NOVATION CIRCUIT EDITOR

MANUAL

ShiftSynth1

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Forward:

This manual was developed to aid in the learning of the Novation Circuit and was made in conjunction with a YouTube video series to complement the information contained here. Please consider using this guide in addition to the educational series available to all online here:

https://www.youtube.com/watch?v=j7Uk2M0Q_GM&index=1&t=0s&list=PLmDbU1S2PJnr2F-H0QzLPB4CZ7eXXPDJB

Source Material

Circuit Editor Manual:
https://isotonikstudios.com/Manuals/Novation%20X%20Isotonik%20Circuit%20Editor%20User%20Guide%20V1.2.1.pdf

Mininova User Guide:
https://d2xhy469pqj8rc.cloudfront.net/sites/default/files/novation/downloads/4635/mininovauserguideen.pdf

MidiSynth Circuit Guide:
http://sigabort.co/Assets/doc/MIDISynth%20Circuit%20Tablet%20User%20Guide.pdf

Setting up:

Midi CC data setup on Circuit: Pressing the power button while holding down shift during power up of the Circuit will result in the bottom two rows lit as follows:

In order to ensure midi control back and forth from the editor to the Circuit, it is best to have all of the settings turned on with the lights on. If you have used the Circuit as a sequencer for external gear, it is likely that you have turned off the center four yellow and red buttons to stop the transmission of Midi CC data to your devices. If this is the case, the Circuit will not interact with the Circuit editor as intended.

Circuit Editor: The Circuit editor consists of 2 windows, the Window Settings and Main Window.

Circuit Editor Window Settings:
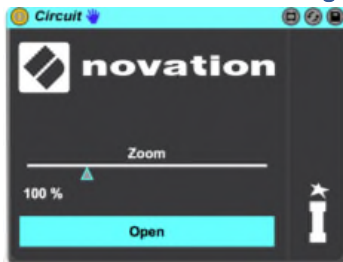
Zoom: Size of the main editor window

Open: Toggles the main editor window on or off

Circuit Editor Main Window:

Midi Port: Choose between Circuits if more than 1 Circuit is connected, when blue the Circuit is synched to the editor

Synth One/ Synth Two: The current Circuit synth being edited, when a synth is selected it becomes highlighted

View Control: Dictates the presentation on screen

> **Skin Color:** Selecting the circle to the right of the "View Control" Title will allow you to change the skin color of the editor.

> **Full:** This will show all macro controls, synthesizer parameters, and mod matrix settings

> **Synth:** This will show only the synth settings

> **Macros:** This will only show the macro settings

> **Mod Matrix:** This will only show the Mod Matrix settings

> **Session:** This will show the session and drum settings

> **Snapshot:** This will temporarily save the current settings of the editor. Clicked once will save the parameters, clicked a second time will revert changes back to the settings at the time of the first click

> **Undo:** Undo the last change made

> **Redo:** Redo the last undo

> **Patch:** This will show the current patch information in the top right corner of the screen

> **Random:** This will show the randomization parameters in the top right corner of the screen

> **Morph:** This will show the morph parameters in the top right corner of the screen
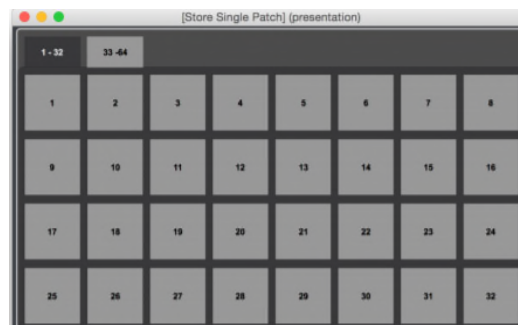

Circuit Controls: Interactive buttons with the Circuit

> **Get Current Patch from Circuit:** This will pull the current patch playing on the Circuit at the moment. This will not pull the patch into the library.

Upload Single Patch to Circuit: This will pull up a menu of 64 patches corresponding to the 64 patches in the Circuit, select a patch slot to import to.

Upload All Patches to Circuit: Uploads all patches to the Circuit that are currently loaded in the sample bank under the Library control.

Library Control: It is important to note that the Library is not a reflection of what is on the Circuit. It is the library of the Circuit editor. This can be uploaded to the Circuit but is not the same as the Circuit's library unless you have curated them to match. If you store a patch to the library it will not automatically be reflected on the Circuit, you must import the files in order to access them on board the Circuit.

