



Waverazor Manual v2.5.2

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Foreword

Thank you so much for adding Waverazor to your music and sound design tool box. We are deeply honored by your choice to take this journey with us, exploring new sonic territory, and making new discoveries. The act of slicing up waveforms and recombining them has never been done in synthesis before, and we're excited that it has already given rise to some interesting and unique new sounds.

The Waverazor oscillator made us re-evaluate what we knew about sound waves. Sine, Saw, Square, Tri... What are those shapes, really? And what do their features actually mean to the harmonic content of the sound? What kind of harmonics does the act of wave slicing generate? And realizing that repetition determines frequency, what waves will you choose to put in each slice? It's a grand experiment.

This sort of questioning and exploration led us to discoveries of new synthesis techniques like Multi Sync, which is the application of hard sync to multiple points within a single cycle or out beyond a single cycle, and Mutant AM, which is a complex matrix of amplitude modulation based on the harmonic content of each wave slice. We are excited to see where your explorations take you!

- Taiho Yamada

When to Use the Quick Start Guide

This Waverazor manual is primarily about the sound editor pages and parameters. If you're looking for installation and setup instructions, or more about the Main Performance Page controls, please reference the Quick Start Guide. You can find the QSG under the "?" button in Waverazor itself, or in the manuals section of the Tracktion website:

www.tracktion.com/training/manuals#waverazor

Editor Patch Select and Global Section



Patch Browser

Clicking on the triangles to the left and right of the patch name will decrement or increment the patch. Or you can click on the patch name to open the browser and select a new patch. Clicking once on a patch will load it, but will also leave the browser open so that further selections can be made. Double clicking will select a new patch and close the browser.

There are three banks of patches that can be selected, giving some options for

organizing the types of patches. Factory patches are singular individualized constructions. Templates are more generic patches that can be used as starting points for sonic exploration. And User patches are any patches that you have saved yourself.

Floppy Icon

Provides a menu for file management functions:

Save - Saves the patch with the current name into the user bank.

Save As - Allows you to name the patch before saving into the user bank.

Rename - Allows you to rename the patch.

Delete - Deletes the current patch.

Import - Allows you to import .wraz patch files to the user bank.

Export - Allows you to export .wraz patch files.

Gear Icon

Provides access to system settings.

Registration - Shows registration status.

Language - Allows you to select from available languages.

Theme - Changes the color scheme for the plug-in.

Number of Voices - Sets a voice limit for the plug-in from 1 to 64 voices.

Voice Buffer - Sets the sample buffer size between 16 and 256 samples.

Question Mark Icon

Provides a menu for opening the Quick Start Guide and Manual documentation, or for navigating to the MOK forum for the latest answers.

Exclamation Point Icon

This is a panic button for turning all notes off if they get stuck.

Undo, History and Redo

Undo - Allows you to undo the last editing action.

History - Allows you to go back through your history of edits and select a point to which you'd like to return.

Redo - Allows you to Redo any editing actions that you've undone.

Meters



CPU Meter

Shows the percentage of processor power used by the plug-in.

Voice Count Meter

Displays the current number of voices playing.

Oscilloscope

Shows the waveform output from the plug-in in real time. The oscilloscope has an associated drop down menu to allow you to choose which Channel you wish to view, Channel 1 (Left) or Channel 2 (Right).

Navigating the Editor



Waverazor uses a groundbreaking contextual editing system that allows you to focus on a specific area of the synth and to see all of the things that are connected to it.

In the top central area, you'll see a voice block diagram, which allows you to jump directly to any module of the synth, such as an oscillator, just by clicking on it.

That brings the module into focus in the lower central area, where all the parameters for that module are available for editing.



Modules in the left column are those that feed into the central focus module, while modules in

the right column are where the outputs of the focus module are going. And these modules change in context depending on which module is in focus.

Modules in the side columns can be brought into central focus by pressing the focus button on the module.



Editing Parameter Values

Knobs and Sliders

Click and drag vertically on a knob to change its value. Go up for higher values and down for lower values. There are also special key modifiers that change the editing behavior.

Shift - Reverses the response so that going up generates lower values and going down generates higher values.

Control - Changes mouse behavior so that slow movements produce fine resolution edits, while fast movements still make large changes in value.

Option (ALT on PC) - Changes mouse behavior so that slow movements produce fine resolution edits, while fast movements still make large changes in value. In addition, holding Option when clicking on a control will reset the parameter value to its default.

Command - Holding Command when clicking on a control will reset the parameter value to its default.

Buttons

Simply click on a button to change its state. Some buttons have more than two states and in those cases, subsequent clicks continue to cycle through parameter values. Key modifiers do not work with buttons.

Menus

Click within a menu field to choose a new setting from a list. Some menu values have sub-menus, indicated by an arrow next to the setting. In that case, when you mouse over your desired setting, a sub-menu will open up with more choices. Also note that some menus may have more items than can be displayed in one window, so automatic scroll arrows are provided at the top and bottom of the window, depending on which direction requires scrolling. You can also scroll with a scroll wheel or standard scrolling gestures. Click on your final selection to change the parameter value and close the menu, or click outside the menu to abort the change.

Editor Column Headers

While the editor is divided into three columns, it is also divided between its top navigation quarter and the lower work area. Three column headers separate the top portion from the bottom and provide additional tabbed viewing options in their respective sections.



The middle column header shows the name of the module that is in focus, but may also display an option with related functionality. For example, you'll be able to select the voice mode module alongside the oscillators, and the hardwired amplitude envelopes alongside the amp module. The header provides additional viewing options whenever it makes sense to do so.

The same is true for the left column where you'll likely see tabs for Mods, Ctls, Inpt and Mtrx

among other options. The left column is contextual so what you see will depend on how the patch was programmed and what you end up focusing on in the center section, but the left header will help sort through the different types of modules feeding the focus module.

The column header on the right side behaves the same way in that it sorts what you are seeing contextually and via tabs, but of course the right column is showing the modules that are destinations for the focus module's output.

Within the middle column header, a drop down menu button appears on the right side when certain voice blocks are selected:

Mods - For voice and global level mods, this menu allows you to add an LFO, Slope Generator, ASR Envelope, ADSR Envelope, or Portamento Generator. The limit is 64 mods in the Voice Mods and 64 in the Global Mods. When the Amp Envs tab is selected, the menu allows you to add an ASR Envelope or ADSR Envelope, for up to 8 additional amp mod sources.

Matrix - For voice and global level matrix routes, this menu allows you to add 1x1 and 1xN mod routes. For the Voice Matrix and Global Matrix, you can have up to 64 routes each, with up to 24 of those 64 being 1xN multiple destination routes (N being a number from 1 to 16).

Amp - When the Amp block is selected and the Amp Envs tab is active, the menu allows you to add an ASR Envelope or ADSR Envelope, for up to 8 additional amp mod sources.

Module Controls

From left to right across the top:



Expand Button

When modules are not in focus, they may only show the most important parameters of the module in a compact view. The Expand Button expands the module to show more parameters if they are available for the module.

Focus Button

When pressed, this button brings the module into focus in the center section.

Mute Button

This button mutes the output of the module, or bypasses it in the case of filters and effects.

Menu

When available, this button opens a drop down menu with options specific to the module in focus.

CTLS Block NAV

This brings the parameters for the macro controls into focus, allowing you to set up custom knobs, buttons and vector XY controls for your patch on the main GUI page.

Focus Column

Control Overview tab - Shows an overview of all the performance page controls and their current values.

Controls tab - Allows you to focus on individual control modules.

Input Column

CTRLS tab - This is a contextual display dependent on what is currently shown in focus. If the control overview is in focus, the left column shows the collapsed versions of the individual control modules. If the individual control modules are in focus, the left column shows a control overview.

Output Column

DESTS tab - When the DESTS tab is selected, the right column shows all the destination modules affected by the current control set via the modulation matrix.

MATRIX tab - When the Matrix tab is selected, the right column shows all the modulation routes with sources originating from the current control set.

Control Module

Menu

Rename - Allows you to name the macro control. The control names will appear on the main performance page for all macro controls except for the vector XY controllers. All control names also appear as source names in the modulation matrix for routing.

Parameters

ctl value - This is a duplicate of the control that appears on the main performance page and in the editor's control overview. It allows you to test the control's behavior while editing.

min val - Sets the Minimum Value range for the Ctl Value parameter. The Min range takes into account the range and range pol settings.

max val - Sets the Maximum Value range for the Ctl Value parameter. The Max range takes into account the Range and Range Pol settings.

range - Sets the general range of the control in powers of 10 (e.g. 1, 10, 100, 1000, etc.). This allows for finer tuning in relation to other parameters if you only set this to the greatest range you will need.

ctl type - Changes the type of control from continuous knobs to toggle or momentary buttons.

zero detent - Sets a percentage of the Ctl Value near zero that behaves as a zero detent. This works in both unipolar and bipolar modes.

step - Defines the number of distinct steps the Target output will have based on the Ctl Value input. Settings:

Float - The target output is a continuous smooth value.

int floor - The Ctl Value is rounded to the NEAREST LOWER integer (e.g. 1.7 rounds down to 1).

int round - The Ctl Value is rounded to the NEAREST integer (e.g. 1.7 rounds to 2).

int ceiling - The Ctl Value is rounded to the NEAREST HIGHER integer (e.g. 1.1 rounds up to 2).

128 ~ 2 - Sets an explicit number of steps for Target output value.

range pol - Sets whether your range should be unipolar (0 to +range) or bipolar (-range to +range).

curve - Sets a curve relationship between the Ctl Value and its remapped modulation output value. It essentially applies a curvature to the control signal that transforms a linear response into a non-linear output response according to the shapes available in the menu.

curve pol - Sets the behavior of the curve if the Ctl Value range is Bipolar. It determines whether the mapping curve is continuous from Min to Max, or a curve mapped from 0 to Max and an inverted curve from Min to 0. If the range polarity is Unipolar, this parameter has no effect.

ctl fill type - This defines how you want a control (knob or button) to have its value graphic displayed when it is being adjusted. There are a bunch of values here, and it's probably best to just experiment with them to see the results.

def val - Sets the default value of the Control. When you double click on a Control in the performance page, the Control will move to its default value. Alternatively, you can return to the default value by using Alt+Click on PC and Linux and Cmd+Click in Mac OS X.

valstr src - Selects where the control text string gets its original value for processing the text.

ctl val - Uses the incoming ctl value as reference value for the control and text.

target val - Uses the resulting target value as reference value for the control and text.

range - Creates a 0.0 to 1.0 range between the min val and max val of the Ctl Value regardless of whether the Ctl Value is bipolar or unipolar.

range bip - Creates a -1.0~0.0 range and 0.0~1.0 range for a bipolar Ctl Value.

If Ctl Value is unipolar, this behaves the same as the Range setting.

range split - Creates a -1.0~+1.0 range for the entire range of Ctl Value, regardless of whether Ctl Value is bipolar or not, or min is at zero, this will put a middle point of 0.0 control text at the current mid-point of the Ctl Value.

valstr type - Formats the type of number used in the control TEXT value.

float.5 ~ float.1 - Displays a floating point number with 1 ~ 5 decimal places (e.g. float.3 will display 1.537, float.2 will display 1.54).

integer - Truncates the value to integer values (e.g. 1.537 will display 1).

hex - Displays the value in integer hex (e.g. the value 5230 will display in hex as 146E).

hex2,4,6,8 - displays the value in integer hex with a truncated number of characters (e.g. the value 5230 will display in hex.8 as 0000146E, and a setting of hex.2 will display the value as 6E).

valstr format - Selects how the metric tags are added to the control text (examples below have metrics on, with valstr tag of 'Hz' and a current value of 1530.0)

vmt - "ValueMetricTag" e.g. 1.53kHz

v(x)mt - "Value MetricTag" e.g. 1.53 kHz

vm(x)t - "ValueMetric Tag" e.g. 1.53k Hz

tvm - "TagValueMetric" e.g. Hz1.53k

t(x)vm - "Tag ValueMetric" e.g. Hz 1.53k

mtv - "MetricTagValue" e.g. kHz1.53

mt(x)v - "MetricTag Value" e.g. kHz 1.53

tmv - "TagMetricValue" e.g. Hzk1.53

tm(x)v - "TagMetric Value" e.g. Hzk 1.53

valstr mult - Multiplier/Divider for the control text value. 1:1 is unity (no change), 2:1~100:1 creates a multiplier from 2x to 100x, and 1:2~1:100 creates a divisor from divide by 2 to divide by 100.

valstr multx - Provides x10 multiples for a quick larger scale Multiplier of the control text value. You can use this in combination with the valstr mult to create larger multiply amounts. For example, if you want to multiply the control text value by 200, you can set multx to 100 and mult to 2:1.

valstr offs - Sets a display value offset amount for the control text value, as a floating point value from -100.0 to +100.0.

base metric - Sets the base metric of the control value. This works in conjunction with macro metric and micro metric to determine how metrics are added to the control value text.

macro metric - Sets the point at which larger (greater than 0) metric labels begin to adjust the control text value, in range from deca[da] to exa[E]. For example, if you set this to [k] kilo, values above 1000.0 will be divided by 1000 and a k will be appended, depending on the valstr format setting (e.g. 1520, will become 1.52k). Set this to the lowest metric value where you want metric conversion to begin affecting the control value text.

micro metric - Sets the point at which smaller (less than 0) metric labels begin to adjust the control text value, in range from deci[d] to atto[a]. For example, if you set this to [m] milli, values below 0.01 will be multiplied by 1000 and an m will be appended, depending on the valstr format setting (e.g. 0.00198, will become 1.98m). Set this to the highest fractional metric value where you want metric conversion to begin affecting the control value text.

MODS Block NAV

This brings all currently available modulation modules into focus, and allows you to add, delete and edit Envelopes, LFOs and other modulation sources.

Focus Column

Voice Mods tab - Shows all the mods at the voice level. Voice level mods are applied

per voice and support polyphonic modulation behavior. They cannot be applied to Global level destinations such as the bus effects.

Global Mods tab - Shows all the mods at the Global level. Global level mods are applied per patch and only support monophonic modulation behavior (i.e. all voices are modulated in the same way). Global mods are also useful for modulating non-polyphonic Global destinations such as the bus effects.

Amp Envs tab - Shows the available amplitude envelopes. You can have up to 8 additional envelope sources hardwired to voice level amplitude destinations.

Menu - Provides drop down menu selections related to Voice Mods, Global Mods, and Amp Envs in context. For Voice and Global, the selections are as follows:

- Add LFO
- Add Slope Generator
- Add Envelope ASR
- Add Envelope ADSR
- Add Portamento Generator

For the Amp Envs tab, the selections are as follows:

- Add Envelope ASR
- Add Envelope ADSR

You can have up to 64 mod sources at the Voice level, 64 mod sources at the Global level, and 8 mod sources at the Amp Envs level. Note that Global modulation sources can be routed to Voice level polyphonic destinations, however Voice level modulation sources cannot be routed to Global destinations such as Effects Send levels.

Input Column

This is a contextual, tabbed display dependent on what is currently shown in focus (Center).

VOICE MODS focus tab selected:

MODS tab - When the Mods tab is selected, the Input column shows the collapsed versions of any controls or modulation sources routed to the Voice focus modules.

MATRIX tab - When the Matrix tab is selected, the Input column shows the collapsed versions of any modulation routes affecting the Voice focus modules.

GLOBAL MODS focus tab selected:

MODS tab - When the Mods tab is selected, the Input column shows the collapsed versions of any controls or modulation sources routed to the Global focus modules.

MATRIX tab - When the Matrix tab is selected, the Input column shows the collapsed versions of any modulation routes affecting the Global focus modules.

AMP ENVS focus tab selected:

OMIX tab - This is a contextual tab displayed only when the AMP ENVS tab is selected for focus. If the OMIX tab is selected, the Input column shows a collapsed version of the Oscillator Mixer, which includes filter routing that feeds the Amp section.

MODS tab - When the Mods tab is selected, the Input column shows the collapsed versions of any controls or modulation sources routed to the Amplitude Envelope modules.

MATRIX tab - When the Matrix tab is selected, the Input column shows the collapsed versions of any modulation routes affecting the Amplitude Envelope modules.

Output Column

This is a contextual, tabbed display dependent on what is currently shown in focus (Center).

VOICE MODS focus tab selected:

DESTS tab - When the DESTS tab is selected, the Output column shows all the destination modules currently affected through the modulation matrix by the Voice modulation sources in focus.

MATRIX tab - When the Matrix tab is selected, the right column shows all the modulation routes stemming from the Voice modulation sources in focus.

GLOBAL MODS focus tab selected:

DESTS tab - When the DESTS tab is selected, the Output column shows all the destination modules currently affected through the modulation matrix by the Global modulation sources in focus.

MATRIX tab - When the Matrix tab is selected, the right column shows all the modulation routes stemming from the Global modulation sources in focus.

AMP ENVS focus tab selected:

VAMP tab - When the VAMP tab is selected, the Output column shows the Voice Amp Mixer controlling the main output levels, as well as the Envelope Mixers that set the oscillator and filter input levels for each Amplitude Envelope.

DESTS tab - When the DESTS tab is selected, the Output column shows all the destination modules currently affected through the modulation matrix by the Amplitude Envelope modulation sources in focus. Note that the Amplitude Envelopes are hardwired to Amplitude, so there is no destination displayed for this.

MATRIX tab - When the Matrix tab is selected, the Output column shows all the modulation routes with sources originating from the Amplitude Envelopes in focus.

Modulation Modules

The modulation modules generate control data that can be routed to modulate nearly every parameter in the synth. These modules each have adjustable parameters as well as additional input destination parameters and output sources that are available only when viewed in the modulation matrix.

Each module has a header that displays the name of the modulator, a Focus button that brings the module into focus in the center column, a Mute button that stops the modulation output, and a drop down Menu that allows you to rename or delete the module.

LFO

The Low Frequency Oscillator primarily provides a cyclical modulation that continues over time.

Parameters

shape - All oscillator wave shapes are available in the LFO. For a full list of waves, see appendix A at the end of this manual.

level - This is a bipolar Level, from -100% to +100%. The negative values simply invert the signal polarity and allow for a smooth thru-zero transition from normal to inverted signal.

speed(f) - Adjusts the frequency of the LFO cycle linearly in Hertz.

range mult - Multiplies the speed parameter between x1 and x100.

sync div - When Clock is set to System, this parameter sets the clock's Note Division in relation to the synchronization reference. This parameter has no effect when Clock is set to Self.

clock - Sets the clock to Self or System (host clock synchronization).

trig mode - Governs how the LFO responds to notes and triggering. Mode settings:

Reset - The LFO will begin at its start phase each time it is triggered (either by Note On, Trigger or Gate input).

Free - The LFO is constantly running and never re-triggered.

Legato - The LFO will reset to its start position only when the first note (or Trigger or Gate input) arrives. If subsequent notes are played legato (overlapping), then the LFO does not reset. If all previous notes and gates have ended and a new note (or Trigger or Gate input) is received, the LFO will reset to its starting phase position.

Random - The LFO has a new starting phase position with every new note (or Trigger or Gate input).

st.phase - Sets the start point of the LFO in degrees from 0 to 360.

mode - Determines the output polarity mode of the LFO. By default, the LFO is bipolar, generating a +1.0 to -1.0 signal (representing positive 100% to negative 100% scale).

Mode settings:

Bipolar: This is the default, with both positive and negative going values. For example, a sine wave will begin at 0, rise to 1, then fall through 0 to -1, and finally return to its beginning at 0.

Unipolar: This will 'normalize' the signal into a positive (0 through 1) value range. For example, a sine wave in Unipolar mode will begin at 0.5, rise to 1, then fall to 0.5, then to 0, and return to its beginning at 0.5.

Rectified: This simply forces a bipolar signal to always be positive. Any negative values will be turned into positive values, creating a positive mirror image of the signal that would otherwise have been negative. For example, the bipolar sine wave shape after being rectified will begin at 0, rise to 1, then fall to 0, then rise to 1 again, then fall to its beginning point of 0.

polarity - Determines a positive or negative polarity of the signal.

duty cycle - Adjusts the duty cycle of the signal. This is the same duty cycle operation as found in the oscillators.

quantizer - Reduces the bit resolution of the LFO wave shape from 0 to 100%. At 0% the wave is normal and unquantized. As you turn the quantizer up, the wave will have fewer bits of resolution. At 100%, the wave shape will become quantized to 2 or 3 bits

of resolution, depending on the wave shape. This amount is tailored to each shape so that the increase in quantization will sound natural, as opposed to normal bit reduction which doesn't sound natural with all wave shapes.

one-shot - When one-shot is active, the LFO will play only one cycle and then stop whenever it is triggered.

freeze - The freeze button is a manual control for the Freeze Gate feature, which causes the Modulator to freeze at its current value. The LFO can also be frozen when it receives a High level Freeze Gate input from an external modulator.

reset - This is a manual button for the Reset Trigger modulation destination that resets the LFO to its start. It can be used to manually reset the LFO phase to its starting position, or to retrigger an LFO in one-shot mode.

LFO Additional Modulation Outputs:

S&H output - The LFO has a built-in Sample & Hold function and this is the output level. Use the S&H trig in to trigger the next sample. (This trigger input is found in the LFO's modulation matrix inputs list.)

gate output - Sends an alternating High and Low output signal with a 50% duty cycle at the frequency of the LFO.

LFO Modulation Destinations Only:

reset trig - Modulation input will reset the LFO to its start point (according to the start phase parameter). Like all Reset Triggers, the input control value must pass below the gate threshold and back above the threshold in order for a new trigger to be actuated. For more on Triggers and Gates, please see Appendix C at the end of this manual.

freeze gate - As long as the Freeze Gate is held above its threshold, the Modulator will freeze at its current value.

speed(s) - Adjusts the Speed in percentage (Scalar). This modulation input is the easiest to use if you want to hear a linear modulation of the speed from your source. This input has no effect if the LFO is synchronized to the System Clock.

sp.adj(f) - Adjusts the Speed in frequency (logarithmic). This modulation input reacts to modulations with the same feel as the Speed(f) knob in the LFO module GUI. This input

has no effect if the LFO is synchronized to the System Clock.

S&H trig in - The LFO has a Sample & Hold on its output. Use the S&H trig in mod destination to trigger the next sample of the Sample & Hold. It is triggered each time the input level goes from low to high.

Slope Generator

The Slope Generator is a simple type of envelope that generates a single stage.

Parameters

slope time - The time it takes for the slope generator to go from its beginning level to its ending level.

slope fine - Fine tune adjustment for the Slope Time parameter.

slope shape - Applies a curvature to the slope.

beg level - Defines the starting level for the slope generator.

end level - Defines the ending level for the slope generator.

trig mode - Governs how the Slope Generator responds to notes and triggering. Mode settings:

Reset - The Slope Generator will start at its beginning level each time it is triggered (either by Note On, Trigger or Gate input).

Legato - The Slope Generator will start at its beginning level only when the first note (or Trigger or Gate input) arrives. If subsequent notes are played legato (overlapping), then the Slope Generator does not reset and continues to its end level as long as a note is held. If all previous notes and gates have ended and a new note (or Trigger or Gate input) is received, the Slope Generator will reset to its beginning level.

key track - Scales the Slope Time according to the note played on the keyboard. With positive key track values, the higher notes have shorter times, and with negative key

track values, higher notes have longer times.

vel crv beg - Provides a velocity curve response for the Vel Scl Beg parameter.

vel scl beg - Allows you to scale the slope's beginning level with note velocity. Positive values give higher output with higher velocity, while negative values cause the output to start at the highest level and modulate lower with higher velocities.

vel crv end - Provides a velocity curve response for the Vel Scl End parameter.

vel scl end - Allows you to scale the slope's ending level with note velocity. Positive values give higher output with higher velocity, while negative values cause the output to start at the highest level and modulate lower with higher velocities.

Slope Generator Additional Modulation Outputs:

S&H output - The Slope Generator has a built-in Sample & Hold function and this is the output level. Use the S&H trig in to trigger the next sample. (This trigger input is found in the Slope Generator's modulation matrix inputs list.)

