of a simulated loudspeaker cabinet.

Cabinet LP       A lowpass filter which controls the high-frequency limit of a simulated cabinet.

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 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.

# FXAlg #727: PolyDistort + EQ

### Eight-stage distortion followed by equalization
*Allocation Units:* 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 LP filter. There is also a one pole LP in front of the first stage. After the distortion there is a 4-band EQ section: Bass, Treble, and two Parametric Mids.
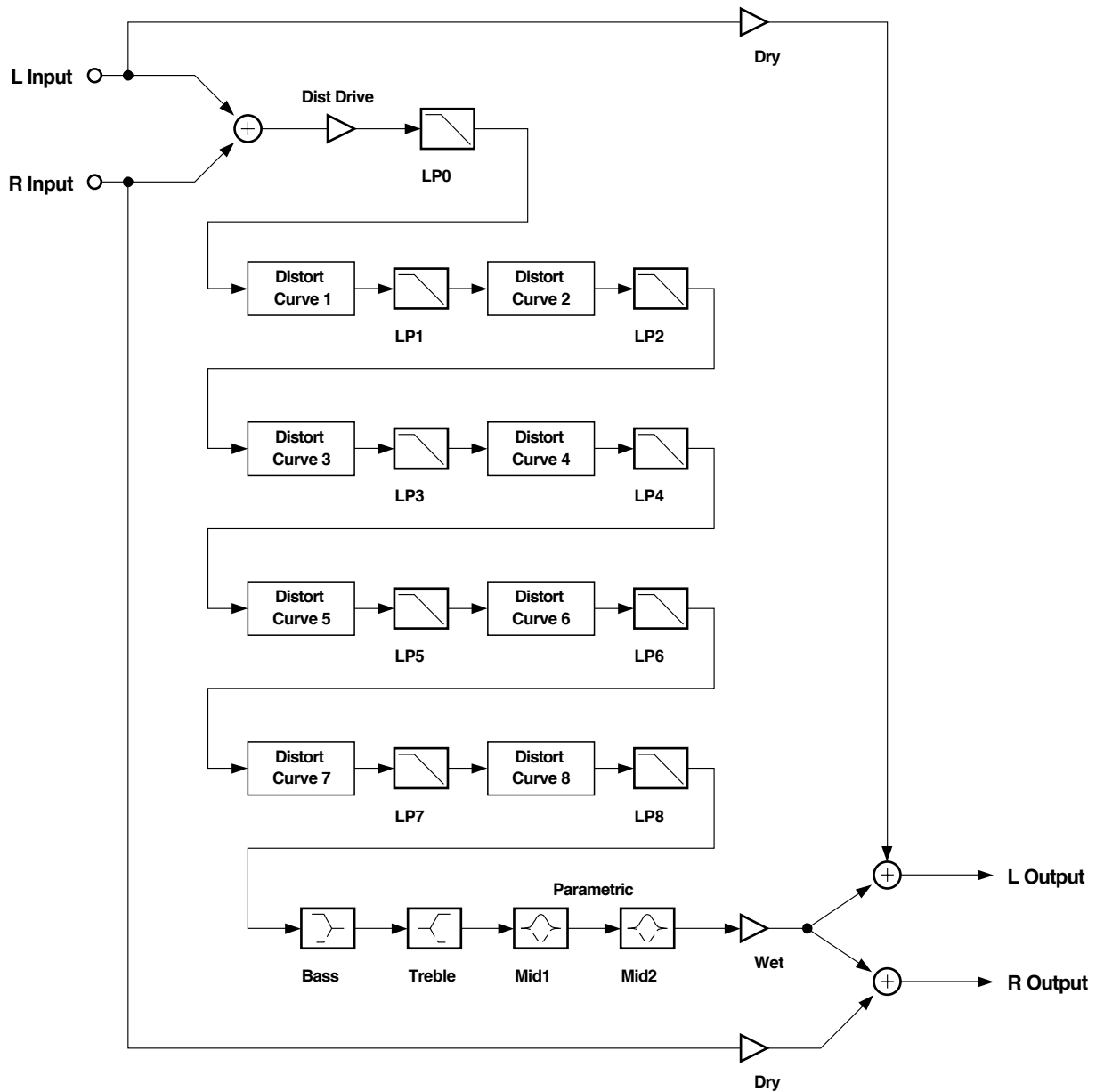
**Block diagram of PolyDistort + EQ**

PolyDistort is an unusual distortion algorithm which 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 the sound. Taken together, they can produce a very harsh heavy metal sound. Between each distortion stage is a lopass filter. The lopass filters work with the distortion stages to help mellow out the sound. Without any lopass 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 lopasses should be turned off if there is no distortion in a section. More than 4 stages seem necessary for lead guitar sounds. For a cleaner sound, you may want to limit yourself to only 4 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 to 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 lopass 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 speakers 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 KDFX Studio INPUT pages, 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 tiring after a while.

## 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 | | |

**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 2**

| LP0 Freq | 16 to 25088 Hz | | |
|---|---|---|---|
| LP1 Freq | 16 to 25088 Hz | LP5 Freq | 16 to 25088 Hz |
| LP2 Freq | 16 to 25088 Hz | LP6 Freq | 16 to 25088 Hz |
| LP3 Freq | 16 to 25088 Hz | LP7 Freq | 16 to 25088 Hz |
| LP4 Freq | 16 to 25088 Hz | LP8 Freq | 16 to 25088 Hz |

**PAGE 4**

| Bass Gain | -79.0 to 24.0 dB | Treb Gain | -79.0 to 24.0 dB |
|-----------|------------------|-----------|------------------|
| Bass Freq | 16 to 25088 Hz | Treb Freq | 16 to 25088 Hz |
| Mid1 Gain | -79.0 to 24.0 dB | Mid2 Gain | -79.0 to 24.0 dB |
| Mid1 Freq | 16 to 25088 Hz | Mid2 Freq | 16 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. For distortion, it is often necessary to turn the output gain down as the distortion drive is turned up. |
| 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. |
| 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 lopass controls. LP0 Freq handles the initial lopass prior to the first distortion stage. The other lopass controls follow their respective distortion stages. With all lopasses 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 of the lopass 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. The bandwidth is specified in octaves. Small values result in a very narrow filter response. Large values result in a very broad response. |

# Tube Amp/Distortion/Delay Combinations

**FXAlg #729: TubeAmp<>MD>Chor**
**FXAlg #730: TubeAmp<>MD>Flan**
**FXAlg #731: PolyAmp<>MD>Chor**
**FXAlg #732: PolyAmp<>MD>Flan**

### Mono distortion circuits in combination with moving delays, and a stereo chorus or stereo flange
*Allocation Units*:3 each

Each of these four algorithms offer 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 re-creating 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 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 4th order roll-offs (24dB/oct) 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" respectively. The chorus and flange have mono inputs and stereo outputs. Each is a standard KDFX single tap dual channel chorus (see FXAlg #150) or flange (see FXAlg #154) 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 Quad MovDelay algorithms (FXAlgs #715-716). 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.
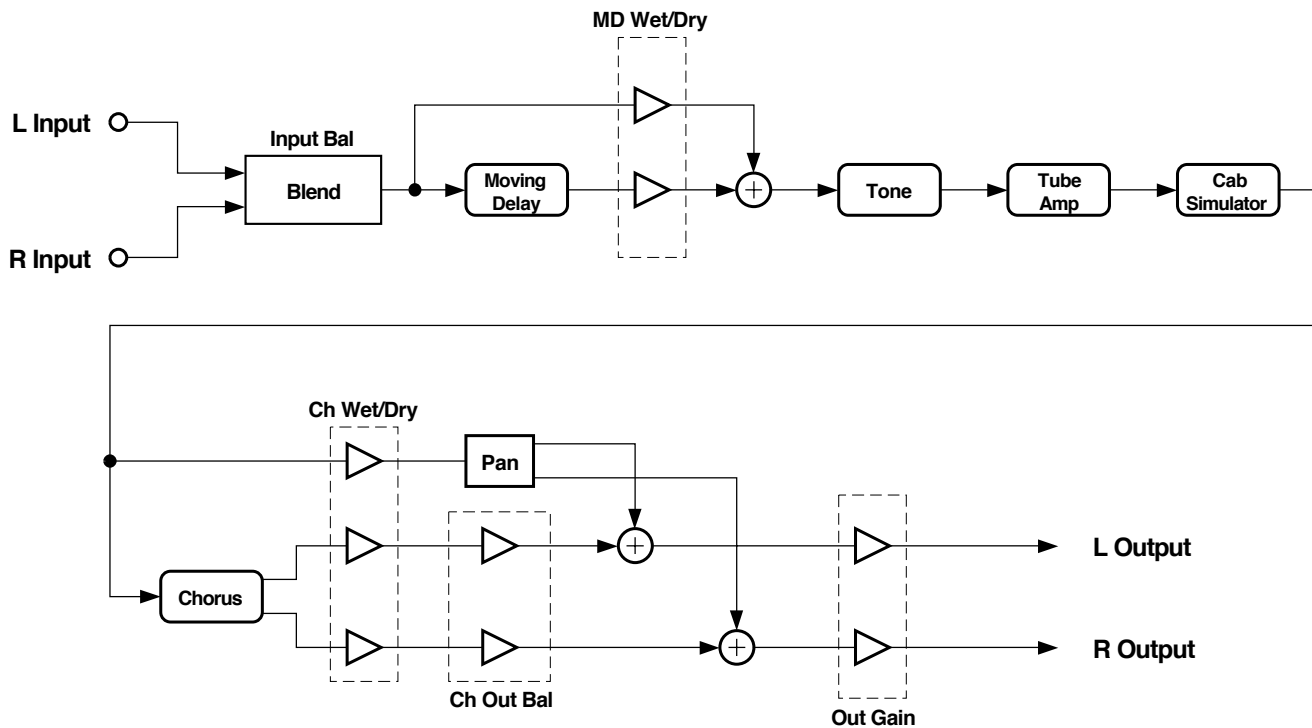


**TubeAmp<>MD>Chor with moving delay inserted PreDist**
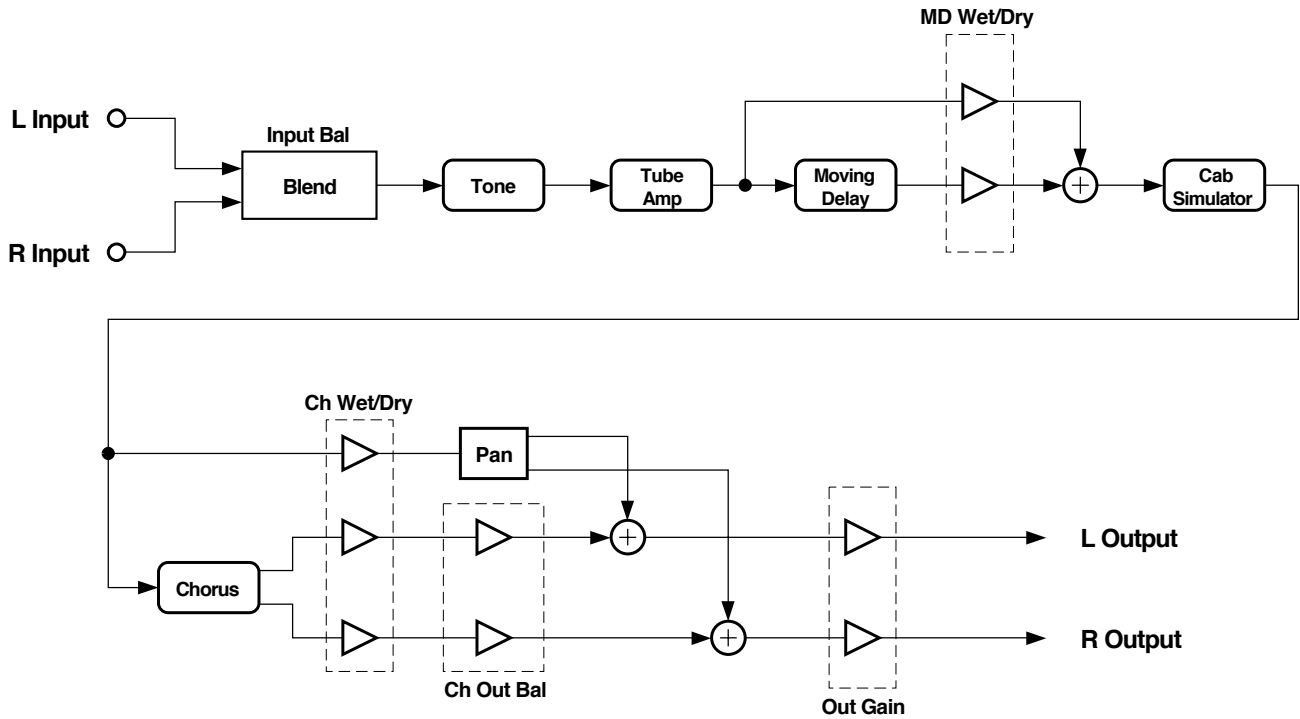
**TubeAmp<>MD>Chor with moving delay inserted PostDist**

## 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 algs)**

| | | Tube Drive | Off; -79.0 to 60.0 dB |
|---|---|---|---|
| | | Warmth | 16 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 2 (PolyAmp algs)**

| | | Poly Drive | 0.0 to 60.0 dB |
|---|---|---|---|
| | | Warmth | 16 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 3

| MD Insert | Post Dist, ... | MD Delay | 0.0 to 1000.0 ms |
|---|---|---|---|
| MD Wet/Dry | 0 to 100% | MD LFOMode | Flange, ... |
| | | MD LFORate | 0.00 to 10.00 Hz |
| | | MD LFODpth | 0.0 to 200.0% |
| | | MD Fdbk | -100 to 100% |

PAGE 4 (Chorus algs)

| 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 cts | Ch Depth R | 0.0 to 100.0 cts |
| Ch Delay L | 0 to 720 ms | Ch Delay R | 0 to 720 ms |
| Ch Fdbk  L | -100 to 100% | Ch Fdbk  R | -100 to 100% |
| Ch PtchEnv | Triangle or Trapzoid | | |
| ChWet/Dry | 0 to 100% | Ch Out Bal | -100 to 100% |

PAGE 4 (Flange algs)

| Fl Rate | 0 to 32 bts | Fl Tempo | System; 1 to 255 BPM |
|---|---|---|---|
| Fl Xcurs L | 0 to 230 ms | Fl Xcurs R | 0 to 230 ms |
| Fl Delay L | 0 to 230 ms | Fl Delay R | 0 to 230 ms |
| Fl Fdbk  L | -100 to 100% | Fl Fdbk  R | -100 to 100% |
| Fl Phase L | 0 to 360 deg | Fl Phase R | 0 to 360 deg |
| 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 3 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) lopass 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.

| | |
|---|---|
| <u>MD Delay</u> | Adjusts the delay time for the moving delay circuit, which is the center of LFO excursion. |
| <u>MD LFOMode</u> | 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. |
| <u>MD LFORate</u> | Adjusts the LFO speed for the moving delay circuit. |
| <u>MD LFODpth</u> | 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. |
| <u>MD Fdbk</u> | Adjusts the level of the moving delay circuit output signal fed back into its own input. Negative values polarity-invert the feedback signal. |
| <u>Ch Wet/Dry, Fl Wet/Dry</u> | 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. |
| <u>Ch Out Bal, Fl Out Bal</u> | 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. |