Extremely important note! You can save a patch being edited on the Circuit editor to your current Circuit session at any time. However, this is not the same as the patch being loaded into the Circuit. This becomes evident by going to the on board Circuit bank, selecting a new patch, and then selecting the original patch slot. If you saved the patch to the session this will result in a different sound being played. If it was saved to the bank then the synth will revert back to the original sound. Please experiment and understand this because it can cause serious confusion and loss of work.

Load Bank from Disc: This will Load a Bank .syx file to the Circuit editor (Shift + Load Bank from Disc will allow you to load multiple single patches to the library)

Load Sing Patch from Disc: This will load a single .syx file to the current selected library slot

Store Patch to: Stores current patch being edited to a selected library slot. When selected it will ask to "Choose Patch in Menu Below," where a drop down menu is presented.

Save Bank to Disc: Saves the bank to your disc (Shift + Save Bank to Disc will allow you to load multiple single patches to your drive)

Save Single Patch to Disc: Saves a single patch

Store Patch: Stores current patch being edited to the current library slot

Name: Current Patch Name

Genre: Genre type for organization

Category: Instrument type for organization

Initialize Current Patch: Returns all patch settings to an initialized state. This is typically used when starting a new patch from scratch.

Oscillators:

2 Oscillators are provided, toggling between the two Oscillators is performed by selecting the tabs above the parameter values. (In the Pro editor edition these two tabs are broken out in the window)

Wave: This selects the oscillator's waveform

Wave Interpol: This parameter sets how smooth the transition is between adjacent waveforms in the same wavetable. A value of 127 will create a very smooth transition, with the adjacent waveforms blending together. With a value of zero the transitions will be abrupt and obvious. With a high value set, it is possible to retain a mix of adjacent waveforms if the modulation value remains fixed. When modulating the wavetable Index (via LFO, etc.), the wavetable interpolation parameter sets how smooth (or not!) the transition is.

PulseW Index: This control has two functions, depending on the waveform selected. With pulse waveforms, it varies the pulse width of the oscillator output. This basic effect can most easily be heard by adjusting this parameter with the Wave set to PW; you will note how the harmonic content varies and at high settings the sound becomes quite thin and metallic. A pulse wave is essentially an asymmetric square wave; when set to zero, the waveform is a normal square wave. This parameter has a different function if the oscillator waveform is set to be one of the 36 Wave Tables. Each Wave Table consists of nine related waveforms, and the setting of PW Index determines which is in use. The total parameter value range of 128 is divided into 9 (approximately) equal segments of 14 value units, so setting the value to anything between -64 and -50 will generate the first of the 9 waveforms, -49 to -35 the second, and so on. See also the Wave Table Interpolation parameter, which can be used to introduce further variation in the way wavetables are used.

Vsynch Depth: Oscillator Sync is a technique of using an additional "virtual" oscillator to add harmonics to the first, by using the virtual oscillator's waveform to retrigger that of the first. This technique produces an interesting range of sonic effects. The nature of the resulting sound varies as the parameter value is altered because the virtual oscillator frequency increases as a multiple of the main oscillator frequency as the parameter value increases. When the Vsync value is a multiple of 16, the virtual oscillator frequency is a musical harmonic of the main oscillator frequency. The overall effect is a transposition of the oscillator that moves up the harmonic series, with values in between multiples of 16 producing more discordant effects.
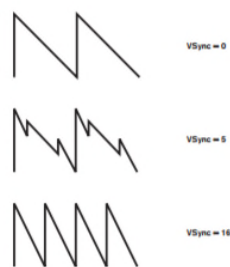
Density: The density parameter effectively adds copies of the oscillator waveform to itself. Up to eight additional virtual oscillators are used for this, depending on the value of the parameter. This produces a "thicker" sound at low to medium values, but if the virtual oscillators are detuned slightly (see Density Detune below), a more interesting effect is obtained.

Density Detune: This parameter should be used in conjunction with the Density control. It detunes the virtual density oscillators, and you will notice not only a thicker sound, but the effect of beating as well.

Semi Tone: This parameter sets the basic per-oscillator tuning. Incrementing its value by 1 shifts the pitch of every note on the keyboard up by one semitone for the selected oscillator only, thus setting it to +12 effectively shifts the oscillator tuning up one octave. Negative values detune in the same manner.

Cents: This parameter lets you make finer adjustments to the tuning. The increments are cents (1/100 of a semitone), and thus setting the value to ±50 tunes the oscillator to a quartertone midway between two semitones.