Gate Out Beg - Will set the Gate level High when the slope starts.

Gate Out End - Will set the Gate level High when the slope ends.

Slope Generator Additional Modulation Destinations:

reset trig - Modulation input will reset the Slope Generator to its beginning point. Like all Reset Triggers, the input control value must pass below the gate threshold and back above the threshold in order for a new trigger to be actuated. For more on Triggers and Gates, please see Appendix C at the end of this manual.

freeze gate - As long as the Freeze Gate is held above its threshold, the Slope Generator will freeze at its current value.

S&H trig in - The Slope Generator has a Sample & Hold on its output. Use the S&H trig in mod destination to trigger the next sample of the Sample & Hold. It is triggered each time the input level goes from low to high.

Envelope ASR

The ASR Envelope is a simple Attack, Sustain, and Release stage envelope, capable of Bipolar operation.

Parameters

trig mode - Governs how the ASR Envelope responds to notes and triggering. Mode settings:

Reset - The ASR Envelope will start at its beginning level each time it is triggered (either by Note On, Trigger or Gate input).

Legato - The ASR Envelope will start at its beginning level only when the first note (or Trigger or Gate input) arrives. If subsequent notes are played legato (overlapping), then the ASR Envelope does not reset and continues through its stages accordingly as long as a note is held. If all previous notes and gates have ended the envelope will release to zero, and when a new note (or Trigger or Gate input) is received, the ASR Envelope will then reset to its beginning.

atk time - The time it takes for the envelope Attack stage to go from its beginning level to the Sustain level.

atk fine - Fine tune adjustment for the Attack Time parameter.

rel time - The time it takes for the envelope Release stage to go from the Sustain level to zero, once the note is no longer held.

rel fine - Fine tune adjustment for the Release Time parameter.

run mode - Governs how the envelope behaves after it is launched. Mode settings:

Normal - The envelope begins its Attack Stage upon Note On. If a Note Off is received before the envelope reaches the full Attack Level, the envelope will immediately jump to the release stage from its current level.

Freerun - The envelope begins its Attack Stage upon Note On. If a Note Off is received before the envelope reaches the full Attack Level, the envelope will complete the full Attack Stage before moving on to the release stage.

key track - Scales the Envelope times according to the note played on the keyboard.

With positive key track values, higher notes have shorter times, and with negative key track values, higher notes have longer times. For reference, with a setting of 100%, the stage times will double in speed with each positive octave.

atk shape - Applies a curvature to the Attack stage.

atk lock - When engaged, the atk lock button will keep the attack time at its current value even if the Time Scale control is adjusting the overall time of the envelope. However, attack time is still available for modulation through the modulation matrix.

rel shape - Applies a curvature to the Release stage.

loop - When set to Atk+Rel, the ASR envelope loops the attack and release stages after a note is released. There must be some Amplitude Envelope release time in order to hear the effect of the ASR in loop mode.

time scale - Adjusts the Overall time of the Envelope to be longer or shorter. This parameter can be used either directly, or as a modulation destination, to increase or decrease the overall envelope time.

level scale - Increases or decreases the overall level of the Envelope stages.

sus pedal - Sets whether or not the Envelope will be held in its Sustain stage when the sustain pedal is depressed.

sus keyon - Sets whether or not the Envelope will be held in its Sustain stage by Key On (Key held down).

vel curve - Provides a velocity curve response for the Vel Scale parameter. It essentially applies a curvature to the velocity signal that transforms a linear response into a non-linear output according to the shapes available in the menu.

vel scale - Allows you to scale the overall envelope level with note velocity. Positive values give higher output with higher velocity, while negative values cause the output to start at the highest level and modulate lower with higher velocities.

ASR Envelope Additional Modulation Outputs:

S&H output - The ASR envelope has a built-in Sample & Hold function and this is the output level. Use the S&H trig in to trigger the next sample. (This trigger input is found in the ASR envelope's modulation matrix inputs list.)

latchout atk - This Gate output level is set High at the end of the Attack stage.

latchout sus - This Gate output level is set High at the end of the Sustain stage.

latchout rel - This Gate output level is set High at the end of the Release stage.

gate out atk - This Gate output level is set High for the duration of the Attack stage only.

gate out sus - This Gate output level is set High for the duration of the Sustain stage only.

gate out rel - This Gate output level is set High for the duration of the Release stage only.

ASR Envelope Additional Modulation Destinations:

reset trig - Modulation input will reset the ASR Envelope to its beginning point. Like all Reset Triggers, the input control value must pass below the gate threshold and back above the threshold in order for a new trigger to be actuated. For more on Triggers and Gates, please see Appendix C at the end of this manual.

freeze gate - As long as the Freeze Gate is held above its threshold, the ASR Envelope will freeze at its current value.

start level - Defines the starting level of the envelope, at the onset of the Attack stage.

S&H trig in - The ASR envelope has a Sample & Hold on its output. Use the S&H trig in mod destination to trigger the next sample of the Sample & Hold. It is triggered each time the input level goes from low to high.

Envelope ADSR

The ADSR Envelope is a more full featured envelope, providing a Delay, Attack, Attack Hold, Decay, Sustain Decay, and Release stage, capable of Bipolar operation.

Parameters

delay time - Defines the time it takes from Note On to the start of the envelope Attack stage.

attack time - The time it takes for the envelope Attack stage to go from its beginning level to the Attack Hold stage.

atkhold time - The time it takes for the envelope Attack Hold stage to complete and then move on to the Decay stage.

decay time - The time it takes for the envelope Decay stage to go from the Attack Hold level to the Sustain Level.

rel time - The time it takes for the envelope Release stage to go from the current Sustain level to zero, once the note is no longer held.

start level - Defines the starting level of the envelope, prior to the Delay stage.

atk level - Sets the ending level of the Attack stage.

sus level - Defines the level at which the Sustain stage begins. If the Sustain Decay stage is inactive, then the envelope will hold at this level until the release stage begins.

sus decay - The Sus Decay parameter sets the amount of time the envelope will decay towards the Release Level during the Sustain portion of the envelope. To allow the Envelope to sustain without decaying, you can set this to its maximum value, which is OFF.

rel level - Sets the ending level of the Release stage.

trig mode - Governs how the ADSR Envelope responds to notes and triggering. Mode settings:

Reset - The ADSR Envelope will start at its beginning level each time it is triggered (either by Note On, Trigger or Gate input).

Legato - The ADSR Envelope will start at its beginning level only when the first note (or Trigger or Gate input) arrives. If subsequent notes are played legato (overlapping), then the ADSR Envelope does not reset and continues through its stages accordingly as long as a note is held. If all previous notes and gates have ended the envelope will proceed to its release level, and when a new note (or Trigger or Gate input) is received, the ADSR Envelope will then reset to its beginning level.

atk shape - Applies a curvature to the Attack stage.

dec shape - Applies a curvature to the Decay stage.

s.dec shape - Applies a curvature to the Sustain Decay stage.

rel shape - Applies a curvature to the Release stage.

run mode - Governs how the envelope behaves after it is launched. Mode settings:

Normal - The envelope begins its Delay Stage upon Note On. If a Note Off is received before the envelope reaches its Release state, it will immediately jump to the Release stage from its current level.

Freerun - The envelope begins its Delay Stage upon Note On. If a Note Off is received before the envelope reaches its Release state, the envelope will continue and complete all other active stages before moving on to the Release stage.

atk lock - When engaged, the atk lock button will keep the attack time at its current value even if the Time Scale control is adjusting the overall time of the envelope. However, attack time is still available for modulation through the modulation matrix.

key track - Scales the Envelope times according to the note played on the keyboard. With positive key track values, higher notes have shorter times, and with negative key track values, higher notes have longer times. For reference, with a setting of 100%, the stage times will double in speed with each positive octave.

dcy lock - When engaged, the dcy lock button will keep the decay time at its current value even if the Time Scale control is adjusting the overall time of the envelope. However, decay time is still available for modulation through the modulation matrix.

loop - From the Loop menu, you can select between two different envelope looping modes.

dly-sus - Loops from the Delay stage to the Sustain stage, while the note is held.

dly-rls - Loops from the Delay stage to the Sustain stage, after the note is released.

time scale - Adjusts the Overall time of the Envelope to be longer or shorter. This parameter can be used either directly, or as a modulation destination, to increase or decrease the overall envelope time.

level scale - Increases or decreases the overall level of the Envelope stages.

sus pedal - Sets whether or not the Envelope will be held in its Sustain stage when the

sustain pedal is depressed.

sus keyon - Sets whether or not the Envelope will be held in its Sustain stage by Key On (Key held down).

vel curve - Provides a velocity curve response for the Vel Scale parameter. It essentially applies a curvature to the velocity signal that transforms a linear response into a non-linear output according to the shapes available in the menu.

vel scale - Allows you to scale the overall envelope level with note velocity. Positive values give higher output with higher velocity, while negative values cause the output to start at the highest level and modulate lower with higher velocities.

ADSR Envelope Additional Modulation Outputs:

S&H output - The ADSR envelope has a built-in Sample & Hold function and this is the output level. Use the S&H trig in to trigger the next sample. (This trigger input is found in the ADSR envelope's modulation matrix inputs list.)

latchout atk - This Gate output level is set High at the end of the Attack stage.

latchout dcy - This Gate output level is set High at the end of the Decay stage.

latchout sus - This Gate output level is set High at the end of the Sustain stage.

latchout rel - This Gate output level is set High at the end of the Release stage.

gate out dly - This Gate output level is set High for the duration of the Delay stage only.

gate out atk - This Gate output level is set High for the duration of the Attack stage only.

gate out dcy - This Gate output level is set High for the duration of the Decay stage only.

gate out sus - This Gate output level is set High for the duration of the Sustain stage only.

gate out rel - This Gate output level is set High for the duration of the Release stage only.

ADSR Envelope Additional Modulation Destinations:

reset trig - Modulation input will reset the ADSR Envelope to its beginning point. Like all Reset Triggers, the input control value must pass below the gate threshold and back above the threshold in order for a new trigger to be actuated. For more on Triggers and Gates, please see Appendix C at the end of this manual.

freeze gate - As long as the Freeze Gate is held above its threshold, the ADSR Envelope will freeze at its current value.

S&H trig in - The ADSR envelope has a Sample & Hold on its output. Use the S&H trig in mod destination to trigger the next sample of the Sample & Hold. It is triggered each time the input level goes from low to high.

Portamento Generator

The Portamento Generator creates a lag time from one modulation level to another based on input note values, and combines some advanced behavioral options with multiple simultaneous outputs for some unusual applications. However, it is typically routed to the Tuning of an Oscillator for the classic Portamento sliding effect.

Parameters

trig mode - Determines the type of slope generation from one note to the next and which keys are used to create the slope distance. Mode settings:

Normal - The latest Note ON played is the reference for subsequent notes. As you hold notes and then play new ones, the generator begins a ramping towards to the new note. If more notes are played while a ramp is in progress, those new notes will begin ramping to their notes from the position of the ramp-in-progress.

First - The first Note ON played is the reference for all subsequent notes. Holding down your first note, all subsequent notes will begin their ramp from that first note.

LegatoX - Same as Legato, except the first Note ON does not ramp from the last

note position. Only Legato notes will ramp.

Reset - Similar to Legato, except subsequently played notes will immediately ramp from the previous notes, whether a ramp-in-progress is happening or not. For example, with a slow slope time, if you hold C2, then hold C4, and then hold C3, the C3 note will begin ramping from C4 even while C4 is still ramping up from C2.

ResetX - Same as Reset, except the first Note ON does not ramp from the last note position. Only Legato notes will ramp.

RandOct - Every Note On will have a random ramp start point within +/- 1 Octave.

RandFull - Every Note On will have a random ramp start point from anywhere within the MIDI note range.

Octave - Every Note On will have a ramp start point 1 Octave away. When a new note is higher than the previous note, the new note will ramp from 1 Octave below. When a new note is lower than the previous note, the new note will ramp from 1 Octave above.

Octave+ - Every Note On will have a ramp start point from 1 Octave above.

Octave- - Every Note On will have a ramp start point from 1 Octave below.

slope shape - Applies a curvature to the portamento slope.

slope time - The time it takes for the Portamento generator to ramp to a new note. See the Time Mode parameter for details on the Constant Time vs. Constant Distance settings and how those modes affect the Slope Time. Note that the Slope Time parameter has no effect when Clock is set to an external source.

slope fine - Fine adjustment (cumulative) of the Slope Time parameter. The Slope Fine parameter has no effect when Clock is set to an external source.

time mode - Determines how ramp time between notes is calculated. Mode settings:

Const.Time - Every Note On will take the same amount of time to ramp from the previous note, regardless of how far apart the notes are. The time is determined by Slope Time when Clock is Internal, otherwise the Clock Settings determine the ramp time between notes.

Distance - The ramp time between notes will be calculated based on the distance between notes. In this mode, Slope Time represents the amount of

time for 1 Octave of ramp time. When Clock is an external source, the Clock Settings determine the ramp time for a 1 Octave distance. Shorter distances result in shorter times, while longer distances result in longer ramp times.

up time scal - Scales the time when a new note is ramping upwards. 100% is default.

up level scal - Adjusts the distance when a new note is ramping upwards. 100% is default. For example, playing C4 followed by C2 would normally ramp 2 octaves upward. If this setting is 50%, the ramp to C2 will begin from C3 instead, as that is 50% of the distance. Settings greater than 100% allow for wider ramps. Negative settings will invert the ramp.

dn time scal - Scales the time when a new note is ramping downwards. 100% is default.

dn level scal - Adjusts the distance of when a new note is ramping downwards. Otherwise it behaves in the same fashion as Up Level Scale.

key track - Based on a Center note of Middle C, higher Notes will result in shorter and shorter ramp times as you move up the keyboard, and lower notes will result in longer and longer ramp times as you move down the keyboard. Negative values will invert this scaling, causing longer times on higher keys, and shorter times on lower keys. The default value of 0% results in the Key played having no additional effect on the ramp time.

vel curve - Applies a curvature to the Vel Scale parameter adjusting how it responds to velocity. It essentially applies a curvature to the velocity signal that transforms a linear response into a non-linear output according to the shapes available in the menu.

vel scale - Allows for the Note On Velocity to affect the ramp time. At 100%, lower velocities will result in slower ramp times, and higher velocities will approach the actual Slope Time setting. Negative settings of this parameter will invert this behavior, allowing for higher velocities to slow down the ramp time. At 0% (the default), velocity has no effect on the ramp time.

clock - When set to Self (the default), the Portamento Generator will use the Slope Time parameter to set the ramp times for portamento. When set to System, the portamento time will synchronize to the host tempo.

sync div - When Clock is set to System, Sync Div determines the note value to use for ramp times. This parameter has no effect when Clock is set to Self.

Portamento Generator Additional Modulation Outputs:

cent out(f) - This is a fractional Cents output, representing the relative tuning position of the ramp to the new note, approaching 0.0 when it reaches the target note's pitch. The range is +/- 6000 cents. You can route it to Oscillator Tuning, using the sum.direct Mod Set Mode, to generate an exact tuning offset value.

semi out - This is a quantized Semitone output, representing the relative semitone position of the ramp to the new note, approaching 0.0 when it reaches the target note pitch. You can route it to Oscillator Tuning, using the sum.direct Mod Set Mode, to generate Glissando sweeps when ramping between notes.

semi whole - Same as Semi Out, but quantized to whole steps (2 semitones).

semi third - Same as Semi Out, but quantized to 3 semitone steps.

note out(f) - This is an absolute value output of the ramp, represented as a Note Number with fractional resolution. This is useful if you want to route an actual Note Value (with fractional note resolution) to your desired target.

note out(q) - Same as NoteOut(f), but without fractional resolution.

S&H output - The Portamento Generator has a built-in Sample & Hold function and this is the output level. Use the S&H trig in to trigger the next sample. (This trigger input is found in the Portamento Generator's modulation matrix inputs list.)

gate out beg - This is a GATE output, which will be HIGH (1.0) when the ramp is moving, and will be LOW (0.0) when the ramp has reached the target note.

gate out end - This is an end GATE output, which will be LOW (0.0) when the ramp is moving, and will be HIGH (1.0) when the ramp has reached the target note.

Portamento Generator Additional Modulation destinations:

reset trig - Modulation input will reset the Portamento Generator to its beginning point. Like all Reset Triggers, the input control value must pass below the gate threshold and back above the threshold in order for a new trigger to be actuated. For more on Triggers and Gates, please see Appendix C at the end of this manual.

freeze gate - As long as the Freeze Gate is held above its threshold, the Portamento Generator will freeze at its current value.

S&H trig in - The ADSR envelope has a Sample & Hold on its output. Use the S&H trig

in mod destination to trigger the next sample of the Sample & Hold. It is triggered each time the input level goes from low to high.

Random Generator

The Random Generator creates random levels at a particular frequency, but if desired the speed can be modulated. The movement between random output levels can also be smoothed for a very organic feel.

Parameters

min level - Sets the minimum level that will be generated from -100% to 0%.

max level - Sets the maximum level that will be generated from 0% to 100%.

speed - Adjusts the frequency at which new random levels are generated. This control is hidden if the clock is set to System.

range mult - This control allows you to multiply the speed setting value by up to 10 times. This control is hidden if the clock is set to System.

sync div - Adjusts the note division value at which the steps play back in relation to the System clock. This control is hidden if the clock is set to Self.

clock - Sets clock synchronization to Self or System (host clock). When the clock is set to System, the speed and range mult controls are replaced by sync div in order to set the note division value for the sequence.

trig mode - Governs how the Step Generator responds to notes and triggering. Mode settings:

Reset: Each Note ON played will start the sequence clock from the beginning and each new Note On will have its own clock start for that voice being played. Essentially every note is independent.

Free: A single free running clock determines the position of the step sequencer. All notes played will simply pick up wherever the free running clock happens to be when the note was played.

Legato: The first Note ON held down begins the clocking of the sequencer. Each subsequent note held down will match the clock position of the first note, so all notes will have the same start point and play together.

Random: Each Note On will start its own Step Generator at a random position.

smoothing - A time can be set to smoothly transition from one generated random number to the next. At 0%, there is no smoothing, resulting in an immediate jump to the next value. Values around 90% take about as long as the clock step, while values above 90% will take longer than a full clock step, which is useful for generating smooth wobbly signals.

sm.shape - Sets the shape (curve) applied when smoothing to the the level of the next generated random number.

sm.keytrk - Modifies the Smoothing Time based on Key Number. Positive values will result in shorter smoothing times with higher keys and longer smoothing times with lower keys.

step trig - This button is intended to be used as a modulation target in the Modulation Matrix, but it can be pressed manually as well. Step Trig will manually advance to the next random number, both when the clock is running and when in Freeze mode. This is useful when using another modulation source to trigger the generation of random values.

freeze - Turning freeze on will stop the clock, and the last generated value will hold in place.

Random Generator Additional Modulation Outputs:

S&H output - The Random Generator has a built-in Sample & Hold function and this is the output level. Use the S&H trig in to trigger the next sample. (This trigger input is found in the Random Generator's modulation matrix inputs list.)

trig out - A brief pulse is emitted each time a new random number is generated. This can be useful in the Mod Matrix to route to destinations that use a Trigger (eg. reset of an envelope, etc.).

square out - A simple square wave that is generated based on the clock that is driving the random number generator.

Slope Gate Beg - This value is High when the smoothing generator has begun and is

moving towards its next target value. This gate output coincides with Trig Out, but it will be High as long as the smoothing takes to complete.

Slope Gate End: This value goes High when the smoothing generator has reached the target value. Since smoothing tracks the clock, this gate value gives you a musical relative offset from the start of the Random Value generation clock.

Random Generator Additional Modulation destinations:

reset trig - High modulation input here will reset the Random Generator's clock. Like all Reset Triggers, the input control value must pass below the gate threshold and back above the threshold in order for a new trigger to be actuated. For more on Triggers and Gates, please see Appendix C at the end of this manual.

freeze gate - As long as the Freeze Gate is held above its threshold, the Random Generator will freeze at its current value.

speed(s) - Adjusts the Speed in percentage (Scalar). This modulation input is the easiest to use if you want to hear a linear modulation of the speed from your source. This input has no effect if the Random Generator is synchronized to the System Clock.

sp.adj(f) - Adjusts the Speed in frequency (logarithmic). This modulation input reacts to modulations with the same feel as the Speed(f) knob in the Random Generator module GUI. This input has no effect if the Random Generator is synchronized to the System Clock.

S&H trig in - The Random Generator has a Sample & Hold on its output. Use the S&H trig in mod destination to trigger the next sample of the Sample & Hold. It is triggered each time the input level goes from low to high.

Step Generator

The Step Generator is an 8-Step Sequencer with a number of features and outputs that can be simultaneously routed to multiple destinations.

Parameters

trig mode - Governs how the Step Generator responds to notes and triggering. Mode settings:

Reset: Each Note ON played will start the sequence clock from the beginning and each new Note On will have its own clock start for that voice being played. Essentially every note is independent.

Free: A single free running clock determines the position of the step sequencer. All notes played will simply pick up wherever the free running clock happens to be when the note was played.

Legato: The first Note ON held down begins the clocking of the sequencer. Each subsequent note held down will match the clock position of the first note, so all notes will have the same start point and play together.

Random: Each Note On will start its own Step Generator at a random position.

steps - Sets the number of steps in the sequence from 1 to 8 steps.

loop - Turns looping on and off for the sequence. If looping is on, when the sequence reaches its last step, it will start again at the beginning. If looping is off, the sequence will stop when it reaches the end.

speed - Adjusts the frequency at which the steps play back. This control is hidden if the clock is set to System.

range mult - This control allows you to multiply the speed setting value by up to 10 times.

sync div - Adjusts the note division value at which the steps play back in relation to the System clock. This control is hidden if the clock is set to Self.

clock - Sets clock synchronization to Self or System (host clock). When the clock is set to System, the speed and range mult controls are replaced by sync div in order to set the note division value for the sequence.

play dir - Determines the direction in which the sequenced notes play. Settings:

forward - Plays the sequence steps from first to last.

reverse - Plays the sequence steps from last to first.

fwd-rev - Plays the sequence steps from first to last, then reverses direction and plays from last to first.

random - Plays the sequence steps in random order.

level A - Scales the values of the A Step Levels. 100% is the default, and no change is being made to the A Step Levels at this value.

level B - Scales the values of the B Step Levels. 100% is the default, and no change is being made to the B Step Levels at this value.

smooth.ktrk - Modifies the Smoothing Times of all A Steps based on Key Number. Positive values will result in shorter smoothing times with higher keys and longer smoothing times with lower keys.

freeze - Stops the clock and the last generated value will hold in place.

step trig - The Step Trig will manually advance to the next step in the sequence, both when the clock is running and when in Freeze mode, as well as when the sequence has completed in non-Looping mode. This momentary switch is intended to be used as a modulation target in the Modulation Matrix.

A - SEQUENCE PARAMETERS:

Enable button - The row of buttons at the top of this section turns its corresponding step on when activated. When a step is disabled (off), the previous step value will just hold longer through the current step length.

Step Value A knobs - Sets the Level to be emitted for each step in a very old school analog way. No numerical values are shown.

Step Smoothing knobs - A smoothing time can be set to smoothly transition from one step to the next. 0% is no smoothing, resulting in an immediate jump to this step's value. Values around 90% take about as long as the clock step, and values above 90% will take longer than the next clock step, which can be combined with Enable-Off for subsequent steps in order to allow one step to smooth over a longer period of time.

Smoothing Shape - Sets the shape (curvature) of the smoothing when transitioning from the previous step to this setting's corresponding step value.

GATE AND TRIGGER OUTPUTS:

Gate Outs - Each sequence step can emit a Gate. A Gate is held high for the duration of a step. The Gate settings are not affected by the Enable settings of the A-Sequence

Steps.

Trigger Outs - Each sequence step can emit a Trigger. A Trigger is a short pulse emitted at the beginning of the step. The Trigger settings are not affected by the Enable settings of the A-Sequence Steps.

