Radical Effects for Pro Tools

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Intro

SoundBlender™ is a pair of software plug-ins, PitchBlender and TimeBlender, for creating radical sound modifications. Each plug-in combines high-quality signal processing algorithms with a powerful modulation structure to create an almost limitless variety of unique effects.

System Requirements

SoundBlender is a pair of software plug-ins for the Digidesign TDM environment. To use SoundBlender, you must have a Digidesign Pro Tools system, version 5.0 or later, equipped with TDM and a 'classic' PCI DSP Farm, MixFarm, or HD card, running on a Macintosh computer with OS 9.1 or later..

Each instantiation of a SoundBlender plug-in processor uses one DSP chip on your DSP farm card. Your current setup must have at least one free DSP.

This manual assumes that you are familiar with the Digidesign Pro Tools TDM environment. If you are uncertain about how DSP plug-ins work within the TDM environment, please read the Digidesign documentation and spend some time getting comfortable with the basic TDM system and the DSP plug-ins that come with it before using this package.

Installation

SoundBlender™ software comes on a UltraTools CD-ROM and uses an iLok USB hardware key device for software *authorization*. The UltraTools CD-ROM contains software for every current Wave Mechanics Pro Tools TDM plug-in in the UltraTools family. The SoundBlender plug-in that is authorized by the iLok will run without restriction whereas other Wave Mechanics plug-ins (not authorized) will run in Demo Mode for a limited period of time.

Install SoundBlender (Macintosh):

- Exit from Pro Tools and any other applications that use the Digidesign 'DAE'.
- Insert the WaveMechanics™ Plug-ins CD-ROM into your computer and double-click the WaveMechanics™ installer application. Click on the *Continue* button with the mouse when the initial 'splash' screen appears.
- Read the license agreement. If you agree to these terms, click the *Accept* button. If you don't agree, click *Decline*, and the installation will be terminated.
- Read the *Read Me* file for last minute updates and important information. Click on the *Continue* button.
- In the installer dialog box, select either the UltraTools product (non-authorized plug-ins will run in demo mode) or use the Custom Install option and select the PurePitch™ product and any other plug-ins you wish to install.
- In the installer dialog box, select the drive on which to install PurePitch™. Select the same drive on which Pro Tools is installed.
- In the installer dialogue box, click on the *Install* button.
- When authorizing any or all of the component plug-ins of the UltraTools bundle, the iLok USB key is used (the older key disks are no longer used for authorization). All Digidesign HD and current Mix and 001 systems ship with the iLok key. If you have purchased an upgrade to UltraTools or any of its component plug-ins, a license card is included in the package and you can proceed to authorize your component plug-ins
- If you purchased UltraTools or any of its component plug-ins for the first time, you must contact Wave Mechanics to register your product and receive the authorization appropriate to your platform. Any of the plug-ins can run, fully functionally, for 30 days until your authorization is completed.

- If you have a Mac with the USB, the iLok is the means of authorization. If your Digidesign system does not already have an iLok key, one can be purchase from Wave Mechanics at a nominal cost. When you register your purchase with Wave Mechanics, an iLok license card will be promptly sent to you to permanently authorize your plug-ins. Be sure to carefully remove the small license card (15mm x 25mm) from the larger plastic card.
- If you have an older, pre-USB Mac then the challenge and response codes are the appropriate means of authorization. To obtain the challenge code, while starting up Pro Tools (or the authorizer application), the challenge code will be displayed. The challenge code can be copied and sent to Wave Mechanics by email (support@wavemechanics.com) if your purchase is already registered with Wave Mechanics or along with your registration. Be sure to include the product name and serial number as listed on your product registration card along with the challenge code. A response that permanently authorizes your plug-ins will be promptly returned.
- If you have an iLok and the iLok license card for the plug-ins you wish to authorize, simply start up Pro Tools or the authorizer application for that product. Displayed is a window with two button options: *Authorize*, or *Quit*. Click on the *Authorize* button to authorize. Confirm that "Use license card" is checked and click on *Next>*. If the license card is not inserted into the iLok, you will be prompted to insert a valid license card for the plug-in purchased, then click on *Next>* and your plug-in is authorized.

If the installation completed successfully, PitchBlender™ and TimeBlender™ should now appear in the *inserts* dialog box next time you start Pro Tools. The installation should have installed the following on your computer:

- A *Wave Mechanics* folder (located in the drive or folder you specified during installation), containing a 'readme' file and the user manual.
- A copy of the PitchBlender™ and TimeBlender™ plug-in, located in your *TDM Plug-Ins* or Plug-Ins folder, located within your *DAE Folder*, which is located in your *System Folder*.
- The PitchBlender™ and TimeBlender™ settings files, located in your *Plug-In Settings* folder, located within your *DAE Folder*, which is located in your *System Folder*.
- If you are upgrading, the previous version of SoundBlender will be moved to the *Removed Items* folder within the *Wave Mechanics* folder.

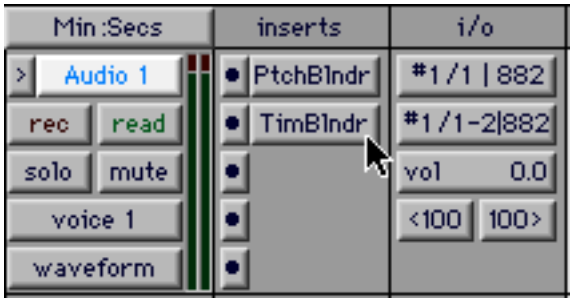
Please take a moment to fill out the enclosed registration card.

Basics

Within Pro Tools, to process a track with one of the SoundBlender plug-ins, 'click' on the Inserts button for that track, and select either the PitchBlender or TimeBlender plug-in from the pop-up menu.

Each time this is done, a new PitchBlender or TimeBlender processor is created. You may create as many PitchBlender or TimeBlender processors as you want, until you run out of DSP resources (each processor uses one of the DSPs available on your DSP farm card).

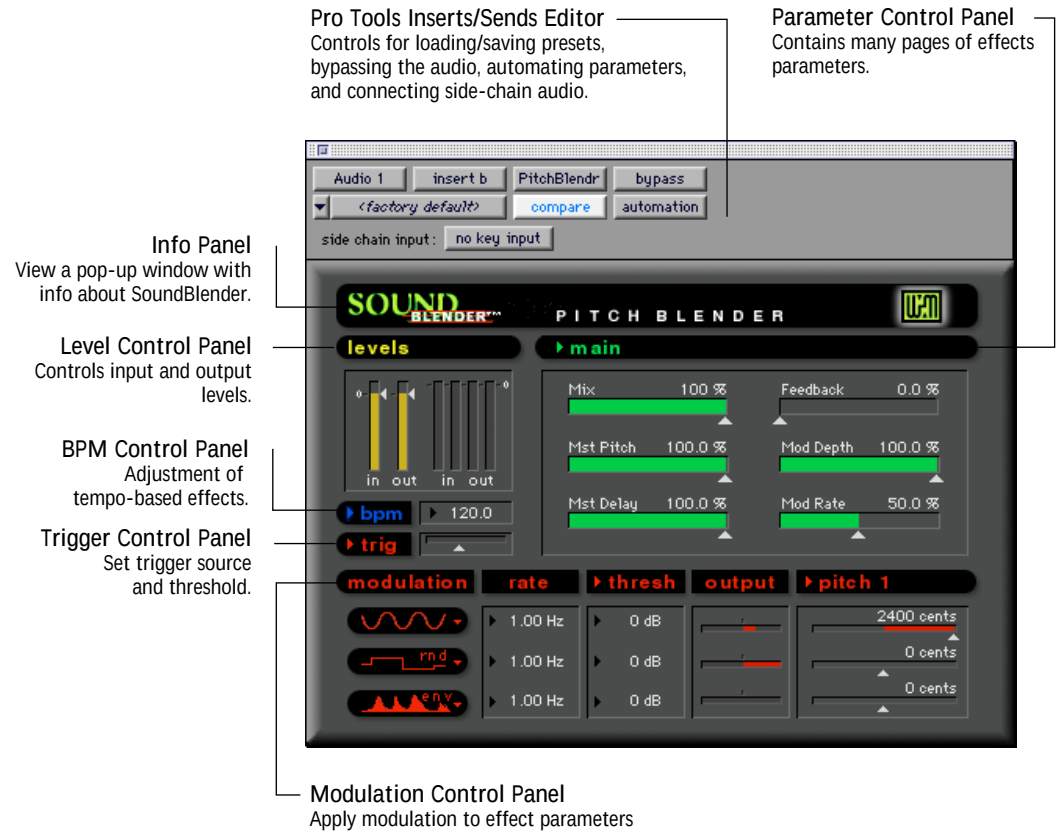
To access the control panel for either PitchBlender or TimeBlender, within Pro Tools, click on the PitchBlender™ or TimeBlender™ insert button, in either the Edit window, or the Mix window.



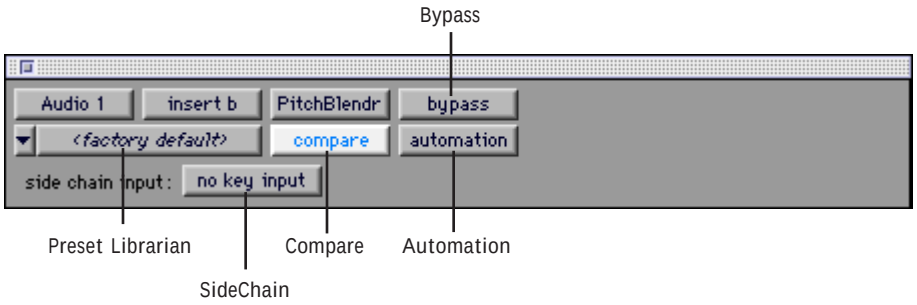
The SoundBlender™ Control Panel

The plug-ins in the SoundBlender family share very similar control panels, and this section will discuss the common features that both PitchBlender and TimeBlender share, and help you to navigate the user-interface. The SoundBlender control panels are designed to be simple and uncluttered, while giving easy access to a huge amount of adjustable parameters.

The control panel for PitchBlender is shown below:



ProTools Inserts/Sends Editor



If you are using SoundBlender within ProTools, you will see the ProTools Inserts/Sends Editor Panel at the top of SoundBlender's control panel.

Preset Librarian

Because of it's extensive programmability, SoundBlender comes with scores of effect presets. We definitely recommend perusing the SoundBlender Preset Library to get a feel for the different sounds that can be created. To load a preset, simply click on the preset librarian button and choose one of the available presets.

Bypass

To compare the difference between the original audio and the SoundBlender processed audio, click on the Bypass button. In most cases, SoundBlender's effect will be pretty obvious — this was not intended to be a subtle plug-in!