# FXAlg #733: VibChor+Rotor 2 · FXAlg #734: VibChor+Rotor 4

## Vibrato/chorus, through optional distortion, into rotating speaker
*Allocation Units:* 2 for VibChor+Rotor 2; 4 for VibChor+Rotor 4

The VibChor+Rotor algorithms contain multiple effects designed for the Hammond B3® emulation (KB3 mode). These effects are the Hammond® vibrato/chorus, amplifier distortion, and rotating speaker (Leslie®). Each of these effects may be turned off or bypassed, or the entire algorithm may be bypassed.
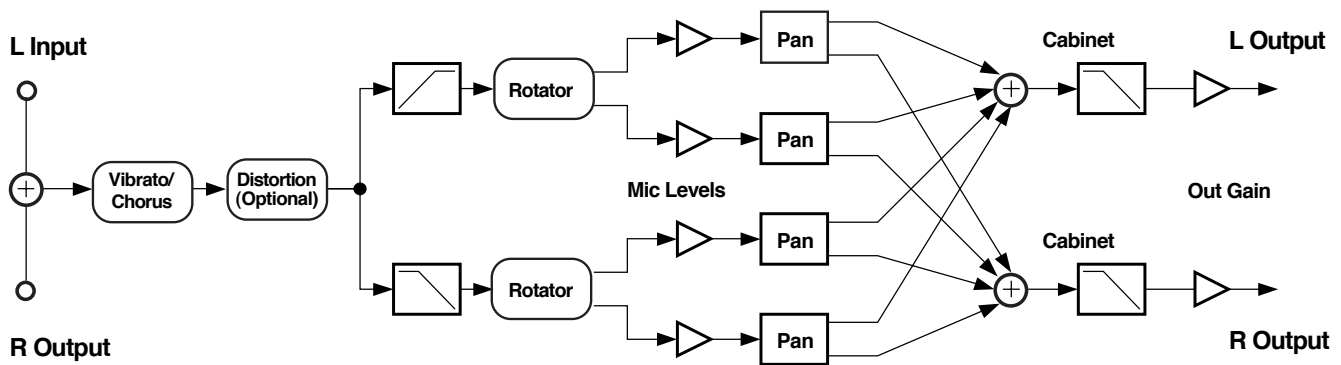


**Block diagram of VibChor+Rotor**

The first effect in the chain is 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 VibChor+Rotor 4, the vibrato chorus has been carefully modeled after the electro-mechanical vibrato/chorus in the B3. The vibrato/chorus in VibChor+Rotor 2 uses 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 VibChor+Rotor 4 vibrato/chorus.

In VibChor+Rotor 4 an amplifier distortion algorithm follows the vibrato/chorus. See the section in this book on FXAlg #724 for more information about the distortion algorithm.

Finally, the signal passes through a rotating speaker routine. 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 a pair of virtual microphones which can be positioned at varying positions (angles) around the rotors. An angle of 0° is loosely defined as the front. You can also control the levels and left-right panning of each virtual microphone. The signal is then passed through a final lowpass filter to simulate the band-limiting effect of the speaker cabinet.
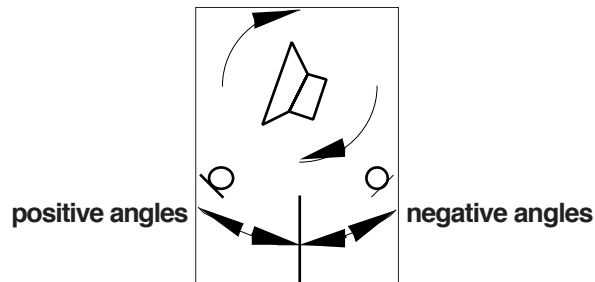
**Rotating speaker with virtual microphones**

For the rotating speakers, you can control the crossover 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 rate, the effective driver size and tremolo can be set. The rotation rate sets how fast the rotating speaker is spinning. 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.
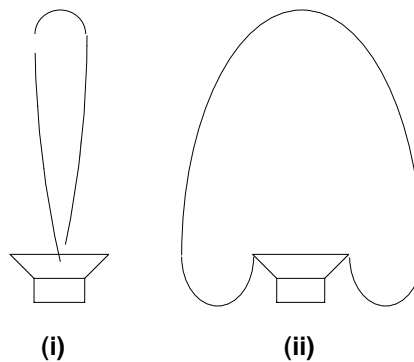


**Acoustic beams for (i) low frequency driver and (ii) high frequency driver.**

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.

## Parameters:

**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 | 16 to 25088 Hz |
| Roto InOut | In or Out | Cabinet LP | 16 to 25088 Hz |

**PAGE 2**

| Xover | 16 to 25088 Hz | | |
|---|---|---|---|
| Lo Gain | Off, -79.0 to 24.0 dB | Hi Gain | Off, -79.0 to 24.0 dB |
| Lo Rate | -10.00 to 10.00 Hz | Hi Rate | -10.00 to 10.00 Hz |
| 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**

| 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% |

**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 | | |

| | |
|---|---|
| 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", "V3", and three choruses "C1", "C2", "C3" |
| Roto InOut | When set to "In" the rotary speaker is active; when set to "Out" the rotary speaker is bypassed. |
| 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. [VibChor+Rotor 4 only] |

| | |
|---|---|
| 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. [VibChor+Rotor 4 only] |
| 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 Gain | The gain or amplitude of the signal passing through the rotating woofer (low-frequency driver. |
| Lo Rate | The rotation rate of the rotating woofer (low-frequency driver). The woofer can rotate clockwise or counter-clockwise. The direction of rotation depends on the sign of the rate parameter. Assuming microphone angles are set toward the front (between -90° and 90°) and microphones at positive angles are panned to the right (positive pan values), then positive rates correspond to clockwise rotation when viewed from the top. |
| 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 Rate | The rotation rate of the rotating tweeter (high-frequency driver). The tweeter can rotate clockwise or counter-clockwise. The direction of rotation depends on the sign of the rate parameter. Assuming microphone angles are set toward the front (between -90° and 90°) and microphones at positive angles are panned to the right (positive pan values), then positive rates correspond to clockwise rotation when viewed from the top. |
| 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). |
| 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 will result in large sample skips (audible as clicks when signal is passing through the effect). There are four of these parameters to include 2 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 2 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 expressed 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.

# FXAlg #734: Distort + Rotary

### Small distortion followed by rotary speaker effect
*Allocation Units:* 2

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. 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.
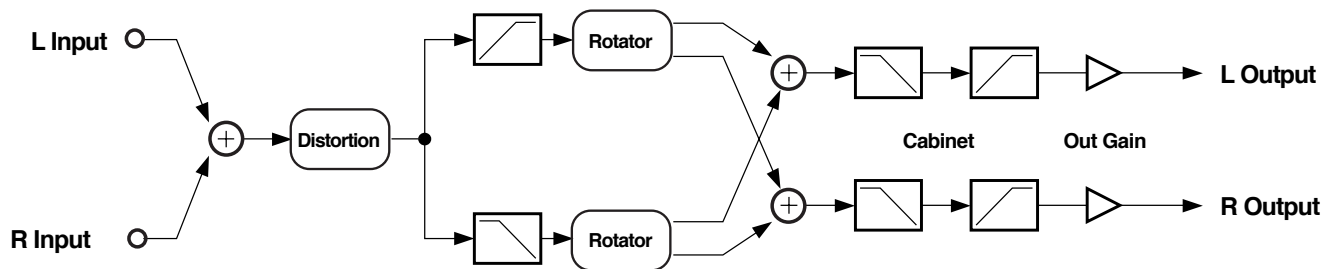


**Block diagram of Distort + Rotary**

The first part of Distort + Rotary is a distortion algorithm. See the section of this book on FXAlg #723 for details.

Next the signal passes through a rotating speaker routine. See the section of this book on FXAlg #733 for details.

## Parameters:

**PAGE 1**

| In/Out | In or Out | Out Gain | Off, -79.0 to 24.0 dB |
|---|---|---|---|
| Cabinet HP | 16 to 25088 Hz | Dist Drive | 0 to 96 dB |
| Cabinet LP | 16 to 25088 Hz | DistWarmth | 16 to 25088 Hz |

**PAGE 2**

| Xover | 16 to 25088 Hz | Mic Angle | 0.0 to 360.0 deg |
|---|---|---|---|
| Lo Gain | Off, -79.0 to 24.0 dB | Hi Gain | Off, -79.0 to 24.0 dB |
| Lo Rate | -10.00 to 10.00 Hz | Hi Rate | -10.00 to 10.00 Hz |
| Lo Size | 0 to 250 mm | Hi Size | 0 to 250 mm |
| Lo Trem | 0 to 100% | Hi Trem | 0 to 100% |

**PAGE 3**

| | | ResH/LPhs | 0.0 to 360.0 deg |
|---|---|---|---|
| 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 |

| | |
|---|---|
| In/Out | When set to "In", the algorithm is active; when set to "Off" 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. |
| 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. |
| 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. |
| Cabinet HP | A hipass filter to simulate the band-limiting of a speaker cabinet. The filter controls the lower frequency limit of the output. |
| 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. |

For details on the rest of the parameters see the previous section (FXAlg #733) of this book.

# FXAlg #735/6: KB3 FX

**Vibrato/chorus into distortion into rotating speaker into cabinet**
*Allocation Units:* 7 for full working effect (4 for KB3 FXBus, 3 for KB3 AuxFX)

The KB3 FXBus and KB3 AuxFX algorithms contain multiple effects designed for the Hammond B3 emulation (KB3 mode). For correct operation, both effects must be running at the same time, with the output of KB3 FXBus feeding the input of KB3 AuxFX. The two algorithms work as one algorithm which use all the available KDFX resources. While the input to KB3 FXBus is stereo (which gets summed to mono) and the output from KB3 AuxFX is stereo, the signals between the two algorithms are the low frequency (left) and high frequency (right) signal bands used to drive the lower and upper rotary speakers. It is possible to run these two algorithms as independent effects, but it is recommended.

These effects are the Hammond vibrato/chorus, amplifier distortion, and rotating speaker (Leslie) emulations. Each of these effects may be turned off or bypassed, or the entire algorithm may be bypassed. To bypass the rotary, the switches in both KB3 FXBus and KB3 AuxFX must be set to "Out".
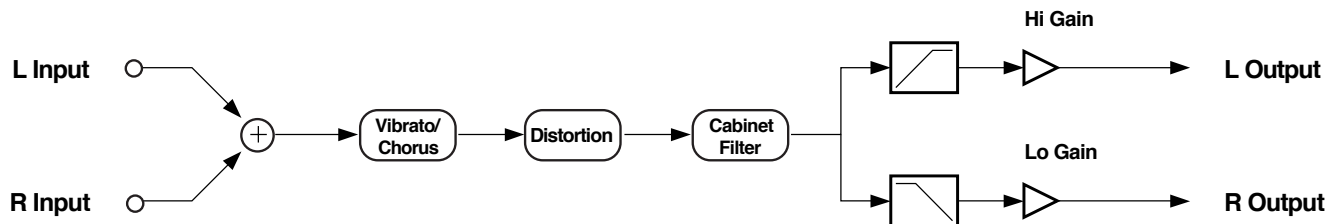


**Block diagram of KB3 FXBus**
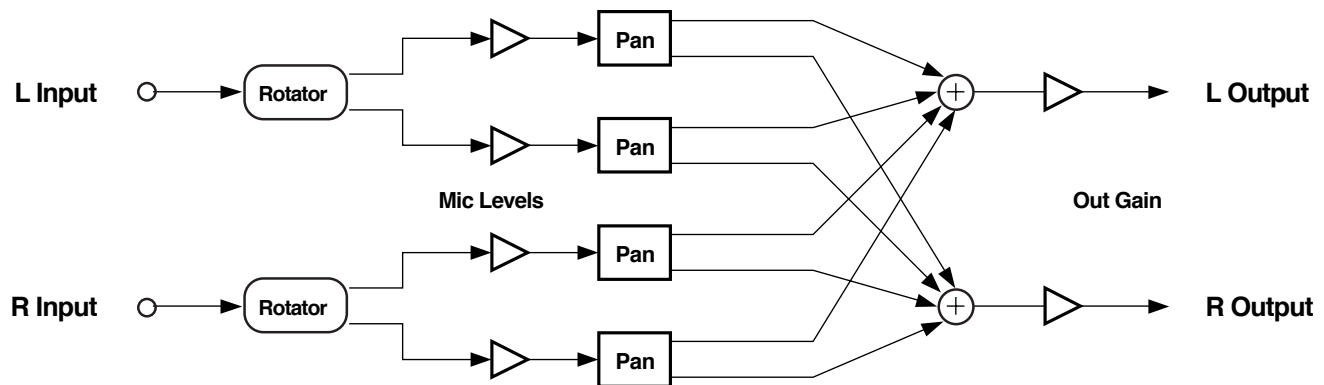


**Block diagram of KB3 AuxFX**

The first effect in the chain is 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. The vibrato chorus has been carefully modeled after the electro-mechanical vibrato/chorus in the B3.

An amplifier distortion algorithm follows the vibrato/chorus. For details, see the section in this book on FXAlg #723.

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 hipass filter which 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.

At this point KB3 FXBus has finished its processing and passes the high and low signals to the KB3 AuxFX algorithm which contains the rotating-speaker routine. See the section in this book on FXAlg #733 for details.

## Parameters (KB3 FXBus):

**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 | 16 to 25088 Hz |

**PAGE 2**

| RotoInOut | In or Out | | |
|-----------|-----------|---------|------------------------|
| Lo Gain | Off, -79.0 to 24.0 dB | Hi Gain | Off, -79.0 to 24.0 dB |
| Lo Xover | 16 to 25088 Hz | Hi Xover | 16 to 25088 Hz |
| Lo HP | 16 to 25088 Hz | Hi LP | 16 to 25088 Hz |

In/Out
: When set to "In", the algorithm is active; when set to "Out" the algorithm is bypassed. For the entire algorithm to be active, KB3 AuxFX must also be active.

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", "V3", and three choruses "C1", "C2", "C3"

Roto InOut
: When set to "In" the rotary speaker is active; when set to "Out" the rotary speaker is bypassed. By bypassing the rotary effect in KB3 FXBus, only the crossover filters are bypassed. You must also bypass KB3 AuxFX to completely bypass the rotary speakers. Likewise, for the entire rotary to be active, KB3 AuxFX must also be active.

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.

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.

Lo Gain
: The gain or amplitude of the signal passing through the rotating woofer (low frequency driver. The control is also available in KB3 AuxFX.

Lo Xover
: The crossover frequency for the low frequency driver. Lo Xover controls a lowpass filter.

Lo HP
: A hipass filter to simulate the band-limiting of a speaker cabinet. The filter controls the lower frequency limit of the output.

Hi Gain
: The gain or amplitude of the signal passing through the rotating tweeter (high frequency driver. The control is also available in KB3 AuxFX.

