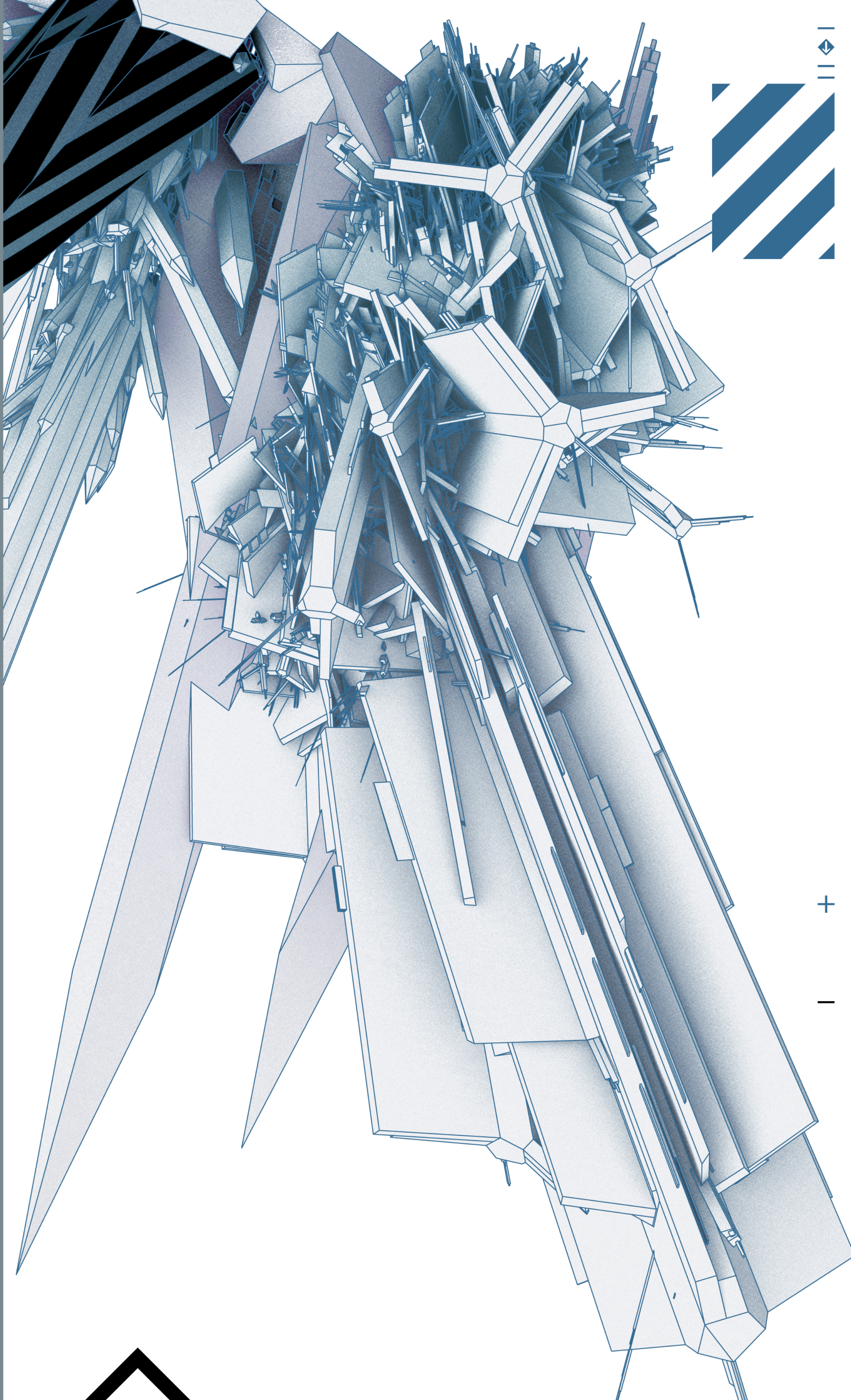
OPERATION MANUAL



STROBE2

fxpansion

Operation Manual

1 Strobe2 Introduction

Introduction

Strobe2 is a performance synthesizer, designed to be easy to program so that you can concentrate on playing and recording great-sounding music! A single-oscillator synth with osc-stacking, sub-osc and osc sync, it also features a versatile multimode filter, dual LFO, ramp generator and two envelopes.

Strobe2 is designed for powerful analogue-style basses, leads, pads and polysynth sounds but its modulation and effects functionality make it great for experimentation and discovering new kinds of timbres.

While being inspired by relatively simple monosynths such as the Roland SH series (especially the SH-09 and SH-101), Oberheim OB-1 and SCI Pro-One, Strobe2 is not an exact model of any particular synth and features hugely improved possibilities over its vintage inspirations: it has been designed to take the performance synthesizer into a new dimension.

Strobe2 highlights

Osc section

At the heart of Strobe2 is a single 'super-oscillator' which allows many of the timbres possible with multiple-oscillator architecture but with far fewer controls, enabling fast programming.

• Each waveform in each osc sub-section: main oscillator, sub oscillator and noise generator - can be heard in parallel using mixer faders to blend between them.

• It features built-in hard-sync and osc stacking functions for complex harmonics, detuning effects and chord sounds.

• Additionally, all 3 osc sub-sections each feature a polyphonic keyboard-tracking Tone control for adjusting spectral balance before the more radical sculpting offered by the V.C.F. section.

V.C.F. (Voltage-controller Filter) section

The detailed filter model features a large variety of modes, leading to a wide range of potential timbres. It also includes realistic overdrive of the circuit, further increasing the tonal range.

V.C.A. (Voltage-controlled Amplifier) section

Strobe's Amp section features a model of an 'O.T.A.' type filter circuit core which produces realistic overdrive and saturation - a very important part of the sound of vintage synths. This section also contains the Analogue control which introduces noise and hum into Strobe2's control circuitry - this can result in musically organic behaviour with realistic hardware noise.

Simple direct modulation

Strobe2 has been designed so that its main parameters can be modulated using dedicated controls - the Osc Pitch, Pulse Width and Filter Cutoff can be directly modulated by the keyboard, LFO or mod envelope using dedicated faders. These modulation techniques are able to create many classic analogue-style sounds.

Advanced but intuitive TransMod modulation

Strobe2 also features the *TransMod* modulation system which goes far beyond these dedicated modulation routings and greatly expands its sound design and performance potential. TransMod modulation offers a huge range of monophonic and polyphonic modulation sources which can be routed to almost any synthesis parameter.

Polyphony and unison

Unlike the monosynths which inspired its design, Strobe2 is capable of polyphony - you can set up as many voices as your CPU can handle and also define a number of unison voices. For example, with 8 voices and 2 unison voices, the result is 4-note polyphony with each note comprising 2 unison voices.

Using the TransMod modulation system, it is possible to vary almost any synthesis parameter for each unison voice - not just the simple unison detuning available on vintage polysynths.

Arpeggiator page

Strobe2's Arpeggiator page turns MIDI note input into rhythmic arpeggio sequences. It also provides a parallel modulation Step Sequencer for modulating parameters via the TransMod system. The Arpeggiator and Step Sequencer can interact in a number of ways.

Effects page

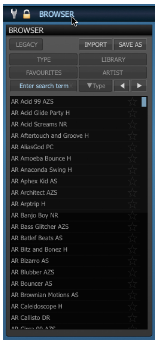
Finally, Strobe2 provides a built-in effects section - 6 instances from 28 available built-in FX devices may be loaded and even modulated via the TransMod system.

Other highlights

Strobe2 also includes a preset switching and morphing system as well as a Randomizer which makes it easy to come up with new sounds.

1.1 Interface overview

Interface overview



Browser

The Browser provides 1-click access to Strobe2's library of presets. It also saves and loads presets created or modified by the user.

The Browser's visibility can be toggled by clicking the **Browser** button above the Browser area.

Just above the Browser are buttons which display the following menus:

 Preferences menu : contains a number of additional Strobe2 settings

 Locks menu : provides the ability to lock groups of controls from being affected by editing, preset-switching, morphing and randomizing

Main Editing area

The main editing area is switchable between 3 different pages:

• Synth page

Synthesis engine

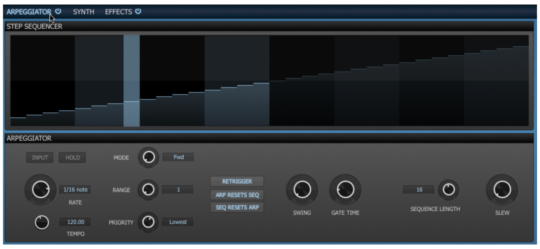
The Synth page contains the synthesis engine parameters for Strobe2 - all parts of the audio path before the effects section - as well as the Visualizer Scope.

Modulation

The Synth page also includes the Dual LFO, Ramp and Envelopes.

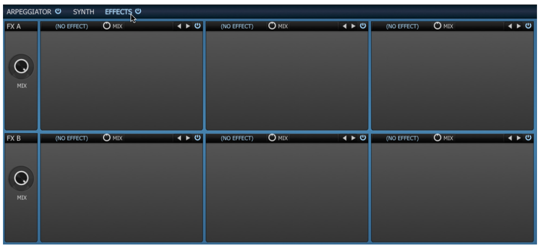
The Scope area can be switched to display the Euclid, Curve and KeyZone modulation processor editors.

• Arpeggiator page



This page contains Strobe2's built-in Arpeggiator as well as a parallel modulation Step Sequencer which can be used with the TransMod system and also interact with the Arpeggiator.

• Effects page



The Effects page contains 2 FX chains (FX A and FX B), each of which features 3 slots in which to insert FX devices. Strobe2 provides 28 built-in FX devices of various kinds.

Global synth controls

This section of the interface is always visible, even when viewing the Arpeggiator or Effects pages.

These settings relate to the way that Strobe2 responds to being played - including polyphony, unison, note priority, master tuning, pitch bend and glide functionality. This section also contains the **CC LRN** button for assigning MIDI CCs to controls.

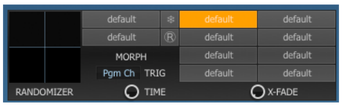
This area also contains the **P1** (Perf1) and **P2** (Perf2) performance controls which provide variation with real-time performance controller input. These must be assigned to available hardware MIDI CC controllers using the MIDI CC Learn mode function.

TransMod modulation slots

The 16 TransMod modulation slots are central to Strobe2's advanced but intuitive TransMod modulation system.

Although Strobe2 already features dedicated Keyboard, LFO and Envelope modulation amount faders for oscillator pitch, pulse width and filter cutoff, the TransMod modulation system goes far beyond these capabilities. Each of the 16 TransMod slots can modulate virtually all Strobe2 parameters from a single modulation source, optionally scaled (multiplied) by another modulation source.

Quick-presets, morphing, Freeze and Randomizer controls



This section allows 8 presets to be loaded into the available Quick-preset slots for fast access and morphing functions which offer radical timbral shifting. Morphing can also be 'frozen' to create new interesting sounds, as can this section's Randomizer functions.

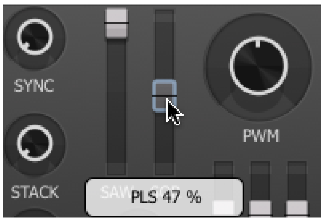
1.2 Adjusting sliders and rotary controls

Adjusting initial values of parameters (No TransMod slot selected)

Sliders

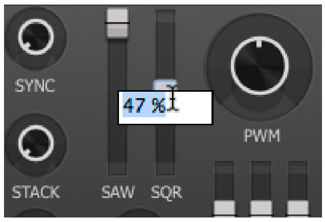
Move the mouse above the slider cap. Notice that it is highlighted. Now either:

• use the mousewheel or trackpad scroll function



• or click and drag the slider cap up/down.

The new value is shown in real time on the tooltip which appears.



• It is also possible to double-click the slider cap, type a new value in the box which appears and press ENTER or RETURN.

Rotary pots

Move the mouse above the rotary control. Notice that it is highlighted. Now either:

• use the mousewheel or trackpad scroll function

• or click and drag the control up/down.

The new value is shown in real time on the tooltip which appears.

• It is also possible to double-click the control, type a new value in the box which appears and press ENTER or RETURN.

Fine control over parameters

Hold down the SHIFT key while adjusting a control for finer resolution.

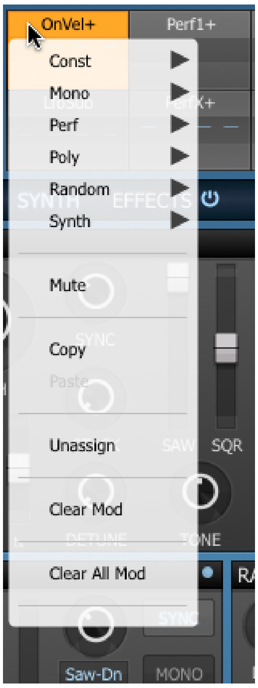
1.3 Adjusting TransMod modulation

Introduction to TransMod modulation

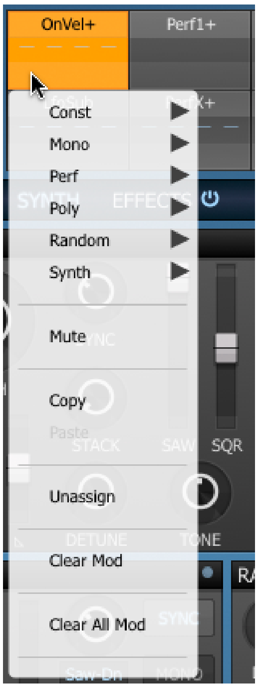
The TransMod system provides an intuitive but very deep modulation system for Strobe2's parameters. See chapter 5 for a full guide to understanding the TransMod system.

Selecting TransMod slots

To begin using TransMod modulation, one of the 16 TransMod slots must first be selected. Each TransMod slot can be assigned to a modulation source which is chosen from an extensive list of monophonic (applied to all playing notes) and polyphonic (applied to each note individually) modulation sources.

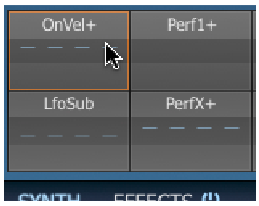


• click on the *upper* part of the slot To select a slot and display its TransMod source menu.



• click on the *lower* part of the slot to select a slot and display its TransMod scalar menu.

• to select a slot without displaying any menus, click the *central* part of the slot.



• to deselect all TransMod slots to work only on initial parameter values as shown in the last section, click its *central* part again.

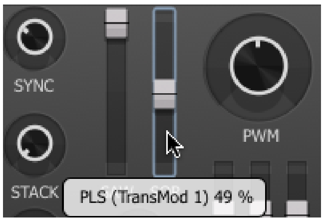
The vast range of TransMod modulation sources (which can also be used as scalars, which scale or multiply a slot's modulation source) are described later in this manual.

When a TransMod slot is selected, TransMod-enabled parameters in Strobe2 display and allow editing for modulation depths on parameters. The vast majority of parameters in Strobe2's Synth, Effects and Arpeggiator pages can be modulated. However, some controls such as buttons and certain other parameters cannot be modulated.

Adjusting TransMod modulation depths: Sliders

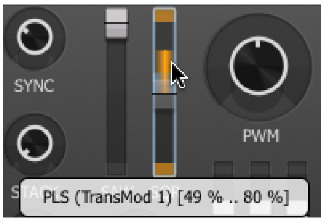
When a TransMod slot is selected, Strobe2's slider-type controls can be manipulated in the following ways:

Creating modulation on sliders



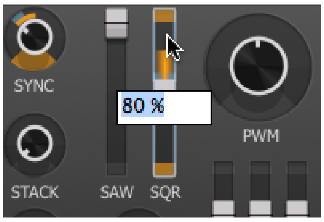
Position the mouse within the slider but *above or below the slider cap*. Notice that the entire slider is highlighted. Now either:

• use the mousewheel or trackpad scroll function



• or click and drag up/down.

The initial value and new modulation value are both shown in real time on the tooltip which appears.

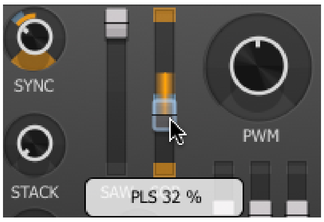


• It is also possible to double-click above/below the slider cap, type a new value in the box which appears and press ENTER or RETURN.

Adjusting the initial value with or without modulation

Move the mouse above the slider cap - notice it is highlighted.

• Use the mousewheel to adjust the initial value - the modulation depth is preserved.



• or click and drag up/down.

The initial value is shown in real time on the tooltip which appears.

• Alternatively, double-click and type a new value.

• It is also possible to hold down the ALT key while clicking and dragging the slider cap (or scrolling with the mousewheel/trackpad) to adjust the initial value *without* preserving the modulation depth.

Adjusting TransMod modulation depths: Rotary controls

Creating modulation on rotary controls

Move the mouse above the *outer ring* of the rotary control. Notice the highlighting around the control. Now either:

• use the mousewheel or trackpad scroll

• or click and drag up/down.

The initial value and new modulation value are both shown in real time on the tooltip which appears.

• It is also possible to double-click in the outer ring, type a new value in the box which appears and press ENTER/RETURN.

Adjusting the initial value with or without modulation

Move the mouse above the main part of the rotary control - notice it is highlighted.

• Use the mousewheel to adjust the initial value - the modulation depth is preserved.

• Or click and drag up/down.

The initial value is shown in real time on the tooltip which appears.

• Alternatively, double-click and type a new value.

• It is also possible to hold down ALT while clicking and dragging the main part of the rotary control (or scrolling with the mousewheel/trackpad) to adjust the initial value *without* preserving the modulation.

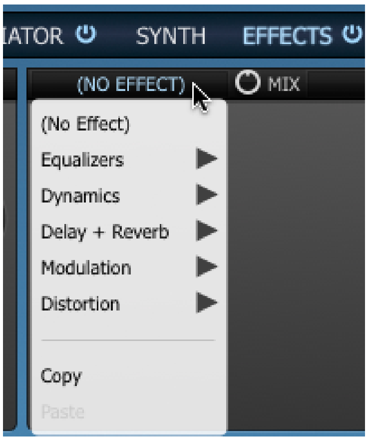
1.4 Other controls and indicators

Rotary selectors and drop-down menus

• Click & drag: click the rotary selector and drag up/down.

• Use the drop-down menu: click on the display that shows the current setting and select the desired setting from the drop-down menu that appears.

Other drop-down menus



Click on the display that shows the current setting and select the desired setting from the drop-down menu that appears.

Numerical text-boxes

There are two ways to adjust these controls:

• Click & drag: click the value and drag it up/down.

• Double-click & type: double-click the value, type a new one and press ENTER/RETURN.

Buttons

Buttons are generally 'toggle' type buttons – click to activate, click again to deactivate. Buttons light up when activated.

Some 'radio button'-style controls also exist, such as the sub-oscillator **Octave** buttons.

Context menus

Many areas of the interface and individual controls within Strobe2 feature a context-menu which is invoked by a right-click (or whichever methods are used for a secondary click on the system, such as a 2-finger tap on a Mac trackpad).

One example is the Parameter context menu which is covered later in this chapter while context menus for various other parts of the Strobe2 interface are mentioned in the relevant parts of this manual.

Indicator LEDs

Strobe2 contains some indicator LEDs such as those at the upper-right of modulators such as the LFO. These indicators represent the current state of the modulator's output in real-time.

1.5 Unit and Snapping modes

Snapping and Unit modes

Controls related to tuning oscillator pitch or filter cutoff frequency offer 4 distinct modes of operation which are accessed via the parameter context menu.

These controls include:

• The Oscillator's 'coarse' **Pitch** and **Sync** controls

• The modulation amount sliders for the Master Pitch from keytracking, LFO and Mod Envelope

• The Filter's **Cutoff** parameter

• The modulation amount sliders for the Cutoff from keytracking, LFO and Mod Envelope

The *Just* and *Harmonic* modes are based on perfect pitch ratios as units, rather than absolute frequency settings.

These modes persist independently for each control. Note that the **Fine** oscillator pitch and **Detune** controls are not affected by snapping modes, which are designed purely for tuning the main oscillator and filter pitch (the latter being especially useful when using self-oscillation at high resonance).

Off

In the *Off* mode, all snapping is deactivated - the control is set in semitones but does not snap to whole semitones.

Just

Just mode uses perfect pitch ratios as units rather than imperfect, equal-tempered pitch. In this mode, pitch or filter cutoff controls are set in harmonics (Hm). There is no snapping to whole harmonics.

Harmonic

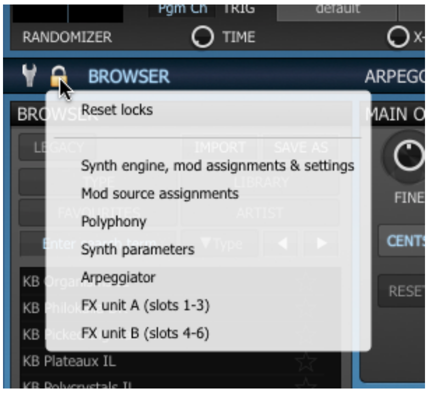
The *Harmonic* mode is similar to *Just*, except that additional snapping to whole harmonics occurs (although SHIFT can be held down for fine control).

Equal

In *Equal* mode controls are set in equal-tempered tuning in semitones and snapping to whole semitones occurs (although SHIFT can be held down for fine control).

1.6 Locks

Locks menu



Strobe2's Locks allow various parts of the state of Strobe2 to remain unaffected when loading new presets. Locks are also very useful when using Quick-preset morphing in situations where very abrupt transitions or clicks may not be desirable.

• Please note that when using Locks, presets may sound very different to how they are intended to sound!

Any currently locked parameters are highlighted in red.

Click the **Locks** button above the Browser area to display a menu showing the various Locks that can be activated or deactivated:

Clear all locks

This function removes all previously activated Locks.

Synth engine, mod assignments & settings

All synth parameters and modulation depths, TransMod slot source and scalar assignments and the following additional controls are locked:

- **Voices**
- **Unison**
- **Bend Up/Bend Down**

Mod source assignments

TransMod slot source and scalar assignments are locked.

Polyphony

The **Voices** and **Unison** settings are locked.

Synth parameters

All synth parameters and modulation depths are locked.

Arpeggiator

The content of the Arpeggiator/Sequencer page is locked.

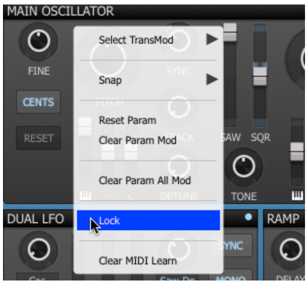
FX unit A (slots 1-3)

The content of the Effects page's FX A chain is locked.

FX unit B (slots 4-6)

The content of the Effects page's FX B chain is locked.

Locking individual controls

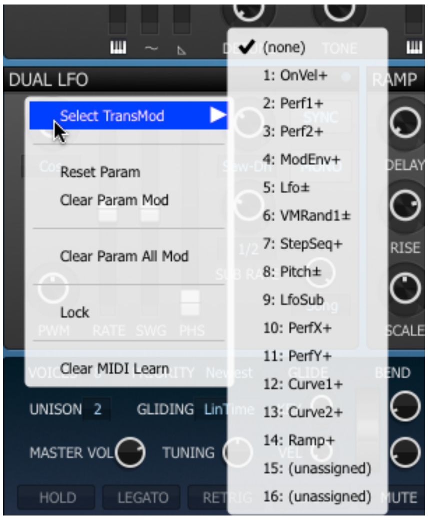


Individual controls can be locked using the Parameter context menu - right-click on a control to display this menu, then use the **Lock** function.

Locked parameters are highlighted in red.

1.7 Parameter context menu

This menu can be invoked with a right-click (or other secondary click) on a parameter or control in Strobe2.



