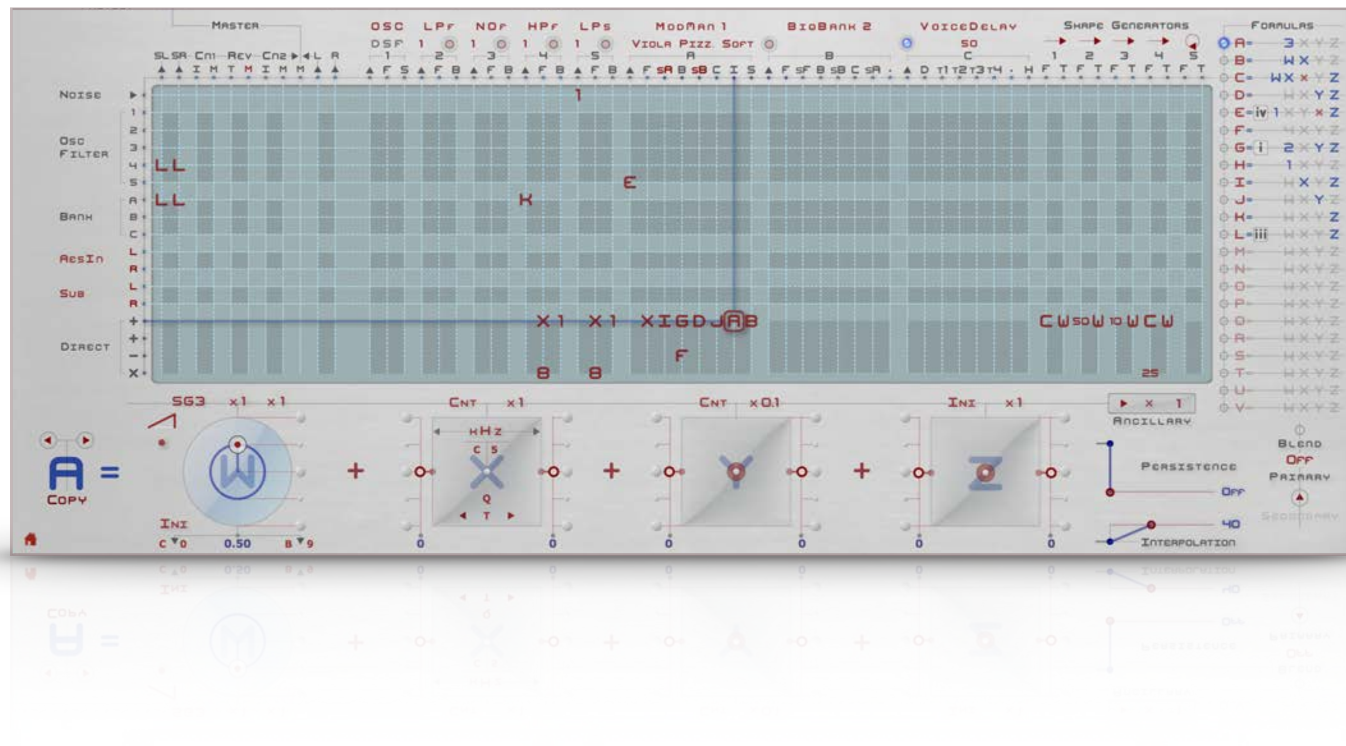
EaganMatrix User Guide

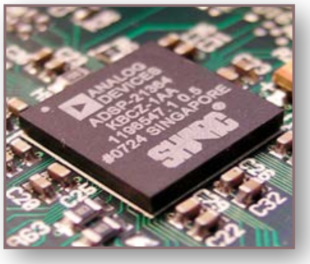
June 20, 2017

Firmware Version 8.50



Edmund Eagan and Lippold Haken

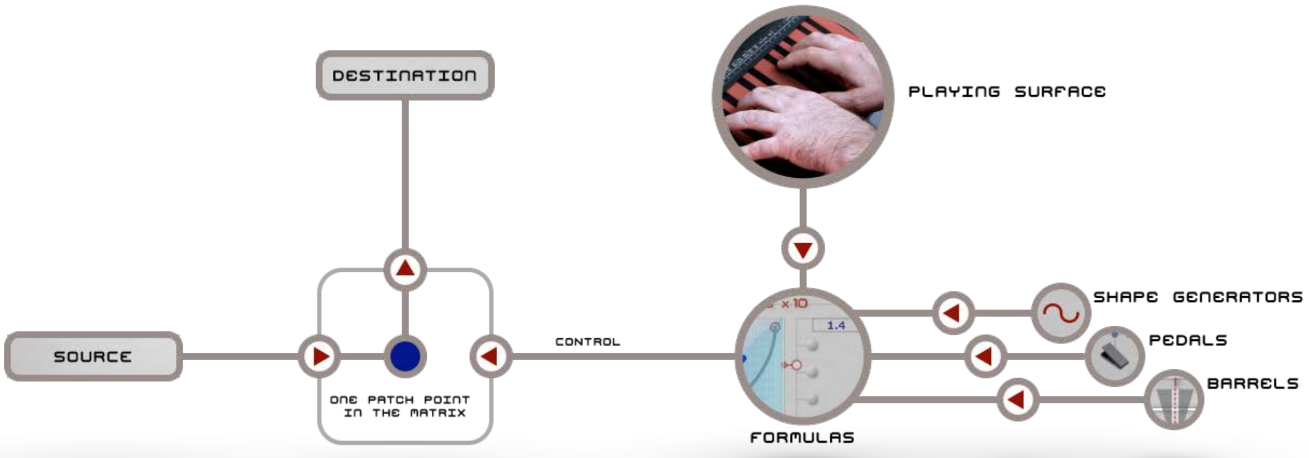
1. Introduction to the EaganMatrix



EaganMatrix is the modular digital synthesizer built into the Continuum Fingerboard. The Continuum Editor program (for Mac and PC) provides the user interface for the EaganMatrix.

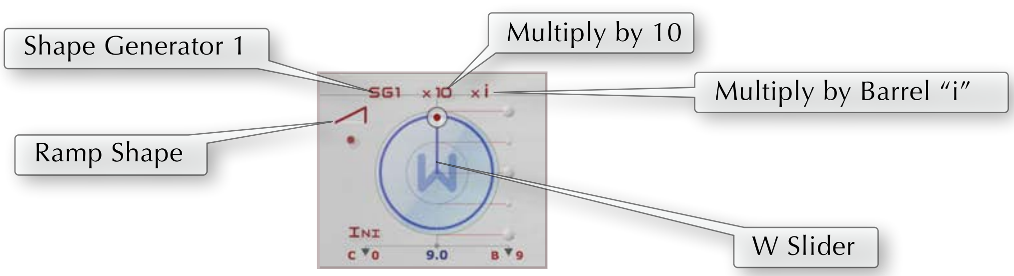
The EaganMatrix allows the user to finely craft musical sound algorithms by connecting audio and control modules via a two dimensional patching matrix. This matrix is inspired by classical modular matrix patching synthesizers such as the EMS VCS3. However, unlike the analog predecessors, the EaganMatrix doesn't use pins to make patch point connections. Instead, the user defines dynamic formulas and places them at matrix patch points. These formulas are a function of the position and pressure of each finger on the playing surface, Shape Generators, and also other formulas. When a formula is assigned to a patch point in the matrix, the formula controls the flow of sound from the source module on the matrix row to the destination module on the matrix column. Each three dimensional performance direction of the Continuum playing surface can influence the final result of every single patch point. And there are many patch points!

This relatively simple concept is powerful. It harnesses the amazing performance response of the Continuum Fingerboard so that the performer can create nuanced, expressive, dynamic, and totally custom sounds. Sounds tailored specifically to your personal musical tastes.



1.1. EaganMatrix Math

The EaganMatrix requires the sound designer to think in a particular mathematical way that is somewhat different from traditional hardware and software synthesizers. Below is an extracted part from an EaganMatrix formula:



An example of the math in the W component of an EaganMatrix formula: Ramp from 0 to 1 (SG 1) × 0.9 (W Slider) × 10 × value of Barrel "i"

This potentially complex mathematical language of the EaganMatrix has been simplified and distilled to best exploit the performance capabilities of the Continuum Fingerboard. The EaganMatrix already includes a comprehensive selection of predefined System Presets that can be further tweaked, aiding in the early exploration of the EaganMatrix design structure. As the sound designer masters this EaganMatrix math, its new capabilities will allow creation of musically satisfying relationships between fingers on the playing surface and the sounds produced, relationships that truly rival the warmth and complexity of acoustic instruments.

1.2. Modules

These modules are available in the EaganMatrix, acting as matrix sources (Src), matrix destinations (Dst), or both sources and destinations:

Src	Dst	Description
•		A wide-band (white) noise generator.
•	•	Five wide-band noise noise generators with seeds.
•	•	Five summation oscillators, each with controls for phase, frequency, and spectral balance. Each oscillator can be set to be a seeded (repeatable) noise source. The oscillators can also act as wave shapers, creating rich harmonic variations to the oscillator input.
•	•	Five Multimode filters, with control of frequency and bandwidth/resonance. Possible filters are Low Pass, High Pass, Band Pass, Low Pass Shelf, High Pass Shelf, Notch, and All Pass. Filters are either 2-pole (12 dB/octave cutoff slope) or 1-pole (6 dB/octave cutoff slope). Each filter can be cascaded 1, 2, 3, or 4 times.
•	•	Two biquad filter banks, which are banks of 48 related but independent biquad filters, with controls for frequency, frequency spread, bandwidth, bandwidth spread, spectral centre, and spectral weighting. This technology is used to create the physical models of winds, strings, and percussion. Four varieties of these filter banks are available: Scaled Resonances, Modal Amplitude Graphs, Vocal Formants, and Modal Manipulators.
•	•	Two sine oscillator banks, each producing 16 sine oscillators under matrix control.
•	•	Two Sine Sprays, which produce sine grains under matrix control.
•	•	Two WaveBanks, each producing five sawtooth, square, or triangle waves under matrix control.
•	•	Two Harmonic Manipulators, which can manipulate the spectra in predefined Spectral Sets, or Live data from the Delay module. Harmonic Manipulators are in addition to Modal Manipulators (mentioned above), which also manipulate Spectral Sets and Live data.
•	•	Four delays, with control of delay times of the main delayed output and additional output taps. The VoiceDelay is suitable for creating chorus and flanging effects, can be used for fast slap delays. The SummedDelay processes an aggregate of all voices in one shared delay. The MicroDelay provides four short single-tap delay buffers for each voice. The FormulaDelay provides long delay times for control data used with EaganMatrix formulas.
	•	8 CVC (Continuum Voltage Convertor) outputs, allowing you to create up to 8 preprocessed control voltages for each voice for analog synthesizers. This requires that a CVC be connected.
•	•	Five independent sub-audio Shape Generators, each with controls for cycle mode, frequency, and trigger. Any formula may use any SG, and each formula independently selects a shape for its use of the SG. Shapes available are Ramp Up, Ramp Down, Pulse, Pulse at End, Triangle, Hann, 4 variations of S Curves, Square, Sine, Sample-and-Hold, and many other shapes attained by formula-specific SG phase modulation. Note: SGs are available as a sources within each formula, and not in the matrix directly.
	•	Soft saturation in the Master Section, for distortion and overloading effects.
	•	A Recirculator which can do digital reverb, modulated delay, or echo processing. The recirculator gives the sound a sense of space.
	•	Two short convolutions, one pre-recirculator and one post-recirculator.
•		AES stereo digital audio input, from the Continuum Fingerboard's AES3 In connection.
•		The submix from the Master Section, available to be reprocessed through the matrix structure.
•	•	Stereo output from other EaganMatrices, to be processed through this matrix (for Combination Presets).

1.3. Source and Destination Locations in the Editor

Master Section

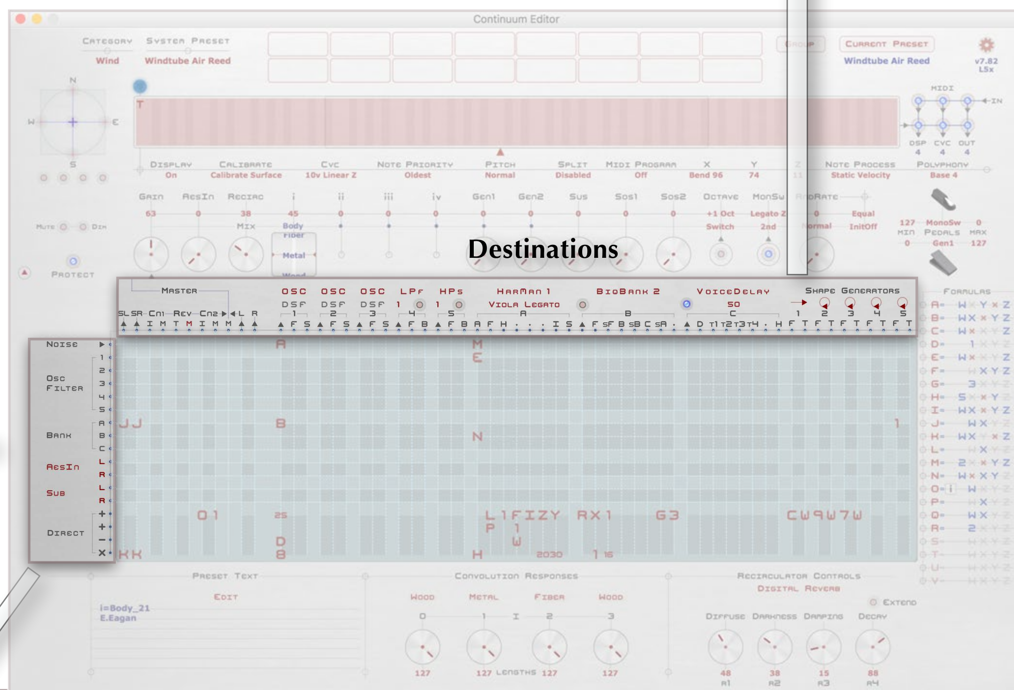
Soft Saturation, Pre-Convolution, Recirculator, Post-convolution, Wet/dry Stereo Output Mix

Summation Oscillators and Filters

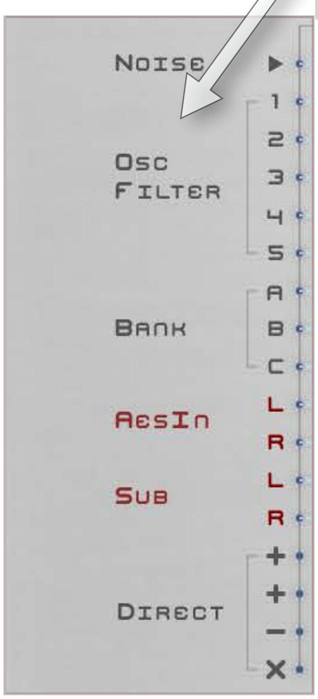
Three Multipurpose Banks

Biquad Filter Banks, Sine Oscillator Banks, Sine Sprays, Manipulators, Delays, CVC Control Voltages

Shape Generators



Sources



Noise

Summation Oscillators, Seeded Noise, and Filters

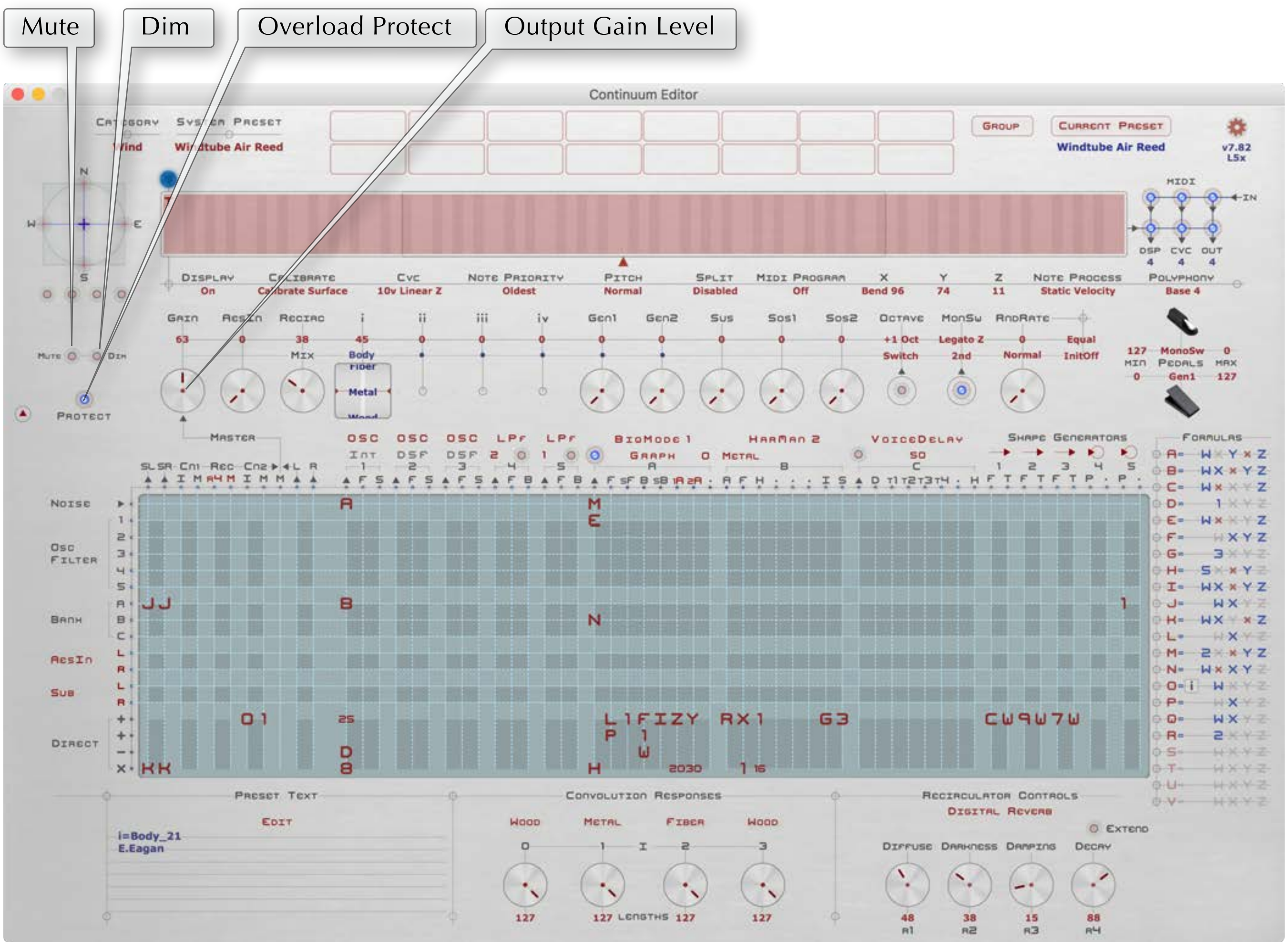
Multipurpose Banks Biquad Filter Banks, Sine Oscillator Banks, Sine Sprays, Manipulators, Delays, CVC Control Voltages

Selectable Sources AES Audio In, Delay Taps, Master Submix

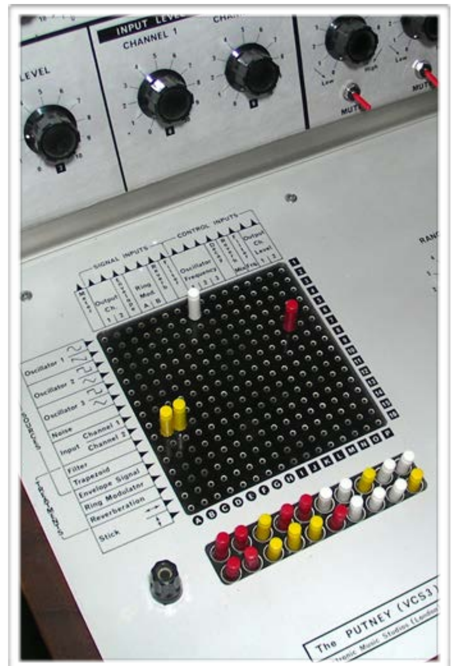
Direct Positive, Negative, Multiply (Formula directly controls destination)

Please Be Careful…

Please be careful with your speakers (and ears) when programming the EaganMatrix! Due to its highly flexible nature, it is possible to create presets that spiral out of control, potentially causing loud output. The EaganMatrix has an Overload Protect mechanism to watch and mute the output when these overloads happen during an editing session, but it can still be heart stopping! Get in the habit of lowering the Output Gain before critical patching explorations. Mute and Dim controls are also available to temporarily lower the output level of the EaganMatrix. Definitely do not get in the habit of "random" patching; successful sound design with the EaganMatrix requires understanding and planning, combined with careful experimentation.