Compare

If you'd like to compare the difference between a stored preset and an edited version, use the compare button. After first instan-

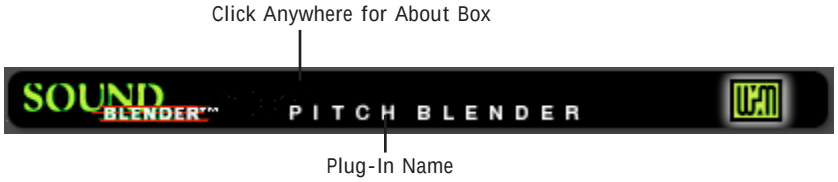
tiating a SoundBlender plug-in, or after loading a preset, the compare button will be grayed out. After adjusting a parameter, the compare button will light up. To hear the difference between the original preset and the changed version, simply click on the compare button. Click again to toggle between the edited and the original versions.

SideChain

SoundBlender has the unique ability to use a *SideChain* to dynamically alter the sound of its effects processing. The side-chain is simply an alternate audio source (other than the track being processed). This alternate audio source can be patched into SoundBlender's modulation section to drive an envelope detector or a gate, which in turn can be patched to any of SoundBlender's modulatable parameters.

To use an audio sidechain, click on the sidechain button, and select the desired source (either an input, or a bus). To use a recorded track as a source for the sidechain, route its output to one of the ProTools busses, or create a send to one of the busses.

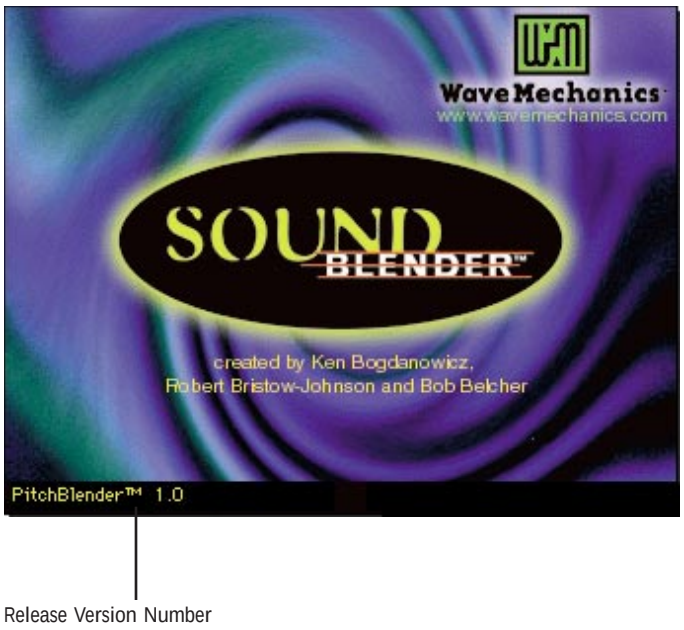
SoundBlender Info Panel



Plug-In Name

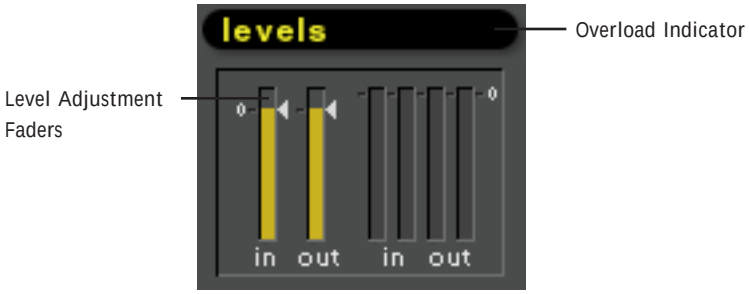
The plug-in name is shown here so that you easily tell which of the SoundBlender family of plug-ins you are currently running.

About Box



Click anywhere in the SoundBlender Info Panel to display the *About Box.* In the about box you'll find the program version, credits, and other info.

Levels Control Panel



Level Adjustment Faders

The level adjustment faders are used to control the overall input and output level. SoundBlender has the ability to add large amounts of gain with its resonant filters and feedback, and it may sometimes be necessary to reduce the input level to prevent digital clipping. The level controls may also be used to add up to 6dB of input or output gain.

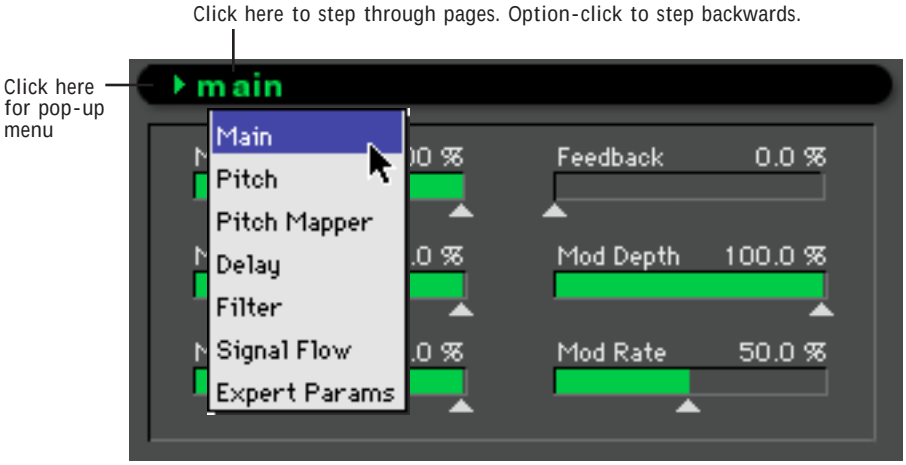
Meters

The input and output meters indicate the signal level relative to the maximum possible digital level, 0 dB. The top, red meter bar indicates possible clipping. With the optional soft-clipping engaged (see the expert control panel), it may be possible, or even desirable to drive the signal into the red range without the nasty artifacts typically associated with digital clipping.

Overload Indicator

The overload indicator will light up if the signal anywhere within the SoundBlender algorithm exceeds the available headroom, resulting in hard-clipping. This *will* produce nasty digital clipping artifacts and should be avoided by reducing the input level.

Parameter Control Panel



The Parameter Control Panel is where you'll find the adjustable effects parameters for all of the SoundBlender plug-ins. The most important thing to know about this control panel is that there are multiple pages of parameters. The pages can be accessed with a pop-up menu by clicking on the ▸ symbol next to the page name. You can also step through the pages by simply clicking on the page name.

Each of the SoundBlender plug-ins has a different set of parameters. Refer to the PitchBlender or TimeBlender manual section for a description of its parameters.

Using the Faders

To adjust a parameter, simply click on the triangular fader 'handle' and drag it to a new position. To get finer adjustment resolution, hold down the command or Ctrl key while dragging the fader handle. Click in the fader 'slot' to move it directly to a new location.

Using the Keyboard

To change a parameter value with the keyboard, click on the displayed value with the mouse, and enter a new value with the keyboard, followed by return or enter. While the parameter value is highlighted, the up/down arrow keys can be used to nudge the parameter value. For non-numeric parameters, the up/down arrow keys will function, but numeric entry will be disabled. The tab key may be used to select a different parameter for keyboard entry.

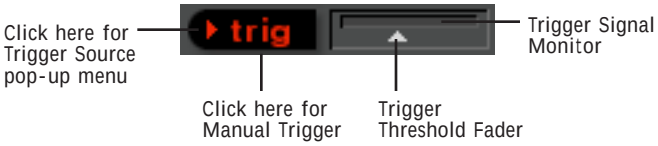
BPM Control Panel

Use the BPM control to to dial in a master tempo for rhythmically correct delay and modulation effects. The tempo is adjusted in beats-per-minute (BPM). Presets that use rhythmic delay and modulation usually have BPM as part of the preset name.

To adjust the BPM setting, click on the displayed value, and enter the new value with the keyboard, followed by enter. Or, click on the ▸ symbol to use a pop-up fader for adjusting the BPM value.

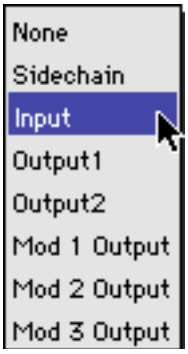
For more information on setting delays and LFOs for rhythmic patterns, see the Modulation Control Panel description, or the delay parameter descriptions in PitchBlender and TimeBlender.

Trigger Control Panel



The SoundBlender modulation section has several modulation types that use an audio signal to trigger a modulation effect, or use the detected level of the audio signal as a modulation source. The trigger control panel is used to select a trigger source, and to adjust the trigger threshold.

Trigger Source



Click on the ▸ in the trigger control panel to view a list of possible trigger sources. Select the desired trigger source from the list. To use the sidechain input, be sure to also select a sidechain source from the ProTools Insert/Sends editor panel (see page 6).

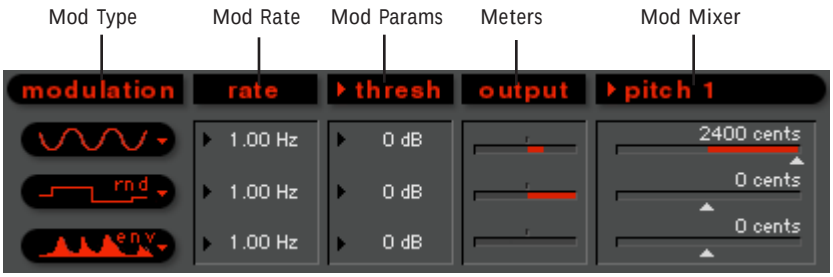
Trigger Threshold

Once a trigger source has been selected, the trigger signal monitor will display the level of the selected signal. Use the Trigger Threshold Fader to adjust the trigger threshold relative to the visual monitor.

Manual Trigger

To manually trigger a modulation effect, use the mouse to click as indicated above. This manual trigger can be recorded as an automation event.

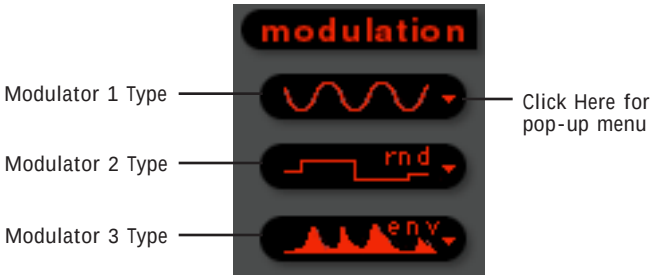
Modulation Control Panel



The Modulation Control Panel is the most interesting, and most complex part of the SoundBlender plug-ins. With modulation, you can dynamically vary effects parameters, using either a built-in oscillator, or in response to changes in the input signal or a sidechain input signal. The ability to smoothly vary effects parameters makes common effects much more interesting, and makes SoundBlender a lot of fun to use.

If you've programmed a synthesizer, much of SoundBlender's modulation section will look familiar. There are three modulation sources which can be mixed and routed to dozens of effects parameters. The control panel is arranged in three rows, with each row corresponding to one of the three modulation sources.