Select TransMod

This sub-menu provides an alternative means of selecting a TransMod modulation slot without having to move the mouse away from a control in the midst of sound design: right-click the parameter, select the desired TransMod slot from the menu and then create TransMod modulation depths on the control (and on any other controls) as desired.

Snap

The **Snap** sub-menu is available for Oscillator **Pitch**, **Sync** and Filter **Cutoff** parameters and specifies the current Unit/ Snapping modes.

Reset Param

This function sets the parameter to its default value.

Clear Param Mod

Use this function to clear any modulation depth that exists for the parameter from the *current TransMod slot*.

Clear Param All Mod

This function clears any modulation depths that exist for the parameter from *all TransMod slots*.

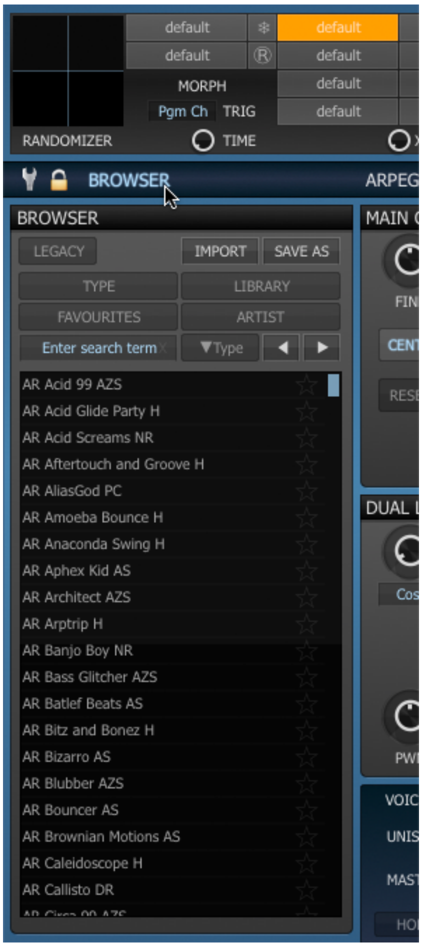
Lock

When activated, the **Lock** function protects the parameter from any changes if a new preset is loaded from the Browser or if a new Quick-preset is selected. This can be very useful for controls which produce abrupt changes or clicks when morphing to a new preset with the Quick-preset controls. Entire sections of parameters can also be locked using the Lock menu.

Clear MIDI Learn

Use the Clear MIDI Learn function to unassign the parameter from any MIDI Learn mapping.

2 Browsing presets



Strobe2's Browser provides quick access to all factory and user presets. It also provides several additional functions for managing and searching for presets.

Show/hide Browser

The Browser can be shown or hidden as needed. To display the Browser if it is not currently visible, click the **BROWSER** button in the upper-left part of the Strobe2 interface. To hide the Browser if it is visible, click the **BROWSER** button again.

Loading and saving Presets

Preset listing

The main part of the Browser area shows the list of available presets. By default all factory-supplied and user-created presets are shown although the list can be narrowed down using the Browser's various filtering and search functions.

• Click any preset in the list to load it.

• To revert to the previous state of Strobe2 before loading the preset, click the **Cancel** button which appears when a preset is modified (or after the initial state of Strobe2 has been modified).

Previous / Next

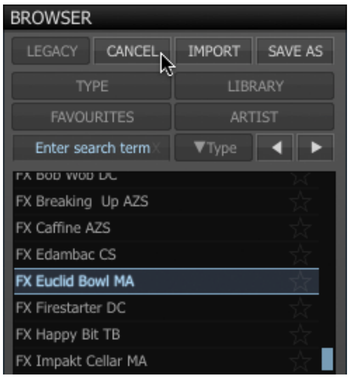
Cycle through the available presets sequentially using the **Previous** and **Next** buttons.

It is also possible to cycle through presets using keyboard cursor-key shortcuts. Note that some DAW/host applications may intercept keyboard input meaning these cannot be used.

↑	Previous preset
↓	Next preset

Import

The **Import** function can load a preset from any location and optionally add it to Strobe's preset database. This may be more convenient than manually copying the preset to the required folder.



Cancel

The **Cancel** button returns to the state of Strobe2 before the last preset was loaded.

Save As

Click the **Save As** button to name and save the current state of Strobe2 as a preset. The saved preset is placed in the user presets location.

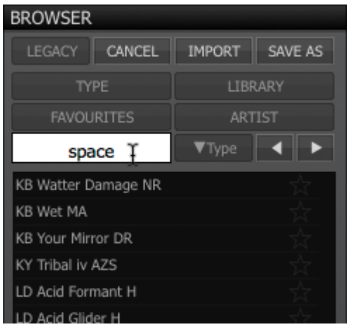
Quick-preset slots

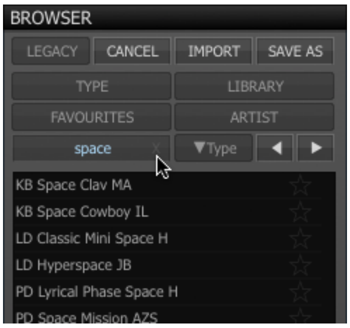
When a preset is loaded from the Browser, it is loaded into the current Quick-preset slot (of which there are 8 available). These slots are explained in the next chapter.

Searching, Sorting and Filtering

The Browser contains a number of additional features for searching or filtering the available presets to find the right sound faster.

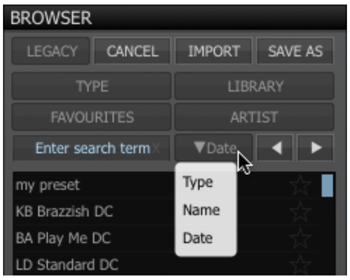
Search





Double-click the **Search** box and enter one or more terms to show only presets whose names match the search term criteria.

Sorting

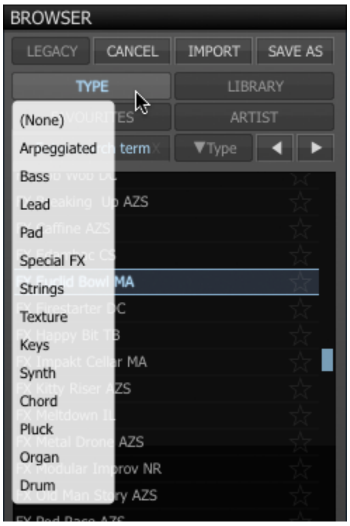


The **Sorting** drop-down menu to the right of the Search box allows the preset listing to be listed by **Type**, by **Name** or by the **Date** it was last saved.

Sorting by **Date** can be very useful as the most recently saved presets are shown at the top of the list.

Filters

The Browser's Filters provide drop-down menus which allow filtering of the preset list according to various criteria.



Type

The **Type** drop-down menu allows the preset list to be filtered to show only 1 of the available preset types. Select *(None)* for the filter to show all preset types. The following preset types are available:

- *(None)*
- *Arpeggiated*
- *Bass*
- *Lead*
- *Special FX*
- *Strings*
- *Texture*
- *Keys*
- *Synth*
- *Chord*
- *Pluck*
- *Organ*
- *Drum*

Library

The **Library** filter allows either *Factory* or *User* presets to be exclusively visible. To show both Factory and User presets in the list, select *(None)* for the filter.

Artist

The **Artist** filter allows presets by a single preset designer to be seen. To show presets by all preset designers, select *(None)* for the filter.

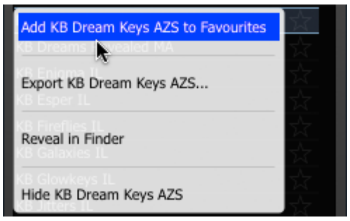
Other controls

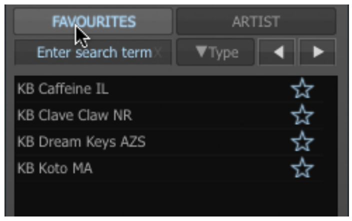
The following buttons toggle between 2 distinct preset views. These views can be combined with search and filter criteria.

Legacy

Activating the **Legacy** button displays only legacy Strobe v1 presets, if DCAM Synth Squad is/was present on the system on which Strobe2 is installed. Deactivate the button to show only Strobe2 presets.

Favourites





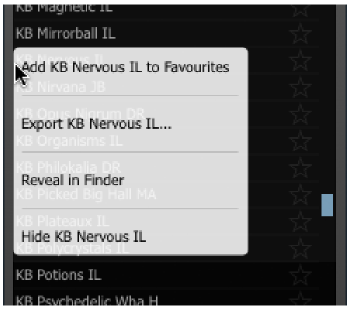
• To mark a preset as a Favourite, activate the  **Favourite** button to the right of the preset name in the preset list by clicking it.

• Alternatively, use the **Add preset to Favourites** function in the Browser context menu (right-click on a preset to display this menu).

• Activating the **Favourites** button displays only presets which have been previously marked as Favourites.

To remove the Favourite status for a preset, deactivate the  by clicking it again or use the **Remove preset from Favourites** function in the Browser context menu.

Browser context menu



Right-click on any preset in the preset list to display the Browser context menu.

Add preset to Favourites
Remove preset from Favourites

These functions are described above.

Export preset...

This function exports the preset to any location on disk. This is useful for sharing a preset without having to navigate to the relevant location on disk to find it.

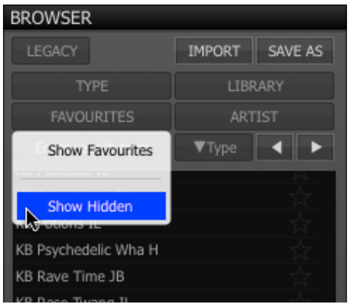
Reveal in Finder... (Mac)
Reveal in Explorer... (Windows)

This function opens a system file window to show the factory or user folder which contains the preset.

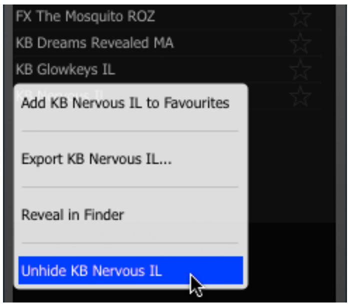
Hide preset
Unhide preset

The Browser provides the ability to hide presets, which can be considered as a counterpart to the Favourites system described above.

To hide a preset from the preset list, use the **Hide preset** function on the Browser context menu. Hidden presets are not deleted - they can be viewed and unhidden.



To view all currently hidden presets, right-click the **Favourites** button and use the **Show Hidden** function on the menu that appears.



Right-click on any hidden preset and use the **Unhide preset** function.

Factory and User preset locations

- Factory presets are stored within the following locations:
  - **Mac OSX:** /Library/Application Support/FXpansion/Strobe2/Presets
  - **Windows:** C:\Program Files (x86)\FXpansion\Strobe2\Presets
- User presets are stored within the following locations:
  - **Mac OSX:** /Users/<your username>/Documents/FXpansion/Strobe2/Presets
  - **Windows:** C:\Users\[Username]\Documents\FXpansion\Strobe2\Presets

Sub-folders within the factory and user preset folders are used as Artist names for the presets within them - use the **Artist** filter to show only the presets within any sub-folder.

3 Synthesis Engine

This chapter describes Strobe2's synthesis sections:

Oscillator Section

Strobe2's Oscillator can be considered as a 'super-oscillator' designed to produce a wide variety of rich oscillator timbres to feed into the Filter section for sculpting. It contains several sub-sections, each of which can be switched on or off via their **Power** buttons.

Main Oscillator

• The **Pitch** and **Fine** controls set the oscillator tuning

• The levels of the main oscillator waveforms are set with the **Saw** and **Sqr** (Square) faders

• The Square waveform features control over its pulse width with the **PWM** parameter

• Faders are provided for setting the amount of direct **Pitch** and **PWM** modulation from the **Keyboard**, **LFO** and **Mod Envelope**

• The **Sync** control increases the pitch of the Main oscillator while forcing it to synchronize to the Master Pitch set by the **Pitch** and **Fine** controls, leading to classic 'hard sync' sounds with added harmonic content

• The **Stack**/**Detune** functions provide classic detuned osc and 'super-saw' sounds and can even produce chords

• The **Reset** (Phase Reset) function allows exactly repeatable osc tones, especially useful for bass sounds

• The **Cents** button sets the detuning mode - with the button deactivated, detuning functions occur after keyboard scaling has been applied, leading to uniform 'beating rates' across the keyboard

Sub Oscillator

The Sub Oscillator is intended to provide a solid low-end foundation for sounds.

• With the **Link** button deactivated, the Sub-osc is not affected by the Main oscillator's **Stack**, **Detune** or **Sync** functions

• A variety of waveform faders are available, with each capable of being tuned 0, 1, 2 or 3 octaves lower than the Main oscillator **Pitch**

• The Square waveform features control over **PWM**

• All Sub-osc waveforms are affected by the **Shape** control which applies a waveshaping function for more complex harmonics in the output

Noise generator

The Oscillator section is completed with a white noise source - the **Noise** fader sets its level in the Oscillator mix.

Tone controls

Each of the above 3 sections contains its own **Tone** control - this is a keyboard-tracking EQ function which is useful for emphasizing or de-emphasizing part of the frequency range before the more radical tonal sculpting provided by the Filter section. Altering the spectral balance in this way can affect how the subsequent filter's resonance behaves over the frequency range, changing the character of filter sweeps.

V.C.F. (Voltage Controlled Filter) Section

The V.C.F. section represents the 'subtractive' part of the term 'subtractive synthesis' - it subtracts parts of the frequency spectrum from the oscillator signal.

• Strobe2's filter model actually utilizes 'pole-mixing' techniques such as those used in the Oberheim Xpander/Matrix 12 synthesizers to create a huge range of unusual filtering responses, accessible via the **Mode** control

• The filter's **Drive** function allows a large amount of tonal variation and is gain-compensated - as it is increased, the output level of the filter is reduced in order to keep levels constant at varying **Drive** settings

• Faders are provided for setting the amount of direct filter **Cutoff** modulation from the **Keyboard**, **LFO** and **Mod Envelope**.

V.C.A. (Voltage Controlled Amplifier) Section

A synthesizer's V.C.A. performs the task of 'articulating' the amplitude of each synth voice. The oscillator is always running internally - the V.C.A. uses a control signal from an envelope to shape the amplitude (or 'loudness') of each voice when notes are played.

• Strobe2's V.C.A. is *always* affected by its Amp Envelope.

• The V.C.A. circuit model provides a saturating overdrive effect at higher settings of the **Amp** control. Reduce the **Level** control to compensate for the gain increase - otherwise the output may clip.

• For cleaner sounds, reduce the **Amp** control and increase the **Level** control

• The **Analogue** control introduces noise and mains hum within Strobe2's control circuitry, resulting in the synth reacting with a more noisy and organic character

Visualizer Scope

The Scope area performs a number of functions in Strobe2.

By default (with the **SCOPE** button activated as shown on the left), this mode performs 2 functions:

1. It displays the Strobe2 logo graphic and a level meter representing the final audio output.

2. When the mouse is moved above or used to adjust certain controls, the Scope area switches to the Visualizer display.

The Visualizer provides a context-sensitive graphical representation of various parts of the Strobe2 synthesis engine. Note that it should not be regarded as an oscilloscope, but rather as an aid to visualising each part of the synth.

For example, when the mouse is used to adjust a control in the Oscillator section, the final waveshape is displayed in the Visualizer.

Using the Filter section results in displaying the current **Mode**'s filter response curve.

Elsewhere, the various parts of the interface feature different Visualizer displays - modulators such as LFOs and envelopes display their respective shapes, for example.

The Visualizer display is updated in real time as any relevant controls are adjusted and also shows the effect of any modulation routed to relevant parameters.

Activating the **EUCLID**/**CURV1**/**CURV2**/**ZONE1**/**ZONE2** buttons in the Scope area displays the corresponding Euclid/Curve/KeyZone processor editor (see below).

Other functions on the Synth page

This chapter covers only functions directly related to the audio synthesis engine. The Synth page also contains a number of editing functions for modulators in Strobe2, which are described in the Modulation chapter:

Dual LFO

Strobe2's full-featured LFO includes a secondary 'sub LFO' which is derived by multiplying/dividing the main LFO rate. The main LFO features direct modulation routings to the Filter **Cutoff** as well as oscillator **Pitch** and **PWM**. The main and sub LFOs can be routed to almost any other parameters in Strobe2 via the TransMod system.

Ramp generator

This modulator provides a variety of uses via the TransMod modulation system and can be optionally looped to provide a polyphonic saw-wave LFO.

Mod Envelope

This envelope features direct routings to the Filter **Cutoff** as well as oscillator **Pitch** and **PWM**. It can also be routed to almost any other parameters in Strobe2 via the TransMod system.

Amp Envelope

This envelope is always directly routed to Strobe2's **Amp** parameter with 100% depth. It can also be routed to almost any other parameters in Strobe2 via the TransMod system.

The Scope area provides 3 additional modulator editors:

Euclid processor

This processor is a combination of an X-Y pad with additional modulation processing with inertia and slew.

Curve processors

These editors provide the ability to remap, quantize and slew the shape of a modulation source.

KeyZone processors

These editors allow custom keytracking responses to be defined.

3.1 Oscillator section

Master Oscillator controls

Pitch, Fine

These controls set the Master Pitch for both the Main Oscillator and the Sub Oscillator.

The **Pitch** control adjusts tuning in semitones while the **Fine** control offers an additional offset of +/- 1 semitone (except when the **Cents** button is deactivated - see below).

The Pitch control can be set in semitones or in just harmonics with optional snapping to whole semitones/ harmonics depending on the current Unit/Snapping mode.

Modulating the Fine control by small amounts with random TransMod modulation sources is useful for creating effects reminiscent of oscillators with unstable tuning.

Please do not confuse the concept of Master Pitch in Strobe2 with the **Tune** (Master Tuning) control in the Global synth controls section. The Tune control sets the overall tuning by specifying the frequency the A note above middle C between 420 and 460 Hz (the default is 440 Hz). The **Pitch** and **Fine** controls used to adjust the Master Pitch are set in semitones and applied *relative to* the **Tune** control setting.

Direct modulation

Osc Pitch can be modulated directly using the following dedicated modulation depth controls:

Keytracking

LFO

Mod Envelope

These controls are affected by the current Unit/Snapping mode.

Reset (Phase Reset)

When the **Reset** button is enabled, the phase of the oscillator is reset to zero when a new voice is played. This is useful when each note onset is required to be consistent rather than with free-running phase - the latter can lead to audible variation which can be undesirable for some types of bass sounds.

Main Oscillator controls

Main Oscillator Power

Click the Main Oscillator's **Power** button to activate/deactivate it.

Saw
Sqr (Square)

These controls represent the levels of the oscillator's **Saw** and **Square** (with variable pulse width) waveforms.

PWM

The Square waveform features variable pulse width which is adjusted using the **PWM** control.

Direct modulation

The main osc's pulse width can be modulated directly with the following modulation depth controls:

Keytracking

LFO

Mod Envelope

Note that these functions only modulate the Pulse Width of the main oscillator's Square wave - the sub osc's Square wave features its own **PWM** control which must be modulated by the TransMod modulation system.

Sync

The Strobe2 oscillator features a 'hard sync' function: increasing the **Sync** control increases the frequency of the main oscillator while always re-synchronizing it on each cycle of the Master Pitch set by the **Pitch**/**Fine** controls. The sub oscillator is also affected by this function (unless the sub osc's **Link** button is deactivated).

The resulting waveform has the same overall pitch as the Master Pitch setting but with added complex harmonics that can create a wide range of timbres and effects, especially when the **Sync** control and other oscillator parameters are modulated.

The Sync control can be set in semitones or just harmonics with optional snapping to whole semitones/ harmonics depending on the current Unit/Snapping mode.

Stack
Detune

These controls provide an oscillator-stacking function which creates classic multi-osc and detuned unison-style sounds using Strobe2's single oscillator - without needing to use additional unison voices.

The **Stack** control sets the number of stacked oscillators which are added above and below the Master Pitch setting as follows:

2nd: Higher
3rd: Lower
4th: Higher
5th: Lower

The **Detune** control spreads these stacked oscillators away from the Master Pitch, creating a detuning effect. The amount of detuning depends on the state of the Cents button:

• With Cents activated, stacked oscillators are spread up to 1 octave away from the Master Pitch

For a 'detuned supersaw' sound, increase the Stack control to 2 or 3 (or any number up to 5!) and then set the Detune control to taste.

To create chord-style tunings, try setting the Detune control to 3, 5 or 7 semitones with the Stack control at 2 or 3.

• With Cents deactivated, stacked oscillators are spread up to 16 Hz away from the Master Pitch

This mode results in a smaller effective detune spread than with Cents activated. However, detuning occurs with a constant rate of beating across the keyboard range (see below).

Cents (Detune mode)

The **Cents** button dictates if any Oscillator detuning functions are applied in terms of absolute pitch (cents of a semi-tone) or in terms of frequency (hertz). The state of this button has an effect on the 'beating' that occurs between detuned oscillators when using Unison voices and detuning the **Fine** control or when using the **Stack** and **Detune** controls.

When the **Cents** button is activated, an absolute pitch deviation in semitones and cents (1% of a semitone) is added to each note on the keyboard *before* keyboard tracking is applied to the oscillator pitch. This leads to a changing 'beating' rate across the keyboard (the higher the note, the faster the beat rate).

When the button is deactivated, the **Fine** pitch control provides a range of +/- 100 Hz, while the **Detune** control provides up to 16Hz of detuning away from the Master Pitch. The specified frequency is added for each note *after* keyboard tracking is applied to the oscillator pitch. This means that using detuning techniques such as those described above leads to uniform 'beating' rates across the entire keyboard range.

Tone

The **Tone** filters within the main Osc, Sub-Osc and Noise sections adjust the spectral emphasis of each of these sections before they are mixed and fed into the Filter. These filters are shelf-style EQ filters for emphasizing or de-emphasizing higher harmonics in the signal before the more radical sculpting offered by the Filter section. Changing the spectral balance in this way can alter the way the filter resonance behaves across the frequency range, giving **Cutoff** sweeps a different character.

Tone filters always track the keyboard so their effect is always consistent throughout the keyboard range, with the centre of the shelf's slope positioned 2 octaves higher than the osc's Pitch setting. Each **Tone** control itself sets the amount of gain (increase or decrease) for the keytracking shelf EQ.

Sub-oscillator controls

Sub-osc Power

Click the Sub Oscillator's **Power** button to activate/deactivate it.

Note that even with the Sub Oscillator deactivated, the Main Oscillator's **Sync** function is still active as normal.

Sub Osc Link

With the **Sub Osc Link** button activated, the Sub Osc is affected by the Main Osc's **Sync**, **Stack** and **Detune** functions. With the button deactivated, the Sub Osc tuning is not affected by these functions. Deactivating this button is useful for heavily-detuned sounds anchored by a tuned sub.

Sine (Sin)
Triangle (Tri)
Saw
Square (Sqr)

These controls represent the levels of the Sub Oscillator's **Sine**, **Triangle**, **Saw** and **Square** waveforms.

The Square waveform is actually a Pulse waveform with adjustable Pulse Width.

Octave

Each of the sub-osc waveforms can be set to the same pitch as the main osc or 1, 2 or 3 octaves below it using the **Octave** buttons.

PWM

This control adjusts the Pulse Width of the sub-oscillator's Square waveform.

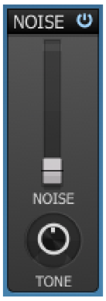
Shape

The **Shape** control adjusts a waveshaper function on the combined mix of the Sub Oscillator's waveforms, resulting in varied timbres due to the added complex harmonics.

Tone

The Sub Oscillator's **Tone** control behaves identically to the Tone control for the Main Oscillator (see above).

Noise Generator controls



Noise Generator Power

 Click the Noise Generator's **Power** button to activate/deactivate it.

Noise

The **Noise** control adjusts the level of Strobe2's Gaussian white noise generator.