Hi Xover
: The crossover frequency for the high frequency driver. Hi Xover controls a hipass filter.

Hi LP
: A lowpass filter to simulate the band-limiting of a speaker cabinet. The filter controls the upper frequency limit of the output.

## Parameters (KB3 AuxFX):

**PAGE 1**

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

**PAGE 2**

| Lo Gain | Off, -79.0 to 24.0 dB | Hi Gain | Off, -79.0 to 24.0 dB |
|---------|----------------------|---------|----------------------|
| Lo Rate | -10.00 to 10.00 Hz | Hi Rate | -10.00 to 10.00 Hz |
| 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**

| 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% |

**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 | | |

In/Out
When set to "In", the algorithm is active; when set to "Off" the algorithm is bypassed. For the entire algorithm to be active, KB3 FXBus must also be active with its Roto InOut parameter set to "In". To completely bypass the rotary, one or both of the In/Out or Roto InOut parameters in KB3 FXBus must also be bypassed.

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

Lo Gain
The gain or amplitude of the signal passing through the rotating woofer (low frequency driver. The control is also available in KB3 FXBus.

Hi Gain
The gain or amplitude of the signal passing through the rotating tweeter (high frequency driver. The control is also available in KB3 FXBus.

For details on the rest of the parameters see the section of this book on FXAlg #733.

# FXAlg #900: Env Follow Filt

## Envelope-following stereo 2-pole resonant filter
### *Allocation Units:* 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 one of low pass (i), highpass (ii), band pass (iii), or notch (iv).



|       |        |
|-------|--------|
| (i)   | (ii)   |
| (iii) | (iv)   |
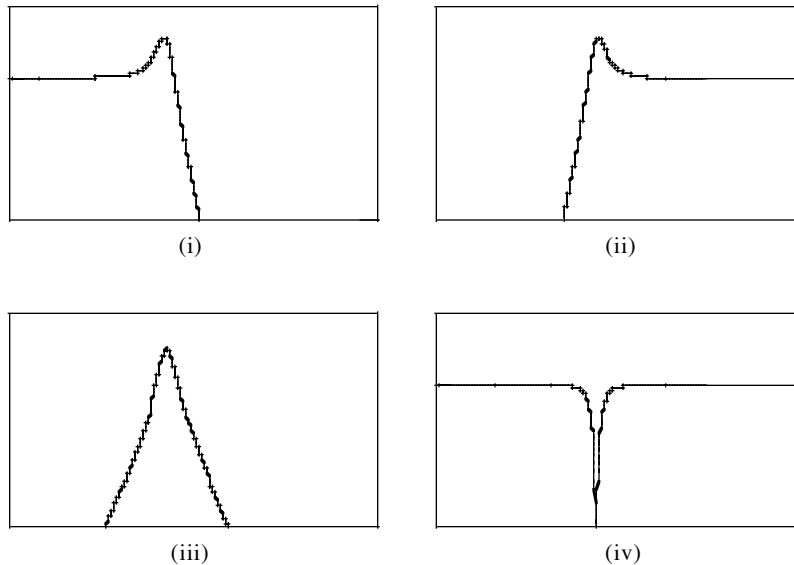
**Resonant Filter Types: (i) lowpass, (ii) highpass, (iii) bandpass, and (iv) notch.**

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 16 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.
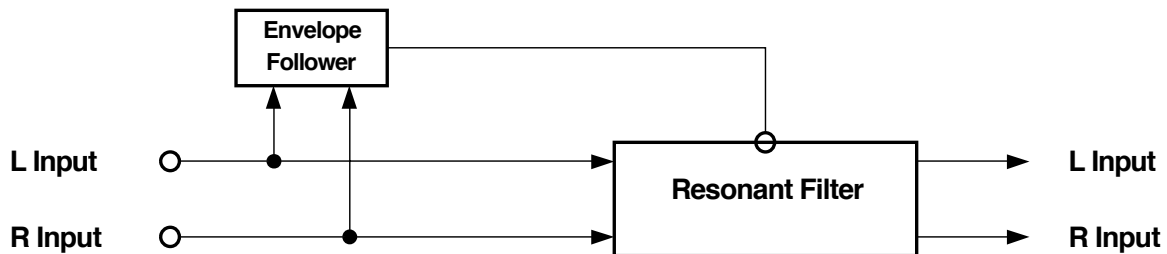


**Block diagram of envelope-following filter**

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 lopass filter which can extend the attack and release times of the envelope follower. A level meter with a threshold 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 | 16  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 |
| E | | | |
| -dB  60   40   *  16  *  8   4   0 | | | |

| | |
|---|---|
| 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 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. In the notch filter, this sets the amount of cut, so 0 dB provides the highest, widest notch, and higher levels make the notch increasingly narrower and shallower. |
| Threshold | 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. |
| 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. |

# FXAlg #901: TrigEnvelopeFilt

### Triggered envelope-following stereo 2-pole resonant filter
*Allocation Units:* 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 one of low pass (i), high pass (ii), band pass (iii), or notch (iv). See the previous section of this book, FXAlg #900, for diagrams of the filter actions.
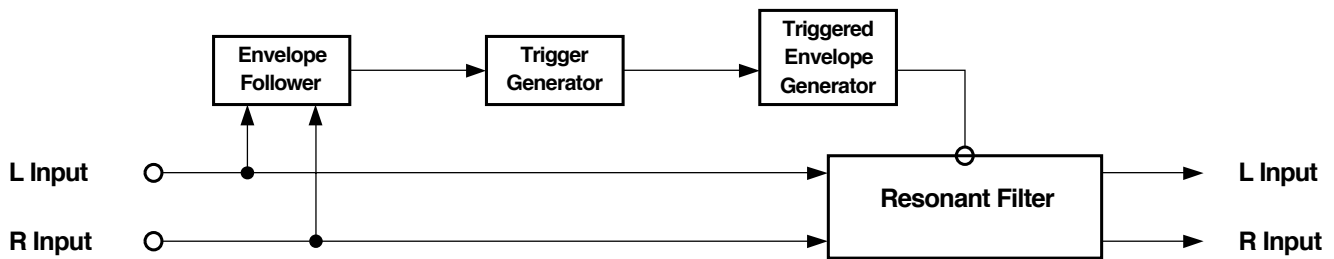
**Block diagram of Triggered Envelope Filter**

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 16 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.

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 | 16 to 8372 Hz |
| F | | Max Freq | 16 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 |
| E | | | |
| -dB   60   40   *   16   *   8   4   0 | | | |

| | |
|---|---|
| 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. In the notch filter, this sets the amount of cut, so 0 dB provides the highest, widest notch, and higher levels make the notch increasingly narrower and shallower. |
| 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 only useful 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) |

# FXAlg #902: LFO Sweep Filter

## LFO-following stereo 2-pole resonant filter
*Allocation Units:* 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 one of low pass (i), high pass (ii), band pass (iii), or notch (iv). See the section of this book on FXAlg #900 for diagrams of the filter actions.

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 16 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. 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 only active 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 sawtooths develops a more gradual slope with smoothing, ending up as triangle waves when set to 100% smoothing.



| Sine | Saw+ | Saw- | Pulse | Tri |
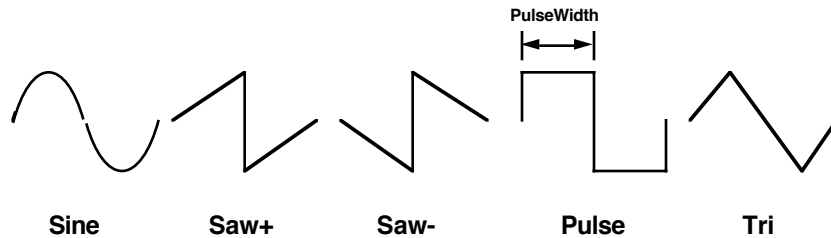
**Configurable Wave Shapes**

## 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 | 16 to 8372 Hz |
|---|---|---|---|
| | | Max Freq | 16 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. When it is set to "System", sources (FUNs, LFOs, ASRs etc.) will have no effect on the Tempo parameter. |
| LFO Period | Sets the LFO rate based on the Tempo determined above: the number of beats corresponding to one period of the LFO cycle. For example, if the LFO Period is set to "4", the LFOs will take four beats to pass through one oscillation, so the LFO rate will be 1/4th of the Tempo setting. If it is set to "6/24" (=1/4), the LFO will oscillate four times as fast as the Tempo. At "0", the LFOs stop oscillating and their phase is undetermined (wherever they stopped). |
| 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 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. |
| 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. In the notch filter, this sets the amount of cut, so 0 dB provides the highest, widest notch, and higher levels make the notch increasingly narrower and shallower. |
| 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. |

# FXAlg #903 Resonant Filter • FXAlg #904 Dual Res Filter

### Stereo and dual-mono 2-pole resonant filters
*Allocation Units:* 1 (each)

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 one of low pass (i), high pass (ii), band pass (iii), or notch (iv). See the section of this book on FXAlg #900 for diagrams of the filter actions.

The resonant frequency of the filter and the filter resonance level may be adjusted.

## Parameters (Resonant Filter):

**PAGE 1**

| Wet/Dry | 0 to 100%wet | Out Gain | Off, -79.0 to 24.0 dB |
|---|---|---|---|
| FilterType | Lowpass | | |
| Frequency | 16  to 8372 Hz | | |
| Resonance | 0 to 50 dB | | |

## Parameters (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 | 16  to 8372 Hz | R Freq | 16  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. In the notch filter, this sets the amount of cut, so 0 dB provides the highest, widest notch, and higher levels make the notch increasingly narrower and shallower. |

# FXAlg #905: EQ Morpher ·
# FXAlg #906: Mono EQ Morpher

### Parallel resonant bandpass filters with parameter morphing
*Allocation Units:* 4 for EQ Morpher, 2 for Mono EQ Morpher

The EQ Morpher algorithms have four parallel bandpass filters acting on the input signal, whose 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 KDFX Studio INPUT pages is used, which includes a width parameter to control the width of the stereo field. Mono EQ Morph uses a standard mono panner for positioning the mono signal between the left and right speakers.
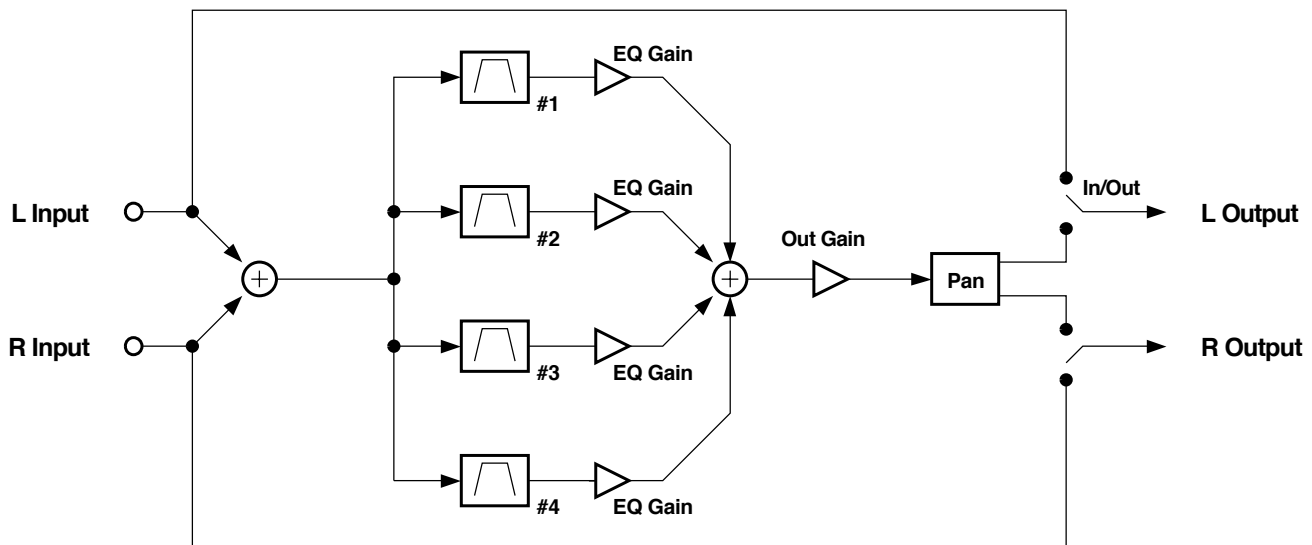


**Mono EQ Morpher (EQ Morpher is similar)**

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 the usual parametric filters you are familiar with. Normal parametric filters 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 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.
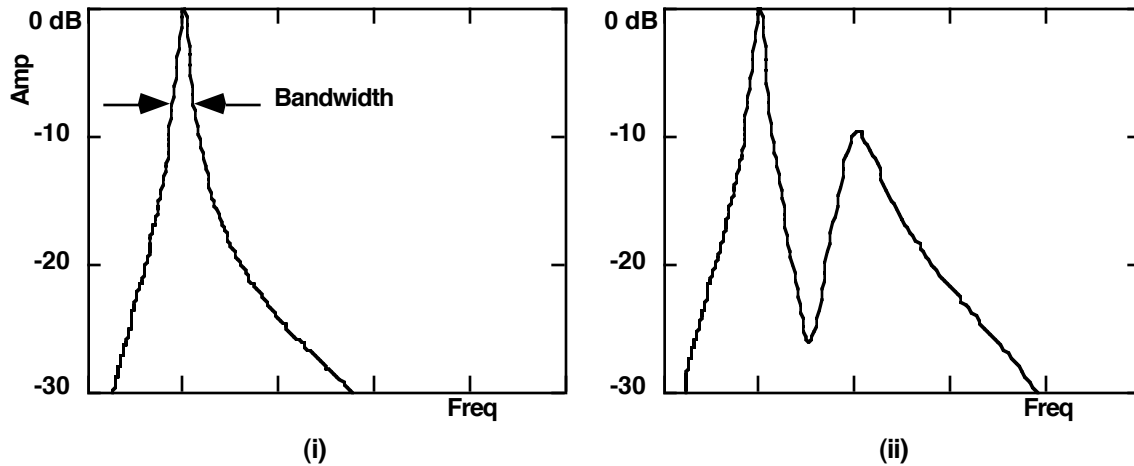
**Frequency response of (i) a single bandpass filter, and (ii) the sum of two bandpass filters**

Now that we've gone through what the algorithm does, the question becomes "Why are we doing this?" With careful thought to parameter settings, EQ Morph does 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 Morphers. 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* | -100 to 100% |
| AFreqScale | -8600 to 8600 ct | BFreqScale | -8600 to 8600 ct |

*EQ Morpher only, not Mono EQ Morpher.

**PAGE 2**

| A Freq  1 | 16 to 25088 Hz | B Freq  1 | 16 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 | 16 to 25088 Hz | B Freq  2 | 16 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 | 16 to 25088 Hz | B Freq  3 | 16 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 | 16 to 25088 Hz | B Freq  4 | 16 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 Morph, this is a stereo panner
                  which pans the entire stereo image, the same way the input sends on a KDFX INPUT
                  page are handled, 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. [EQ Morpher
                  only]

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 A to B settings, the A filter #1 will change to the B filter
                  #1, A filter #2 moves to B filter #2, and so on. This is an excellent candidate for
                  assignment to a real-time KDFX Modulation.