Pitch Bend: (Note: This is not applicable to the Circuit at this time) The pitch wheel can vary the oscillator pitch by up to an octave, up or down. The units are in semitones, so with a value of +12, moving the pitchwheel up increases the pitch of the notes being played by one octave, and moving it down takes them down an octave. Setting the parameter to a negative value has the effect of reversing the operating sense of the pitch wheel.

LFO:

2 LFOs are provided for modulation, toggling between the two LFOs is performed by selecting the tabs above the parameter values. (In the Pro editor edition these two tabs are broken out in the window)

Wave: The Circuit's LFOs are able to generate not only the familiar sine, sawtooth, triangle and square waveforms for modulation purposes, but are also able to produce a wide range of preset sequences of various lengths and random waveforms. A common use of an LFO is to modulate the main oscillator(s), and with many of the sequenced waveforms, setting the Depth parameter in the Modulation Matrix Menu to either 30 or 36 will ensure that the resulting oscillator pitches will be musically associated in some way.

LFO Waveforms Table

DISPLAY	WAVEFORM	EXTRA INFO
Sine	Traditional LFO shapes	
Triangle		
Sawtooth		
Square		
Rand S/H		Jumps to random values every cycle of the LFO
Time S/H		Jumps to min and max value each held for a random amount of time
PianoEnv		A curved sawtooth shape
Seq 1	These are sequences that jump to different values, holding each for a sixteenth of the LFO cycle rate.	
Seq 2		
Seq 3		
Seq 4		
Seq 5		
Seq 6		
Seq 7		
Altern 1	These are sequences that jump between a minimum and a maximum value, each value held for a varying interval of time.	
Altern 2		
Altern 3		
Altern 4		
Altern 5		
Altern 6		
Altern 7		
Altern 8		
Chromat	These are "melodic" sequences of various kinds. When modulating oscillator pitch, to obtain chromatic results, set Modulation Depth to either ±30 or ±36.	
Major		
Major 7		
Minor 7		
MinArp 1		
MinArp 2		
Diminish		
DecMinor		
Minor3rd		
Pedal		
4ths		
4ths x12		
1625 Maj		
1625 Min		
2511		

Phase Offset: This control is only active if Key Sync is set On. It determines the start point of the LFO waveform when the key is pressed. A complete waveform has 360º, and the control's increments are in 3º steps. Thus a half-way setting (180º) will cause the modulating waveform to start at half-way through its cycle
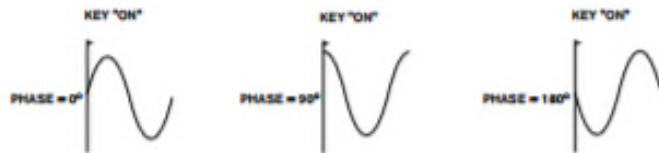
Slew Rate: Slew has the effect of modifying the shape of the LFO waveform. Sharp edges become less sharp as Slew is increased. The effect of this can be easily observed by selecting Square as the LFO waveform and setting the rate fairly low so that the output when a key is pressed alternates between just two tones. Increasing the value of Slew will cause the transition between the two tones to become a "glide" rather than a sharp change. This is caused by the vertical edges of the square LFO waveform being slewed.
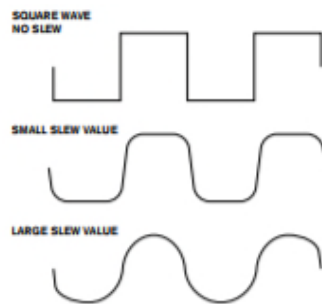
Delay: LFO Delay is a time parameter whose function is determined by Fade Mode (see below).

Delay Sync: When this parameter is set to off, the LFO delay is controlled by the Delay parameter. At all other settings Delay becomes inoperative, and the LFO delay is derived from the internal/external MIDI clock. You will notice that the Synch Value Table from the Mininova manual differs slightly from the Circuit. However, the logic is consistent and clear for 21+ beats which is where the two differ.