Tone

The noise generator's **Tone** control behaves identically to the Tone control for the Main Oscillator (see above).

3.2 Filter (V.C.F.) section

Strobe2's filter is its main source of tonal variation, especially when modulated. While there are direct modulation routings to the filter frequency cutoff from keyboard pitch, LFO and Mod Envelope, the filter's controls can be modulated with the TransMod system for creative and varied effects which go beyond these basic modulation routings.

The filter model is based on an OTA (operational transconductance amplifier) cascaded core, with a diode-based clipper circuit in the feedback section. The modelled diodes in the circuit are slightly mismatched, leading to the characteristic growl of a real analogue filter.

Filter Power

Click the Filter's **Power** button to activate/deactivate it.

When the filter is deactivated, the audio from the Oscillator section passes through the filter unaffected.

Cutoff
Res (Resonance)

The **Cutoff** control determines the cutoff frequency of Strobe2's filter.

The **Res** control adds emphasis around the cutoff frequency. At extreme Res settings, the filter circuit exhibits self-oscillating behaviour.

The Cutoff control is adjusted in pitch-based units, allowing it to be tuned just like an oscillator - this can be especially useful when the filter is self-oscillating and effectively behaving as a sine oscillator. It can be set in semitones or just harmonics with optional snapping to whole semitones/harmonics depending on the current Unit/Snapping mode.

Direct modulation

The **Cutoff** parameter can be modulated directly using the following dedicated modulation depth controls:

Keytracking

LFO

Mod Envelope

Setting the Keytracking depth control to maximum allows the filter to be played musically from the keyboard.

These controls are affected by the current Unit/Snapping mode.

Drive

The **Drive** control increases the gain of the signal going into the filter. This overloads the filter's components, drastically changing its sound and character.

This Drive control is gain-compensated - as it is increased, the output level of the filter is reduced so that, while the timbre changes, the level stays constant, meaning that it is unneccessary to compensate increases in Drive by reducing the V.C.A. **Amp** parameter (and vice versa).

Note that the effective resonance of the filter circuit is reduced as the Drive amount is increased.

Leak

The **Leak** control allows an adjustable amount of the original signal through, mixed with the filtered output, simulating the effect of a bleeding filter circuit. It allows further timbral options even with the more complex filter **Mode** settings.

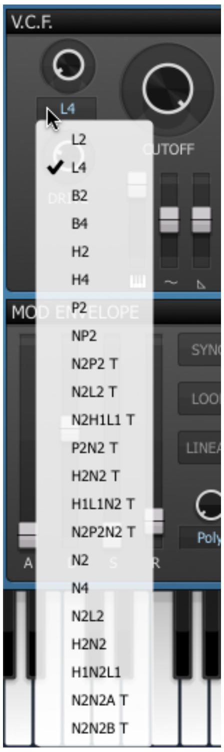
Mode

A large variety of filter modes are available by adjusting the **Mode** control.

Note that filters denoted as 'T' feature 2 or more filters with an octave between the frequency of each. The first filter in the name is at the cutoff frequency, with subsequent filters each an octave higher than the last. The only exceptions are the last 2 filter modes - see the table below.

In other combination filters, all filters act at the cutoff frequency.

See <u>later in this manual</u> for graphical representations of the frequency response curves for each Mode.



L2, L4	2-pole (12 dB/oct.) & 4-pole (24 dB/oct.) Low-pass filters
B2, B4	2-pole & 4-pole Band-pass filters
H2, H4	2-pole & 4-pole High-pass filters
P2	2-pole Peak filter
NP2	1-pole Notch and 2-pole Peak filters
N2P2 T	2-pole Notch and 2-pole Peak filters, an octave apart
N2L2 T	2-pole Notch and 2-pole Low-pass filters, an octave apart
N2H1L1 T	2-pole Notch, 1-pole High-pass and 1-pole Low-pass filters, each an octave apart
P2N2 T	2-pole Peak and 2-pole Notch filters, an octave apart
H2N2 T	2-pole High-pass and 2-pole Notch filters, an octave apart
H1L1N2 T	1-pole High-pass, 1-pole Low-pass and 2-pole Notch filters, each an octave apart
N2P2N2 T	2-pole Notch, 2-pole Peak and 2-pole Notch filters, each an octave apart
N2, N4	2-pole & 4-pole Notch filters
N2L2	2-pole Notch and 2-pole Low-pass filters
H2N2	2-pole High-pass and 2-pole Notch filters
H1N2L1	1-pole High-pass, 2-pole Notch and 1-pole Low-pass, each an octave apart
N2N2A T	2-pole Notch with 2-pole Notch an octave higher than the cutoff frequency
N2N2B T	2-pole Notch with 2-pole Notch an octave lower than the cutoff frequency

3.3 Amp (V.C.A.) section

The Amp section represents the final output of the synth circuit before it enters the FX processing blocks. It articulates each synth voice in terms of its output level and its position within the stereo field.

Amp

The **Amp** parameter is always directly modulated by the Amp Envelope - its value represents the amplitude at the maximum value of the AmpEnv (at the end of its attack stage).

The Amp section features a modelled VCA circuit which can be overloaded like that of a real analogue synth VCA (voltage-controlled amplifier) resulting in non-linear distortion characteristics.

For a cleaner sound, keep the **Amp** control at low settings and increase the **Level**.

When overloading the V.C.A. by increasing the Amp parameter, remember to turn down the Level control. Otherwise the output of Strobe2 may clip.

In order to achieve a velocity-sensitivity response for amplitude (so that harder velocities result in a louder sound), modulate the **Amp** parameter with velocity (the OnVel+ source) using the TransMod system - this is already set up in TransMod slot 1 in the Strobe2 Init preset.

Pan

The **Pan** parameter sets the position of the voice in the stereo field.

Try modulating this control with the TransMod system using a Voice or Unison modulation source – this leads to a rich stereo spread of chords or multiple unison voices.

Level

This parameter sets the final **Level** of each voice before it is summed with all other active voices. If this parameter's value is set too high, the output of Strobe2 entering the Effects section may clip.

Suitable settings for this parameter depend upon context - care should be taken especially with polyphonic sounds to allow enough headroom for the output not to clip when multiple voices are mixed.

Please note that this parameter is *monophonic* and sets the level for all active voices - to achieve amplitude modulation per-voice, modulate the **Amp** control rather than the **Level**.

Analogue

The **Analogue** parameter simulates the effect of noise and mains hum in certain parts of the audio and control signal paths, something that always occurs in real analogue synth circuits. At lower settings, it leads to a subtly gritty and slurring character, while higher settings create a more unstable and noisy sound.

4 Global Synth Controls

This section of the interface is always visible, even when viewing the Arpeggiator or Effects pages. It is used for various global settings and controls.

Polyphony settings

The **Voices** and **Unison** settings are covered in the next section.

Keyboard input, Glide and Retrigger controls

The **Priority**, **Gliding**, **Key Glide**, **Vel Glide**, **Legato** and **Retrigger** controls are involved in adjusting various aspects of how Strobe2 responds to keyboard input. These are covered later in the chapter.

MIDI CC Learn mode

Activating the **CC LRN** button initiates MIDI CC Learn mode which is used to assign Strobe2 controls to external MIDI CC controller messages.

Keyboard and Transpose buttons

The on-screen keyboard allows playing notes with no hardware keyboard available. It is useful for verifying that Strobe2 is operating correctly.

The **Transpose** buttons adjust the MIDI note range shwon within the 4 visible octaves.

Bend (Pitch Bend)

This on-screen control represents the MIDI pitch bend controller - it is always assigned to the MIDI pitch bend controller messages and is provided on the interface for creating these messages with no hardware pitch-wheel available. MIDI pitch bend messages are fed into a dedicated TransMod source (MnBend) which remains active even if the **Bend Up** / **Bend Dn** controls are set to 0.

Bend Up
Bend Dn (Down)

These controls set the pitch bend sensitivity – the amount of pitch modulation from MIDI pitch bend messages (and the on-screen **Pitch Bend** control described above). The amount of upward and downward pitch displacement can be specified separately, to a maximum of 12 semitones in each direction.

P1 (Perf1)
P2 (Perf2)

These controls represent the 2 available performance controllers - the presence of these controllers is due to most users typically having access to 2 expressive controllers when playing a keyboard - for example, mod-wheel and aftertouch. To map these controls to the hardware controllers available to you, assign them using MIDI CC Learn mode and then use the **Save preferences** function in the Preferences menu.

Master Vol (Volume)

This control sets the final output level of Strobe2, after the Effects stages. It allows the volume to be boosted by up to 12dB or attenuated to -∞ dB.

Tune (Master Tuning)

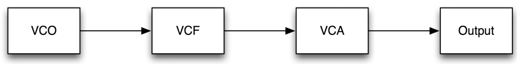
This control sets the master tuning - it specifies the frequency for the A note above middle C. The default is 440 Hz and is adjustable between 420 and 460 Hz.

Mute

The **Mute** button shuts down any playing notes/voices - it can be considered as a 'panic' button.

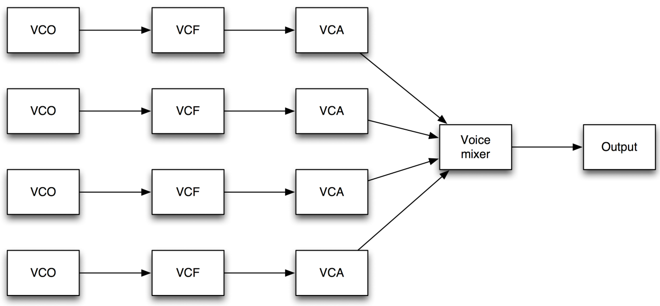
4.1 Polyphony: Voice and Unison settings

Polyphony overview



A synth voice can be considered to be an entire monophonic synth that plays a single note at a time – a monophonic synth features 1 voice.

• Monosynth architecture



• Polysynth architecture

A polyphonic synth features multiple voices, meaning that it is comprised of multiple identical monosynths with a simple logic circuit for distributing received notes amongst them.

The most obvious example of the latter is the original family of Oberheim 2-voice / 4-voice / 8-voice polyphonic synths. These were simply multiple discrete Oberheim SEM units housed together with note-distribution and some other global programming functions.

While the original Oberheim polysynths are perhaps the most obvious examples for visualizing the architecture of a polyphonic synth, this architecture is common to all conventional polyphonic analogue synths. Usually it is implemented in the form of 'voice cards' which contain the circuitry for each individual voice along with a single control panel which distributes parameter edits to the voice cards.

The same principles are used within Strobe2.

Voices
Unison

These numerical text-boxes set the number of active **Voices** for the current patch and the number of **Unison** voices if required.

If more simultaneous notes are played than the current voice and unison settings allow, voice-stealing functions are applied, the behaviour of which is specified by the **Priority** control (see the next section).

Voices

With Unison set to 1, the **Voices** setting represents the maximum number of simultaneous notes (polyphony) that can be played.

However, when using a **Unison** setting higher than 1, the polyphony is the number of Voices divided by the Unison setting.

Unison

The number of **Unison** voices is a sub-set of the maximum number of voices - it determines how many available voices to stack for each note. For example:

• specifying 8 voices and 2 unison voices results in 4-note polyphony, with each note comprising 2 stacked unison voices

• specifying 12 voices and 3 unison voices also results in 4-note polyphony, with each note comprising 3 stacked unison voices

Voices, Unison and TransMod modulation

In traditional analogue synths, unison voices could usually only be detuned against each other. The possibilities in Strobe2 are far more advanced - using the polyphonic Unison TransMod sources in Strobe2, any synth parameter - including but not limited to oscillator pitch - can be spread across a range for each unison voice. Similarly, a number of Voice TransMod sources are also provided in order to create per-voice modulation with regular polyphonic playing (1 voice per note).

4.2 Keyboard Input, Glide and Retrigger controls

Keyboard input settings

Priority

The **Priority** control dictates the voice stealing priority if more notes are played than the number of voices available and can be set to *Newest*, *Oldest*, *Highest*, *Lowest*, *Hardest* or *Softest*.

The default setting is *Newest*, which means that new notes are prioritized over old notes. If **Voices** is set to 1 (see the last section for details of this setting), playing a note while another note (the older note) is already playing results in the new note 'stealing' the voice as it has a higher priority. When this new note is released, the older note is played again.

The *Lowest* setting results in the lowest played note always being given priority - this is similar behaviour to that in the original Minimoog. The *Hardest* and *Softest* settings give priority to higher and lower velocity notes respectively.

Hold

Engaging the **Hold** button results in incoming MIDI note *off* messages being ignored until the button is deactivated. Therefore, if the Hold button is enabled, any keys already playing or subsequently played are sustained indefinitely, even after the keys are released.

Please remember that held notes are not saved with presets!

Glide and Retrigger settings

Classic analogue synth glide essentially smooths transitions between keyboard-input note intervals, leading to the typical 'portamento'-style rising or falling pitch effects.

Strobe2 can additionally apply Glide to Velocity TransMod modulation. When activated, any modulation which is derived from the OnVel+ source (for example, in the first TransMod slot in the Strobe2 factory presets) is smoothed over time with new incoming events.

Note that glide behaviour is also affected by the **Legato** button (see below).

Key (Pitch Glide)

The **Key Glide** control sets the glide time towards the pitch of new note events.

The control can be modulated with the TransMod system - try using a VRand source to randomize the glide time on each note!

Vel (Velocity Glide)

The **Velocity Glide** control sets the glide time towards new MIDI note velocity (OnVel+ TransMod source) modulation depths on each key-on event.

Since note velocity is perhaps the most immediate performance control method, Velocity Glide offers an easy way of injecting additional variation and dynamic response to a synth performance. It is also polyphonic so occurs independently on each played voice.

For example, if the OnVel+ TransMod source is set to modulate the filter's **Cutoff** parameter so that it increases with higher velocity, playing a harder note after a soft note would cause the cutoff frequency to glide to the higher value over the **Velocity Glide** time period. If the Velocity Glide control is set to zero, the cutoff frequency is fully modulated by a new note as soon as it is played.

Like the **Key Glide** control, the **Velocity Glide** control can itself be modulated by the TransMod modulation system. Again, a VRand source can be very effective, or try a Unison source in a unison-enabled patch so that each playing voice features varying glide times.

Gliding (Glide Mode)

The **Gliding** control changes the glide time response between Linear (*LinTime*, *LinRate*) and Exponential (*Exp1*, *Exp2*) settings. Each setting results in a different shape and resulting playing feel for the 'curve' of glide transitions.

Legato
Retrig

These buttons provide further options for Glide and modulator retriggering.

The **Legato** button relates to glide: activating it results in a 'fingered' glide – meaning that glides only occur when 2 notes 'overlap'. This applies to both Velocity Glide and Pitch Glide.

The **Retrig** button forces gated modulators (Ramp/LFO/ModEnv/AmpEnv) to retrigger when voice stealing occurs. This is especially useful with monosynth patches.

4.3 MIDI CC Learn mode

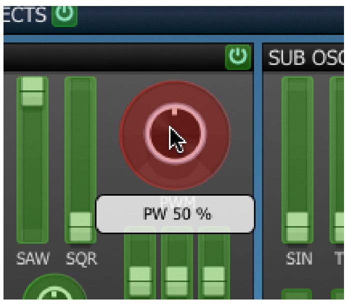
Creating MIDI CC assignments

The **CC LRN** button provides access to Strobe2's MIDI CC Learn mode, which allows parameters to be assigned to hardware MIDI CC controllers.

1

Click the **CC LRN** button to enter MIDI CC Learn mode. Parameters in Strobe2 which can be mapped to MIDI CCs are highlighted in green.

2

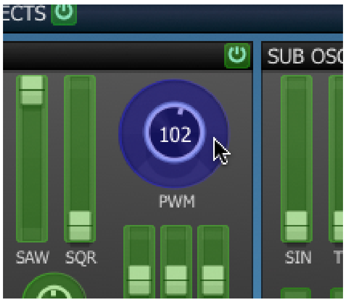


Click the parameter to map on the interface.

3

Move the desired physical MIDI CC knob, slider or other controller.

4



The parameter is now mapped to the controller. The MIDI CC number is overlaid on the control.

5

Click the **CC LRN** button again to exit MIDI CC Learn mode.

Removing MIDI CC assignments

To remove MIDI CC Learn assignments, right-click on the desired control to display the Parameter context menu and use the **Clear MIDI Learn** function.

This operation can be performed whether MIDI CC Learn mode is currently activated or deactivated.

5 Modulation

Strobe2 features 2 ways of modulating parameters:

• Direct modulation of certain important parameters by keytracking, LFO and Mod Envelope

• TransMod modulation, which offers a hugely increased array of modulation possibilities

1. Direct modulation of Osc Pitch, PWM, V.C.F. Cutoff and V.C.A. Amp parameters

Strobe2 features direct modulation routings and depth controls for selected destination parameters - Oscillator **Pitch**, main oscillator Square wave **PWM** and Filter **Cutoff** - from the following sources:

Keytracking

This source tracks the keyboard (MIDI input note range).

LFO

This source allows modulation from Strobe2's LFO.

Mod Envelope

This source allows modulation from Strobe2's Mod Envelope.

Amp Envelope to V.C.A. Amp parameter

This is a direct modulation routing which is always active - see below.

Basic functional modulation

Modulation can be considered as the movement of a parameter over time. In an initialized Strobe2 state (i.e. with the Init preset loaded), despite the simple output of the synth, several types of modulation are already active:

Amp envelope to VCA Amp

This modulation routing is <u>always active with 100% depth</u> - this means that the Amp Envelope always acts to open the VCA to allow Strobe2's output to be heard.

Keytracking to Osc Pitch

The **Osc Keytrack** slider underneath the Oscillator's **Pitch** control has a setting of *+100%* by default. This means that MIDI note input produces chromatic pitches from the oscillator.

Keytracking to Filter Cutoff

The **Filter Keytrack** slider underneath the Filter's **Cutoff** control has a setting of +100% by default. This means that Strobe2's output remains harmonically constant throughout the MIDI note range and produces musical chromatic pitch when in a self-oscillating state (with the **Resonance** turned up to a high setting).

Using an envelope or LFO to add variation

Using an envelope to shape the timbre over time: Envelope to Filter Cutoff

1. If the Filter **Cutoff** is currently turned up, turn it down so that the signal from the oscillator is filtered.

2. Increase the **Envelope** slider underneath the Filter **Cutoff** control.

Using the LFO to add vibrato: LFO to Osc Pitch

1. Increase the **LFO** slider underneath the Oscillator **Pitch** control.

Delaying an envelope

1. Set the Mod Envelope's **Gate source** to *Ramp* instead of *Poly*.
2. Deactivate the **Sync** and **Loop** buttons on the Ramp modulator if they are activated and make sure its **Gate mode** is set to *Poly*.
3. Increase the **Delay** time on the Ramp modulator to around *500ms*.

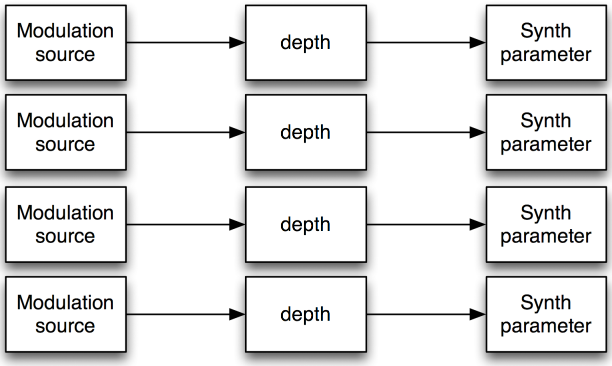
The Mod Envelope is now triggered around 500ms after each MIDI note is received.

See the next section in this chapter for more details of modulator Gate modes.

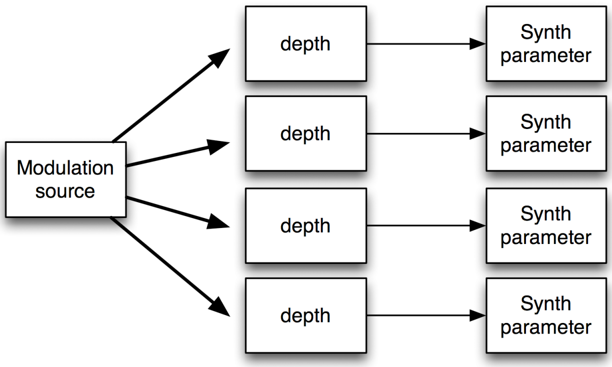
2. TransMod modulation

While the above direct modulation routings allow quick programming of many classic analog-style sounds, Strobe2 also features the TransMod modulation system which provides hugely increased versatility and complex modulation effects.

A wide variety of modulation sources is provided, from variations of the built-in modulators to numerous other monophonic and polyphonic modulation sources.



Traditional modulation matrix



Strobe 2 TransMod system

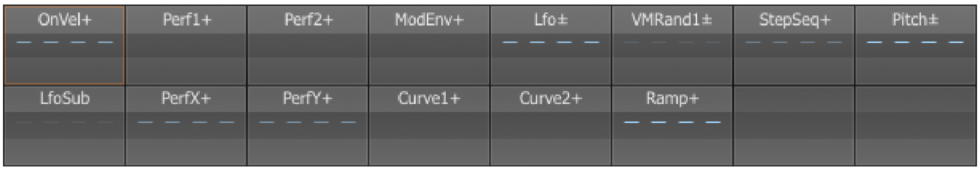
Essentially, TransMod is similar to the well-known 'mod-matrix' found in many synthesizers. However, instead of simply connecting a modulation source to a single synthesis parameter and setting the depth or amount of modulation, TransMod allows a single modulation source to be routed to multiple synth parameters with varying depths for each routing.

The above diagram illustrates how a single modulation source, such as an LFO, is routed to destination parameters in a traditional mod-matrix and in DCAM: Synth Squad's TransMod system:

The TransMod modulation system is covered in depth in the next section.

5.1 Using TransMod modulation

The TransMod modulation system centres around the 16 TransMod 'slots' at the top part of the Strobe2 interface.

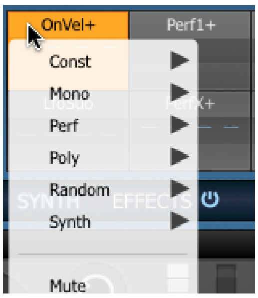


• The default setup for the 16 TransMod slots in Strobe2's Init (initialized) preset

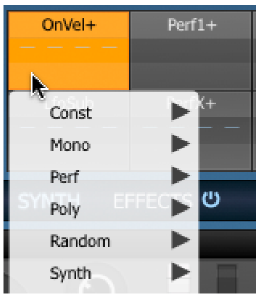
With no slot selected, adjusting Strobe2's parameters adjusts the 'initial state' of the synth - before any TransMod modulation is applied. Note that, as mentioned previously, Strobe2 contains certain direct modulation routings that may already be active within this 'initial state'.

Selecting a TransMod slot

To begin using the TransMod system, a TransMod slot must first be selected.

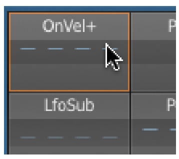


• click on the *upper* part of the slot to select a slot and display its TransMod source menu.



• click on the *lower* part of the slot to select a slot and display its TransMod scalar menu - these are explained later in this section.

• to select a slot without displaying any menus, click the *central* part of the slot.



• to deselect all TransMod slots, click the selected slot's *central* part again.

TransMod slots can also be selected via the Parameter context menu or via the following keyboard shortcuts (in DAW/host applications which do not intercept keyboard input):

TAB or **]**	Next TransMod slot
[	Previous TransMod slot

When a TransMod slot is selected Strobe2's parameters display any modulation 'amounts' that may exist from the slot to each control. The modulation amounts can be edited on the controls themselves using the outer ring for rotary controls or the slider path for vertical sliders.

With no TransMod slots selected, editing *only* occurs on the initial state of Strobe2's parameters.