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.This, too, is an excellent
                  candidate for assignment to a real-time KDFX Modulation.

For the two filter sets A & 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 boosted nor 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.

# FXAlg #907: Ring Modulator

### A configurable ring modulator
*Allocation Units:* 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's and why's 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 $\phi$ 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 trigometric identity for this, or we can just look 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):
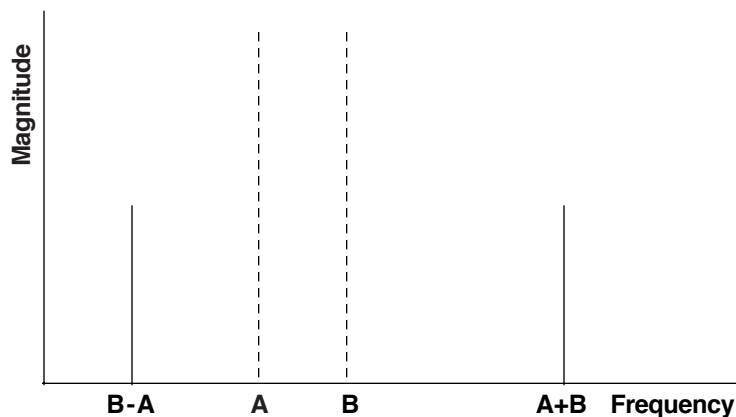


**Result of Modulating Two Sine Waves, A and B**

This algorithm has two operating modes, 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. The following figure shows the signal flow when in "L\*R" mode:
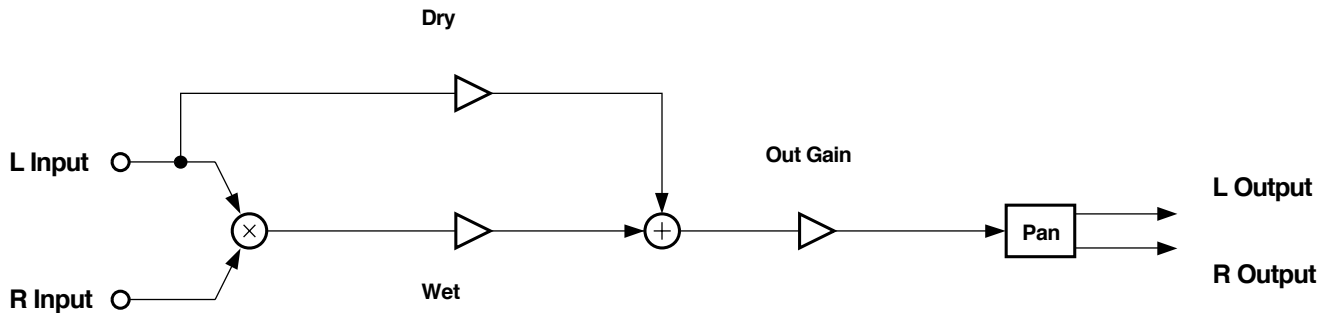
**Ring Modulator in "L*R" Mode**

The other modulation mode is "Osc". In this mode, the algorithm inputs and outputs are stereo, and the carrier signal for both channels is generated inside the algorithm.
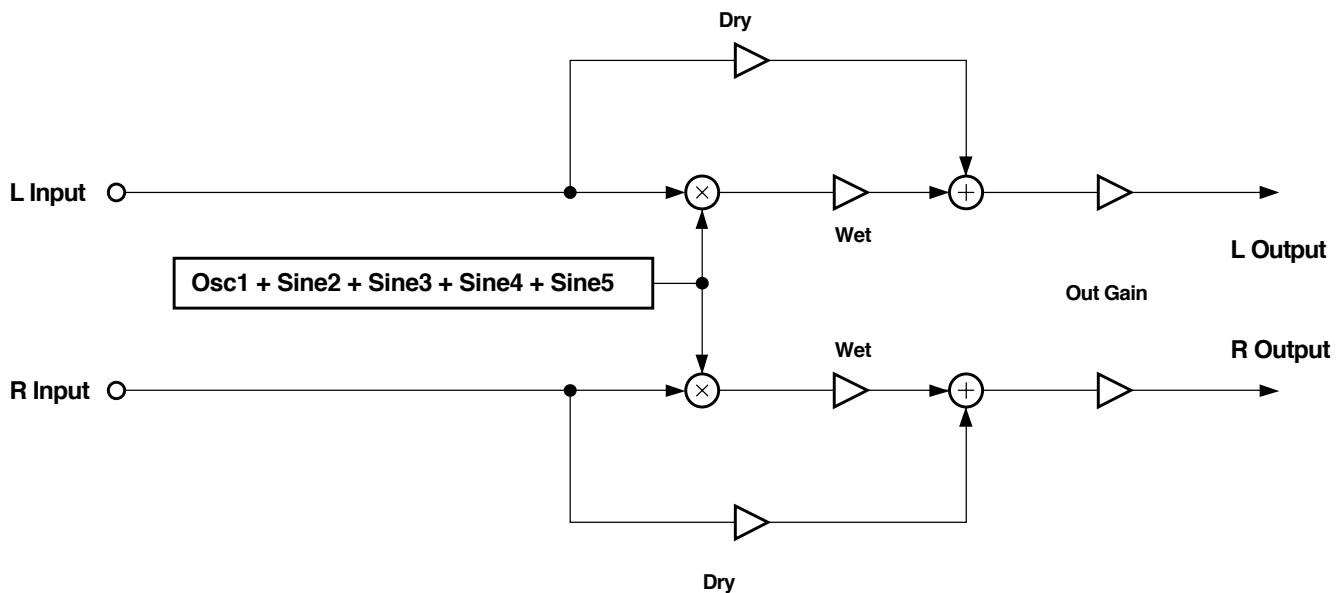
**Ring Modulator in "Osc" Mode**

The carrier signal is the sum of five oscillators. On all of the oscillators, level and frequency may be set. Four of the oscillators are simple sine waves, while the fifth may be configured to one of a variety of wave shapes. Available shapes are Sine, Saw+, Saw-, Pulse, Tri and Expon. 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 only active when the Pulse waveform is selected. Saw+ and Saw- produce rising and falling sawtooth waveforms.

Smoothing is available to reduce the upper partials of the Pulse and Saw waveforms. The pulse wave, at 50% width, turns into a sine wave when set to 100% smoothing. The sudden change in amplitude of a sawtooth turns into a more gradual slope with smoothing, ending up as a triangle wave when set to 100% smoothing.

**PulseWidth**

| Sine | Saw+ | Saw- | Pulse | Tri | Expon |
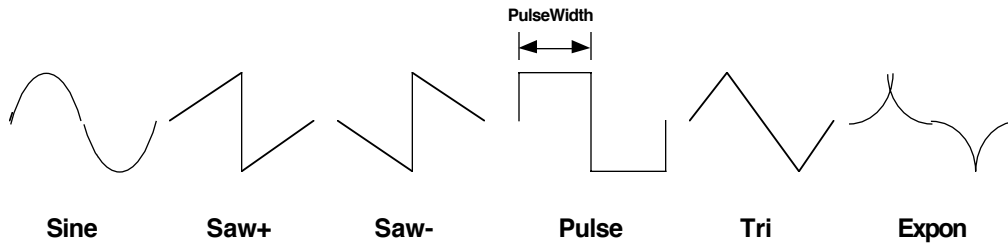
**Configurable Wave Shapes (Osc1 only)**

## Parameters:

| 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% |

| Osc1 Lvl | 0 to 100% | Osc1 Freq | 16 to 25088 Hz |
|----------|-----------|-----------|----------------|
| Osc1 Shape | Sine | | |
| Osc1PlsWid | 0 to 100% | | |
| Osc1Smooth | 0 to 100% | | |

| Sine2 Lvl | 0 to 100% | Sine2 Freq | 16 to 25088 Hz |
|-----------|-----------|------------|----------------|
| Sine3 Lvl | 0 to 100% | Sine3 Freq | 16 to 25088 Hz |
| Sine4 Lvl | 0 to 100% | Sine4 Freq | 16 to 25088 Hz |
| Sine5 Lvl | 0 to 100% | Sine5 Freq | 16 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. 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 16-25088 Hz with 1 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 16-25088 Hz with 1 semitone resolution. This parameter is active only in "Osc" mode.

# FXAlg #908: Pitcher

## Creates pitch from pitched or non-pitched signal
*Allocation Units:* 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 several 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].
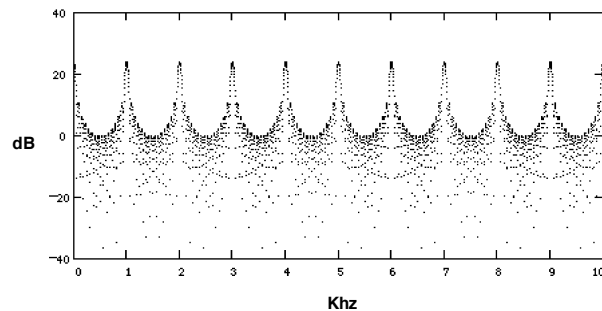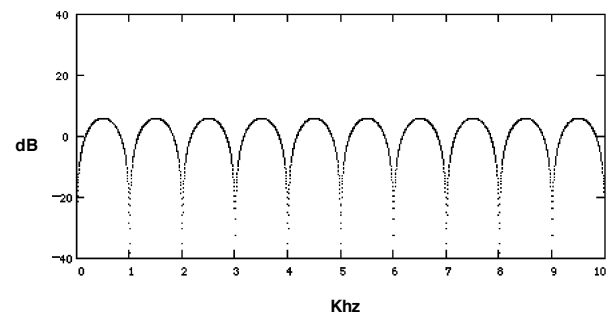


**[OPQH=100, 100, 100, 100] All peaks are exact multiples of the fundamental frequency set by the Pitch parameter. This setting gives the most "pitchiness" to the output.**
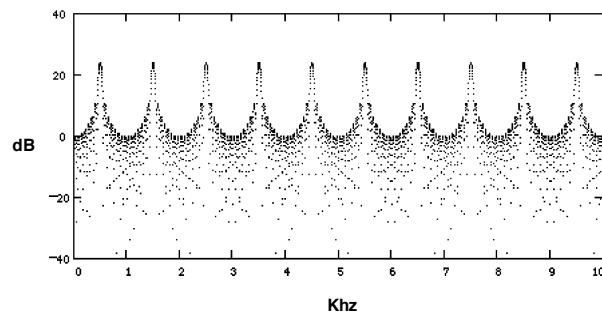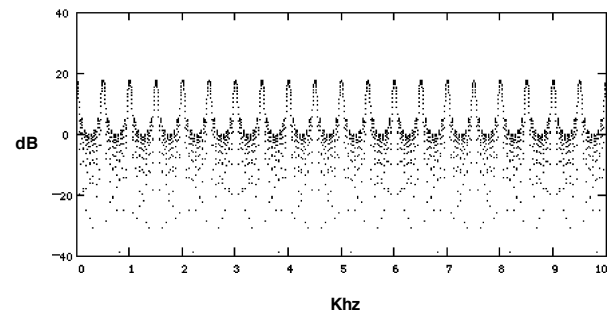


**[OPQH= -100, 100, 100, 100] Peaks are odd multiples of a frequency one octave down from the Pitch setting. This gives a hollow, square-wavey sound to the output.**
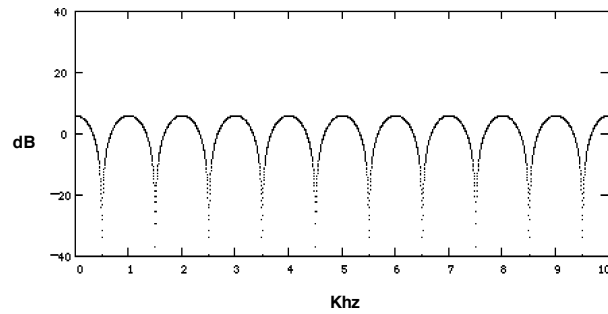
**[OPQH=100, 0, 0, 0] Deeper notches between wider peaks**
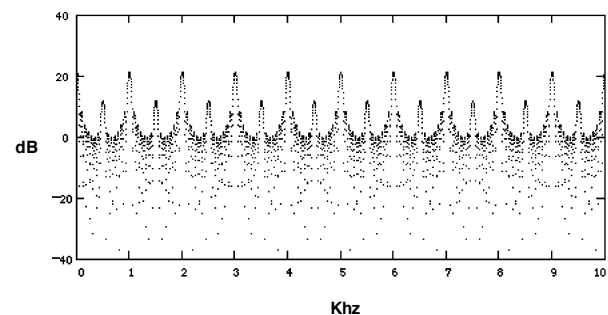


**[OPQH= -100, 0, 0, 0] Peaks on odd harmonic multiples and notches on
even harmonic multiples of a frequency one octave down from the Pitch setting.**



**[OPQH=0, 100, 100, 100] Like [100,100,100,100], except that all the peaks are at
(all) multiples of half the Pitch frequency.**



**[OPQH=50,100,100,100] Halfway between [0,100,100,100] and [100,100,100,100].**
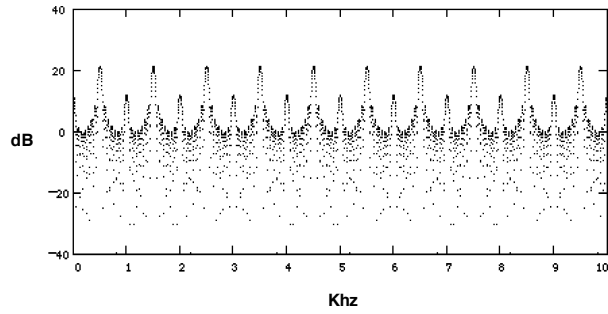
**[OPQH= -50,100,100,100] Halfway between [0,100,100,100] and [-100,100,100,100].
If the "Odd" parameter is modulated with an FXMOD, then one can morph smoothly between the
[100,100,100,100] and [-100,100,100,100] curves.**
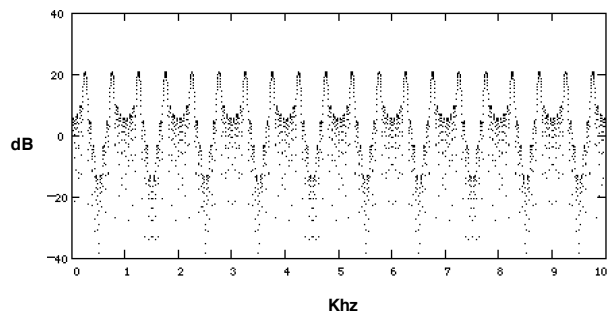


**[OPQH=100, -100, 100, 100]**
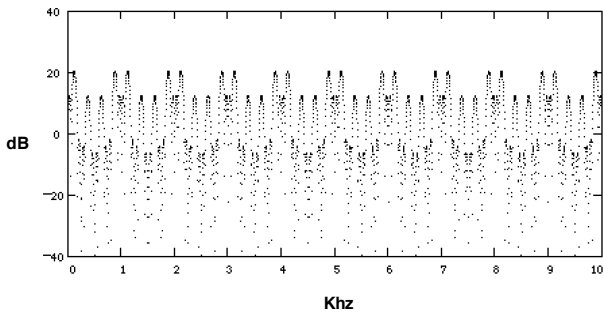


**[OPQH=100, 100, -100, 100]**



**[OPQH=100, 100, 100, -100]**

The other 1,632,240,792 response curves have been omitted in the interest of brevity.