Mod Type



Click on one of the three Modulation Type buttons to select the desired modulation type. A pop-up menu will appear listing the available types:

Sine, Square, Triangle, SawTooth Oscillators

The Oscillator modulators will produce continuous, periodic modulation signals, like the standard LFO section on a synthesizer. The rate of modulation can be changed by adjusting the Modulation Rate parameter, described below. The waveshape will also be affected by the Duty Cycle, Attack Time, Decay Time, and Polarity parameters, also described below.

When a trigger occurs, the oscillators will reset to their starting value. If you don't wish to have the input trigger affect the oscillators in this way, set the Trigger Source (described below) to none.

Triggered Square, Ramp, Triangle

The triggered modulation types will generate some sort of modulation in response to a trigger signal. The signal used for triggering will depend on the setting of the Trigger Source parameter, described below.

When the signal level goes above the trigger threshold, the modulator will produce a single cycle of the specified waveshape. The time taken to complete this cycle is the same as for the

Hint:

To insure perfectly repeatable modulation in a ProTools session, use the manual trigger to synchronize the oscillators. Record the trigger event with ProTools automation at the desired synchronization point.

Oscillator modulation types, and is controlled by Modulation Rate parameter.

Triggered Random

The Triggered Random type will generate a new, constant random modulation value in response to each trigger event.

Toggle Square, Toggle Ramp

The toggle modulation types alternate between maximum and minimum modulation values in response to successive triggers. The Toggle Square type will abruptly switch between the on and off states. The Toggle Ramp will smoothly ramp between the two states. In this case, the transition speed is controlled by the modulation rate.

Envelope Detector

The envelope detector generates a modulation signal that varies with the level of an audio signal. The signal used for envelope detection will depend on the setting of the Trigger Source parameter, described below.

When the signal level is below the trigger threshold, the envelope detector will produce no output. As the level increases above the trigger threshold, the amount of modulation will increase.

The attack and decay time modulation parameters are important when using the envelope detector. Typically, fast to moderate attack times and moderate to slow decay times will give the most desirable results.

Gate

Like the envelope detector, the Gate modulation type also uses a trigger source as input. When the level of the trigger source is below the trigger threshold, the modulation output will be zero. When the level goes above the threshold, the modulation output

will change to its maximum value.

The attack time and decay time parameters are used to control the transition speed of the gate's modulation output. Small settings of attack and decay will produce abupt changes in the modulation output. Larger values will produce slower and smoother changes.

Random Numbers, Random Square, Random Ramp

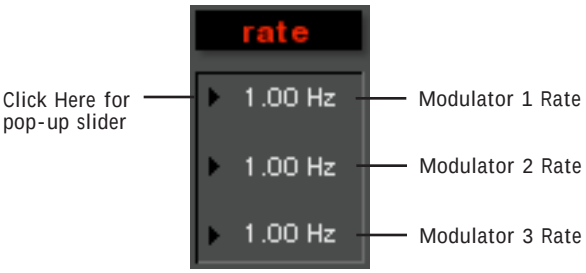
The Random modulation types are used to vary effects parameters in a constantly changing and unpredictable way. The Random Numbers type produces a new random value at a regular rate, determined by the modulation rate setting. For example, if the rate is set to 2 Hertz (which means 2 cycles per seconds), a new random modulation value will be produced twice each second.

Random Square produces a new random value each cycle, but it's output will be limited to either fully on, or fully off. Random Ramp smoothly ramps to each new random modulation value.

Always On, Always Off

These modulation types produce a constant-valued output. To turn off one of the three modulators, use the Always Off modulation type. To modulate by a constant amount, use the Always On modulation type (this *can* occasionally be useful).

Mod Rate



The Mod Rate control adjusts the speed of the modulation effect for the Oscillator, Triggered, and Random modulation types. The adustment can be in either Hertz or in Beats, depending on the setting of the Beats/Hertz parameter below. For settings in Hertz, this parameter controls how many modulations will occur per second. A setting of 2 Hertz means that the modulation will be swept twice per second.

When the Mod Rate adjustment mode is in beats, the rate will be displayed as a fractional number of beats, i.e. 8/24. A setting of 24/24 corresponds to a value of 1 beat (remember your fractions?). Rather than explain the logic of this, here's a simple conversion guide:

4 beats	96/24
half note (2 beats)	48/24
dotted half (3 beats)	72/24
quarter note (1 beat)	24/24
dotted quarter	36/24
quarter triplet	16/24

Hint:

Increasing Hertz increases the rate of modulation. Increasing Beats decreases the rate of modulation.

eighth note	12/24
dotted eighth	18/24
eighth triplet	8/24
sixteenth note	6/24
dotted sixteenth	9/24
sixteenth triplet	4/24
thirty-second note	3/24
thirty-second triplet	2/24

Mod Parameters

The mod parameters section of the Modulation Control Panel also uses a paged interface, like the main parameter area. Click on the ▸ to view and select a modulation parameter. You can also click on the parameter name to step through the list of parameters. The state of the Mod Parameters page will be saved with any preset you create, making it easy to edit parameters that are important to that preset.

Trigger Source

The Trigger Source parameter selects the signal used to trigger the triggered modulation types, or for the envelope detector or gate modulation types. When this parameter is set to Master Trigger, the trigger control panel will control the triggering function. In some cases it may be desirable to use different trigger sources for different modulators, or to disable the triggering effect for a specific modulator. To do this, choose a modulation source other than Master Trigger.

Threshold

When the Master Trigger is not used as the trigger source, the Threshold control is used to set individual trigger thresholds for the three modulators. The adjustment is in dB relative to the maximum signal level of 0 dB.

Attack Time, Decay Time

The Attack and Decay Time parameters are used to smooth out abrupt changes in the modulation signal. The Attack Time parameter controls how fast the modulation signal will increase, and the Decay Time parameter control how fast the modulation signal will decrease. Small Attack/Decay Time settings will result in fast changes in the modulation output and larger settings will result in slower changes in the modulation output.

Sync Mode

The Sync Mode control is used to control how the oscillators are synchronized, and is most useful when trying to create rhythmic effects. Modulator 1 is always the master oscillator, and Modulators 2 and 3 can be slaved to it. To synchronize a Modulator with Modulator 1, set its sync mode to *Slave.* To unlock an oscillator, set its sync mode to *FreeRun.*

Phase Offset

When a pair of Modulators are synchronized and running at the same frequency, their outputs will rise and fall at exactly the same time. However it may be useful for one Modulator to rise while the other Modulator is falling, or for one to rise slightly after the other rises. This type of adjustment can be done by controlling the *phase* relationship of the oscillators.

If the phase of both oscillators is set to zero, they will be in perfect sync, rising and falling at the same time. If the phase of one of the oscillators is set to 180 degrees, they will be perfectly out of phase, with one rising while the other falls. Settings between 0 and 180 degrees will produce varying degrees of phase offset.

Duty Cycle

The duty cycle control is used to alter the shape of the waveforms used for the Oscillator and Triggered Modulation Types. This control alters the relative proportions of the beginning and ending portions of the selected waveshape. The default setting of 50 percent puts equal weight on both portions of the waveshape, leaving the waveshape unchanged. Smaller settings 'speed up' the first part of the waveform, and 'slow down' the later part of the waveform. Settings larger than 50 percent have the opposite effect.

Beats/Hertz

The Beats/Hertz mode switch controls whether the modulation rate is displayed in units of Hertz, or in Beats, relative to the master BPM setting.

Polarity

The Polarity switch controls the range of the Modulator output. When the Polarity is set to Unipolar, the Modulator output is always positive, ranging from 0 to 100 percent. When the Polarity is set to bipolar, the modulator will range from -100 to +100 percent.

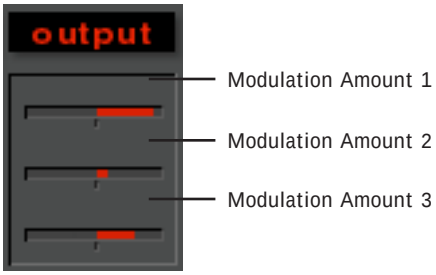
Mapping

The Mapping switch allows an optional non-linear mapping to be applied to the output of the Modulators. The mapping choices are *Linear, Exponential,* or *Logarithmic.* The default setting of *Linear* has no effect on the Modulation output. The *Exponential* setting will tend to create smaller output changes for small modulation values, and larger output changes for large modulation values. The *Logarithmic* setting will have the opposite effect.

Output Level

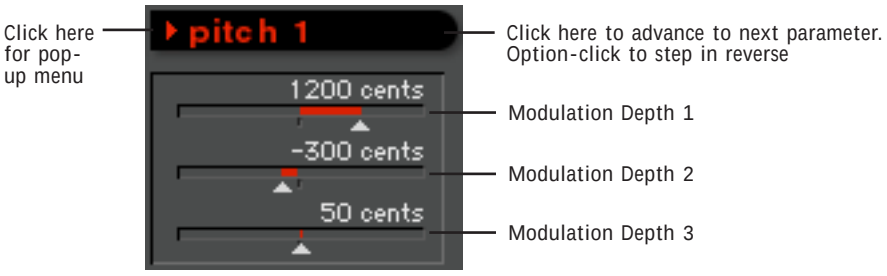
The Output Level control is used to reduce the output level of a single Modulator. This provides an easy way to change the amount of modulation applied to all parameters tied to a specific Modulator. However, to reduce the modulation amount for all Modulators simultaneously, it is usually easier to use the Mod Amount parameter. This parameter is also quite useful when modulating the output level of a modulator.

Mod Output Meters



The Modulation Output Meters provide a convenient way of monitoring the output of the three Modulators. Positive modulation output will be indicated as a bar to the right, negative as a bar to the left. The metering is especially useful for creating synced rhythmic effects, or when using the gate, envelope detector, and triggered modulation types.

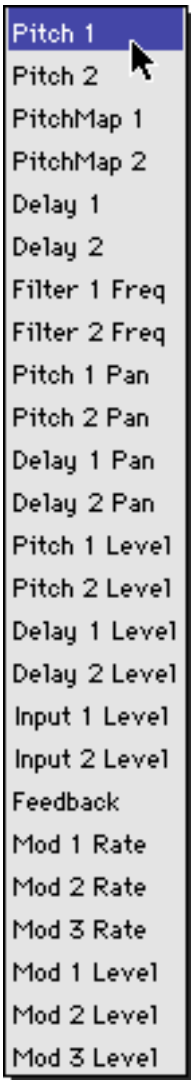
Mod Mixer



The modulation mixer is where the outputs of the three modulators are combined and routed to the various DSP parameters.

Click on the ▸ to view a pop-up menu of parameters available for modulation. The three faders control how the three Modulators will affect the selected parameter.

In the above example, the pitch shift 1 amount is modulated by 1200 cents with modulator 1, -300 cents with modulator 2 and by 50 cents with modulator 3.