Creating modulation amounts

Once a TransMod slot is selected, amounts of modulation can be created directly on Strobe2's parameters, as shown earlier in this manual.

These modulation amounts or 'depths' represent the greatest amount of movement of the parameter away from its initial value when the TransMod slot's modulator sends out a maximum value.

For 'continuous' modulation sources - such as envelopes, LFOs, and so on - the modulation depths represent the 'travel' of movement over time. While the modulation source varies over time, the values of any modulated parameters change proportionally within the defined range according to the values they receive from the modulation source.

For other modulation sources - such as Random, Voice and Unison sources - the modulation depths represent a range throughout which values are distributed or 'spread' at the initiation of each voice.

Compared to a traditional mod matrix, the TransMod system offers some advantages. It is easier to keep track of large numbers of modulated parameters. There is also no abstracted, table-based relationship between depths and parameters – all depths are shown visually, directly on the parameters themselves using the dual-action controls.

Almost all synth parameters can be modulated using the TransMod system. The only exceptions are the following:

• buttons (including Sub-oscillator **Octave** controls)

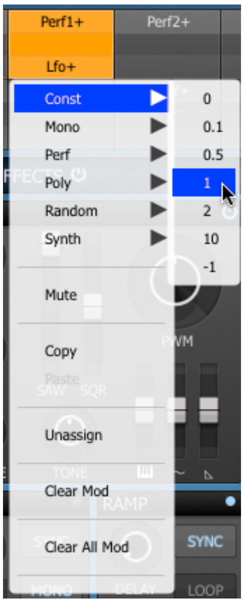
• **Gate source** controls for gated Modulators (LFO, Envs etc)

• all parameters in the Global controls section except the **Vel Glide** and **Pitch Glide** controls

• Quick-preset and morphing controls

Scaling

Each TransMod slot features a multiplying scalar in the lower part of the slot in addition to a source in the upper part of the slot.



By default, this is set to 1 (a numeric constant) meaning that the modulation source is mapped 1:1 with any modulation amounts on destination parameters. In other words the source is unchanged by the scaling function.

Destination modulation depths can be multiplied by any available monophonic or polyphonic modulation source - an easy way to create more complex modulation behaviours.

Selecting a modulation scalar

Click on the lower segment of a TransMod slot to display the TransMod scalar menu - this is identical to the TransMod source menu.

Navigate to and click the desired scalar to assign it to the slot.

In the example on the left, the scalar has already been set to the LFO+ source - to remove the scalar, set it to a *Const* (Constant) source of 1.

Scaling examples

• Note-on velocity (Poly/OnVel+)

Scale any modulation source by the OnVel+ source to dynamically affect the amount of modulation applied to each note with varying MIDI key velocity.

• Performance controller (Perf/Perf1+) or monophonic controller such as mod-wheel (Mono/Mod+)

Scale an LFO with the Mod+ source or a Performance controller mapped to the mod-wheel. This allows the mod-wheel to control the amount of LFO modulation.

• Use the Ramp (Synth/Ramp+) to delay and fade in an LFO

Scale an LFO with the Ramp+ source in order to apply a delay (with the Ramp's **Delay** parameter) and fade-in (using the **Rise** parameter).

Simple TransMod examples

Using the TransMod system to create velocity-based modulation

1. Select the first TransMod slot by clicking its central section. In the Strobe2 Init patch, this slot features the OnVel+ modulation source which represents MIDI note-on velocity.

This slot already features modulation for the V.C.A.'s **Amp** parameter, meaning that each voice's amplitude is modulated by incoming MIDI note velocity.

2. Turn down the V.C.F.'s **Cutoff** parameter to a low value.

3. Draw an upwards modulation depth on the outer ring of the **Cutoff** control.

Now incoming MIDI note-on velocity dictates the value of the **Cutoff** as well as the **Amp** parameter for each played note.

Try creating modulation depths from this TransMod slot for any other parameters - for example, try the LFO **Rate** or envelope **Attack** and **Decay** times!

Using the TransMod system to scale modulators

Using the Ramp to delay and 'fade in' an LFO

1. Deactivate the **Sync** and **Loop** buttons on the Ramp modulator if they are activated.

2. Increase the Delay parameter to around *500ms* and increase the **Rise** parameter to around *1 second*.

3. Select the LFO+ source in the upper part of a TransMod slot

4. Select the Ramp+ source in the lower part of the same TransMod slot

5. Now set a TransMod modulation depth on a parameter such as the Filter **Cutoff** or Main Oscillator **PWM** control.

Managing TransMod modulation

Modulation indicators and slot/control highlighting

Strobe2 provides several ways of keeping track of controls that are being modulated from the various TransMod slots. The following example, with TransMod modulation amounts set up from 3 slots, demonstrates these functions:

• Slot 1

• Slot 2

• Slot 3

Active slot highlighting

• If a TransMod slot features any modulation depths, it is highlighted with an *orange* outline. This allows easy identification of which slots are currently modulating parameters.

Note that in the above examples, all TransMod slots except 1-3 do not feature any orange outlines - this indicates that they do not possess any modulation amounts on parameters.

Active control highlighting

• If any controls are modulated from the current slot, they are highlighted in orange in addition to the orange modulation indicators within the outer ring or slider path (beneath each modulated rotary control and above/below each slider). This is to aid visibility of very small amounts of modulation on controls.

• If any controls are modulated from any other slots, they are highlighted in blue - again, beneath each modulated rotary control and above/below each slider.

Real time parameter modulation indicators

• Modulation depths are animated in real time on the interface, to show an indicative representation of the modulation currently occurring.

Real time slot output indicators

• The central part of each TransMod slot shows an animated indicator representing its modulation signal output.

Parameter to slot highlight

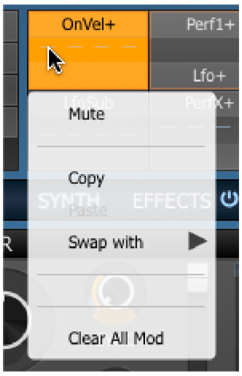
Hover the mouse cursor over a parameter - any TransMod slots which are modulating the parameter are highlighted with a *bright orange* outline.

Slot to parameter highlight

Hover the mouse cursor over any other TransMod slot than that which is currently selected - any modulation amounts from the slot are temporarily shown on the controls.

Managing TransMod slots

Strobe2 features a number of additional functions for managing TransMod modulation slots using the TransMod slot context menu.



• Functions for TransMod slots

The following functions are available by right-clicking in the central part of a TransMod slot and apply to the entire slot. They can also be accessed from the Source and Scalar menus (click in the upper or lower part of the slot).

Mute

When activated, the **Mute** function stops the TransMod slot from having any effect. This can be very useful during sound design for analysing or modifying aspects of existing patches.

Copy/Paste, Swap With...

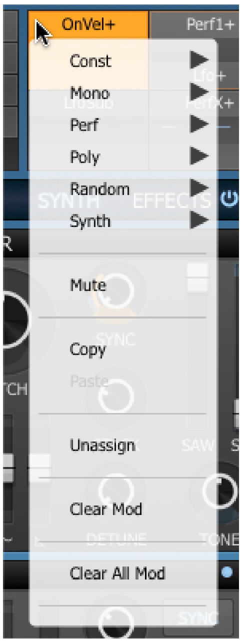
These functions allow the contents of TransMod Slots to be interchanged.

The **Copy** and **Paste** functions operate on the currently selected slot.

The **Swap With** function opens a sub-menu allowing the contents of the current slot to be exchanged with any of the other TransMod slots.

Clear All Mod

The **Clear All Mod** function clears all current modulations depths in all TransMod Slots.



- **Functions for the Source and Scalar within a TransMod slot**

The following functions can be accessed within the source and scalar menus for a TransMod slot - click in the upper or lower part of the slot.

Clear Mod

The **Clear Mod** function clears any modulation depths specified for the TransMod Slot.

Unassign

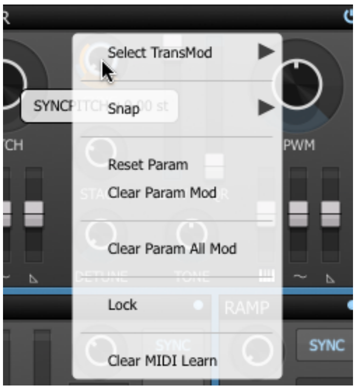
Use the **Unassign** function to remove the source or scalar assignment from the TransMod slot. Any modulation depths for the slot are unaffected. If the slot is subsequently assigned to another TransMod source, the existing modulation depths become active again and target parameters respond to the newly assigned source.

Managing modulation with the Parameter context-menu

Right-click on a parameter in Strobe2 to display its context menu which contains the following functions relating to the TransMod system:

Reset Param

The **Reset Param** function resets the control's initial value to its default setting.



Clear Param Mod

The **Clear Param Mod** function clears any modulation amount that exists for the control from the *current* TransMod slot.

Clear Param All Mod

The **Clear Param All Mod** function clears any modulation amounts that exist for the control from *all* TransMod slots.

Select TransMod

This sub-menu offers another way of selecting TransMod slots.

5.2 Modulator gating/triggering

The adjustable modulation sources shown on the Strobe2 interface - LFO, Ramp, Mod Envelope and Amp Envelope - are known as *gated modulators*.

This term refers to the fact that these modulators require an event - a 'gate' or 'trigger' - to function. A 'gate' can be considered as an on/off event message with a duration - essential for ADSR envelopes gated by MIDI note-on and note-off messages. On the other hand, LFO and Ramp modulators are 'triggered' with an 'on' message only - for example with MIDI note-on messages.

The most common gate/trigger source is MIDI note input. However, a number of additional gating/ triggering modes are provided for more advanced patch design.

While some gated modulators feature direct routings and depth controls for commonly used parameters in Strobe2, the Ramp and additional versions of the LFOs and Envelopes (such as the sub LFO as well as unipolar/inverted versions) must be utilized via the TransMod system.

• *Above: the Dual LFO's Gate modes*

Poly

The *Poly* setting is the default setting for all gated modulators in Strobe2. The modulator is gated or triggered with polyphonic MIDI note input.

When all available voices are exceeded, subsequent notes (which are played as a result of voice stealing) do not trigger the modulator unless the **Retrig** button is activated.

PolyOn

The *PolyOn* mode disregards the **Retrig** button setting entirely - the modulator is always gated or triggered by key-on events for each new voice that is generated.

Mono

In *Mono* mode, a gated modulator is gated/triggered by the first key-on when all notes are off – subsequent keys that are played without releasing all keys do not re-gate the modulator.

This behaviour occurs regardless of the current number of active voices.

Important note

It is important to be aware of the difference between the *Mono* gate mode and the **Mono** button on the LFO:

• Activating the **Mono** button forces the LFO to monophonic rather than polyphonic operation (in other words, a single LFO is available for all voices rather than an LFO being available per voice).

• In *Mono* gate mode, with the **Mono** button deactivated, the LFO is polyphonic but is only retriggered from its starting phase by the first key-on after all notes are off.

Song

The main function of the *Song* gate type is to provide a 'free-running' LFO which is not triggered by key-on events or other modulators.

When playback is started in the host/DAW, the LFO is triggered and runs freely afterwards. This allows free-running behaviour but with repeatability – each time the song is played the LFO runs identically.

When playback is stopped in the host, the LFO simply runs freely, continuously. In such a situation, it behaves just like a standalone synth that features a free-running LFO.

This gate mode can also be used for the Ramp, ModEnv and AmpEnv modulators. This is mainly to facilitate situations such as using the **Loop** envelope mode as a free-running LFO or to create a long Ramp that rises from 0 over the course of a song.

Please note that using this mode with the AmpEnv is a special case - while it is resynchronized by the Song phase, it is also always re-triggered by polyphonic note input in addition.

Other gated modulators

Gated modulators can be gated or triggered by a selection of other modulators – the gating or triggering occurs when the gate source reaches a value of 1.

For example:

• when setting the ModEnv to be gated by the LFO, the ModEnv is retriggered every time the LFO reaches its highest point (1).

• when setting the LFO to be gated by the ModEnv, the LFO is retriggered every time the Envelope reaches its highest point (at the end of the Attack stage).

5.3 Dual LFO

Strobe2's LFO provides sources of movement for modulating parameters and is particularly interesting when its own parameters are modulated with the TransMod system. The **Mode** control offers many waveforms including several types of random shapes. The **PWM** and **Swing** controls provide additional control over the waveform shape.

The secondary 'sub LFO' is derived by clock-dividing the main LFO for additional movement rhythmically linked to the LFO. The sub LFO features its own **Sub Mode** control for setting its shape.

The LFO can operate polyphonically (per-voice) or monophonically (a single LFO for all voices as found in many vintage polysynths) depending on the state of the **Mono** button.

Direct modulation routings

Osc Pitch, Pulse Width, Filter Cutoff

Note that these routings utilize a bi-polar main LFO signal (equivalent to the LFO+- TransMod source).

Gate modes

The LFO features a number of Gate mode settings. For polyphonic 'key-on reset' behaviour, use the *Poly* gate mode. For 'free-running' operation, use the *Song* mode.

Gate mode settings: Poly, PolyOn, Mono, Ramp, ModEnv, Song

Main LFO controls

• *Setting up a free-running monophonic LFO with the **Mono** button and the* Song ***Gate mode***

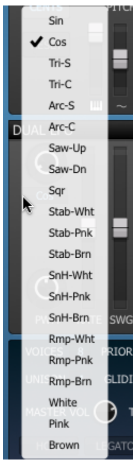
Rate
Sync

The **Rate** parameter controls the rate or speed of the LFO's cycle.

If the **Sync** button is activated, BPM-based units are used when setting the **Rate**. With the Sync button deactivated, the Rate control is adjusted in Hz.

Mono

Enabling the **Mono** button forces the LFO into monophonic operation. In monophonic mode, there is only 1 global LFO instead of an LFO for each voice.



Mode

The **Mode** control selects the LFO shape and can be modulated via the TransMod system. Note that modulating this control continuously (as opposed to, say, with a key-on random source for example) is liable to result in audible clicks.

Standard LFO shapes

- Sine
- Cosine
- Tri-S
- Tri-C
- Arc-S
- Arc-C
- Saw-Up
- Saw-Down
- Square

Random Stab LFOs

- Stab-White
- Stab-Pink
- Stab-Brown

The various noise types provide random points to which the LFO 'jumps'. After it reaches the next point, it drops down vertically to the position of the previous point. This process then repeats. The end result is a series of sharp 'stab' shapes.

Random Sample+Hold LFOs

- SnH-White
- SnH-Pink
- SnH-Brown

These modes are sample+hold values taken from noise sources. They create 'steppy' sounding random LFOs, because each value stays constant until the next value, at which point the LFO immediately travels vertically to the new value.

Random Ramp LFOs

- Ramp-White
- Ramp-Pink
- Ramp-Brown

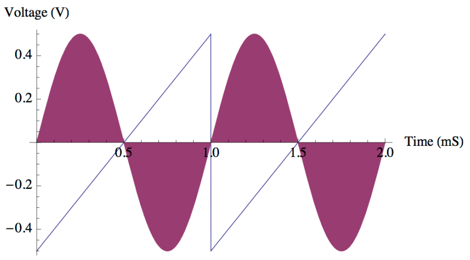
In these modes, the LFO ramps up or down to the next value – the end result is somewhat like a complex triangle shape but with random amplitudes.

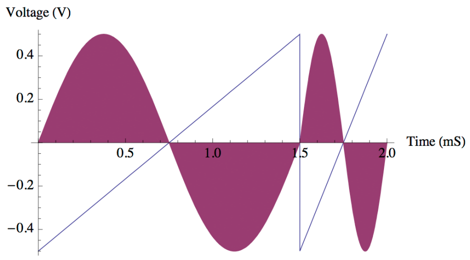
Noise LFOs

- White noise
- Pink noise
- Brown noise

These are straightforward noise sources for use as an LFO that output noise values continuously at control-rate.

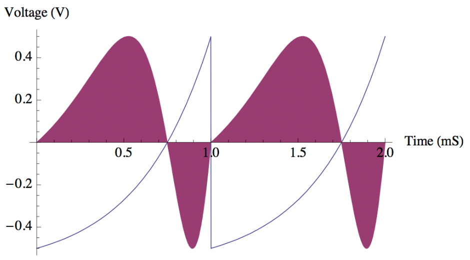
Swg (Swing)





This is an unusual control for an LFO to possess. It creates a 'swung' feel by changing the length of each part of 2 LFO cycles so that each cycle has a different length, but their combined length remains the same.

PWM (Pulse Width)



Again, this control is not often found on LFOs. This function changes the curve of each LFO cycle in a similar way to changing the pulse width on a pulse oscillator. It works on all shape modes.

The diagram illustrates the function at a setting of 75% on a sine shape.

Phs (Phase)

The **Phase** parameter allows the phase of the LFO to be adjusted between 0 and 360 degrees. Phase can be modulated in real time, and always resets when the LFO is gated.

Sub LFO controls

Try using combinations of the LFO and sub LFO TransMod sources to modulate the same parameters. By using different combinations of LFO **Sub Rate** and **Sub Mode** settings it is possible to create fascinating 'additive' LFO waveform shapes. The LFO+Sub TransMod source can achieve similar effects but by using separate LFO and Sub LFO sources, even more variation can be achieved by using positive and negative directions for modulation amounts and also by using scaling to further vary each source.

Sub Rate

The Sub LFO offers a divided or multiplied version of the LFO - it is always synchronized to the LFO but allows additional modulation variation which is rhythmically linked to the LFO.

It can also have an entirely different waveform shape (specified by the **Sub Mode** parameter).

The Sub LFO can run at the following ratios of the LFO speed:

1/8	*1/3*	*1*	*2*	*4*
1/6	*1/2*	*1 1/3*	*2 1/3*	*5*
1/5	*2/3*	*1 1/2*	*2 1/2*	*6*
1/4	*3/4*	*1 2/3*	*3*	*8*

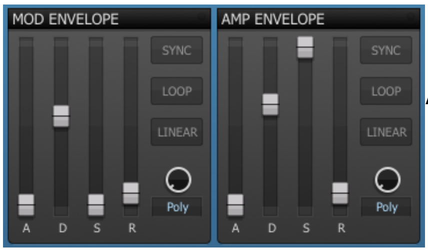
Sub Mode

The **Sub Mode** control above the **Sub Rate** specifies the shape of the Sub LFO - the waveform shapes available are identical to those for the main LFO's **Mode** control.

TransMod sources (Synth sub-menu)

LFO^{+}_{-}	Bi-polar LFO
LFO+	Uni-polar LFO
LFOSub	Bi-polar Sub LFO
LFOSubUni	Uni-polar Sub LFO
LFO+Sub	The LFO and Sub LFO signals mixed together
LFOGate+	Gate output of LFO (square wave)

5.4 Amp and Mod Envelopes

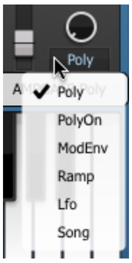


An envelope allows a sound to be shaped over time, from its beginning to its end. It is sometimes called a 'contour'.

Amp Envelope

The **Amp Envelope** is directly routed internally to the Amp parameter in Strobe2 with a fixed depth of modulation. This is to avoid having to make a dedicated modulation connection between them purely for engaging the amp so that notes can be heard.

The TransMod system allows the Amp Envelope to be routed to almost all synth parameters.



Direct modulation routings

VCA Amp (fixed at 100% depth)

Gate modes

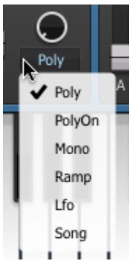
The Amp Envelope features a number of Gate mode settings.

Gate mode settings: Poly, PolyOn, ModEnv, Ramp, LFO, Song

Mod Envelope

The **Mod Envelope** can be used for a variety of purposes. It is similar to the second envelope typically found on many monophonic and polyphonic synths which are often routed to the filter cutoff.

The TransMod system allows the Mod Envelope to be routed to almost all synth parameters.



Direct modulation routings

 Osc Pitch, Pulse Width, Filter Cutoff

Gate modes

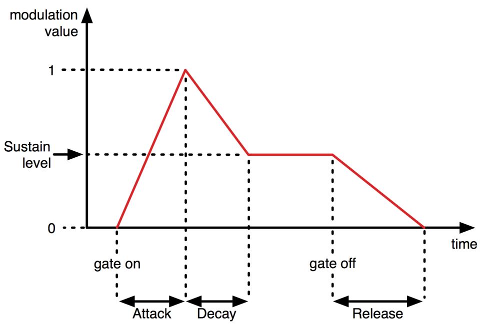
The Mod Envelope features a number of Gate mode settings.

Gate mode settings: Poly, PolyOn, Mono, Ramp, LFO, Song

ADSR controls

Atk (Attack), **Dcy (Decay)** and **Rel (Release)** are time periods, while **Sus (Sustain)** is a level, expressed in %.

When the envelope is gated, the following processes occur:



- The envelope level rises from 0 to 1 over the defined **Attack** time
- After this has been reached, it decays towards the level defined by the **Sustain** control, over a time period defined by the **Decay** parameter (if **Sustain** is at 100%, there is effectively no Decay stage).
- This occurs as long as the envelope is gated (while the gate signal is 'on'). Whenever the gate is released (when the gate signal returns to an 'off' state), no matter which stage has been reached, the envelope's level falls to 0 over the time defined by the **Release** parameter.

All of these parameters can be modulated with the TransMod system.

Sync

With the **Sync** button activated, the time-based envelope controls (**Attack**, **Decay**, **Release**) are set in BPM units.

With the Sync button deactivated, these controls operate in seconds.

Loop

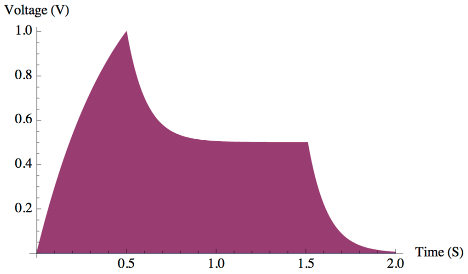
Activating the **Loop** button causes the envelope to repeat after it has completed the **Attack** and **Decay** phases. After the Decay time is complete, the envelope starts again - this continues while the envelope is gated. If the **Sustain** control is higher than 0, the envelope travels towards the sustain level and remains at this level for the duration of the Decay time.

When the gate is released, the level of the looping Attack and Decay stages is scaled down to 0 over the **Release** time.

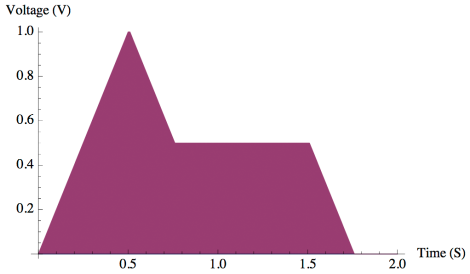
The level of the envelope returns to 0 when it is gated, and at the end of each 'loop' (after each decay stage).

The Loop function is useful for simulating echo/delay effects on gate release or for providing an additional LFO capable of alternative shapes.

Linear



By default, the **Attack**, **Decay** and **Release** phases of envelopes react exponentially.



When the **Linear** button is enabled, their behaviour changes to a linear response.

TransMod sources (Synth sub-menu)

ModEnv+	Mod Envelope (positive)
ModEnv-	Inverted Mod Envelope - emulates the invert switch available on many synth envelopes
AmpEnv+	Amp Envelope (positive)
AmpEnv-	Inverted Amp Envelope - emulates the invert switch available on many synth envelopes

5.5 Ramp

The Ramp generator is a versatile gated modulator that is intended for use with the TransMod system or as a gate source for other gated modulators.

When triggered, its value drops immediately from its maximum value of 1 to 0 (minimum value). Then, after a period of time defined by the **Delay** parameter (during which its value remains at 0), it rises to 1 over a period defined by the **Rise** parameter.