## Parameters:

**PAGE 1**

| 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% |

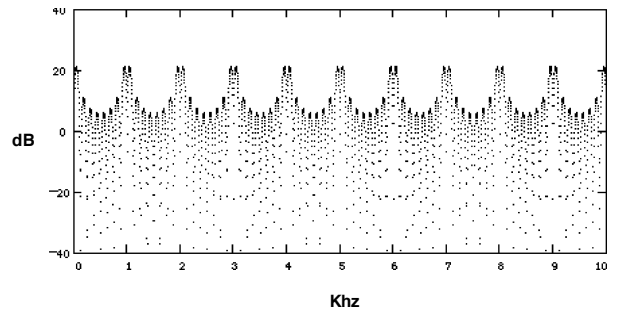Wet/Dry                  The relative amount of input signal and effected signal that is to appear in the final effect output mix. When set to 0%, the output is taken only from the input (dry). When set to 100%, the output is all wet.

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

Pitch                    The fundamental pitch imposed upon the input. Values are in MIDI note numbers.

Ptch Offst               An offset from the pitch frequency in semitones. This is also available for adding an additional continuous controller mod like pitch bend.

Odd Wts, Pair Wts, Quartr Wts, Half Wts       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.

# FXAlg #909: Super Shaper

## Ridiculous shaper
*Allocation Units:* 1

The Super Shaper algorithm packs two and a half times the number of shaping loops, and 8 times the gain of the shaper found in VAST. Refer to the section on shapers in the *K2500 Performance Guide* for an overview of VAST shaper.

Setting Super Shaper amount under 1.00x will produce the same non-linear curve as that found in the VAST shaper. At values above 1.00x where the VAST 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.

**1.00x**

**2.50x**

**4.00x**
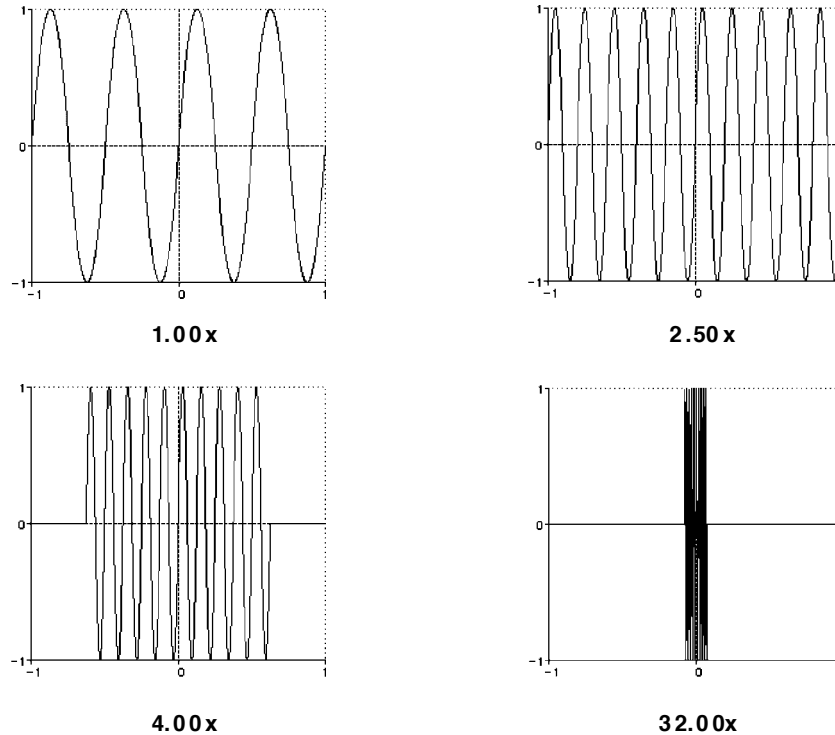
**32.00x**

**Super Shaper at various Amounts.**

## Parameters:

**PAGE 1**

| 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.

# FXAlg #910: 3 Band Shaper

### 3-band shaper
*Allocation Units:* 2

The 3 Band Shaper non-destructively splits the input signal into 3 separate bands using 1 pole (6dB / oct) filters, and applies a VAST-type shaper to each band separately. Refer to the *K2500 Performance Guide* for an overview of VAST 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 2 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 VAST 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.

Lo Enable, Mid Enable, Hi Enable        Turns processing for each band on or off. Turning each of the 3 bands Off results in a dry output signal.

Lo Amt, Mid Amt, Hi Amt        Adjusts the shaper intensity for each band.

Lo Mix, Mid Mix, Hi Mix        Adjusts the level that each band is summed together as the wet signal. Negative values polarity-invert the particular bands signal.

# FXAlg #911: Mono LaserVerb • FXAlg #912: LaserVerb Lite • FXAlg #913: LaserVerb

## A bizarre reverb with a falling buzz
*Allocation Units:* 1 for Mono LaserVerb; 2 for LaserVerb Lite; 3 for LaserVerb

LaserVerb is a new kind of reverb sound that has to be heard to be believed! When it is fed an impulsive sound such as a snare drum, 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.)

**t = 0**                                                                 **time**
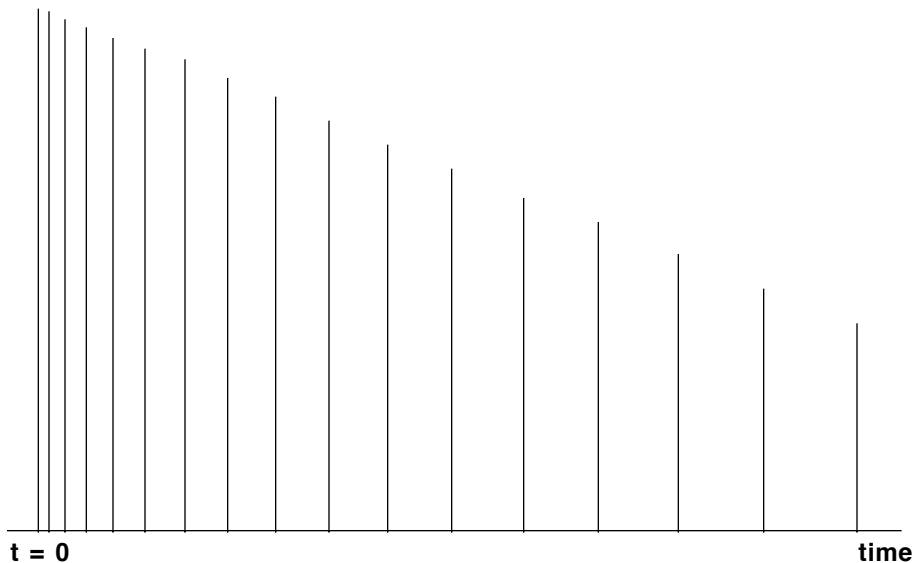
**Simplified Impulse Response of LaserVerb**

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 and decays slowly. As the Contour value is reduced, the decay becomes shorter and the sound takes longer to build up. At a setting of zero, 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 go down.

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 2-PAU version is a sparser version than the 3-PAU version. Its buzzing is somewhat coarser. The 1-PAU version is like the 2-PAU version except the two input channels are summed and run through a single mono LaserVerb. The 1-PAU version does not have the cross-coupling control but does have output panning.
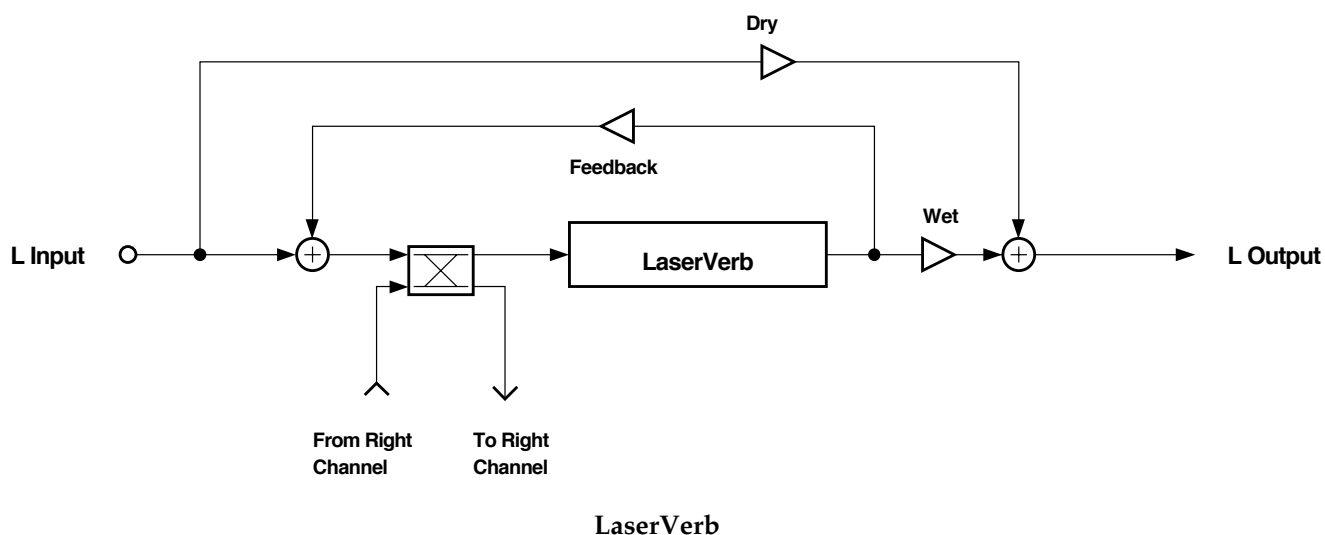
**LaserVerb**

## Parameters:

**PAGE 1 - 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 | 16 to 25088Hz | | |

**PAGE 1 - Mono LaserVerb**

| 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 | 16 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 & 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 (3 PAU) or 0 to 1.3 seconds (2 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 zero, LaserVerb is reduced to a simple delay. |

# FXAlg #950: HardKneeCompress • FXAlg #951: SoftKneeCompress

### Stereo hard- and soft-knee signal-compression algorithms
*Allocation Units:* 1

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. 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 between infinity 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.
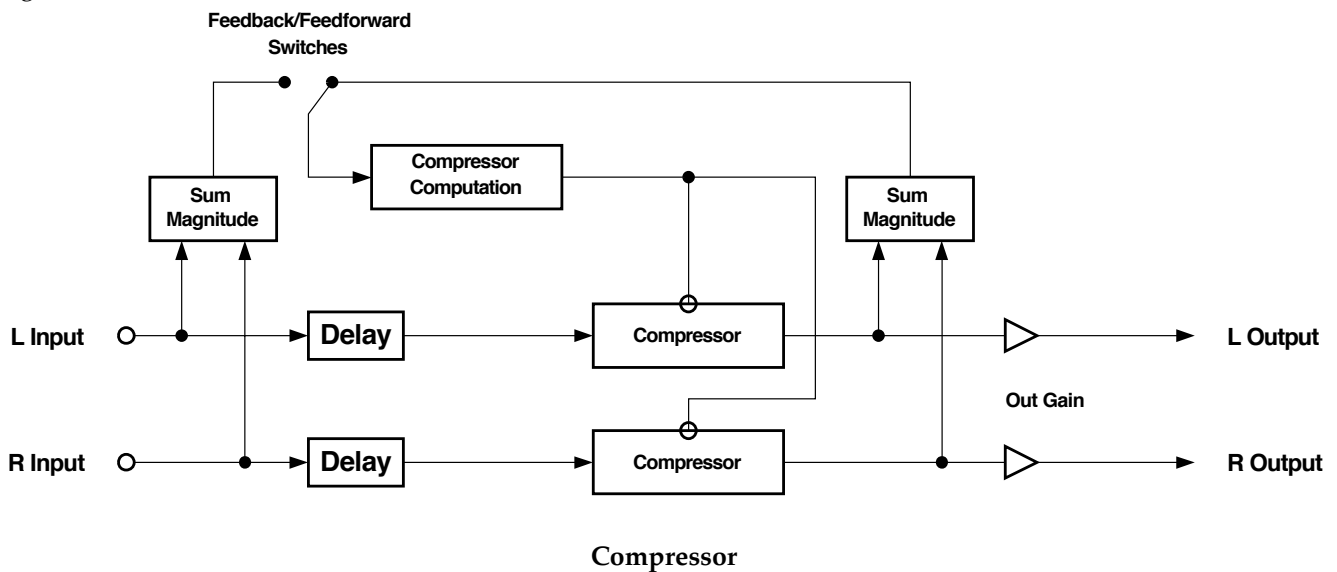


**Compressor**

In the hard-knee compressor, there is a sudden transition from uncompressed to compressed at the compression threshold. In the soft-knee compressor there is a more gradual transition from compressed to unity gain.



**Hard-Knee (left) and Soft-Knee (right) 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: "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 in feed-forward or feedback configuration. 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 feed-forward 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 feedback configuration, the delay affects both the main signal and the side chain, and so is of limited usefulness.