B SEQUENCE LEVELS:

Step Value B knobs - Sets the Level to be emitted for each step in a very old school analog way. No numerical values are shown. The B-Sequence Levels are a second sequence line of independent outputs. These are simple outputs with no smoothing, and they are not affected by the Enable setting of the A-Sequence Steps.

Step Generator Additional Modulation Outputs:

Output - This is a standard -1.0 through +1.0 output from the A-Sequence.

Output B - This is the -1.0 through +1.0 output from the B-Sequence.

cur step -

S&H A output - There is a built-in Sample & Hold function for the A-Sequence, and this is the output level. Use the S&H A trig in to trigger the next sample. (This trigger input is found in the Step Generator's modulation matrix inputs list.)

S&H B output - There is a built-in Sample & Hold function for the B-Sequence, and this is the output level. Use the S&H B trig in to trigger the next sample. (This trigger input is found in the Step Generator's modulation matrix inputs list.)

Semitone Out - This Output presents the A-Sequence Levels quantized to +48 through -48 semitones. This is intended to be routed to the Tuning destination of an oscillator using the "Sum Direct" modulation mode for correct quantization to equal tempered note values.

Trig Out Start - A brief pulse is emitted each time the sequence starts from the first step.

Trig Out Step - A brief pulse is emitted each time the sequence moves to a new step.

Trig Out Seq - A brief pulse is emitted based on the Trigger Out setting for each step.

Gate Out Seq - Sends a signal that is High while the sequence is in a step that has its Gate Out turned on.

Square Out Step - A simple square wave that is generated based on the clock that is driving the steps. A wave cycle is the length of each step.

Slope Gate Beg - This value is High when the smoothing generator has begun, and while it is moving towards the next step's value. This coincides with the Trig Step timing, but Slope Gate Beg will be High for as long as the smoothing takes to complete its operation.

Slope Gate End: This value goes High when the smoothing generator has reached the next step's target value.

Step Generator Additional Modulation destinations:

reset trigger - Modulation input will reset the Step Generator to its beginning step. Like all Reset Triggers, the input control value must pass below the gate threshold and back above the threshold in order for a new trigger to be actuated. For more on Triggers and Gates, please see Appendix C at the end of this manual.

freeze gate - As long as the Freeze Gate is held above its threshold, the Step Generator will freeze at its current value.

speed(s) - Adjusts the Speed in percentage (Scalar). This modulation input is the easiest to use if you want to hear a linear modulation of the speed from your source. This input has no effect if the Step Generator is synchronized to the System Clock.

sp.adj(f) - Adjusts the Speed in frequency (logarithmic). This modulation input reacts to modulations with the same feel as the Speed(f) knob in the Step Generator module GUI. This input has no effect if the Step Generator is synchronized to the System Clock.

S&H (A, B) trig in - The Step Generator has a Sample & Hold on each of its outputs. In the Step Generator's modulation matrix sources, you'll see the S&H A output and S&H B output that each send out Sample & Hold signal levels corresponding to samples of their respective sequencer lanes. Use the S&H trig in mod destinations to trigger the next sample of the Sample & Hold. It is triggered each time the input level goes from low to high.

MTRX Block NAV

This brings all currently available modulation routes into focus, and allows you to add, delete and edit modulation routes.

Focus Column

Voice Matrix tab - Shows all the mod routes at the voice level. Voice level mods are applied per voice and support polyphonic modulation behavior. They cannot be applied to Global level destinations such as the bus effects.

Global Matrix tab - Shows all the mod routes at the Global level. Global level mods are applied per patch and only support monophonic modulation behavior. Global mods are also useful for modulating non-polyphonic Global destinations such as the bus effects.

Menu Button - Provides drop down menu selections related to the Voice Matrix and Global Matrix tabs in context. For Voice and Global, the selections are as follows:

Add 1x1 Route

Add 1xN Route

You can have up to 64 mod routes at the Voice level and 64 mod routes at the Global level. In each case, up to 24 of those 64 routes can be 1xN routes, which are single source routes that target up to 16 destinations simultaneously.

Input Column

This is a contextual, tabbed display dependent on what is currently shown in focus (Center).

VOICE MATRIX focus tab selected:

CTLS tab - When the Ctls tab is selected, the Input column shows the collapsed

versions of any controls connected to the Voice Matrix modulation routes.

MODS tab - When the Mods tab is selected, the Input column shows the collapsed versions of any modulation sources connected to the Voice Matrix modulation routes.

MATRIX tab - When the Matrix tab is selected, the Input column shows the collapsed versions of any modulation routes affecting the Voice Matrix modulation routes.

GLOBAL MATRIX focus tab selected:

CTLS tab - When the Ctls tab is selected, the Input column shows the collapsed versions of any controls connected to the Global Matrix modulation routes.

MODS tab - When the Mods tab is selected, the Input column shows the collapsed versions of any modulation sources connected to the Global Matrix modulation routes.

MATRIX tab - When the Matrix tab is selected, the Input column shows the collapsed versions of any modulation routes affecting the Global Matrix modulation routes.

Output Column

This is a contextual, tabbed display dependent on what is currently shown in focus (Center).

VOICE MATRIX focus tab selected:

DESTS tab - When the DESTS tab is selected, the Output column shows all the destination modules currently affected by the Voice Matrix modulation routes in focus.

MATRIX tab - When the Matrix tab is selected, the right column shows all the modulation routes affected by the Voice modulation routes in focus.

GLOBAL MODS focus tab selected:

DESTS tab - When the DESTS tab is selected, the Output column shows all the

destination modules currently affected by the Global Matrix modulation routes in focus.

MATRIX tab - When the Matrix tab is selected, the right column shows all the modulation routes affected by the Global modulation routes in focus.

Modulation Routes

Modulation routes can connect modulation sources to nearly every parameter in the synth. These routes each have adjustable parameters, and will reveal additional mod sources and destinations that are available only when viewed within the modulation matrix. You can find a list of these special mod sources in Appendix D, while the special destinations are found in the descriptions of each synthesis module.

Aside from the continuous modulations available from control sources and modulators like LFOs and Envelopes, triggers and gates can also be sent via the matrix to make events happen throughout the synth. For more on triggers and gates, please see Appendix C at the end of this manual.

Each route has a header that displays the type of route (either 1x1 or 1xN), a Focus button that brings the module into focus in the center column, a Mute button that stops the modulation output, and a drop down Menu that allows you to delete the route. For a 1xN route, the Menu also presents an option for adding destinations.

Each destination has a triangular button to its left that, when clicked, will expand the destination to show its associated parameters. (These parameters affect how the modulation behaves.) In the case of a 1xN route, there is also an "X" button to the right of the destinations that can be used to delete its associated destination from the modulation route.

1x1 Route

The 1x1 Mod Route is a single source to single destination route, sending a modulation signal to one target parameter.

1xN Route

The 1xN Mod Route is a single source to multiple destination route, used to send a modulation signal to between 1 and 16 target parameters.

Parameters

Source Level - Amount to scale the source signal. This is a bipolar parameter, so negative values will invert the modulation polarity. Note that the final output will be limited to the minimum and maximum value range of the destination parameter.

Source - Menu for selecting categorized modulation sources. Note that some sources have multiple outputs, such as triggers and gates as well as levels. Even some System level functions such as CPU Usage can be used as modulation sources. (For more information on the System sources, please see Appendix D - System Modulation Sources.)

Source Modifier Select - Chooses between six control sources to scale the Source Level amount. Choices are:

None (off)

Mod Wheel

Pitch Wheel

Aftertouch

Sustain Pedal

Velocity

Scale - Sets the amount of scaling applied by the Source Modifier.

Bias - Applies an offset to the Source Modifier scaling amount.

Destination Level - Amount to scale the signal output to the destination. This is a bipolar parameter, so negative values will invert the modulation polarity. Note that the final output will be limited to the minimum and maximum value range of the destination parameter.

Destination - Menu for selecting categorized modulation destinations. Note that some target modules have additional modulation destinations beyond their visible parameters. These “hidden” destinations often provide alternate functionality when modulated.

Polarity - Changes the polarity mode of the modulation signal. Mode settings:

Bipolar - Allows both positive and negative going values. For example, a sine wave will begin at 0, rise to 1, then fall through 0 to -1, and finally return to 0.

Unipolar - This will shift the signal into a positive (0 through 1) value range. For example, a sine wave in Unipolar mode will begin at 0.5, rise to 1, then fall to 0.5, then to 0, and finally return to 0.5.

Rectified - This forces a bipolar signal to always be positive. Any negative values will be flipped into a positive mirror image of their signal, while leaving positive values alone. For example, the bipolar sine wave shape after being rectified will begin at 0, rise to 1, then fall to 0, then rise to 1 again, then fall to its beginning point of 0.

Recti-Chop - This forces a bipolar signal to always be positive. Any negative values will be simply chopped off, while leaving positive values alone. For example, the bipolar sine wave shape after being recti-chopped will begin at 0, rise to 1, then fall to 0, then stay flat for half a cycle, then rise to 1, then fall back to its beginning point of 0.

Recti-Flip - This forces a bipolar signal to always be positive. Any negative values will be offset into a fully positive signal, while leaving positive values alone. For example, the bipolar sine wave shape after being recti-flipped will begin at 0, rise to 1, then fall to 0, then jump to 1, fall to 0, then rise to 1, then jump back to its beginning point of 0.

Curve - Applies a curvature to the modulation output signal that transforms a linear response into a non-linear output response according to the shapes available in the menu.

DC - Amount of DC offset (bias) applied to the source signal after Level adjustment. Using this parameter allows for sliding the final output value up or down, which is useful for positioning a modulation signal into a particular range of the destination parameter. The DC offset amount is bipolar, so negative or positive bias can be applied. Note that

the final scaling will be limited to the minimum and maximum value range of the destination parameter.

Set Min - Sets a minimum modulation value that the route can pass to the destination. Any values below this value will be clipped.

Set Max - Sets a maximum modulation value that the route can pass to the destination. Any values above this value will be clipped.

Mod Set Mode - This parameter determines the behavior of the modulation signal and how it is applied to the destination parameter. These modes are very specific and different in their operation. For most applications, the default mode will work fine, however it is advisable to understand how these modes work, as that knowledge will allow you to achieve your exact desired modulation behavior. Mode settings:

sum.ranged - The scaled source is normalized between -1.0 to +1.0 and is then “fitted” to the Set Min and Mod Set Max values, to create a full modulation range, which is then summed to the current destination parameter value. This allows for modulation of the target that re-scales as you change the Min/Max modulation range.

sum.direct - The direct source value is added to the destination parameter. This allows for direct value modulation of the target. For example, if you have a source that generates an exact number of semitones, the value can be added, without scaling or shaping, to the destination. **NOTE:** This mode may have no perceivable effect in many cases. For example, an LFO can generate bipolar +/-1.0 output values, and if you route this using sum.direct to a Filter Frequency target, you will be modulating the filter cutoff +/-1Hz in total... This will likely not be perceivable. During the summation of the modulation and destination values, the resulting value will not be allowed to exceed the Set Min and Set Max boundary values.

sum.scaled - The scaled source is normalized between -1.0 to +1.0 and then “fitted” to the destination parameter's full range, which is then summed to the current destination parameter value. The resulting destination parameter value will not be allowed to exceed the Set Min and Set Max boundary values.

mul.direct - Same as sum.direct, but with multiplication instead of summation. **NOTE:** Multiplication can be tricky... If your destination parameter is at ZERO, all modulation values will result in a destination value of zero as well because

anything multiplied by zero is zero. Therefore, no modulation will be heard. The same thing will happen if your modulation source is at zero.

mul.scaled - Same as sum.scaled, but with multiplication instead of summation. (See NOTE in mul.direct above.)

mul.ranged - Same as sum.ranged, but with multiplication instead of summation. (See NOTE in mul.direct above.)

sum(-)scaled - The scaled source value is fitted to a range between the destination parameter's minimum and current values, and then modulates a percentage of that range. The resulting destination parameter value will not be allowed to exceed the Set Min and Set Max boundary values.

sum(+)scaled - The scaled source value is fitted to a range between the destination parameter's current and maximum values, and then modulates a percentage of that range. The resulting destination parameter value will not be allowed to exceed the Set Min and Set Max boundary values.

sum(-)ranged - The scaled source value is fitted to a range between the Set Min value and destination parameter's current value, and then it modulates a percentage of that range. The resulting destination parameter value will not be allowed to exceed the Set Min and Set Max boundary values. NOTE: If Set Min is set to a value that exceeds the destination parameter's current value, no modulation will occur.

sum(+)ranged - The scaled source value is fitted to a range between the destination parameter's current value and the Set Max value, and then it modulates a percentage of that range. The resulting destination parameter value will not be allowed to exceed the Set Min and Set Max boundary values. NOTE: If Set Max is set to a value below the destination parameter's current value, no modulation will occur.

set.direct - The destination parameter is simply set to the value of the direct modulation signal. The current value of the destination parameter is ignored, and therefore user adjustment of the target parameter will have no effect since the mod route will be setting it directly.

set.scaled - The scaled source is normalized between -1.0 to +1.0 and "fitted" to the destination parameter's full range, and then the destination is set to this value. The current value of the destination parameter is ignored, and therefore

user adjustment of the target parameter will have no effect since the mod route will be setting it directly.

set.ranged - The scaled source value is “fitted” to the Set Min and Set Max values, and the destination parameter is then set to this value. The current value of the destination parameter is ignored, and therefore user adjustment of the destination will have no effect since the mod route will be setting it directly.

Destination Modifier Select - Chooses between six control sources to scale the Destination Level amount. Choices are:

None (off)

Mod Wheel

Pitch Wheel

Aftertouch

Sustain Pedal

Velocity

Scale - Sets the amount of scaling applied by the Destination Modifier.

Bias - Applies an offset to the Destination Modifier scaling amount.

Lag Up - Creates a lag time (glide) only when modulation values go from low to high.

Lag Down - Creates a lag time (glide) only when modulation values go from high to low.

INPT Block NAV

This brings the audio input module into focus, and allows you to adjust levels throughout its audio path and even add a filter module to tweak the sound.

Focus Column

Audio Input Filter - The audio input focus only has one module, which is used to insert a filter, effect or audio routing block on the input. It has a header that is a drop down menu for selecting which DSP function should populate the module, and a Mute button that bypasses the module. The available DSP functions are:

chorus	ladder	env follow
delay	sk35 filter	gain
delay dual	X pole	resonator
flanger	ladder	ring math
reverb	X pole sv	saturator
1 pole	all pass	shaper
1 pole dual	bit offen	tap line 1
1 pole va	comb	tap line 5
2 pole	comb lpf	tap line 8
2 pole sv	compressor	tap line 8X
bi-quad	dc blocker	wavefolder
diode	decimator	

The efx split and efx bank selections are actually structures that can hold multiple DSP

functions within them. The amount of processor power allocated to the audio input filter is limited, so if you exceed the maximum amount you won't be able to add any more DSP functions. Note that some DSP functions in the list require more processor power than others. More information on individual Filter and Effects can be found in their respective sections in this manual.

Input Column

AUDIO INPUT is the single tab available and it shows the Audio Input level control, which is also available on the main performance page.

Output Column

This is a tabbed display with three tabbed choices available in relation to the Audio Input.

OMIX tab - When the OMIX tab is selected, the Output column shows a condensed version of the Oscillator Mixer that focuses only on the level controls related to the Audio Inputs as they are directly feeding the synthesizer's Filter section.

VAMP tab - When the VAMP tab is selected, the Output column shows a condensed version of the Voice Amp Mixer so that main output levels can be adjusted in relation to the Audio Input Level.

PREMIX tab - When the PREMIX tab is selected, the Output column shows a condensed version of the Pre FX Mixer which adjusts the Audio Input Level against the voice outputs at a point after the amplitude envelopes and prior to the effects.

OSC 1-3 Block NAV

These navigation buttons bring your desired oscillator module (1-3) into focus. Oscillators 1 and 2 are monophonic, but can be panned at the Amp stage. Oscillator 3 is stereo and its waveforms, while monophonic in themselves, can be panned at the oscillator. Oscillator 3 will also support a stereo audio input.

Focus Column

OSCILLATOR 1-3 tab - Displays the chosen oscillator module for editing.

VOICE MODE tab - Shows the common voice mode parameters that govern how the synth voices will play.

Input Column

This is a contextual, tabbed display, the contents of which are dependent on what is currently shown in focus (Center).

OSCILLATOR 1-3 focus tab selected:

MODS tab - When the MODS tab is selected, the Input column shows the collapsed versions of any modulation sources routed to the oscillator in focus.

CTLS tab - When the CTLS tab is selected, the Input column shows the collapsed versions of any main performance page controls routed to the oscillator in focus.

INPT tab - When the INPT tab is selected, the Input column shows the main performance page Audio Input level control and filter setting. The Audio Input is available as a wave shape selection to any of the oscillators.

MTRX tab - When the MTRX tab is selected, the Input column shows the collapsed

versions of any modulation routes affecting the oscillator in focus.

VOICE MODE focus tab selected:

When Voice Mode is in focus, the Input column is blank because there is no audio input to this module and it cannot be modulated.

Output Column

This is a contextual, tabbed display dependent on what is currently shown in focus (Center).

OSCILLATOR 1-3 focus tab selected:

OMIX tab - When the OMIX tab is selected, the Output column shows a condensed version of the Oscillator Mixer where you can control the parallel mix of the focus oscillator into the three filters. NOTE: When the Oscillator Mixer module is in focus, you can also access the serial mix levels of the three filters.

FILT tab - When the FILT tab is selected, the right column shows condensed versions of the filter modules in the filter section of the synth.

VOICE MODE focus tab selected:

When Voice Mode is in focus, the Output column is blank because there is no audio output from this module and it cannot be used for modulation.

OSCILLATOR Modules

The oscillator modules slice up waveforms in order to generate new output wave shapes.

These modules have adjustable parameters for all individual wave slices, as well as for the “Razor” itself.

Each module has a header that displays the name of the oscillator and a Mute button that stops the oscillator audio output. In the body of the module is a waveform graphic display that shows the wave slices and their contents. Directly below the waveform display are navigation tabs that select the Razor (R) or wave slices (between 1 and 16, depending on the number of steps selected) for editing.

Oscillator Parameters

Level - Sets the output level for the entire oscillator, from -100% to 100%. Negative values will invert the polarity of the waveform, e.g. -100% is the same level as 100%, but with inverted polarity.

Pan (Only available for Oscillator 3) - Sets the output panning from LEFT -100%, to CENTER 0, to RIGHT 100%.

Razor Parameters

The Razor is the unique component of the oscillator that sequences the individual wave slices into a full waveform.

num steps - Defines the number of waveform slices available for creating the composite waveform output.

step size - Changes the segment size, from fractions of a cycle to multiples of a cycle.

step frac - Fractionally increases or decreases the Step Size. Under most circumstances, adjusting this parameter will make the overall wave cycle fractionally unaligned, creating inharmonic frequencies.

step fine - Provides fine tuning of the Step Frac parameter. This only has an effect when Step Frac is set to something other than 100%.

tune octave - This is a quantized octave tuning adjustment. It is useful for when you want to adjust or modulate only the octave setting of the Razor.

tune semi - This is a quantized semitone tuning adjustment with a range of +/- 1 octave. It is useful for when you want to adjust or modulate only the semitone (note) setting of the Razor.

tune cents - This is a small range (+/- 1 semitone) in cents for fine tuning adjustment. It is useful for making small changes to the Razor tuning.

window duty - Allows adjustment of duty cycle of alternating steps of the Razor segments.

step cycle - This parameter aligns the overall waveform sequence, based on its number of steps and step size, to the Razor cycle. Settings:

off - Disables cycle alignment.

integral - Aligns the overall wave cycle to the best-fit integral value based on the num steps and step size settings.

nearest, nearer, near - These settings behave the same as integral, but they allow for more rounding in the cycle alignment.

1 - 64 cycles - Forces the wave cycle alignment to a specific number of cycles. This can sometimes have an interesting effect when aligning a waveform to a number of cycles that is at odds with the number of steps and step size.

For example, If Num Steps is set to 4 and Step Size is 1/3, the overall waveform will take four cycles to repeat itself. The "Integral" setting would align itself to four cycles and ensure that the waveform sequence remains synchronized to this pattern. However, if you set Step Cycle to a value of 1, this will force an alignment to one cycle (instead of four) and create a discontinuous transition which generates additional harmonics.

NOTE: The function of the Step Cycle parameter is not always obvious and it can cause glitches or unexpected side-bands at certain frequencies, notes, and/or filter cutoffs. Therefore it is recommended to leave this set to "integral" if you want to continue with normal operation.

trig mode - Governs how the oscillator begins oscillating. Mode settings:

Reset - The Razor will always start at wave segment 1, and the wave segments begin according to their Start Phase settings.

Free - The oscillator will always begin at a position based on a free-running reference oscillator.

Legato - The Razor will start at wave segment 1, and the wave segments begin according to their Start Phase settings when the first note is received, but do not reset as long as notes are played legato (overlapping). When all notes have been let up, the oscillator will reset again when the next note is received.

Random - The oscillator will begin at a random start position and phase.

key track - Adjusts the Key Tracking of the Razor, relative to its pitch. 100% is the default value, meaning that the pitch will move one octave with each octave on the keyboard. Smaller values will result in smaller pitch changes for each keyboard note movement, and larger values result in larger changes. To reverse the pitch to keyboard relationship, you can set Key Track to negative values. Key Tracking is centered around C3.

key track on - Enables or disables Key Track for the Razor.

wave tracking - Enables or disables key tracking for any of the wave segments that are set to follow the main oscillator's key tracking. This has no effect on segments that are set to Self or Off.

polarity - Flips the polarity of the oscillator's waveform output.

bend range - Sets the amount that the Pitch Wheel affects Oscillator Tuning. The default is 200 cents (2 semitones). Positive Values will result in a pitch increase during Pitch Wheel Up, and a pitch decrease with Pitch Wheel down. To invert the behavior of the Pitch Wheel, you can use negative Bend Range values. If you want to create different ranges for the Bend Up and Down, you can use Bend Range in combination with the Bend Add Up and Bend Add Down parameters, which extend the Bend Range in their respective directions.

bend add up - This parameter Adds (or Subtracts) a number of cents to (from) the Bend Range parameter for Pitch Wheel Up only. For example, if you want a Bend Range of 200 Cents Down, and 500 Cents Up, you can set the Bend Range to 200, and then set Bend Add Up to 300.

bend add dn - This parameter Adds (or Subtracts) a number of cents to (from) the Bend Range parameter for Pitch Wheel Down only. For example, if you want a Bend Range of 500 Cents Down, and 200 Cents Up, you can set the Bend Range to 200, and then set this parameter to 300.

vel scale - Allows you to scale the overall oscillator output level with note velocity. Positive values give higher output with higher velocity, while negative values cause the output to start at the highest level and modulate lower with higher velocities.

vel curve - Provides a velocity curve response for the Vel Scale parameter. It essentially applies a curvature to the velocity signal that transforms a linear response into a non-linear output according to the shapes available in the menu.

Razor Additional Modulation destinations:

reset trig - This Trigger Input resets the oscillator to its Start Phase setting, regardless of the Trigger Mode of the oscillator.

num waves - Defines the number of waveforms available for creating the composite waveform output. If the number of waveforms available is less than the number of steps in the Razor, the difference in slices at the end of the sequence will still play back, but they will be empty.

tuning - This is the overall Tuning, in cents, for the Razor. It is a useful Modulation destination parameter when you want to sweep the pitch of the Razor over a large, un-quantized range.