2. Basic Design Concepts



In analog synthesizers matrices are a compact way to maximize the number of input-to-output connection points in a small and concise area, rather than the more customary use of patch cables. The matrix idea has existed in analog modular synthesizers for some time, for instance in the 1970s EMS VCS3 analog synthesizer, its matrix pictured to the right.

For these traditional matrices, shorting pins are used to connect sources (labelled on the left) to the destinations (labelled on the top). A pin is inserted at the desired connection point in the matrix, creating a signal flow from source to destination.

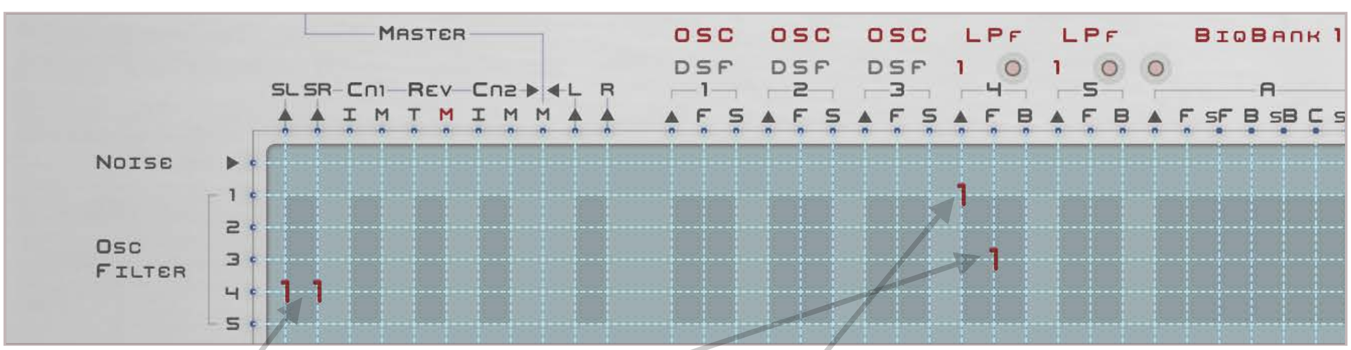
In the VCS3 matrix to the right, a white shorting pin is connecting the output of Oscillator 1's sawtooth wave to the input of the Filter. A red shorting pin connects the Oscillator 3 triangle output to the frequency cutoff control of the Filter. Finally, two yellow shorting pins connect the Filter output to Output Channels 1 and 2.

These shorting pins pass the outputs unaltered to the connected input. Finer signal control is achieved by supplying various control knobs on the synthesizer (i.e. a knob to control the shape of the oscillator waveform or another to control the oscillator's fundamental frequency).

Like the VCS3, the heart of the EaganMatrix is its matrix, but unlike the VCS3, the execution of its matrix has been greatly expanded by adding the ability to place a powerful mathematical formula directly inside a connection point.

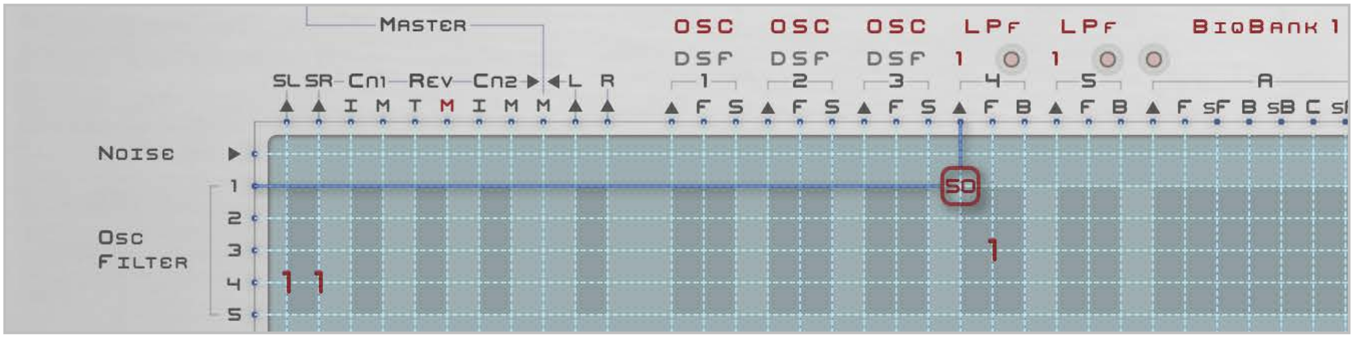
2.1. The Matrix in the EaganMatrix

Pictured here is a similar patch to the previous one shown in the EMS VCS3, except this time executed in the EaganMatrix:

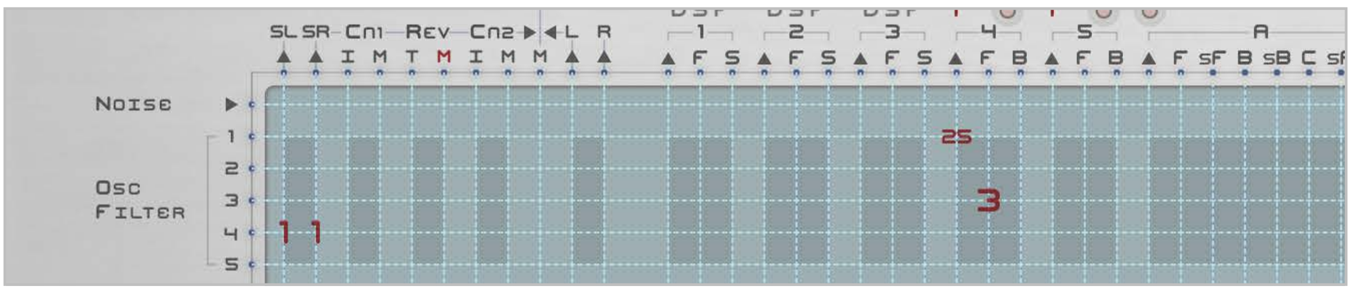


Again, sources are labelled on the left and destinations are labelled on the top. Instead of pins as used in the VCS3, red numbers are at the location where a VCS3 shorting pin would be, this red number representing the connection from source to destination. Notice that a number 1 is connecting the output of Oscillator 1 to the input of one of the filters in the EaganMatrix. This number **1** represents a unity gain from source to destination. Another number **1** is connecting the Oscillator 3 output to the frequency cutoff control of the Filter. Finally, two number **1**s are connecting the Filter output to Output Channels Left and Right.

In the EaganMatrix, the simplest virtual pin connection is represented by a numerical constant, like these number 1s above. However, patch points can be modified in more subtle or complex ways. For instance, to have half the amplitude (volume) of Oscillator 1 go into the input of the Filter, the number 1 at that connection point can be changed to the constant .50 (half of 1), as pictured below:

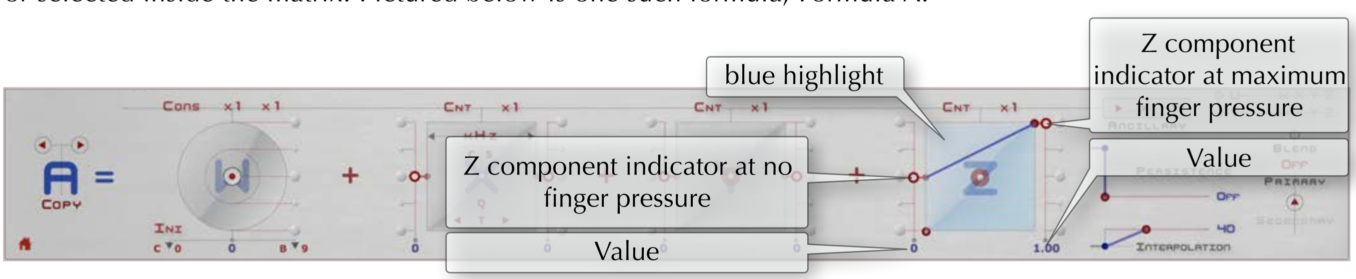


Of course, other numbers can be added. For instance, to have 1/4 of the amplitude of Oscillator 1 go into the Filter, and to triple the influence of Oscillator 3 on the cutoff of the Filter, use numbers .25 and 3, respectively:

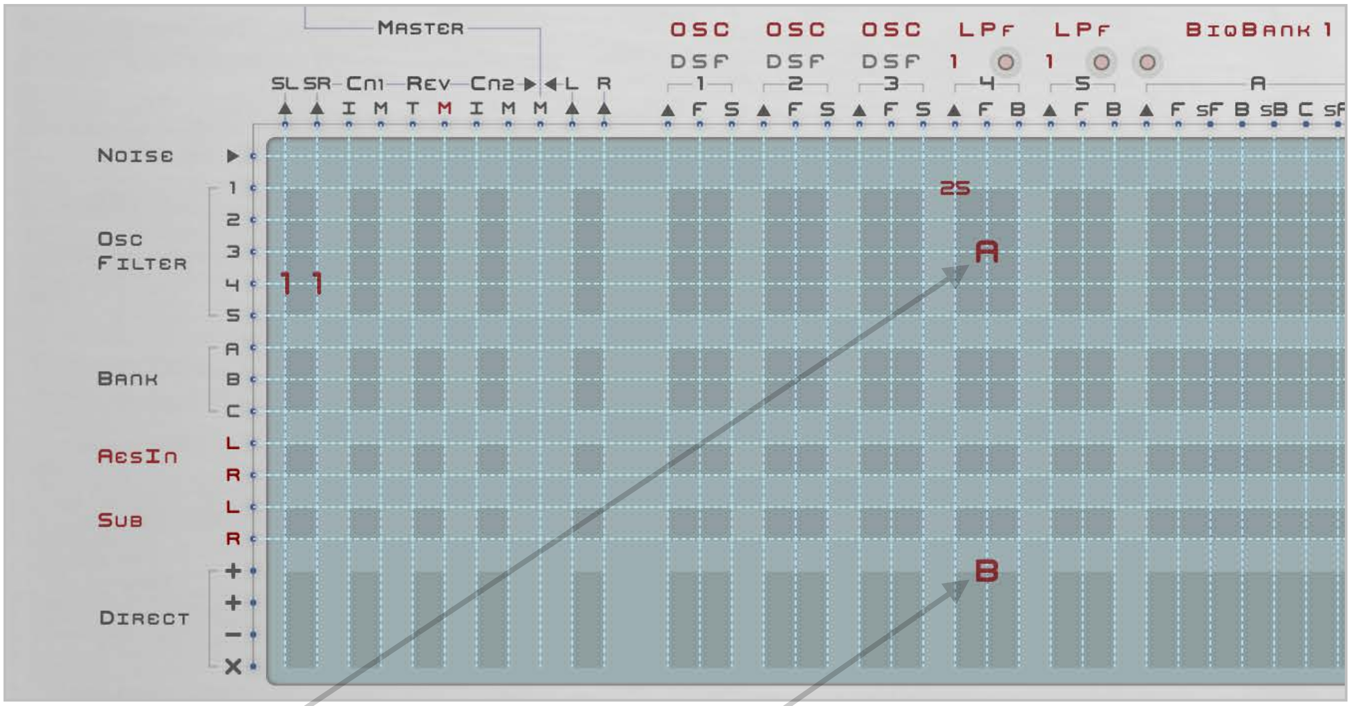


2.2. Playing Surface to Formula Connection

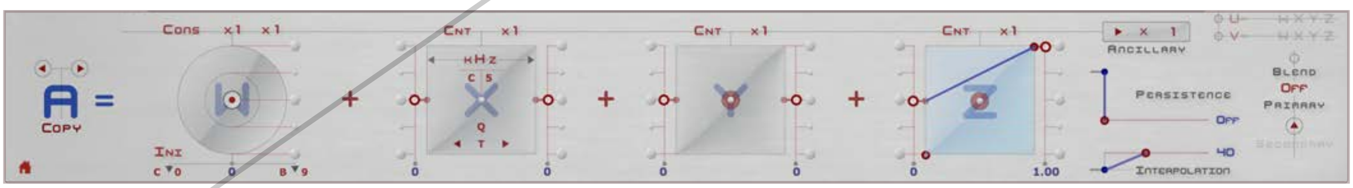
This is just the beginning of the patching possibilities in the EaganMatrix. Connection points can be dynamic rather than static, in the sense that they can change over time and through external influence. For the EaganMatrix, this external influence is primarily the Continuum playing surface. These dynamic matrix patching pins are formulas, graphically represented by the letters A through V and the letters W, X, Y, and Z. The user-programmable structure of the formulas A through V automatically becomes visible when that formula is placed or selected inside the matrix. Pictured below is one such formula, Formula A:



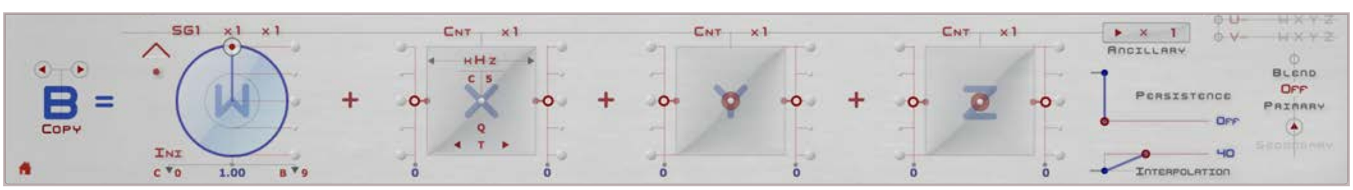
Formulas can mathematically represent input in three dimensions from the playing surface, as well as interactions with Shape Generators, pedals, barrels, and other formulas within an EaganMatrix preset. This formula structure is truly what makes the EaganMatrix powerful. In the formula pictured above, only the Z component (pressure input from the Continuum surface) is active. The active components of a formula are represented by a **blue highlight**.



With this **Formula A** placed into the Matrix, the frequency of a filter (osc/filter 4) is now controlled by an oscillator (osc/filter 3 via finger pressure, and only finger pressure, as the Z component of Formula A is the only part of the formula that is active. This finger pressure controls the amount that osc/filter 3's output is influencing the frequency of osc/filter 4. Formula A's Z component is at 0 with no finger pressure, and at 1 when there is maximum finger pressure.

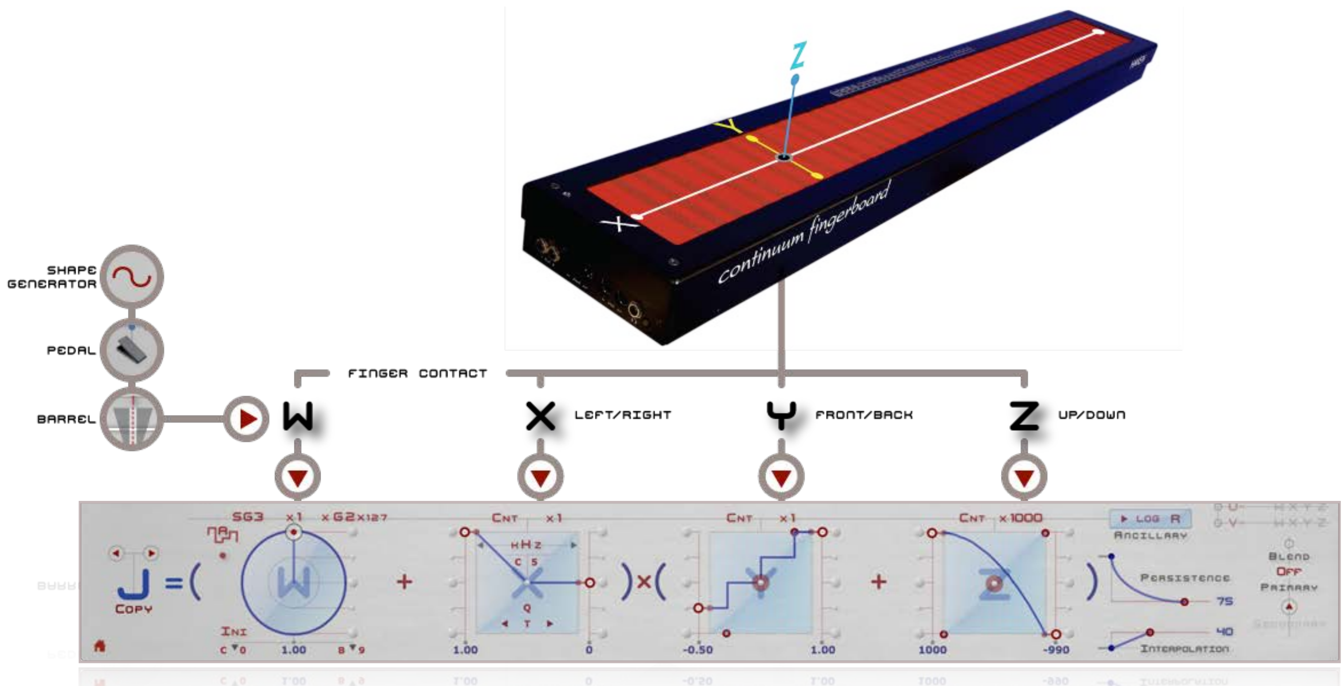


Formula B has also been placed at the intersection of filter 1 frequency column and a Direct+ row.



A Direct+ row has a constant value of 1, instead of a variable value like Osc3's output. Formula B has only the W component active, set to a value of 0 to 1 based on the current value of Shaper Generator 1. In a filter frequency column, this means it has a varying value from 0 to 1 kHz. This is added to the value of Formula A to determine the actual setting for the frequency of filter 1. (Osc 3 * Formula A) + (1 * Formula B) = Filter 1 Frequency.

2.3. WXYZ - The Four Components of an EaganMatrix Formula



A formula is divided into four components, labelled in the EaganMatrix formula graphic as W, X, Y, and Z.

The W component is active when a finger is in contact with the playing surface, or optionally active all the time. The W component can include influence from Shape Generators, sample-and-holds, barrels, foot pedals, and other formulas.

The X, Y, and Z components of the formula correspond to a particular playing direction in the three dimensional space on the Continuum surface. The X, Y, and Z components each have a programmable transfer function (a.k.a. "input-output mapping"). In formula J above, the Y component has a three-step transfer function, and the Z component has a squared transfer function.

The four components, W, X, Y, and Z, can be combined by various methods involving a selection of additions and multiplications. This streamlined yet flexible design architecture means that formulas can be tailored to respond to countless blended combinations of performance data from the Continuum playing surface.

3. Exploring the EaganMatrix

Due to the radically new design of the EaganMatrix, we highly recommend you go through the tutorials in this User Guide. The tutorials will familiarize you with the fundamental programming aspects of the EaganMatrix. If you have not done so already, please read the Continuum User Guide prior to starting the tutorials.

