State Zero

Reference Manual

Contents

Connections

Power

The synthesiser is powered by a regulated 9V DC power supply, connected by a 2.1mm DC plug tip positive polarity. The power supply needs to be tightly regulated and should not exceed 10V at any time. Earthing is not recommended.

Display

An SXGA video output is available on a standard HD-15 connector. The 1024x1280 pixel image is oriented 90 degrees counter-clockwise from the typical format, and is often described as the portrait orientation (as from the typical landscape orientation). Connections to this socket should only be made when the power is switched off at the wall for both State Zero and the monitor.

Audio

The stereo inputs and outputs are via balanced ¼” TRS plugs. Both inputs and outputs are DC coupled to allow interfacing with CV signals to/from modular synthesisers.

Midi

Midi input and output is on standard 5 pin 180 degree DIN connectors. The synthesiser only responds to messages on channel 1.

Patchfield

The patch points of the synthesiser are connected with 3.5mm TS plugs. They do not provide or accept analog signals and should not be patched to or from any other piece of equipment. Within their field, connecting outputs together will cause no damage but may patch to unpredictable locations. Also each output has infinite fanout and can be connected to as many inputs as needed without buffering.

Module Reference

Introduction

State Zero represents a new class of synthesiser, bringing together the complete flexibility in the signal path of patchable systems with modern polyphony and memory capabilities.

Signal flow is handled in the traditional method using physical patch cables to connect modules together. The entire system follows the convention of signal flow being from left to right, with inputs on the left of modules and their output being on the right.

Patching is simplified by the absence of “multiples”, rather every input and output is a pair of connected jacks allowing a signal to be endlessly chained between inputs.

Most modules have several parameters that are continuously variable, each of these are available as a pair of knobs and a signal input. Called “control voltages” in conventional synthesisers there are no restrictions to signal routing and these will be referred to as modulation inputs. The pair of knobs set the base level of the parameter (right) and the amount by which the modulation input signal will vary the parameter (left).

Oscilloscope

The most prominent function of the video output is a dual timebase, 3 channel oscilloscope.

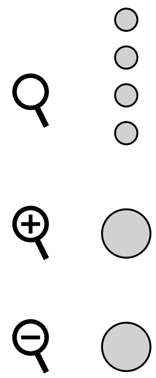
The primary channel (green trace) follows whatever patch connection was last made, allowing visualisation of any signal within the system. Triggering is fixed and occurs on positive going zero crossings, while the timebase is automatically adjusted to maintain several cycles on the screen at any time. If the signal is too slow to effectively display as a periodic waveform, the oscilloscope will ignore triggering and operate in roll mode.

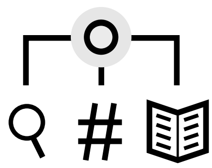
A period counter is integrated with the primary channel to allow frequency measurement or tuning, and operates from the same positive going zero crossing trigger. The output is always displayed in microseconds, and is disabled while the oscilloscope is in roll mode.

Further, a simple spectrum analyser (red trace) is available on the primary channel. Operating from the waveform captured on screen it is scaled in decibels vertically and octaves horizontally. Again it is disabled while the oscilloscope is in roll mode.

The second timebase of the oscilloscope displays the left audio output (blue trace) right audio output (yellow trace). Triggering is again fixed and occurs on positive going zero crossings of the left audio output. An automatic timebase the same as the primary channel is used, but operating independently.

These two channels are also applied to an X-Y trace (red) in the upper left of the display, where the X and Y directions are driven by the left and right channels respectively. This can be used for tuning oscillators by Lissajous curves, or plotting transfer functions by viewing the relationship of the input and output of a module.





Master

Zoom

Displays the zoom level of the last knob adjusted.

Increment/Decrement

Zoom

Increases/decreases the zoom level of the last knob adjusted. Pressing both together zeros the value.