Wave Segment Parameters

shape - Selects which wave shape to use in the segment. Aside from a variety of waveform variations, choices also include “none” and the Audio Inputs.

wav# level - Sets the output level for the wave segment, from -100% to 100%. Negative values will invert the polarity of the waveform, e.g. -100% is the same level as 100%, but with inverted polarity.

wav# phase - Wav Phase sets the starting phase of the wave segment. You can choose a setting from 0 to 360 degrees in relation to the full waveform cycle. This parameter functions normally when the oscillator Trigger Mode is set to RESET. With the FREE Trigger Mode, the Start Phase will be an offset from the current free-running phase, and with RANDOM Trigger modes, start phase will have no effect. However, when a Reset Trigger is received it overrides any Trigger Mode setting and the wave segment will reset to its Start Phase value.

wav# duty - Allows adjustment of waveform duty cycle (the amount of positive cycle time versus negative cycle time) for ANY wave shape. The range is from 5% to 95%.

wav# dc - Adjusts the amount of DC offset (bias) applied to the wave segment. The DC offset amount is bipolar, so negative or positive bias can be applied.

wav# octave - This is a quantized octave tuning adjustment. It is useful for when you want to adjust or modulate only the octave setting of the wave segment.

wav# semi - This is a quantized semitone tuning adjustment with a range of +/- 1 octave. It is useful for when you want to adjust or modulate only the semitone (note) setting of the wave segment.

wav# cents - This is a small range (+/- 1 semitone) in cents for fine tuning adjustment. It is useful for when you are creating small changes to wave segment tuning.

wav# key track - Adjusts the Key Tracking of the wave segment, relative to its pitch. 100% is the default value, meaning that the pitch will move one octave with each octave on the keyboard. Smaller values will result in smaller pitch changes for each keyboard note movement, and larger values result in larger changes. To reverse the pitch to keyboard relationship, you can set Key Track to negative values. Key Tracking is centered around C3.

wav# track mode - Selects whether Key Track for the segment follows the Main Oscillator, its own key track parameter (self), or is turned off.

wav# velcurve - Provides a velocity curve response for the Vel Scale parameter. It essentially applies a curvature to the velocity signal that transforms a linear response into a non-linear output according to the shapes available in the menu.

wav# velscale - Allows you to scale the wave segment level with note velocity. Positive values give higher output with higher velocity, while negative values cause the output to start at the highest level and modulate lower with higher velocities.

wav# bend on - Enables pitch bend for the wave segment.

wav# constpol - With Constpol (short for Constant Polarity) set to On, it keeps the wave energy in the polarity that it's meant to occupy, according to its normal phase relationship to the step configuration of the Razor and its frequency. If the wave crosses over the zero boundary it is wrapped back into the correct polarity.

wav# sync - Resets the wave in the segment to its starting phase when active. Mode settings:

none - The wave segment starts according to the Voice Mode parameter settings.

cycle - Resets the wave segment when a Razor cycle is completed. Completion of the Razor cycle is dependent on the number of steps, in relation to step size, and is defined as when the waveform sequence repeats itself.

wave - The segment will reset on every wave period. This will keep the segment in the same phase position with each wave period even when the cycle length is greater than 1, as in the case of odd step and size combinations.

step - The segment will reset on every segment step. The wave will begin from its start phase, in every step in which it appears.

Wave Segment Additional Modulation Destinations:

tuning - This is the overall Tuning, in cents, for the wave segment. It is a useful Modulation destination parameter when you want to sweep the pitch of the wave segment over a large, un-quantized range.

COMMON VOICE MODE Module

The Voice Mode module contains parameters that govern how the synth plays its voices in relation to MIDI input.

Parameters

voice mode - Selects between Poly and Mono operation. Poly allows for playing more than one voice at a time, while Mono limits the synth to one voice at a time no matter how many MIDI notes are input.

num voices - Sets the upper limit to the number of voices that can be played simultaneously.

voice steal - Defines how the voices are stolen when the upper limit of polyphony is reached. Modes:

direct - The voice is stolen immediately.

release - The amplitude envelope of the voice to be stolen immediately plays its release stage, briefly overlapping with the new voice.

steal order - Defines which voices are stolen when the upper limit of polyphony is reached. Modes:

stage - Selects the voice that is farthest along in its amplitude envelope for stealing.

oldest - Selects the oldest sounding voice for stealing.

newest - Selects the latest playing voice for stealing.

MIX Block NAV (Oscillator Mixer)

The Oscillator Mixer is responsible for mixing all of the available audio routings from oscillators, audio input, and feedback paths into the filters. It's also used for mixing the filter routings in series, parallel, or in a combination of both.

Focus Column

OMIX tab - Displays the Oscillator Mixer module (OMIX) for editing.

Input Column

This is a contextual, tabbed display, the contents of which are dependent on what is currently shown in focus (Center).

OSCS tab - When the OSCS tab is selected, the Input column displays collapsed versions of Oscillators 1, 2 and 3.

MODS tab - When the MODS tab is selected, the Input column shows the collapsed versions of any modulation sources routed to the Oscillator Mixer module.

Output Column

FILT tab - Displays collapsed versions of Filters 1, 2 and 3 for editing.

Oscillator Mixer Module (OMIX)

The Oscillator Mixer is responsible for mixing all of the available audio routings into the filters. Each parameter is the mix level for an audio route with a range from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity. These are the channels available for mixing:

Filter 1 Mix

osc1 ->flt1 - Sets the level sent from Oscillator 1 to Filter 1.

osc2 ->flt1 - Sets the level sent from Oscillator 2 to Filter 1.

osc3L ->flt1 - Sets the level sent from Oscillator 3 Left to Filter 1.

osc3R ->flt1 - Sets the level sent from Oscillator 3 Right to Filter 1.

audInL->flt1 - Sets the level sent from Audio Input Left to Filter 1.

audInR->flt1 - Sets the level sent from Audio Input Right to Filter 1.

fbackL->flt1 - Sets the level sent from Feedback Left to Filter 1.

fbackR->flt1 - Sets the level sent from Feedback Right to Filter 1.

Filter 2 Mix

osc1 ->flt2 - Sets the level sent from Oscillator 1 to Filter 2.

osc2 ->flt2 - Sets the level sent from Oscillator 2 to Filter 2.

osc3L ->flt2 - Sets the level sent from Oscillator 3 Left to Filter 2.

osc3R ->flt2 - Sets the level sent from Oscillator 3 Right to Filter 2.

audInL->flt2 - Sets the level sent from Audio Input Left to Filter 2.

audInR->flt2 - Sets the level sent from Audio Input Right to Filter 2.

fbackL->flt2 - Sets the level sent from Feedback Left to Filter 2.

fbackR->flt2 - Sets the level sent from Feedback Right to Filter 2.

Filter 3 Mix

osc1 ->flt3L - Sets the level sent from Oscillator 1 to Filter 3 Left.

osc1 ->flt3R - Sets the level sent from Oscillator 1 to Filter 3 Right.

osc2 ->flt3L - Sets the level sent from Oscillator 2 to Filter 3 Left.

osc2 ->flt3R - Sets the level sent from Oscillator 2 to Filter 3 Right.

osc3L ->flt3L - Sets the level sent from Oscillator 3 Left to Filter 3 Left.
osc3L ->flt3R - Sets the level sent from Oscillator 3 Left to Filter 3 Right.
audInL->flt3L - Sets the level sent from Audio Input Left to Filter 3 Left.
audInL->flt3R - Sets the level sent from Audio Input Left to Filter 3 Right.
fbackL->flt3L - Sets the level sent from Feedback Left to Filter 3 Left.
fbackL->flt3R - Sets the level sent from Feedback Left to Filter 3 Right.
osc3R ->flt3L - Sets the level sent from Oscillator 3 Right to Filter 3 Left.
osc3R ->flt3R - Sets the level sent from Oscillator 3 Right to Filter 3 Right.
audInR->flt3L - Sets the level sent from Audio Input Right to Filter 3 Left.
audInR->flt3R - Sets the level sent from Audio Input Right to Filter 3 Right.
fbackR->flt3L - Sets the level sent from Feedback Right to Filter 3 Left.
fbackR->flt3R - Sets the level sent from Feedback Right to Filter 3 Right.

Filter Configuration Mix

flt1->flt2 - Sets the level sent from Filter 1 to Filter 2.
flt1->flt3L - Sets the level sent from Filter 1 to Filter 3 Left.
flt1->flt3R - Sets the level sent from Filter 1 to Filter 3 Right.
flt2->flt3L - Sets the level sent from Filter 2 to Filter 3 Left.
flt2->flt3R - Sets the level sent from Filter 2 to Filter 3 Right.

FLT 1-3 Block NAV

These navigation buttons bring your desired main filter block (1-3) into focus. Filter blocks 1 and 2 are monophonic, but can be panned at the Amp stage. Filter block 3 is stereo.

Focus Column

FILTER 1-3 tab - Displays the chosen filter module for editing.

Input Column

This is a contextual, tabbed display, the contents of which are dependent on what is currently shown in focus (Center).

OMIX tab - When the OMIX tab is selected, the Input column displays a collapsed version of the Oscillator Mixer module, showing only the mix inputs to the filter.

MODS tab - When the MODS tab is selected, the Input column shows the collapsed versions of any modulation sources routed to the filter in focus.

CTLS tab - When the CTLS tab is selected, the Input column shows the collapsed versions of any main performance page controls routed to the filter in focus.

MTRX tab - When the MTRX tab is selected, the Input column shows the collapsed versions of any modulation routes affecting the filter in focus.

Output Column

This is a contextual, tabbed display dependent on what is currently shown in focus (Center).

VAMP tab - When the VAMP tab is selected, the Output column displays a condensed version of the Voice Amp Mixer showing only the main output mix controls. It also displays any active Amp Envelopes.

Filter Modules

The Filter modules allow you to add filters and simple effects to the audio path. The three available filter blocks can be routed in series and/or parallel, but can also be populated with sub configurations that allow multiple filters to be routed within a single block for extremely powerful filtering options.

Each module has a header that displays the name of the filter and a Mute button that bypasses the filter.

Note: Filters have an Input and Output Boost level amount to provide additional (or scaled down) gain amounts to the input and output levels. These are provided separately to allow for a gain structure that drives a filter with makeup gain, but still allows the Input and Output levels to be modulated independently of the boost gain structure.

1 pole

A 1 Pole Filter capable of Hi Pass or Low Pass operation.

Parameters

in level - Sets the input level to the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

cutoff(s) - This is the filter cutoff frequency scaled with a natural feeling curve. Setting

this parameter overrides and updates the Tune Note and Tune Cents parameters.

mode - Selects whether the filter operates as a Hi Pass or Low Pass filter.

key track - Adjusts the Key Tracking of the filter, relative to its cutoff frequency. At a setting of 0%, the cutoff of the filter will be the same, regardless of the note number that triggered the voice. At 100%, the cutoff frequency will double with each octave on the keyboard. To reverse the cutoff to keyboard relationship, you can set key track to a negative value. Key Tracking is centered around C3.

tune note - Allows you to set the cutoff frequency based on a musical note. This parameter works in conjunction with the tune cents parameter, and setting either will override and update the cutoff(s) parameter.

tune cents - Adjusts the tuning of the cutoff frequency in musical terms. This parameter works in conjunction with the tune note parameter, and setting either will override and update the cutoff(s) parameter.

1 pole Additional Modulation sources:

tune note - Provides a quantized modulation output based on the current filter cutoff. The quantization is an absolute note value.

tune octave - Provides a quantized modulation output based on the current filter cutoff. The quantization is in octaves, but can be offset from true octave tuning.

tune semi - Provides a quantized modulation output based on the current filter cutoff. The quantization is in semitones, but can be offset from true semitone tuning.

tune cents - Provides a modulation output based on the current filter cutoff in cents.

1 pole Additional Modulation destinations:

cutoff(f) - Allows non-scaled direct modulation of frequency.

tune octave - Allows you to modulate the cutoff frequency based on musical octaves in relation to the current cutoff setting.

tune semi - Allows you to modulate the cutoff frequency based on musical semitones in relation to the current cutoff setting.

1 pole dual

A pair of 1 Pole Filters that can be configured independently.

Parameters

in level - Sets the input level to the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

1 cutoff(s) - This is the filter cutoff frequency scaled with a natural feeling curve. Setting this parameter overrides and updates the 1 Tune Note and 1 Tune Cents parameters.

1 mode - Selects whether the filter operates as a Hi Pass or Low Pass filter.

1 key track - Adjusts the Key Tracking of the filter, relative to its cutoff frequency. At a setting of 0%, the cutoff of the filter will be the same, regardless of the note number that triggered the voice. At 100%, the cutoff frequency will double with each octave on the keyboard. To reverse the cutoff to keyboard relationship, you can set key track to a negative value. Key Tracking is centered around C3.

1 tune note - Allows you to set the cutoff frequency based on a musical note. This parameter works in conjunction with the 1 tune cents parameter, and setting either will override and update the 1 cutoff(s) parameter.

1 tune cents - Adjusts the tuning of the cutoff frequency in musical terms. This parameter works in conjunction with the 1 tune note parameter, and setting either will override and update the 1 cutoff(s) parameter.

2 cutoff(s) - This is the filter cutoff frequency scaled with a natural feeling curve. Setting this parameter overrides and updates the 2 Tune Note and 2 Tune Cents parameters.

2 mode - Selects whether the filter operates as a Hi Pass or Low Pass filter.

2 key track - Adjusts the Key Tracking of the filter, relative to its cutoff frequency. At a setting of 0%, the cutoff of the filter will be the same, regardless of the note number that triggered the voice. At 100%, the cutoff frequency will double with each octave on the keyboard. To reverse the cutoff to keyboard relationship, you can set key track to a negative value. Key Tracking is centered around C3.

2 tune note - Allows you to set the cutoff frequency based on a musical note. This parameter works in conjunction with the 2 tune cents parameter, and setting either will override and update the 2 cutoff(s) parameter.

2 tune cents - Adjusts the tuning of the cutoff frequency in musical terms. This parameter works in conjunction with the 2 tune note parameter, and setting either will override and update the 2 cutoff(s) parameter.

1 pole dual Additional Modulation sources:

1 tune note - Provides a quantized modulation output based on the current filter 1 cutoff. The quantization is an absolute note value.

1 tune octave - Provides a quantized modulation output based on the current filter 1 cutoff. The quantization is in octaves, but can be offset from true octave tuning.

1 tune semi - Provides a quantized modulation output based on the current filter 1 cutoff. The quantization is in semitones, but can be offset from true semitone tuning.

1 tune cents - Provides a modulation output based on the current filter 1 cutoff in cents.

2 tune note - Provides a quantized modulation output based on the current filter 2 cutoff. The quantization is an absolute note value.

2 tune octave - Provides a quantized modulation output based on the current filter 2 cutoff. The quantization is in octaves, but can be offset from true octave tuning.

2 tune semi - Provides a quantized modulation output based on the current filter 2 cutoff. The quantization is in semitones, but can be offset from true semitone tuning.

2 tune cents - Provides a modulation output based on the current filter 2 cutoff in cents.

1 pole dual Additional Modulation destinations:

1 cutoff(f) - Allows non-scaled direct modulation of frequency.

1 tune octave - Allows you to modulate the cutoff frequency based on musical octaves in relation to the current cutoff setting.

1 tune semi - Allows you to modulate the cutoff frequency based on musical semitones in relation to the current cutoff setting.

2 cutoff(f) - Allows non-scaled direct modulation of frequency.

2 tune octave - Allows you to modulate the cutoff frequency based on musical octaves in relation to the current cutoff setting.

2 tune semi - Allows you to modulate the cutoff frequency based on musical semitones in relation to the current cutoff setting.

2 pole

A 2 Pole Filter capable of Hi Pass, Low Pass, or Band Pass operation.

Parameters

in level - Sets the input level to the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

cutoff(s) - This is the filter cutoff frequency scaled with a natural feeling curve. Setting this parameter overrides and updates the Tune Note and Tune Cents parameters.

mode - Selects whether the filter operates as a Hi Pass, Low Pass, or Band Pass filter. In Band Pass mode, the filter uses one of the poles for low pass and the other pole for high pass, with matching cutoff frequencies.

key track - Adjusts the Key Tracking of the filter, relative to its cutoff frequency. At a setting of 0%, the cutoff of the filter will be the same, regardless of the note number that triggered the voice. At 100%, the cutoff frequency will double with each octave on the keyboard. To reverse the cutoff to keyboard relationship, you can set key track to a negative value. Key Tracking is centered around C3.

tune note - Allows you to set the cutoff frequency based on a musical note. This parameter works in conjunction with the tune cents parameter, and setting either will override and update the cutoff(s) parameter.

tune cents - Adjusts the tuning of the cutoff frequency in musical terms. This parameter works in conjunction with the tune note parameter, and setting either will override and update the cutoff(s) parameter.

2 pole Additional Modulation sources:

tune note - Provides a quantized modulation output based on the current filter cutoff. The quantization is an absolute note value.

tune octave - Provides a quantized modulation output based on the current filter cutoff. The quantization is in octaves, but can be offset from true octave tuning.

tune semi - Provides a quantized modulation output based on the current filter cutoff. The quantization is in semitones, but can be offset from true semitone tuning.

tune cents - Provides a modulation output based on the current filter cutoff in cents.

2 pole Additional Modulation destinations:

cutoff(f) - Allows non-scaled direct modulation of frequency.

tune octave - Allows you to modulate the cutoff frequency based on musical octaves in

relation to the current cutoff setting.

tune semi - Allows you to modulate the cutoff frequency based on musical semitones in relation to the current cutoff setting.

1 pole va

Another 1 Pole Filter capable of Hi Pass or Low Pass operation. This filter has a more “virtual analog” character than the other 1 pole filter.

Parameters

in level - Sets the input level to the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

cutoff(s) - This is the filter cutoff frequency scaled with a natural feeling curve. Setting this parameter overrides and updates the Tune Note and Tune Cents parameters.

mode - Selects whether the filter operates as a Hi Pass or Low Pass filter.

key track - Adjusts the Key Tracking of the filter, relative to its cutoff frequency. At a setting of 0%, the cutoff of the filter will be the same, regardless of the note number that triggered the voice. At 100%, the cutoff frequency will double with each octave on the keyboard. To reverse the cutoff to keyboard relationship, you can set key track to a negative value. Key Tracking is centered around C3.

tune note - Allows you to set the cutoff frequency based on a musical note. This parameter works in conjunction with the tune cents parameter, and setting either will override and update the cutoff(s) parameter.

tune cents - Adjusts the tuning of the cutoff frequency in musical terms. This parameter works in conjunction with the tune note parameter, and setting either will override and update the cutoff(s) parameter.

1 pole va Additional Modulation sources:

tune note - Provides a quantized modulation output based on the current filter cutoff. The quantization is an absolute note value.

tune octave - Provides a quantized modulation output based on the current filter cutoff. The quantization is in octaves, but can be offset from true octave tuning.

tune semi - Provides a quantized modulation output based on the current filter cutoff. The quantization is in semitones, but can be offset from true semitone tuning.

tune cents - Provides a modulation output based on the current filter cutoff in cents.

1 pole va Additional Modulation destinations:

cutoff(f) - Allows non-scaled direct modulation of frequency.

tune octave - Allows you to modulate the cutoff frequency based on musical octaves in relation to the current cutoff setting.

tune semi - Allows you to modulate the cutoff frequency based on musical semitones in relation to the current cutoff setting.

2 pole sv

A 2 Pole State Variable Filter capable of Hi Pass, Low Pass, Band Pass, and Notch, as well as a simultaneous mix of all outputs.

Parameters

in level - Sets the input level to the filter, from -100% to 100%. Negative values will

invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

cutoff(s) - This is the filter cutoff frequency scaled with a natural feeling curve. Setting this parameter overrides and updates the Tune Note and Tune Cents parameters.

resonance - Controls the level of emphasis of the signal at the cutoff frequency.

key track - Adjusts the Key Tracking of the filter, relative to its cutoff frequency. At a setting of 0%, the cutoff of the filter will be the same, regardless of the note number that triggered the voice. At 100%, the cutoff frequency will double with each octave on the keyboard. To reverse the cutoff to keyboard relationship, you can set key track to a negative value. Key Tracking is centered around C3.

tune note - Allows you to set the cutoff frequency based on a musical note. This parameter works in conjunction with the tune cents parameter, and setting either will override and update the cutoff(s) parameter.

tune cents - Adjusts the tuning of the cutoff frequency in musical terms. This parameter works in conjunction with the tune note parameter, and setting either will override and update the cutoff(s) parameter.

mode - Selects whether the filter operates as a Hi Pass, Low Pass, Band Pass, Notch, a Mix of all the filters, or as a Morph between the filters. When mixed is selected, the separate filter mix level parameters will be enabled. When morph-LBH is selected, the morph knob is enabled and the filters crossfade between Low Pass, Band Pass and Hi Pass modes. When morph-LNH is selected, the morph knob is enabled and the filters crossfade between Low Pass, Notch and Hi Pass modes.

morph - Enabled when Mode is set to morph-LBH or morph-LNH. Crossfades between filter modes Low Pass, Band Pass and Hi Pass when morph-LBH is selected, and Low Pass, Notch and Hi Pass when morph-LNH is selected.

mix lp - Enabled when Mode is set to Mix. Adjusts the amount of the Low Pass output

from the filter.

mix hp - Enabled when Mode is set to Mix. Adjusts the amount of the Hi Pass output from the filter.

mix bp - Enabled when Mode is set to Mix. Adjusts the amount of the Band Pass output from the filter.

mix notch - Enabled when Mode is set to Mix. Adjusts the amount of the Notch output from the filter.

res2level - The Resonance-to-Level parameter can be used to decrease the Filter's operating level when Resonance is increased. Higher Res2Level values will intensify this effect. This is a useful parameter in cases where you are modulating resonance and want to keep the filter under control during large and/or fast changes during resonance modulation.

over-samp - This causes the filter to iterate during its calculation, which results in a bit more smoothness. Higher settings increase the range of the cutoff frequency, and the Cutoff parameter range will update when the oversampling is changed, in order to reflect the current possible range. The tradeoff is that higher oversampling values will require more CPU.

freq calc - Experimentation option with different formulae to calculate the cutoff frequency. Generally, it is best to use the most efficient calculation that will achieve the sound you are looking for, as the more efficient calculations will use less CPU when the filter cutoff is being modulated. Mode settings:

simple - A faster and simpler calculation, however slightly less accurate.

ideal - A slightly less simple calculation, with a bit more accuracy.

2 pole sv Additional Modulation sources:

tune note - Provides a quantized modulation output based on the current filter cutoff. The quantization is an absolute note value.

tune octave - Provides a quantized modulation output based on the current filter cutoff. The quantization is in octaves, but can be offset from true octave tuning.

tune semi - Provides a quantized modulation output based on the current filter cutoff.

The quantization is in semitones, but can be offset from true semitone tuning.

tune cents - Provides a modulation output based on the current filter cutoff in cents.

2 pole sv Additional Modulation destinations:

cutoff(f) - Allows non-scaled direct modulation of frequency.

tune octave - Allows you to modulate the cutoff frequency based on musical octaves in relation to the current cutoff setting.

tune semi - Allows you to modulate the cutoff frequency based on musical semitones in relation to the current cutoff setting.

bi-quad

The bi-quad filter is a standard 2 pole recursive filter, typically used for (but not restricted to) building EQs.

Parameters

in level - Sets the input level to the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

cutoff(s) - This is the filter cutoff frequency scaled with a natural feeling curve. Setting this parameter overrides and updates the Tune Note and Tune Cents parameters.

key track - Adjusts the Key Tracking of the filter, relative to its cutoff frequency. At a

setting of 0%, the cutoff of the filter will be the same, regardless of the note number that triggered the voice. At 100%, the cutoff frequency will double with each octave on the keyboard. To reverse the cutoff to keyboard relationship, you can set key track to a negative value. Key Tracking is centered around C3.

tune note - Allows you to set the cutoff frequency based on a musical note. This parameter works in conjunction with the tune cents parameter, and setting either will override and update the cutoff(s) parameter.

tune cents - Adjusts the tuning of the cutoff frequency in musical terms. This parameter works in conjunction with the tune note parameter, and setting either will override and update the cutoff(s) parameter.

mode - Selects whether the bi-quad operates as a Hi Pass, Low Pass, Band Pass, Notch, Peaking, Low Shelf, or Hi Shelf filter.

q - The Q Factor sets the sharpness of the cutoff frequency slope, and it is similar to Resonance in other filters. The Q parameter is not available when the Shelf modes are selected, and this state is denoted by the appearance of an (n) after the parameter name.

peak gain - Adjusts the gain or attenuation at the cutoff frequency. Peak Gain is only enabled for the Peak and Shelf modes. Peak gain's disabled state is denoted by the appearance of an (n) after the parameter name.

bi-quad Additional Modulation sources:

tune note - Provides a quantized modulation output based on the current filter cutoff. The quantization is an absolute note value.

tune octave - Provides a quantized modulation output based on the current filter cutoff. The quantization is in octaves, but can be offset from true octave tuning.

tune semi - Provides a quantized modulation output based on the current filter cutoff. The quantization is in semitones, but can be offset from true semitone tuning.

tune cents - Provides a modulation output based on the current filter cutoff in cents.

bi-quad Additional Modulation destinations:

cutoff(f) - Allows non-scaled direct modulation of frequency.

tune octave - Allows you to modulate the cutoff frequency based on musical octaves in relation to the current cutoff setting.

tune semi - Allows you to modulate the cutoff frequency based on musical semitones in relation to the current cutoff setting.

diode ladder

A classic Diode Ladder filter implementation capable of Low Pass operation.