Sync Values Table

DISPLAY	DETAILS	CHORUS SYNC LFO RATE SYNC LFO DELAY SYNC PAN SYNC	ARP SYNC GATOR SYNC FX DELAY SYNC
32nd T	48 cycles per 1 bar	✔	✔
32nd	32 cycles per 1 bar	✔	✔
16th T	24 cycles per 1 bar	✔	✔
16th	16 cycles per 1 bar	✔	✔
8th T	12 cycles per 1 bar	✔	✔
16th D	8 cycles per 3 beats / 32 cycles per 3 bars	✔	✔
8th	8 cycles per 1 bar	✔	✔
4th T	6 cycles per 1 bar	✔	✔
8th D	4 cycles per 3 beats / 16 cycles per 3 bars	✔	✔
4th	4 cycles per 1 bar	✔	✔
1 + 1/3	3 cycles per 1 bar	✔	✔
4th D	2 cycles per 3 beats / 8 cycles per 3 bars	✔	✔
2nd	2 cycles per 1 bar	✔	✔
2 + 2/3	3 cycles per 2 bars	✔	✔
3 Beats	1 cycle per 3 beats / 4 cycles per 3 bars	✔	✔
4 Beats	1 cycles per 1 bar	✔	✔
5 + 1/3	3 cycles per 2 bars	✔	✔
6 Beats	1 cycle per 6 beats / 2 cycles per 3 bars	✔	✔
8 Beats	1 cycle per 2 bars	✔	✔
10 + 2/3	3 cycles per 4 bars	✔	
12 Beats	1 cycle per 12 beats /1 cycle per 3 bars	✔	
13 + 1/3	3 cycles per 10 bars	✔	
16 Beats	1 cycle per 4 bars	✔	
18 Beats	1 cycle per 18 beats /2 cycles per 9 bars	✔	
18 + 2/3	3 cycles per 8 bars	✔	
20 Beats	1 cycle per 5 bars	✔	
21 + 1/3	3 cycles per 16 bars	✔	
24 Beats	1 cycle per 6 bars	✔	
28 Beats	1 cycle per 7 bars	✔	
30 Beats	2 cycles per 15 bars	✔	
32 Beats	1 cycle per 8 bars	✔	
36 Beats	1 cycle per 9 bars	✔	
42 Beats	2 cycles per 21 bars	✔	
48 Beats	1 cycle per 12 bars	✔	
64 Beats	1 cycle per 16 bars	✔	

Display list (right side):
✓ off
32nd Triplets
32nd
16th Triplets
16th
8th Triad
16th Dotted
8th
4th Triad
8th Dotted
4th
1 + 1/3 beats
4th Dotted
2nd
2 + 2/3 beats
3 beats
4 beats
5 + 1/3
6 beats
8 beats
21
22
23
24
25
26
27
28
29
30
31
32
33
34
35
36

Rate: Rate is the LFO's frequency. A value of zero stops the LFO, and most musical effects are likely to use values in the 40~70 range, though higher or lower values may be appropriate for certain sound effects.

LFO RATE = X   LFO RATE = 0
OFFSET
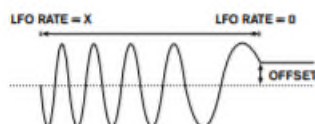
Rate Sync: This control allows the frequency of the LFO to be synchronized to an internal/external MIDI clock. When set to Off, the LFOs run at a frequency set by the Rate parameter. At all other settings Rate becomes inoperative, and the LFO rate is determined by Rate Sync, which in turn is derived from the MIDI clock. The sync table is the same as the Delay Sync, see above for the reference table.

Key Sync: Each LFO runs continuously, 'in the background'. If the Key Sync setting is Off, there is no way of predicting where the waveform will be when a key is pressed. Consecutive presses of a key will inevitably produce varying results. Setting Key Sync to On re-starts the LFO at the same point on the waveform every time a key is pressed. The actual point is set by the Phase Offset parameter.
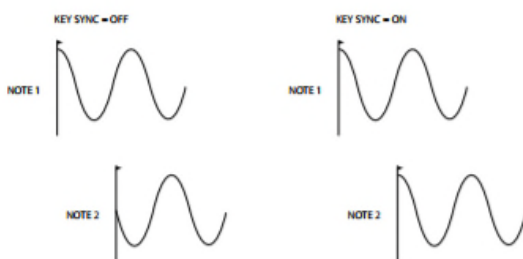
Common Sync: When LFOs are in use for pitch modulation (their most common application), Common Sync is only applicable to polyphonic voices. It ensures that the phase of the LFO waveform is synchronized for every note being played. When to set Off, there is no such synchronization, and playing a second note while one is already pressed will result in an unsynchronized sound as the modulations will be out of time.
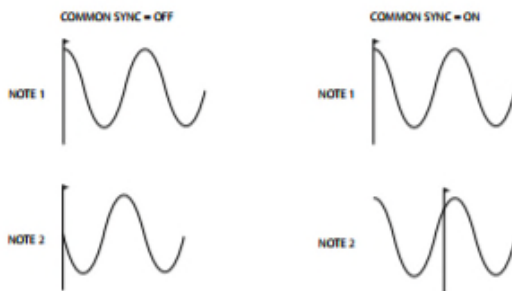
One-Shot: As its name suggests, setting this parameter to On causes the LFO to generate just a single cycle of its waveform. Note that a full waveform cycle is always generated regardless of the setting of LFO Phase; if LFO Phase is set to 90°, the one-shot waveform will start at the 90° point, execute a full cycle, and end at 90°