Keyboard Shortcuts

Here's a summary of the keyboard shortcuts available in SoundBlender:

Global

Global keyboard commands are always active in SoundBlender:

Control-b (Mac), 'B' (Win) . . . . . Toggle Bypass.

Parameter Edit Fields

These are available after clicking on an editable text field:

Tab . . . . . Advance to Next Editable Param.

Return . . . . . Confirm Entry and Deselect Text

Enter . . . . . Confirm Entry, Allow Further Input

↑, ↓ . . . . . Nudge Parameter Up/Down

Fader Adjustments

These are used before clicking on a fader or pop-up fader:

⌘(Mac), Ctrl (Win) . . . . . Extra-Fine Fader Mode

Option (Mac), Alt (Win) . . . . . Return to Default Value

⌘-Ctrl-0ption (Mac) . . . . . Enable/Disable Automation

⌘-Ctrl (Mac) . . . . . View Automation Playlist

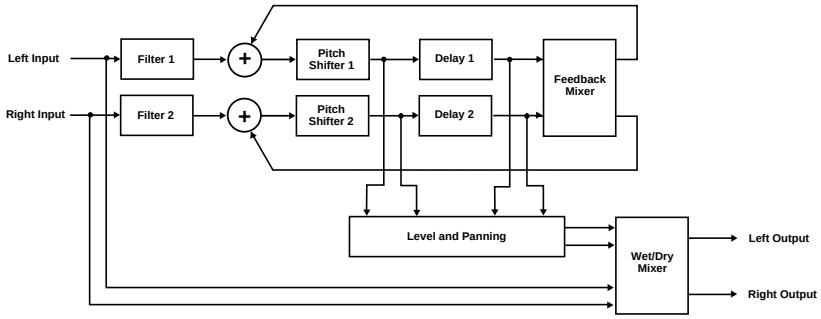
Ctrl-⊞-Alt (Win) . Enable/Disable Automation and display dialog

Overview

The PitchBlender plug-in is two-channel effects processor that combines pitch-shifting, delay, filtering, panning and modulation.

Because the basic effects-processing modules in this plug-in can be re-wired and modulated in many different ways, almost endless varieties of signal processing effects can be created.

PitchBlender contains two pitch shifters, two digital delays, two filters, a mixer, and a feedback matrix. The diagram below shows a typical configuration of these basic modules.



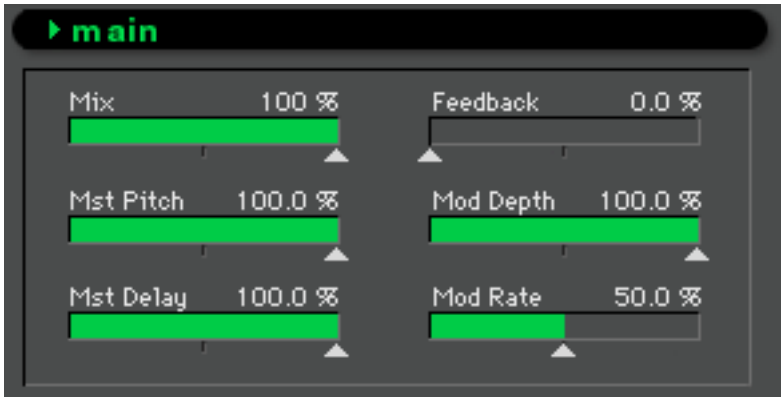
The simplicity of the above diagram, and of the PitchBlender control panel hides much of the depth and power of this plug-in. Many other configurations are possible, and are created simply by changing parameters on the Signal Flow page. Virtually every parameter of the signal processing can be modulated by a wide variety of modulation sources, making it easy to create unique effects that evolve and respond to changes in the input signal.

The best way to get a feel for the range of PitchBlender's effects is to try out some of the presets. Experiment with different parameters and listen to how they affect the sound.

The signal processing parameters for PitchBlender are grouped into pages, and are accessed by either clicking on the ▸ or on the menu name text. This is described in more detail in the Using SoundBlender section.

Main Parameters

The Main parameter page contains the most frequently used parameters for quickly tweaking the sound of a particular preset. Most of the parameters on this page function as *master* controls, and their function will typically



depend on parameter settings found on other pages.

Mix

The Mix parameter controls the mixing of the dry, unprocessed signal and the wet, processed signal. A setting of zero will pass only the dry signal and a setting of 100 percent will pass only the wet signal.

Feedback

The Feedback parameter controls the amount of effected signal to be fed back into the input. Non-zero settings can create regenerative digital delay effects, and interesting pitch sweeps. The adjustment is in percent, where a setting of zero produces no feedback. A setting of 100 percent will produce maximum feedback, and will typically result in near infinite regeneration. The Feedback setting is also affected by the Feedback Balance, Feedback Mix, and Algorithm settings on the Signal Flow page.

Master Pitch

The Master Pitch control is used to simultaneously change the amount of pitch shift for both pitch shifters. It functions as a global control, scaling the Shift 1 and Shift 2 settings on the Pitch page. When Master Pitch is set to 100 percent, the pitch shift amount will be equal to the Shift 1 and Shift 2 settings. A setting of 50 percent will reduce the amount of pitch shift by half. A setting of 0 for Master Pitch will produce no pitch shift.

Master Delay

The Master Delay control is used to simultaneously change the amount of delay for both channels. Like Master Pitch, it functions as a global control, scaling the Delay 1 and Delay 2 settings on the Delay page. The Master Delay control will only affect a delay setting if the delay is programmed in *Time.* When the delay adjustment mode is set to *Beats,* the Master Delay control will have no effect.

Mod Rate

The Mod Rate parameter is a global rate control for the frequencies of the three modulators, and functions similar to the Master Pitch and Master Delay controls. Like the Master Delay control, the Mod Rate parameters will only affect the modulation frequency if its adjustment mode is in *Hertz.* When the adjustment mode is set to *Beats,* the Master Rate control will have no effect for that modulator.

Mod Depth

The Mod Depth parameter is a global control for the output levels of the three modulators. It is an easy way to control the intensity of any modulation-based effects.

Pitch Parameters

PitchBlender contains two channels of pitch shifting, which can be used to create detune and chorus effects, harmony generation, arpeggiation, and wild pitch modulations.

Unlike PurePitch, the pitch shifters in SoundBlender will *not* preserve the formant structure of vocal source material. In other words, large shift amounts *will* cause vocalists to sound like

chipmunks. The shifters have been designed to shift most sources well for small shift amounts (less than 100 cents), and to shift monophonic sources well for larger shift amounts. For difficult material, the pitch shift quality can be optimized by adjusting the parameters on the Expert page.

Shift

The Shift parameters control the pitch shift amount for Pitch Shifter 1 and 2. The adjustments are in cents of pitch change, where 100 cents equals a semitone, and 1200 cents equals an octave. To create double-tracking or detuning effects, use Shift amounts in the range of +/- 5 to 20 cents.

Pan

The output of the two pitch shifters may be panned anywhere within the stereo field by using the Pan 1 and Pan 2 parameters. A Pan setting of -100 will pan the pitch shifter to the left output, a setting of 0 will pan center, and a setting of +100 will pan to the right. A constant power panning algorithm is used to maintain a constant loudness across the stereo field.

Level

The Level parameters are used to adjust the output levels of the two pitch shifters. The adjustment is in decibels, where 0 dB represents unity gain through the pitch shifters, or maximum output level.

Pitch Mapper Parameters

The Pitch Mapper adds intelligent harmony and arpeggiation features to the two pitch shifters. For intelligent harmony generation, the Pitch Mapper analyzes the pitch of the input signal, and dynamically adjusts the pitch shift interval depending on the detected pitch, the selected key, and the selected pitch shift interval.

To create an arpeggiation pattern using the pitch mapper, it is necessary to modulate PitchMap1 or PitchMap2 with one of the modulation sources. If you are unfamiliar with using modulation, see the Modulation Mixer section in Using SoundBlender. To create a straightforward arpeggiation pattern, use a triangle waveform for the modulation source. The amount of modulation will affect the range of the arpeggiation pattern. If the amount of modulation is set to 1200 cents (1 octave), a 1 octave arpeggiation pattern will be created. Lesser amounts of modulation will produce arpeggiation patterns that span a smaller range.

Note that the pitch mapping function is *added* to the amount of pitch shift specified by the Shift 1 and Shift 2 parameters. This allows arpeggiation patterns or intelligent pitch shift intervals to be further detuned or modulated.

PitchMap

The PitchMap parameters determine which pitch mapping function is used for pitch shifter 1 and pitch shifter 2. To disable the pitch mapping functions, set this parameter to *none*. The following pitch maps are available:

none

Disables the pitch mapping function.

chromatic, pentatonic, diatonic, blues, w. african, indian, iranian, iraqi, korean, kukuya, lebanese, mambuti, marimba, mbira, mboko, pelog, renteng, siamese, slendro1, slendro2, thailand scales

The *scale* pitchmaps will map an equal-tempered, chromatic input pitch (like a piano) into the specified scale, in the specified key. When no modulation is used, the pitch of the input is shifted to the closest note from the selected scale. These can be used to create exotic-sounding melodies from a western instrument.

When the pitchmap parameter is modulated, other pitch intervals will be produced, with the output pitch remaining in the chosen key and scale.

major scale

For a steady tonal input, the major scale pitchmap will produce a major scale arpeggio whose root is based on the input pitch. This is different from the *diatonic scale* setting where the output pitch will always be in the specified key. In this case, the key setting is ignored. To produce an arpeggio, the pitchmap parameter must be driven by at least one of the modulators.

diatonic, pentatonic triads

The *diatonic and pentatonic triad* pitchmaps produce an arpeggiated triad in the specified key, with the root of the chord based

on the input pitch. The shifted note will always be a note from the scale of the specified key. To produce an arpeggio, the pitchmap parameter must be driven by at least one of the modulators.

6th, 7th, sus4th chords

The *chordal* pitchmaps are designed to produce an arpeggiated chord in the specified key, with the root based on the input pitch. To produce an arpeggio, the pitchmap parameter must be driven by at least one of the modulators.

pedal chromatic, pentatonic, diatonic, blues, w. african, indian, iranian, iraqi, korean, kukuya, lebanese, mambuti, marimba, mbira, mboko, pelog, renteng, siamese, slendro1, slendro2, thailand scales

The *pedal scale* pitchmaps produce scalar arpeggios with the specified key as the root. To produce an arpeggio, the pitchmap parameter must be driven by at least one of the modulators.

pedal major, minor, augmented, diminished, min 6th, maj 6th, min 7th, maj 7th chords

The *pedal chord* pitchmaps produce chordal arpeggios with the specified key as the root. To produce an arpeggio, the pitchmap parameter must be driven by at least one of the modulators.

major triad, minor triad, diminished triad, augmented triad

The above pitchmaps produce a chordal arpeggios with the input pitch as its root. To produce an arpeggio, the pitchmap parameter must be driven by at least one of the modulators.

diatonic unison, 2nds, 3rds, 4ths, 5ths, 6ths, 7ths, 8ves

The *diatonic* pitchmaps will shift the input pitch by the specified interval, altering the amount of shift so that the output pitch is in the selected key.