In its most basic form, the Ramp can be seen as a simple inverted envelope with an instant attack and a variable decay, although it is triggered like an LFO rather than gated like an envelope and has a definite period rather than an indefinite period of sustain.

Direct modulation routings

The Ramp does not feature any direct modulation routings.

Gate modes

The Ramp features a number of Gate mode settings.

Gate mode settings: Poly, PolyOn, Mono, ModEnv, LFO, Song

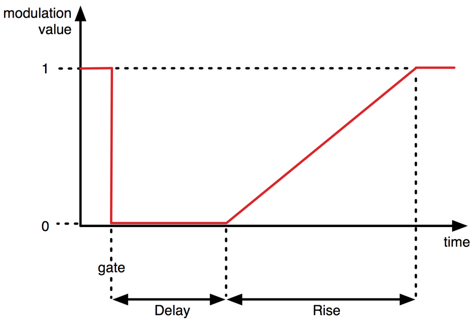
These are the main intended uses for the Ramp:

• A delay before the LFO is triggered – set the LFO Gate to Ramp, and use the Ramp's **Rise** time to set a delay time

• A simple, fixed decay envelope using the Ramp- or RampTrans TransMod sources (the **Rise** control sets the decay)

• A ramp-up scaling function – use the Ramp as a scalar in a TransMod slot to scale an LFO over time (**Rise**) from 0 to 1 - this method can also be used to set a delay on an LFO, using the Ramp's **Delay** parameter

• A polyphonic LFO-style modulator with a saw-wave shape when the **Loop** button is activated (try using a Curve processor to transform it into a different shape)



Dly (Delay)

The **Delay** parameter sets the delay after the ramp value drops from 1 to 0, and before it begins rising to 1. It is scaled by the value of the **Mult** control.

Rise

The **Rise** control adjusts the time taken for the ramp to rise from 0 to 1. It is scaled by the **Scale** control.

Sync

With the **Sync** button activated, the **Delay** and **Rise** times are set in BPM units. With the Sync button deactivated, these parameters are set in seconds.

Scale

This control is a multiplier for the Ramp's **Delay** and **Rise** times. By default, it is set at 100%, meaning that the Delay and Rise times occur exactly as set on the respective controls. The **Scale** control multiplies these times between 0% and 200%.

Loop

With the **Loop** button activated, the Ramp repeats its **Rise** period indefinitely. It therefore provides a useful additional polyphonic LFO-style modulator with a saw-wave shape.

TransMod sources (Synth sub-menu)

Ramp+	Ramp (positive)
Ramp-	Inverted ramp This is very useful as a triggered envelope shape (it is not gated like an envelope).
RampTrans	This is a version of Ramp- but with a slewing function applied which smooths its shape to a curve.

5.6 Keytracking

Keytracking is a modulation source which simply represents pitch input from the keyboard - unlike the LFO/Envelopes/Ramp it does not feature any adjustable parameters. It does, however, include dedicated depth controls for modulating the Osc Pitch, Pulse Width and Filter Cutoff.

In order to play the oscillator 'musically' with the keyboard, the **Keytrack** depth control for the Osc Pitch must be set to 100% (as it is in Strobe2's INIT patch).

In order to play the filter musically (especially useful with the **Resonance** control turned up so that the filter self-oscillates), set the Filter's **Keytrack** depth control to 100%.

Direct modulation routings

 Osc Pitch, Pulse Width, Filter Cutoff

TransMod source (Poly sub-menu)

As well as the direct routings and depth controls specified above, keytracking can also be achieved for any other parameter by using the Pitch source in the TransMod system.

Pitch$^{+}_{-}$	Derived from keys played on the keyboard: increases by 1 for each octave going up the keyboard. Modulate parameters by 12 semitones for full tracking.

KeyZones

Strobe2 provides 2 KeyZone editors for situations when more flexible behaviour is required than that offered by the Pitch TransMod source.

Unlike the Pitch TransMod source, which provides a linear keyboard tracking response, the KeyZone editors allow an arbitrary keyboard tracking curve with which to achieve any type of response across the keyboard range. KeyZone processors are very similar to Curve processors but are designed especially for keyboard tracking purposes.

The KeyZone editors are shown in place of the Scope/Curve/Euclid panels - to access the Keyzone editors, click the **ZONE1** or **ZONE2** buttons. To return to the Visualizer Scope or the Curve Processors, click the **SCOPE** or **CURV1**/**CURV2** buttons.

The default curve for each KeyZone is identical to that of the Pitch TransMod source - a linear response. Click/drag on the curve display in order to draw a customized response.

KeyZone TransMod sources (Poly sub-menu)

KeyZone1+	The output of the Keyzone1 editor
KeyZone1$^{+}_{-}$	As above but bi-polar
KeyZone2+	The output of the Keyzone2 editor
KeyZone2$^{+}_{-}$	As above but bi-polar

ScaleNote TransMod source (Poly sub-menu)

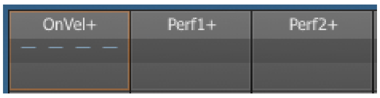
The ScaleNote+ source provides a way of tracking notes of an octave of the 12-tone scale. It ranges from 0 at C to 1 at B, increasing by $1/_{12}$ for each note.

This source can be especially useful when used in conjunction with the Curve processors to apply a distinct arbitrary modulation value for each note in the 12-tone scale.

For example, try modulating the Curve1 processor's **Input** control fully with the ScaleNote+ source. Set its **XStep** control to around 3.55 so that 12 steps are visible. Now click/drag individual steps up/down to create an arbitrary value for each step and use the Curve1+ TransMod source to modulate any desired parameter(s).

ScaleNote+	C=0 for each octave, increasing by $1/_{12}$ for each semitone up to B=5.5

5.7 Velocity and Performance controllers



Velocity and performance controllers are special cases in Strobe2. They are the default sources for the first 3 TransMod slots in the default Strobe2 Init preset and are present within the available TransMod slots in all Strobe2 factory presets.

The presence of these controllers is based on the philosophy that most users typically have access to 3 main performance controllers when playing a keyboard.

The first of these is *MIDI key velocity* which offers polyphonic (per-note) velocity variation for each note.

Other performance controllers tend to be monophonic (affecting all playing notes):

- *modulation wheel*
- *aftertouch*
- *expression pedal*
- *breath controller*

The supplied presets are designed to provide satisfying and dynamic variation using 2 such monophonic controllers - implemented in the P1 (Perf1) and P2 (Perf2) performance controllers alongside polyphonic MIDI note velocity modulation.

Scaling with Velocity and performance controllers

A fast and easy way to inject expressive control into an existing Strobe2 preset is to use velocity and performance controllers as TransMod scalars in existing slots - for example, try using a performance controller mapped to a mod-wheel as a scalar for a TransMod slot containing an LFO source.

Velocity

In most cases, this is the most immediate control available to users with a MIDI keyboard. It requires no extra features or hardware, and is available with 'digital piano' keyboards, which usually lack even a mod wheel. It is also polyphonic so acts on each individual note that is played.

The TransMod system allows almost any parameter to be modulated with velocity, meaning that incredibly dynamic sounds are possible.

TransMod sources (Poly sub-menu)

OnVel+	Key-on velocity: default Init preset source for TransMod slot 1 Modulate the **Amp** parameter with this source in order to vary amplitude with velocity (this is already set up in the Init preset). The **Velocity Glide** control creates glide times to new notes' OnVel modulation values, allowing a further expressive element to performances.
OffVel+	Key-off velocity

Velocity Glide

While a note velocity value is received when a key-on event occurs, Strobe2 also includes the Velocity Glide function - this special glide function smooths out over time any transitions to the modulation depths defined in slots driven by the OnVel+ source.

It can be considered in a similar way to regular pitch glide which smooths transitions to new pitch values, but it acts on any synthesis parameter to which velocity modulation has been assigned.

Performance controllers (Perf1 and Perf2)

Performance controllers are monophonic – they represent a single controller which is applied to all voices. They are represented on the interface to the left of the on-screen keyboard, but they are intended to be assigned to hardware controllers.

Performance controllers are monophonic – they represent a single controller which is applied to all voices. They are represented on the interface to the left of the on-screen keyboard, but they are intended to be assigned to hardware controllers using the built-in MIDI CC Learn functionality.

Strobe2 also features an X-Y pad in the Euclid processor which can be used as a performance controller. The **Base X** and **Base Y** values are available in the TransMod system as the PerfX+ and PerfY+ sources.

To map the Perf1, Perf2, BaseX and BaseY controls to MIDI controllers, assign the on-screen controls using Strobe2's built-in MIDI CC Learn system. For the assignments to persist in future sessions, use the **Save preferences** function in the Preferences menu.

TransMod sources (Perf sub-menu)

Perf1+	Performance controller 1: default Init preset source for TransMod slot 2
Perf2+	Performance controller 2: default Init preset source for TransMod slot 3

5.8 Monophonic sources

Monophonic TransMod sources operate globally on all active voices - unlike polyphonic sources which modulate each voice independently.

Typical examples are continuous MIDI messages such as MIDI CCs, mono pressure (channel aftertouch) or mod wheel - these all represent a single stream of signals which are distributed to all active voices.

With the exception of the monophonic sources described below, Performance controller sources and monophonic gate/trigger sources, all other TransMod sources described in this chapter are polyphonic.

Common MIDI controller sources (Mono sub-menu)

These sources are provided to facilitate direct control of a TransMod slot from common MIDI performance controllers.

Strobe2 also provides the dedicated Perf1 and Perf2 controllers, which are intended to act as the 2 main performance controls. The Perf1 and Perf2 controls are generally intended to be assigned to 2 of the first 4 controls in the following list.

See the previous section for more details on the Perf1 and Perf2 controls.

Mod+	Modulation wheel: MIDI CC #1
Breath+	Breath controller: MIDI CC #2
Expr+	Expression: MIDI CC #11
MnPress+	Mono pressure (channel aftertouch)
MonoBend$^{+}_{-}$	Pitch bend

Step-sequencer sources (Mono sub-menu)

Strobe2's Arpeggiator page features a Step-sequencer modulation lane that corresponds to a monophonic TransMod source (available in uni-polar and bi-polar versions).

As the sequence is stepped through by the timing clock the specified values for each step are output to the StepSeq TransMod sources.

Please note that several aspects of the Step-sequencer are dependent on the Arpeggiator settings.

StepSeq$^{+}_{-}$	Step-sequencer modulation lane (bi-polar).
StepSeq+	As above but uni-polar).

Song phase (Mono sub-menu)

These sources provide a saw-up ramp synchronized to the host tempo. Usually, song phase is used for mono free-running gated modulators with repeatable behaviour, but they are also provided in the TransMod system for extra mono LFOs synced to 1 or 4 beats of the host tempo.

Phase1$^{+}_{-}$	Song phase: a saw-up ramp from 0 to 1 per beat, based on song position
Phase1+	As above, but uni-polar
Phase2$^{+}_{-}$	Song phase: a saw-up ramp from 0 to 1 over 4 beats, based on song position
Phase2+	As above, but uni-polar

Numeric constants (Const sub-menu)

Numeric constants are very useful scalars for the modulation from a TransMod slot.

0, 0.1,0.5, 1, 2, 10, -1	1 is the default scalar for all TransMod slots, meaning that each source is unchanged before modulating the destination parameters.

5.9 Voice and Unison sources

Voice and Unison sources allow polyphonic modulation effects which spread parameter values within the modulation depth range according to the number of active voices. Active voice counts are converted into a set of values from -1 to 1 (bi-polar) and 0 to 1(uni-polar). Each source is provided in 2 different varieties of value distribution between the minimum and maximum.

The Unison sources allow any parameter to be varied across unison voices as opposed to most vintage synths which were limited to detuning the pitch of unison voices although some examples (such as the Oberheim OB-8) featured trim pots to adjust the pan position of each voice.

Unison and Voice sources in the TransMod system enable effects more reminiscent of the original Oberheim polysynths based on multiple discrete SEM modules, which allowed the programming of a totally different patch for each voice.

Voice count TransMod sources (Poly sub-menu)

Voice1$^+_-$	Each active voice generates a value proportionally distributed between -1 and 1
Voice1+	As above but uni-polar
Voice2$^+_-$	As Voice1 but distributed in a different order
Voice2+	As above but uni-polar

Random voice TransMod sources (Random sub-menu)

The following per-voice random sources can also be used:

VMRand1$^+_-$	A random number per voice, per parameter, that is generated at synth load time, generated using white noise This source is useful for having a set of random values that stays constant throughout the current session.
VMRand1+	As above but uni-polar
VMRand2$^+_-$	Identical to VMRand1$^+_-$ but using pink noise
VMRand2+	As above but uni-polar

Alternating voice TransMod sources (Poly sub-menu)

The Alt sources output alternating values for each voice used in sequence:

Alt1$^+_-$	Alternating source per key-on: -1, 1, -1, 1...
Alt1+	As above but positive only: 0, 1, 0, 1...
Alt2$^+_-$	Alternating source per key-on: -1, 0, 1, -1, 0, 1...
Alt2+	As above but positive only: 0, 0.5, 1, 0, 0.5, 1...

Unison TransMod sources (Poly sub-menu)

The Unison sources allow any parameter to be varied across unison voices – not just the pitch detuning found in classic synths.

Unison1$^+_-$	Each active unison voice outputs a value proportionally distributed between -1 and 1 This source is especially suited to panning voices across the stereo field.
Unison1+	As above, but uni-polar
Unison2$^+_-$	As Unison1 but distributed in a different order This distribution can be more suitable for unison detuning with osc pitch modulation.
Unison2+	As above, but uni-polar

Examples and ideas for using voice and unison TransMod sources

• Modulate the **Pan** control for a stereo spread of voices during chords or unison notes.

• Modulate the V.C.F. **Mode** control for a different filter type for each voice!

• Subtle voice and unison modulation on envelope, filter, osc **Pitch** and other controls can result in more variation during performances and emulate the slightly uneven, unpredictable feel of a vintage synthesizer containing voice cards with slight differences in calibration or pitch-tracking.

• Modulate **Pitch Glide** and **Velocity Glide** controls to give each voice its own glide time.

• Use the Voice/Unison sources as TransMod scalars to inject per-voice variation for other modulation sources.

5.10 Curve processors

Strobe2 provides 2 Curve Processors which can be used for remapping a modulation source to an arbitrary stepped curve with adjustable slew (smoothing). This modulation processing is polyphonic - it occurs per-voice.

The Curve Processor editors are shown in place of the Visualizer Scope - click the **CURV1** or **CURV2** buttons to display them. To return to the Visualizer Scope display, click the **SCOPE** button.

Curve Processor controls

Input

The **Input** control is intended to be modulated by a TransMod source.

XStep

The **XStep** control adjusts the number of steps in the X-axis (horizontal) of the curve.

YStep

The **YStep** control adjusts the number of steps in the Y-axis (vertical) of the curve.

Slew

The **Slew** control adjusts the amount of smoothing over time between steps on the curve - a setting of 0 results in no smoothing and the curve output is quantized to discrete values.

Curve Editor

Each step on the curve can be dragged up/down to set its value. The value is set when the mouse button is released or when the cursor is moved over an adjacent step. In this way, curves can be 'drawn' directly onto the display.

• *Using the **Reset** function*

• Hold down SHIFT while click/dragging to draw straight lines
• Hold down CTRL and SHIFT to draw a straight horizontal line

To reset the curve to the default linear shape, right-click in the Curve Editor and use the **Reset** function.

To use more intricate curve shapes, set the **XStep** and **YStep** parameters to high values.

The Curve Editor display shows real-time feedback of the **Input** modulation.

TransMod sources (Poly sub-menu)

Curve1+	Output from Curve Processor 1
Curve2+	Output from Curve Processor 2

Examples and ideas for the curve processors

• Freely-drawn LFO shape

Use the LFO or Sub LFO to modulate the Curve processor's **Input** control, turn up the **Step** control to the maximum setting and draw a custom LFO shape as the curve. Then set the **Slew** control to smooth the output as required.

Alternatively, try the same method with the Ramp with **Loop** button activated. Set the **Delay** parameter to *0* and draw the required curve shape for the Ramp's **Rise** travel.

• Freely-drawn Ramp or Envelope shapes

Try the same as above but with a Ramp-, RampTrans or ModEnv source to draw custom Ramp transient and Mod Envelope responses.

• Slewing any TransMod modulation source

Modulate the Curve processor **Input** with the LFO+ source, with the LFO shape set to *Sqr*. Then adjust the **Slew** control - higher settings result in a sine-like shape. This method can also be very useful for creating a smoother shape from a looping Ramp source.

• Multiplying 3 TransMod sources

With a maximum **Step** setting, modulate Curve1's **Input** control from a TransMod slot with a source and scalar specified. Then, in a second TransMod slot, set Curve1+ as the source - this source effectively outputs the result of the original slot's source multiplied by the scalar. The resulting modulation from the Curve1+ source can now be multiplied by another scalar in the second TransMod slot.

• Applying different modulation to each note in the 12-tone scale

When used with the ScaleNote+ source, Curve processors can be used to apply a distinct arbitrary modulation value for each note of the 12-tone scale.

5.11 Euclid processor

The Euclid processor utilizes Euclidean geometry and a spring model to provide an interesting way of creating chaotic and complex modulation signals for use in the TransMod system. It can also be used as an X-Y pad (mapped to an external X-Y pad controller if desired) with adjustable inertia.

The X-Y position is set by the **Base** and **Mod** controls while the **Slew**, **Rate** and **Damp** functions dictate the way that movements to new positions occur.

The Euclid processor editor is shown in place of the Visualizer Scope/ Curve/Keyzone processors - click the **EUCLID** button to display it. To return to any other display, click its respective button.

Base X
Base Y

These controls set the base position on the X-Y area - they can be adjusted either by moving the rotary controls themselves or by clicking anywhere on the X-Y area. **Base X** and **Base Y** are very similar to the **Perf1** and **Perf2** controls - they are intended to be monophonic performance controllers and are perfect for assigning to a hardware X-Y pad controller. In fact, their direct signals are available with the PerfX+ and PerfY+ TransMod sources - these output the Base X and Base Y values without any of the Euclid processor's additional functions being applied.

Mod X
Mod Y

The **Mod X** and **Mod Y** controls apply an offset to the X and Y axes and can be modulated using the TransMod system in order to create new modulation signals by processing existing ones.

Slew

This control adjusts the amount of **Slew**, or lag, applied to movement of the X-Y position.

Rate

The **Rate** control adjusts the speed of movement of the X-Y position.

Damp

The **Damp** control adjusts the amount of damping applied to the X-Y position's inertia. The inertia is derived from a spring model - increasing the Damp control 'tightens' the spring so that less elasticity is evident for X-Y position movements.

TransMod sources (Poly sub-menu)

EuclidX	X-axis output from Euclid processor
EuclidY	Y-axis output from Euclid processor
EuclidRadius	The distance from the centre of the X-Y area to the current value
EuclidAngle	The angle from the centre of the X-Y area to the current value
PerfX+	This source outputs the **Base X** value (without any further Euclid processing)
PerfY+	This source outputs the **Base Y** value (without any further Euclid processing)

Examples and ideas for the Euclid processor

• Changing LFO shapes

Set the LFO to a *Sqr* shape, modulate the **Mod X** or **Mod Y** controls with the LFO+ source, set the **Damp** control to 100%, adjust the **Slew** and **Rate** controls and use the EuclidX or EuclidY TransMod sources to modulate parameters with the result. Setting **Slew** at approx. 70% and **Rate** at 50% results in an output very similar to a sine LFO.

• Creating chaotic LFOs

Modulate the **Mod X** control with the LFO+ source and modulate the **Mod Y** control with the LFOSubUni source (with the Sub LFO's division set to 1/3 for example). Use the EuclidRadius source to modulate parameters with the result.

• Slewing any TransMod modulation source

Modulate the **Mod X** or **Mod Y** controls with any TransMod modulation source, increase the **Slew** control and use the EuclidX or EuclidY TransMod sources to modulate parameters with the result.

For example, using a PlGate+ or PlOnGate+ source can result in a slow ASR envelope shape suitable for a pad sound. The envelope response can be further refined with the **Rate** and **Damp** controls.

Other sources to try include:

• the LFO set to one of its random shapes

• the RampTrans or Ramp- sources, especially useful with the Ramp's **Loop** button activated.

5.12 Random sources

The TransMod system contains a range of polyphonic random modulation sources of various types, from simple noise sources to more specific ways of generating random values. With the exception of the Noise sources (the first 4 entries below), each of the random sources acts on key-on events: these are not constantly changing/evolving random sources. For the latter, the LFO with its various random modes should be used or, alternatively, creative use of the Euclid and Curve processors.

TransMod sources (Random sub-menu)

Noise1$^{+}_{-}$	Fast-varying white noise random value
Noise1+	As above, but uni-polar
Noise2$^{+}_{-}$	Fast-varying pink noise random value
Noise2+	As above, but uni-polar
Rand1$^{+}_{-}$	A single random value for all destination parameters, generated by key-on sample-and-hold on white noise
Rand1+	As above but uni-polar
Rand2$^{+}_{-}$	A single random value for all destination parameters, generated by key-on sample-and-hold on pink noise
Rand2+	As above but uni-polar
MRand1$^{+}_{-}$	Individual random values for all destination parameters (multiple random values), generated by key-on sample-and-hold on white noise. This source is useful for modulating each note with a different random value every time it is played.
MRand1+	As above but uni-polar
MRand2$^{+}_{-}$	Identical to MRand1$^{+}_{-}$ but using pink noise
MRand2+	As above but uni-polar
VMRand1$^{+}_{-}$	A random number per voice, per parameter, that is generated at synth load time, generated using white noise. This source is useful for having a set of random values that stays constant throughout the current session.
VMRand1+	As above but uni-polar
VMRand2$^{+}_{-}$	Identical to VMRand1$^{+}_{-}$ but using pink noise
VMRand2+	As above but uni-polar
Drift$^{+}_{-}$	Slowly changing random LFO This can be used to simulate subtle parameter drift in hardware analogue synths ñ for example. small amounts of modulation on osc pitch can simulate drifting VCOs.

5.13 Gate/Trigger sources

Normally, gate and trigger sources are used for various types of gated modulator behaviour. However, they are also provided as TransMod sources for complex modulation operations (try using them as scalars), or as a way of achieving quick pulses for filter stabs and so on without having to use an envelope.

These sources simply output a value of 1 or 0 depending on several conditions.

Monophonic gate/trigger sources (Mono sub-menu)

MnGate+	Outputs 1 if one or more keys are held down, otherwise 0
MnOnGate+	Outputs 1 if one or more keys are held down, but is retriggered on every key-on (output returns to 0 and immediately to 1)
MnTrig+	Outputs 1 then immediately 0 on key-on when a note is played after all notes are off
SongGate+	Outputs 0 if the host song is not playing, and 1 if it is playing: this gates on the nearest beat

Polyphonic gate/trigger TransMod sources (Poly sub-menu)

These are very similar to the monophonic gate/trigger sources, except that they operate polyphonically (they can occur multiple times depending on the number of available voices).

PlGate+	Outputs 1 if one or more keys are held down, otherwise 0 Retriggering occurs according to the state of the Retrig button (returns to 0 and immediately to 1).
PlOnGate+	1 if one or more keys are held down, but retriggers on every key-on (returns to 0 and immediately to 1)
PlTrig+	This is the same as PlGate+, except that it outputs a 1 and then returns to 0 immediately when keys are played.

6 Arpeggiator page

Strobe2's Arpeggiator and modulation sequencer functions are located on the Arpeggiator page - click the **Arpeggiator** button on the interface in order to access it.

The Arpeggiator uses keyboard notes to build a rhythmic arpeggio, described in this chapter as an *Arpeggiator sequence*.

The Step Sequencer runs in parallel with the Arpeggiator - it runs from the same 'clock'. However, it is not used to produce notes - instead, it outputs a rhythmic sequence of modulation values to the TransMod modulation system which can be used to modulate parameters using the StepSeq+ and StepSeq^{+}_ TransMod sources.