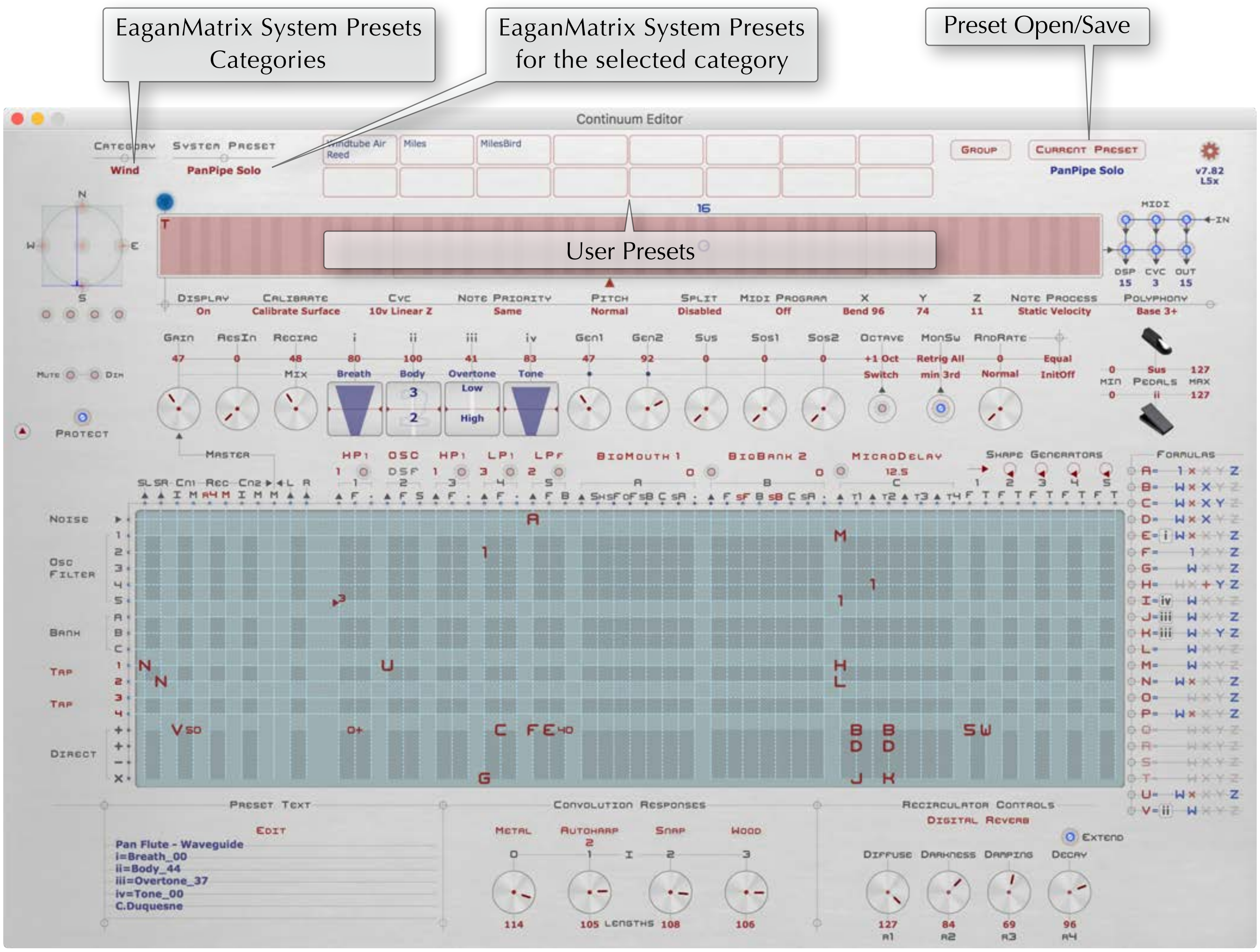
After finishing the tutorials, you may learn advanced techniques by studying the designs of the EaganMatrix system presets. You can create user-defined presets using system presets as starting points for further sonic explorations.

The picture below shows the location for accessing System Presets and where to work with custom Presets. The EaganMatrix System Presets are listed via categories, then alphabetically. Selecting a system preset will load it into the Continuum as well as the Editor.

In addition to the System Presets, 16 User Presets are stored inside the Continuum Fingerboard. These User Presets can simply be copies of System Presets, or customized versions of System Presets, or Presets created from scratch.

Additional Presets can be opened from your computer's file system by clicking on the "Current Preset" button and choosing "Open". A standard dialog box will appear prompting for a file location.

When you work on a User Preset of your own, you should not only store it as a User Preset inside the Continuum, but also use "Save" to store it to your computer's file system. When you use "Save", you will have the opportunity to choose a name for your preset.



4. Tutorials

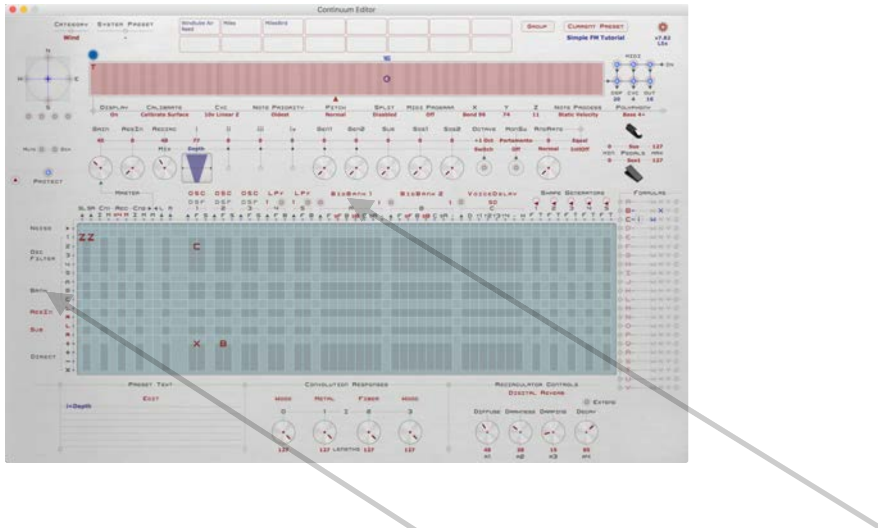
The tutorials require presets that are not stored inside the Continuum. In order to perform the tutorials, locate and open the Preset files "Simple FM Tutorial.mid" (Tutorial 1) or "Resonant Drum Tutorial.mid" (Tutorial 2) which reside in the Tutorial folder within the Archives folder.

4.1. Tutorial 1: Simple FM

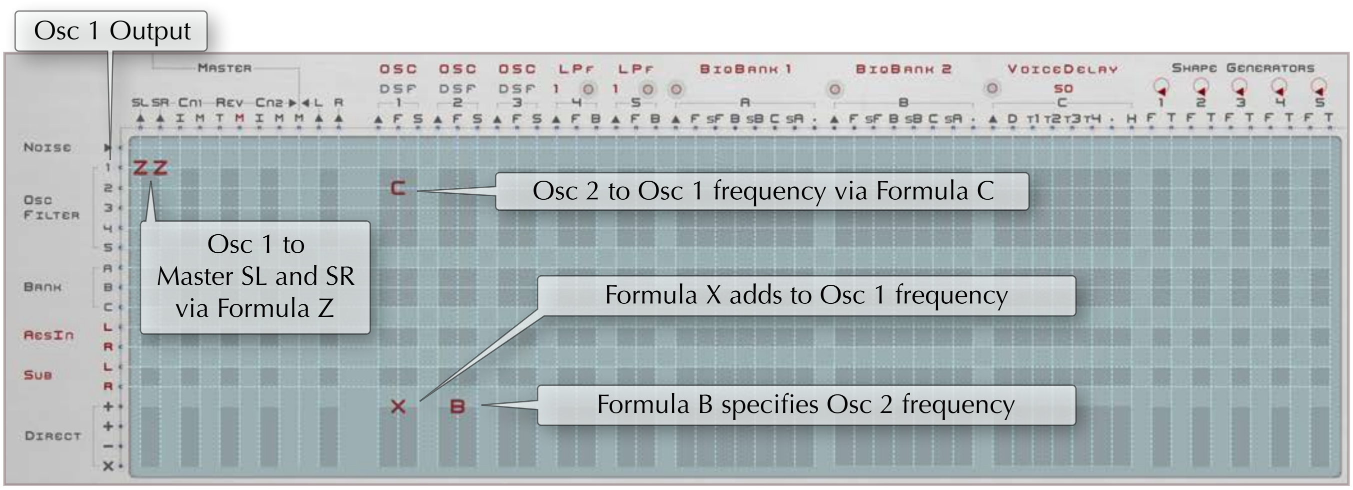
Run the Continuum Editor and open the Preset file "Simple FM Tutorial.mid". We'll use this preset to look at the data flow and formulas used inside the EaganMatrix. When the preset is selected the Continuum page of the Editor should look like this:

This preset uses only one of the four possible barrels. The designer of this preset has used barrel "i" to allow the performer to make sonic variations of the sound. Play this preset, and listen carefully as you try it out. This preset is a basic two operator FM sound, using oscillator sine waves. When a finger presses on the Continuum's playing surface while this preset is loaded, the finger's X (left to right) position controls pitch, the finger's Z (pressure) controls amplitude, and the finger's Y (front to back) position does nothing. Play a note and move the barrel "i", also labelled "Depth", and notice how the timbre changes. This barrel has an effect on the depth of the FM modulation, in this case Oscillator 2's modulation depth of Oscillator 1.

Click on the Large Edit Window button (red triangle in a circle) and select Matrix View to see the EaganMatrix. Or use the keyboard shortcut, Command+L (Mac) or Control+L (Windows). After the EaganMatrix is displayed, Full Screen can be toggled using the Large Edit Window button or by Command+F (Mac) or Control+F (Windows).



The large edit window shows the Matrix Area of the Editor. **Sources** are labeled on the left, and **destinations** are labeled on the top. Signals flow from the sources on the left through an active matrix point, then to the destinations on the top. These destinations will in turn compute new source values for the next sample interval. The red alphanumerics in the matrix are active matrix patch points.



In this particular preset, two oscillators are used, Oscillator 1 and Oscillator 2.

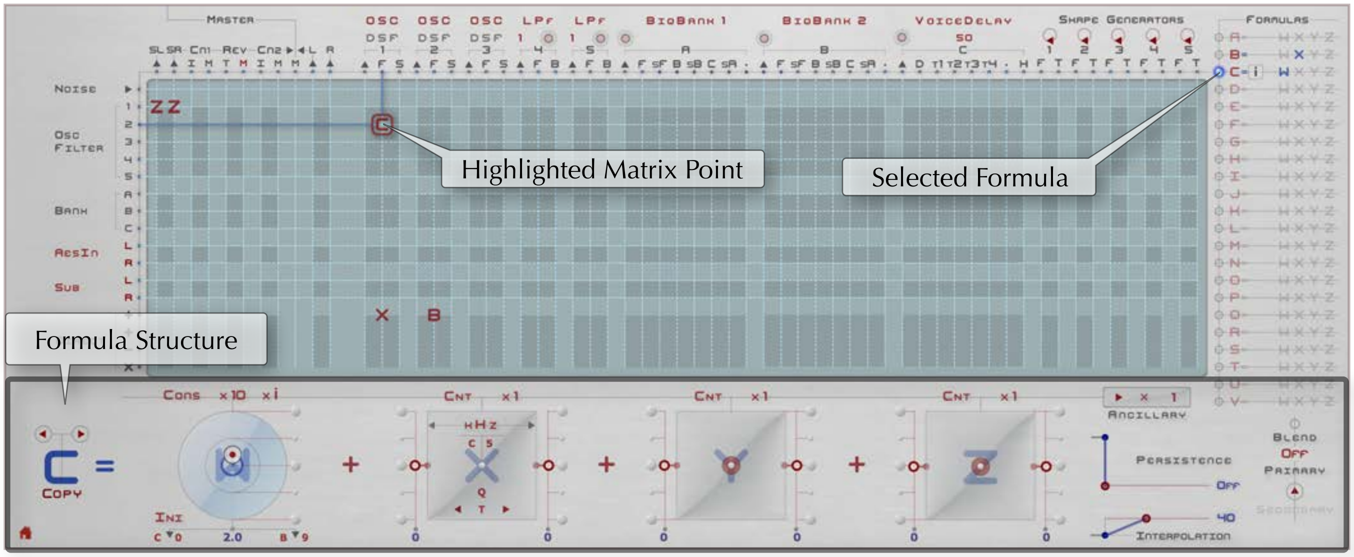
The EaganMatrix has four pre-defined Formulas, W, X, Y, and Z; and twenty-two user-defined formulas, A through V. This preset uses pre-defined Formulas X and Z, and user-defined Formulas B and C.

Oscillator 1 (the FM carrier) is routed to the Master Section (the EaganMatrix output section) using the pre-defined Formula Z. Formula Z implements a squared function of finger pressure. Typically a squared-pressure function like this is ideal for finger control of amplitude. This is why it is being used here.

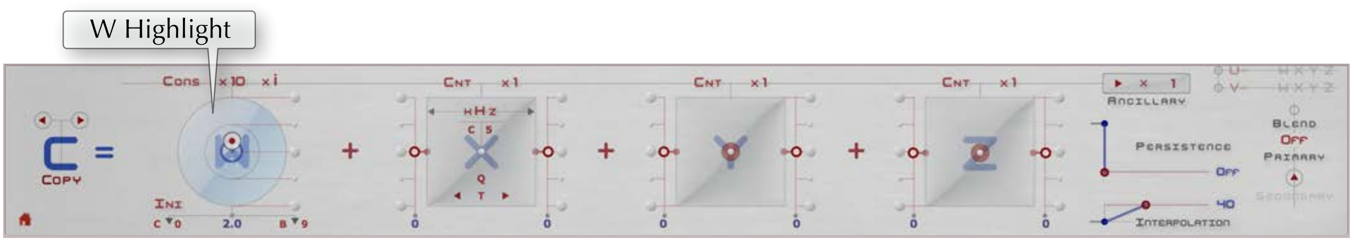
Oscillator 1's frequency is being controlled by the influence of two formulas, X and C. Pre-defined Formula X is in a Direct+ row, so it will directly affect the frequency of Oscillator 1; the Direct+ rows do not route audio signals, instead formulas in the Direct+ rows directly add into the matrix column. Formula X will cause the frequency of Oscillator 1 to track the X direction (left to right finger position) of the Continuum surface, so that the pitch goes higher to the right and lower to the left. Formula C controls the amount of modulation added to this frequency by Oscillator 2.

Formula B controls the frequency of Oscillator 2 (the FM modulator).

This creates a classic FM style of synthesis. Click on the C in the matrix. The screen will update, highlighting the selected matrix point and showing the selected formula below the matrix.

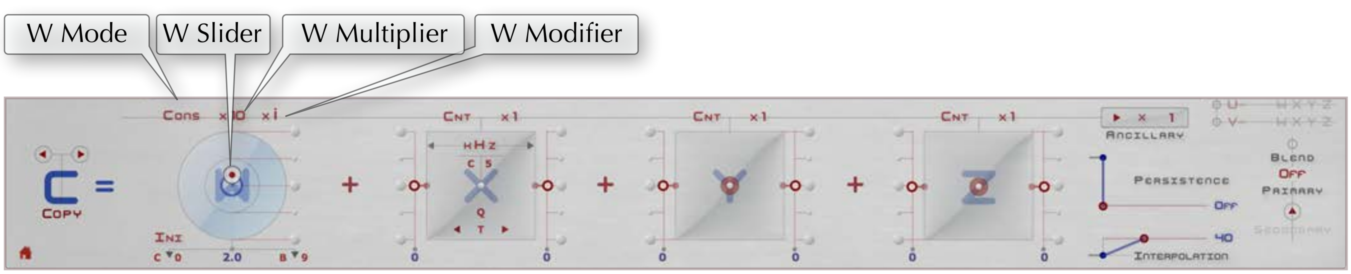


Look at the formula structure. This formula is being used to control the output of Oscillator 2 into the frequency of Oscillator 1. The W circle is highlighted, meaning that the W component contributes in the formula. The X, Y, and Z components are not highlighted, indicating they do not contribute to the formula. The Continuum surface X (left to right), Y (front to back), and Z (pressure) will have no effect on changing this formula's output, since those components each have value zero in this formula.



The W stands for Window. This W component of the formula can be used to generate a non-zero value while the finger is in contact with the surface, or optionally it can generate a value independent of finger contact. The math for the W component of this formula is above the round W icon.

The math for the W component of EaganMatrix formulas is: **W Mode** × **W Slider** × **W Multiplier** × **W Modifier**.



The **W Mode** in the example above is "Constant" (abbreviated Cons), always having value 1. This means that the W component's value will be persistent, keeping its value even after the finger is lifted from the playing surface. Other choices are:

"Gated" with a finger in contact, W Mode has value 1; when the finger is lifted, W Mode has value 0,

"SG1" W Mode value depends on the state of Shape Generator 1, "SG2" W Mode value depends on the state of SG 2,

"SG3" W Mode value depends on the state of SG 3,

"SG4" W Mode value depends on the state of SG 4,

"SG5" W Mode value depends on the state of SG 5,

"SG1B" W Mode value depends on the state of SG 1B (if SG1 has "Dual" selected),

"SG2B" W Mode value depends on the state of SG 2B (if SG2 has "Dual" selected),

"SG3B" W Mode value depends on the state of SG 3B (if SG3 has "Dual" selected),

"SG4B" W Mode value depends on the state of SG 4B (if SG4 has "Dual" selected,

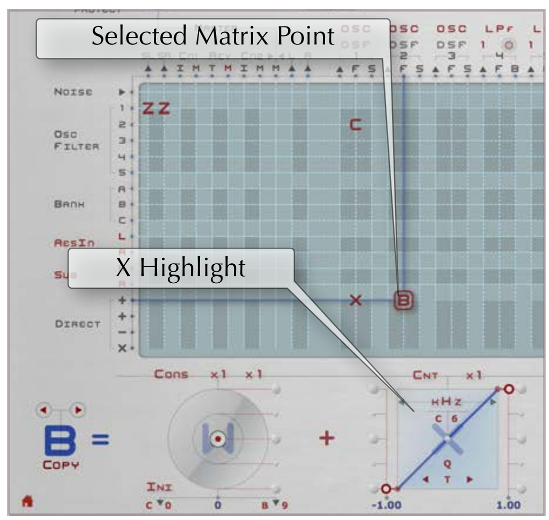
"SG5B" W Mode value depends on the state of SG 5B (if SG5 has "Dual" selected),

"FormulaDelay" W Mode value depends on an output of the FormulaDelay.

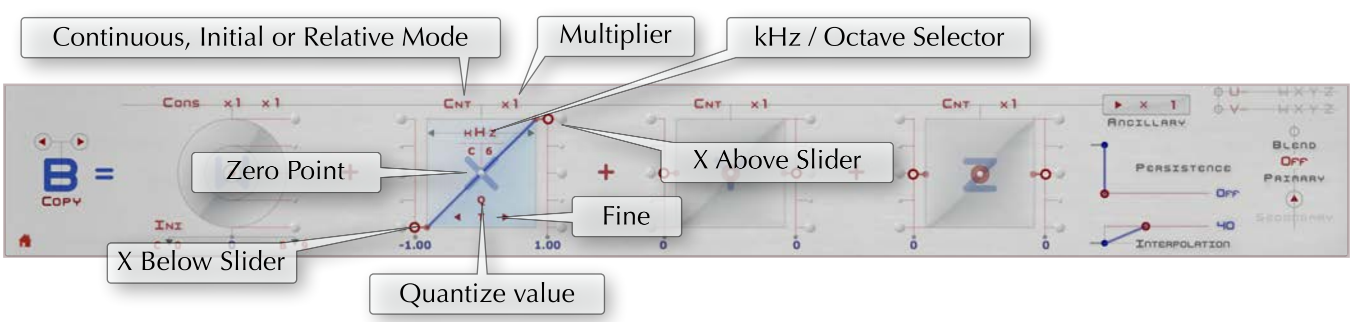
The **W Slider** has a range from -1 to 1; in the example above it is 0.2. The **W Multiplier** has a range from 0.001 to 1000; in the example above, it is 10. Finally the **W Modifier** allows the W component to be modified from an external controller, such as the i, ii, iii, iv barrels or the Gen1, Gen2 dials.

In general, barrel values are scaled 0 to 1 when used in EaganMatrix formulas. In this W component, barrel "i" is multiplied by 0.2 (W Slider) and by 10 (W Multiplier). So the modulation of Oscillator 2 into Oscillator 1's frequency will be scaled by a value from 0 to 2, depending on barrel "i". Incidentally, if a foot pedal connected to the Continuum's pedal jack controls barrel "i", it will be at much higher resolution than the normal 7-bit Midi resolution.

Now click on the B in the matrix to have a look at that formula.



Formula B is being used to control the frequency of Oscillator 2. B is in a Direct+ row, which means it's not passing information from Source to Destination, but instead is directly controlling the Destination. In Formula B, only the X component of the formula is highlighted, indicating that the W, Y, and Z components do not contribute to the formula value.



The **X Mode** is "Continuous" (abbreviated to Cnt), which means that the fingerboard's X position is being tracked and updated continuously as long as the finger is in contact with the surface. Other less common X

Modes are "Initial" (which means that the X value is sampled and held when the finger first comes in contact with the surface) or "Relative" (which means that the X value is relative to the position at which finger first comes in contact with the surface).

The X component has a **Zero Point**, which is the position on the Continuum surface where the X component of the formula has an output of zero for "Continuous" and "Initial" mode. In a pitch sense, this is where Middle C will be.