Number

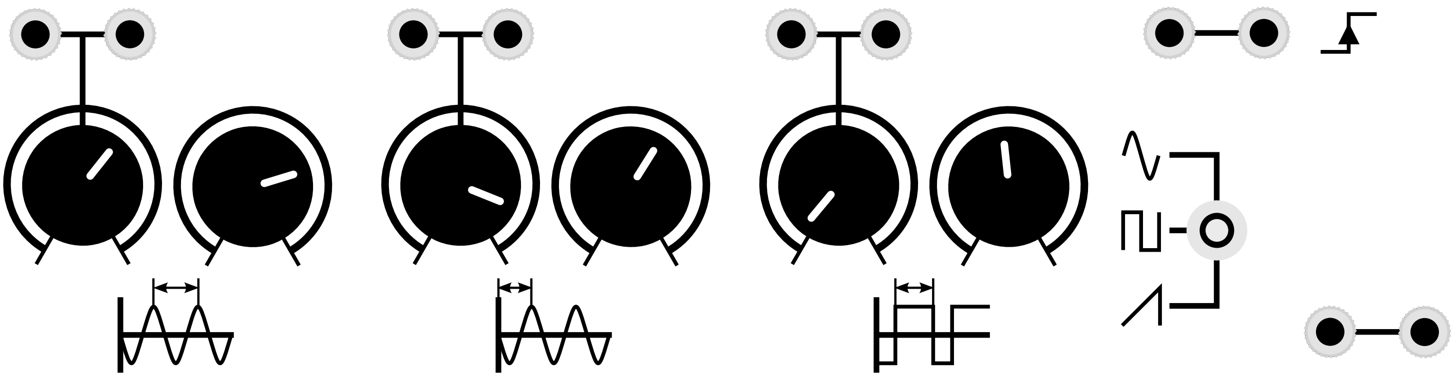
Increases/decreases the value of the last knob adjusted by a single step, regardless of zoom control. Pressing both together zeros the value.

Program

Selects program number.

Mode

Selection of zoom, number, and program modes.



Oscillator

Frequency

Exponential response, modulation of 50% from the master pitch source achieves natural scaling and A440 tuning. Offset is displayed in Octave/Semitone/Cent quantities.

Phase

Advances or retards phase of the oscillator up to 1 period. Offset is displayed as a percentage.

Symmetry

Square:
Pulse width from 0 to 1, expressed as a percentage of full range from 0.5.

Sawtooth:
Ramp shape from negative going sawtooth to positive going sawtooth, expressed as a percentage of full range from symmetric triangle.

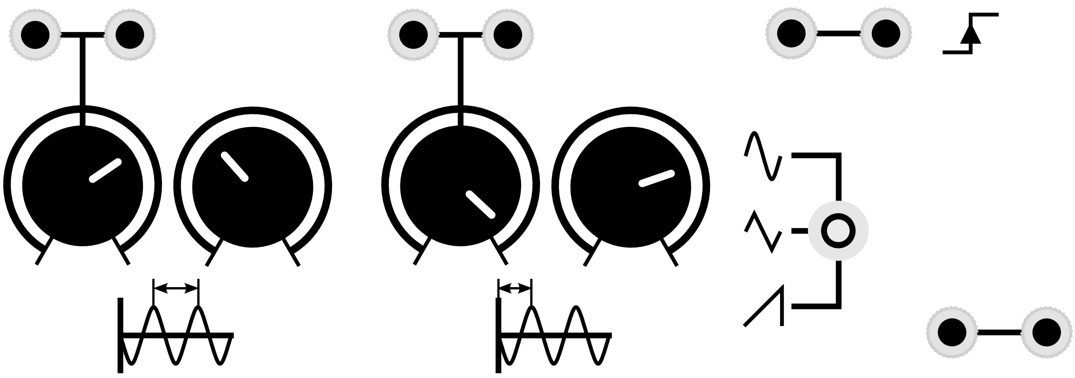
Sinusoid:
No effect.

Waveform

Output selection of sinusoid, ramp, or square waveform.