Parameters

in level - Sets the input level to the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

cutoff(s) - This is the filter cutoff frequency scaled with a natural feeling curve. Setting this parameter overrides and updates the Tune Note and Tune Cents parameters.

resonance - Controls the level of emphasis of the signal at the cutoff frequency.

gain comp -

key track - Adjusts the Key Tracking of the filter, relative to its cutoff frequency. At a setting of 0%, the cutoff of the filter will be the same, regardless of the note number that triggered the voice. At 100%, the cutoff frequency will double with each octave on the keyboard. To reverse the cutoff to keyboard relationship, you can set key track to a negative value. Key Tracking is centered around C3.

tune note - Allows you to set the cutoff frequency based on a musical note. This parameter works in conjunction with the tune cents parameter, and setting either will override and update the cutoff(s) parameter.

tune cents - Adjusts the tuning of the cutoff frequency in musical terms. This parameter works in conjunction with the tune note parameter, and setting either will override and update the cutoff(s) parameter.

diode ladder Additional Modulation destinations:

cutoff(f) - Allows non-scaled direct modulation of frequency.

tune octave - Allows you to modulate the cutoff frequency based on musical octaves in relation to the current cutoff setting.

tune semi - Allows you to modulate the cutoff frequency based on musical semitones in relation to the current cutoff setting.

sk35 filter

An emulation of a famous Sallen-Key filter implementation, known for its character when configured as Low Pass and Hi Pass filter modes in series.

Parameters

in level - Sets the input level to the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

mode - Selects whether the filter operates as a Hi Pass or Low Pass filter.

cutoff(s) - This is the filter cutoff frequency scaled with a natural feeling curve. Setting this parameter overrides and updates the Tune Note and Tune Cents parameters.

resonance - Controls the level of emphasis of the signal at the cutoff frequency.

key track - Adjusts the Key Tracking of the filter, relative to its cutoff frequency. At a setting of 0%, the cutoff of the filter will be the same, regardless of the note number that triggered the voice. At 100%, the cutoff frequency will double with each octave on the keyboard. To reverse the cutoff to keyboard relationship, you can set key track to a negative value. Key Tracking is centered around C3.

tune note - Allows you to set the cutoff frequency based on a musical note. This parameter works in conjunction with the tune cents parameter, and setting either will override and update the cutoff(s) parameter.

tune cents - Adjusts the tuning of the cutoff frequency in musical terms. This parameter works in conjunction with the tune note parameter, and setting either will override and update the cutoff(s) parameter.

sk35 filter Additional Modulation destinations:

cutoff(f) - Allows non-scaled direct modulation of frequency.

tune octave - Allows you to modulate the cutoff frequency based on musical octaves in relation to the current cutoff setting.

tune semi - Allows you to modulate the cutoff frequency based on musical semitones in relation to the current cutoff setting.

x pole ladder

A classic Ladder Filter capable of Hi Pass and Low Pass with a variable number of Poles, between 1 and 32.

Parameters

in level - Sets the input level to the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

cutoff(s) - This is the filter cutoff frequency scaled with a natural feeling curve. Setting this parameter overrides and updates the Tune Note and Tune Cents parameters.

resonance - Controls the level of emphasis of the signal at the cutoff frequency.

key track - Adjusts the Key Tracking of the filter, relative to its cutoff frequency. At a setting of 0%, the cutoff of the filter will be the same, regardless of the note number that triggered the voice. At 100%, the cutoff frequency will double with each octave on the keyboard. To reverse the cutoff to keyboard relationship, you can set key track to a negative value. Key Tracking is centered around C3.

tune note - Allows you to set the cutoff frequency based on a musical note. This parameter works in conjunction with the tune cents parameter, and setting either will override and update the cutoff(s) parameter.

tune cents - Adjusts the tuning of the cutoff frequency in musical terms. This parameter works in conjunction with the tune note parameter, and setting either will override and update the cutoff(s) parameter.

poles - Sets the number of poles for the filter. The default is 4, for the classic 24dB ladder sound. A higher number of Poles requires higher CPU usage. The very high numbers (e.g. above 10) are primarily here for experimentation, but if you happen to find a sound that can only be achieved with these higher poles... well, great! Note that resonance can get out of control pretty quickly with a high number of poles, and the Res Scale parameter can help keep the filter under control.

mode - Selects whether the filter operates as a Hi Pass or Low Pass.

freq calc - An experimentation option with different formulae to calculate the cutoff frequency. Generally, it is best to use the most efficient calculation that will achieve the sound you are looking for, as the more efficient calculations will use less CPU when the filter cutoff is being modulated. Mode settings:

empirical - The fastest and simplest calculation, however slightly less accurate. It becomes less accurate as number of poles increases above 4.

sinepi - A slightly less simple calculation, with a bit more accuracy. It becomes less accurate as number of poles increases above 4.

mokwarp - A reasonably efficient calculation that keeps the cutoff mostly accurate at all pole number settings.

mokwhop - A bit more complex calculation that keeps the cutoff accurate at all pole number settings. This is the most accurate, but also the most CPU intensive formula.

res scale - Scales the effective amount of resonance when adjusting the Resonance parameter. Resonance behaves differently depending on the number of poles, and this parameter can help you shape the Resonance so that it behaves properly, particularly when Resonance will be modulated.

x pole ladder Additional Modulation sources:

tune note - Provides a quantized modulation output based on the current filter cutoff. The quantization is an absolute note value.

tune octave - Provides a quantized modulation output based on the current filter cutoff. The quantization is in octaves, but can be offset from true octave tuning.

tune semi - Provides a quantized modulation output based on the current filter cutoff. The quantization is in semitones, but can be offset from true semitone tuning.

tune cents - Provides a modulation output based on the current filter cutoff in cents.

x pole ladder Additional Modulation destinations:

cutoff(f) - Allows non-scaled direct modulation of frequency.

tune octave - Allows you to modulate the cutoff frequency based on musical octaves in relation to the current cutoff setting.

tune semi - Allows you to modulate the cutoff frequency based on musical semitones in relation to the current cutoff setting.

x pole sv

A state Variable Filter capable of Hi Pass, Lo Pass, Band Pass, and Notch with a variable number of Poles between 2 and 32.

Parameters

in level - Sets the input level to the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

cutoff(s) - This is the filter cutoff frequency scaled with a natural feeling curve. Setting this parameter overrides and updates the Tune Note and Tune Cents parameters.

resonance - Controls the level of emphasis of the signal at the cutoff frequency.

key track - Adjusts the Key Tracking of the filter, relative to its cutoff frequency. At a setting of 0%, the cutoff of the filter will be the same, regardless of the note number that triggered the voice. At 100%, the cutoff frequency will double with each octave on the keyboard. To reverse the cutoff to keyboard relationship, you can set key track to a negative value. Key Tracking is centered around C3.

tune note - Allows you to set the cutoff frequency based on a musical note. This

parameter works in conjunction with the tune cents parameter, and setting either will override and update the cutoff(s) parameter.

tune cents - Adjusts the tuning of the cutoff frequency in musical terms. This parameter works in conjunction with the tune note parameter, and setting either will override and update the cutoff(s) parameter.

mode - Selects whether the filter operates as a Low Pass, Hi Pass, Band Pass or Notch.

poles - Sets the number of pole pairs (in multiples of 2) for the filter. A higher number of Poles requires higher CPU usage. The very high numbers (e.g. above 10) are primarily here for experimentation, but if you happen to find a sound that can only be achieved with these higher poles... Woot! Note that resonance can get out of control pretty quickly with a high number of poles, and the Res2Level parameter can help keep the filter under control.

over-samp - This causes the filter to iterate during its calculation, which results in a bit more smoothness. Higher settings increase the range of the cutoff frequency, and the Cutoff parameter range will update when the oversampling is changed, in order to reflect the current possible range. The tradeoff is that higher oversampling values will require more CPU.

freq calc - Experimentation option with different formulae to calculate the cutoff frequency. Generally, it is best to use the most efficient calculation that will achieve the sound you are looking for, as the more efficient calculations will use less CPU when the filter cutoff is being modulated. Mode settings:

simple - A faster and simpler calculation, however slightly less accurate.

ideal - A slightly less simple calculation, with a bit more accuracy.

res2level - The Resonance-to-Level parameter can be used to decrease the Filter's operating level when Resonance is increased. Higher Res2Level values will intensify this effect. This is a useful parameter in cases where you are modulating resonance and want to keep the filter under control during large and/or fast changes during resonance modulation.

x pole sv Additional Modulation sources:

tune note - Provides a quantized modulation output based on the current filter cutoff. The quantization is an absolute note value.

tune octave - Provides a quantized modulation output based on the current filter cutoff. The quantization is in octaves, but can be offset from true octave tuning.

tune semi - Provides a quantized modulation output based on the current filter cutoff. The quantization is in semitones, but can be offset from true semitone tuning.

tune cents - Provides a modulation output based on the current filter cutoff in cents.

x pole sv Additional Modulation destinations:

cutoff(f) - Allows non-scaled direct modulation of frequency.

tune octave - Allows you to modulate the cutoff frequency based on musical octaves in relation to the current cutoff setting.

tune semi - Allows you to modulate the cutoff frequency based on musical semitones in relation to the current cutoff setting.

all pass

An All Pass filter passes all frequencies equally in gain, but changes the phase relationship among various frequencies. It is not typically useful on its own, but when combined with a non-linear filter, such as a distortion, it can add a lot of timbral color. All Pass filters are also used as building blocks for spatial effects like reverbs, or for phasers.

Parameters

in level - Sets the input level to the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

filt shift - This is like the Cutoff Frequency in normal filters. It adjusts the reference point the All Pass filter uses to shift the harmonics in the incoming signal.

order - Determines the number of filters in use, adjustable from 1st to 8th order filtering. The 2nd and 3rd order settings are the most effective for a smooth phase adjustment.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

bit offen

The Bit Offen is a bit manipulation filter designed to modify the bit field of a signal by turning bits off and on, both individually and in groups. Bits can be enabled or disabled for manipulation, and ranges of active bits can be defined using the "Offen" range parameters.

Parameters

in level - Sets the input level to the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the effect, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

all ena trig - This trigger will set all Bit Parameters to their "ENABLED" state.

all off trig - This trigger will set all Bit Parameters to their "OFF" state.

all on trig - This trigger will set all Bit Parameters to their "ON" state.

bit 01-24 - Sets the enable state of each bit. The Enable state determines how the bit can be manipulated by an Offen range. Settings:

OFF - This forces the bit off so it ignores Offen state commands.

ENABLED - This allows the bit to be controlled by an Offen group.

ON - This forces the bit on so it ignores Offen state commands. This is useful for maintaining important bits in the sonic field while an Offen group is modulating a range of bits.

bit offen A-D - There are FOUR Offen groups (A,B,C,D). These operate together and set up groups and ranges of bits to be turned off. The bit offen parameter determines the first bit of the Offen range that will be turned OFF. If this is set to 'none', the Offen group has no effect.

bit width A-D - Sets the number of bits to be turned OFF, starting from the Bit Offen value. For example, if Bit Offen is set to 7 and Bit Width is set to 3, then Bits 7,8, and 9 will be turned off.

bit modus A-D - The method of operation governing the Bit range. Mode settings:

normal - All bits from Bit Offen through Bit Width will be turned off.

evens - Every other bit, starting from Bit Offen and the following bit, through the Bit Width range will be turned off. For example, if Bit Offen is set to 7 and Bit Width is set to 6, then Bits 7, 8, 10, and 12 will be turned off. If Bit Offen is set to 8 and Bit Width is set to 6, then Bits 8, 9, 11, and 13 will be turned off. The Bit Offen bit is always included in the range.

odds - Every other bit, beginning from Bit Offen through Bit Width will be turned off. For example, if Bit Offen is set to 7 and Bit Width is set to 6, then Bits 7, 9, and 11 will be turned off. If Bit Offen is set to 8 and Bit Width is set to 6, then Bits 8, 10, and 12 will be turned off. The Bit Offen bit is always included in the range.

thirds - Every third bit from Bit Offen thru Bit Width will be turned off. For example, if Bit Offen is set to 7 and Bit Width is set to 8, then Bits 7, 10, and 13 will be turned off. If Bit Offen is set to 8 and Bit Width is set to 13, then Bits 8, 11, 14, 17, and 20 will be turned off. The Bit Offen bit is always included in the range.

fourths - Every fourth bit from Bit Offen through Bit Width will be turned off. For example, if Bit Offen is set to 7 and Bit Width is set to 8, then Bits 7 and 11 will be turned off. If Bit Offen is set to 8 and Bit Width is set to 13, then Bits 8, 12, 16, and 20 will be turned off. The Bit Offen bit is always included in the range.

bit invrt A-D - Inverts the field of bits to be turned off. For example, if Bit Offen, Width, and Modus created a field of Bits like so: 8, 9, 10, 11, 12, 13, 14, 15, and 16, and then Bit Invert is turned on ('Inverted'), the field of bits that turn off becomes 1, 2, 3, 4, 5, 6, 7, 17, 18, 19, 20, 21, 22, 23, and 24.

comb

A filter implemented by adding a delayed version of a signal to itself, causing cancellations and accentuations at different frequencies, normally in a series of regularly spaced notches.

Parameters

in level - Sets the input level to the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the filter, from -100% to 100%.

mode - Chooses the type of comb filter. Mode settings:

Juju - Standard Feedforward Comb that is commonly used in Reverb algorithms. This is the basic one.

Floi - A variation on Juju that gives a different effect when modulating the delay line. Also, the damping will have a slightly different effect.

Wifwd - A textbook feed-forward Comb Filter.

Wirev - A textbook feedback Comb Filter.

damping - This allows you to control the feedback (or feed forward in most cases) of the comb. It keeps the comb under control, or out of control if you like.

feedback - Adjusts the level of signal being added back into the delay line, and can be positive or negative in polarity.

delay time - Sets the delay time in milliseconds which affects the cancellations of the comb filter.

clip thresh - Sets a threshold level above which the signal is clipped.

key track - Adjusts the Key Tracking of the filter, relative to its delay time. At a setting of 0%, the center frequency will be the same, regardless of the note number that triggered the voice. At 100%, the center frequency will double with each octave on the keyboard. To reverse the frequency to keyboard relationship, you can set key track to a negative value. Key Tracking is centered around C3.

comb lpf

This is a slight variation on the normal Comb Filter described previously. The main difference is the addition of a Low Pass Filter (LPF) embedded in the feedback path of the algorithm. This type of comb filter is commonly used in comb filtered Reverb designs.

Parameters

in level - Sets the input level to the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the filter, from -100% to 100%.

lpf feedback - The amount of the delayed signal that is fed back into the circulating delay line, consisting of the feedback signal through a low pass filter, which eliminates the need for a separate damping control.

delay time - Sets the delay time in milliseconds which affects the cancellations of the comb filter.

clip thresh - Sets a threshold level above which the signal is clipped.

key track - Adjusts the Key Tracking of the filter, relative to its delay time. At a setting of 0%, the center frequency will be the same, regardless of the note number that triggered the voice. At 100%, the center frequency will double with each octave on the keyboard. To reverse the frequency to keyboard relationship, you can set key track to a negative value. Key Tracking is centered around C3.

compressor

The compressor reduces the volume of loud sounds, thus compressing the dynamic range of the audio signal.

Parameters

in level - Sets the input level to the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the effect, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

mode - Governs the behavior of the effect. Mode settings:

peak comp - Responds to the instantaneous level of the input signal. Results in

a more evident compression and sometimes even distortion.

rms comp - Performs root-mean-square averaging on the input signal before comparing the level to the threshold. Results in a more natural sounding dynamic control.

peak limit - Signals below the threshold pass unaffected while peaks that exceed the threshold are attenuated.

threshold - Sets the level at which the effect is engaged.

ratio - Sets the amount of gain reduction. A ratio of 2:1 means that if input level is 2 dB over the threshold, the output signal level is reduced to 1 dB over the threshold.

attack - Sets the amount of time it takes to reach the full compression ratio after the signal goes over the threshold.

release - Sets the amount of time it takes to return to zero compression after the signal falls below the threshold.

rms window - Set the amount of time that is averaged when comparing the input signal to the threshold.

neg.ratio - An experiment with ratios that make no sense. Nasty.

Compressor Additional Modulation sources:

comp level - Modulation output from the overall compression envelope.

env level A - Modulation output from the stereo left compression envelope. Only active within the stereo filter 3 block.

env level B - Modulation output from the stereo right compression envelope. Only active within the stereo filter 3 block.

dc blocker

A fixed structure high pass filter that removes DC signals.

Parameters

in level - Sets the input level to the dc blocker effect, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

block val - Sets the cutoff frequency of the high pass filter.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the dc blocker effect, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

decimator

Performs sample rate and bit reduction.

Parameters

in level - Sets the input level to the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the effect, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

sample rate - Reduces sample rate in a range from 100% down to 1%.

sr keytrack - Adjusts the Key Tracking of the sample rate, relative to the Sample Rate

parameter. At a setting of 0%, the sample rate stays the same, regardless of the note number that triggered the voice. At 100%, the sample rate perfectly tracks note frequency. To reverse the sample rate to keyboard relationship, you can set key track to a negative value. Key Tracking is centered around C3.

hi pass bits - Reduces bits from the bottom up, starting at bit 1 and ending at bit 24.

lo pass bits - Reduces bits from the top down, starting at bit 24 and ending at bit 1.

env follow

The envelope follower takes a high-frequency signal input and generates a modulation output, which is the amplitude envelope of the original signal.

Parameters

in level - Sets the input level to the dc blocker effect, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

attack - Used to smooth the rise time of the follower.

atk trig - Sets a rising input level threshold for starting the gate modulation output.

release - Used to smooth the fall time of the follower.

rel trig - Sets a falling input level threshold for ending the gate modulation output.

env follow Additional Modulation sources:

env levelA - Modulation output from the stereo left envelope follower, or from the overall envelope if used in a mono filter block.

env levelB - Modulation output from the stereo right envelope follower. Only active within the stereo filter 3 block.

trig out A - Gate output from the stereo left envelope follower, or from the overall envelope if used in a mono filter block.

trig out B - Gate output from the stereo right envelope follower. Only active within the stereo filter 3 block.

gain

Provides amplification and attenuation with options for waveshaping.

Parameters

in level - Sets the input level to the gain effect, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

out pan (only available in the FLT3 filter block) - Pans the output of the Gain module between the left and right channels of the stereo voice buss.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the gain effect, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

clip mode - Defines how the signal will be clipped. Modes:

chop - Any signal above the threshold is simply chopped off.

wrap - Any signal above the threshold wraps back and starts at zero again.

fold - Folds to center and back to max.

foldZ - Folds toward zero. Once the fold reaches zero, it stays there.

zero - Any value above the clip threshold just becomes zero level.

clip thresh - Sets a threshold level above which the signal is clipped.

resonator

The Resonator is a Filter that is designed only to Resonate at a specific frequency. Resonators have been used to create room and box tones to simulate standing waves, as well as for pipe and tube emulation for physical modeling.

Parameters

in level - Sets the input level to the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the effect, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

cutoff(s) - This is the filter cutoff frequency scaled with a natural feeling curve. Setting this parameter overrides and updates the Tune Note and Tune Cents parameters.

damping - This controls the amount of Resonance to apply to the Filter. Positive values are normally used to increase the resonance until the filter hits oscillation around 100%. Negative values are purely experimental, and once in a while might yield a surprisingly unexpected chorus of noisy weirdness.

damp fine - Fine adjustment of the Damping amount.

key track - Adjusts the Key Tracking of the filter, relative to its cutoff frequency. At a setting of 0%, the cutoff of the filter will be the same, regardless of the note number that triggered the voice. At 100%, the cutoff frequency will double with each octave on the keyboard. To reverse the cutoff to keyboard relationship, you can set key track to a negative value. Key Tracking is centered around C3.

Resonator Additional Modulation Sources:

tune note - Allows you to set the cutoff frequency based on a musical note. This parameter works in conjunction with the tune cents parameter, and setting either will override and update the cutoff(s) parameter.

tune cents - Adjusts the tuning of the cutoff frequency in musical terms. This parameter works in conjunction with the tune note parameter, and setting either will override and update the cutoff(s) parameter.

Resonator Additional Modulation Destinations:

tune note - Provides a quantized modulation output based on the current filter cutoff. The quantization is an absolute note value.

tune octave - Provides a quantized modulation output based on the current filter cutoff. The quantization is in octaves, but can be offset from true octave tuning.

tune semi - Provides a quantized modulation output based on the current filter cutoff. The quantization is in semitones, but can be offset from true semitone tuning.

tune cents - Provides a modulation output based on the current filter cutoff in cents.

ring math

Ring Math is based on the idea of Ring Mod with a number of additional modes for mathematically shaping the character of the Ring Mod effect.

Parameters

in level - Sets the input level to the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

out boost - Provides additional gain or attenuation to the output level.

clip mode - Selects how the signal will clip when it exceeds the clipping threshold.

Clip - The signal will hard clip when it reaches the clipping threshold.

Wrap - When the signal reaches the clipping threshold, it will wrap around to the opposite polarity and continue.

clip thresh - Sets a dB threshold where clipping will begin.

out level - Sets the output level from the effect, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

dry level - This is the mix amount of dry, unprocessed signal from the input. Dry level is bi-polar and can be useful for adding the fundamental signal back to the processed Ring Mod signal, or negatively added back to the signal to phase cancel parts of the output signal.

wet level - This is the mix amount of the processed signal.

osc level - The Ring Mod effect uses an oscillator to modulate the incoming signal. This level adjustment allows you to listen to the direct output of the oscillator. Useful if you want to mix the modulating signal, either directly (positively values), or for phase cancellation (negative values), into the final output.

math level - A special control that tailors the amount of signal applied in each math mode, typically this parameter scales the amount of modulating oscillator applied to the source signal. However, in the mag math modes, this parameter is used to favor the source signal in the value comparisons as math level is decreased.

math mode - Traditional Ring Mod is a multiplication between the modulating oscillator and the source signal. In addition to multiplication, a number of other modes of combining the *oscillator* and *source* signal are possible.

None - disables the combination signal

Multiply - multiplies the source signal and the modulating oscillator. This is traditional ring modulation.

Mult ABS - Multiplies the absolute value of the oscillator with the bipolar source signal. The result always maintains the same polarity as the source, as there will never be two negative values creating polarity changes.

ABS Mult - Multiplies the absolute value of the source signal with the bipolar oscillator. The result always maintains the same polarity as the modulating oscillator, as there will never be two negative values creating polarity changes.

Mag MAX - Whichever signal, the source signal or the modulating oscillator, has the greatest absolute value will be selected as the result signal. In this case, the math level parameter will scale the modulating oscillator.

Mag MIN - Whichever signal, the source signal or the modulating oscillator, has the minimum absolute value will be selected as the result signal. In this case, the math level parameter will scale the modulating oscillator.

Mag2 MAX - A variation of mag max where only the positive values of each signal are used for comparison. The math level parameter is used to blend the original signal back into the output.

Mag2 MIN - A variation of mag min where only the positive values of each signal are used for comparison. The math level parameter is used to blend the original signal back into the output, which also favors the original source signal.