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 | | |

**PAGE 3**

| 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". |

| | |
|---|---|
| <u>FdbkComprs</u> | A switch to set whether the compressor side chain is configured for feed-forward (Out) or feedback (In). |
| <u>Atk Time</u> | The time for the compressor to start to cut in when there is an increase in signal level (attack) above the threshold. |
| <u>Rel Time</u> | The time for the compressor to stop compressing when there is a reduction in signal level (release) from a signal level above the threshold. |
| <u>SmoothTime</u> | 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. |
| <u>Signal Dly</u> | For the feed-forward setting, 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 pre-delay). This allows the compression to appear to take effect just before the signal actually rises. For feedback compression, this parameter causes both the side-chain and main signal path to be delayed together. |
| <u>Ratio</u> | The compression ratio. High ratios are highly compressed; low ratios are moderately compressed. |
| <u>Threshold</u> | The threshold level in dBFS (decibels relative to full scale) above which the signal begins to be compressed. |
| <u>MakeUpGain</u> | 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. |

# FXAlg #952: Expander

## A stereo expansion algorithm
*Allocation Units:* 1

This algorithm expands the signal (reduces 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. (Maximum expansion for this expander is 1:17.) 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.
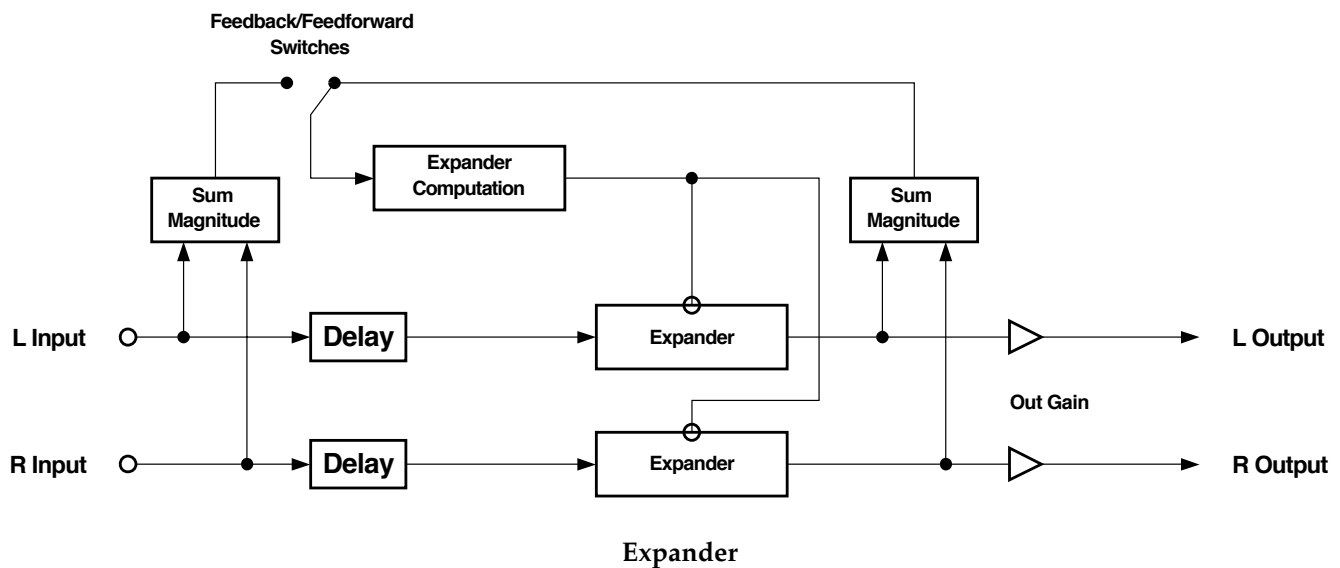


**Expander**

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.
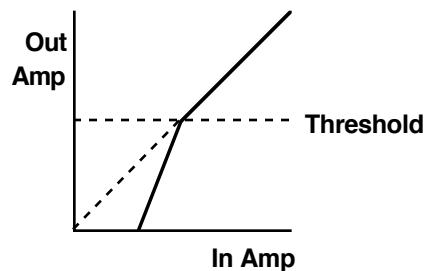


**Expansion Transfer Characteristic**

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 | | |
| **Reduction** | | | |
| **-dB  40   20  12   8   6   4   2   0** | | | |

| | |
|---|---|
| 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 pre-delay). This allows the expansion to appear to turn off just before the signal actually rises. |
| Ratio | The expansion ratio. High values (≤1:17) are highly expanded, low values (>1:1.0) 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. |

# FXAlg #953: Compress w/SC EQ

### Stereo soft-knee compression algorithm with filtering in the side chain
*Allocation Units:* 2

The Compress w/SC EQ algorithm is the same as the SoftKneeCompress algorithm 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.
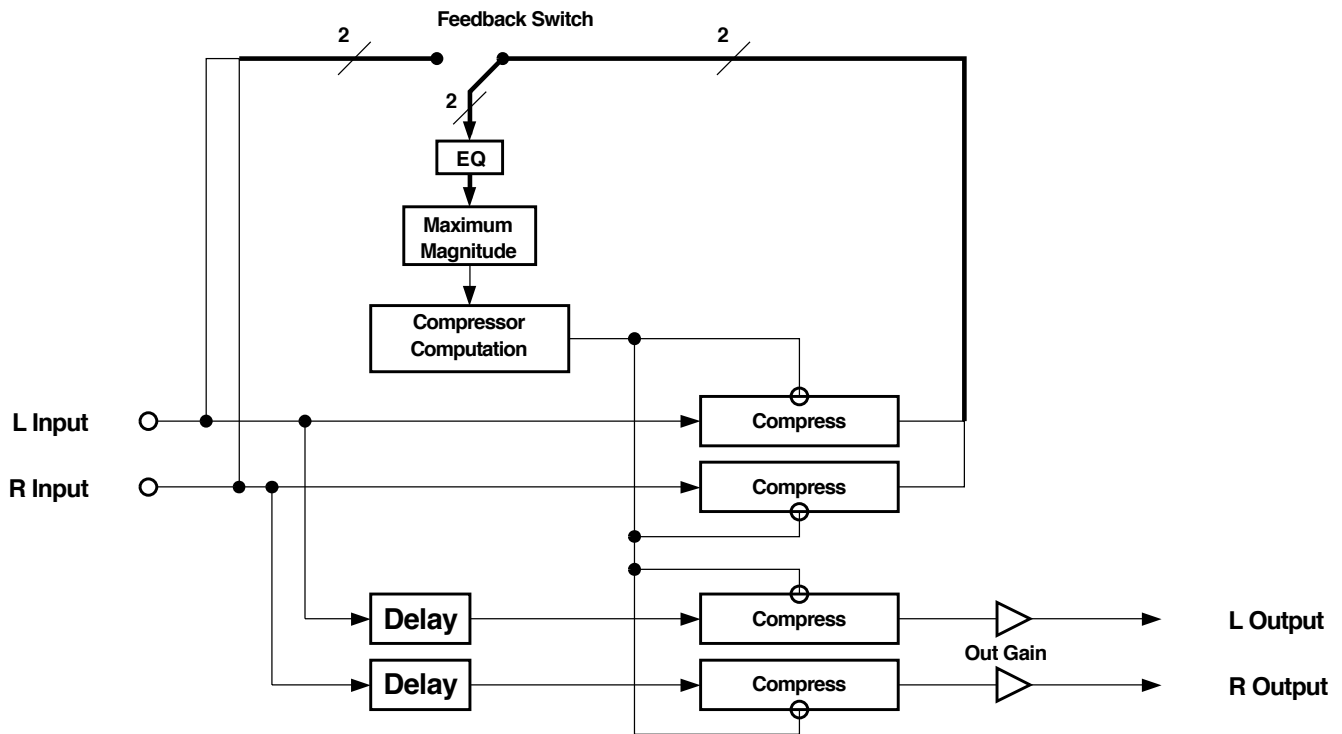


**Compressor with side chain equalization.**

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 are a lot of high frequencies present. This technique is often called "de-essing", because it is useful for removing excess sibilance from vocals.

## Parameters:

**PAGE 1**

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

**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 | 16 to 25088 Hz | SCTrebFreq | 16 to 25088 Hz |
| SCMidGain | -79.0 to 24.0 dB | | |
| SCMidFreq | 16 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). |
| 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 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 pre-delay). 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. |

# FXAlg #954: Compress/Expand · FXAlg #955: Comp/Exp + EQ

### A stereo soft-knee compression and expansion algorithm with and without equalization

*Allocation Units:* 2 for Compress / Expand; 3 for Cmp / 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.

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 compressor is a soft-knee compressor, so the transition from compressed to linear is quite 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.



**Compressor/Expander (optional EQ not shown)**
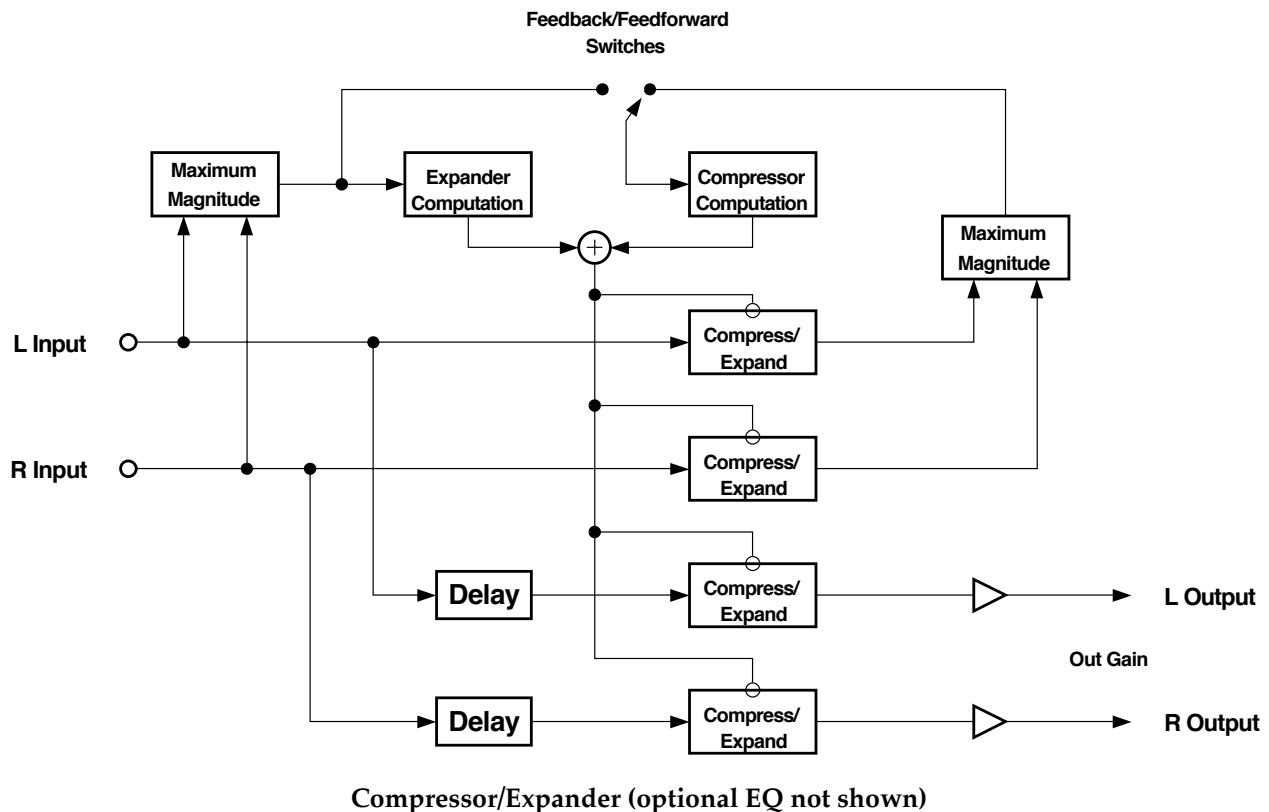
To determine how much to compress or expand the signal, the compressor/expander must measure the signal level. Since musical signal levels will change over time, the compression and expansion amounts must change as well. You can control how fast the compression or expansion changes in response to changing 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).
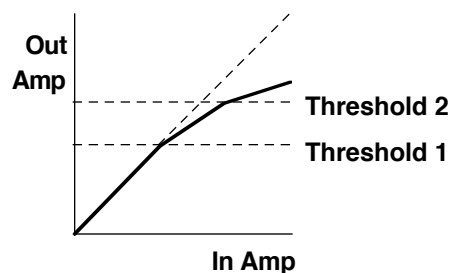


**Two Segment Compression Characteristic**

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.
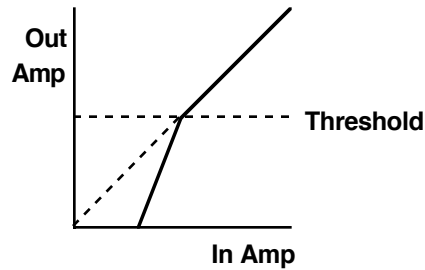


**Expansion Transfer Characteristic**

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 | | |
| Reduction | | | |
| -dB   40   20   12   8   6   4   2   0 | | | |

**PAGE 4 (Comp/Exp + EQ only)**

| Bass Gain | -79.0 to 24.0 dB | Treb Gain | -79.0 to 24.0 dB |
|-----------|------------------|-----------|------------------|
| Bass Freq | 16 to 25088 Hz | Treb Freq | 16 to 25088 Hz |
| Mid Gain | -79.0 to 24.0 dB | | |
| Mid Freq | 16 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 pre-delay). 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.
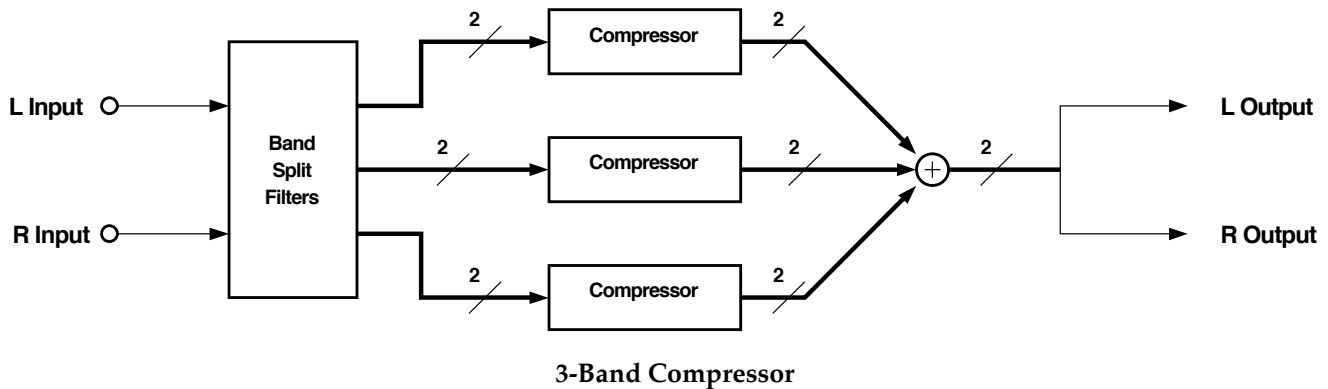
## Comp/Exp + EQ:

| | |
|---|---|
| <u>Bass Gain</u> | 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. |
| <u>Bass Freq</u> | The center frequency of the bass shelving filter in intervals of one semitone. |
| <u>Treb Gain</u> | 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. |
| <u>Treb Freq</u> | The center frequency of the treble shelving filter in intervals of one semitone. |
| <u>Mid Gain</u> | 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. |
| <u>Mid Freq</u> | 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. |
| <u>Mid Wid</u> | 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. |

# FXAlg #956: Compress 3 Band

### Stereo soft-knee 3-frequency band compression algorithm
*Allocation Units:* 4

The 3-band compressor divides the input stereo signal into 3 frequency bands and runs each band through its own stereo 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.

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.



**3-Band Compressor**

For details about the compression action, see the section in this book on FXAlg #950.