Synchronization

Resets the phase of the oscillator to zero plus/minus phase offset. Triggered by a positive going edge through zero.



Low Frequency Oscillator

Frequency

Exponential response, modulation of 50% from the master pitch source achieves natural scaling and A440 tuning down 6 octaves. Modulation of 50% from the tempo rate source achieves an eighth note period. Offset is displayed in Octave/Semitone/Cent quantities.

Phase

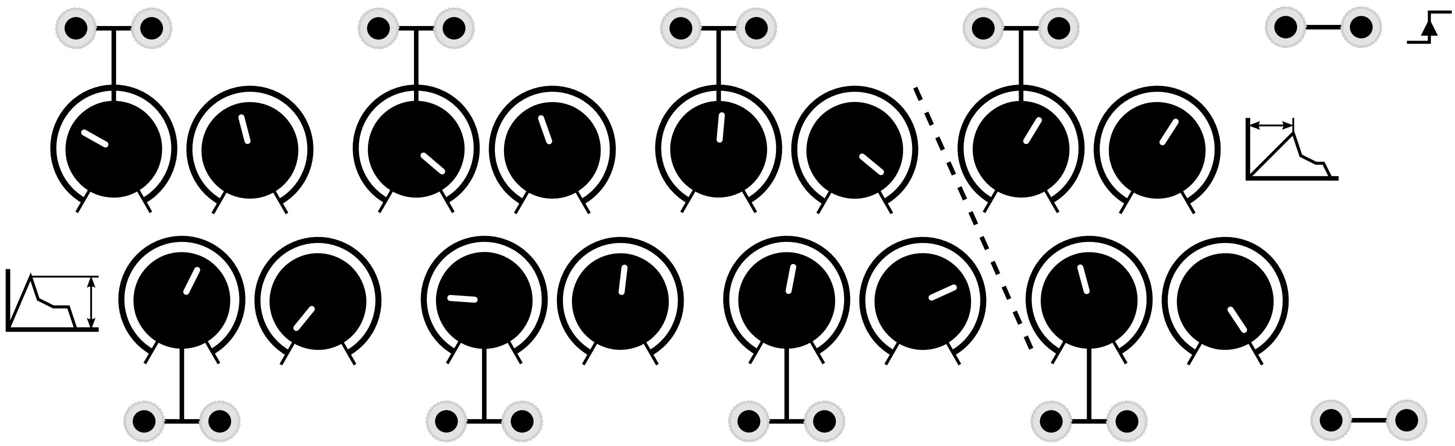
Advances or retards phase of the oscillator up to 1 period. Offset is displayed as a percentage.

Waveform

Output selection of sinusoid, triangle, or positive going sawtooth waveform.

Synchronization

Resets the phase of the oscillator to zero plus/minus phase offset. Triggered by a positive going edge through zero.



Envelope

Rate

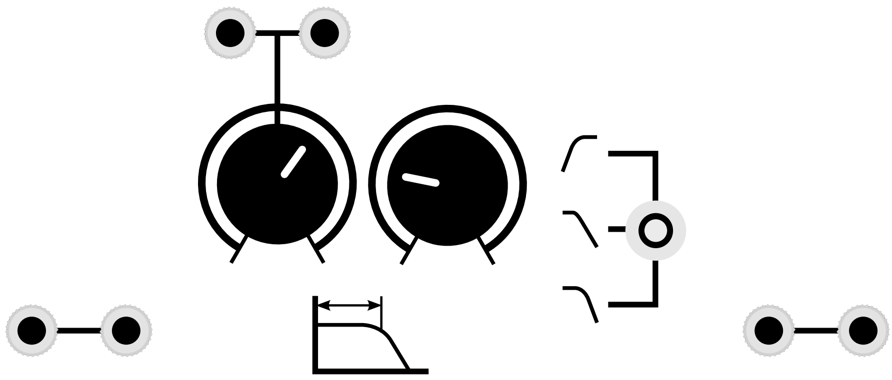
Exponential response, expressed in nominal units from 0 (infinite hold) to full scale per sample.