5ths and octaves

The 5ths and octaves pitchmap will produce pitch shifts of unison, fifths, and octaves when the pitchmap parameter is driven by a modulation source.

Key

The Key parameter should be set to the key that you are playing in. This is only needed when using the pitchmapping feature.

Tuning

By default, the pitchmapping function is tuned to A440. To use a different reference pitch, adjust the Tuning parameter. The adjustment is in cents, where 100 cents is equal to one half-step. Positive settings will adjust the tuning to be higher than A440, and negative settings will adjust the tuning to be lower.

Attack, Decay

The Attack and Decay parameters control the speed of pitch variation caused by the pitchmappers. Attack adjusts the speed for upward changes in the amount of pitch shift. Decay adjusts the speed for downward changes. Attack and Decay may be adjusted to smooth rapid changes in pitch shift, or to create portamento effects.

Delay Parameters

PitchBlender contains two digital delay lines that are used to create a variety of delay effects like echo, slap delays, and rhythmic effects. By modulating the delay values, chorus, flange, vibrato, and extreme modulation effects may be created.

Delay

The Delay parameters control how much the audio is delayed for the two digital delay lines. The delay time can be controlled in either *Time* or in *Beat*s, depending on the Delay Mode setting on the Expert page. When the mode is *Time*, the setting is in milliseconds, where 1000 milliseconds equals 1 second. For settings in *Beats*, the time value of 1 beat will depend on the setting of the BPM control on the main panel. Look at the Mod Rate control description on page 16 for an explanation of how the beat parameters are adjusted.

Pan

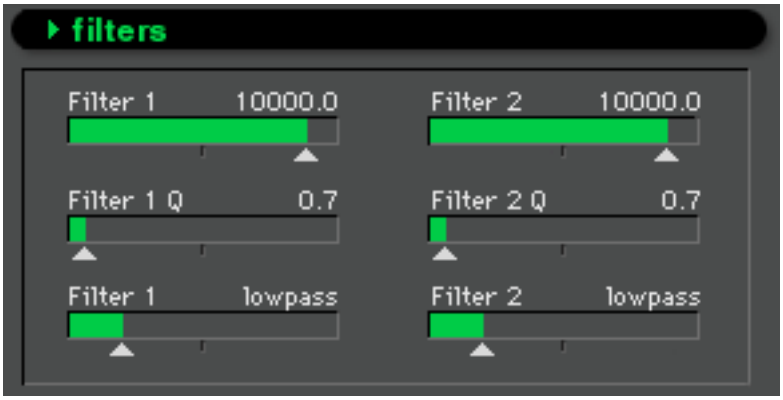
The output of the two delay lines may be panned anywhere within the stereo field by using the Pan 1 and Pan 2 parameters. A Pan setting of -100 will pan the delay output to the left, a setting of 0 will pan center, and a setting of +100 will pan to the right. A constant power panning algorithm is used to maintain a constant loudness across the stereo field.

Level 1 and 2

The Level parameters are used to adjust the output levels of the two delay lines. The adjustment is in decibels, where 0 dB represents unity gain through the delay lines, or maximum output level.

Filter Parameters

PitchBlender contains two filters, which are used to alter the frequency content of the audio, much like the filter section on an analog synthesizer. The filters can be programmed to reduce high



or low frequencies, eliminate or enhance a band of frequencies, or to create highly resonant effects. By modulating the filter frequencies, many interesting synth-like effects can be created.

Frequency

The frequency control is used to adjust the frequency range that is modified by the filters. The exact type of modification will depend on the setting of the Filter Type parameter. For example, if the filter type is 'lowpass', audio frequencies below the frequency setting will be unchanged, while frequencies above will be reduced in level. This is often referred to as the cutoff frequency or center frequency for the filter.

The function of the filters will also be affected by the Filter Config control in the Signal Flow page.

The filter frequencies can be modulated to create dramatic swept filter effects. The modulation adjustment is in octaves, and is always relative to the current frequency setting. For example, if Frequency 1 is set to 100 Hertz, and the modulation is set for 1 octave, the frequency will be modulated from 100 Hertz to 200 Hertz. If Frequency 1 is changed to 500 Hertz, the frequency will be modulated from 500 Hertz to 1000 Hertz.

Filter Q

The Filter Q parameter is used to control the shape of the filter, and its exact function depends on the Filter Type setting. In general, higher settings of Q will produce, narrower, steeper, and more resonant filters. Lower settings of Q will produce gentler filters.

Filter Type

The filters can perform different functions, depending on the setting of the Filter Type parameter:

None . . . . . Bypass.

Lowpass . . . . . Reduce high frequencies.

Peak BPF . . . . . Emphasize a band of frequencies.

Norm BPF . . . . . Pass a band of frequencies.

Highpass . . . . . Reduce low frequencies.

Notch . . . . . Reduce a band of frequencies.

Lowpass

The Lowpass Filter Type is used to reduce the amount of high frequencies in a signal. Frequencies above this setting will be reduced, and frequencies below will be unchanged. Small settings of the Q parameter will produce a gentle lowpass filter. Settings

of Q above 0.7 will produce a resonance, boosting the level near the cutoff frequency.

Peak BPF

The Peak BPF Filter Type is used to emphasize specific range of frequencies in a signal. The band will be centered around the Frequency setting, or center frequency. Small settings of the Q parameter will produce a gentle bandpass filter. Settings of Q above 0.7 will boost the audio level at the center frequency, and can produce very high gain.

Norm BPF

The Norm BPF Filter Type is also a bandpass filter. It is different from the Peak BPF in that the audio at the center frequency will always be at unity gain. High settings of Q for this filter type will produce a narrower bandpass, but will not produce any gain. This is especially useful when used in combination with the Loop Filter Configuration, and will prevent unstable feedback from being produced.

HighPass

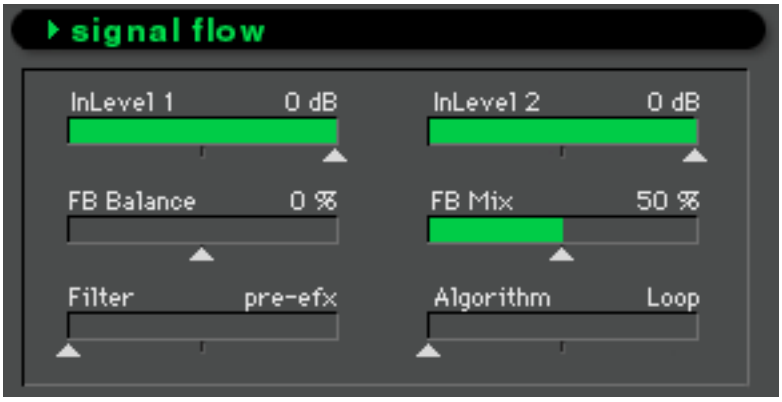
The Highpass Filter Type is used to reduce the amount of low frequencies in a signal. Frequencies below the Frequency setting will be reduced, and frequencies above will be unchanged. Small settings of the Q parameter will produce a gentle highpass filter. Settings of Q above 0.7 will produce a resonance, boosting the level near the cutoff frequency.

Notch

The notch filter will removing a range of frequencies around the center frequency. Higher Q settings for this filter type will produce narrower notches. Lower Q settings will produce wider notches.

Signal Flow Parameters

The parameters on the Signal Flow page are used to re-patch and otherwise alter the flow of audio within the PitchBlender



processing plug-in. These settings can have drastic effects on how the audio is processed.

Input Level

The Input Level parameters are used to independantly control the input level for the two channels of PitchBlender. This can be used for fine-tuning the audio levels, or can be used to turn off the input to one or both of the processing channels.

FB Balance

The Feedback Balance parameter is used to control the relative amount of feedback for the two channels, and works together with the Feedback control on the Main page. A setting of 0 will produce equal amounts of feedback for both channels. A setting of -100 will produce feedback only for channel 1. A setting of 100 will produce feedback only for channel 2. Other settings will smoothly crossfade between these extremes.

FB Mix

The Feedback Mix parameter is used to allow varying degrees of feedback to be mixed between the left and right channels. A setting of 0 will not mix the feedback paths, i.e. the channel 1 output will feed back to the channel 1 input and the same for channel 2. The default setting of 50 percent will produce an equal mix of channels 1 and 2 which is fed back to both channels equally. This can be used to create extremely dense chorus effects, and even reverb-like sound when used together with pitch-shifting and delay modulation. A setting of 100 percent will produce a perfectly crossed feedback path, with the output of channel 1 feeding the input of channel 2 and vice-verse.

Filter Config

The Filter Config parameter controls at what point in the signal chain the filters have their effect.

pre-efx . . . . . Filter after input.

pre-efx+. . . . . Filter after input, summed.

post-pitch . . . . . Filter pitch-shift output.

post-delay . . . . . Filter delay output.

post-efx . . . . . Filter final output.

loop . . . . . Filters in feedback loop.

pre-efx

The pre-efx filter configuration places the filters at the very input of the PitchBlender algorithm, before the pitch shifters and delay lines. Filter 1 operates on the left input channel, and filter 2 operates on the right input channel.

pre-efx+

The pre-efx+ configuration is similar to the above, except that the output of the two filters is mixed before being sent on for further processing. This can be used to create a formant filter effect with the two filters.

post-pitch

The post-pitch setting places the filters at the output of the pitch shifters, before panning and mixing. Filter 1 operates on the output of pitch shifter 1 and filter 2 operates on pitch shifter 2.

post-delay

The post-delay setting is similar to above, placing the filters at the delay line outputs.

post-fx

The post-fx setting places the filters at the very end of the signal chain, after the pitch shifters, delays, and mixing.

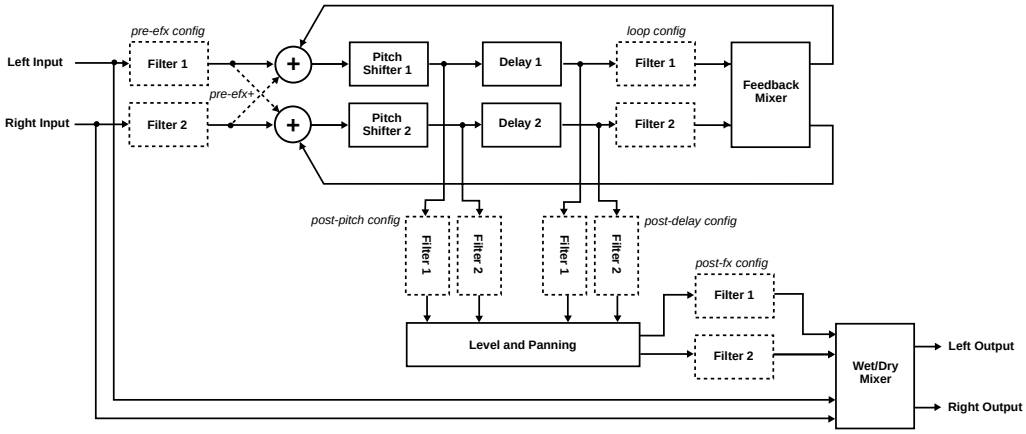
loop

The loop configuration places the filters in the feedback path. This configuration is useful to attenuate high frequencies when using feedback with delay lines or pitch shifters. Be careful when using high Q values in this configuration, as this can easily produce unstable feedback, and potentially very loud sounds.