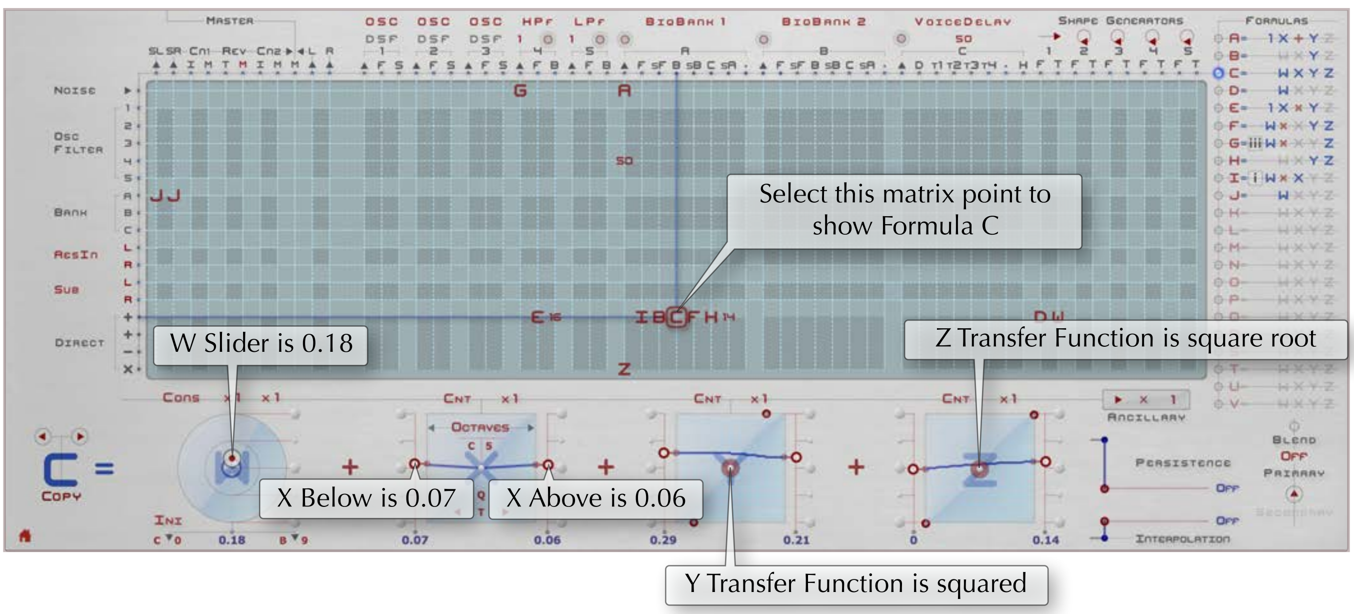
To the left of the Zero Point, X is scaled by the **X Below Slider**. To the right of the Zero Point, X is scaled by the **X Above Slider**.

Since the oscillators and filters in the EaganMatrix are controlled in kHz units, **kHz** is normally selected. Alternatively, **Octaves** can be selected for octave units.

In this preset, Formula B causes the frequency of Oscillator 2 to track the Continuum surface one octave above concert pitch, due to the setting of the Zero Point at C6 instead of the default concert pitch setting of C5.

4.2. Tutorial 2: Resonant Drum

Open the Preset file "Resonant Drum Tutorial.mid". This is a copy of the EaganMatrix preset sound "Resonant Drum", one of the more complex EaganMatrix sounds, in particular with some of the formulas used for the Biquad Bank:



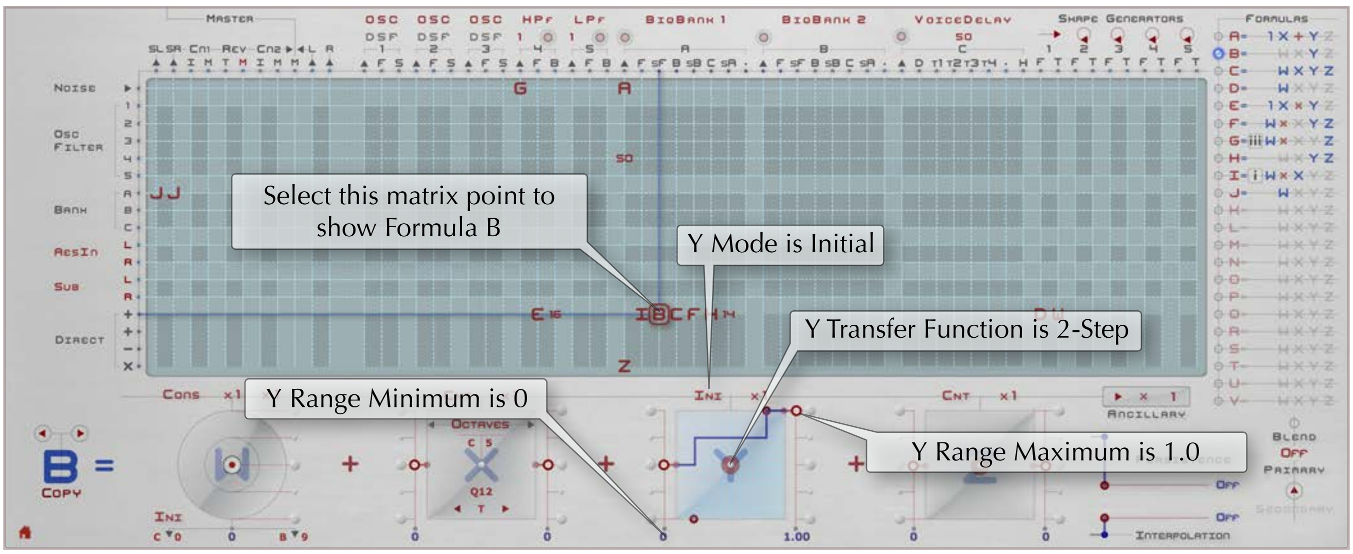
Select Formula C in the matrix, as shown above. In this Resonant Drum sound, Formula C is used for subtle control of the bandwidth parameter of the Biquad Bank. All four components (W, X, Y and Z) are used in Formula C to control the amount of ringing of the resonant drum. It uses a careful blend of X (left/right), Y (front/back), and Z (pressure) finger movements to influence the width of the filters in Biquad Bank 1. Lower value output from this formula will cause narrower bandwidths in the biquad bandpass filters, resulting in more sustained resonance in the sound.

The W component is a constant, adding 0.18 to the overall formula. Try modifying the formula by increasing the value of **W Slider** from 0.18 to 0.28. The overall effect will be a more muted drum because of the overall increase in the formula output value, resulting in wider bandwidths. Return the W Slider to 0.18.

The X component has **Octave** units. The X component smoothly increases by 0.07 each octave below middle C, and by 0.06 each octave above middle C. This X component ensures that the farther the finger's pitch is from middle C, the more the drum resonance is muted.

The Y component uses a squared transfer function, with *lower* values as Y *increases*. This reverse-Y effect is achieved by making the **Y Range Minimum** *larger* than the **Y Range Maximum**.

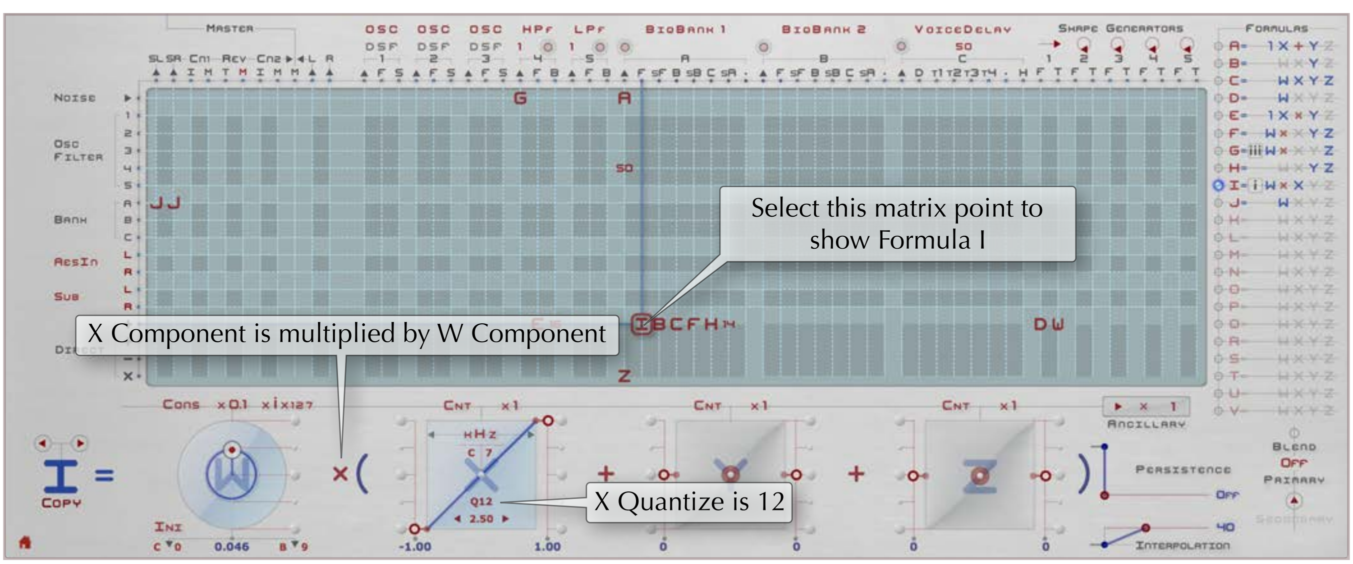
The Z component is of particular interest. Having the **Z Range Maximum** set at 0.14 is what makes the drum sound ring when played with a sharp attack and release, and muted when played held down. This mimics the behaviour of an acoustic drum head, which mutes with sustained harder contact.



Select the formula used to control the frequency of the Biquad Bank, Formula I, as shown below.

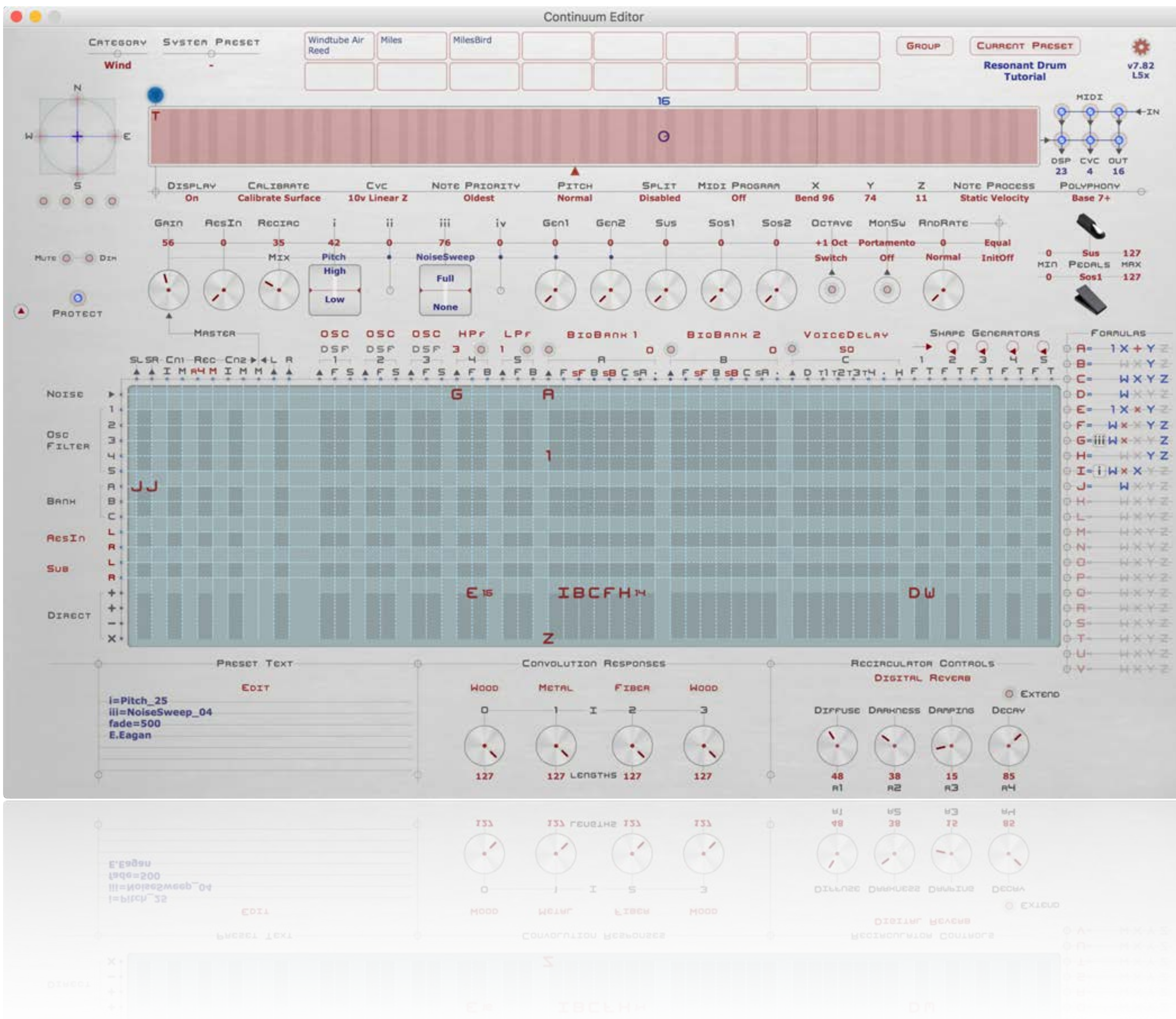
Formula I is used for control the centre frequencies of the biquad bandpass filters. The **X Quantize** is 12 to make the pitch is the same within each octave from C to C. It then jumps an octave in the next octave range of the playing surface. This simulates having a separate virtual tuned drum within each octave span of the Continuum. The X component is multiplied by the W component. The **W Modifier** is barrel "i×127". This allows for global pitch of the sound to be set using the barrel. The version of barrel assignment is the "times 127" variety, which allows for equal divisions using 7-bit values (e.g., doubling the barrel setting will cause an exact doubling in pitch).

Select the formula used to control the frequency spread of the Biquad Bank, Formula B.



Only the Y component is used in this formula. The **Y Mode** is set for "Initial" (abbreviated Ini), so it reads only the initial Y value (the value of Y when the finger first touches the playing surface). The 2-step transfer function is selected; this means the Y component has only three values: front, middle, and back, set at 0, 0.5, and 1 respectively. The middle value is exactly halfway between the front and back values. This Y component creates three apparent shifts to the pitch of the drum sound due to the varying spreads of the frequency centres of the filters in the biquad bank. As the finger continues to move on the surface, this value will not change due to the "Initial" mode setting.

Explore the other formulas to deduce their influences on the sonic output of the Resonant Drum.



5. EaganMatrix Sources Reference

Any EaganMatrix Source (matrix row) may be multiplied by a formula and added to an EaganMatrix Destination (matrix column) by placing the formula at the intersection of the row and column.

5.1. Sources

Source		Description of Matrix Row for each Source
Noise		Noise audio out.
Osc/ Filter	1	Audio from oscillator/filter 1. Possible oscillators are DSF, Integrated DSF, and Expose Phase. Possible filters are Low Pass, High Pass, Band Pass, Low Pass Shelf, High Pass Shelf, Notch, and All Pass. There is also a Noise from Seed available. These are described in Section 7.2.
	2	Audio out from oscillator/filter 2.
	3	Audio out from oscillator/filter 3.
	4	Audio out from oscillator/filter 4.
	5	Audio out from oscillator/filter 5.
Bank	A	Audio from multipurpose Bank A. One of the following may be selected for Bank A: a biquad bandpass filter bank (BiqBank Scaled Resonances, BiqMode Mode Amplitude Graphs, or BiqMouth Vocal Formants), a generator bank (SineSpray Sine Grains, SineBank Sine Oscillators, or WaveBank Wave Generators), or a Manipulator (HarMan Harmonic Manipulator, or ModMan Modal Manipulator). See Section 7.
	B	A second multipurpose bank like Bank A, giving access to a second set of modules identical to those in Bank A. In addition, the CVC Control Voltage module is available in Bank B.
	C	A third multipurpose bank. One of the following may be selected for Bank C: a delay (VoiceDelay, SummedDelay, MicroDelay, or FormulaDelay), or modules from Bank B. Bank C can use modules from Bank B when Bank B has different (and compatible) modules selected.
AesIn	L	The audio from the left channel of the AES3 input into the Continuum. Alternatively, this row can be Tap 1 from VoiceDelay or SummedDelay (Bank C), or a mono sum of the stereo AES3 input. In a CEE Combination Preset, this row can supply the left channel output of DSP 1, or the mono sum of DSP 1's output, to DSP 2 and/or DSP 3 — allowing for post-processing of DSP 1's output by another DSP.
	R	The audio from the right channel of the AES3 input into the Continuum. Alternatively, this row can be Tap 2 from VoiceDelay or SummedDelay, or a mono sum of Tap 1 and Tap 2 from the Delay. In a CEE Combination Preset, this row can supply the right channel output of DSP 1 to DSP 2 and/or DSP 3 — allowing for post-processing of DSP 1's output by another DSP.
Sub	L	The audio submix from the left channel of the Master (SL column after saturation, convolutions and recirculator). Alternatively, this row can be Tap 3 from VoiceDelay or SummedDelay, or a mono sum of the Master's submix. In a CEE Combination Preset, this row can supply the left channel output of DSP 2, or the mono sum of DSP 2's output, to DSP 3 — allowing for post-processing of DSP 2's output by DSP 3.
	R	The audio submix from the right channel of the Master (SR column after saturation, convolutions and recirculator). Alternatively, this row can be Tap 4 from VoiceDelay or SummedDelay, or a mono sum of Tap 3 and Tap 4. In a CEE Combination Preset, this row can supply the right channel output of DSP 2 to DSP 3 — allowing for post-processing of DSP 1's output by DSP 3.

5.2. Direct Sources

Direct	+	Adds formulas or constants directly into a matrix column (without multiplying by an audio signal).
	+	Adds formulas or constants directly into a matrix column.
	-	Subtracts formulas or constants directly from a matrix column.
	×	Multiplies a matrix column by a formula or constant.

6. EaganMatrix Destinations Reference

Any EaganMatrix Source (matrix row) may be multiplied by a formula and added to an EaganMatrix Destination (matrix column) by placing the formula at the intersection of the row and column.

6.1. Master

Master (Output Section of the EaganMatrix)			
Master	SL	Left	Audio input into the left channel of the Master Section, to be processed by the Master's soft saturation, convolution, recirculator, and second convolution. The Master Section processes the stereo audio sum of all voices.
	SR	Right	Audio input into the right channel of the Master Section, to be processed by the Master's soft saturation, convolution, recirculator, and second convolution. The Master Section processes the stereo audio sum of all voices.
Cnv1	I	Interp	Values 0 to 3, controls an interpolation between four convolution responses. Select the responses, as well as a length and tune value (spectral shift) for each response, in the Convolution area below the matrix.
	M	Mix	Dry/wet mix of the pre-recirculator convolution output, 0 = dry, 1 = 100% wet.
Recirc	R1 R2 R3 R4 M	R1 R2 R3 Time Mix	Each of the two Recirc matrix columns contributes to a recirculator control: R1, R2, R3, R4, or Mix. The control is selectable by clicking on the column's heading. The column value (scaled 0..1) will be added to the corresponding Recirculator Control Dial value (cc20..cc24 scaled 0..127; see Continuum User Guide). The first Recirc column defaults to "R4", which is the time control for all the recirculator algorithms. The second Recirc column defaults to "M", the dry/wet mix of the recirculator output (0 = dry, 1 = 100% wet).
Cnv2	I	Interp	Values 0 to 3, controls an interpolation between four convolution responses. Select the responses, as well as a length and tune value (spectral shift) for each response, in the Convolution area below the matrix.
	M	Mix	Dry/wet mix of the post-recirculator convolution output, 0 = dry, 1 = 100% wet.

Master (Output Section of the EaganMatrix)			
Out	M	Mix	Controls the final mix: SL SR processed by convolutions and recirculator (value 0), and unprocessed L R (value 1).
Master	L	Left	Input into left channel of the Master. This input is not processed by saturation, convolutions, and recirculator.
	R	Right	Input into right channel of the Master. This input is not processed by saturation, convolutions, and recirculator.

Notes for Master Section:

In the Master Section, columns SL, SR, L, and R, are evaluated independently for each voice. The formulaic output for all other Master Section columns is derived from a pressure-weighted average of all fingers in contact with the playing surface.

The Master produces a stereo **submix** as follows:

Stereo sum (SL, SR) from all voices → Saturation Waveshaper
→ Pre-Recirculator Convolution → Recirculator → Post-Recirculator Convolution → Submix

The Master's submix is fed back into the matrix on the **Sub L** and **Sub R** matrix rows, for optional further processing.