Diff Modes (Diff 1, Diff 2, Diff 3, Diff DYNA, Diff ABS, ABS Diff) : These modes use polarity of the source signal to create addition and subtraction combinations of the source and modulating oscillator.

Sum Modes (EXCL[+] / [-], DYNA, AND, NAND, OR[+] / [-], BOOL[m] / [+] / [-]) : These modes use polarity of the source signal to create addition combinations of the source and modulating oscillator.

Mult Modes (EXCL[+] / [-], DYNA, AND, NAND, OR[+] / [-], BOOL[m] / [+] / [-]) : These modes use polarity of the source signal to create multiplication combinations of the source and modulating oscillator.

Gate A Modes (Gate A, AND, NAND, OR, BOOL) : These modes use the modulating oscillator to gate the source signal. The processed output is derived solely from the gated source signal.

Gate B Modes (Gate B, AND, NAND, OR, BOOL) : These modes use the source signal to gate the modulation oscillator. The processed output is derived solely from the gated oscillator.

dc blocker - Enables a DC Blocking filter. Some of the math modes in combination with shapes of the modulating oscillator can generate DC in the resulting signal. The DC Blocker filters out that DC if you do not intend to pass that DC along to the next stage. Modulate this matrix destination high in order to activate the DC Blocker.

SOURCE OSCILLATOR PARAMETERS:

The modulating oscillator's parameter set is based on the general parameters of the Waverazor oscillator's individual wave segment. The descriptions of these parameters can be found in the Oscillator Wave Segment section of this manual.

saturator

A basic Saturation stage with adjustable dB rails and width.

Parameters

in level - Sets the input level to the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the effect, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

drive - Controls the amount of additional Gain to drive the source signal into the rails.

rail - Similar to analog voltage rails, this control sets the absolute maximum dB clipping point.

width - Sets a range of saturation, starting below the clipping level and increasing the saturation until it hard clips at the rail. A setting of 0% will put the saturation and clipping points at the same place, effectively just giving you hard clipping.

shaper

A waveshaper that changes audio by applying a mathematical function in order to create distortion.

Parameters

in level - Sets the input level to the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the effect, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

shape sel - Selects a shape for the waveshaper transfer function, which distorts the audio signal.

shape neg - Selects a shape for the waveshaper transfer function in the negative portion of the wave only. This is used for asymmetrical distortion.

clip mode - Defines how the signal will be clipped. Modes:

clip - Any signal above the threshold is clipped.

wrap - Any signal above the threshold wraps back and starts at zero again.

tap line 1

A single delay tap.

Parameters

in level - Sets the input level to the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the effect, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

dry level - The level of the dry input signal mixed in with the tap line delay effect.

clip thresh - Sets a threshold level above which the signal is clipped.

feed rate - Controls the amount of sample rate reduction on the delay line.

feed keytrack - Adjusts the Key Tracking of the feed rate, relative to the feed rate parameter. At a setting of 0%, the feed rate stays the same, regardless of the note number that triggered the voice. At 100%, the feed rate perfectly tracks note frequency. To reverse the feed rate to keyboard relationship, you can set key track to a negative value. Key Tracking is centered around C3.

tap 1 time - Sets the delay time in milliseconds, up to 10.24 mSec.

tap 1 level - Sets the mix level of the tap 1 delay.

tap 1 feedback - Sets the feedback amount for tap 1.

tap 1 keytrack - Adjusts the Key Tracking of the delay time, relative to the Tap 1 Time parameter. At a setting of 0%, the delay time stays the same, regardless of the note

number that triggered the voice. At 100%, the delay time perfectly tracks note frequency. To reverse the delay time to keyboard relationship, you can set key track to a negative value. Key Tracking is centered around C3.

tap line 5

Five delay taps.

Parameters

in level - Sets the input level to the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the effect, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

dry level - The level of the dry input signal mixed in with the tap line delay effect.

clip thresh - Sets a threshold level above which the signal is clipped.

num taps - Sets the number of delay taps between 0 and 5.

feed rate - Controls the amount of sample rate reduction on the delay line.

feed keytrack - Adjusts the Key Tracking of the feed rate, relative to the feed rate parameter. At a setting of 0%, the feed rate stays the same, regardless of the note number that triggered the voice. At 100%, the feed rate perfectly tracks note frequency. To reverse the feed rate to keyboard relationship, you can set key track to a negative value. Key Tracking is centered around C3.

tap 1-5 time - Sets the delay time in milliseconds, up to 10.24 mSec.

tap 1-5 level - Sets the mix level of the tap delay.

tap 1-5 feedback - Sets the feedback amount for the tap.

tap 1-5 keytrack - Adjusts the Key Tracking of the delay time, relative to the Tap 1-5 Time parameter. At a setting of 0%, the delay time stays the same, regardless of the note number that triggered the voice. At 100%, the delay time perfectly tracks note frequency. To reverse the feed rate to keyboard relationship, you can set key track to a negative value. Key Tracking is centered around C3.

tap line 8

Eight delay taps.

Parameters

in level - Sets the input level to the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

dry level - The level of the dry input signal mixed in with the tap line delay effect.

num taps - Sets the number of delay taps between 0 and 8.

clip thresh - Sets a threshold level above which the signal is clipped.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the effect, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

feed rate - Controls the amount of sample rate reduction on the delay line.

feed keytrack - Adjusts the Key Tracking of the feed rate, relative to the feed rate parameter. At a setting of 0%, the feed rate stays the same, regardless of the note number that triggered the voice. At 100%, the feed rate perfectly tracks note frequency. To reverse the feed rate to keyboard relationship, you can set the key track

to a negative value. Key Tracking is centered around C3.

taps keytrk - Adjusts the Key Tracking of all tap times, relative to their individual tap time settings. At a setting of 0%, the original tap times stay the same, regardless of the note number that triggered the voice. At 100%, the relative tap times perfectly track note frequency. To reverse the tap tracking to keyboard relationship, you can set this parameter to a negative value. Key Tracking is centered around C3. Taps keytrk only affects taps that have their key trk parameter set to On.

time scale - Scales all tap times at once. The default setting is 100% which produces no change, while the maximum setting of 200% doubles the tap times from their current value. The minimum setting of 001% scales the tap times to 1% of their original value. Time scale will only affect taps that have their time trk parameter set to On.

time spread - Increases all of the delay taps simultaneously and in a relative way, spreading Tap 1 the furthest and Tap 8 the least. Time spread will only affect taps that have their time trk parameter set to On.

fdbk scale - Scales all feedback times at once. The default setting is 100% which produces no change, while the maximum setting of 200% doubles the tap feedback from their current value. The minimum setting of 001% scales the tap times to 1% of their original value. Fdbk scale will only affect taps that have their fdbk trk parameter set to On.

tap smooth - Slows down the tap time adjustments. It is useful if you are modulating the tap times, or using Time Spread or Time Scale, and you want to modulate smoothly (lag) to the new time value.

tap 1-8 level - Sets the mix level of the tap delay.

tap 1-8 time - Sets the delay time in milliseconds, up to 10 mSec.

timebase - Changes the delay time setting between millisecond and tuned ratio modes. The tuned ratios are Octaves : Semitones : Cents. These are helpful if you are using the taps in relative tuning modes with key tracking.

tap 1-8 feedback - Sets the feedback amount for the tap.

tap 1-8 key trk - When engaged, defeats the effect of the taps keytrk parameter.

tap 1-8 time trk - When engaged, defeats the effect of the time scale and time spread parameters.

tap 1-8 fdbk trk - When engaged, defeats the effect of the fdbk scale parameter.

wavefolder

The wavefolder is a 16-stage classic folding algorithm, but with a few interesting additions.

Parameters

in level - Sets the input level to the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the effect, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

fold mode - Determines how the input signal is folded when it reaches the fold point (+ or - maximum value for the signal).

bi-fold - This is the typical fold back mode. When the signal reaches the fold point, it will bounce back in the opposite direction until it reaches the next maximum fold point, at which time it will reverse direction again.

polar-fold - This is the same as bi-fold, except the signal is only folded within its original polarity. For example, as a signal exceeds the positive maximum it will bounce back in order to begin a downward fold, however it will only go as far as zero (mid-point), before folding back again in a positive direction. This creates wavefolding that maintains the original fundamental orientation of the source signal.

bi-flip - Instead of folding the signal as it reaches the fold points, the signal will pass thru to the opposite polarity. This is similar to wrapping. For example, as

a signal reaches 1.0 (maximum), it will be wrapped to -1.0 and continue its upward trajectory.

polar-flip - This is the same as Bi-Flip, except the signal will be wrapped to the midpoint, giving a similar effect, but maintaining the fundamental orientation of the original source signal.

fold mode- - Allows for setting a separate folding mode for the negative polarities. When set to "---", the positive Fold Mode parameter determines the mode for both positive and negative folding. When this is set to something else, the positive and negative polarities of the source signal will each have their own folding modes.

drive - Adds gain to the input signal to create more folding.

bias - Adds DC Offset (positive or negative) to the Drive signal so the folding can be pushed more into one polarity or the other.

drive+ gain - Allows for a separate gain amount to be applied to the positive side of the source signal.

drive- gain - Allows for a separate gain amount to be applied to the negative side of the source signal.

fold+ gain - Allows for a decrease in gain after folding has occurred, and is applied only to the positive side of the source signal.

fold- gain - Allows for a decrease in gain after folding has occurred, and is applied only to the negative side of the source signal.

stage (1,2,3,4) gain - Four stage gain controls adjust the amount of gain or reduction of the signal after each folding stage. This allows for removing gain from one stage or another. For example, it is possible to use a large drive amount to get rapid folding, but simultaneously reduce gain in Stage 1 in order to prevent subsequent rapid folding. Stage gain can also be used to increase folding only in later stages. Note that the wavefolder is 16 stages, and the Stage 4 Gain control sets the level for Stages 4 through 16.

filter split

A filter configuration sub-structure that can be loaded into a main filter block to create more complex filter routings. In filter blocks 1 and 2, when a filter split is loaded, the mono signal is split into two channels for separate processing and then summed back together for output. In filter block 3, the left and right channels are processed separately and remain independent stereo signals on output.

To populate the split, click in the top slot (A) to see a drop down menu of filter choices, or use the module drop down menu to add a filter to the highlighted slot. Once the slot A filter is added, a filter for slot B can be selected. When both slots are populated, you can highlight either filter for editing by clicking on it, or go to the module drop down menu to remove either filter. If you remove the filter from slot A, the slot B filter will move up into slot A's place.

filter block

A filter configuration sub-structure that can be loaded into a main filter block to create more complex filter routings. The Filter Block configuration contains 3 slots that can be routed in several ways.

To populate the sub filter block, click in any slot to see a drop down menu of filter choices. When any of the slots are populated, you can highlight a filter for editing by clicking on it, or go to the module drop down menu to remove the highlighted filter, or all filter slots at once.

A Filter Block has different parameters depending on which main filter block it is populating:

FLT1-2

order - Selects from the available mono configurations:

Serial - 3 filters arranged in series, and the 3 slots process the mono signal one after another.

Parallel - The mono input signal is split and processed independently over 3 separate channels before being summed back into a single output.

Pre A|B - The mono signal runs through the first filter slot, then the signal is split into 2 channels for separate processing before being summed back together at the output.

A|B Post - The mono signal is split into 2 channels for separate processing before being summed back together for the final filter slot and passed to the output.

out level - Sets the output level from the Filter Block, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

FLT2 only

flt1->flt2 - Input level from the main filter block 1 to filter block 2. By using the filter block input levels and the filter mix controls that feed the Amplitude Envelopes, the main filter blocks can be configured in series, parallel, or some combination of the two.

FLT3 only

flt1->flt3L - Input level from the main filter block 1 to the left channel of filter block 3. By using the filter block input levels and the filter mix controls that feed the Amplitude Envelopes, the main filter blocks can be configured in series, parallel, or some combination of the two.

flt1->flt3R - Input level from the main filter block 1 to the right channel of filter block 3.

flt2->flt3L - Input level from the main filter block 2 to the left channel of filter block 3.

flt2->flt3R - Input level from the main filter block 2 to the right channel of filter block 3.

order - Selects from the available stereo configurations:

Serial - 3 filters arranged in series, and the 3 slots process the signal one after another, maintaining a stereo signal path.

Parallel - The stereo input signal is split and processed independently over 3

separate stereo channels before being summed back into a single stereo output.

Pre A|B - The stereo signal runs through the first slot, and then the stereo channels are split, with the left signal going to one slot and the right signal going to the other slot for independent processing. On output, the two channels are rejoined as a stereo pair.

A|B Post - The stereo signal is split with the left signal going to one slot and the right signal going to the other slot for independent processing before being rejoined as a stereo pair for the final filter slot and passed to the output.

out level - Sets the output level from the Filter Block, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

filter bank

A filter configuration sub-structure that can be loaded into a main filter block to create more complex filter routings. The Filter Bank configuration contains 8 slots that can be routed in several ways.

To populate the sub filter block, click in any slot to see a drop down menu of filter choices. When any of the slots are populated, you can highlight a filter for editing by clicking on it, or go to the module drop down menu to remove the highlighted filter, or all filter slots at once.

A Filter Bank has different parameters depending on which main filter block it is populating:

FLT1-2

proc order - Selects from the available mono configurations:

Serial - 8 filters arranged in series, and the 8 slots process the mono signal one after another.

Parallel - The mono input signal is split and processed independently over 8

separate channels before being summed back into a single output.

A|B - The mono signal is split into 2 channels for separate processing by 4 serial filter slots on either side, and then summed back together at the output.

I-A|B-O - The input signal passes through two filter slots in series, and then splits into two channels for separate processing by two serial filter slots on either side, and then summed back together for additional processing by two filter slots in series before output.

out level - Sets the output level from the Filter Bank, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

FLT2 only

flt1->flt2 - Input level from the main filter block 1 to filter block 2. By using the filter block input levels and the filter mix controls that feed the Amplitude Envelopes, the main filter blocks can be configured in series, parallel, or some combination of the two.

FLT3 only

flt1->flt3L - Input level from the main filter block 1 to the left channel of filter block 3. By using the filter block input levels and the filter mix controls that feed the Amplitude Envelopes, the main filter blocks can be configured in series, parallel, or some combination of the two.

flt1->flt3R - Input level from the main filter block 1 to the right channel of filter block 3.

flt2->flt3L - Input level from the main filter block 2 to the left channel of filter block 3.

flt2->flt3R - Input level from the main filter block 2 to the right channel of filter block 3.

order - Selects from the available stereo configurations:

Serial - 8 filters arranged in series, and the 8 slots process the signal one after another, maintaining a stereo signal path.

Parallel - The stereo input signal is split and processed independently over 8 separate stereo channels before being summed back into a single stereo output.

A|B - The stereo signal is split into 2 channels for separate processing by 4 serial filter slots on either side, and then rejoined as a stereo pair for output.

I-A|B-O - The stereo input signal passes through 2 filter slots in series, and then splits the signal into 2 channels for separate processing by 2 serial filter slots on either side, and then rejoined as a stereo pair for additional processing by two filter slots in series before stereo output.

out level - Sets the output level from the Filter Block, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

AMP Block NAV

This navigation button brings the Voice Amp Mixer module into focus along with Waverazor's hardwired Amplitude Envelopes. These Envelopes are hardwired to amplitude, but they have versatile input level mixers, and can be routed to additional modulation destinations if desired.

The Voice Amplifier Matrix Mixer, or VAmP for short, is responsible for mixing any of the voice's source signal combinations (oscillators, filters, audio input, feedback) into any of the Amp Envelopes, and providing output bus routing for each envelope. The VAmP essentially takes a matrix of audio signal inputs and creates the main bus and effects stereo outputs under amplitude envelope control.

Focus Column

VOICE AMP tab - Displays the Voice Amp Mixer with levels, panning and polarity controls available for how the totality of the Amplitude Envelopes feed into the Effects and Main Output. The Envelope Mixers also appear here to provide access to each Amp section envelope's hardwired mixing parameters. For each instantiated envelope, you have input level controls for the oscillator, filter, audio input, and feedback levels, as well as output level controls for how the specific envelope feeds the Main and EFX 1-3 buses.

AMP ENVS tab - Shows the modules for all active Amplitude Envelopes. You can have up to 8 Amplitude Envelopes, which are always hardwired to the Amp section, but can have additional modulation routing destinations through the modulation matrix.

Input column

This is a contextual, tabbed display, the contents of which are dependent on what is currently shown in focus (Center).

VOICE AMP focus tab selected:

OMIX tab - When the OMIX tab is selected, the Input column shows a collapsed version of the Oscillator Mixer.

CTLS tab - When the CTLS tab is selected, the Input column shows the collapsed versions of any main performance page controls routed to the Voice Amp Mixer in focus.

MODS tab - When the MODS tab is selected, the Input column shows the collapsed versions of any modulation sources routed to the Voice Amp Mixer in focus.

AMP ENVS focus tab selected:

CTLS tab - When the CTLS tab is selected, the Input column shows the collapsed versions of any main performance page controls routed to any Amplitude Envelope(s) in focus.

MODS tab - When the MODS tab is selected, the Input column shows the collapsed versions of any modulation sources routed to any Amplitude Envelope(s) in focus.

Menu Button - Only active when the Amp Envs tab is selected. Provides drop down

menu selections for adding up to 8 amplitude envelopes. The selections are as follows:

Add Envelope ASR

Add Envelope ADSR

Output Column

This is a contextual, tabbed display dependent on what is currently shown in focus (Center).

VOICE AMP focus tab selected:

PREMIX tab - When the PREMIX tab is selected, the Output column shows a condensed version of the Pre FX Mixer where you can control the levels of the Audio Input and Feedback (in parallel to the Amp output), as they feed the Main Bus and Effects Buses.

AENV tab - When the AENV tab is selected, the right column shows condensed versions of any active Amplitude Envelopes.

AMP ENVS focus tab selected:

When the AMP ENVS tab is selected, the Output column shows a condensed version of the Voice Amp Mixer where you will find level controls for the Main Mix bus. Any active Amp Envelope mixers will also be displayed, providing oscillator and filter mixing for the envelope inputs.

Voice Amp Mixer Module

The Voice Amp Mixer module provides control over the mix feeding into the effects buses and main output bus. The parameters are arranged in vertical channel strips with Volume, Balance

and Polarity controls for each channel going from top to bottom. The available mix buses are:

EFX1 SND - Controls the send to Effects Bus 1.

EFX2 SND - Controls the send to Effects Bus 2.

EFX3 SND - Controls the send to Effects Bus 3.

MIX BUS - Controls the Main Mix bus in relation to the Effects Sends.

Overlap - Increasing this parameter helps to control the voice count of sounds with long release times. When more and more voices overlap each other as they are played on the same note, this setting works to fade the voices out sooner than they would otherwise from their normal release times. Explanation of settings:

OFF - No rejection of overlapping notes.

0.01% through 99.9% - This value drives a 10x ratio that reduces the Release envelope for voices that have been overlapped. It is recursive, so each time a voice is overlapped, its Release envelope is sped up by another 10x factor. For example:

- Voice 1 has a 10 second release envelope, and Overlap is set to 10%.
- Another Voice 2 is triggered from the same note. Voice 1 gets a 100% (10% x 10) decrease in its release time, so now it begins moving through its release with a 5 second duration. (How long until the amplitude envelope reaches zero depends on when the note was actually released.)
- Another Voice 3 is triggered from the same note. Voice 1 gets a 100% (10% x 10) decrease in its release time, so now it begins moving through its release with a 2.5 second duration. Simultaneously, Voice 2 is modified to a 5 second release time.

And this continues for each voice triggered on the same note, so you can see that a value of 10% is actually quite a lot, and you have a range of 0.01% through 99.9%... For most purposes we recommend using the range sparingly as the higher values can be quite aggressive.

IMMEDIATE - Overlapping voices immediately kill the previous voices.

Envelope 1-8 Mixer Modules

Each hardwired Amplitude Envelope that you add (up to 8 available) has an input and output mixer associated with it. This allows you to have detailed control over which signals are routed through the envelope and which buses the envelope feeds. The parameters are:

Input Levels

osc1 - The Oscillator 1 direct output amount mixed into this Envelope.

osc2 - The Oscillator 2 direct output amount mixed into this Envelope.

osc3 - The Oscillator 3 direct output amount mixed into this Envelope.

filt1 - The Filter 1 output amount mixed into this Envelope.

filt2 - The Filter 2 output amount mixed into this Envelope.

filt3 - The Filter 3 output amount mixed into this Envelope.

audio in - The amount of Audio Input mixed into this Envelope.

fdback - The amount of Main Feedback mixed into this Envelope.

Output Levels

EFX1 - The Envelope Output amount sent to Effects bus 1.

EFX2 - The Envelope Output amount sent to Effects bus 2.

EFX3 - The Envelope Output amount sent to Effects bus 3.

MAIN - The Envelope Output amount sent to the Main Mix bus.

Input Pan and Polarity

osc1 (Pan) - Oscillator 1 input panning to this Envelope.

osc2 (Pan) - Oscillator 2 input panning to this Envelope.

osc3-bal (Balance) - Oscillator 3 input balance to this Envelope.

filt1 (Pan) - Filter 1 input panning to this Envelope.

filt2 (Pan) - Filter 2 input panning to this Envelope.

filt3-bal (Balance) - Filter 3 input balance to this Envelope.

auin-bal (Balance) - Audio Input balance to this Envelope.

fdbk-bal (Balance) - Main Feedback balance to this Envelope.

osc1 (Polarity) - Oscillator 1 input polarity to this Envelope.

osc2 (Polarity) - Oscillator 2 input polarity to this Envelope.

filt1 (Polarity) - Filter 1 input polarity to this Envelope.

filt2 (Polarity) - Filter 2 input polarity to this Envelope.

osc3L (Pan) - Oscillator 3 Left post-balance panning to this Envelope.

osc3R (Pan) - Oscillator 3 Right post-balance panning to this Envelope.

filt3L (Pan) - Filter 3 Left post-balance panning to this Envelope.

filt3R (Pan) - Filter 3 Right post-balance panning to this Envelope.

auin-L (Pan) - Audio Input Left post-balance panning to this Envelope.

auin-R (Pan) - Audio Input Right post-balance panning to this Envelope.

fdbk-L (Pan) - Main Feedback Left post-balance panning to this Envelope.

fdbk-R (Pan) - Main Feedback Right post-balance panning to this Envelope.

osc3L (Polarity) - Oscillator 3 Left input polarity to this Envelope.

osc3R (Polarity) - Oscillator 3 Right input polarity to this Envelope.

filt3L (Polarity) - Filter 3 Left input polarity to this Envelope.

filt3R (Polarity) - Filter 3 Right input polarity to this Envelope.

auin-L (Polarity) - Audio Input Left polarity to this Envelope.

auin-R (Polarity) - Audio Input Right polarity to this Envelope.

fdbk-L (Polarity) - Main Feedback Left input polarity to this Envelope.

fdbk-R (Polarity) - Main Feedback right input polarity to this Envelope.

Output Balance and Polarity

EFX1 (Balance) - The Envelope Output stereo balance sent to Effects bus 1.

EFX2 (Balance) - The Envelope Output stereo balance sent to Effects bus 2.

EFX3 (Balance) - The Envelope Output stereo balance sent to Effects bus 3.

MAIN (Balance) - The Envelope Output stereo balance sent to the Main Mix bus.

EFX1 (Polarity) - The Envelope Output signal polarity sent to Effects bus 1.

EFX2 (Polarity) - The Envelope Output signal polarity sent to Effects bus 2.

EFX3 (Polarity) - The Envelope Output signal polarity sent to Effects bus 3.

MAIN (Polarity) - The Envelope Output signal polarity sent to the Main Mix bus.

MIX Block NAV (Pre FX Mixer)

The Pre FX Mixer is responsible for mixing the direct Audio Input and Feedback paths into the Effects buses in relation to the output of the Amp block. It's also used for mixing the Effects bus routings in series, parallel, or in some combination of both.

Focus

PREMIX tab - Displays the Pre FX Mixer with levels, panning and polarity controls for the Audio Input, Feedback and Effects routing.