## Parameters:

**PAGE 1**

| In/Out | In or Out | Out Gain | Off, -79.0 to 24.0 dB |
|---|---|---|---|
| FdbkComprs | In or Out | Crossover1 | 16 to 25088 Hz |
| Signal Dly | 0.0 to 25.0 ms | Crossover2 | 16 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 Band 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". |
| 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 pre-delay). This allows the compression to appear to take effect just before the signal actually rises. |
| Crossover*N* | 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 expander's envelope detector. Smoothing will affect the attack or release times when the smoothing time is longer than one of the other times. |
| Ratio*Band* | (*Band* is Low, Mid or High) The compression ratio. High ratios are highly compressed; low ratios are moderately compressed. |
| Thres*Band* | (*Band* is Low, Mid or High) The threshold level in dBFS (decibels relative to full scale) above which the signal begins to be compressed. |

# FXAlg #957: Gate ·
# FXAlg #958: Super Gate

### Signal gate algorithms
*Allocation Units:* 1 for Gate; 2 for Super Gate

Gate and Super Gate do 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.
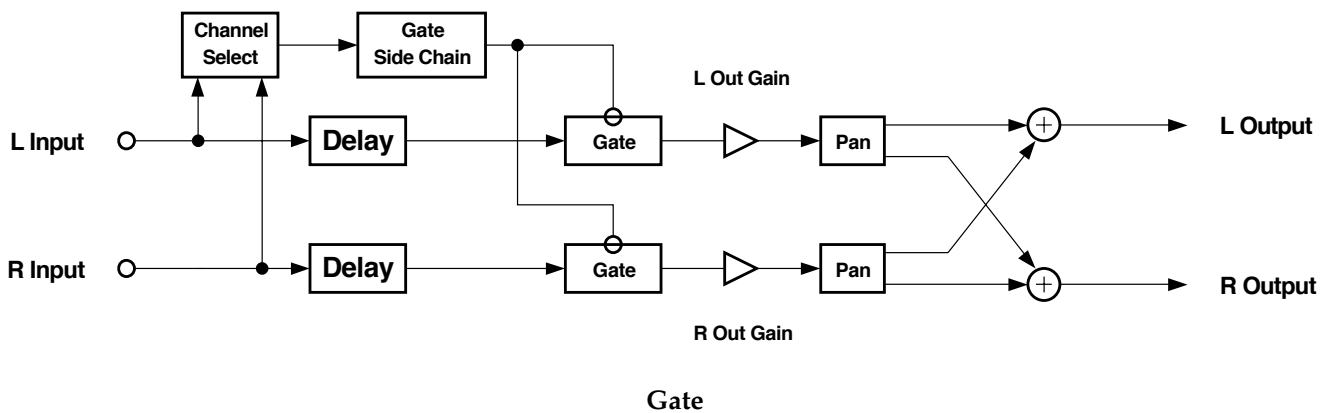


**Gate**

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.

Super Gate will behave differently depending on whether the Retrigger parameter is set to off or 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 on for percussive sounds. For the simpler Gate, there is no Retrigger parameter, and it is as if Retrigger is always on.
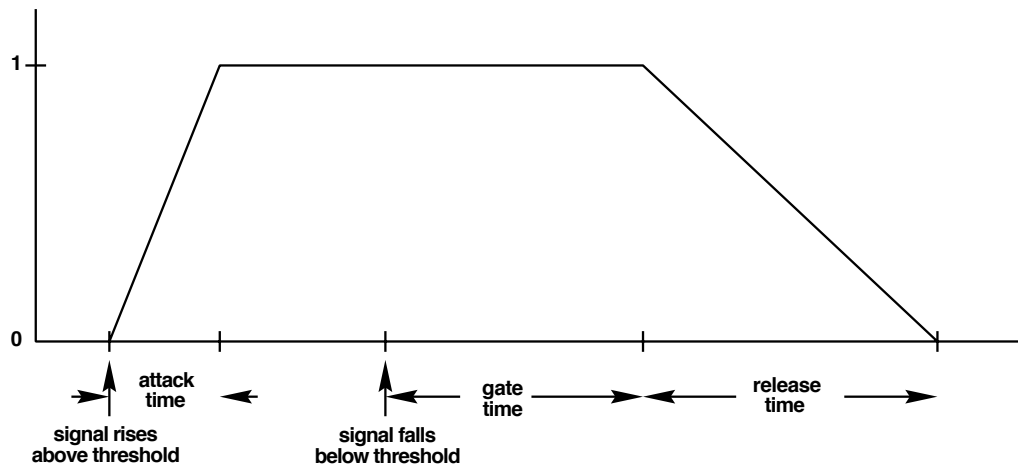
**Signal envelope for Gate and Super Gate when Retrigger is "On"**

If Retrigger is off (Super Gate 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.
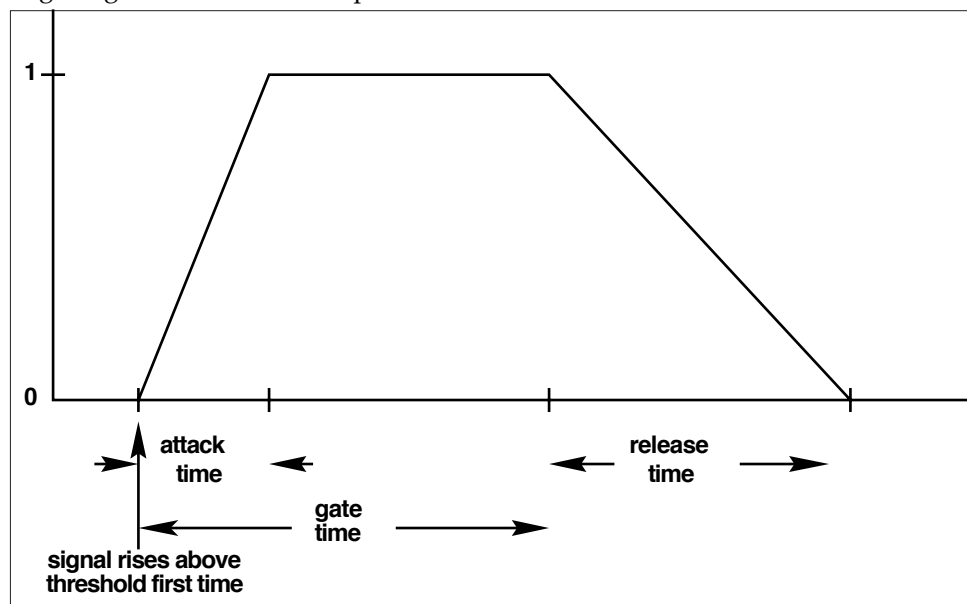


**Super Gate signal envelope when Retrigger is "Off"**

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 Super Gate (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:

**PAGE 1**

| 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 Pan | -100 to 100% | R Pan | -100 to 100% |
| SC Input | (L+R)/2 | | |

**PAGE 2**

| Threshold | -79.0 to 24.0 dB | Gate Time | 0 to 3000 ms |
|---|---|---|---|
| Ducking | On or Off | Atk Time | 0.0 to 228.0 ms |
| Retrigger* | On or Off | Rel Time | 0 to 3000 ms |
| Env Time* | 0 to 3000 ms | Signal Dly | 0.0 to 25.0 ms |
| Reduction | | | |
| -dB    60    40    *    16    *    8    4    0 | | | |

*Super Gate only

**PAGE 3 - (Super Gate)**

| SCBassGain | -79.0 to 24.0 dB | SCTrebGain | -79.0 to 24.0 dB |
|---|---|---|---|
| SCBassFreq | 16 to 25088 Hz | SCTrebFreq | 16 to 25088 Hz |
| SCMidGain | -79.0 to 24.0 dB | | |
| SCMidFreq | 16 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.

L/R Out Gain    The separate output signal levels in dB for the left and right channels. 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.

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. [Super Gate only]

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.

## Super Gate:

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.

# FXAlg #959: 2 Band Enhancer

### 2-band spectral modifier
*Allocation Units:* 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 two frequency bands using 6 dB/oct hipass and lopass filters. The hipassed 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 effects without adding any gain. In this algorithm, delaying the lopass signal relative to the hipass signal brings out the high-frequency transient of the input signal giving it more definition. Conversely, delaying the hipass signal relative to the lopass signal brings out the low-frequency transient information which can provide punch.

The transfer applied to the hipass 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 | 16 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 hipass band and a lopass 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 hipass signal is delayed.

Hi Mix                    Adjusts the output gain of the hipass signal.

Lo Delay                  Adjusts the number of samples the lopass signal is delayed.

Lo Mix                    Adjusts the output gain of the lopass signal.

# FXAlg #960: 3 Band Enhancer

### 3-band spectral modifier
*Allocation Units:* 2

The 3 Band Enhancer modifies the spectral content of the input signal by boosting existing spectral content, or stimulating new content. First, the input is non-destructively split into 3 frequency bands using 6 dB/oct hipass and lopass filters. 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 at 24 kHz sampling rate. Using what we know about psychoacoustics, phase shifting or delaying certain frequency bands relative to others can have useful effects 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.
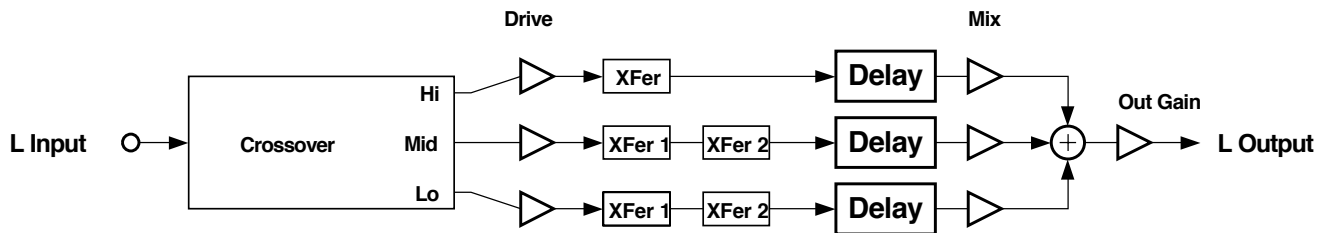


**One channel of 3 Band Enhancer**

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 effect 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.

Lo Enable, Mid Enable, Hi Enable — Turns processing for each band on or off. Turning each of the 3 bands off results in a dry output signal.

Lo Drive, Mid Drive, Hi Drive — Adjusts the input into each transfer. Increasing the drive will increase the effects.

Lo Xfer, Mid Xfer1, Mid Xfer2, Hi Xfer1, Hi Xfer2 — Adjusts the intensity of the transfer curves.

Lo Delay, Mid Delay, Hi Delay — Adjusts the number of samples the each signal is delayed.

Lo Mix, Mid Mix, Hi Mix — Adjusts the output gain of each band.

# FXAlgs #961/962: Tremolo and Tremolo BPM

### A stereo tremolo or auto-balance effect.
*Allocation Units:* 1

Tremolo and Tremolo BPM are 1-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, as follows:
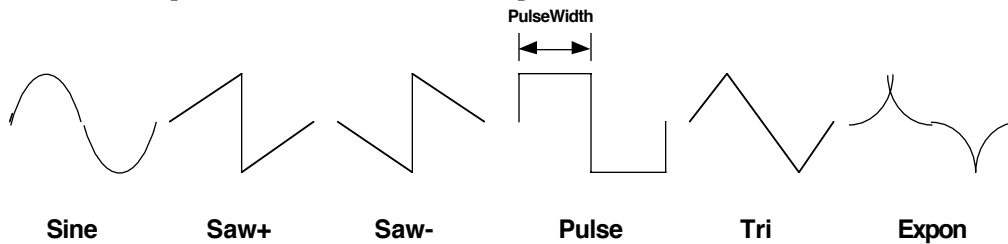


**LFO Shapes available for Tremolo and Tremolo BPM**

An additional shape modifier is "50% Weight". This bends the LFO shape up or down relative to its -6dB point. 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.
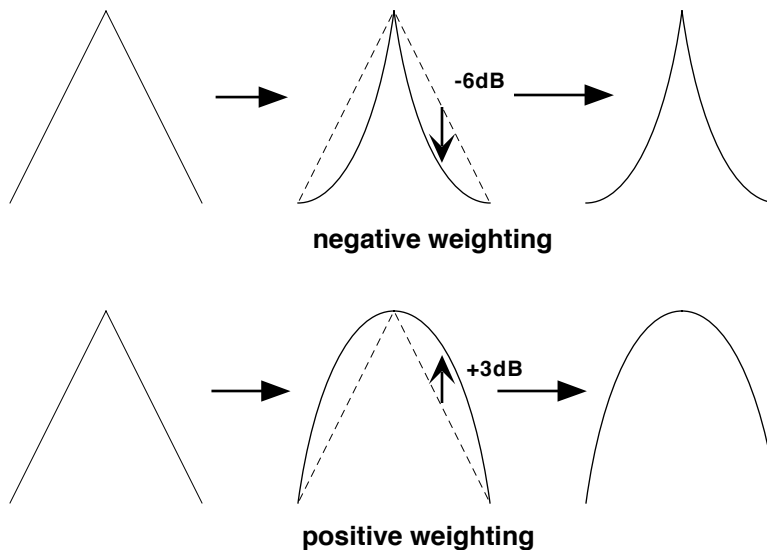


**Action of the "50% Weight" parameter**

Other features include "L/R Phase", which flips the LFO phase of the left channel for auto-balancing applications, and LFO metering, which can be viewed on the bottom of the PARAM2 page.

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

## Parameters (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 (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 (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. When it is set to "System", sources (FUNs, LFOs, ASRs etc.) will have no effect on the Tempo parameter. |
| LFO Rate (Tremolo) | The speed of the tremolo LFO in cycles per second. |
| LFO Rate (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; etc... |
| Rate Scale (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 (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.

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.

# FXAlg #963: AutoPanner

### A stereo auto-panner
*Allocation Units:* 1

"AutoPanner" is a 1-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. This effect provides six different LFO shapes (see the previous section of this book, FXAlgs #961-962), 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.
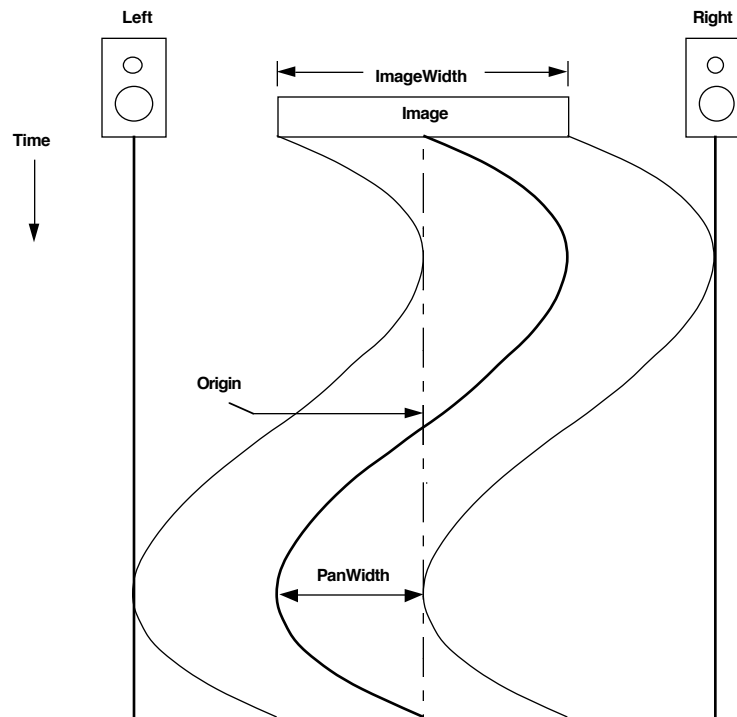


**Concept of stereo autopanning with ImageWidth set to 50%, LFO Shape set to Sine, Origin set to 0%, and PanWidth set to 100%**

## Parameters:

**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% |
| 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 Rate | The speed of the panning motion. |
| Rate Scale | This multiplies the speed of the LFO rate into the audio range. When above 19x, the values increment in semitone steps. These steps are musically accurate when the 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. |