The Master's **final mix** is controlled by the Master's rightmost M column. For the final mix, the submix is mixed with the sum of L and R from all voices. This final mix is scaled by the Gain dial and sent to the Continuum Fingerboard's headphone and AES3 outputs.

For saturation and overload effects, use large formula values (typically larger than 1) in the Master SL and SR columns, and use the Gain dial to attenuate the final output level. The Master Section will waveshape high amplitude audio from the SL and SR columns, or soft clip if Via Limiter (see Ancillary Operator Mechanism) is selected in the Master SL and SR formulas. Waveshaping and clipping in the Master Section can be completely avoided by using small formula values (typically less than 1) in the Master SL and SR columns, and increasing the Gain dial to compensate. Also, no waveshaping is done on the Master L and R columns. Technical detail: No loss of accuracy results from scaling values up or down, because the EaganMatrix is based on floating-point computations.

Examples of Master Section usage in EaganMatrix System Presets:

Sine Wave: Very simple sound; Oscillator 1 is connected to Master Section's SL and SR columns using predefined Formula Z.

Pulsar: Combination of Bank A and Filter 1 outputs are summed in the Master Section.

Sputnik's Dream: Uses formulas to control the Master Section's recirculator and post-recirculator convolution.

Left Hand Filter Echo: Makes use of the L and R columns of the Master Section to bypass the Master Section's recirculator and convolutions.

6.2. Oscillators / Multimode Filters

Oscillators			
Osc1	▲	Audio Input	Oscillator 1 phase modulation input, 0 = 0 radians, 1 = 2π radians. Use this for modulation or waveshaping, or as a seed value for Noise from Seed. Leave this blank for an oscillator without modulation.
	F	Frequency	Oscillator 1 frequency control, in kHz. Leave this blank for a waveshaper, and for Noise from Seed.
	S	Spectral Balance	Oscillator 1 spectral balance, 0 for pure sine, >0 to add harmonics, 1 for max harmonic amplitudes. For waveshaping, this provides a continuous real-time control of the the nonlinear filter's transfer function.
	S	Spread	If Expose Phase is selected, this is the Frequency Spread (in Hz) for the oscillator's 7 phase generators.
	T	Trigger	If Noise from Seed is selected, a transition from T≤0 to T>0 triggers use of the ▲ input as a new random noise seed value.
Osc2..5			Four more oscillators like Oscillator 1.

Notes for Oscillators:

1. The EaganMatrix has five 3-column Oscillator/Filter slots; each of the five is configurable as Oscillator or MultiMode Filter.
2. When the Audio Input (phase modulation) and Spectral Balance (S) are both equal to zero, this functions as a pure sine wave source at frequency F. The oscillator is most efficient when the S column is left blank due to optimization of the sine algorithm.
3. An oscillator can act as a waveshaper when audio is routed into the Audio Input (phase), and the F (frequency) column is blank (0 kHz). The S column can be used to dynamically alter the waveshaping function, making for unique nonlinear filtering effects.
4. Select Normal DSF for harmonic amplitudes that decrease with harmonic number in an exponential fashion determined by the S column. Select Integrated DSF for harmonic amplitudes proportional to the inverse of harmonic number; S=0.8 has the same harmonic amplitudes as a sawtooth. (Refer to the graphs below for Normal DSF and Integrated DSF waveforms.) Select Expose Phase to output the oscillator's 7 internal phase generators, with the S column controlling frequency spread in Hz. Expose Phase is equivalent to a sum of 7 non-bandlimited sawtooth waveforms (for bandlimited saws, see "WaveBank"). Select Noise from Seed for a random noise generator based on a seed value. If the same seed value is used for each note (a constant in the ▲ input column and W in the T column), the sequence of random numbers will be the same for each note.

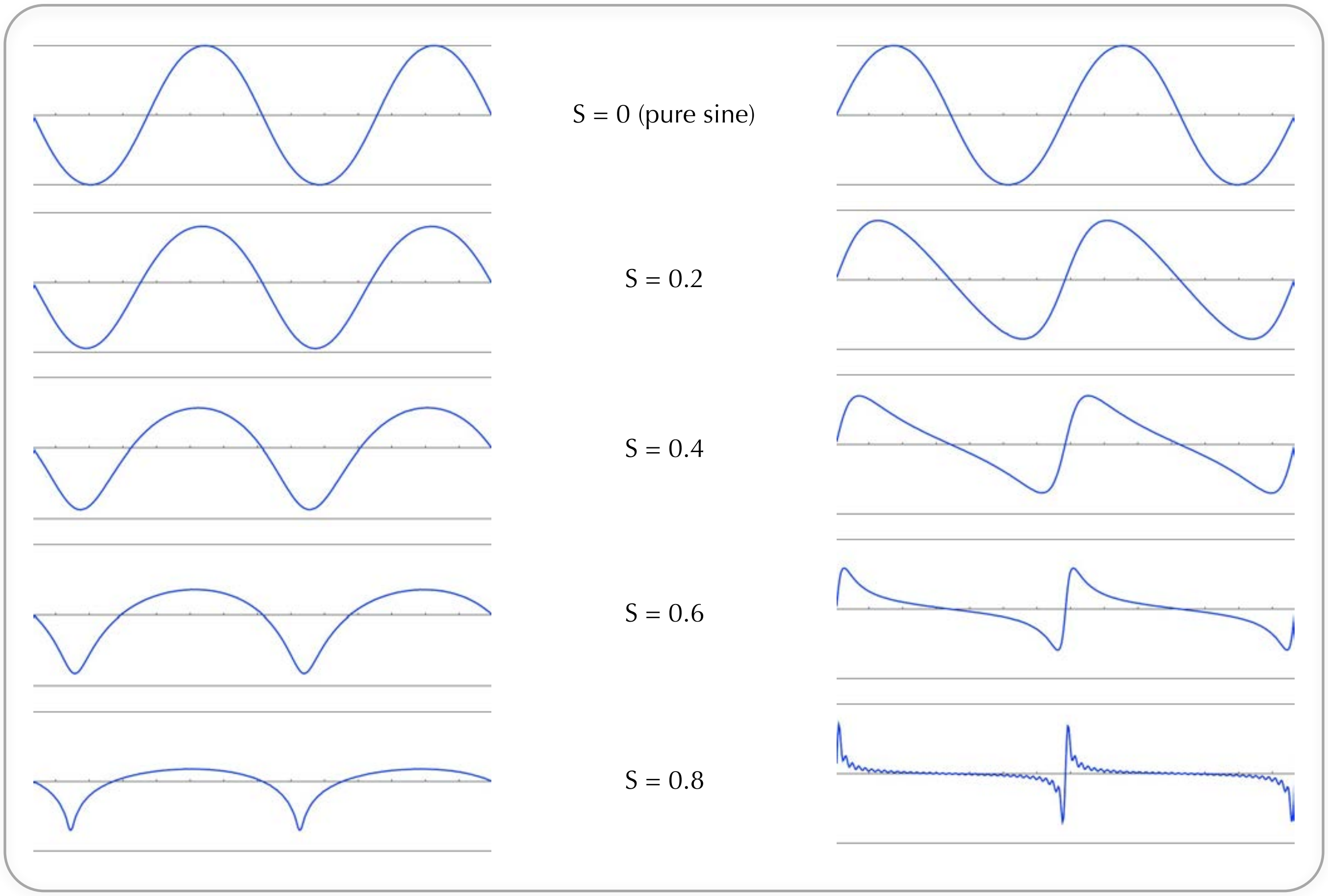
Examples of Oscillator usage in EaganMatrix System Presets:

FM Bells: FM synthesis technique.

Harmonic Break: Waveshaping using the phase input of an Oscillator.

Meccano: Changes the frequency of an Oscillator with a Sample-and-Hold (Formula A's sample-and-hold of Formula B is triggered by SG1).

Sawz: Combines two bandlimited Oscillators (with Integrated DSF) and a non-bandlimited Oscillator (with Exposed Phase) to create an ensemble sound over the whole pitch range, without audible aliasing.



Integrated DSF *and* ***Normal (non-integrated) DSF*** *Oscillator Waveforms for different S values*

Note: The 0.8 integrated waveform has harmonic amplitudes equal to a sawtooth (6 dB/octave rolloff).

Mini Shepard: Crossfades two Oscillators to create the illusion of an ever-rising pitch.

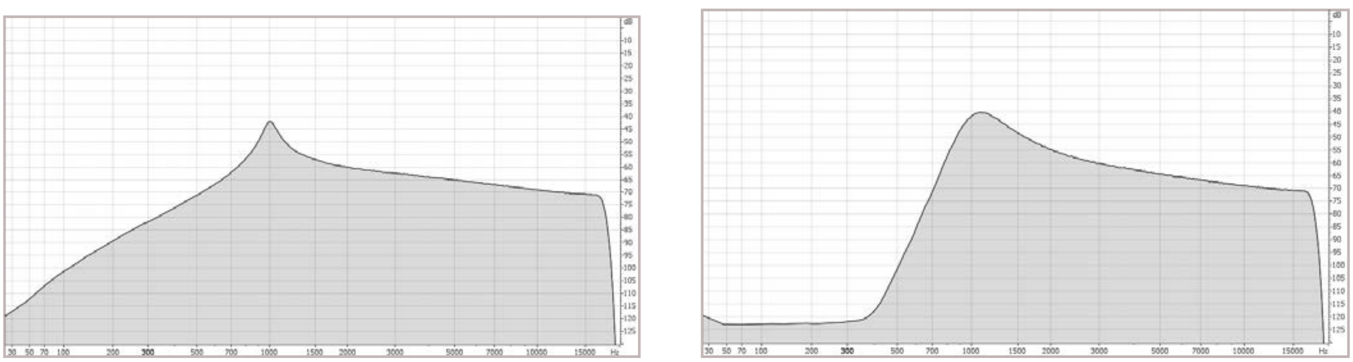
Pad Tie: Changes the spectrum of Oscillators dynamically (Formula E in the S column).

TriMod: A combination of frequency modulation and waveshaping.

Multimode Filters			
1 (LPF, HPF, BPF, LPS, HPS, NOF, APF, LP1, HP1)	▲	Audio Input	Filter audio input. Possible 2-pole filters (12 dB/octave cutoff slope) are Low Pass, High Pass, Band Pass, Low Pass Shelf, High Pass Shelf, Notch, and All Pass. Possible 1-pole (6 dB/octave) filters are Low Pass and High Pass.
	F	Cutoff Frequency	Filter frequency control, in kHz.
	B	Bandwidth	Filter bandwidth control, smaller values narrow the bandwidth and increase resonance.
2..5			Four more filters like Multimode Filter 1.

Notes for Multimode Filters:

1. The EaganMatrix has five 3-column Oscillator/Filter slots; each of the five is configurable as Oscillator or MultiMode Filter.
2. For MultiMode filters, **Cascade** choices are 1, 2, 3, or 4, corresponding to 12, 24, 36, or 48 dB/octave cutoff slope for 2-pole filters, and 6, 12, 18, or 24 dB/octave cutoff slope for 1-pole filters. Technical detail: When cascading 2-pole filters to create 4, 6, and 8 pole filters, the bandwidths are adjusted to $B^{1/2}$, $B^{1/3}$, and $B^{1/4}$ so that the same perceived cutoff frequency is maintained at each cascade value. See example filter responses below.
3. Select **Normal** to recompute filter coefficients once per millisecond, based on F and B column values. Select **Extreme** to recompute filter coefficients every sample interval. If the F or B columns have audio-rate signals, Extreme should be selected.



Resonant Low Pass filter responses for F = 1.0, B = 0.1, cascade = 1 (left image) and cascade = 4 (right image). In both images, reducing B will make a narrower and more pronounced resonance at the cutoff frequency; increasing B (0.1 < B < 1) will make a wider and shallower resonance; B = 1 will result in no resonance, and, as B is increased beyond 1, the slope near the cutoff frequency will be reduced, making for a gradual filter.

Examples of Multimode Filter usage in EaganMatrix System Presets:

Ping Pong: Extensive filtering using two Band Pass filters.

Geiger Insects: An emulation of imaginary insects, using High Pass filters controlled by a cyclical Shape Generator.

Meccano: Changes the frequency of an Oscillator with a Sample-and-Hold.

Mini Shepard Breathing: Low Pass Shelf filter on noise.

Mutate Looper: Uses a Low Pass Filter to modify the output of a delay.

Razor Loops: Extensive filtering using two All Pass filters.

Spit Tube: Modulates the cutoff frequency of a filter at audio rates, with "Extreme" selected for the filter.

Swept Delay: Uses High Pass Shelf filters to modulate a sustained delay loop.

Three Sines: Uses a fixed-frequency Low Pass filter to reduce possible aliasing artifacts.

VelVlaCelBass: A High Pass filter adds high frequency energy into the sound, based on finger pressure.

6.3. Multipurpose Banks: BiqBank

Multipurpose Banks: BiqBank (Scaled Resonances)			
BiqBank **1**	▲	Audio Input	Audio input into the biquad bandpass filters in the BiqBank.
	F	Frequency	Frequency (in kHz) for the first bandpass filter in the BiqBank.
	sF qF	Frequency Spread	sF; 1 for harmonic spacing of bandpass filters, >1 for stretched spacing, <1 for compressed spacing. qF; 0 for harmonic spacing, >0 for offsets proportional to harmonic-number-squared.
	B	Bandwidth	Bandpass bandwidth control, smaller values increase resonance.
	sB qB	Bandwidth Spread	sB linear, qB quadratic by mode number. 0 means each mode has the same bandwidth, >0 means modes farther from Centre (C column) have increased bandwidth, <0 means modes farther from Centre have decreased bandwidth.
	C	Centre	0 means first bandpass filter is Centre, 1 means 8th bandpass filter is Centre.
	sA	Amplitude Spread	0 for uniform amplitude bandpass filters, >0 reduced amplitude farther from Centre.
2			A second filter bank like BiqBank 1, available in Bank B.

Notes for BiqBank:

BiqBank, BiqMode, BiqMouth, and ModMan are four different ways to control the same underlying technology: a set of biquad bandpass filters.

BiqBank provides bandpass filters on an audio-rate input, with filter parameters controlled by the matrix as described in the table above. Each BiqBank may consist of 8 or 48 bandpass filters. The indicator above the first BiqBank matrix column is off for Normal (8 bandpass filters), on for Extended (48 bandpass filters). Click on the Indicator to change its state.

The Deflation Limit (above the last BiqBank matrix column) avoids excessive buildup of energy inside the BiqBank, but has no effect under normal playing conditions. Set the Deflation Limit to the largest value (between 0 and 7) that does not affect the timbre of the sound. Click on the Deflation Limit to change its value.

Examples of BiqBank usage in EaganMatrix System Presets:

Bouncer: A narrow bandwidth BiqBank creates a sound like a bouncing tonal object, using noise for the BiqBank's audio input.

Icicles: A narrow bandwidth BiqBank creates a glass-like repeating tonal object, using noise for the BiqBank's audio input.

Golliclock: Oscillators processed though a BiqBank.

Marlin Perkins: A tongue-drum simulation.

Metal Rainstick: A wide bandwidth BiqBank creates a breathy repetitive tonal object.

Model String Wind: A string and wind simulation, using an Oscillator to waveshape audio data in a BiqBank feedback loop.

Resonant Drum 1 and 2: Pitched drum emulations.

Rub String: Novel use of the update noise from a formula to resonate within a pair of BiqBanks. To produce the update noise, interpolation has been set to zero in Formula A and Formula F.

6.4. Multipurpose Banks: BiqMode

Multipurpose Banks: BiqMode (Mode Amplitudes)			
BiqMode **1**	▲	Audio Input	Input to the mode filters in the BiqMode.
	F	Frequency	Frequency (in kHz) for the first mode in the BiqMode.
	sF	Frequency Spread	1 for harmonic spacing of modes, >1 stretched harmonics, <1 compressed.
	B	Bandwidth	Bandwidth control, smaller values increase mode resonance.
	sB	Bandwidth Spread	0 means each mode has the same bandwidth, >0 means higher modes have increasing bandwidth, <0 higher modes have decreasing bandwidth.
	1A 1F	1 Graph Scale	Scale factor for 1 Graph, offsetting Mode dB Amplitudes. See below. Scale factor for 1 Graph, offsetting Mode % Frequencies. See below.
	2A 2B	2 Graph Scale	Scale factor for 2 Graph, offsetting Mode dB Amplitudes. See below. Scale factor for 2 Graph, offsetting Mode dB Bandwidth. See below.
2			A second filter bank like BiqMode 1, available in Bank B.

Notes for BiqMode:

BiqBank, BiqMode, BiqMouth, and ModMan are four different ways to control the same underlying technology: a set of biquad bandpass filters.

BiqMode is suitable for Modal Physical Modeling. In Modal Modeling terminology, each biquad bandpass filter is called a "mode". The amplitudes for each mode are computed from dB amplitude values specified in three amplitude graphs. The three graphs are called A Graph (dB amplitude values), 1 Graph (first set of dB amplitude offsets), and 2 Graph (second set of dB amplitude offsets). These graphs are combined under control of matrix columns, resulting in time-varying mode amplitudes, as explained on the next page.

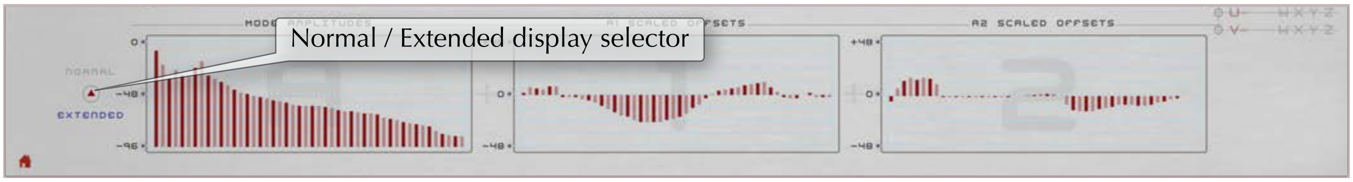
Alternatively, by selecting 1F instead of 1A for the matrix column, the 1 Graph may be used for Scaled % Frequency Offsets. By selecting 2B instead of 2A for the matrix column, the 2 Graph may be used for Scaled dB Bandwidth Offsets.

Each BiqMode may consist of 8 or 48 modes. The indicator above the first BiqMode matrix column is off for Normal (8 modes), on for Extended (48 modes). Click on the Indicator to change its state.

The Deflation Limit (above the last BiqMode matrix column) avoids excessive buildup of energy inside the BiqMode, but has no effect under normal playing conditions. Set the Deflation Limit to the largest value (between 0 and 7) that does not affect the timbre of the sound. Click on the Deflation Limit to change its value.

BiqMode Graphs: Click on the the word Graph to inspect and edit three graphs: Mode Amplitudes (A graph), first set of dB amplitude offsets (1 Graph), and second set of dB amplitude offsets (2 Graph). In Slide Wind EM, all three graphs are used to determine mode amplitudes. The dB amplitude of each of the modes is computed from the Mode Amplitude A graph, plus matrix column 1A times the 1 Graph, plus matrix column 2A times the 2 Graph. These are the three graphs for the 8 modes used by the Slide Wind EM sound:

If you are using a 48 mode BiqBank (if the indicator located above the first BiqMode matrix column is lit), click to the left of the graphs to switch from showing the first 8 modes to all 48 modes.



Edit the graphs by clicking with the mouse to change one mode's amplitude; or by pressing left arrow or right arrow to change the mode at the mouse cursor by 1 dB; or by dragging the mouse to change the amplitudes of many modes with one motion.

Examples of BiqMode usage in EaganMatrix System Presets:

French Sax: Uses an Oscillator as a waveshaper in a BiqMode feedback loop, to create a cross between a sax and harmonica.

Slide Wind EM: Uses an Oscillator as a waveshaper in a BiqMode feedback loop, to create a changing-body-size wind instrument.

Space Flute: Uses an Oscillator as a waveshaper in a BiqMode feedback loop, to create a flute-like sound with spacey echo repeats.