Arpeggiator Page Power

The **Arpeggiator Power** button provides a global activation function for the Arpeggiator page - with the button activated, the Arpeggiator and Step sequencer are also active.

With the button deactivated, the entire Arpeggiator page is disabled.

Please note that for the Arpeggiator to operate, the **Input** button must be activated.

The Step sequencer always runs with the **Arpeggiator Page Power** button activated and while it receives a clock signal from the DAW/host (with Strobe2 running as a plugin) or the Strobe2 standalone application.

Sequencer Lane

The sequencer lane allows step values to be drawn with the mouse - click and drag up/down on each step to set its value, additionally dragging left-right across multiple steps if required.

Sequence Length

The length of the Sequence is determined by the **Sequence Length** parameter. Note that other factors can affect the length of the Sequencer, such as the **Arp Resets Seq** and **Seq Resets Arp** buttons.

Slew

The **Slew** control applies an adjustable amount of lag, or glide, between values output by the Step Sequencer to the StepSeq+ and StepSeq^{+}_ TransMod sources.

Arpeggiator functions

Input

When the **Input** button is activated, keyboard input is routed to the Arpeggiator - in this mode, all keyboard notes contribute to the Arpeggiator sequence. With the button deactivated, the Arpeggiator is disabled and all keyboard input is routed directly to the Strobe2 synthesis engine.

Hold

With the **Hold** button activated, MIDI note-off messages are ignored, thereby allowing the Arpeggiator sequence to be repeated indefinitely without having to manually hold down the notes on the keyboard.

Rate

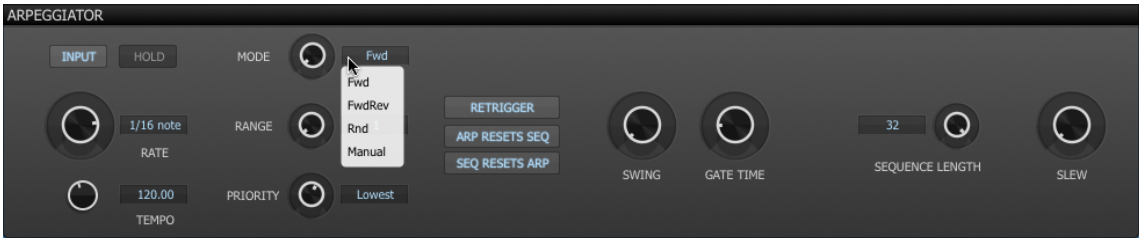
The **Rate** control sets the speed of the clock used for the Arpeggiator and modulation Sequencer - it is set as a rhythmic division of the **Tempo**.

Tempo

When running Strobe2 as a plugin within a DAW/host, it is not possible to adjust the **Tempo** control as Strobe2 is always locked to the host's tempo. However, in the standalone application version of Strobe2, this control can be adjusted.

Please note that this control cannot be modulated by the TransMod system.

Mode



This setting, in conjunction with the **Priority** setting, dictates how keyboard notes are arranged for the Arpeggiator sequence.

Fwd

Notes are arranged according and played according to the Priority setting. After the sequence completes, it starts again from the beginning.

FwdRev

In the FwdRev Mode, the Arpeggiator sequence is arranged according to the Priority control, plays forward as in the Fwd mode and then plays in reverse before starting again from the beginning.

Rnd

In this Mode, notes are arranged in a random order to form the Arpeggiator sequence.

Manual

This Mode uses the each Sequencer step's value to dictate which note in the Arpeggiator sequence should be played. The value range in each Sequencer step is divided by the number of notes in the Arpeggiator sequence - each step's value plays the corresponding note in the Arpeggiator sequence.

Range

By default, the Arpeggiator sequence is created from the notes played from the keyboard. The **Range** control allows the Arpeggiator sequence is extended by repeating it over ascending octaves, up to a maximum of 4 octaves. Note that the final Arpeggiator sequence includes all generated notes including those added by using the **Range** control.

Priority

The **Priority** control dictates the order in which keyboard notes held down simultaneously are inserted into the Arpeggiator sequence - the sequence plays *towards* the specified setting.

Newest

The *Newest* note (that is, the last played note), is always placed at the end of the Arpeggiator sequence. This setting creates the Arpeggiator sequence according to the order in which notes are played.

Oldest

The *Oldest* note is always placed at the end of the Arpeggiator sequence.

Highest

The *Highest* note is always placed at the end of the Arpeggiator sequence - this setting always results in classic 'ascending' arpeggios.

Lowest

The *Lowest* note is always placed at the end of the Arpeggiator sequence - this always results in descending arpeggios.

Hardest
Softest

These settings use the velocity of input MIDI notes to dictate the order of the Arpeggiator sequence. In *Hardest* mode, the highest velocity note is placed at the end of the Arpeggiator sequence, while in *Softest* mode, the lowest velocity note is always placed at the end of the Arpeggiator sequence.

Swing

The **Swing** control adjusts the amount of syncopation applied to 'off-beat' steps (steps 2, 4, 6, 8, 10 and so on).

Gate Time

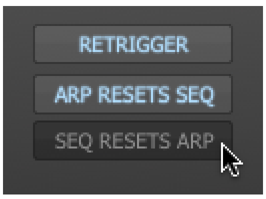
The **Gate Time** setting dictates the length of the notes created in the Arpeggiator sequence. With a setting of 100%, adjacent notes in the sequence 'overlap', meaning that it is possible to use **Legato** glide behaviour.

Arpeggiator and Step Sequencer Interaction

The Arpeggiator and Step-sequencer are able to interact with each other in a number of ways.

Retrigger

The Sequencer lane always runs while it receives a clock signal from the DAW/host or standalone application. With the **Retrigger** button activated, the Sequencer is reset to step 1 when the keyboard is used to play an Arpeggiator sequence (the **Input** button must be active).



Arp Resets Seq

When this function is active the Sequencer lane is reset to step 1 on the first note of the Arpeggiator sequence.

Seq Resets Arp

With this function active the Arpeggiator sequence is reset when playback reaches step 1 of the Sequencer lane.

Arp Resets Seq and Seq Resets Arp active simultaneously

With the **Arp Resets Seq** and **Seq Resets Arp** settings both activated, both behaviours described above are active simultaneously and occur whenever the relevant conditions are met as described above.

7 Effects page

Using Strobe2's Effects section

Strobe2's Effects section is located on the Effects page - click the **Effects** button on the interface in order to access it.

Effects Power

The **Effects Power** button provides a global activation function for the entire Effects section - with the button activated, the Effects section is also activated.

With the button deactivated, the entire Effects section is bypassed.

FX A and FX B Chains

Strobe2's Effects section comprises the FX A and FX B Chains which are arranged in series with FX A feeding into FX B. Each chain features 3 FX Slots for inserting available FX devices.

Chain Mix controls

Each FX Chain features a **Chain Mix** control that can be modulated by the TransMod system. The Chain Mix control blends between the input and the FX Chain's output.

FX A Mix

The **FX A Mix** control blends between Strobe2's raw output and the output of the FX A chain.

FX B Mix

The **FX B Mix** control adjusts the blend between the output of the FX A and FX B chains.

In the above screenshot, the FX A Chain is entirely bypassed and the FX B Mix control results in a 50/50 mix between the dry Strobe2 output and the Delay/Reverb chain.

FX Slot controls

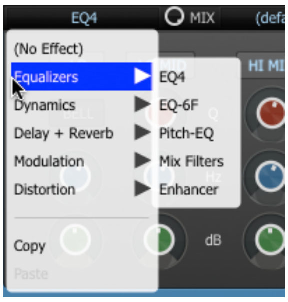
Each of the 3 FX Slots in each FX Chain features the following common controls:

Power

 The Slot's **Power** button activates/deactivates the device loaded into the FX Slot.

Mix

Each FX Slot features a **Mix** control for blending between the incoming signal (*0*) and the fully-processed output (*100%*) from the device loaded into the FX Slot.



Device menu

This drop-down menu selects an FX device to load into the FX Slot.

It also contains functions to **Copy** and **Paste** the contents of the Slot.

Device Preset controls

This set of controls browses through the available presets for the loaded Effect device.

The **Preset menu** provides a drop-down menu of the available presets for the device.

The **Previous** and **Next** buttons cycle through the available presets. The currently loaded preset name is shown on the **Preset menu**.

Reordering FX Chains

The sequence of FX devices within the 6 Slots can be changed using drag and drop. To move an FX device, simply drag it to the desired Slot - other devices are rearranged within the available Slots accordingly.

It is also possible to use the **Copy** and **Paste** functions on each Slot's **Device menu** in order to copy FX devices with their settings to other Slots.

Effects and TransMod modulation

Strobe2's FX devices can be modulated with the TransMod system in a similar way to the Synth controls.

Effects section and polyphony

The crucial difference when modulating the Effects section is that it is *monophonic* - when using more than 1 voice in the Strobe2 Synth section, the multiple voices are mixed down to a stereo signal and then fed into the Effects section which then acts on the entire signal.

When using *polyphonic* TransMod modulation sources, the Effects section applies the modulation signals from the *last voice played*. In other words, it features *Newest* note priority.

7.1 EQ

EQ

The EQ device provides 4 EQ bands including two parametric mid bands.

L (Low) and H (High) bands

The Low and High bands are switchable between a bell with a fixed **Q** of 2.5 octave (with the **Bell** button activated) and a shelving response with a fixed rolloff of 12 dB per octave (with the **Bell** button deactivated).

• Low band **Freq** range: 40 to 600 Hz

• Low band **Gain** range: +/- 16.5 dB

• High band **Freq** range: 600 Hz to 14 kHz

• High band **Gain** range: +/- 20 dB

LM (Low Mid) and HM (High Mid) parametric bands

The Mid bands feature bell curves with adjustable Q.

• Low Mid Band **Freq** range: 200hz to 2 kHz

• High Mid band **Freq** range: 800hz to 7 kHz

• **Q** range: 0.5 octave to 2.5 octave

Band Power

Each band has its own **Power** button - turning off a band reduces CPU usage.

EQ-6F

The EQ-6F device is similar to the EQ device with a few differences:

• 4 parametric mid bands instead of 2

• High-pass and Low-pass filters as well as Shelf/Bell Low and High bands.

Pitch EQ

The Pitch EQ is tuned to musical notes and features 3 EQ bands including a parametric mid band.

Mix Filters

The EQ-Filter effect provides non-resonant high-pass and low-pass filters for broad tonal shaping of signals.

HP Frequency

The -18dB/oct high-pass filter removes frequencies below the specified frequency which can be set within a range between 20 Hz and 500 Hz.

LP Frequency

The -12 dB/oct low-pass filter removes frequencies above the specified frequency, which can be set within a range of 35 Hz to 3 kHz.

Power buttons

The **HP In** and **LP In** buttons are power buttons for each filter – when activated, the filter is enabled, and when deactivated, the filter is bypassed.

Enhancer

The Enhancer device is intended to add high-frequency harmonics to emphasize these frequencies in the signal. It is useful for adding 'sizzle' to sounds to help them cut through dense mixes.

Frequency

The Enhancer adds harmonics at and above the **Frequency** defined with this control.

Mode

The **Mode** control adjusts how the Enhancer circuit reacts to transients in the incoming signal. *I* is the slowest setting while *IV* is the fastest.

Amount

The **Amount** control adjusts the level of the harmonics which are added to the signal before the output.

7.2 Dynamics

Comp Chan

Comp Chan (channel compressor) is derived from a classic 'feedback-based FET limiting amplifier' design.

It features a 'fixed-threshold' design – the threshold at which compression starts is not adjustable. In practice, this means that the **Input** and **Output** controls may need adjustment when changing ratios.

Input & Output

The **Input** control adjusts the level of the signal entering the compressor. Once the input level has reached the internal threshold, compression begins. When this happens, use the **Output** control to turn down the increased input. The **Input** control ranges from -20 to +40 dB, while the **Output** control ranges from -40 dB to +20 dB.

Attack

The **Attack** control has a range between 0.02 ms and 1.2 ms.

Release

The **Release** control has a range between 50ms and 1.2 seconds.

Ratio

This control sets the compression **Ratio** to 4:1 , 8:1 , 12:1, 20:1 or 'Nuke', which is an emulation of the 'all buttons' mode on a classic limiting amplifier design. It results in a particularly brutal type of compression with accompanying distortion artifacts.

Comp Bus

This device is based on a classic bus compressor design from the centre section of a well-known British large-format mixing console.

It is most commonly used to add 'glue' and power to a drum or mix bus. However, it also works very well as an instrument compressor in its own right and offers a different flavour of compression to that of the Comp Chan.

Key HP (Key signal High-pass)

The **Key HP** control adjusts a variable high pass filter on the key signal that is used for the compressor's amplitude detection. It is applied whether the main input or the SideChain input is being used to drive the compressor. No filtering is applied to the audible signal - only to that being used to drive the peak detection circuit.

This control is useful when there is too much low-end in the sidechain signal, resulting in the compressor reacting more heavily than desired.

Attack

Six **Attack** times are available: 0.1 ms, 0.3 ms, 1 ms, 3 ms, 10 ms, 30 ms.

Release

Five **Release** settings are available: 0.1 ms, 0.3 ms, 0.6 ms, 1.2 ms and Auto.

Ratio

Three **Ratio** settings are available: 2:1, 4:1 and 10:1.

Threshold

Unlike the Comp Chan, the Comp Bus device allows the **Threshold** to be adjusted. The Threshold represents the signal level at which the compressor begins to react.

Output

The **Output** control allows the overall output level to be increased after the compressor circuit has applied gain reduction to the input signal.

Limit

The **Limit** button applies analogue non-linearities to the input to the compressor's amplitude detection circuit (while not affecting the input signal itself). This results in a more transparent character to the compression effect, especially on attack phases of transients.

NoiseGate

A noise gate is a type of dynamics processor that attenuates the input signal until its amplitude exceeds an adjustable threshold level, at which time the gate 'opens' to allow audio through at its actual level.

It can be useful for 'chopping' effects with a sound whose amplitude is being modulated by an LFO, or for helping to create 'gated reverb' effects.

When setting very fast **Attack** and **Release** times, it is common to hear 'clicks' in the audio when the gate opens and closes, especially with sounds predominantly comprised of low frequencies such as kicks and toms. This behaviour is completely normal - these times simply need to be increased slightly to overcome the problem.

Attack

The **Attack** control adjusts the speed at which the gate opens once the **Threshold** has been exceeded by the input signal's amplitude.

Hold

The **Hold** parameter controls the amount of time the gate remains open after the input signal has dropped below the **Threshold** level.

Release

The **Release** control adjusts the speed at which the gate closes at the end of the hold time.

Threshold

The **Threshold** control sets the level at which the gate starts to open. When the input signal amplitude exceeds the level specified by the Threshold control, the gate starts to open to allow audio through.

LPF and HPF Input Filters

This control provides high-pass (**HPF**) and low-pass (**LPF**) filters to process the input signal used to trigger the gate while leaving the actual processed signal *unfiltered*. This allows certain frequencies in the input to be isolated to improve the gating response – for example, excessive low frequencies in the input can make the gate react more sensitively than required.

Hysteresis

Noise gates have a tendency to open and close very quickly when the input signal's amplitude remains close to the threshold level for longer periods, something that can result in 'gate chatter'.

Increasing the **Hysteresis** control smooths out the gate response to reduce this problem, although the gate becomes less sensitive to small changes around the threshold level.

Mix

With the Mix control at 100%, the NoiseGate mutes the signal completely when closed. To let some of the original signal through at a low level, decrease the Mix control as desired.

EnvShaper

Env Shaper offers an alternative approach to dynamics processing by adjusting the intensity of attack and sustain portions of transients.

Attack

The **Attack** control adjusts the intensity of the attack phase of detected peaks in the audio signal. Increase the control to intensify attack transients, and decrease it to soften transients.

Sustain

The **Sustain** control adjusts the intensity of release portions of detected peaks in the audio signal, which increases or decreases the apparent sustain of sounds in the signal.

Increase this control for more sustain, and decrease it for less sustain.

This effect can also be useful for creative emphasis or de-emphasis of reverb effects earlier in the FX chain.

Signal Bias

The **Signal Bias** control adjusts the sensitivity and release characteristics of DCAM EnvShaper. At low settings it is more sensitive to short transients while at higher settings it is more sensitive to longer transients.

Gain

The Gain device is a simple tool for increasing or decreasing a channel's level, useful as a final make-up gain or attenuation stage before Strobe2's output.

Gain Amp

The **Gain** control increases the channel's gain up to 18 dB, or decrease it up to -inf dB.

Linear Amp

This control provides a linear-response attenuator for the channel's gain - at a setting of 100% the signal is unaffected while at a setting of 0% the signal is fully attenuated.

Pan

The **Pan** control adjusts the balance between the left and right channels in the stereo field.

7.3 Delay and Reverb

Delay

Time

The delay **Time** can be set either in absolute time values in seconds or in tempo-based values (depending on the the **Sync** setting).

Sync

With the **Sync** button activated, the delay time is set in tempo-based values relative to Strobe2's current tempo. Possible values range from 64th note to 2 bars, including dotted and triplet variations. With the button deactivated, the delay time is set as an absolute time value, ranging from 31 ms to 4 seconds.

Swing

This control opens up a wide variety of delay grooves. The Delay effect features 2 taps, which are played at the same time with the swing control at the centre position.

By turning down the **Swing** control, the left tap is moved before the right tap up to a distance of half the delay time. By turning up the control, the right tap is moved up to a distance of half the delay time before the left tap.

Feedback

The **Feedback** control sets the amount of delay regenerations, caused by feeding the delayed signal back into the input. Higher values feed the delayed signal back in at a higher amplitude, leading to more regenerations of the input signal. Settings of 100% lead to indefinite regenerations until the value is reduced, and with good use of the built-in filtering can result in classic psychedelic, dubby analogue delay effects.

Hi (High-pass)
Lo (Low-pass)

These controls provide high-pass (**Hi**) and low-pass (**Lo**) filtering to each delay regeneration.

FX-Verb

FX-verb is a high-end algorithmic reverb device.

Pre-delay

The **Predelay** control introduces a delay between the dry sound and the reverberated output, creating a sense of space and distance.

Room Size

This control adjusts the size of the virtual reverberation chamber. Increasing the **Room Size** leads to a more pronounced and longer reverberation effect.

Decay

This control adjusts the length of the reverberation tail (which is also affected by the **Room Size** control).

Density

The **Density** control adjusts the density of reflections in the generated reverberation.

Damping Gain

The **Damping Gain** control adjusts the amount of damping applied by the Damping EQ before the output of the FXverb device. Increasing this control leads to more attenuation at the **Damping Freq**.

Damping Freq

The **Damping Freq** control adjusts the centre frequency of the Damping EQ positioned before the final output.

Output Early

The **Early** control adjusts the level of early reflections within the reverb output signal.

Output Late

The **Late** control adjusts the level of late reflections within the reverb output signal.

Pattern Delay

This effect features 2 delay lines - the first is a multitap delay sets the 'pattern' and the second is a simple stereo delay with feedback. The length of the 2 delay lines is distinct although they are both set as multiples of the host tempo using the **Step Length** setting.

• **Pattern delay line (multitap, no feedback)**: maximum length = 1/4 note * 16, min length = 1/64T * 16

• **Repeat/Regen delay line (simple stereo, feedback)**: max. length 1/4 note * 16 / minimum 1/64T * 1

The Pattern delay line cannot be fed back on itself without engaging the Repeat/Regen delay line.

Controls for each of 4 taps

Steps (1-16)

Each of the 4 taps can be placed on 1 of 16 'sequence' steps which, combined with the Step Length and Repeat parameters, define the 'pattern' in the Pattern Delay. This pattern comprises the first delay line.

Click the number of each tap to enable/disable it in the sequence.

Level
Pan
Pitch
Res

These controls set the volume, stereo pan position and filter settings for each of the 4 taps.

Blend

The Blend control crossfades between the following signals for each of the 4 delay taps:

Unfiltered - Low-pass - Band-pass - High-pass - Unfiltered

Try using modulation to sweep this control between the various filter types while delay taps are active and feeding back with the Repeat/Regen controls.

Global controls

Repeat

This parameter defines the length of the Repeat/Regen delay line which exists to feed the output back into the Pattern delay line's multiple taps. It is set in steps so its length is dependent on the **Step Length** parameter setting.

• With a **Repeat** setting of *8* and a **Step Length** of *1/16*, the pattern is fed back on itself every half-bar.

• With a **Repeat** setting of *8* and a **Step Length** of *1/8*, the pattern is fed back on itself every bar.

• With a **Repeat** setting of *8* and a **Step Length** of *1/32*, the pattern is fed back on itself every quarter-bar.

Deactivate the **Repeat** button to entirely disable the Repeat/Regen delay line.

Regen

This parameter sets the amount of feedback for the Repeat/Regen delay line.

RG Mix (Regen Mix)

The **RG Mix** parameter controls the signal mix which is routed to the feedback circuit - it mixes between the Pattern Delay line and the Repeat/Regen delay line signals.

HP, LP

These High-pass (**HP**) and Low-pass (**LP**) filters exist within the Repeat/Regen delay line's feedback path and are be used to filter out low and high frequencies in the feedback signal.

Step Length

The **Step Length** parameter defines the length of each Step, in terms of a BPM-division of the master tempo, used in both delay lines.

TinCanVerb

This effect is a recreation of a low-end room reverb unit, perfect for emulating 'cheap and nasty' onboard synth FX. Use the FX-Verb device for higher quality reverb.

Size

The **Size** control adjusts the size of the virtual reverberation room. Smaller rooms offer subtle ambience, while large rooms result in a more 'cavernous' and reflective effect.

Decay time

This control adjusts the **Decay time** of the reverb effect. Use shorter settings for subtle small room effects and longer times for special effects.

Damp

Increasing the **Damp** control results in less high frequencies in the effected signal, leading to a darker reverb sound. At least some damping is essential to avoid overly tinny and fatiguing results.

Pinch
Squeeze

The **Pinch** and **Squeeze** controls manipulate the shape of the virtual reverb room, leading to a variety of resulting effects. They make the reverb sound a lot more artificial and are useful for special effects.

Freeze

As the name suggests, this button 'freezes' the current reverb buffer and loops it indefinitely until the button is disabled. Modulating the **Freeze** control is useful for dubby special effects.

Freezer

This is a granular buffer-looping/freezing effect that is useful for glitchy and stuttery sounds.

Sync

With the **Sync** button activated, the **Length** and **Grain Size** parameters are set in tempo-based values relative to Strobe2's current tempo. Possible values range from 64th note to 2 bars, including dotted and triplet variations. With the button deactivated, these parameters are set in seconds.

Gate
Length

Once the **Gate** control is turned to the *On* position the Freezer effect starts recording audio from the input into a buffer, the size of which is dictated by the **Length** control (1-16 beats in **Sync** mode, up to 2 seconds with Sync deactivated).

The loop buffer is filled until the end of the time period defined by the **Length** parameter, after which subsequent incoming audio is ignored until the buffer is re-gated (**Gate** control set to *Off* and *On* again).

Grain Size

The audio in the loop buffer is divided up into slices, the size of which is dictated by the **Grain Size** parameter. This can be set in seconds or BPM units depending on the setting of the **Sync** button.

Smooth

The **Smooth** control introduces crossfading to each grain, relative to the size of the grain. At a setting of 100% the crossfading occurs over the entire length of the grain. At a setting of 0, no crossfading occurs.

Speed
Jump Manual
Jump Rand

Once the buffer has been filled, the Freezer loops the first grain – it 'freezes' the grain – while the **Speed**, **Jump Rand** and **Jump Manual** parameters are set to 0.