Delay Trigger: This parameter works in conjunction with the Fade/Gate parameters set by Fade Mode. In Re-Trig mode, every note played has its own delay time, as set by Delay (or MIDI clock if Delay Sync is active). In Legato mode, it is only the first note of a legato-style passage which triggers the delay – i.e., second and subsequent notes do not re-trigger the Delay function. For the Legato setting of Delay Trigger to be operative, mono voicing must be selected – it will not work with polyphonic voicing.

Fade Mode: The function of the four possible settings of Fade Mode are as follows:

Fade In: The LFO's modulation is gradually increased over the time period set by the Delay parameter.

Fade Out: The LFO's modulation is gradually decreased over the time period set by the Delay parameter, leaving the note with no LFO modulation.

Gate In: The onset of the LFO's modulation is delayed by the time period set by Delay parameter, and then starts immediately at full level.

Gate Out: The note is fully modulated by the LFO for the time period set by the Delay. At this time, the modulation stops abruptly.

Mixer:

Osc1: This parameter sets the amount of Oscillator 1's signal present in the overall sound.

Osc2: This parameter sets the initial amount of Oscillator 2's signal present in the overall sound.

Ring Mod: In its simplest form, a Ring Modulator is a processing block with two inputs and one output, which effectively "multiplies" the two input signals together. Depending on the relative frequencies and harmonic content of the two inputs, the resulting output will contain a series of sum and difference frequencies as well as the fundamentals.
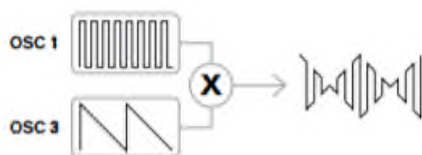
Noise Level: This parameter sets the amount of Noise present in the overall sound.

Pre-FX: The summed mixer inputs are routed through the FX block (even if no effects are active) at a level determined by Pre-FX level. This control should be adjusted with care to avoid overloading the FX processing.

Post-FX: This parameter adjusts the level returned from the output of the FX processor. Thus both Pre-FX level and Post-FX level will alter the signal level even when all FX slots in the FX block are bypassed.

Envelope:

The Circuit has 3 envelopes per synthesizer: A dedicated Amp Env, a dedicated Filter Env, and a Flexible 3rd Env for modulation. (In the Pro editor edition these two tabs are broken out in the window)
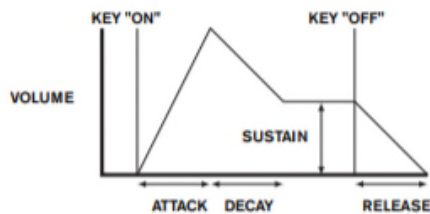
Amp Velocity: Amp Velocity does not modify the shape of the ADSR amplitude envelope in any way, but adds touch sensitivity to overall volume, so that with positive parameter values, the harder you play the keys, the louder will be the sound. With Amp Velocity set to zero, the volume is the same regardless of how the keys are played. The relationship between the velocity at which a note is played and volume is determined by the value. Note that negative values have the inverse effect.

Amp Attack: This parameter sets the note's attack time. With a value of 0 the note is at its maximum level immediately the key is pressed; with a value of 127, the note takes over 20 seconds to reach its maximum level. At the mid-setting (64), the time is approx. 250 ms.

Amp Decay: This parameter sets the notes' decay time. Decay time only has meaning if Amp Sustain (see below) is set to less than 127, as the Decay phase will be inaudible if the sustain level is the same as the level reached during the attack phase. At the mid-setting (64), the time is approx. 150 ms.

Amp Sustain: The value of the Sustain parameter sets the volume of the note after the completion of the decay phase. Setting a low value will obviously have the effect of emphasizing the start of the note; setting it to zero will render the note silent after the decay phase has elapsed.