Algorithm

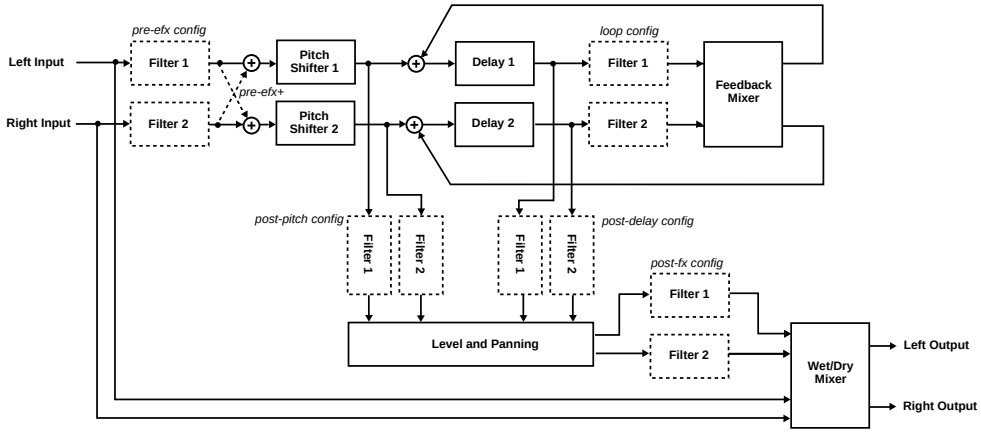
The algorithm setting is used to re-patch the various processing elements within the PitchBlender plug-in. The diagrams below illustrate the possible processing configurations:

Loop Algorithm



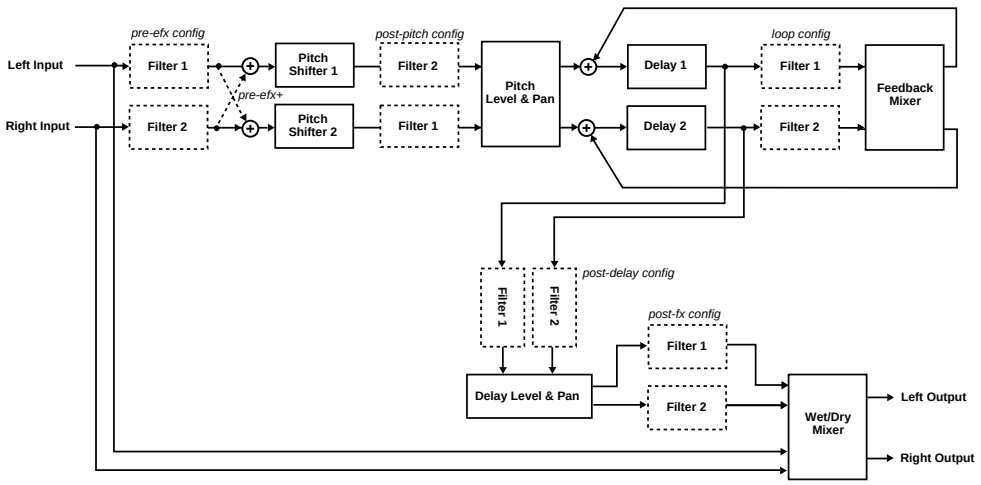
The loop algorithm is used to create delayed, regenerative pitch effects. By feeding back the delayed pitch-shifted audio, the audio is re-processed through the pitch-shifters, creating interesting pitch effects. If very small amounts of pitch-shifting are used, this configuration can be used to create very rich chorus effects, and to create some unique reverb effects.

Tapped Algorithm



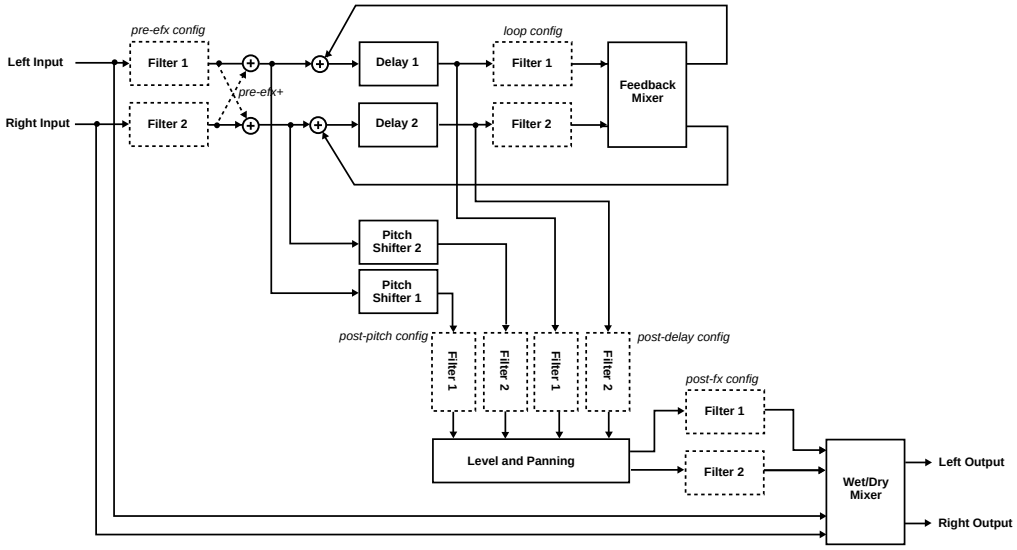
The tapped algorithm is similar to the Loop algorithm, except that the feedback loop is arranged so that only the delayed signal is regenerated. This is useful for creating pitch-shift effects that are then further delayed, echoed, and reverberated. Note that in order to hear any delay effects, the delay levels must be turned on. They are normally defaulted to *off.*

Series Algorithm



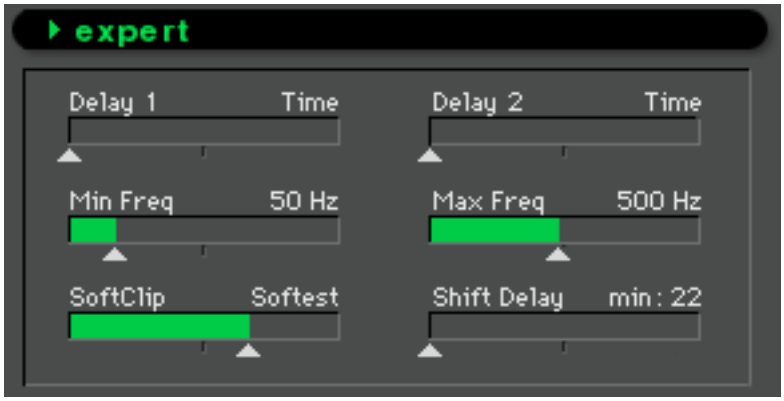
The Series algorithm has the same delay feedback structure as the tapped algorithm. The difference in this case is that the pitch level and panning controls are used to control how much much pitch shifted output is fed into the two delay lines. In this algorithm, to hear any output at all, it is necessary to have both the pitch and the delay levels turned up. This algorithm was designed primarily to create percussive effects by modulating the output levels of the pitch-shifters as they feed into the delay and feedback structure.

Parallel Algorithm



The parallel algorithm is designed so that the pitch shifters and delay lines may be used independantly. In the other three algorithms, the output of the pitch-shifters always feeds into the delay lines. In this case, the delay lines operate on the input, and the pitch-shifting can be used independantly. This algorithm is particularly useful for creating thick chorusing effects.

Expert Parameters



The expert page contains parameters that are used infrequently, but can be used to optimize the audio processing of the PitchBlender plug-in.

Delay Mode

The Delay Mode parameter is used to control how the delay time for the two delay lines is adjusted. When this parameter is set to *Time*, the delay time is adjusted in milliseconds. When this parameter is set to *BPM*, the delay time is adjusted in beats, and is affected by the BPM control on the SoundBlender panel.

Min Freq

The Min Freq parameter is used to optimize the audio quality for the two pitch shifters. This parameter should be set to roughly the lowest frequency present in the audio that you are processing. In general, the pitch-shifters will sound pretty good on most material for small shift amounts, and on single-line material for most shift amounts.

If you are trying to shift polyphonic material over a large shift range, you will get best results if this parameter is set very low.

This will have the side effect of increasing the processing delay through the pitch shifters.

Max Freq

The Max Freq parameter also optimizes the operation of the pitch shifters. This should typically be set to the highest frequency expected to be processed. However, on occasion, smaller settings of this parameter will produce better results.

SoftClip Mode

This parameter controls the operation of the soft-clipping function. When turned to off, soft-clipping will be disabled, and any audio that is boosted beyond the maximum digital level will be clipped in the typically harsh digital way. When soft-clipping is turned on, the audio will be much more gradually clipped, producing more pleasant harmonics, and a more analog sound.

Three soft-clipping modes are available – *soft*, *softest*, and *asymmetrical.* The *soft* mode will preserve the maximum linear range while still clipping softly. The *softest* mode will produce the softest clipping possible, but will also introduce very small amounts of distortion over much of the audio range. The *asymmetrical* mode will clip one side of the audio waveform first, producing a higher proportion of even harmonics.

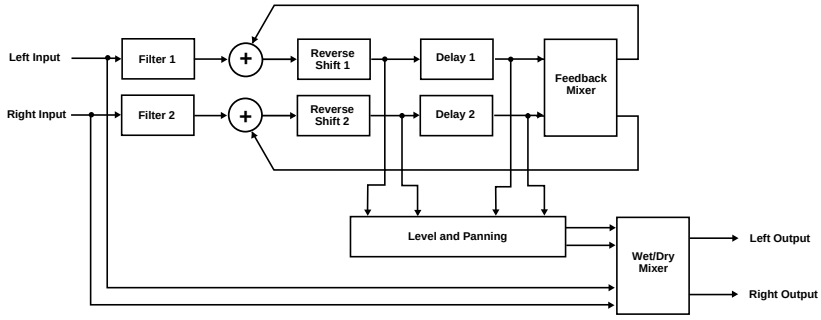
Shift Delay

The Shift Delay parameter is used to optimize the sound of the pitch-shifter by controlling the amount of processing delay. Two choices are possible, minimum and optimum. Minimum will produce minimum delay, at the possible expense of pitch-shifting quality. Optimum will produce the best pitch-shifting, at the expense of processing delay. This parameter will also display the processing delay of the pitch shifters, in milliseconds. This amount is also affected by the Min Freq parameter.