Increasing the **Speed** control plays through the grains sequentially starting with a grain between 0 and 16, set using the **Jump Manual** parameter, as long as **Jump Rand** = 0. A setting of 100% is normal speed.

Increasing the **Jump Rand** parameter results in the Freezer jumping to random grains instead of playing through them sequentially.

Scratch

The **Scratch** parameter scales the pitch of the loop just like a record on a turntable - the loop can be played forwards and backwards and everywhere in between.

7.4 Modulation FX

Phaser

The Phaser uses phase cancelling techniques (with the use of all-pass filtering) to create a series of peaks across the frequency spectrum. When these peaks are moved over time, a psychedelic sweeping effect is created.

Mode

The **Mode** control selects between a number of phaser responses. 4, 6, 8 and 12 stage phaser types are available, with positive or negative feedback. The number of stages refers to the number of all-pass filters within the algorithm.

Pitch

The **Pitch** control adjusts the centre frequency of the all-pass filters used in the Phaser algorithm.

Resonance

The **Resonance** control adjusts the amount of resonance (feedback) in the all-pass filters.

Sync

When the **Sync** parameter is set to *BPM*, the **Rate** control is set in tempo-based values relative to Strobe2's current tempo. With Sync deactivated, the Rate control is set in seconds.

Rate

The **Rate** control adjusts the speed of the internal LFO. In **Sync** mode, possible values range from 64th note to 2 bars, including dotted and triplet variations. With Sync deactivated, the available range is 31ms to 4 seconds.

Depth

The **Depth** control adjusts the amount of Pitch modulation from the Phaser device's internal LFO that modulates the **Pitch** parameter for classic phaser effect sounds.

Phase

The **Phase** control adjusts the phase between the dry signal and the all-pass filtered signal.

Flanger

The Flanger effect is a short modulated delay line with feedback to the input. It is used for a sense of movement and for psychedelic effects from the subtle to the extreme. It features a built-in sine LFO for modulating the flanging delay line.

Freq (Frequency)

The **Freq** control affects the speed of sine LFO modulation of the Flanger's delay time.

Depth

The **Depth** control adjusts the amount of modulation of the delay time.

Length

This control introduces an additional fixed delay time to the Flanger's delay line. It is a very short delay, ranging from 0 ms to 15 ms.

Feedback

This control adjusts the amount of the flanged signal that is fed back into the input.

Higher **Feedback** settings result in a more pronounced flanging effect. Settings over 50% lead to extreme comb filter type effects.

Spread

This control varies the panning of the left and right channel processed signals.

Phase

The **Phase** control offsets the phase of the internal LFOs for the left and right channels.

Invert

By default (with the **Invert** button deactivated), the flanged signal is in positive phase with the input signal.

Activating the **Invert** button inverts the flanged signal's phase in relation to the input signal.

A positive phase setting tends to lead to a more obvious flanging effect.

Chorus

The Chorus effect is a modulation effect that is pitch-based. It is used for thickening up sounds.

Rate

The **Rate** control adjusts the speed of pitch modulation by the built-in sine LFO.

Depth

The **Depth** control adjusts the amount of modulation away from the input signal's original pitch.

Spread

This control varies the panning of the left and right pitch-modulated signals.

Autofilter

This device features a state-variable multimode filter with a response of 12 dB/oct, capable of self-oscillation and featuring input and output drive stages.

It has a built-in envelope follower to modulate the filter's cutoff frequency relative to the input's amplitude. This function is modelled on an analogue full-wave rectified envelope follower. Additionally, the filter frequency can be modulated at audio rate by the input signal.

Mode

4 filter Mode settings are available, each offering different filtering functions relative to the cutoff frequency.

Low (Low-pass)

Low-pass mode allows through only frequencies below the cutoff frequency.

High (High-pass)

High-pass mode allows through only frequencies above the cutoff frequency.

Band (Band-pass)

Band-pass mode allows through only a band of frequencies around the cutoff frequency.

Notch

This is the opposite of a band pass - it allows through all other frequencies except a band of frequencies around the cutoff frequency.

Modulating a notch filter can give phaser-like results.

Drive

The **Drive** control sets the amount of signal gain before the filtering stage. As well as increasing the gain, using more drive results in a rich and complex interaction with the filter's resonance due to modelled non-linearities within it.

Out Drv (Out Drive)

The **Out Drive** control adjusts the gain of an OTA-type non-linear amp function at the output, in order to boost and colour the filtered signal.

Cutoff

The **Cutoff** control adjusts the cutoff frequency of the filter. It is measured in semitones.

Res (Resonance)

This control adjusts the resonance of the filter, accentuating the frequencies around the cutoff frequency set by the **Pitch** control.

High **Resonance** settings cause the filter to self-oscillate.

Attack

This control changes how quickly the envelope follower section responds to transients in the input signal. Longer **Attack** times mean that the filter takes a longer time to respond to amplitude changes.

Release

The **Release** control changes how quickly the envelope follower causes the filter to return to its original position when the input signal decreases after a transient.

Depth

The **Depth** control adjusts the amount of modulation of the filter's cutoff frequency by the built-in envelope follower.

FM

The **FM** control sets the amount of audio-rate modulation of the cutoff frequency by the input.

Amber Chorus

Amber Chorus provides a range of chorus algorithms derived from classic bucket-brigade delay (BBD) chorus circuits found in string-synthesizers and stomp boxes. These algorithms impart a psychedelic and thickening effect to the source signal and results in a 'blended' ensemble effect with a very different character to the regular Chorus effect.

Mode

6 different models are available using the **Mode** control with each providing their own unique sonic character:

- 1975
- 1977
- 1981
- 1984
- 1985
- 1986

Speed

This control adjusts the rate of Pitch modulation.

Spread

This control varies the panning of the left and right pitch-modulated signals.

Bright

The algorithms in Amber Chorus were derived from classic ensemble string synth chorus circuits. Such devices usually integrated additional fixed filtering to shape the tone of the output to be more string-like. Activating the Bright button bypasses these filters. With the button deactivated, the filters are applied and a darker sound is created at the output.

Amber Formants

Amber Formants is a 4-band formant filter consisting of 4 bandpass filters. It is useful for vocal-style filtering and for imparting string-like characteristics to sounds.

The formant filter is capable of self-oscillation – use the final mixer/amplifier controls to tame the sound so that it does not distort at higher resonance settings.

Freq (Frequency)
Gain

The formant filter comprises 4 band-pass filters, each with their own **Freq** (frequency) and **Gain** controls.

Resonance

The resonance of all bands is controlled by a single **Resonance** parameter

Scale

The **Scale** control adjusts the frequency of all 4 bands simultaneously (relative to their individual **Freq** settings).

Notch

The **Notch** button switches all bands to notch filter mode – in this mode, the **Gain** controls are inactive.

RingMod

Ring modulators are used for radical timbral shifts and experimental effects. It multiplies the input signal with an internal oscillator, creating interesting sidebands and inharmonic timbral changes in the output signal.

Mode

The **Mode** control adjusts the waveshape of the internal oscillator which is multiplied with the audio input to the effect. *Sine*, *Triangle*, *Saw*, *Square* and *Parabolic* oscillator shapes are available, as well as *White* or *Pink* noise.

Pitch

The **Pitch** control tunes the internal oscillator within a range of 1 octave.

Drive

The **Drive** control introduces an adjustable amount of distortion on the input signal. Overdriving the signal in this way changes the waveshape of the input, leading to further variations in the resulting effect.

Nonlinear RingMod

This device provides an alternative ring modulator circuit. Again, the input signal is multiplied with an internal oscillator to create complex sidebands and dissonant effects.

Drive

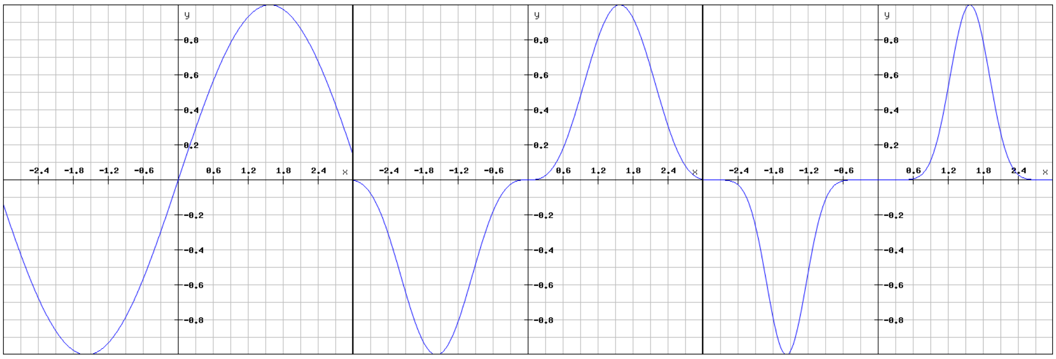
The **Drive** control adjusts the amount of overdrive on the input signal. Higher amounts of Drive affect the waveshape and harmonics of the input signal and lead to further variation in the timbre of the output.

Pitch

The **Pitch** control tunes the internal oscillator by +/- 6 octaves.

Mode

The **Mode** control adjusts the waveshape of the internal oscillator between *Sine*, *Triangle*, *Sine3* and *Sine9* shapes. The latter shapes represent $\sin(x)^3$ and $\sin(x)^9$ functions respectively. The following graphs show how their shapes differ from a regular sine shape.



Sine | Sine3: $\sin(x)^3$ | Sine9: $\sin(x)^9$

Tone

The **Tone** control introduces and adjusts a low-pass filter function to the output signal. Increase the control to increase the amount of filtering.

PhaseMod Array

The PhaseMod Array device provides 4 sine oscillators whose phase is modulated at audio-rate by the incoming signal. It is designed to provide audio-rate FM effects. When using higher amounts of the **Spread** and **Depth** controls, the device often produces best results with a monophonic synth source.

Pitch

The **Pitch** control adjusts the base tuning of 4 sine wave oscillators.

Stack
Spread

The **Stack** control dictates how many of the 4 sine oscillators are heard while the **Spread** control spreads the pitch of the 4 sine oscillators apart according to the **Mode** setting. These functions behave in a very similar way to the **Stack** and **Detune** functions in Strobe2's oscillator.

Width

The **Width** control spreads the 4 sine oscillators apart in the stereo field.

Mode

The **Mode** control changes the way the **Spread** control affects the tuning of each of the 4 sine oscillators.

The following table shows the deviation from the base tuning (set by the **Pitch** control) of each sine introduced by the **Stack** control with the **Spread** control at 100%:

	Mode A	Mode B	Mode C
Stacked sine 1	+0	+0	+0
Stacked sine 2	-12	-11	+11
Stacked sine 3	+12	+12	-12
Stacked sine 4	-5	+7	-4

Depth

The **Depth** control sets the amount of audio-rate phase modulation applied to the 4 sine oscillators from the audio input signal.

FreqShift (Frequency Shifter)

The FreqShift effect changes the pitch of a signal without preserving the harmonic information, resulting in very alien, abstract and clangorous timbres.

Pitch

The **Pitch** control adjusts the amount of frequency shifting by +/- 3 octaves.

Gain

The **Gain** control adjusts the level of the output signal.

7.5 Distortion FX

Bitcrusher

The Bitcrusher effect provides a type of digital distortion that occurs when the sample-rate and bit-depth of the audio is reduced. It simulates the sound of early samplers, useful for underground hip-hop and other 'lo-fi' styles.

Bits

The **Bits** control reduces the bit depth from a maximum of 16 bits to a minimum of 1 bit, which is effectively digital noise. The digital noise generated by the bit-reduction process is referred to as quantisation noise.

Frequency

The **Frequency** control adjusts the sample rate frequency of the audio processed by the effect and ranges from a maximum of 100 kHz to a minimum of 1 Hz.

Lower sample rates result in an aliasing effect on the processed audio.

Drive

The **Drive** control adjusts the amount of drive in an OTA-type distortion stage after the crossover filters. This allows gain and colour to be added to the signal before it is processed by the **Bit** and **Freq** processes.

LPF (Low-pass filter)
HPF (High-pass filter)

These controls provide high-pass and low-pass for isolating a part of the signal before the sample-rate and bit-depth reduction stages.

These filters act as crossover filters – the active frequency band is processed by the distortion circuit with its level adjustable via the **Dirty** control. The frequencies that are filtered out before the distortion stage are accessible via the **Clean** control.

Dirty & Clean

The **Dirty** control sets the amount of post-distortion signal that is heard at the output.

The **Clean** control sets the amount of the clean signal, which is comprised of the signals filtered out by **LPF** and **HPF** filters.

Please note that these are not 'wet' and 'dry' controls – use the standard **Mix** control at the upper-right of the effect interface in order to mix between the pre- and post-effect signals.

Tone

The **Tone** control provides a simple -6 dB/oct low-pass filter for the **Dirty** signal after the bitcrushing process. It rolls off unwanted high frequencies that may have been generated in the signal as part of the distortion effect.

Drive

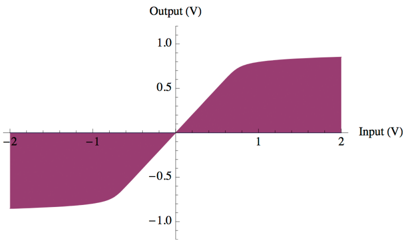
Drive is a versatile DCAM-modelled overdrive/distortion effect.

Drive

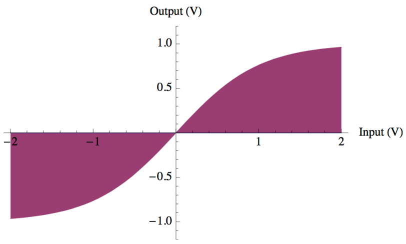
The **Drive** control sets the amount of distortion that takes place.

Mode

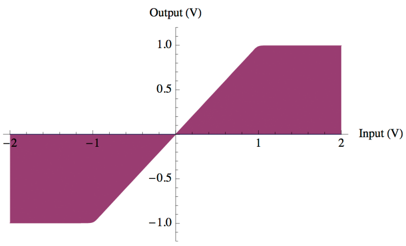
The **Mode** drop-down menu selects between 4 distortion types:



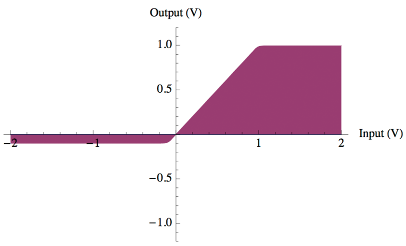
- *Diode*



- *OTA*



- *OpAmp*



- *HalfRect (half-rectified)*

LPF (Low-pass filter)
HPF (High-pass filter)

These controls provide high-pass and low-pass filters before the distortion stage to shape the tonal characteristics of the signal going into the drive circuit.

For example, it may be useful to distort the high end of a sound while leaving the low end unchanged.

These filters act as crossover filters – the active frequency band is processed by the distortion circuit with its level adjustable via the **Dirty** control. The frequencies that are filtered out before the distortion stage are accessible via the **Clean** control.

Dirty & Clean

The **Dirty** control sets the amount of post-distortion signal that is heard at the output.

The **Clean** control sets the amount of the clean signal, which is comprised of the signals filtered out by **LPF** and **HPF** filters.

Please note that these are not 'wet' and 'dry' controls – use the standard **Mix** control at the upper-right of the effect interface in order to mix between the pre- and post-effect signals.

Tone

The **Tone** control provides a simple -6 dB/oct low-pass filter for the processed signal after the distortion stage. It rolls off unwanted high frequencies that may have been generated in the signal as part of the distortion effect.

DirtyDAC

This effect emulates the behaviour of an old Digital-to-Analogue Converter with low-fidelity characteristics. It offers an alternative processing algorithm to the Bitcrusher device and imparts a similar type of effect, although the character is very different.

Frequency

The **Frequency** control sets the sample-rate frequency of the DAC model. It can be adjusted between 1kHz (fully anti-clockwise) and 10kHz. Lower settings lead to more pronounced aliasing effects.

Anti-alias

The **Anti-alias** control adjusts the amount of anti-aliasing filtering for the output signal. Turning up the control reduces the cutoff frequency of a low-pass filter, reducing audible sidebands and aliasing effects caused by the sample-rate reduction process.

Bits

The **Bits** control sets the bit-depth of the modelled DAC - it can be set between 6 and 16 bits. Lower values introduce more noise into the signal and reduce the dynamic range.

8 Quick-presets, Morphing and Randomizing

Quick-presets overview

The upper-left part of the Strobe2 interface contains the Quick-preset, Morph/Freeze and Randomizer controls.

• Loading a new preset to the current slot

These functions involve a total of 10 slots which each hold an entire preset (state of the Strobe2 instrument). Quick-preset slots can be considered as 'edit buffers' - any editing of Strobe2 always applies to the currently selected Quick-preset slot. 1 of these 10 slots is always active as the *currently selected Quick-preset slot* and highlighted in *yellow*.

• When a preset is loaded from the Browser, it is loaded into the currently selected Quick-preset slot.

Main 8 Quick-preset slots

The 8 main Quick-preset slots can each be used to hold a full Strobe2 preset. Each slot can either be filled in either of the following ways:

• Select a Quick-preset slot by clicking it and load a preset from Strobe2's Browser - the preset is loaded to the Quick-preset slot, leaving other slots unaffected

• Copy contents between Quick-preset slots using the Quick-preset context menu (right-click on the slot)

Freeze slot

The Freeze slot is used for creating new sounds using the morphing functions with sounds within the 8 main Quick-preset slots.

It also features a **History** function on its Quick-preset context menu which forms a multi-level undo function for its contents.

Randomizer slot

The Randomizer slot is used for creating new sounds using the Randomizer pad. Like the Freeze slot, it features a **History** function to return to previous states.

Using Quick-presets

Quick-presets are intended to be used for a variety of functions.

Fast switching between 8 presets for live use scenarios

Quick-presets allow extremely fast switching between sounds. If switching between Quick-presets with the same Effects page contents (or when using the Lock function to prevent them being changed), the transition to the new preset occurs extremely quickly compared to loading presets from disk using the Browser (under 1ms vs. 10+ ms).

• Quick-preset slots can be selected using MIDI messages (a range of notes, program changes or MIDI CC #0).

• Try also creating copies of a sound to other slots using the Quick-preset context menu. Then program variations which can subsequently be recalled in real time via MIDI.

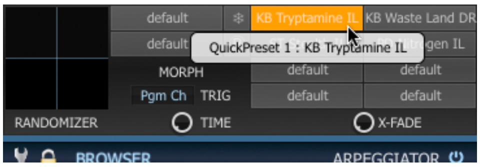
Exploring variations of sounds using the morphing and Freeze functions

With the morphing and Freeze functions, it is possible to morph between 2 Quick-preset slots, creating new sounds with Freeze operations during mid-morph states. The results can be subtle or radical depending on whether similar or wildly different presets are used.

Exploring variations of sounds using the Randomizer functions

The Randomizer provides a different way of exploring new sounds - it creates 4 random seeded variations which can then be explored using the Randomizer Pad.

Selecting Quick-preset slots with the mouse or computer keyboard



• Selecting another slot with the mouse

Click any Quick-preset slot to switch to it, selecting it as the current slot. If the **Time** control is set to *0*, the preset within the slot is recalled immediately.

If the **Time** control is increased, a morph occurs for the duration of the Time setting. During this period, parameters morph to the values in the new preset. Morphing is discussed in detail in the next section.

To select a Quick-preset slot with the mouse immediately regardless of the **Time** control setting, select it while holding down the ALT key.

Quick-preset slots can also be selected using the following keyboard cursor shortcuts (in DAW/host applications which do not intercept keyboard input):

←	Previous Quick-preset slot
→	Next Quick-preset slot

Selecting Quick-presets via MIDI

Quick-presets can also be selected using the following MIDI messages, selectable using the **Trig** menu.

• *Pgm Ch* (Program change)

MIDI Program change messages 1 to 10.

• *CC #0* or *#1* (MIDI continuous controller #0 or #1)

MIDI CC#0 or #1 is used to address Quick-preset slots 1-10 proportionally over the CC value range (0-127).

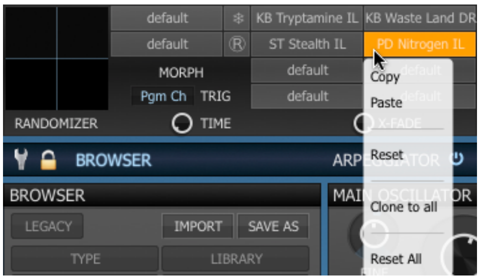
• *Oct -1 (12) / Oct 0 (24) / Oct 1 (36)*

Notes C to A in 1 of 3 octaves, starting from MIDI note number 12, 24 or 36, on MIDI channel 1.

• *Ch 16 (36)*

Notes C1 to A1 starting from MIDI note number 36, on MIDI channel 16.

Quick-preset context menu



Right-click on any Quick-preset slot (or the **Freeze** button) to display the Quick-preset context menu.

Copy

This function copies the Quick-preset slot's contents and stores them in the clipboard.

Paste

This function overwrites the Quick-preset slot's contents with the contents of the clipboard.

Reset

This function clears the Quick-preset slot and replaces its contents with the Init patch.

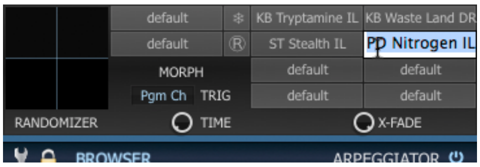
Clone to all

This function copies the contents of the Quick-preset slot to all 7 other main Quick-preset slots.

Reset All

This function clears the contents of all 8 main Quick-preset slots.

Renaming the preset within a Quick-preset slot



Double-click any Quick-preset slot (including the Freeze and Randomizer slots) to enter a new name for the preset within it. Note that this action does not save the preset - select the slot and use the **Save As** function in the Browser or Preferences menu in order to save the slot's contents as a preset.

Saving/recalling Quick-preset setups

Because they involve compiling multiple single Strobe2 presets, Quick-preset setups exist *outside Strobe2's preset format* - any time a preset is saved using the Save As function in Strobe2's Browser or Preferences menu, only the currently active Quick-preset slot's contents are saved.

However, Quick-preset setups can be saved and recalled in the following situations:

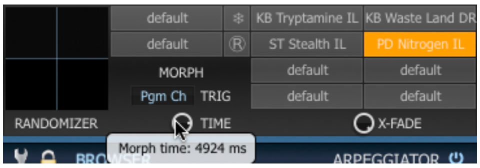
• Within the host project file when Strobe2 plugin instances are used in a host/DAW.

• Within host/DAW preset files - such as *.FXP in Cubase/Nuendo or *.AUpreset in Logic for example)

8.1 Morphing and Freezing

Morphing

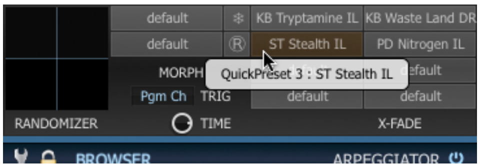
Timed morphing using the Time (Morph Time) control



• Setting the **Morph Time**

When a Quick-preset slot is selected, the transition from the current state to the slot's contained preset - called the *morph destination* - occurs over a period of time defined by the **Time** (**Morph Time**) control.

During this period, Strobe2's parameters are moved continuously in real time to new positions as dictated by the new preset.



• A timed morph in progress

If Strobe2 is played via MIDI (or with a locked Arpeggiator) during this period, the effect of the parameters moving to new positions is heard.

When the Quick-preset slot is selected - via MIDI as described above or by clicking it as shown on the left - the yellow highlight for the newly selected slot fades in over the duration of the **Morph Time**.

Interrupting a timed morph

• If any controls are adjusted while a timed morph is in progress, they are excluded from the morph (which otherwise continues as normal) and their final positions are added to the state stored in the *morph destination* Quick-preset slot.

• If a new preset is loaded from the Browser while a timed morph is in progress, the morph process is cancelled and the new preset is loaded to the *morph destination* Quick-preset slot.