Amp Release: Many sounds acquire some of their character from the notes remaining audible after the key is released; this "hanging" or "fade-out" effect, with the note gently dying away naturally (as with many real instruments), can be very effective. A setting of 64 gives a Release Time of approx. 360 ms. The Circuit has a maximum release time of over 20 seconds (with Amp Release set to 127), but shorter times will probably be more useful! Note that the relationship between the parameter value and Release Time is not linear

Filter Velocity: As Amplitude Velocity adds touch sensitivity to volume, so Filter Velocity can be set to make filter action touch sensitive. With positive parameter values, the harder you play the keys, the greater will be the effect of the filter. With Filter Velocity set to zero, the sound's characteristics are the same regardless of how the keys are played. Note that negative values have the inverse effect.

Filter Attack: This parameter sets how the filter section acts during the note's Attack phase. The higher the value, the longer it takes for the filter to react during this phase

Filter Decay: This parameter sets how the filter section acts during the note's Decay phase. Again, the higher the parameter value, the longer the period for which filtering is applied.

Filter Sustain: The frequency of the filter (cut-off or center, depending on filter type) "settles" at a value set by the Filter Sustain Level. Thus, once the Attack and Decay stages of the envelope are completed, the harmonic content that will be most evident in the sound will be determined by this parameter. Remember that if the filter frequency parameter (as set in the Filter Menu) is set at a too low or too high a value, the envelope's effect will be limited.

Filter Release: As Filter Release is increased in value, the note undergoes increasingly more filter action once the key is released.

Env 3: The assignment of Envelopes 3 to other synth parameters is performed in the Modulation Matrix Menu. In order to audition their effects, you must first open the Mod Matrix Menu and set a Mod Slot Source to Env3 and the Destination to a parameter of your choice.

Filter:

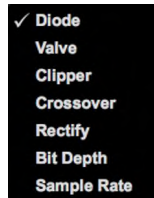**Routing:** Circuit provides three possible combinations of the filter block.

Normal: Both oscillators are routed through the filter

Osc 1 Bypass: Only oscillator 2 is routed through the filter, oscillator 1 bypasses it.

Osc 1 + Osc 2 Bypass: Both oscillators bypass the filter

Drive: The filter section includes a dedicated drive (or distortion) generator; this parameter adjusts the degree of distortion treatment applied to the signal. The basic 'type' of drive added is set by Filter Drive Type. The drive is added pre the filter.

Drive Type: Diode, Valve, Clipper, XOver, Rectify, BitsDown, RateDown The drive processor for each filter is located immediately before the filter section itself. The type of drive (or distortion) generated can be selected with the Filter Drive Type parameter.

Diode: Simulation of analogue Circuitry producing distortion where the waveform is progressively "squared-off" as the amount of distortion is increased.

Valve: Simulation of analogue Circuitry producing distortion similar to Diode, but at extreme settings alternate half-cycles of the waveform are inverted.

Clipper: Simulation of a digital overload.

XOver: Simulation of the crossover distortion generated by bi-polar analogue Circuitry, e.g., amplifier output stages.

Rectify: All negative-going half-cycles are inverted, simulating the effect of rectification.

BitsDown: Reproduces the "grainy" quality associated with lower bit rates, as found in older digital devices.

RateDown: Gives the effect of reduced definition and HF loss, similar to the use of a low sample rate.

Filter Type: The Circuit filter sections offer 6 different types of filter: two hi-pass and two lowpass, and two band-pass filters of various types. Each filter type differentiates between frequency bands in a different way, rejecting some frequencies and passing others, and thus each imposes a subtly different character on the sound.
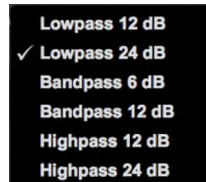
Filter Res: This parameter adds gain to the signal in a narrow band of frequencies around the frequency set by the filter frequency. It can accentuate the swept-filter effect considerably. Increasing the resonance parameter is very good for enhancing modulation of the cut-off frequency, creating a very edgy sound. Increasing Resonance also accentuates the action of the Filter Frequency parameter, so that as the filter parameter is moved, a more pronounced effect is obtained.