Input column

This is a contextual display, and its contents are dependent on what is currently shown in focus (Center).

VAMP tab - The Input column shows a collapsed version of the Voice Amp Mixer, and the mixers for any active Amplitude Envelopes.

Output Column

This is a contextual display, and its contents are dependent on what is currently shown in focus (Center).

EFX tab - The Output column shows collapsed versions of the three Effects buses. Each module provides quick controls for effect selection and mixing.

Pre FX Mixer Module

The Pre FX Mixer is responsible for mixing the direct Audio Input and Feedback paths into the Effects buses in relation to the output of the Amp block. It's also used for mixing the Effects bus routings in series, parallel, or in a combination of both. The available mix controls are:

audio in lev - Controls the Audio Input level to the Pre FX Mixer block.

audio in bal - Controls the Audio Input balance to the Pre FX Mixer block.

audio in pol - Controls the Audio Input polarity to the Pre FX Mixer block.

auin->MIX lev - Controls the Audio Input level to the Main Mix bus.

auin->MIX bal - Controls the Audio Input balance to the Main Mix bus.

auin->EFX1 lev - Controls the Audio Input level to Effects bus 1.

auin->EFX1 bal - Controls the Audio Input balance to Effects bus 1.

auin->EFX2 lev - Controls the Audio Input level to Effects bus 2.

auin->EFX2 bal - Controls the Audio Input balance to Effects bus 2.

auin->EFX3 lev - Controls the Audio Input level to Effects bus 3.

auin->EFX3 bal - Controls the Audio Input balance to Effects bus 3.

feedback lev - Controls the Feedback level to the Pre FX Mixer block.

feedback bal - Controls the Feedback balance to the Pre FX Mixer block.

feedback pol - Controls the Feedback polarity to the Pre FX Mixer block.

fdbk->MIX lev - Controls the Feedback level to the Main Mix bus.

fdbk->MIX bal - Controls the Feedback balance to the Main Mix bus.

fdbk->EFX1 lev - Controls the Feedback level to Effects bus 1.

fdbk->EFX1 bal - Controls the Feedback balance to Effects bus 1.

fdbk->EFX2 lev - Controls the Feedback level to Effects bus 2.

fdbk->EFX2 bal - Controls the Feedback balance to Effects bus 2.

fdbk->EFX3 lev - Controls the Feedback level to Effects bus 3.

fdbk->EFX3 bal - Controls the Feedback balance to Effects bus 3.

EFX 1-3 Block NAV

These navigation buttons bring your desired effects bus (1-3) into focus. The contents of the bus can be an individual effect or a multi-effect Split or Bank configuration. All three effect sends are stereo.

Focus

EFX 1-3 tab - Displays the chosen Effect bus for editing.

Input column

This is a contextual, tabbed display, the contents of which are dependent on what is currently shown in focus (Center).

PREMIX tab - When the PREMIX tab is selected, the Input column displays collapsed versions of the Pre FX Mixer, Voice Amp Mixer, and the mixers for any active Amplitude Envelopes.

MODS tab - When the MODS tab is selected, the Input column shows the collapsed versions of any modulation sources routed to the effect in focus.

CTLS tab - When the CTLS tab is selected, the Input column shows the collapsed versions of any main performance page controls routed to the effect in focus.

MTRX tab - When the MTRX tab is selected, the Input column shows the collapsed versions of any modulation routes affecting the effect in focus.

Output Column

This display shows a collapsed version of the Main Output Mixer.

MAIN tab - The Output column displays a condensed version of the Main Output Mixer showing only the main output level controls.

Effect Modules

The Effect modules are effects that you add to the audio path in the effect blocks. The three available effect blocks can be routed in series and/or parallel, but can be populated with sub configurations that allow multiple effects to be routed within a single block for extremely powerful effect combinations.

Each module has a header that displays the name of the effect, or effect configuration, and a Mute button that bypasses the module. The EFX Split and EFX Bank modules have an additional drop down menu for managing the contents of their effect configuration slots.

Effects modules have an Input and Output Boost level amount to provide additional (or scaled down) gain amounts to the input and output levels. These are provided separately to allow for a gain structure that drives an effect with makeup gain, but still allows the Input and Output levels to be modulated independently of the boost gain structure.

Note: All Filter types are also available for use in Effects configurations. For more details on the individual Filter modules, please see the Filter section of this manual.

chorus

A chorus effect with the following parameters:

in level - Sets the input level to the effect, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

dry level - The amount of the original input signal to mix with the effected output. Polarity can be inverted with negative levels.

wet level - The amount of processed signal to mix to the Output. Polarity can be inverted with negative levels.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the effect, with a range of -100% to 100%. Negative values will invert the polarity of the signal.

shape - Selects the shape for the LFO. All of Waverazor's oscillator wave shapes are available to this effect's LFO, except for the Audio Input selections.

rate - Adjusts the LFO Rate in Hertz.

rate fine - Fine adjustment of the LFO Rate.

depth - The amount that the LFO will move through the delay line (within the Delay Min and Max settings). Negative values will move in reverse direction. Note that negative Depth values may not be audible if your LFO Shape is symmetrical, however the Depth polarity will be obvious during LFO Reset Triggers with non-symmetric LFO Shapes (like Sawtooth).

width - This is the amount of additional depth to create on the right channel, making a wider stereo image. The Delay line on the right channel is 50% larger than the Left, in order to accommodate a wider delay line sweep. Negative values will create a smaller sweep on the Right Channel than on the left.

sweep type - The Sweep type allows for different methods of using the LFO motion to move along the delay line. The available Sweep Type selections are:

Centered - The Bipolar LFO moves from the Center (controlled by Center Bias) of the Delay Line in positive (to Delay Max) and negative (to Delay Min) directions.

Center Lo - The LFO is Unipolar and moves from Delay Min to the Center (controlled by Center Bias) of the Delay line.

Center Hi - The LFO is Unipolar and moves from Delay Center (controlled by Center Bias) to the Delay Max of the Delay line.

Full Lo - The LFO is Unipolar and moves from Delay Min to Delay Max.

Full Hi - The LFO is Unipolar and moves from Delay Max to Delay Min.

feedback - Standard Delay line feedback, which can be Positive or Negative.

center bias - Allows the center point of the Delay line to be changed. Normally the center of the Delay line is midway between Min and Max, and the LFO will sweep evenly between those points, however you can skew the center point, transforming the

LFO results by moving closer to one side of the delay line or the other. Note: The Center Bias has no effect when the Sweep Type is set to Full Lo or Full Hi.

delay min - This parameter sets the minimum position, and has an effect on the overall size, of the delay line used for modulation. The Delay Min default value for this Chorus is 20 ms. There are MANY different sizes and ranges used in popular chorus and flanger algorithms, so you can dial in the settings that work for you and even modulate them.

delay max - This parameter sets the maximum position, and has an effect on the overall size, of the delay line used for modulation. The default value for this Chorus Delay Max position is 40 ms.

time smooth - This adjusts how smoothly (and slowly) transitions from one delay position to another occur, thus avoiding clicks and glitchy sounds when adjusting a delay time.

Chorus Additional Modulation sources:

lfo A out - Output value from LFO A (left).

lfo B out - Output value from LFO B (right).

lfo A gate - Output gate from LFO A (left).

lfo B gate - Output gate from LFO B (right).

Chorus Additional Modulation destinations:

lfo AB lock - This keeps the phase in sync between the left and right LFO channels by using the Left LFO only for modulation.

lfo A phase - Sets the start phase of the LFO used for the Left (A) Channel.

lfo B phase - Sets the start phase of the LFO used for the Right (B) Channel. LFO B only has an effect when the signal is stereo and LFO AB Lock is off.

rst trig - This will reset the LFOs (both A and B channels) back to their start phase.

rst trig A - This will reset LFO A (left channel) back to its start phase.

rst trig B - This will reset LFO B (right channel) back to its start phase.

frz gate - This will freeze the LFOs (both A and B channels) at their current position. This can be useful for creating sweep stuttering effects.

frz gate A - This will freeze LFO A (left channel) at its current position.

frz gate B - This will freeze LFO B (right channel) at its current position.

delay

A delay effect with the following parameters:

in level - Sets the input level to the effect, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

dry level - The amount of the original input signal to mix with the effected output. Polarity can be inverted with negative levels.

wet level - The amount of processed signal to mix to the Output. Polarity can be inverted with negative levels.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the effect, with a range of -100% to 100%. Negative values will invert the polarity of the signal.

clock - Sets the delay's clock synchronization to Self or System (host clock). When the clock is set to System, the time parameter values change to set the clock's note division.

time - Sets the delay time from 0, up to 3 seconds. However, if the clock is set to System, the time parameter values change to set the clock's note division.

feedback - Standard Delay line feedback, which can be Positive or Negative.

LR spread - Increases the stereo effect by scaling the delay times on one channel or the

other. At a value of 0%, there is no change. Negative percentages scale the delay time of the left channel up to 2x its original length, while positive percentages scale the delay time on the right channel up to 2x its original length.

time smooth - This adjusts how smoothly (and slowly) transitions from one delay position to another happen, avoiding clicks and glitchy sounds when modulating delay time. You can use this as a slew effect as well, controlling the sharpness in movement of the delay line, as low values will have very little smoothing and high values will have a VERY slow transition from one delay point to another.

Delay Additional Modulation sources:

dlypos A out - The current write position of the Left (A) Delay line can be retrieved, allowing use of the delay line timing as a modulation source.

dlypos B out - The current write position of the Right (B) Delay line can be retrieved, allowing use of the delay line timing as a modulation source.

dlypos A trig - A Trigger is issued when the Left (A) Delay line wraps back around to its beginning (as a Delay is circular buffer).

dlypos B trig - A Trigger is issued when the Right (B) Delay line wraps back around to its beginning.

Delay Additional Modulation destinations:

time (%) - Modulates the Delay time from its minimum value (0%) to its maximum value (100%), with a simple smooth curved adjustment.

time adj - Fine modulation of Time (Sec). It adds or subtracts up to 100 ms.

time fast - A non-smoothed delay time adjustment. The other Time adjustments use a lag process to sweep the delay smoothly (but slowly) from one time setting to another. The Time Fast mod destination allows you to skip the smoothing if you just need to quickly jump/set a new delay time without a sweep in between.

rst pos trig - The Reset Position Trigger resets the Delay line write position to the beginning, regardless of where it currently is located. This can be used for an effect, although it is kind of weird when you think about what is happening here.

rst clr trig - The Reset Clear Trigger resets the Delay line write position to the beginning AND clears the contents of the delay line.

delay dual

A quad delay line effect, set up with 2 delays on the left channel and 2 delays on the right. Delay dual provides the following parameters:

in level - Sets the input level to the effect, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

dry level - The amount of the original input signal to mix with the effect output. Polarity can be inverted with negative levels.

wet level - The amount of processed signal to mix to the Output. Polarity can be inverted with negative levels.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the delay effect, with a range of -100% to 100%. Negative values will invert the polarity of the signal.

time scale - Scales all the delay times from 0% to 200%, with 100% producing no change in the delay time values.

A-B spread - Increases the stereo effect by scaling the delay times on one channel or the other. At a value of 0%, there is no change. Negative percentages scale the delay times on the left channel up to 2x their original length, while positive percentages scale the delay times on the right channel up to 2x their original length.

feedbk scale - Scales all the delay feedback levels from -100% to 100%.

time smooth - This adjusts how smoothly (and slowly) transitions from one delay position to another happen, avoiding clicks and glitchy sounds when modulating delay time. You can use this as a slew effect as well, controlling the sharpness in movement of the delay line, as low values will have very little smoothing and high values will have a

VERY slow transition from one delay point to another.

rev smooth - This controls the amount of smoothing (ramping) when the reverse delay buffer jumps back to the starting point. Without smoothing, this jump can cause a glitch in the audio.

A:pre-clock - Sets the left channel pre-delay's clock synchronization to Self or System (host clock). When the pre-clock is set to System, the pre-time parameter values change to set the clock's note division.

A:pre-time - Sets the time of the left channel pre-delay from 0, up to 3 seconds. However, if its pre-clock is set to System, the pre-time parameter values change to set the clock's note division.

A:pre-feedback - Sets the feedback level for the left channel pre-delay. Feedback can be Positive or Negative.

A:clock - Sets the left channel post-delay's clock synchronization to Self or System (host clock). When the post-clock is set to System, the post-time parameter values change to set the clock's note division.

A:time - Sets the time of the left channel post-delay from 0, up to 3 seconds. However, if its clock is set to System, the time parameter values change to set the clock's note division.

A:feedback - Sets the feedback level for the left channel post-delay. Feedback can be Positive or Negative.

A:pre-drymix - The amount of the original input signal to mix with its effected output into the post-delay.

pre-reverse - Activating reverse takes the delay buffer and plays it backwards. The incoming audio is written forwards, but the data is read backwards. The effect of this will depend on the length of the delay. The pre-reverse in the "A" channel section affects the first of the two delays on the left channel.

pre-freeze - Activating freeze stops writing new audio into the delay line, so it will keep playing whatever was in the delay buffer when it was initiated. The pre-freeze in the "A" channel section affects the first of the two delays on the left channel.

post-reverse - Activating reverse takes the delay buffer and plays it backwards. The incoming audio is written forwards, but the data is read backwards. The effect of this

will depend on the length of the delay. The post-reverse in the "A" channel section affects the second of the two delays on the left channel.

post-freeze - Activating freeze stops writing new audio into the delay line, so it will keep playing whatever was in the delay buffer when it was initiated. The post-freeze in the "A" channel section affects the second of the two delays on the left channel.

A:drymix - The amount of the original input signal to mix with the post delay's effected output.

A:out level - Sets the output level from the delay's left channel, with a range of -100% to 100%. Negative values will invert the polarity of the signal.

A:out pan - Pans the output of the left channel delay between the left and right channels of the stereo output buss.

B:pre-clock - Sets the right channel pre-delay's clock synchronization to Self or System (host clock). When the pre-clock is set to System, the pre-time parameter values change to set the clock's note division.

B:pre-time - Sets the time of the right channel pre-delay from 0, up to 3 seconds. However, if its pre-clock is set to System, the pre-time parameter values change to set the clock's note division.

B:pre-feedback - Sets the feedback level for the right channel pre-delay. Feedback can be Positive or Negative.

B:clock - Sets the right channel post-delay's clock synchronization to Self or System (host clock). When the post-clock is set to System, the post-time parameter values change to set the clock's note division.

B:time - Sets the time of the right channel post-delay from 0, up to 3 seconds. However, if its clock is set to System, the time parameter values change to set the clock's note division.

B:feedback - Sets the feedback level for the right channel post-delay. Feedback can be Positive or Negative.

B:pre-drymix - The amount of the original input signal to mix with its effected output into the post-delay.

pre-reverse - Activating reverse takes the delay buffer and plays it backwards. The incoming audio is written forwards, but the data is read backwards. The effect of this will depend on the length of the delay. The pre-reverse in the "B" channel section affects the first of the two delays on the right channel.

pre-freeze - Activating freeze stops writing new audio into the delay line, so it will keep playing whatever was in the delay buffer when it was initiated. The pre-freeze in the "B" channel section affects the first of the two delays on the right channel.

post-reverse - Activating reverse takes the delay buffer and plays it backwards. The incoming audio is written forwards, but the data is read backwards. The effect of this will depend on the length of the delay. The post-reverse in the "B" channel section affects the second of the two delays on the right channel.

post-freeze - Activating freeze stops writing new audio into the delay line, so it will keep playing whatever was in the delay buffer when it was initiated. The post-freeze in the "B" channel section affects the second of the two delays on the right channel.

B:drymix - The amount of the original input signal to mix with the post delay's effected output.

B:out level - Sets the output level from the delay's right channel, with a range of -100% to 100%. Negative values will invert the polarity of the signal.

B:out pan - Pans the output of the right channel delay between the left and right channels of the stereo output buss.

Delay Additional Modulation sources:

dlypos preA out - The current write position of the first delay in the Left (A) channel can be retrieved, allowing use of the delay line timing as a modulation source.

dlypos preB out - The current write position of the first delay in the Right (B) channel can be retrieved, allowing use of the delay line timing as a modulation source.

dlypos A out - The current write position of the second delay in the Left (A) channel can be retrieved, allowing use of the delay line timing as a modulation source.

dlypos B out - The current write position of the second delay in the Right (B) channel

can be retrieved, allowing use of the delay line timing as a modulation source.

dlypos preA trig - A Trigger is issued when the first delay of the Left (A) channel wraps back around to its beginning (as a Delay is circular buffer).

dlypos preB trig - A Trigger is issued when the first delay of the Right (B) channel wraps back around to its beginning (as a Delay is circular buffer).

dlypos A trig - A Trigger is issued when the second delay of the Left (A) channel wraps back around to its beginning (as a Delay is circular buffer).

dlypos B trig - A Trigger is issued when the second delay of the Right (B) channel wraps back around to its beginning (as a Delay is circular buffer).

Delay Additional Modulation destinations:

A:pre-time fast - A non-smoothed delay time adjustment for the first delay of the Left channel. The other Time adjustments use a lag process to sweep the delay smoothly (but slowly) from one time setting to another. The Time Fast mod destination allows you to skip the smoothing if you just need to quickly jump/set a new delay time without a sweep in between.

A:time fast - A non-smoothed delay time adjustment for the second delay of the Left channel.

B:pre-time fast - A non-smoothed delay time adjustment for the first delay of the Right channel. The other Time adjustments use a lag process to sweep the delay smoothly (but slowly) from one time setting to another. The Time Fast mod destination allows you to skip the smoothing if you just need to quickly jump/set a new delay time without a sweep in between.

B:time fast - A non-smoothed delay time adjustment for the second delay of the Right channel.

rst pos trig - The Reset Position Trigger resets the Delay line write position to the beginning, regardless of where it currently is located. This can be used for an effect, although it is kind of weird when you think about what is happening here.

rst clr trig - The Reset Clear Trigger resets the Delay line write position to the beginning AND clears the contents of the delay line.

rst rev all - This will reset the Reverse Read Position of every delay line. (It has no effect if the Delay line is not in Reverse mode.) The use for this is if you have a triggered sound and you want your Reverse Delay to line up with some trigger, like a percussive sound, then you can use the reset to make sure the Reverse playback aligns with it, especially if you have anything that is also modulating the delay time.

Normally a reverse delay just writes sound into a buffer and it is read out of that buffer backwards, but that reading process also has to loop back to the beginning of the buffer, so the synchronization of the delay length and the position of the backwards reader will lose any correlation when you start adjusting delay times. Using this trigger, you can force the correlation to happen when you want it to happen.

rst rev preA - This will reset the Reverse Read Position of the first delay of the Left (A) channel. (It has no effect if the Delay line is not in Reverse mode.)

rst rev preB - This will reset the Reverse Read Position of the first delay of the Right (B) channel. (It has no effect if the Delay line is not in Reverse mode.)

rst rev A - This will reset the Reverse Read Position of the second delay of the Left (A) channel. (It has no effect if the Delay line is not in Reverse mode.)

rst rev B - This will reset the Reverse Read Position of the second delay of the Right (B) channel. (It has no effect if the Delay line is not in Reverse mode.)

flanger

A flanger effect with the following parameters:

in level - Sets the input level to the effect, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

dry level - The amount of the original input signal to mix with the effected output. Polarity can be inverted with negative levels.

wet level - The amount of processed signal to mix to the Output. Polarity can be inverted with negative levels.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the effect, with a range of -100% to 100%. Negative values will invert the polarity of the signal.

shape - Selects the shape for the LFO. All of Waverazor's oscillator wave shapes are available to this effect's LFO, except for the Audio Input selections.

rate - Adjusts the LFO Rate in Hertz.

rate fine - Fine adjustment of the LFO Rate.

depth - The amount that the LFO will move through the delay line (within the Delay Min and Max settings). Negative values will move in reverse direction. Note that negative Depth values may not be audible if your LFO Shape is symmetrical, however the Depth

polarity will be obvious during LFO Reset Triggers with non-symmetric LFO Shapes (like Sawtooth).

width - This is the amount of additional depth to create on the right channel, making a wider stereo image. The Delay line on the right channel is 50% larger than the Left, in order to accommodate a wider delay line sweep. Negative values will create a smaller sweep on the Right Channel than on the left.

sweep type - The Sweep type allows for different methods of using the LFO motion to move along the delay line. The available Sweep Type selections are:

Centered - The Bipolar LFO moves from the Center (controlled by Center Bias) of the Delay Line in positive (to Delay Max) and negative (to Delay Min) directions.

Center Lo - The LFO is Unipolar and moves from Delay Min to the Center (controlled by Center Bias) of the Delay line.

Center Hi - The LFO is Unipolar and moves from Delay Center (controlled by Center Bias) to the Delay Max of the Delay line.

Full Lo - The LFO is Unipolar and moves from Delay Min to Delay Max.

Full Hi - The LFO is Unipolar and moves from Delay Max to Delay Min.

feedback - Standard Delay line feedback, which can be Positive or Negative.

center bias - Allows the center point of the Delay line to be changed. Normally the center of the Delay line is midway between Min and Max, and the LFO will sweep evenly between those points, however you can skew the center point, transforming the LFO results by moving closer to one side of the delay line or the other. Note: The Center Bias has no effect when the Sweep Type is set to Full Lo or Full Hi.

delay min - This parameter sets the minimum position, and has an effect on the overall size, of the delay line used for modulation. The Delay Min default value for this flanger is 10 ms. There are MANY different sizes and ranges used in popular chorus and flanger algorithms, so you can dial in the settings that work for you and even modulate them.

delay max - This parameter sets the maximum position, and has an effect on the overall size, of the delay line used for modulation. The default value for this flanger Delay Max position is 20 ms.

time smooth - This adjusts how smoothly (and slowly) transitions from one delay position to another occur, thus avoiding clicks and glitchy sounds when adjusting a delay time.

Flanger Additional Modulation sources:

lfo A out - Output value from LFO A (left).

lfo B out - Output value from LFO B (right).

lfo A gate - Output gate from LFO A (left).

lfo B gate - Output gate from LFO B (right).

Flanger Additional Modulation destinations:

lfo AB lock - This keeps the phase in sync between the left and right LFO channels by using the Left LFO only for modulation.

lfo A phase - Sets the start phase of the LFO used for the Left (A) Channel.

lfo B phase - Sets the start phase of the LFO used for the Right (B) Channel. LFO B only has an effect when the signal is stereo and LFO AB Lock is off.

rst trig - This will reset the LFOs (both A and B channels) back to their start phase.

rst trig A - This will reset LFO A (left channel) back to its start phase.

rst trig B - This will reset LFO B (right channel) back to its start phase.

frz gate - This will freeze the LFOs (both A and B channels) at their current position. This can be useful for creating sweep stuttering effects.

frz gate A - This will freeze LFO A (left channel) at its current position.

frz gate B - This will freeze LFO B (right channel) at its current position.

reverb

The reverb is a comb filter based reverb algorithm with a few basic controls, and a bunch of advanced controls for the adventurous sound designer and modulation enthusiast. It is a mono-input, stereo-output design. When you send discreet left/right signals into the reverb, they will be combined and processed together. For a description of which parameters are involved in the stereo imaging of the reverb, see the Advanced Parameters section below.

dry level - This controls the amount of the original input signal to mix into the output. The polarity of this signal can be inverted using negative level values.

wet level - This controls the amount of processed signal to mix into the output. The polarity of this signal can be inverted using negative level values.

pre delay - Sets the reverb pre delay time in milliseconds.

decay - Adjusts the overall reverb decay time in seconds.

param setup - This is a convenient macro menu that will change the advanced parameters in order to set you up for different types of reverb. It provides a quick way to get the parameter values close to a desired reverb sound and you can continue adjusting the parameters from there.