Overview

The TimeBlender plug-in is a two-channel effects processor that combines reverse pitch-shifting with delay, filtering, panning and modulation. A reverse pitch-shifter is an effect found in some hardware effects devices, and was originally created to model the backwards tape effect popularized by Jimi Hendrix. Obviously, playing completely backwards, in realtime, is impossible, because we would need to predict what notes you were planning on play. A reverse pitch-shifter samples small snippets of audio and time-reverse each snippets. The end result is not quite what you would expect, and when combined with feedback, modulation, and filtering is downright stunning.

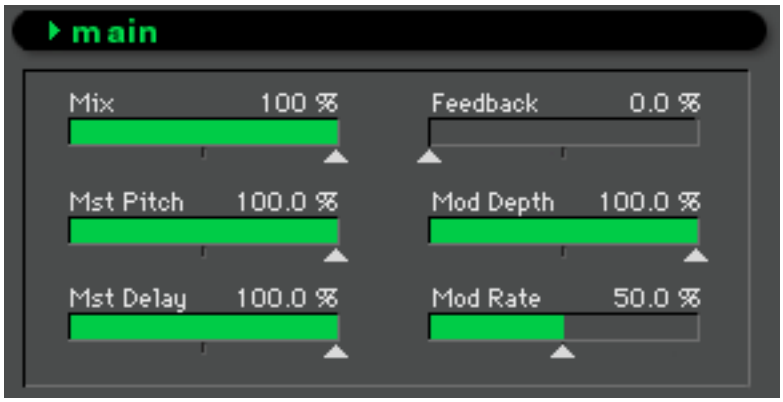
Here is a basic diagram of TimeBlender's processing configuration:



The best way to get a feel for the range of TimeBlender's effects is to try out some of the presets. Experiment with different parameters and listen to how they affect the sound.

The signal processing parameters for TimeBlender are grouped into pages, and are accessed by either clicking on the ▸ or on the menu name text. This is described in more detail in the Using SoundBlender section.

Main Parameters



The Main parameter page contains the most frequently used parameters for quickly tweaking the sound of a particular preset. Most of the parameters on this page function as *master* controls, and their function will typically depend on parameter settings found on other pages.

Mix

The Mix parameter controls the mixing of the dry, unprocessed signal and the wet, processed signal. A setting of zero will pass only the dry signal and a setting of 100 percent will pass only the wet signal.

Feedback

The Feedback parameter controls the amount of pitch-shifted, delayed signal to be fed back into the input. Non-zero settings will create regenerative digital delay effects, and interesting pitch sweeps. The adjustment is in percent, where a setting of zero produces no feedback. A setting of 100 percent will produce maximum feedback, and will typically result in near infinite regeneration. The Feedback setting is also affected by the Feedback Balance, Feedback Mix, and Algorithm settings on the Expert page.

Master Pitch

The Master Pitch control is used to simultaneously change the amount of pitch shift for both revserse shifters. It functions as a global control, scaling the Shift1 and Shift 2 settings on the Pitch and Delay page. When Master Pitch is set to 100 percent, the pitch shift amount will be equal to the Shift 1 and Shift 2 settings. A setting of 50 percent will reduce the amount of pitch shift by half. A setting of 0 for Master Pitch will produce no pitch shift.

Master Delay

The Master Delay control is used to simultaneously change the amount of delay for both channels. Like Master Pitch, it functions as a global control, scaling the Delay 1 and Delay 2 settings on the Delay page. The Master Delay control will only affect a delay setting if the delay is programmed in *Time*. When the delay adjustment mode is set to *Beats*, the Master Delay control will have no effect.

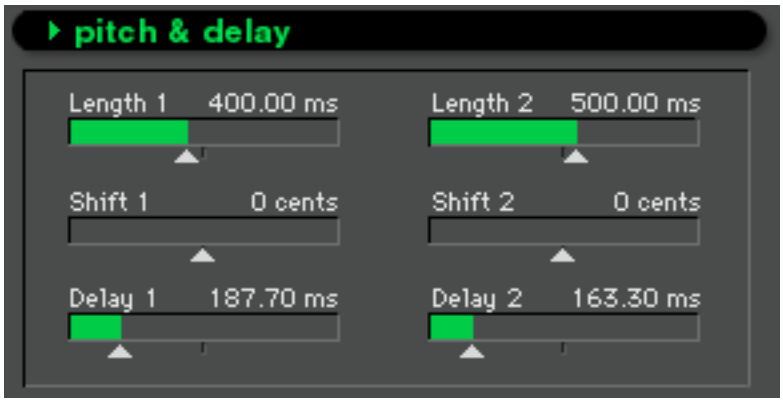
Mod Rate

The Mod Rate parameter is a global rate control for the frequencies of the three modulators, and functions similar to the Master Pitch and Master Delay controls. Like the Master Delay control, the Mod Rate parameters will only affect the modulation frequency if it's adjustment mode is in *Hertz*. When the adjustment mode is set to *Beats*, the Master Rate control will have no effect for that modulator.

Mod Depth

The Mod Depth parameter is a global control for the output levels of the three modulators. It is an easy way to control the intensity of any modulation-based effects.

Pitch and Delay



This page contains the main parameter adjustments for TimeBlender's two reverse pitch shifters and two delay lines.

Length

Each reverse pitch shifter continuously samples small segments of audio, and plays the sampled segments in reverse. The length and the playback pitch of these segments can be varied.

The Length parameters control the length of the audio segments that are sampled for reverse shifting. The playback length of the segments will vary with the pitch shift amount, and is equal to the length setting for a pitch shift amount of 0 cents. Positive shift amounts will reduce the playback length and negative shift amounts will increase the playback length, much like a keyboard sampler's effect when transposing samples. Large values will create the most convincing revserse playback effect. Very small values of Length can be used to create some interesting modulation effects.

Shift

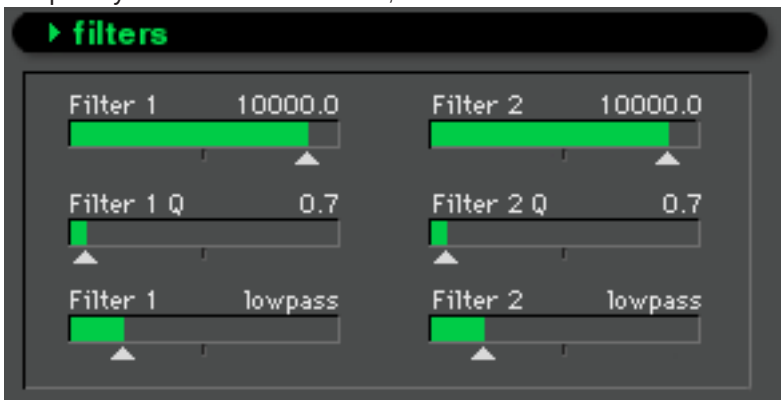
The Shift parameters are used to adjust the playback pitch of the reverse pitch shifters' sampled audio segments. The adjustment is in cents, where 1200 cents is equal to an octave of pitch shift, and 100 cents equals one semtone of pitch shift.

Delay 1 and 2

TimeBlender contains two digital delays are typically used to create recirculating delay effects. The Delay values can be adjusted in either milliseconds or in beats, depending on the Delay Mode setting on the Expert page. For settings in milliseconds, 1000 milliseconds equals 1 second. For settings in beats, the time value of 1 beat will depend on the setting of the BPM control on the main panel. Please see page 16 for an detailed description of tempo-based parameter adjustments.

Filter Parameters

PitchBlender contains two filters, which are used to alter the frequency content of the audio, much like the filter section on an



analog synthesizer. The filters can be programmed to reduce high or low frequencies, eliminate or enhance a band of frequencies, or to create highly resonant effects. By modulating the filter frequencies, many interesting synth-like effects can be created.

Frequency

The frequency control is used to adjust the frequency range that is modified by the filters. The exact type of modification will depend on the setting of the Filter Type parameter. For example,

if the filter type is ‘lowpass’, audio frequencies below the frequency setting will be unchanged, while frequencies above will be reduced in level. This is often referred to as the cutoff frequency or center frequency for the filter.

The function of the filters will also be affected by the Filter Config control in the Signal Flow page.

The filter frequencies can be modulated to create dramatic swept filter effects. The modulation adjustment is in octaves, and is always relative to the current frequency setting. For example, if Frequency 1 is set to 100 Hertz, and the modulation is set for 1 octave, the frequency will be modulated from 100 Hertz to 200 Hertz. If Frequency 1 is changed to 500 Hertz, the frequency will be modulated from 500 Hertz to 1000 Hertz.

Filter Q

The Filter Q parameter is used to control the shape of the filter, and its exact function depends on the Filter Type setting. In general, higher settings of Q will produce, narrower, steeper, and more resonant filters. Lower settings of Q will produce gentler filters.

Filter Type

The filters can perform different functions, depending on the setting of the Filter Type parameter:

None . . . . . Bypass.

Lowpass . . . . . Reduce high frequencies.

Peak BPF . . . . . Emphasize a band of frequencies.

Norm BPF . . . . . Pass a band of frequencies.

Highpass . . . . . Reduce low frequencies.

Notch . . . . . Reduce a band of frequencies.

Lowpass

The Lowpass Filter Type is used to reduce the amount of high frequencies in a signal. Frequencies above the Frequency setting will be reduced, and frequencies below will be unchanged. Small settings of the Q parameter will produce a gentle lowpass filter. Settings of Q above 0.7 will produce a resonance, boosting the level near the cutoff frequency.

Peak BPF

The Peak BPF Filter Type is used to emphasize specific range of frequencies in a signal. The band will be centered around the Frequency setting, or center frequency. Small settings of the Q parameter will produce a gentle bandpass filter. Settings of Q above 0.7 will boost the audio level at the center frequency, and can produce very high gain.

Norm BPF

The Norm BPF Filter Type is also a bandpass filter. It is different from the Peak BPF in that the audio at the center frequency will always be at unity gain. High settings of Q for this filter type wil produce a narrower bandpass, but will not produce any gain. This is especially useful when used in combination with the Loop Filter

Configuration, and will prevent unstable feedback from being produced.

HighPass

The Highpass Filter Type is used to reduce the amount of low frequencies in a signal. Frequencies below the Frequency setting will be reduced, and frequencies above will be unchanged. Small settings of the Q parameter will produce a gentle highpass filter. Settings of Q above 0.7 will produce a resonance, boosting the level near the cutoff frequency.

Notch

The notch filter removes a range of frequencies around the center frequency. Higher Q settings for this filter type will produce narrower notches. Lower Q settings will produce wider notches.

Level Parameters

The Levels page is used to control the individual input levels to the TimeBlender processor, and the individual output levels of the delay lines and the reverse pitch shifters.