QNorm: This parameter alters the bandwidth of the peak created by the resonance control. The value of filter res has to be set to something other than zero for this parameter to have any effect. This feature enables the Filter section to emulate many of the filter responses found on various classic analogue and digital synths
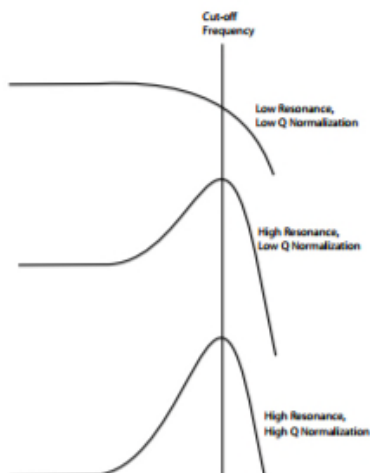
Env to Freq: The filter's action may be triggered by Envelope Generator 2. Envelope 2's own menu provides comprehensive control over precisely how this shape of the envelope is derived. Env2 lets you control the "depth" and "direction" of this external control; the higher the value, the greater the range of frequencies over which the filter will sweep. Positive and negative values make the filter sweep in opposite directions, but the audible result of this will be further modified by the filter type in use.

Voice Controls:

Portamento: With Portamento active, notes played sequentially glide from one to the next, rather than immediately jumping to the desired note pitch. The synth remembers the last note played and the glide will start from that note even after the key has been released. The Portamento parameter is the duration of the glide, and a value of 115 equates to approximately 1 second. Portamento is primarily intended for use in a mono Mode, where it is particularly effective. It can also be used in a Poly mode, but its operation can be unpredictable, particularly when chords are played. Note that Pre Glide must be set to zero in order for Portamento to be operative.

Pre Glide: Pre Glide takes priority over Portamento, though it does use the Portamento parameter to set its duration. Pre Glide is calibrated in semitones, and each note played will actually begin a on a chromatically-related note up to an octave above (value = +12) or below (value = -12) the note corresponding to the key pressed, and glide towards the 'target' note. This differs from Portamento in that, e.g., two notes played in sequence will each have their own Pre Glide, related to the notes played, and there will be no glide 'between' the notes.

Keyboard Octave: Sets the default octave the synth will load in at in the Circuit.

Polyphony Mode:
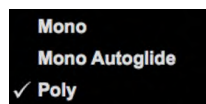
Mono: this is standard monophonic mode; only one note sounds at a time, and the "last played" rule applies
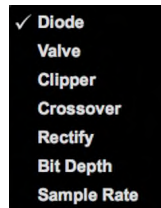
MonoAG: AG stands for Auto-Glide. This is an alternative mono mode, which differs from Mono in the way Portamento and Pre-Glide work. In Mono mode, Portamento and PreGlide apply both if notes are played separately, or in a legato style (when one note is played when another is already held down). In MonoAG mode, Portamento and Pre-Glide only work if the keys are played in a legato style; playing notes separately produces no glide effect.

Poly1: in this polyphonic mode successively playing the same note(s) uses separate voices and the notes are therefore 'stacked', so the sound gets louder as more notes are played. The effect will only be evident on patches with a long amplitude release time

Distortion:

Type: Select the type of distortion applied to the sound

Diode: Simulation of analogue Circuitry producing distortion where the waveform is progressively "squared-off" as the amount of distortion is increased.

Valve: Simulation of analogue Circuitry producing distortion similar to Diode, but at extreme settings alternate half-cycles of the waveform are inverted.

Clipper: Simulation of a digital overload.

XOver: Simulation of the crossover distortion generated by bi-polar analogue Circuitry, e.g., amplifier output stages.

Rectify: All negative-going half-cycles are inverted, simulating the effect of rectification.

BitsDown: Reproduces the "grainy" quality associated with lower bit rates, as found in older digital devices.

RateDown: Gives the effect of reduced definition and HF loss, similar to the use of a low sample rate.

Comp: Distortion Compensation only has an effect on Diode and Valve distortion types. Increasing compensation reduces the harshness of the distortion effect.

Level: Level of distortion applied to the sound

Chorus:

Type: Configures the FX processor as either a Chorus or Phaser

Rate Sync: Chorus Rate may synchronized to the internal or external MIDI clock, using a wide variety of tempos.