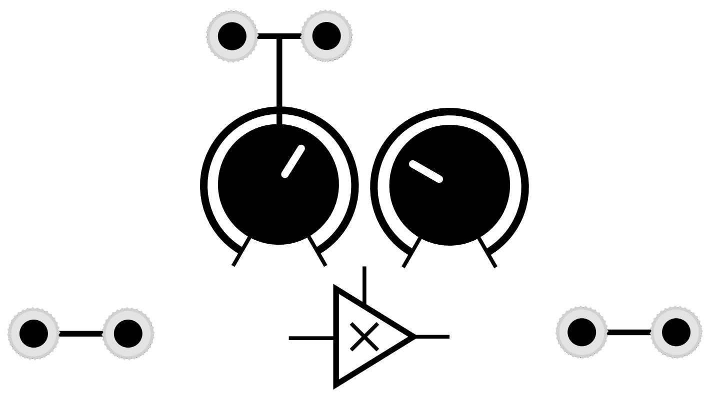
Level

Final level of envelope stage expressed as a percentage of full scale from zero.

Synchronization

The first stage of the envelope is triggered on a positive going edge through zero and two stages follow it. The final stage is triggered on a negative going edge through zero.





Filter

Frequency

Exponential response, modulation of 49.249% from the master pitch source achieves approximately natural scaling about A440 down 1 octave 9 semitones and 23 cents while self oscillating at 24dB/octave. Offset is displayed in Octave/Semitone/Cent quantities.

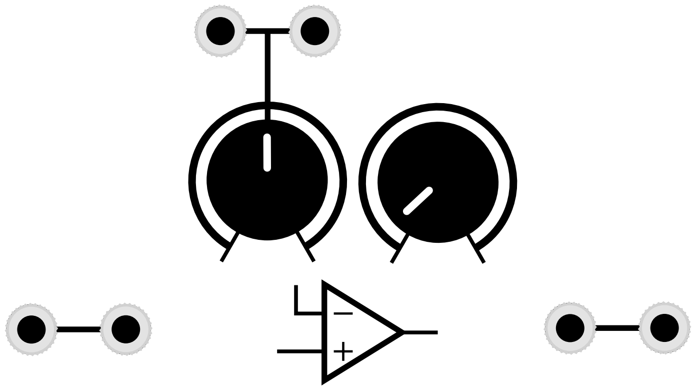
Slope

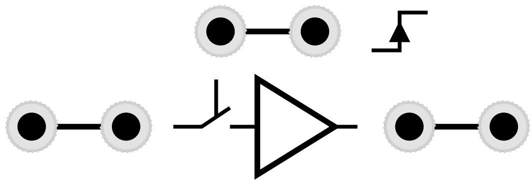
Output selection of 24dB/octave high pass, 12dB/octave low pass, or 24dB/octave low pass.

Amplifier

Level

Exponential response, expressed as a percentage over its range.