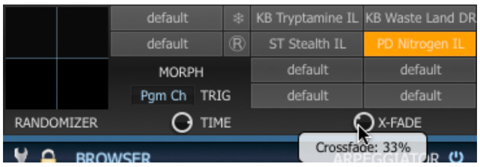
Manual morphing with the X-Fade control

The main 8 Quick-preset slots are arranged in 2 adjacent banks of 4.

If a Quick-preset slot in the left bank is currently selected and a slot on the right is subsequently selected, the **X-Fade** control is turned fully to the right at the end of the morph (and vice versa). It is always fully turned towards the currently selected Quick-preset slot and is visible *only* when 1 of the 8 main Quick-preset slots are selected - it cannot be used when either the Freeze or Randomizer slots are selected.

The **X-Fade** control can be used to manually morph towards most settings of the Quick-preset slot immediately to the right or left of the currently selected slot. This manual morphing process excludes both of the following:

• TransMod slot assignments (but not modulation amounts within TransMod slots)
• Effects page contents



The selected Quick-preset slot *does not change* - the original slot remains selected even with the **X-Fade** control turned fully away from it.

While a manual morph is in progress (when the **X-Fade** control is at any setting except fully turned towards the currently selected Quick-preset slot):

• Using the **Freeze** button adds the current state of Strobe2 to the Freeze slot and the Freeze slot is selected as the current Quick-preset slot

• Using the **Save As** function adds the current state of Strobe2 to the Freeze slot, the Freeze slot is selected as the current Quick-preset slot and its contents are saved as a preset

• Moving any Strobe2 parameters results in the new state being added to the Freeze slot and the Freeze slot is selected as the active Quick-preset slot

• Loading a new preset from the Browser results in the preset being loaded to the currently selected Quick-preset slot and the **X-Fade** control is fully turned towards it

See below for more details of the Freeze slot and its operation.

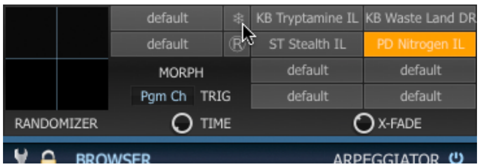
The **X-Fade** knob is not available while a timed morph is already in progress.

Selecting Quick-preset slots without morphing

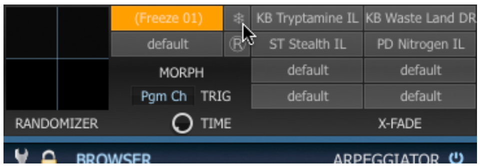
To switch to a preset without morphing (and without reducing the **Morph Time** control to 0), click the Quick-preset slot while holding down the ALT key.

Using the Freeze slot

The Freeze slot (which also features a corresponding **Freeze** button to its right) is a special Quick-preset slot which stores 'frozen' mid-morph states in order to create new sounds. The most immediate way to use the Freeze slot is as follows:



Create a manual **X-Fade** towards the adjacent Quick-preset as shown previously and then click the **Freeze** button.



The current state of Strobe2 is 'frozen', stored in the Freeze slot and named *(Freeze 01)*. The **X-Fade** control is not accessible with the Freeze slot selected.

The Freeze slot is filled with the current state of Strobe2 and selected as the active Quick-preset slot in all the following situations:

• The **Freeze** button is clicked while a manual **X-Fade** morph is in progress (when the **X-Fade** control is at any setting except fully turned towards the currently selected Quick-preset slot) - as shown above

• The **Freeze** button is clicked while a timed morph towards a Quick-preset is in progress

• The **Save As** function is used while an **X-Fade** morph is active - the frozen state is also the state which is saved

• A Strobe2 parameter is adjusted while a **X-Fade** morph is in progress - the current state of Strobe2 is saved to the Freeze slot along with any manually adjusted controls' final positions

Selecting the Freeze slot

The Freeze slot can be selected at any time, like any other Quick-preset slot, in order to recall its current contents and without creating a new Freeze state.

While the Freeze slot is selected, any new preset loaded from the Browser is loaded into it.

Freeze slot History

When the Freeze slot is filled with a new state, the previous state is stored in the Freeze slot History. Right-click on the Freeze buffer slot to to display a list of sequentially stored Freeze slot states - click any state to recall it.

Using Freeze slot contents

When the Freeze slot is selected, its contents can be saved as a preset in the same way as any other Quick-preset slot. Click the **Save As** button or use the **Save As** function in the Preferences menu to proceed.

It is also very useful to copy Freeze slot contents to other Quick-preset slots in order to compile variations or to perform further editing.

Using Locks during morphing

Please note that while transitions of continuous parameters are smoothly interpolated during morphing operations, switched controls and differing TransMod slot assignments may produce abrupt changes or clicks. It is also possible to encounter clicks when the contents of the Effects page are morphed.

Using Locks on certain parts of Strobe2 such as the Effects page and TransMod slot setups can be very useful during morphing operations as some causes of abrupt changes and clicks can be avoided - these 2 elements are always locked during manual **X-Fade** morphing as mentioned above.

When simply using the morphing functions to experiment with creating new sounds, clicks produced during the process may not be a concern.

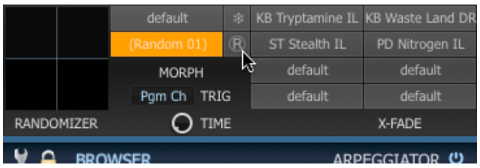
8.2 Randomizer

Using the Randomizer slot

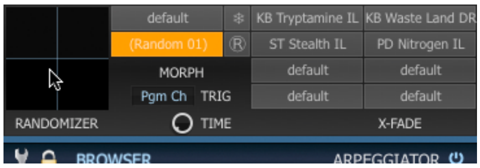
The Randomizer slot (which also features a corresponding **Randomizer** button to its right) is another special Quick-preset slot which is used alongside the Randomizer Pad for creating randomized variations of sounds.

To use the Randomizer:

1



Either click the Randomizer button...



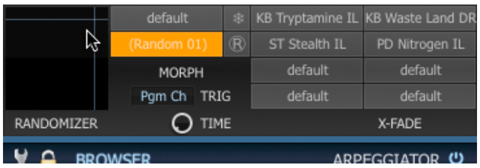
... or click and keep the mouse button held down on the Randomizer Pad

At this point:

• The current state of Strobe2 is copied to the Randomizer slot and given a new *(Random)* name, in a similar way to clicking the **Freeze** button to populate the Freeze slot. This creates an 'undo' point to return to the original state using the History menu if desired after creating a random variation.

• The Randomizer slot is selected as the active Quick-preset slot

2



Click and hold on the Randomizer Pad if this was not already done in step 1.

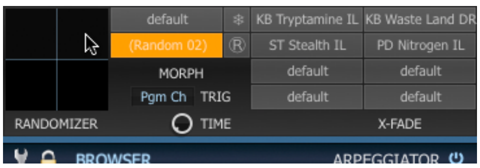
Then, keeping the mouse button held down, drag within the Randomizer pad to explore random variations.

Strobe2 can, of course, be played during this process.

As soon as the Randomizer pad is clicked, the 4 corners of the pad are re-seeded with new random parameter values, with the original state located at the centre.

Dragging the mouse towards each corner gradually morphs towards its random values from the original state.

3



Release the mouse at the desired point to store the live state as the new contents of the Randomizer slot, adding the previous contents of the slot to its slot History.

Selecting the Randomizer slot

Like the Freeze slot and any other Quick-preset slot, the Randomizer slot can be selected at any time in order to recall its current contents and without creating a new Randomized variation.

While the Randomizer slot is selected, any new preset loaded from the Browser is loaded into it.

Randomizer slot History

When the Randomizer slot is filled with a new state, the previous state is stored in the Randomizer slot History. Right-click on the Randomzer slot to to display a list of sequentially stored Randomizer states - click any state to recall it.

Using Randomizer slot contents

• When the Randomizer slot is selected, its contents can be saved as a preset in the same way as any other Quick-preset slot - click the **Save As** button or use the **Save As** function in the Preferences menu to proceed.

• It is also very useful to save Randomizer slot contents to other Quick-preset slots using the Quick-preset context menu in order to compile variations or to perform further editing.

Using Locks during Randomizer operations

Locks can be very useful during Randomizer operations - for example, it may be convenient to keep the Pitch control, TransMod slot assignments or Effects page contents locked.

9 Preferences

The preferences store global settings that are not stored within presets. These also include MIDI CC Learn mode assignments.

Click the **Preferences** button to show the Preferences menu.

For any changes to persist when Strobe2 is launched in future, use the **Save preferences** function.

Save as

This function duplicates the **Save as** button in the Browser.

Reset to default
Save preferences
Reset all preferences

All preferences settings can be made for the current session without affecting their default states when Strobe2 is launched in the future. If any of the settings for any Strobe2 instance in the current session differ from the defaults, these settings are saved per-instance in the host/DAW session.

However, they are *not* saved with Strobe2 presets using the **Save As** function.

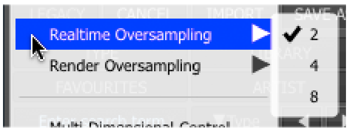
To apply the current preference settings to future instances when Strobe2 is launched, use the **Save preferences** function.

This function also saves the current MIDI Learn assignments for Strobe2's controls (including the **Perf1**, **Perf2**, **Base X** and **Base Y** controls).

If the preference settings are changed from the defaults, to switch back to the saved default states without relaunching Strobe2, use the **Reset to default...** function.

The **Reset all preferences** function resets the preferences to their factory default values.

Realtime and Render Oversampling



These settings relate to the oversampling in the synth engine. Higher oversampling sounds better but uses more CPU!.

The **Realtime Oversampling** setting relates to the oversampling used for normal realtime operation. Set this as high as the system's CPU (and the number of voices) allows.

The **Render Oversampling** setting relates to 'offline' mixdowns and 'freeze' operations in hosts that provide this feature. This setting can be set to very high levels regardless of the system CPU for extremely high sound quality. However, higher settings result in longer render times for offline mixdown/freeze operations.

Multi Dimensional Control

When this function is activated, a special mode becomes active which supports controllers which transmit 1 note per MIDI channel to facilitate messages such as polyphonic pitch-bend and also additional multi-dimensional controllers such as polyphonic aftertouch if desired.

Example controllers include the ROLI Seaboard and Roger Linn Linnstrument.

In this mode, notes received on multiple MIDI channels are each assigned to an individual voice (up to 16 MIDI channels can be used, with single-note polyphony on each channel). In addition, the following MIDI messages are accepted per MIDI channel and applied discretely to each individual voice to which the MIDI channel is assigned:

• Pitch-bend (also hard-wired to drive the PerfX+ TransMod source as well as the Euclid processor's **Base X** value)

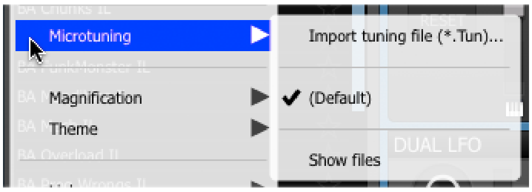
• MIDI CC #74 (hard-wired to drive the PerfY+ TransMod source as well as the Euclid processor's **Base Y** value)

• Mod-wheel (hard-wired to drive the **Perf1** control and Perf1+ TransMod source)

• Mono and Poly pressure (channel and poly aftertouch, hard-wired to drive the **Perf2** control and Perf2 + TransMod source)

When in this mode, MIDI Learn mode can be used for the **Perf1**, **Perf2**, **Base X** and **Base Y** controls in order to additionally assign these controls to monophonic controllers on a single MIDI channel. However, the Pitch-bend, Expression, Mod-wheel or Mono/Poly pressure messages cannot be used for this.

Microtuning

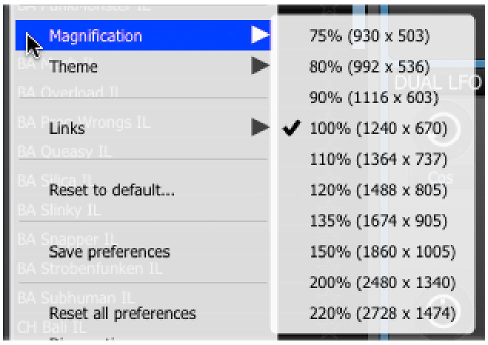


The **Microtuning** sub-menu allows several operations for using microtonal tuning files (.Tun files).

The **Import tuning files** function imports .Tun files to Strobe2's internal database of tuning files. The menu of scales allows selection from the available imported files.

To return to regular 12-tone scale operation, select *(Default)* in the menu. The **Show files** function opens the folder on disk in which tuning files are stored.

Note that Strobe2's internal handling of pitch is always in an equal-temperament 12-tone scaling - it can be considered in a similar way to the 1 volt per octave system used in modular synthesizers. Microtuning is applied during the point at which MIDI input is converted into 'pitch volts' - control signals that dictate keytracking values which set the osc pitch when being assigned to a voice.



Any additional adjustment and modulation within Strobe2 of pitch and other keytracked parameters is conducted within its normal equal-temperament 12-tone scaling.

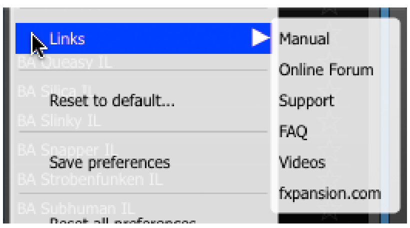
Magnification

Strobe2 features a vector-drawn interface which can be scaled to a variety of sizes, selectable in the **Magnification** sub-menu.

Theme

This sub-menu allows toggling between 2 available interface themes for the Strobe2 interface - *Dark Theme* and *Light Theme*.

Links

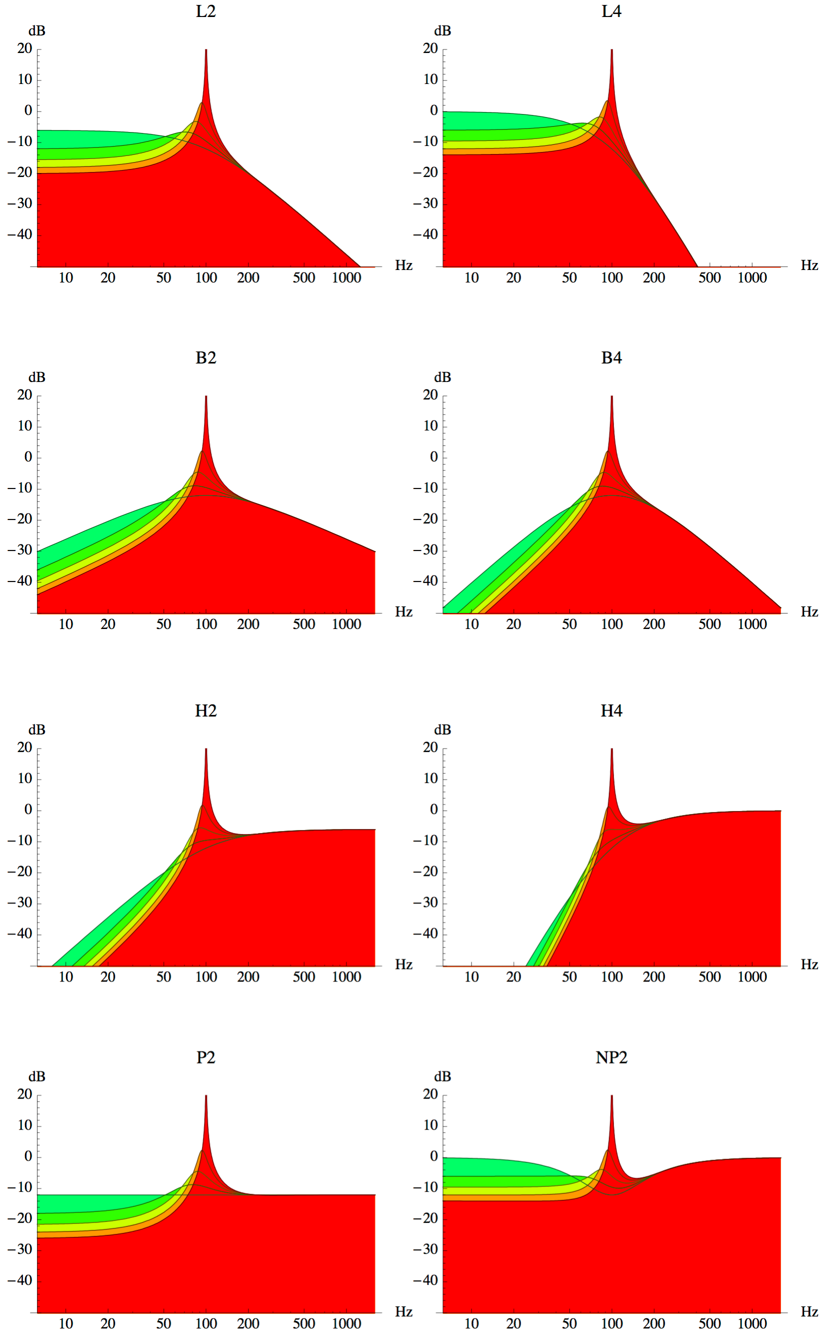


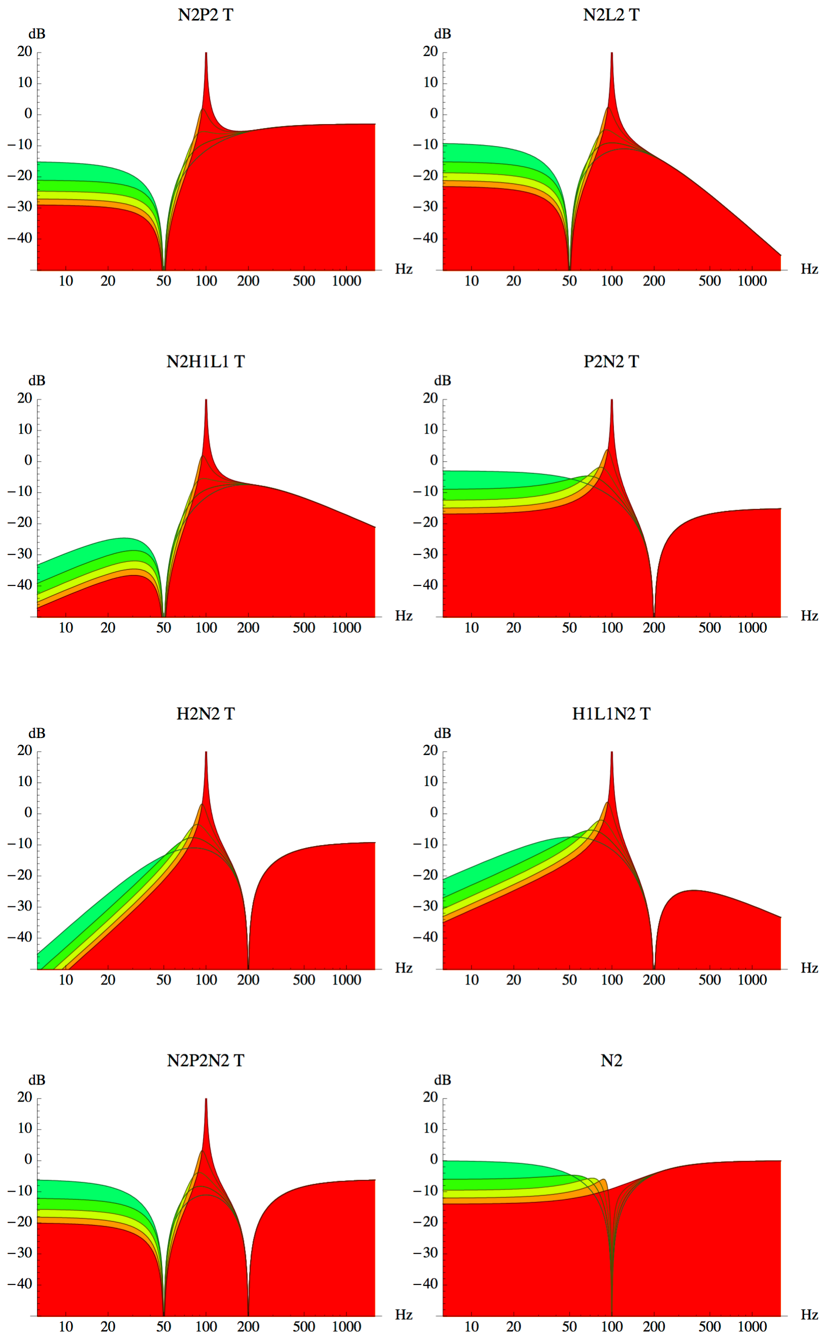
This sub-menu provides various web-links which open in the system's default browser.

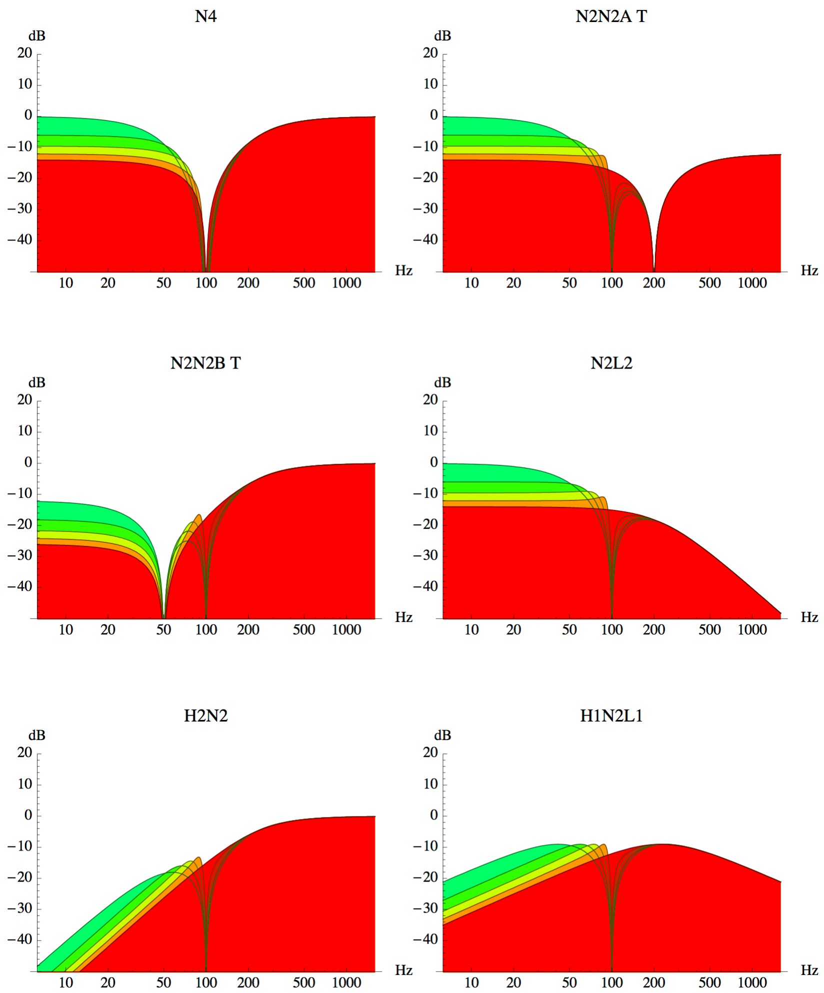
Diagnostics...

This function reports CPU capabilities and performance - our tech support staff may instruct to use this function during troubleshooting.

10 V.C.F. Mode responses




N2P2 T
N2L2 T
N2H1L1 T
P2N2 T
H2N2 T
H1L1N2 T
N2P2N2 T
N2
dB
20
10
0
−10
−20
−30
−40
Hz
10
20
50
100
200
500
1000


N4
N2N2A T
N2N2B T
N2L2
H2N2
H1N2L1
dB
Hz
20
10
0
−10
−20
−30
−40
10
20
50
100
200
500
1000