Input Level

The Input Level parameters are used to independantly control the input level for the two channels of TimeBlender. This can be used for fine-tuning of levels, or can be used to turn off the input to one or both of the processing channels.

Shift Level

Use the Shift Level parameters to adjust the output levels of the two reverse shifters. The adjustment is in decibels, where 0 dB represents unity gain through the delay lines.

Delay Level

Use the Delay Level parameters to adjust the output levels of the two delay lines. The adjustment is in decibels, where 0 dB represents unity gain through the delay lines.

Panning Parameters

The Panning page contains parameters for panning the delay and reverse pitch shifter outputs, and parameters for controlling the feedback balance and mixing.

Shift Pan, Delay Pan

The outputs of the two delay lines and the two reverse shifters may be panned anywhere within the stereo field by using the Pan controls. A Pan setting of -100 will pan an output to the left, a setting of 0 will pan center, and a setting of +100 will pan to the right. A constant power panning algorithm is used to maintain a constant loudness across the stereo field.

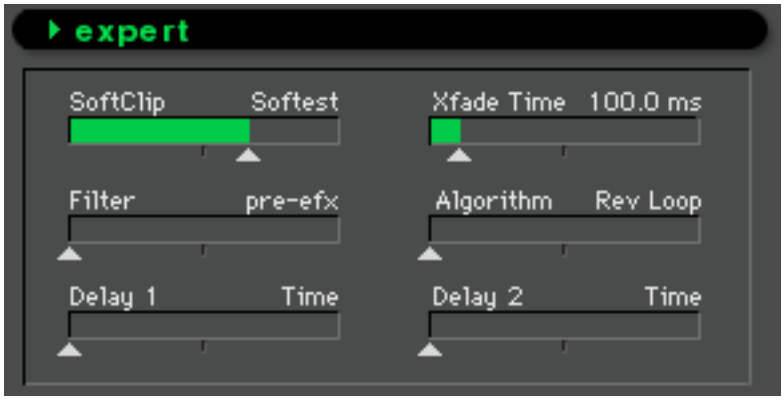
FB Balance

The Feedback Balance parameter is used to control the relative amount of feedback for the two channels, and works together with the Feedback control on the Main page. A setting of 0 will produce equal amounts of feedback for both channels. A setting of -100 will produce feedback only for channel 1. A setting of 100 will produce feedback only for channel 2. Other settings will smoothly crossfade between these extremes.

FB Mix

The Feedback Mix parameter is used to allow varying degrees of feedback to be mixed between the left and right channels. A setting of 0 will not mix the feedback paths, i.e. the channel 1 output will feed back to the channel 1 input and the same for channel 2. The default setting of 50 percent will produce an equal mix of channels 1 and 2 which is fed back to both channels equally. This can be used to create extremely dense chorus effects, and even reverb-like sound when used together with pitch-shifting and delay modulation. A setting of 100 percent will produce a perfectly crossed feedback path, with the output of channel 1 feeding the input of channel 2 and vice-verse.

Expert Parameters



The expert page contains parameters for re-configuring the signal flow and optimizing the audio processing within the TimeBlender plug-in.

SoftClip Mode

When SoftClip Mode is turned to off, soft-clipping will be disabled, and any audio that is boosted beyond the maximum digital level will be clipped in the typically harsh digital way. When

soft-clipping is turned on, the audio will be much more gradually clipped, producing more pleasant harmonics, and a more analog sound.

Three soft-clipping modes are available – *soft*, *softest*, and *asymmetrical.* The *soft* mode will preserve the maximum linear range while still clipping softly. The *softest* mode will produce the softest clipping possible, but will also introduce very small amounts of distortion over much of the audio range. The *asymmetrical* mode will clip one side of the audio waveform first, producing a higher proportion of even harmonics.

Xfade Time

The Xfade Time control is used to adjust the speed of the cross-fade used by reverse shifters. Because the reverse shifters time-reverse short segments of audio, it is necessary to perform a cross-fade to prevent clicks at the segment boundaries. Longer values of cross-fade time will produce smoother transitions, and shorter times will produce more abrupt transitions.

Filter Config

The Filter Config parameter controls at what point in the signal chain the filters have their effect.

pre-efx . . . . . Filter after input.

pre-efx+. . . . . Filter after input, summed.

post-pitch . . . . . Filter pitch-shift output.

post-delay . . . . . Filter delay output.

post-efx . . . . . Filter final output.

loop . . . . . Filters in feedback loop.

pre-efx

The pre-efx filter configuration places the filters at the very input of the TimeBlender algorithm, before the pitch shifters and delay lines. Filter 1 operates on the left input channel, and filter 2 operates on the right input channel.

pre-efx+

The pre-efx+ configuration is similar to the above, except that the output of the two filters is mixed before being sent on for further processing. This can be used to create a formant filter effect with the two filters.

post-pitch

The post-pitch setting places the filters at the output of the revserse pitch shifters, before panning and mixing. Filter 1 operates on the output of pitch shifter 1 and filter 2 operates on pitch shifter 2.

post-delay

The post-delay setting is similar to above, placing the filters at the delay line outputs.

post-fx

The post-fx setting places the filters at the very end of the signal chain, after the revserse pitch shifters, delays, and mixing.

loop

The loop configuration places the filters in the feedback path. This configuration is useful to attenuate high frequencies when using feedback with delay lines or revserse pitch shifters. Be careful when using high Q values in this configuration, as this can easily produce unstable feedback, and potentially very loud sounds.

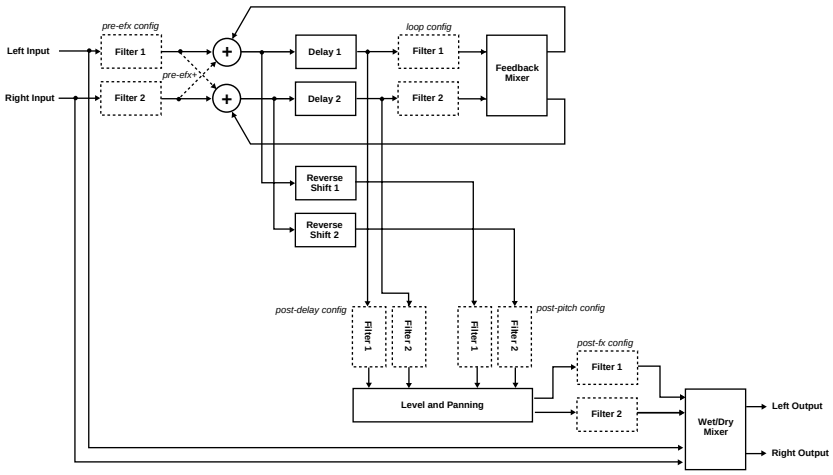
Delay Mode

The Delay Mode parameter is used to control how the delay time for the two delay lines is adjusted. When this parameter is set to *Time,* the delay time is adjusted in milliseconds. When this parameter is set to *BPM*, the delay time is adjusted in beats, and is affected by the BPM control on the SoundBlender panel.

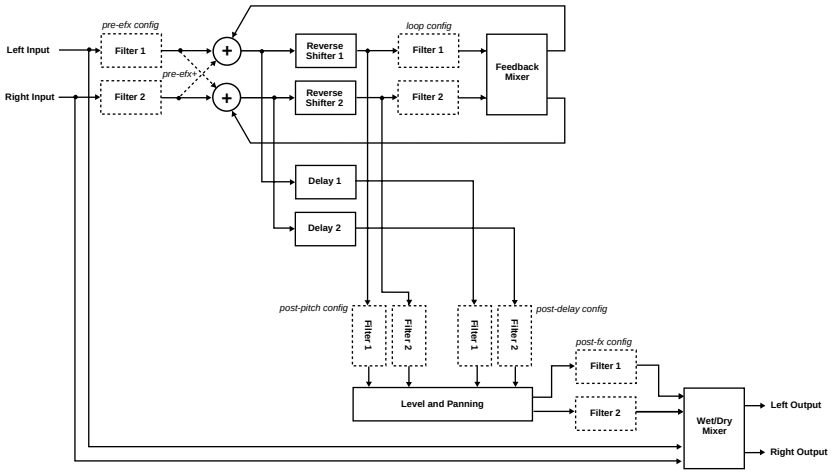
Algorithm

The algorithm setting is used to alter TimeBlender's feedback path. The Rev Loop setting patches the outputs of the reverse shifters into the feedback path, causing the time-reversing effect to be recycled. This configuration is used in the Crystallizer presets. The Dly Loop algorithm patches the delay outputs into the feedback path, and is used to create delay loop effects, and reverse reverb effects. The two configurations are shown below:

Delay Loop



Reverse Loop



Because there are so many presets for SoundBlender and there will be many more to come, the presets are described in the file SoundBlender Presets.pdf located in the Wave Mechanics folder created when you installed SoundBlender and also on the Wave Mechanics Plug-ins CD-ROM. You will need Adobe Acrobat Reader (which is free and available on the web at www.adobe.com) to read the file.

Newer presets can be obtained, as they become available, on-line from the Wave Mechanics web site www.wavemechanics.com and the SoundBlender Presets.pdf file on that site will be updated to contain descriptions of any newer presets.

How do I get finer resolution on parameter adjustments?

Mac- Hold down the ⌘ key while dragging the parameter slider.

Win- Hold down the Ctrl key while dragging the parameter slider.

How do I return to the default value for a parameter control?

Mac- Hold down the option key and click on the parameter control.

Win- Hold down the Alt key and click on the parameter control.

The pitch shifted output sounds chopped up or 'glitchy'.

Make sure that you are only trying to process a monophonic, or single instrument sound. Also, make sure that the track is as dry as possible, which means making sure that any reverb or other effects are inserted after SoundBlender. Adjust the Min Frequency and Max Frequency parameters.

Pro Tools says that the DSPs are 'maxed out' when trying to insert PitchBlender or TimeBlender.

You've run out of DSPs on your DSP farm card. You must either remove some other DSP plug-ins from your current session, or add more DSP power by purchasing another DSP farm card from Digidesign.

Wave Mechanics offers free technical support for all registered users. We love to hear from our users, but if you are having problems, first try to look in the manual for an answer. Also, check our web site for technical notes and product updates. If you are still stumped, please e-mail us with the following info (keep in mind that we won't be able to help you with questions about Pro Tools, Apple, or Windows hardware or software, or any other non-Wave Mechanics stuff):

- The product version and serial number.
- The version number of your Pro Tools system, and type of hardware (e.g. HD, Mix, or 'classic' PCI.)
- Your computer type and operating system version number (e.g. system 9.1, etc.)
- A detailed description of the problem.

The e-mail address for support is:

support @ wavemechanics.com

If you don't have access to e-mail, please fax the same info to:

fax: 802-951-9799

If you don't have e-mail or a fax, you can call us at:

phone: 802-951-9700

Finally, if you are completely un-wired, you can write to us at:

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