Comparator

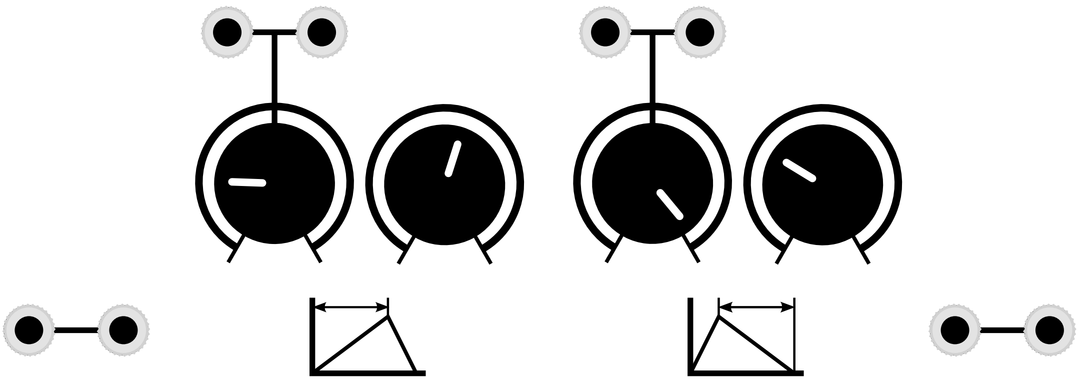
Level

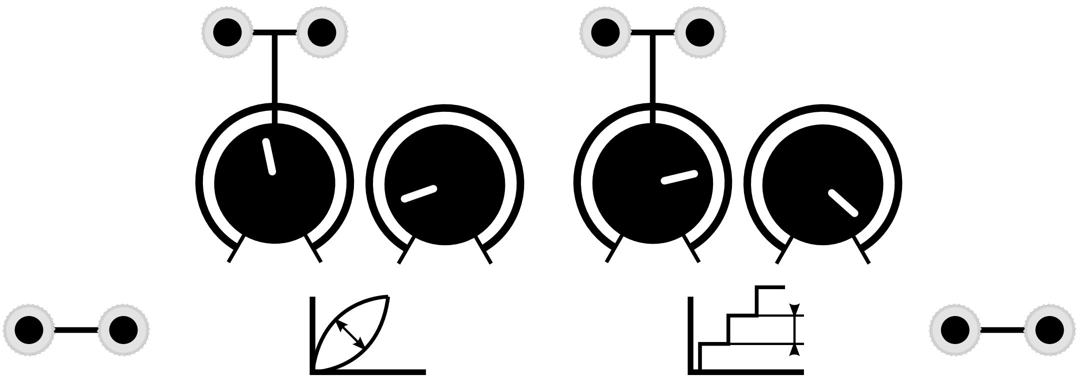
Threshold for comparison, output is positive full scale for input > threshold and negative full scale for input < threshold.

Sample and Hold

Synchronization

The input value is captured on a positive going edge through zero.





Slew Rate Limiter

Up

Exponential response, expressed in nominal units from 0 (infinite hold) to full scale per sample.

Down

Exponential response, expressed in nominal units from 0 (infinite hold) to full scale per sample.