6.5. Multipurpose Banks: BiqMouth

Multipurpose Banks: BiqMouth (Vocal Formants)			
BiqMouth **1**	▲	Audio Input	Audio input into the vocal formant filter bank, a series of five biquad formant filters.
	Sh	Mouth Shape	A series of mouth shapes, selectable 0 to 4.8 at 0.1 intervals with smooth morphing between adjacent shapes. See note below for mouth shape listing.
	sF	Frequency Spread	Multiplier on formant frequencies, 1.0 = normal.
	oF	Frequency Offset	Offset on formant frequencies, 0.0 (blank column) = normal.
	sB	Bandwidth Spread	Formant bandwidth multiplier, 1.0 means each formant has default bandwidth, > 1.0 means formants farther from Centre (see below) have increased bandwidth, < 1.0 means formants farther from Centre have decreased bandwidth.
	C	Centre	BiqMouth Centre control, 0 means first formant is centre, 0.625 means last (5th) formant is centre.
	sA	Amplitude Spread	BiqMouth amplitude spread, 0 means no amplitude scaling of formants, > 0 means reduced amplitude of formants farther from Centre.

Multipurpose Banks: BiqMouth (Vocal Formants)			
2			A second filter like BiqMouth 1, available in Bank B.

Notes for BiqMouth (Vocal Formants):

BiqBank, BiqMode, BiqMouth, and ModMan are four different ways to control the same underlying technology: a set of biquad bandpass filters.

BiqMouth is suitable for vocal filtering of an audio-rate input. In voice modeling terminology, each biquad bandpass filter is called a "formant". The matrix columns control the parameters of 5 formants, to simulate a variety of mouth shapes.

Each mouth shape is implemented using 5 vocal formants. Five biquad bandpass filters are set at the appropriate frequencies, amplitudes, and bandwidths to imitate the resonances of the mouth shape.

The Deflation Limit (above the last BiqMouth matrix column) avoids excessive buildup of energy inside the BiqMouth, but has no effect under normal playing conditions. Set the Deflation Limit to the largest value (between 0 and 7) that does not affect the timbre of the sound. Click on the Deflation Limit to change its value.

BiqMouth Mouth Shapes:

Mouth Shapes	Symbol	Male Primary "Sh" Column Value	Female Primary "Sh" Column Value
F**a**ther	**ä**	0	3.5
M**a**de	**ā**	0.1	3.6
F**oo**d	**o͞o**	0.2	3.7
G**o**	**ō**	0.3	3.8
P**a**d	**a**	0.4	n/a
M**e**rry	**e**	0.5	3.9
M**u**d	**ə**	0.6	n/a

Mouth Shape Selection Chart: The table below shows how the same mouth shapes are selectable with several different "Sh" column values. Mouth shapes have been repeated in different combinations to allow for a maximum of adjacent morphing possibilities. Male mouth shapes have "M" prefix, female mouth shapes have "F" prefix. Male mouth shapes are shown with a yellow background and female mouth shapes are shown with a purple background.

Sh Value	0	0.1	0.2	0.3	0.4	0.5	0.6	0.7	0.8	0.9
0	M:ä	M:ā	M:o͞o	M:ō	M:a	M:e	M:ə	M:ä	M:o͞o	M:a
1	M:ə	M:ā	M:ō	M:e	M:ä	M:ō	M:ə	M:a	M:ä	M:e
2	M:ā	M:a	M:ə	M:o͞o	M:e	F:ä	M:ä	F:ā	M:ā	F:o͞o
3	M:o͞o	F:ō	M:ō	F:e	M:a	F:ä	F:ā	F:o͞o	F:ō	F:e
4	F:ä	F:o͞o	F:e	F:ā	F:ō	F:ä				

Examples of BiqMouth usage in EaganMatrix System Presets:

Larynx Horn: A vocal-sounding brass instrument.

Mantra Voice: A dreamy ambient vocal sound.

Tibetan Throat Stick: Vocal formant processing of a harmonic manipulation (HarMan) of data recorded into the VoiceDelay.

6.6. Multipurpose Banks: SineBank

Multipurpose Banks: SineBank (Sixteen Sine Wave Oscillators)			
SineBank **1**	W	Weighting	Weighting of the amplitudes of the sixteen sine wave oscillators in the SineBank. 0 is equal weighting. Values higher than 0 up to 1 favour even harmonics, and values lower than 0 to -1 favor odd harmonics.
	F	Frequency	Frequency (in kHz) for the first sine oscillator in the SineBank.
	sF	Frequency Spread	Frequency spread, 1 for harmonic spacing of sines, >1 stretched, <1 compressed.
	P	Phase	Phase modulation input for first sine oscillator.
	sP	Phase Spread	Phase modulation spread between consecutive sine oscillators.
	C	Centre	Centre control, 0 means first sine oscillator is centre, 1 means last is centre.
	sA	Amplitude Spread	Amplitude spread, 0 means no amplitude scaling of sine oscillators, >0 means reduced amplitude of oscillators farther from Centre.
2			A second bank like SineBank 1, available in Bank B.

Notes for SineBank:

Unlike other oscillators in the EaganMatrix, SineBank oscillators use simple table lookup for sine values. As a result, the SineBank may have audible table lookup noise at very low frequencies. Use phase modulation or some other technique to mask such noise, or use a different method to generate sinusoids (e.g. BiqBank with narrow bandwidths).

Examples of SineBank usage in EaganMatrix System Presets:

SinebankFM String: Uses two SineBanks to frequency modulate Oscillators.

Sinebank Horn: A simulation of a brass-like texture.

SinePhases: Creates texture changes using SineBank phase manipulation.

6.7. Multipurpose Banks: SineSpray

Multipurpose Banks: SineSpray (Sine Grains)			
SineSpray **1**	A	Amplitude	Amplitude of Sine Spray. If the amplitude is 0, no spray is generated (conserves DSP processing).
	F	Frequency	Frequency of Sine Spray's fundamental, in kHz. It is common to add randomness to this value to create audible clusters within the Sine Spray.
	H	Harmonic Truncation	Harmonic truncation, in kHz. Small H creates long sine grains within the spray. Larger H shortens the grains and adds harmonics. As H increases, high harmonics increase and low harmonics get truncated. Try half the S column value.
	P	Phase	Phase offset for sinusoid within the Spray.
	S	Sine Frequency	Frequency for sinusoid within the Spray, in kHz.
2			A second SineSpray like SineSpray 1, available in Bank B or in Bank C.

Examples of SineSpray usage in EaganMatrix System Presets:

Bird Echoer and Bird Whistler: Creates a looped harmonic shift using a SineSpray, reminiscent of a bird call.

Harmonic Break: Controls harmonic timbre shifts with front-to-back finger movement.

Sonogram: Creates a diffuse-to-focussed noise texture, using noise modulation of a SineSpray.

SpaceJaw: A thin band-passed texture, using two SineSprays to create the stereo texture.

Three Cycles: Subtle modulations in the SineSpray create a cyclical spinning texture.

6.8. Multipurpose Banks: WaveBank

Multipurpose Banks: WaveBank (Waveform Generator Bank)			
WaveBank **1**	A	Amplitude	Amplitude of WaveBank. If the amplitude is 0, no output is generated (conserves DSP processing).
	F1	Frequency	Frequency of first wave in WaveBank, in kHz.
	F2 - F5	Frequency	Frequency of second through fifth wave in WaveBank. If a column is left blank, fewer waves will be generated by the WaveBank.
	D	Duty	Duty cycle for waves generated by the WaveBank. Values range from 0 to 1 (or equivalently, 0 to -1). Value 0 is normal; if the WaveBank has Square selected, 0 is a 50% duty cycle square, and nonzero is a pulse wave.
	W	Weighting	Weighting of the amplitudes of the waves generated by the WaveBank. 0 is equal weighting. Values higher than 0 up to 1 favour even numbered waves, and values lower than 0 to -1 favor odd numbered waves.
	oD	Duty Offset	Offset for the D column's Duty Cycle value. The offset is added once to the second wave in the WaveBank, twice to the third, etc.
2			A second WaveBank like WaveBank 1, available in Bank B or C.

Notes for WaveBank:

The WaveBank popup menu provides three choices: Sawtooth, Square, or Triangle wave.

The delay memory of VoiceDelay, SummedDelay, MicroDelay, and FormulaDelay is used by the WaveBank, HarMan, and ModMan.

Note: When you use a WaveBank, it occupies half the memory of the VoiceDelay, SummedDelay, MicroDelay, and FormulaDelay. If you have one WaveBank (in Bank A or Bank B), you can also have a VoiceDelay, SummedDelay, MicroDelay, and FormulaDelay in Bank C (which will operate using half its normal memory). If you have two WaveBanks (both Bank A and Bank B) as well as a Delay in Bank C, both WaveBanks must specify the same kind of wave (Sawtooth, Square, or Triangle wave).

Examples of WaveBank usage in EaganMatrix System Presets:

Boson Particles: Uses a WaveBank to generate triangle waves. The WaveBank's W column value changes with finger Y position.

String Machine: Uses a WaveBank to generate sawtooth waves, to emulate an old-style analogue string ensemble synthesizer. Finger X position affects the duty cycle, producing a more nasal sound for lower strings.

Squares in Space: Uses a WaveBank to generate four square (pulse) waves with slowly changing duty cycles.

6.9. Multipurpose Banks: HarMan

Multipurpose Banks: HarMan (Harmonic Manipulator)			
HarMan **1**	A	Amplitude	Amplitude of Harmonic Manipulator. If the amplitude is 0, no output is generated (conserves DSP processing).
	F	Frequency	Frequency of Harmonic Manipulator's fundamental, in kHz.
	H	Harmonic Truncation	Harmonic truncation, in kHz. More of the low harmonics will be truncated as this value increases. Try .16 times the S column value. (Larger H shortens time duration of each spectrum.)
	I	Interpolation Index	Interpolation between spectra selected in the Spectral Set popup menu. Value 0 to 1: 0 = first spectrum in the Set, 1 = last spectrum. Additional special values when "Live" is selected in the Spectral Set popup menu: -1 = quantize I to align with 2 * H, -2 = align when S > 1.
	S	Spectral Dilation	1 means no change in spectral envelope, > 1 stretches the spectral envelope, < 1 compresses. Normally the S column and the H column differ only by a scale factor.
2			A second Harmonic Manipulator like HarMan 1, available in Bank B or C.

Notes for HarMan:

The delay memory of VoiceDelay, SummedDelay, MicroDelay, and FormulaDelay is used by the WaveBank, HarMan, and ModMan. For HarMan, a popup menu allows you to select a predefined **Spectral Set** to load into the Delay memory, or **Live** to manipulate time-domain data (312.5 Hz fundamental) you record into the the SummedDelay.

Note: When you select a Spectral Set, it occupies half the memory of the VoiceDelay, SummedDelay, MicroDelay, and FormulaDelay. If you have one Manipulator with Spectral Set (in Bank A or Bank B), you can also have a VoiceDelay, SummedDelay, MicroDelay, and FormulaDelay in Bank C (which will operate using half its normal memory). If you have two Manipulators (both Bank A and Bank B) as well as a Delay in Bank C, both Manipulators must specify the same Spectral Set.

Examples of HarMan usage in EaganMatrix System Presets:

Bell Rub: Uses finger pressure to control a cyclical traversal of bell spectra.

Cycle Kalimba: A moving kalimba-like sound.

GrainSilo Woodwind: A generic woodwind-like timbre with a wide pitch range.

SiloString Pizz: Transforms spectra from a viola pizzicato into a guitar string.

Spinning Duet: Creates a stereo spinning texture by cycling through spectra from a flute.

VlaVlaCelBass: A full-range solo orchestral string sound, created by manipulating spectra from a single viola tone.

6.10. Multipurpose Banks: ModMan

Multipurpose Banks: ModMan (Modal Manipulator)			
ModMan **1**	▲	Audio Input	Audio input into all of the modes of the ModMan.
	F	Frequency	Frequency (in kHz) for the first mode in the ModMan.
	 sF qF sA	 Frequency Spread Quadratic Frequency Amplitude Spread	Select this column's function by clicking on the column's heading. sF=1 for harmonic spacing of modes, >1 stretched, <1 compressed. qF times mode-number-squared is each mode's frequency offset. sA>0 reduces amplitude of modes farther from Centre (see below).
	B	Bandwidth	Bandpass bandwidth, smaller values increase mode resonance.
	 sB qB	 Bandwidth Spread Quadratic Bandwidth	Select this column's function by clicking on the column's heading. sB>0 increase bandwidth farther from Centre (see below), <0 decrease. qB times distance-from-Center-squared is bandwidth offset.
	C	Centre	0 means first mode is Centre, 1 means the 8th mode is Centre.
	I	Interpolation Index	Interpolation between spectra selected in the Spectral Set popup menu. Value 0 to 1: 0 = first spectrum in the Set, 1 = last spectrum.
	S	Spectral Dilation	1 for unchanged spectral envelope, >1 stretched, <1 compressed. Normally the S and H columns differ only by a scale factor.
2			A second filter bank like BiqBank 1, available in Bank B.

Notes for ModMan:

The delay memory of VoiceDelay, SummedDelay, MicroDelay, and FormulaDelay is used by the WaveBank, HarMan, and ModMan. For ModMan, a popup menu allows you to select a predefined **Spectral Set** to load into the Delay memory, or **Live** to manipulate spectral envelope data (spectral period 620.69 Hz) you record into the SummedDelay.

The Deflation Limit (above the last ModMan matrix column) avoids excessive buildup of energy inside the ModMan, but has no effect under normal playing conditions. Set the Deflation Limit to the largest value (between 0 and 7) that does not affect the timbre of the sound. Click on the Deflation Limit to change its value.

Note: When you select a Spectral Set, it occupies half the memory of the VoiceDelay, SummedDelay, MicroDelay, and FormulaDelay. If you have one Manipulator with Spectral Set (in Bank A or Bank B), you can also have a VoiceDelay, SummedDelay, MicroDelay, and FormulaDelay in Bank C (which will operate using half its normal memory). If you have two Manipulators (both Bank A and Bank B) as well as a Delay in Bank C, both Manipulators must specify the same Spectral Set.

Examples of ModMan usage in EaganMatrix System Presets:

Harp: Uses the Vln PIzz Soft spectra to create a sustained harp-like sound.

Koto: Uses the Vln PIzz Snap spectra to create a sustained koto-like sound.

6.11. Multipurpose Banks: CVC Control Voltages

Multipurpose Banks: CVC (Continuum Voltage Converter)		
W	1	CVC control voltage output 1.
X	2	CVC control voltage output 2.
Y	3	CVC control voltage output 3.
Z	4	CVC control voltage output 4.
W	5	CVC control voltage output 5.
X	6	CVC control voltage output 6.
Y	7	CVC control voltage output 7.
Z	8	CVC control voltage output 8.

Notes for CVC:

Select CVC in Bank B or in Bank C to create preprocessed Continuum Voltage Converter outputs using the EaganMatrix.

If the EaganMatrix is supplying CV voltages, the Editor will display CVC via Matrix in the Editor. Otherwise, the Editor will display the Standard CV Definition being used. See the CVC User Guide for details.

When a CVC bank is in use, the CVC will be controlled by the EaganMatrix, with eight control voltages per note (two voices in one CVC). The columns in the EaganMatrix CVC bank specify voltage values.

Two of the Continuum Fingerboard's split modes will override the EaganMatrix control of the CVC:

Internal Sound Below Split - the EaganMatrix sound will be below the split point and the CVC control will be above the split point using the selected Standard CV Definition.

Internal Sound Above Split - the EaganMatrix sound will be above the split point and the CVC control will be below the split point using the selected Standard CV Definition.

6.12. Multipurpose Banks: VoiceDelay

Multipurpose Banks: VoiceDelay (Audio Delay Buffer, each voice separate)			
VoiceDelay	▲	Audio Input	Audio input into the VoiceDelay. If the VoiceDelay is operating at reduced sample rate, this input passes through an optional anti-aliasing filter.
	D	Delay Time	Delay time. 0 = no delay, 1 = delay by max amount selected in the Delay menu. The primary output of the VoiceDelay is available on the Bank C row of the EaganMatrix.
	T1	Tap 1	Delay time for Tap 1. This is a multiplier on the D column value. 0 = no delay, 1 = delay by D value. The Tap 1 and 2 outputs are available as an alternative on the AesIn matrix rows.
	T2	Tap 2	Delay time for Tap 2, values 0 to 1. 0 = no delay, 1 = delay by D value.
	T3	Tap 3	Delay time for Tap 3. 0 = no delay, 1 = delay by D value. The Tap 3 and 4 outputs are available as an alternative on the Sub matrix rows.
	T4	Tap 4	Delay time for Tap 4. 0 = no delay, 1 = delay by D value.
	H	Hold	Greater than zero stops recording new data into the VoiceDelay.

Notes for VoiceDelay:

VoiceDelay, SummedDelay, MicroDelay, and FormulaDelay are four different ways to use the one underlying delay memory. This delay memory is also used by the WaveBank, HarMan, and ModMan.

The **VoiceDelay** provides a multi-tap audio delay buffer, separate for each voice. VoiceDelay is suitable for creating chorus and flanging effects, can be used for fast slap delays. VoiceDelay is available in Bank C.

The Delay popup menu has a choice of maximum delay times for the VoiceDelay.

The sample rate of the VoiceDelay memory is limited by two variables: maximum delay and polyphony. Longer delays or higher polyphony may result in reduced sample rate for the delay memory. The **Anti-Aliasing Low Pass Filter** indicator above the first Bank C column will light up when the delay memory is at reduced sample rate. If you do not want an Anti-Aliasing filter at reduced sample rates, click on the indicator to disable the filter.

Note: When a Manipulator with Spectral Set is in use (in Bank A or Bank B), the VoiceDelay will still operate, using half the normal amount of delay memory.

Examples of VoiceDelay usage in EaganMatrix System Presets:

Echo 8va: An echo that combines elements not in the original dry note.

Echo Star: Uses an echo to modify the pitch of an Oscillator.

Follower: Delays a control-level signal to create a distinct single echo, which follows the pitch of the dry note with an alternate texture.

Mutate Looper: Sonically modifies an extended delay by filters and oscillators.

Ping Pong: Subtle delay creates stereo depth.

Razor Loops: Short delay values create a pitched-noise texture.

Swept Delay: Sonically modifies a delay with slow filter sweeps.

Waterphone Strings: Uses VoiceDelay tap outputs to create a chorusing effect.

6.13. Multipurpose Banks: SummedDelay

Multipurpose Banks: SummedDelay (Audio Delay Buffer, shared by all voices)			
SummedDelay	▲	Audio Input	This voice's audio input contribution to the shared SummedDelay. If the SummedDelay is operating at reduced sample rate, this input passes through an optional anti-aliasing filter.
	D	Delay Time	Delay time. 0 = no delay, 1 = delay by max amount selected in Delay menu. The primary output of the SummedDelay is available on the Bank C row of the EaganMatrix.
	T1	Tap 1	Delay time for Tap 1. This is a multiplier on the D value. 0 = no delay, 1 = delay by D value. The Tap 1 and 2 outputs are available as an alternative on the AesIn matrix rows.
	T2	Tap 2	Delay time for Tap 2. 0 = no delay, 1 = delay by D value.
	T3	Tap 3	Delay time for Tap 3. 0 = no delay, 1 = delay by D value. The Tap 3 and 4 outputs are available as an alternative on the Sub matrix rows.
	T4	Tap 4	Delay time for Tap 4. 0 = no delay, 1 = delay by D value.
	H	Hold	Greater than zero stops recording new data into the SummedDelay.

Notes for SummedDelay:

VoiceDelay, SummedDelay, MicroDelay, and FormulaDelay are four different ways to use the one underlying delay memory. This delay memory is also used by the WaveBank, HarMan, and ModMan.

The **SummedDelay** provides a single multi-tap audio delay buffer shared by all voices. SummedDelay is a summed aggregate of all voices in one shared delay. The SummedDelay is available in Bank C.

The Delay popup menu has a choice of maximum delay times for the SummedDelay.

The sample rate of the SummedDelay memory is limited by the maximum delay selected in the Delay menu. The **Anti-Aliasing Low Pass Filter** indicator above the first Bank C column will light up when the delay memory is at reduced sample rate. If you do not want an Anti-Aliasing filter at reduced sample rates, click on the indicator to disable the filter.

Note: When a WaveBank or Manipulator with Spectral Set is in use (in Bank A or Bank B), the SummedDelay will still operate, using half the normal amount of delay memory.

Examples of SummedDelay usage in EaganMatrix System Presets:

Left Hand Filter Echo: Creates echo control for the left hand, while the note is produced with the right hand.

Voyager: Creates a pulse delay by modifying the recirculator output level with a Shape Generator.