# FXAlg #964: Dual AutoPanner

**A dual mono auto-panner**
*Allocation Units:* 2

"Dual AutoPanner" is a 2-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 (see the section of this book on FXAlgs #961-962), 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.
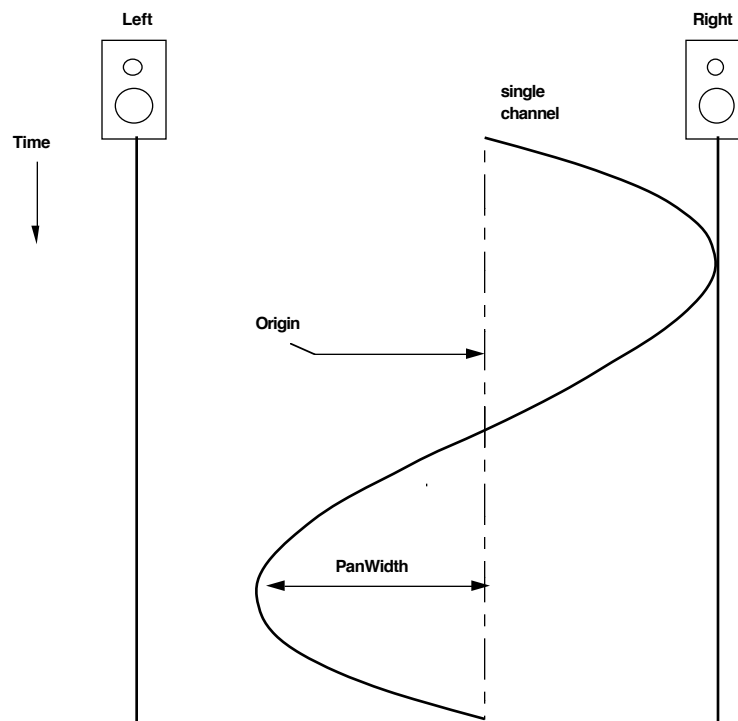


**Concept of mono auto-panning with LFO Shape set to Sine,
Origin set to 15%, and PanWidth set to 100%**

## 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          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 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. |

# FXAlg #965: SRS

## Licensed "Sound Retrieval System®" or SRS™ effect
*Allocation Units:* 1

The SRS™ algorithm has been licensed from SRS Labs, Inc. The following is from an SRS Labs press release:

SRS, the Sound Retrieval System, is based on the human hearing system. It produces a fully immersive, three-dimensional sound image from any audio source with two or more standard stereo speakers. Whether the signal is mono, stereo, surround sound or encoded with any other audio enhancement technology, SRS expands the material and creates a realistic, panoramic sound experience with no "sweet spot" or centered listening position. SRS is single-ended, requiring no encoding or decoding, and uses no artificial signal manipulation such as time delay or phase shift to produce its natural, true-to-life sound image.

The four SRS parameters control the ambience of the image, and may have different optimal settings depending on the amount of stereo content in the inputs. To match the optimal settings specified by SRS Labs, the bass and treble gains should be set to 0 dB. This algorithm will have no effect on mono signals.

## Parameters:

**PAGE 1**

| In/Out | In or Out | Out Gain | Off, -79.0 to 24.0 dB |
|--------|-----------|----------|------------------------|
| Center | Off, -79.0 to 24.0 dB | Bass Gain | -79.0 to 24.0 dB |
| Space | Off, -79.0 to 24.0 dB | Treb Gain | -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. Out Gain is not applied to the signal when the effect is bypassed. |
| Center | The amount of "center channel" can be varied with this control. |
| Space | The width of the image is controlled with this parameter. |
| Bass Gain | The amount of ambience added to the Bass frequencies in the signals. A setting of 0 dB gives a best match to the optimizations of SRS Labs. |
| Treb Gain | The amount of ambience added to the Treble frequencies in the signal. A setting of 0 dB gives a best match to the optimizations of SRS Labs. |

# FXAlg #966: Stereo Image

## Stereo enhancement with stereo channel correlation metering
*Allocation Units:* 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.
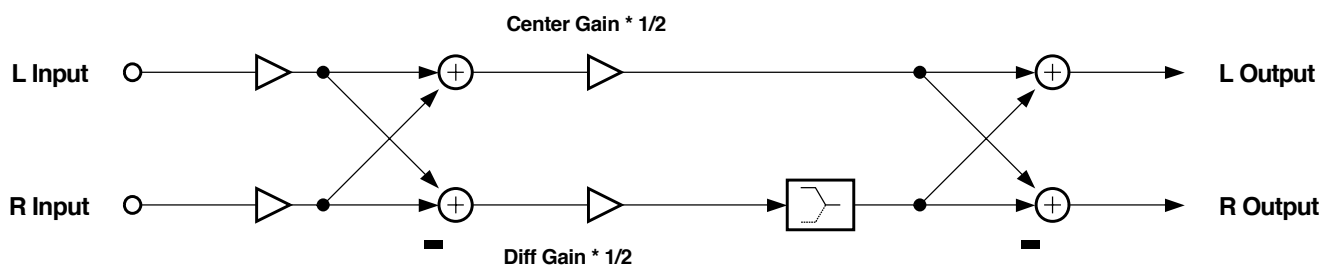


**Block diagram of Stereo Image algorithm**

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 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.

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 (FXAlg #999) 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 | 16 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.

# FXAlg #967: Mono -> Stereo

## Stereo simulation from a mono input signal
### *Allocation Units:* 1

Mono -> Stereo is an algorithm which 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 the mono input, or you may choose to sum the left and right inputs.
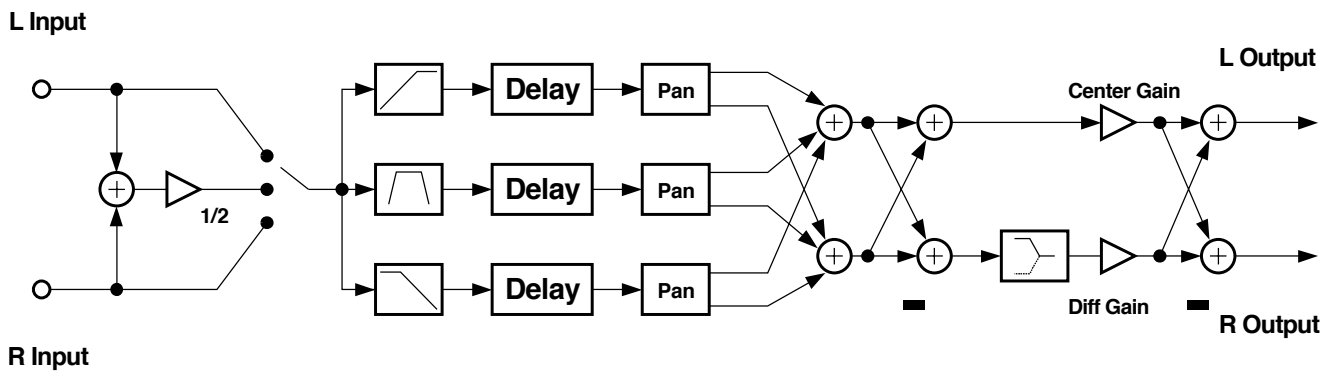


**Block diagram of Mono -> Stereo effect.**

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 the 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 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 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 | 16 to 25088 Hz |

**PAGE 2**

| Crossover1 | 16 to 25088 Hz | | |
|---|---|---|---|
| Crossover2 | 16 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 *Band* — There are three pan parameters: one each for Low, Mid, and High frequency bands. The panning of each band is separately controllable. -100% is fully left and 100% is fully right.

Delay *Band* — There are three delay parameters: one each for Low, Mid, and High frequency bands. The delays are set in milliseconds (ms).

# FXAlg #968: Graphic EQ · FXAlg #969: Dual Graphic EQ

## Dual mono 10-band graphic equalizers
*Allocation Units:* 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 the Figure 1. The dual graphic equalizer has a separate set of controls for the two mono channels.
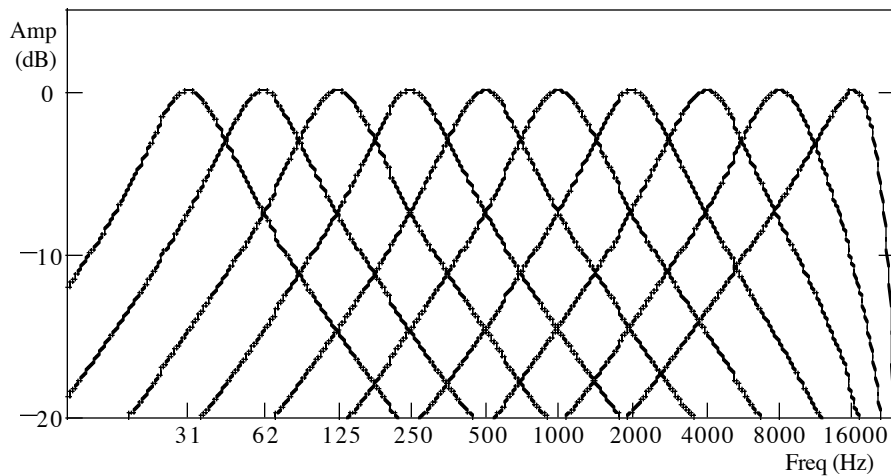


**Filter Response of Each Bandpass Filter**

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 2). To minimize the EQ ripple, you should attempt to center the overall settings around 0 dB.
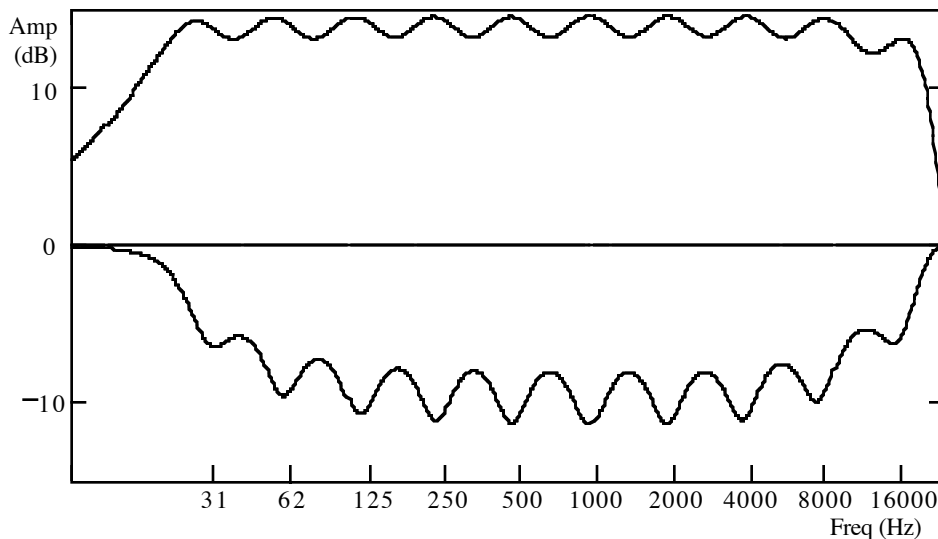


**Overall Response with All Gains Set to +12 dB, 0 dB and -6 dB**

## Parameters (Graphic EQ):

**PAGE 1**

| In/Out | In or Out | | |
|---|---|---|---|

**PAGE 2**

| 31Hz G | -12.0 to 24.0dB | 1000Hz G | -12.0 to 24.0dB |
|---|---|---|---|
| 62Hz G | -12.0 to 24.0dB | 2000Hz G | -12.0 to 24.0dB |
| 125Hz G | -12.0 to 24.0dB | 4000Hz G | -12.0 to 24.0dB |
| 250Hz G | -12.0 to 24.0dB | 8000Hz G | -12.0 to 24.0dB |
| 500Hz G | -12.0 to 24.0dB | 16000Hz G | -12.0 to 24.0dB |

## Parameters (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 24.0dB | L 1000Hz G | -12.0 to 24.0dB |
|---|---|---|---|
| L  62Hz G | -12.0 to 24.0dB | L 2000Hz G | -12.0 to 24.0dB |
| L  125Hz G | -12.0 to 24.0dB | L 4000Hz G | -12.0 to 24.0dB |
| L  250Hz G | -12.0 to 24.0dB | L 8000Hz G | -12.0 to 24.0dB |
| L  500Hz G | -12.0 to 24.0dB | L16000Hz G | -12.0 to 24.0dB |

**PAGE 3**

| R  31Hz G | -12.0 to 24.0dB | R 1000Hz G | -12.0 to 24.0dB |
|---|---|---|---|
| R  62Hz G | -12.0 to 24.0dB | R 2000Hz G | -12.0 to 24.0dB |
| R  125Hz G | -12.0 to 24.0dB | R 4000Hz G | -12.0 to 24.0dB |
| R  250Hz G | -12.0 to 24.0dB | R 8000Hz G | -12.0 to 24.0dB |
| R  500Hz G | -12.0 to 24.0dB | R16000Hz G | -12.0 to 24.0dB |

In/Out
: In Graphic EQ, when set to "In" the equalizer is active; when set to "Out" the equalizer is bypassed.

L In/Out, R In/Out
: In Dual Graphic EQ, when set to "In" the specified channel's equalizer is active; when set to "Out" the specified channel's equalizer is bypassed.

*freq* G
: In Graphic EQ, gain of the *freq* band in dB.

L *freq* G, R *freq* G
: In Dual Graphic EQ, gain of the left-channel or right-channel *freq* band in dB.

# FXAlg #970: 5 Band EQ

### Stereo bass and treble shelving filters and 3 parametric EQs
*Allocation Units:* 3

This algorithm is a stereo 5 -band equalizer with 3 bands of parametric EQ and with bass and treble tone controls. The user has control over 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 controls for the two stereo channels are ganged.

## Parameters:

**PAGE 1**

| In/Out | In or Out | | |
|---|---|---|---|
| Bass Gain | -79.0 to 24.0 dB | Treb Gain | -79.0 to 24.0 dB |
| Bass Freq | 16 to 25088 Hz | Treb Freq | 16 to 25088 Hz |

**PAGE 2**

| Mid1 Gain | -79.0 to 24.0 dB | Mid2 Gain | -79.0 to 24.0 dB |
|---|---|---|---|
| Mid1 Freq | 16 to 25088 Hz | Mid2 Freq | 16 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 | 16 to 25088 Hz | | |
| Mid3 Width | 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.

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 filters 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 filters in intervals of one semitone.

Mid$n$ 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.

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

Mid$n$ 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.

# FXAlg #998: FXMod Diagnostic

### FXMod source-metering utility algorithm
*Allocation Units:* 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 VAST 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 through #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 through #8 provide better resolution, but the range is limited to 0 though +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. To use the algorithm, save a Multieffect and Studio containing the algorithm, then go to one of the FXMod pages of your Program or Setup (with the Studio selected). Select the FX bus which contains the Multieffect using the FXMod Diagnostic algorithm, and choose one of the meter parameters (Bipole N or Monopole N). 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 re-enter the Studio and Multieffect editor where you can view the meters on parameter page 2.

### 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. |

# FXAlg #999: Stereo Analyze

### Signal metering and channel summation utility algorithm
*Allocation Units:* 1

Stereo Analyze is a utility algorithm which 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, 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/or differences between 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 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 values delay the left channel, while negative values delay 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). |