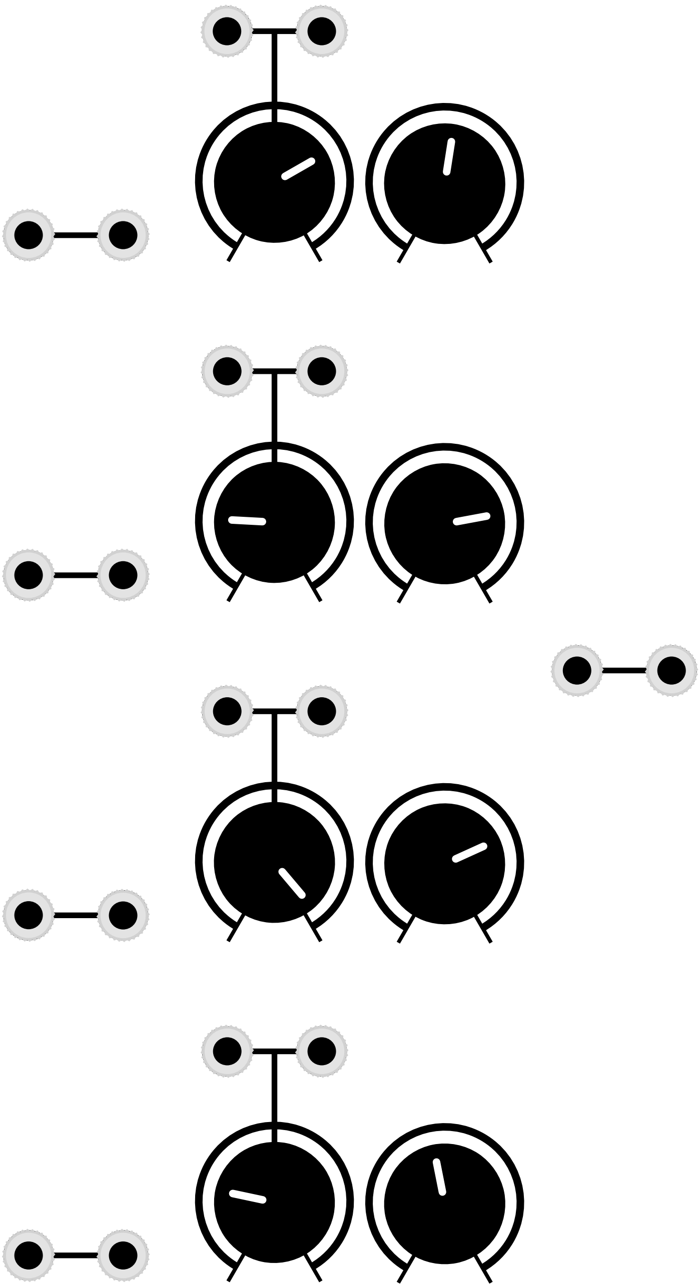
Curve and Quantize

Curve

Variable transfer function from soft clipping, through logarithmic, linear, and exponential, to crossover distortion. Expressed as a percentage of full range from linear.

Quantize

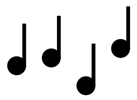
Step size of quantization from zero to full scale, expressed as a percentage.

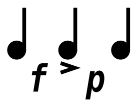


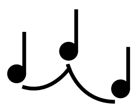
Mixer

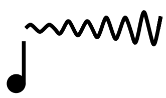
Level
Amplitude of input expressed as a percentage.

Note, the sum of the 4 channels is saturated to full scale.









Sources

Pitch
Master pitch source.

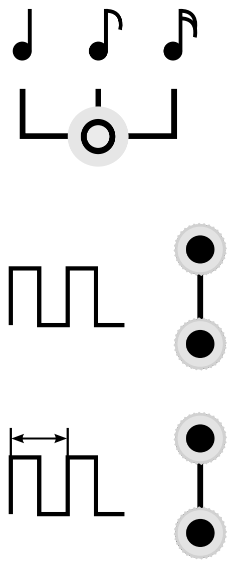
Gate
Square wave of key press.

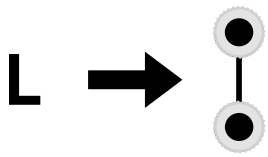
Accent
Velocity of key press.

Pitch bender
Value of pitch bend wheel.

Modulation
Value of modulation wheel.

Noise
Uniformly distributed white noise source.

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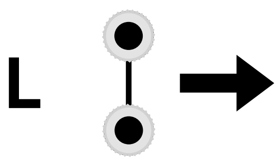
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Tempo

Note Length
Tempo clock period selection of quarter, eigth, or sixteenth note length.

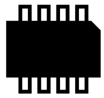
Clock
Square wave clock with 0.5 duty cycle at the selected rate.

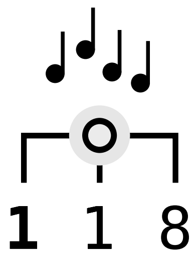
Rate
Tempo rate independent of clock period selection.

Audio

Audio Input
Source supplied from audio inputs.

Audio Output
Destination for audio outputs.

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Memory

Save

Holding down this button while sending a program change message will store the current voice to that location. Alternatively press and hold the save button then select the save location using the increment/decrement keys while they are in program mode, the voice will be stored on release of the save button.

Pressing the save button without changing the program number or sending a program change command will load the voice from the current program number. To load a specific voice select its program number with the increment/decrement keys in program mode then press and release the save button.

Voice

Polyphony

Selection of monophonic unison, monophonic, or polyphonic note assignment.