6.14. Multipurpose Banks: MicroDelay

Multipurpose Banks: MicroDelay (Four Short Audio Delay Buffers for Each Voice)			
MicroDelay	▲	Audio Input 1	Audio input into MicroDelay's first single-tap delay buffer. If the MicroDelay is operating at reduced sample rate, the audio inputs pass through an optional anti-aliasing filter.
	T1	Tap 1	Delay time for MicroDelay's first delay buffer. 0 = no delay, 1 = delay by max value (selected in the Delay menu). The Tap 1 output is available as an alternative on the Sub matrix rows. The sum of Taps 1..4 are available on the Bank C matrix row.
	▲	Audio Inputs 2-4	Three additional audio inputs like Audio Input 1.
	T2-T4	Taps 2-4	One delay time for each audio input 2 through 4. 0 = no delay, 1 = delay by max value. The Tap 2..4 outputs are available as a alternative on the Sub and AesIn matrix rows.

Notes for MicroDelay:

VoiceDelay, SummedDelay, MicroDelay, and FormulaDelay are four different ways to use the one underlying delay memory. This delay memory is also used by the WaveBank, HarMan, and ModMan.

The **MicroDelay** provides four single-tap audio delay buffers for each voice. The MicroDelay is available in Bank C. The outputs of MicroDelay's four delay buffers are available on the Tap 1..Tap4 matrix rows (alternative choices for the Sub and AesIn matrix rows). The sum of all four delay buffers outputs (Tap 1 + Tap 2 + Tap 3 + Tap 4) is available on the Bank C matrix row.

The Delay popup menu has a choice of maximum delay times for the MicroDelay.

The sample rate of the delay memory is limited by two variables: maximum delay and polyphony. The **Anti-Aliasing Low Pass Filter** indicator above the first Bank C column will light up when the delay memory is at reduced sample rate. If you do not want an Anti-Aliasing filter at reduced sample rates, click on the indicator to disable the filter.

Note: When a WaveBank or Manipulator with Spectral Set is in use (in Bank A or Bank B), the MicroDelay will still operate, using half the normal amount of delay memory.

Examples of MicroDelay usage in EaganMatrix System Presets:

SoundBoard: Uses four audio delay buffers with relatively prime delay lengths, coupled sequentially.

6.15. Multipurpose Banks: FormulaDelay

Multipurpose Banks: FormulaDelay (Delay Buffers for Control Information, for use in Formulas)			
FormulaDelay	B1	Buffer 1 Input	Input to FormulaDelay's first delay buffer. This buffer's output is available by selecting Buffer 1 in the W component of a formula.
	B2	Buffer 2 Input	Input to FormulaDelay's second delay buffer. This buffer's output is available by selecting Buffer 2 in the W component of a formula.
	D	Delay for Buffers 1 & 2	Delay time for FormulaDelay's delay buffers 1 & 2. 0 = no delay, 1 = max delay (selected in the Delay menu).
	H	Hold for Buffers 1 & 2	H ≤ 0 is normal operation. H > 0 stops recording new data into buffers 1 & 2. Transition from H ≥ 0 to H < 0 clears buffers 1 & 2.
	B3	Buffer 3 Input	Input to FormulaDelay's third delay buffer. This buffer's output is available by selecting Buffer 3 in the W component of a formula.
	B4	Buffer 4 Input	Input to FormulaDelay's fourth delay buffer. This buffer's output is available by selecting Buffer 4 in the W component of a formula.
	D	Delay for Buffers 3 & 4	Delay time for FormulaDelay's delay buffers 3 & 4. 0 = no delay, 1 = max delay (selected in the Delay menu).
	H	Hold for Buffers 3 & 4	H ≤ 0 is normal operation. H > 0 stops recording new data into buffers 3 & 4. Transition from H ≥ 0 to H < 0 clears buffers 3 & 4.

Notes for FormulaDelay:

VoiceDelay, SummedDelay, MicroDelay, and FormulaDelay are four different ways to use the one underlying delay memory. This delay memory is also used by the WaveBank, HarMan, and ModMan.

The **FormulaDelay** provides four delay buffers for each voice, for delaying control data used in Matrix formulas. Typical values to feed through a FormulaDelay might be X, Y, Z, or a pedal. The FormulaDelay is available in Bank C. No output from FormulaDelay appears directly on matrix rows; instead, the delay buffer outputs are available by selecting Buffer 1..4 in the W component of formulas.

The Delay popup menu has a choice of maximum delay times for the FormulaDelay.

The sample rate of FormulaDelay's delay memory is normally at the formula evaluation rate (1 kHz), but it can be limited by two variables: maximum delay and polyphony. Very long delays or high polyphony may result in reduced sample rate for the delay memory. Consider filtering or slewing the data inputs to FormulaDelay when using long delay times in the Delay popup menu, to avoid aliasing the control signals being delayed through FormulaDelay.

Note: When a WaveBank or Manipulator with Spectral Set is in use (in Bank A or Bank B), the FormulaDelay will still operate, using half the normal amount of delay memory.

6.16. Shape Generators

SGs (Shape Generators)			
1	F	Frequency	Frequency (in Hz) of the SG. This value should be a sub-audio rate < 30 Hz.
	P	Phase	For "Phase from Amplitude", this column controls the phase.
	T	Trigger	Triggers SG when T changes from T≤0 to T>0. Stops SG when T<0.
	F	Frequency	For "Dual", this controls the frequency (in Hz) of the secondary SG.
2 - 5			Four more SGs like SG 1.

Notes for Shape Generators:

Shape Generators are intended for use at sub-audio rates. Highest quality results are obtained with frequencies less than 30 Hz.

At the top of each SG column, you may choose Continuous Cycle, Single Cycle, Dual, or Phase from Amplitude.

Single Cycle and Continuous Cycle SGs are Low Frequency Oscillators (LFOs), with Frequency and Trigger input columns in the matrix:

- **Continuous Cycle** - the SG triggers on the positive edge of T (T≤0 to T>0), and continues as long as T≥0.
- **Single Cycle** - the SG triggers on the positive edge of T, and completes one cycle unless T<0 stops it.

Dual SGs are a pair of Low Frequency Oscillators (LFOs), with a pair of Frequency columns in the matrix:

- **Dual** - both SGs trigger on finger touch, and complete one cycle.

Phase from Amplitude SGs are Low Frequency Waveshapers (LFWs), with a Phase input column in the matrix:

- **Phase from Amplitude** - the SG's phase is supplied directly by the P column.
  (Technical detail: The SG phase is the RMS value of the P column computed each millisecond.)

It is possible to retrigger Continuous Cycle and Single Cycle SGs without stopping them first. For example, a Continuous Cycle SG with this sequence of T values retriggers without stopping: T=0,1,0,1. This sequence stops the SG before triggering it again: T=0,1,-1,1.

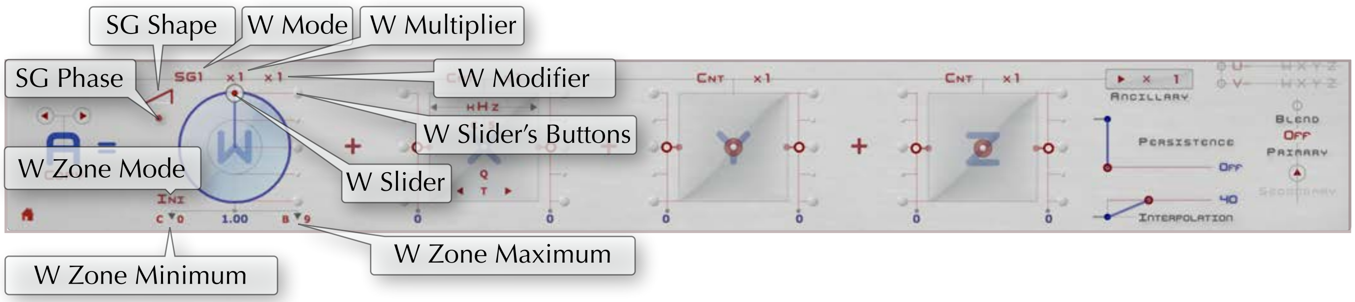
The Continuous Cycle and Single Cycle SG's F and T columns control the frequency and the triggering of each SG, but not the shape of the SG. Any SG shape can be selected by a formula that uses an SG; each formula using the same SG can select different SG shapes. Also, each formula may select different phase offsets and phase modulations for its SG, for quadrature operation and shape modulation.

Phase from Amplitude SGs can be used to implement an amplitude follower. Connect the audio-rate signal to be followed to the SG's P column. Then select this SG in a formula, select a Ramp SG shape, and adjust the formula's persistence and interpolation controls for smooth following.

The Thumbnails tell you which formulas use SGs. If a formula uses an SG, the W in the thumbnail is replaced by the SG number used.

7. EaganMatrix Formula Reference

7.1. W Component of the Formula

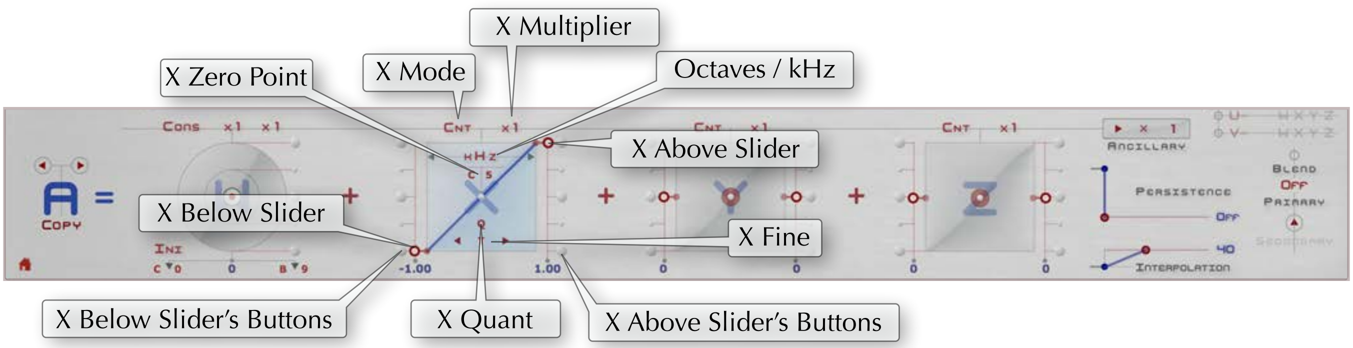


The math for the W component of EaganMatrix formulas is: **W Mode** × **W Slider** × **W Multiplier** × **W Modifier**.

Name	Function
W Slider	A value from -1 to 1.27.
W Slider’s Buttons	Five buttons allow you to quickly position the W Slider at -1, -0.5, 0, 0.5 or 1.
W Mode	If W Mode is Constant (abbreviated Cons), its value is always 1. If W Mode is Gated, its value is 1 as long as the finger is touching the surface, otherwise its value is 0. If W Mode is SG1..SG5, its value depends on the selected Shape Generator as well as SG Shape and SG Phase. (The triggering, rate, and repetition of the Shape Generator is determined by the Shape Generator’s matrix columns, but SG Shape and SG Phase is specific to this formula’s use of the Shape Generator.) If W Mode is FormulaDelay, its value depends on the delay tap selected in the W Tap control.
SG Shape	This control only appears if W Mode selects a Shape Generator. Possible SG shapes: increasing ramp (0..1), decreasing ramp (1..0), pulse (1 on, 0 off), pulse at end (0 off, 1 on), triangle (0..1..0), Hann (raised cosine 0..1..0), gentle s-curve up (0..1), steep s-curve up (0..1), gentle s-curve down (1..0), steep s-curve down (1..0), square (1 then -1), sine (min..max -1..1), SampleAndHold matrix row, SampleAndHold Formula A..V. The SampleAndHold are triggered when the Shape Generator initially triggers, and (if it is Continuous Cycle) at each repetition.
SG Phase	This control only appears if W Mode selects a Shape Generator. Select an optional phase offset for this formula’s use of the SG: .25 (90 degrees), .5 (180 degrees), .75 (270 degrees), or a value provided by another formula.
W Tap	This control only appears if W Mode is FormulaDelay. W Tap selects output 1..4 in Bank C’s FormulaDelay.
W Multiplier	Multiplier on the W component value; choices are .001, .01, .1, 1, 10, 100, 1000.
W Modifier	W Modifier optionally multiplies by barrel i, ii, iii, iv, Gen1, Gen2, or by voice number. W Modifier “If Voice 1” will equal 1 on voice 1, and 0 on all other voices. W Modifier “Polyphony Scaler” will scale by 1/p, where p is the DSP’s polyphony.
W Zone Min/Max	Minimum and Maximum finger pitch for W Zone. The W component’s value will be forced to 0 if the finger’s pitch is outside of the W Zone. To define a “dead zone”, set Minimum>Maximum; then W will be forced to 0 when Maximum≤pitch≤Minimum.

Name	Function
W Zone Mode	If W Zone Mode is set to Continuous, the finger pitch movements are checked against the W Zone as long as the finger is in contact with the surface. If W Zone Mode is set to Initial, the finger pitch is checked against the W Zone only once, when the finger first comes in contact with the surface.

7.2. X Component of the Formula



The details of the math for the X component are as follows:

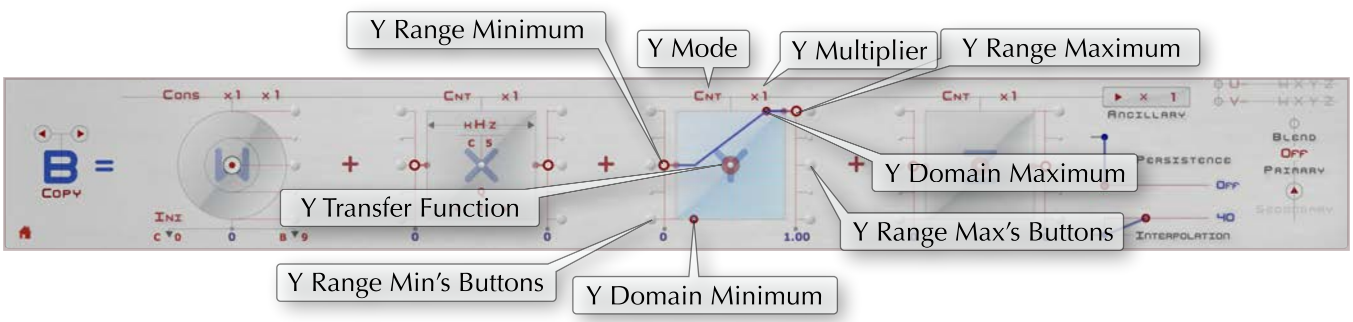
when Octaves is selected **(*quant*(nn - nn_0)) × (X Slider + X Fine) × X Multiplier) / 12.0**
when kHz is selected **_nnToKHz_(60.0 + _quant_(nn - nn_0)) × (X Slider + X Fine) × X Multiplier)**
where: nn = pitch of finger in note number units (for example 60.102 is 10.2 cents above middle C)
nn_0 = X Zero Point in Midi note number units
quant() = optional quantization to Q half steps
X Slider = X Above Slider when nn > nn_0, otherwise negative of X Below Slider
nnToKHz() = converts note number units (60.0 = middle C) to kHz (.26162557 = Middle C)

transfer function selected **_trans_(_xx_) × (X Above Slider - X Below Slider) + X Below Slider) × X Multiplier**
where: xx = x position of finger; transfer function extends from 3 octaves below Middle C to 3 above
trans() = X Transfer Function

Name	Function
X Zero Point	Position on the Continuum surface where the X component of the formula is zero. This is available when Octaves or kHz is selected.
X Below Slider	Change in the X component when the performer's finger is to the left of the X Zero Point. If an X Transfer function is used, this is the value 3 octaves below Middle C.
X Below Buttons	Five buttons allow you to quickly position the X Below Slider at -1, -0.5, 0, 0.5 or 1.
X Above Slider	Change in X component when the performer's finger is to the right of the X Zero Point. If an X Transfer function is used, this is the value 3 octaves above Middle C.
X Above Buttons	Five buttons allow you to quickly position the X Above Slider at -1, -0.5, 0, 0.5 or 1.
X Quant	X Quant allows you to quantize the X finger position to implement regularly-sized regions on the playing surface. X Quant is normally off. (Note: X Quant is not related to the Round Rate tuning mechanism.)

Name	Function
Octaves / kHz / Transfer Function	The X value may be converted to kHz by selecting "kHz". This is normally selected if the formula appears in a matrix column that controls frequency in kHz. If the formula appears in CVC columns, "Octaves" is used (together with appropriate X Slider values) to generate CVC control voltages in 1 or 1.2 volts-per-octave units. Transfer Functions are available for situations where the formula controls non-pitch synthesis parameters; the available functions are Linear, S, Squared, Square Root, 2 Step, and 3 Step. The twelve horizontal divisions in the transfer function graph correspond to tritones, and the centerline is Middle C; thus the transfer function domain is 3 octaves below middle C to 3 octaves above.
X Fine	Fine adjust on the X Slider values. This is normally 0, but can be used to slightly offset the X Slider values in .05 cent increments.
X Mode	If X Mode is set to Continuous (abbreviated Cnt), the finger X position is tracked and updated continuously as long as the finger is in contact with the surface. If X Mode is set to Initial, the X value is sampled and held when the finger first comes in contact with the surface. If X Mode is Relative, the X value will start at 0 and deviate from that value depending on the current X finger position relative to the starting X finger position. If X Mode is Derivative, the X value will the finger's X velocity (positive or negative) when the finger is moving, or 0 if the finger stops moving.
X Multiplier	Multiplier on X component value. The multiplier may be .001, .01, .1, 1, 10, 100, 1000.

7.3. Y Component of the Formula



The details of the math for the Y component are as follows:

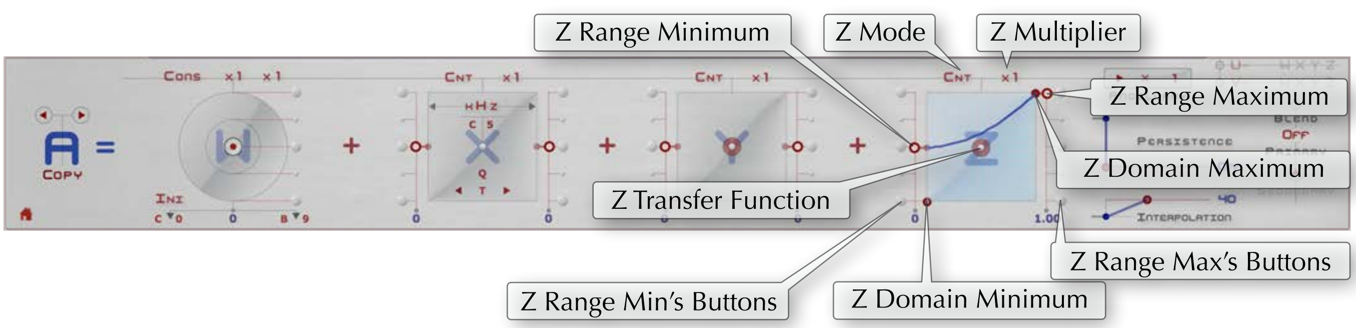
(*trans*(*domain*(yy)) × (Y Range Maximum - Y Range Minimum) + Y Range Minimum) × Y Multiplier

where: yy = front-to-back position of finger, 0.0 is front, 1.0 is back
domain() = scaling based on Y Domain Minimum and Y Domain Maximum
trans() = Y Transfer Function

Name	Function
Y Range Minimum	Specifies minimum value for Y transfer function.
Y Range Min's Buttons	Five buttons allow you to quickly position the Y Range Minimum at -1, -0.5, 0, 0.5 or 1.

Name	Function
Y Range Maximum	Specifies maximum value for Y transfer function.
Y Range Max's Buttons	Five buttons allow you to quickly position the Y Range Maximum at -1, -0.5, 0, 0.5 or 1.
Y Domain Minimum	Specifies the finger's Y value at which the Y transfer function will reach minimum. This value is quantized to twelve choices.
Y Domain Maximum	Specifies the finger's Y value at which the Y transfer function will reach maximum. This value is quantized to twelve choices.
Y Transfer Function	The Y Transfer Function affects the slope of the minimum to maximum Y value. The available slopes are Linear, S, Squared, Square Root, 2 Step, and 3 Step.
Y Mode	If Y Mode is set to Continuous (abbreviated Cnt), the finger Y position is tracked and updated continuously as long as the finger is in contact with the surface. If Y Mode is set to Initial, the Y value is sampled and held when the finger first comes in contact with the surface. If Y Mode is Relative, the Y value will start at 0 and deviate from that value depending on the current Y finger position relative to the starting Y finger position. If Y Mode is Derivative, the Y value will the finger's Y velocity (positive or negative) when the finger is moving, or 0 if the finger stops moving.
Y Multiplier	Multiplier on Y component value. The multiplier may be .001, .01, .1, 1, 10, 100, 1000.