ADVANCED PARAMETERS:

In this section, the inner workings of a comb-delay based reverb are made available for editing and routing in the Mod Matrix. For example, you can create shimmering effects by routing an LFO to the comb filter delay times, or to the all pass filter's shift frequency... chaos is certainly lurking in this section, along with potential happy accidents.

delay smooth - All of the delay lines have smoothing capability in order to keep parameter changes glitch-free. When you are modulating the delay lines, you may want to adjust the smoothing amount. Higher values result in slower time transitions, and lower values will have faster transitions to new delay time settings.

pre level - This is the level to the pre-delay, as well as the input level to the reverb, in general.

input lpf - This controls a low pass filter on the Input to the reverb.

lpf out L - This controls a low pass filter on the left channel output of the reverb.

lpf out R - This controls a low pass filter on the right channel output of the reverb.

apf (1-4) delay - There are four all pass filters used in this design. 1 and 2 are part of the mono input processing section, and 3 and 4 are used for the left and right output channels respectively. Each apf delay control sets the delay time of its all pass filter. The speed at which the delay time is updated when adjusted or modulated is determined by the delay smooth control.

apf (1-4) shift - Sets the shift frequency of the all pass filter.

COMB FILTERS:

There are eight comb filters in the reverb. Combs 1 through 4 are for the left channel, and combs 5 through 8 are for the right Channel. The differences between these two banks of filters is the key to getting a stereo image.

Comb delay times (1L, 2L, 3L, 4L, 5R, 6R, 7R, 8R) - Sets the delay time of the comb filter. This creates the repeating echoes that will eventually smear together with the other combs in the series in order to give you a reverb sound. Setting the delay times to avoid mathematical relationships between the combs is helpful when you want to avoid 'ringing' in the sound. The speed at which the delay time is updated when adjusted or modulated is determined by the delay smooth control.

fdbk scale (for combs 1L, 2L, 3L, 4L, 5R, 6R, 7R, 8R) - The feedback of the comb filter is what determines the decay time of the reverb. These all follow the main decay parameter, however you can change each comb's feedback relationship to each other by changing the individual feedback scaling. At 100%, the comb feedback amount will track directly with the main decay control. Higher values will increase the feedback beyond the main decay setting.

Comb feedback polarity (the buttons at the bottom of the module associated with combs 1L, 2L, 3L, 4L, 5R, 6R, 7R, 8R) - These buttons allow for inverting the feedback polarity of the individual comb filters.

efx split

The EFX Split configuration module allows Waverazor's stereo EFX block to be turned into two parallel Mono FX Buses. Two Effects or Filter types can be chosen for the sub-block slots. The first slot becomes the Left Bus, the second slot becomes the Right Bus.

Effects or filters can be added to either slot by clicking in its empty field, or on the menu button and then choosing from a drop down list. When a slot is filled and selected for editing, the choices under the menu button expand to include options for removing an unwanted effect.

Once the slots are filled, you can click on any slot to select a sub-module and view its parameters. Each of the sub-modules can be muted individually by pressing their respective mute buttons, or the whole configuration can be bypassed by pressing the mute button on the EFX Split module itself. If a slot is empty, the audio simply passes through unaffected.

efx bank

The EFX Bank allows for 16 effects (or filter types) to be added as sub-blocks and then routed in six different ways.

proc order - Changes how the DSP sub-blocks are configured. The available routings are:

Serial - Up to 16 stereo effects or filters routed one after another in series.

Parallel - Up to 16 stereo effects or filters routed side by side and then summed to a stereo output.

A|B Stereo - 2 chains of up to 8 serial stereo effects or filters each, both running parallel to each other and summed at the end.

A|B m/split - 2 chains of up to 8 serial mono effects or filters each, both running parallel to each other and maintaining stereo separation on input

and output.

I-A|B-O - Up to 4 serial stereo effects or filters in series, which then split into 2 chains of up to 4 serial mono effects each, and then feed another 4 serial stereo effects.

I-A|B|C|D-O - Up to 2 serial stereo effects or filters in series, which then split into 4 chains of up to 3 serial mono effects each, with 2 chain in parallel on the left and 2 in parallel on the right, finally feeding another 2 serial stereo effects.

out level - Sets the output level for the entire EFX Bank, from -100% to 100%. Negative values will invert the polarity of the waveform, e.g. -100% is the same level as 100%, but with inverted polarity.

Effects or filters can be added to any slot by clicking in its empty field, or on the menu button and then choosing from a drop down list. When a slot is filled and selected for editing, the choices under the menu button expand to include options for removing an unwanted effect.

Once the slots are filled, you can click on any slot to select a sub-module and view its parameters. Each of the sub-modules can be bypassed individually by pressing their respective mute buttons, or the whole configuration can be bypassed by pressing the mute button on the EFX Split module itself. If any slot is empty, the audio simply passes through it unaffected.

tap line 8X

Eight delay taps.

Parameters

in level - Sets the input level to the filter, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

in boost - Provides additional gain or attenuation to the input level.

dry level - The level of the dry input signal mixed in with the tap line delay effect.

num taps - Sets the number of delay taps between 0 and 8.

clip thresh - Sets a threshold level above which the signal is clipped.

out boost - Provides additional gain or attenuation to the output level.

out level - Sets the output level from the effect, from -100% to 100%. Negative values will invert the polarity of the signal, e.g. -100% is the same level as 100%, but with inverted polarity.

feed rate - Controls the amount of sample rate reduction on the delay line.

feed keytrack - Adjusts the Key Tracking of the feed rate, relative to the feed rate parameter. At a setting of 0%, the feed rate stays the same, regardless of the note number that triggered the voice. At 100%, the feed rate perfectly tracks note frequency. To reverse the feed rate to keyboard relationship, you can set the key track to a negative value. Key Tracking is centered around C3.

taps keytrk - Adjusts the Key Tracking of all tap times, relative to their individual tap time settings. At a setting of 0%, the original tap times stay the same, regardless of the note number that triggered the voice. At 100%, the relative tap times perfectly track note frequency. To reverse the tap tracking to keyboard relationship, you can set this parameter to a negative value. Key Tracking is centered around C3. Taps keytrk only affects taps that have their key trk parameter set to On.

time scale - Scales all tap times at once. The default setting is 100% which produces no change, while the maximum setting of 200% doubles the tap times from their current value. The minimum setting of 001% scales the tap times to 1% of their original value. Time scale will only affect taps that have their time trk parameter set to On.

time spread - Increases all of the delay taps simultaneously and in a relative way, spreading Tap 1 the furthest and Tap 8 the least. Time spread will only affect taps that have their time trk parameter set to On.

fdbk scale - Scales all feedback times at once. The default setting is 100% which produces no change, while the maximum setting of 200% doubles the tap feedback from their current value. The minimum setting of 001% scales the tap times to 1% of their original value. Fdbk scale will only affect taps that have their fdbk trk parameter

set to On.

tap smooth - Slows down the tap time adjustments. It is useful if you are modulating the tap times, or using Time Spread or Time Scale, and you want to modulate smoothly (lag) to the new time value.

tap 1-8 level - Sets the mix level of the tap delay.

tap 1-8 time - Sets the delay time in milliseconds, up to 3000 mSec.

timebase - Changes the delay time setting between millisecond, tuned ratio, and tempo modes. The tuned ratios are Octaves : Semitones : Cents. These are helpful if you are using the taps in relative tuning modes with key tracking.

tap 1-8 feedback - Sets the feedback amount for the tap.

tap 1-8 key trk - When engaged, defeats the effect of the taps keytrk parameter.

tap 1-8 time trk - When engaged, defeats the effect of the time scale and time spread parameters.

tap 1-8 fdbk trk - When engaged, defeats the effect of the fdbk scale parameter.

OUT Block NAV

The OUT block contains controls for the final mix to the Audio Output, and the mix to the Feedback path leading back to various points in the voice architecture.

Focus

MAIN tab - Displays the Main Output Mixer module.

Input Column

This is a contextual, tabbed display, the contents of which are dependent on what is currently shown in focus (Center).

PREMIX tab - When the PREMIX tab is selected, the Input column displays collapsed versions of the Pre FX Mixer.

MODS tab - When the MODS tab is selected, the Input column shows the collapsed versions of any modulation sources routed to the Main Output Mixer.

CTLS tab - When the CTLS tab is selected, the Input column shows the collapsed versions of any main performance page controls routed to the Main Output Mixer.

MTRX tab - When the MTRX tab is selected, the Input column shows the collapsed versions of any modulation routes affecting the Main Output Mixer.

Output Column

This display shows the AUDIO OUTPUT module, containing meters and a level slider for the final audio output from Waverazor.

Main Output Mixer Module

The Main Output Mixer is responsible for mixing the effect returns and the main bus either into the Main Audio Output or into a Feedback path that sends signal back to multiple points in the voice architecture. The available mix controls are:

Audio Output

mix bus lev - Controls the Main Mix Bus level to the Main Out Level.

mix bus bal - Controls the Main Mix Bus balance to the Main Out Level.

mix bus pol - Controls the Main Mix Bus polarity to the Main Out Level.

EFX1 bus lev - Controls the EFX1 return level to the Main Out Level.

EFX1 bus bal - Controls the EFX1 return balance to the Main Out Level.

EFX1 bus pol - Controls the EFX1 return polarity to the Main Out Level.

EFX2 bus lev - Controls the EFX2 return level to the Main Out Level.

EFX2 bus bal - Controls the EFX2 return balance to the Main Out Level.

EFX2 bus pol - Controls the EFX2 return polarity to the Main Out Level.

EFX3 bus lev - Controls the EFX3 return level to the Main Out Level.

EFX3 bus bal - Controls the EFX3 return balance to the Main Out Level.

EFX3 bus pol - Controls the EFX3 return polarity to the Main Out Level.

main out lev - Controls the level of the final mix to the Main Audio Output.

main out bal - Controls the balance of the final mix to the Main Audio Output.

main out pol - Controls the polarity of the final mix to the Main Audio Output.

clip limit - Sets the level at which the signal is clip limited.

dc blocker - Activates a fixed structure high pass filter that removes DC signals from the main output.

Feedback Bus Sends

mix fdback lev - Controls the Feedback send level from the Main Mix Output Bus.

mix fdback bal - Controls the Feedback send balance from the Main Mix Output Bus.

EFX1 fdback lev - Controls the Feedback send level from the EFX1 Output Bus.

EFX1 fdback bal - Controls the Feedback send balance from the EFX1 Output Bus.

EFX2 fdback lev - Controls the Feedback send level from the EFX2 Output Bus.

EFX2 fdback bal - Controls the Feedback send balance from the EFX2 Output Bus.

EFX3 fdback lev - Controls the Feedback send level from the EFX3 Output Bus.

EFX3 fdback bal - Controls the Feedback send balance from the EFX3 Output Bus.

fdback boost L - Boosts or attenuates the feedback send level for the Left channel.

fdback boost R - Boosts or attenuates the feedback send level for the Right channel.

fdback send L - Controls the Feedback send level to the Left channel of the Main Feedback Bus.

fdback send R - Controls the Feedback send level to the Right channel of the Main Feedback Bus.

fdback timesep - Turning this on allows you to set different delay times for the left and right channels of the feedback delay line. When set to off, both delay times use the value from the Left channel setting.

fdback time L - Sets the delay time for the Left channel of the feedback delay line.

fdback time R - Sets the delay time for the Right channel of the feedback delay line.

fdback clip L - Sets the level at which the Left channel of the feedback bus hard clips.

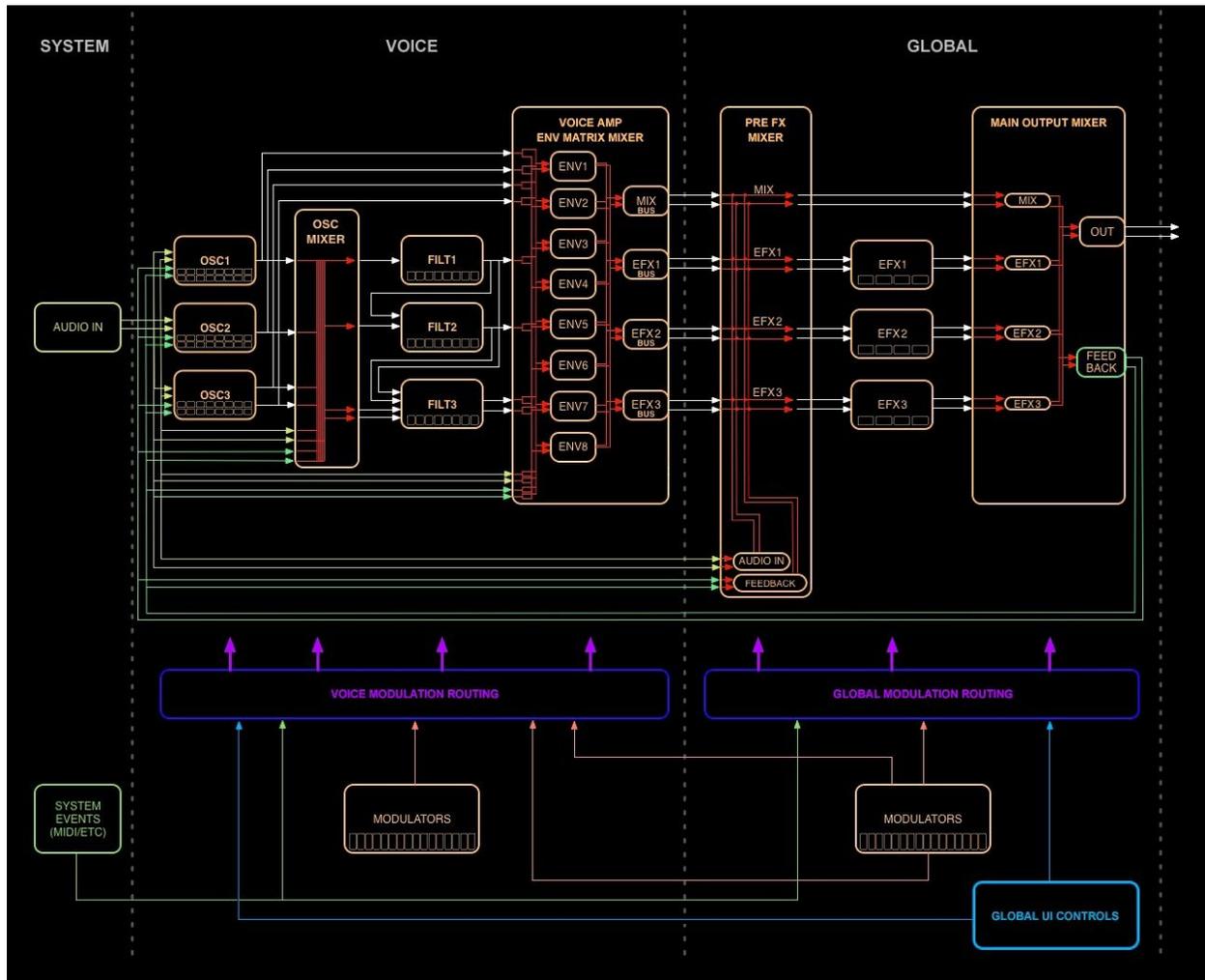
fdback clip R - Sets the level at which the Right channel of the feedback bus hard clips.

 Appendix A - Wave Shapes

None	ExLog 3	LogSawUp 1
Sine	ExLog 4	LogSawUp 2
SineInv	ExLog 5	LogSawUp 3
SineEx	ExLog 6	LogSawUp 4
SineLog	ExLog 7	LogSawUp 5
Triangle	ExLog 8	LogSawUp 6
Exp 1	ExLog 9	LogSawUp 7
Exp 2	LogEx 1	LogSawUp 8
Exp 3	LogEx 2	LogSawUp 9
Exp 4	LogEx 3	Saw Down
Exp 5	LogEx 4	ExpSawDn 1
Exp 6	LogEx 5	ExpSawDn 2
Exp 7	LogEx 6	ExpSawDn 3
Exp 8	LogEx 7	ExpSawDn 4
Exp 9	LogEx 8	ExpSawDn 5
Log 1	LogEx 9	ExpSawDn 6
Log 2	Saw Up	ExpSawDn 7
Log 3	ExpSawUp 1	ExpSawDn 8
Log 4	ExpSawUp 2	ExpSawDn 9
Log 5	ExpSawUp 3	LogSawDn 1
Log 6	ExpSawUp 4	LogSawDn 2
Log 7	ExpSawUp 5	LogSawDn 3
Log 8	ExpSawUp 6	LogSawDn 4
Log 9	ExpSawUp 7	LogSawDn 5
ExLog 1	ExpSawUp 8	LogSawDn 6
ExLog 2	ExpSawUp 9	LogSawDn 7

LogSawDn 8	Rand R1	Noise R2
LogSawDn 9	Rand R2	Noise R3
RipSawUp	Rand R3	Noise R4
RipSawDown	Rand R4	Noise R5
Chop Saw	Rand R5	Noise R6
Chop Sin	Rand R6	Noise R7
SawTri Up	Rand R7	Noise R8
SawTri Dn	Rand R8	Noise R9
Square	Rand R9	
Pulse	Noise S0	(Oscillator only)
Rand S0	Noise S1	AudIn L
Rand S1	Noise S2	AudIn R
Rand S2	Noise S3	AudInL+R
Rand S3	Noise S4	(Curvatures only - Controls, Mod Routes, Envelopes, Slope Gen, Portamento, Shaper)
Rand S4	Noise S5	
Rand S5	Noise S6	
Rand S6	Noise S7	Boolean
Rand S7	Noise S8	BoolSoft
Rand S8	Noise S9	BoolSftr
Rand S9	Noise R0	Fixed
Rand R0	Noise R1	

Appendix B - Voice Architecture



Appendix C - Triggers and Gates

Aside from continuous modulation signals routed and shaped through the modulation matrix, you can also send triggers and gates to actuate various synthesis functions. These special modulators are found in the modulation matrix source and destination lists of various synthesis modules.

Trigger - A Trigger almost always targets an input and will have an effect when the signal transitions from Low (unset) to High (set). An example would be using an LFO source to reset an Envelope. As the LFO transitions from low to high (crossing a threshold of 50%), the Envelope will reset. The envelope will not reset again until the input signal from the LFO transitions back to low (below 50%) and then transitions high again (above 50%).

Gate - A Gate is commonly used for both inputs and outputs. A Gate signal has a function that operates while a signal is held in a high state. For an output example, Envelopes can generate a Sustain Gate, where the output signal is High while the Envelope is in the sustain stage, and Low at all other times. For an input example, LFOs have a function where the LFO will freeze at its current position while the input signal to its Freeze Gate destination is High.

Latch - A Latch is a type of Gate that holds its state, and is almost always used for outputs. For example, an Envelope Sustain Latch output will have a High state once the envelope reaches its sustain stage, and the signal will remain High until the end of the Envelope, or until the Envelope is reset. Unlike the Envelope Sustain Gate, which is High only during sustain, this signal can be used for changing a state at, or beyond, the Sustain Stage in an Envelope.

Appendix D - Additional Modulation Sources

In addition to traditional modulation sources like LFOs and Envelopes, Waverazor provides additional modulation sources that are only selectable from within a mod matrix route.

System

level in L - The left channel level into the engine from the audio input, post level scaling.

level in R - The right channel level into the engine from the audio input, post level scaling.

level out L - The left channel level out from the engine, post main out level.

level out R - The right channel level out from the engine, post main out level.

cpu usage - The level of CPU usage. A practical use might be to shorten envelope release times if CPU usage increases, thereby ending voices earlier than normal and reducing CPU load.

voice count - A level derived from the number of voices (polyphony) currently sounding.

sample rate - A level based on Waverazor's current sample rate.

in channels - A level derived from the sum of the input channels.

out channels - A level derived from the sum of the output channels.

proc bufsize - A level based on the voice buffer value in the system settings page.

voice limit - A level based on the "num voices" value in the system settings page.

host sr - A level based on the host application sample rate.

host sr conv - A level based on Waverazor's conversion of the host application sample rate, multiplied or divided to fall within a 40 kHz to 192 kHz range.

x.clkstate - A level based on system tempo sync status. (0=Off, 1=Sync Tempo, 2=Sync Tempo and Beat Clock) This is useful if you want to modulate something different based on whether or not Waverazor is synchronizing to the host.

x.tempo - A level based on the system tempo, which can change if externally synchronized.

x.timesig num - A level based on the numerator of the host's time signature, which can change (if supported by the host).

x.timesig den - A level based on the denominator of the host's time signature, which can change (if supported by the host).

o.tempo - A level based on the system tempo parameter value. This source doesn't change if tempo is synchronized to the host.

o:cur bar - (Range 0 - 2048) This source counts upwards as the bar progresses and then wraps around.

o:cur beat - (Range 0 - 64) This source counts up and resets, depending on the time signature. The max value implies a Max time signature numerator of 64.

o:cur beat64 - (Range 0 - 64) This source counts up and resets, depending on the time signature, and is always locked to be 1/64th of a whole note. The whole note value is derived from the time signature.

o:cur beatX - (Range 0 - 64) This source counts up and resets, depending on the time signature. The X value is currently fixed to reset on 1/4 notes.

o:trig bar - A pulse trigger is generated at each bar.

o:trig beat - A pulse trigger is generated at each beat.

o:trig beat64 - A pulse trigger is generated at each 1/64 of a beat.

o:trig beatX - A pulse trigger is generated at each 1/4 note. The X value is currently fixed to reset on 1/4 notes.

The o:cur bar, o:cur beat, o:cur beat64, and o:cur beatX sources are levels that increase over time and then wrap around depending on time signature. These are all '0' based... for example in 3/4 time, BeatX will generate a value pattern like this: 0,1,2,0,1,2,0... in 8/4 time, BeatX would generate 0,1,2,3,4,5,6,7,0,1,2,3,4,5,6,7,0...

rand once - Generates a random number, per voice, when the voice is launched and holds that random number.

Modulator: Global Control State

chan vol - Relays the MIDI volume CC#7 value.

mod wheel - Relays the MIDI modulation wheel CC#1 value.

pitch bend - Relays the 14 bit MIDI pitch bend value.

pan - Relays the MIDI balance (also known as pan) CC#8 value.

chan press - Relays the MIDI channel pressure (also known as aftertouch) value.

sus pedal - A modulation value generated from the MIDI sustain pedal state.

random - Generates a random number.

rand stat - Generates a random number, but steps through the random number table each time a note is launched, so the random number sequence will repeat. A new sequence is generated each time Waverazor is instantiated.

rand once - Generates a random number for each voice and holds that number until a new voice generates a new random number.

sys ctl A - An additional control input available for input from the host's automation lane.

sys ctl B - An additional control input available for input from the host's automation lane.

sys ctl C - An additional control input available for input from the host's automation lane.

sys ctl D - An additional control input available for input from the host's automation lane.

sys ctl E - An additional control input available for input from the host's automation lane.

sys ctl F - An additional control input available for input from the host's automation lane.

sys ctl G - An additional control input available for input from the host's automation lane.

sys ctl H - An additional control input available for input from the host's automation lane.

first Note (ch) - The modulation value equals the first received MIDI note number on the track MIDI channel.

first Vel (ch) - The modulation value equals the first received MIDI velocity value on the track MIDI channel.

last Note (ch) - The modulation value equals the last received MIDI note number on the track MIDI channel.

last Vel (ch) - The modulation value equals the last received MIDI velocity value on the track MIDI channel.

first Note (any) - The modulation value equals the first received MIDI note number on any channel.

first Vel (any) - The modulation value equals the first received MIDI velocity value on any channel.

last Note (any) - The modulation value equals the last received MIDI note number on any channel.

last Vel (any) - The modulation value equals the last received MIDI velocity value on any channel.

num Notes On - Relays a value between 0 and 128 equivalent to the number of notes currently being played.

bool Notes On - Generates a value that is 0 or 1, depending on whether any notes are currently being played. A state with no notes results in 0, while any notes played result in a value of 1.

Appendix E - US Patent

Waverazor's wave slicing oscillator algorithm is protected by US Patent, number 10,262,646.

Waverazor Community

For additional information, please join the conversation at the MOK User Forum:

<https://shrine.mok.com/>

To keep up with the latest MOK news, please join our no-spam mailing list at:

<http://mok.com/>

Or like us on Facebook, follow us on Twitter, and subscribe to us on YouTube.

We are here for you!

- Rob, Taiho and Chris

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