Synthesis Techniques

Normalling

On powering up the synthesiser the voice stored in memory location 0 is loaded, allowing a set of normalled connections to be automatically setup with common patches and parameter values. Equally, additional normalled schemes could be stored in other memory locations.

These settings will remain until modified by creating a patch to an input or adjusting a knob. Removing a patch will not return the normalled state but rather then connect that input to zero.

If you are using extensive normalling to save time it is highly recommended to mark these connections in coloured pen on the front panel. Historically modular synthesisers offered a ring around each knob so that settings could be recorded by marking the desired position of the pointer, and the choice of a white face plate for State Zero is no coincidence. STAEDTLER Lumocolor non-permanent pens in the medium or broad size are perfect for this application.

Portamento

No dedicated portamento function is provided, instead giving the user a choice of methods.

Portamento (exponential)

Passing the pitch signal through a low pass filter provides exponential portamento with a constant time for the pitch to arrive at the new note.

Glide (linear)

Passing the pitch signal through a slew rate limiter provides linear portamento with a constant rate at which the pitch changes to the new note.

Legato

Playing overlapping notes in one of the monophonic modes will provide a new trigger for each key press. To achieve legato voicing where the amplitude (if not other) envelopes do not retrigger for overlapped notes, pass the gate signal through a slew rate limiter.

With the limiters down rate set slow enough, notes played close together will not cause the gate signal to cross zero and the envelope will remain held from the previous note. The up rate should be as fast as possible to maintain accurate triggering of note on events.

Fingered Portamento

Similar to legato voicing, fingered portamento only slides between notes that are played overlapping. Again the simplest way to achieve the discrimination between notes that are overlapped or not is to pass the gate signal through a slew rate limiter or low pass filter.

Setting both the up and down rates to the correct speed will keep the output signal negative for notes played without overlap and positive for those played overlapped. This signal can then be used to control the portamento rate, such that as notes are played further apart the portamento time will reduce to zero.

This provides a smooth transition from portamento to none, unlike traditional synthesiser fingered portamento modes where it is either on or off. To achieve that effect, the output of the slew rate generator would be passed through a comparator before being used to control the portamento rate.

Filter Resonance

The filter modules of State Zero do not have a built in resonance control, instead resonance can be added by mixing the output of the filter back to its input. Positive feedback gains will be unstable and negative gains too large will self oscillate uncontrollably.

Self oscillation at the rolloff frequency can be stabilized in several ways, the simplest using the curve and quantize module in the feedback path to reduce gain for larger signals. This will greatly increase the gain for small signals and the feedback gain will need to be adjusted accordingly.

Due to the delay in the feedback loop high frequency oscillations can occur above the rolloff frequency of the filter when the feedback gain is set very high. As these oscillations only occur at higher frequencies it is possible to eliminate them by having the feedback gain negatively modulated by the filter modulation.

Envelope Curves

The envelopes are linear in operation, but feeding back the output to modulate the rate of a section allows it to take on a logarithmic or exponential shapes.

Envelope Variations

The 4 stage point-to-point envelopes provide numerous variations on the traditional ADSR envelope shapes. Some examples include:

ADSR

With the 3rd stage rate set to zero it is disabled, combined with the 1st and final stages levels set to 100% and 0% respectively a conventional ADSR envelope is formed.

AHDSR

Particularly effective for bass voices an almost flat top between the attack and decay sections can be achieved with a very low rate setting for the 2nd stage, and its level close in value to that of the 1st stage.

Staccato

Setting the 1st stage to a level of 0 and a very fast rate and using the 2nd stage as the attack component, the envelope will always rise from 0 for well defined staccato voicing.

Envelope Hold

Envelopes will always retrigger on a positive going edge through zero at their synchronization input. If a true hold time is required during which a retrigger cannot occur, the envelope will need to be triggered by an AD envelope, which is triggered by the gate pulse. The AD envelope should have an attack rate as fast as possible and the decay time adjusted to that it reaches 0 at the desired hold time.