7.4. Z Component of the Formula



The details of the math for the Z component are as follows:

(***trans***(***domain***(**zz**)) × **(Z Range Maximum - Z Range Minimum) + Z Range Minimum)** × **Z Multiplier**

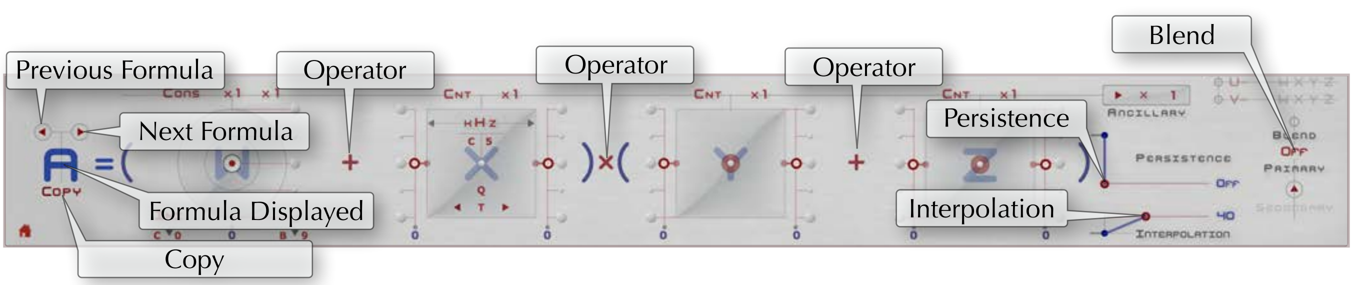
where: zz = pressure of finger, 0.0 to 1.0

domain() = scaling based on Z Domain Minimum and Z Domain Maximum

trans() = Z Transfer Function

Name	Function
Z Range Minimum	Specifies minimum value for Z transfer function.
Z Range Min's Buttons	Five buttons allow you to quickly position the Z Range Minimum at -1, -0.5, 0, 0.5 or 1.
Z Range Maximum	Specifies maximum value for Z transfer function.
Z Range Max's Buttons	Five buttons allow you to quickly position the Z Range Maximum at -1, -0.5, 0, 0.5 or 1.
Z Domain Minimum	Specifies the finger's Z value at which the Z transfer function will reach minimum. This value is quantized to twelve choices.
Z Domain Maximum	Specifies the finger's Z value at which the Z transfer function will reach maximum. This value is quantized to twelve choices.
Z Transfer Function	The Z Transfer Function affects the slope of the minimum to maximum Z value. The available slopes are Linear, S, Squared, Square Root, 2 Step, and 3 Step.
Z Mode	If Z Mode is set to Continuous (abbreviated Cnt), the finger Z position is tracked and updated continuously as long as the finger is in contact with the surface. If Z Mode is set to Initial, the Z value is sampled and held when the finger first comes in contact with the surface. If Z Mode is Derivative, the Z value will the finger's Z velocity (positive or negative) when the finger is moving, or 0 if the finger stops moving. If Z mode is Release Only, a release pulse is generated when the finger is lifted (select "finger lift") or when sustain/sostenuto is ended (select "sus/sost pedal end"). A formula can use Release Only combined with high persistence to create finger-lift noise.
Z Multiplier	Multiplier on Z component value. The multiplier may be .001, .01, .1, 1, 10, 100, 1000.

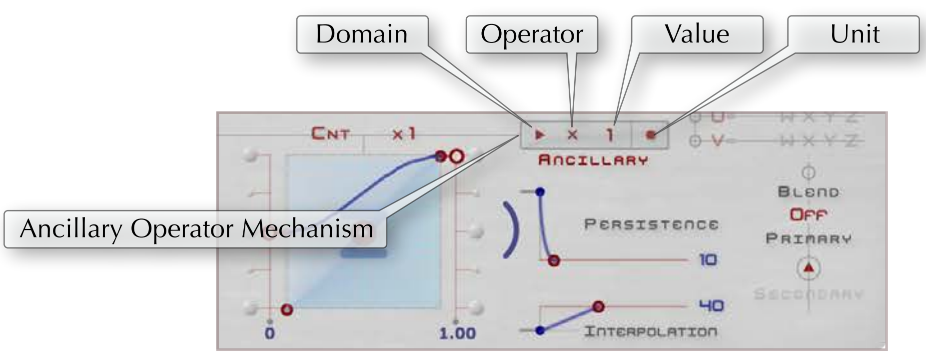
7.5. Other Parts of the Formula



Name	Function
Previous Formula	Display previous formula.

Name	Function
Next Formula	Display next formula.
Formula Displayed	Identifies the formula currently displayed. To choose a different formula, click on the matrix, on the thumbnails, or on the Previous/Next Formula selectors.
Show Value	A numeric readout of the formula's value is toggled by clicking the red Show Value circle. For Continuums with CEE, only the first DSP is plays notes when Show Value is active.
Copy	Copy another formula definition. This will replace the formula definition displayed with a copy of another formula definition. You can copy the user-defined formulas (A-V), pre-defined formulas (W-Z), or a constant.
Operator	Determines what math is used to combine the components of the formula. The most common choice is W + X + Y + Z. Other choices are W × (X + Y + Z), (W + X + Y) × Z, (W + X) × (Y + Z), and (W × X) + (Y × Z).
Persistence	Slows down decreases in formula value. This only affects decreases in formula value, not increases. If Persistence is off (0), the formula decreases are not slowed down. If the Persistence is large, the formula decreases slowly.
Interpolation	Smooths changes in formula value. Unlike Persistence, this affects both decreases and increases in formula value. If Interpolation is off (0), the formula values are not smoothed at all (stepwise evaluation), and in many cases noise or aliasing will result. Most commonly the Interpolation is set around 40, to give fast response without introducing stepping artifacts. If the Interpolation is large, the formula changes slowly.
Blend	Specifies a blend between primary and secondary values for the formula's W Slider, X Range Min/Max, Y Range Min/Max, Z Range Min/Max, Persistence, and Modulation. The Blend may be controlled by user-defined formulas A..V, by barrels i..iv, by gen1 or gen2, or by External Note (0 if internal from playing surface, 1 if Midi In using External Note Mode). If no blend is specified, only primary values are used. When a blend is specified, the secondary values may be viewed by clicking on the primary/secondary toggle.

7.6. Ancillary Operator Mechanism



The Ancillary Operator Mechanism consists of four menus:

- The Domain menu selects which components of the formula are affected (all components, or only W, X, Y, or Z). It also has options to evaluate the formula (on all voices) using Voice 1's WXYZ values, or using the Touche's WXYZ values.
- The Operator menu selects a mathematical operation.
- The Value menu selects a constant, formula, or the Midi clock (normalized to 1.0 for 120 quarters per minute).
- The Unit menu selects "No Unit Conversion" (no change to the formula's value), or conversion from note number (nn=60 for middle C) to kHz (0.2616 for Middle C).

If A stands for the formula being edited, and B stands for the constant/formula/Midi clock value, these are possible:

$A \times B$, $|A| \times B$, $(1 \div A) \times B$, $A+B$, A^B, $\log_B A$, A mod B, A quant B, A crosses B, min(A,B), max(A,B)

where: A mod B is the remainder of A divided by B.
A quant B is A-(A mod B), or "A quantized to B".
A crosses B is 1 if A>B when previously A<B, or A<B when previously A>B.

The Ancillary Operator Mechanism has no effect with "× 1" (multiply by constant 1), the default for a formula.

Via Limiter: Selecting "Via Limiter" will result in special processing for any matrix point where the formula appears: the formula×row value will be hard-clipped to -1/4..1/4.

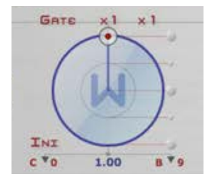
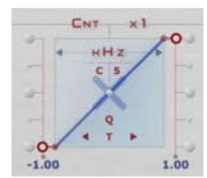
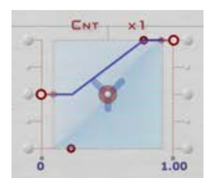
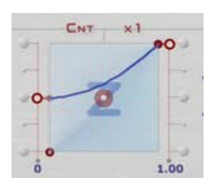
A special use for Via Limiter is in the first two matrix columns, which feed into the Master Section's saturation waveshaper. The limiter together with this waveshaper results in soft clipping, with a sine $-\pi/2..\pi/2$ transfer function. Without the limiter, very loud EaganMatrix output in the first two columns will be sinusoidally waveshaped, also a useful overload saturation effect.

Technical details about evaluating each column in a matrix: First, each formula in the column that has "Via Limiter" is multiplied by its matrix row; the sum of these is clipped to the range -1/4..1/4; then other formulas in the column are multiplied by their matrix rows; the sum of these is added to the Limiter result; finally that result is multiplied by the formula (if any) in the column's Multiply row. For example, if formulas f_1 and f_2 have "Via Limiter", f_3 and f_4 are other formulas in the column, and f_5 is in the column's Multiply row:

$$\text{columnValue} = (\text{clip}(f_1 \times \text{row}_1 + f_2 \times \text{row}_2) + f_3 \times \text{row}_3 + f_4 \times \text{row}_4) \times f_5$$

7.7. Pre-defined Formulas

There are four pre-defined formulas available in the EaganMatrix. Each is a simple way to quickly assign the window, X direction, Y direction, or Z direction of the playing surface in a useful way. When one of these formulas is selected, no formula information will appear below the matrix. These are hard-coded optimized formulas, and are not editable.

Predefined Formulas		
Formula	Function	User-defined Formula equivalent
W	Gated value of 0 when finger is off surface, 1 when finger is in contact with surface. This generates a standard trigger. It can, for instance, be used to trigger a Shape Generator to start at the beginning of a note.	
X	Standard pitch control from the X direction per octave (in kHz). Useful in the frequency column of the oscillators, filters, and multipurpose banks.	
Y	Shelved Y value from 0 (at the front) to 1 (at the back).	
Z	Pressure via a squared function from 0 to 1. Useful as the amplitude control of an audio source going into the master section.	

8. Sound Design with the EaganMatrix

In order to create an original EaganMatrix sound design, start with one of the following options:

- New (create an empty matrix): select Empty Matrix in the Internal Sound Popup Menu.
- Modify (existing preset sound): select one of the other EaganMatrix preset sounds in the EaganMatrix section of the Internal Sound Popup Menu, or open a Preset file from your Mac or PC. Preset files are available in the Archives folder within your Continuum Editor folder, or anywhere else you have previously saved Preset files.

Select a point in the matrix by clicking on the desired matrix point. When a matrix point is selected, it will be highlighted with a red outline.

8.1. Editing the Selected Matrix Point

To set the selected matrix point to a constant between 1 and 9, type the digit. To set the selected matrix point to a two digit constant between .01 and .99, type a decimal point, then two digits. To set the selected matrix point to a four digit constant between 0.0001 and 0.9999, type a decimal point then four digits.

To set the selected matrix point to a user-defined formula A..V, type the letter. The formula will be displayed below the matrix. To set the matrix point to a predefined formula W, X, Y, or Z, type the letter.

To delete (clear out) the selected matrix point, press Delete.

To unselect the the matrix point, press Escape, or select a different point in the matrix.

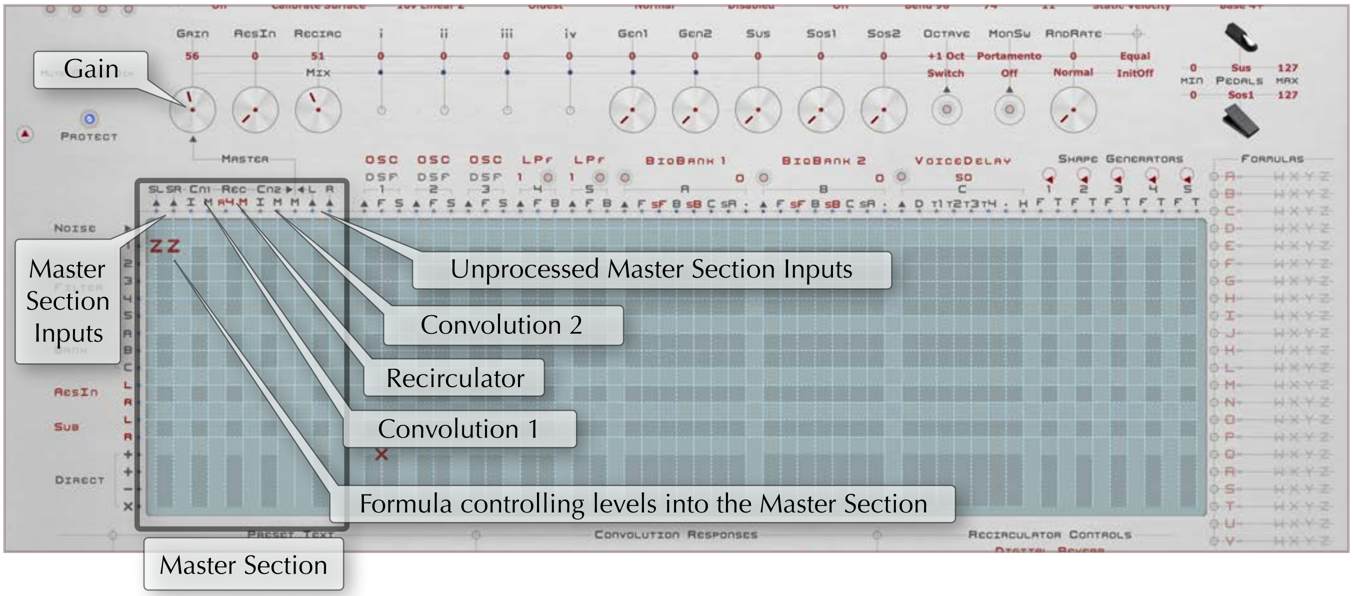
These are all the ways to edit the matrix:

Increase / Decrease Selected Point	Right Arrow / Left Arrow
Edit Selected Matrix Point	Type: a..v for user-defined formulas A..V, w..z for pre-defined formulas W..Z, 1..9 for constant values 1..9, decimal point then two digits for constants .01 .. .99, decimal point then four digits for constants .0001 .. .9999, @ or # for row-value-squared or row-value-cubed. Press Delete to remove the matrix point (disconnect row from column).
Move Selected Matrix Point	Drag to a new matrix location (Alt-Drag to copy)
Deselect Matrix Point and Show Description/Convolution/Recirculator	Escape, or click on Home symbol at bottom left.
Kenton Mini Controller	Assign the Tweak function to a Kenton Rotary (using the Kenton option in the Cogwheel menu) for convenient changes to the selected point.

8.2. Four-digit Constants in the Matrix

Four digit constants will be displayed in a shortened form within the matrix, due to display space constraints. Only the first digit followed by a + will be displayed. For example, ".5623" will show "5+". To see all four digits, click on the matrix point.

8.3. Master Section (Output Section of the EaganMatrix)



The output level of an EaganMatrix sound is influenced by two things. One is the setting of the Gain dial. The other is the value of formulae in the first two columns of the EaganMatrix. These SL and SR columns are inputs to the Master Section of the EaganMatrix. The Master Section features a built-in limiter and a saturator, a stereo convolution, recirculator, and a second (post-recirculator) convolution.

To have less saturation, select larger values with the Gain dial, and lower values in the SL and SR columns (formula J above). For higher saturation values, experiment with high values in the L and R columns (higher than 1), and lower Gain dial settings. Increased saturation will occur with more notes sounding at the same time, since the Master Section works on the stereo sum of all the voices.

8.4. "Reduce Gain" Message

If Overload Protect is active, extreme output levels will be suppressed and the Editor will display a message to "Reduce Gain". This detection and suppression is only done while the Editor is running, and only when the Overload Protect indicator is lit. The Overload Protect indicator is to the left of the Gain control. Click on the indicator to change its state.

8.5. "Reduce Polyphony" Message

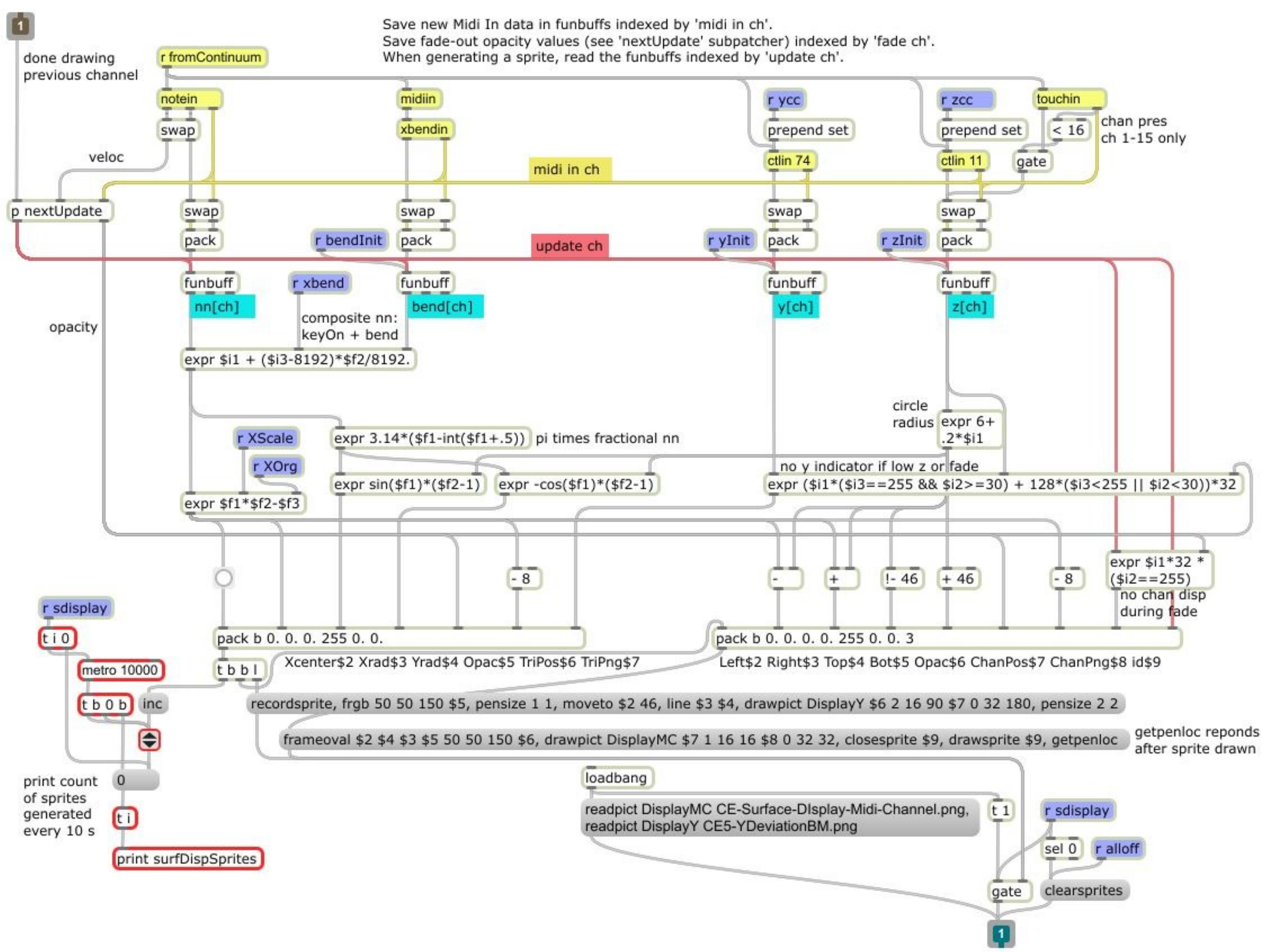
While the Continuum Editor is running, excessive polyphony values (resulting in excessive computations) will display a message to "Reduce Polyphony". This automatic detection may not catch borderline situations. If you notice sluggish behaviour - such as sluggish response to pedal changes as you play - try reducing the polyphony by 1, or try simplifying your design.

8.6. Save your Work

Your EaganMatrix designs can be saved to Preset files and opened at a later time (see the Open and Save topic in the Continuum User Guide). When you save your Preset, you will have the opportunity to name it. Save your work often, and remember to back up your computer's file system.

9. Need more information?

Feel free to contact Haken Audio with any questions concerning the EaganMatrix.



Part of Continuum Editor program, written in Cycling74 Max.