Sync Values Table

DISPLAY	DETAILS	CHORUS SYNC LFO RATE SYNC LFO DELAY SYNC PAN SYNC	ARP SYNC GATOR SYNC FX DELAY SYNC
32nd T	48 cycles per 1 bar	✓	✓
32nd	32 cycles per 1 bar	✓	✓
16th T	24 cycles per 1 bar	✓	✓
16th	16 cycles per 1 bar	✓	✓
8th T	12 cycles per 1 bar	✓	✓
16th D	8 cycles per 3 beats / 32 cycles per 3 bars	✓	✓
8th	8 cycles per 1 bar	✓	✓
4th T	6 cycles per 1 bar	✓	✓
8th D	4 cycles per 3 beats / 16 cycles per 3 bars	✓	✓
4th	4 cycles per 1 bar	✓	✓
1 + 1/3	3 cycles per 1 bar	✓	✓
4th D	2 cycles per 3 beats / 8 cycles per 3 bars	✓	✓
2nd	2 cycles per 1 bar	✓	✓
2 + 2/3	3 cycles per 2 bars	✓	✓
3 Beats	1 cycle per 3 beats / 4 cycles per 3 bars	✓	✓
4 Beats	1 cycles per 1 bar	✓	✓
5 + 1/3	3 cycles per 2 bars	✓	✓
6 Beats	1 cycle per 6 beats / 2 cycles per 3 bars	✓	✓
8 Beats	1 cycle per 2 bars	✓	✓

Rate: The Chorus Rate parameter controls the frequency of the Chorus processor's dedicated LFO. Lower values give a lower frequency, and hence a sound whose characteristic changes more gradually. A slow rate is generally more effective.

FeedBack: The Chorus processor has its own feedback path between output and input, and a certain amount of feedback will usually need to be applied to get an effective sound. Higher values will generally be needed when Phaser mode is selected. Negative values of Feedback mean that the signal being fed back is phase-reversed.

Mod Depth: The Depth parameter determines the amount of LFO modulation applied to the Chorus delay time, and thus the overall depth of the effect. A value of zero produces no effect

Delay: Chorus Delay is the actual delay that is used to generate the chorus/phaser effect. Dynamically altering this parameter will produce some interesting effects, though the difference in sound between different static settings is not marked, unless Chorus Feedback is at a high value. The overall effect of Chorus Delay is more pronounced in Phaser mode.

Level: The level of Chorus applied to the sound

EQ:

Low Freq: The equalizer is a "swept" type, which means that in addition to being able to increase or decrease the treble, mid or bass, you can also control the frequency band over which the Cut/Boost controls are effective – i.e., just what is meant by "bass", "mid" and "treble". This gives you much more accurate control over the frequency response. Increasing the value of Low Freq increases the frequency below which Low Gain is effective, so in general, Low Gain will have more effect on the sound the higher the value of Low Freq. Decreasing the value of Low Freq will lower the frequency below which the cut/boost control is effective with a value of 0 corresponding to approximately 140 Hz. The maximum value of 127 corresponds to about 880 Hz and the default value of 64 to about 500 Hz.

Low Gain: This parameter controls the LF response of the equalizer; a value of 0 gives a flat response in the LF region, positive values will give a rise in the LF response – i.e., more bass, and negative values will have the opposite effect. The range of adjustment is ±12 dB

Mid Freq: Increasing the value of this parameter increases the "center" frequency of the MF response. The center frequency is the one that gets the maximum amount of cut or boost as you adjust Mid Gain, and this control will have a proportionately decreasing effect on frequencies above and below the center frequency. The range of adjustment is from 440 Hz (value = 0) to 2.2 kHz (value = 127). The default value of 64 corresponds to approximately 1.2 kHz.

Mid Gain: This parameter controls the MF response of the equalizer; a value of 0 gives a flat response in the MF region, positive values will give a rise in the MF response – i.e., more midfrequencies (the voice region of the audio spectrum), and negative values will accordingly reduce the MF response. The range of adjustment is ±12 dB

High Freq: Decreasing the value of High Freq decreases the frequency above which High Gain is effective, so in general, High Gain will have more effect on the sound the lower the value of High Freq. Increasing the value of High Freq will raise the frequency above which the cut/ boost control is effective, with a value of 127 corresponding to about 4.4 kHz. A value of 0 corresponds to about 650 Hz and the default value of 64 to about 2 kHz.

High Gain: This parameter controls the HF response of the equalizer; a value of 0 gives a flat response in the HF region, positive values will give a rise in the HF response – i.e., more treble, and negative values less treble. The range of adjustment is ±12 dB

Placebo PRO: This does nothing.

Patch Store: Link to the Isotronics website

Macro Controls: The 8 Macro knobs correspond to the 8 macro control knobs on the Circuit. Each control knob can be programmed to change 4 parameters (1 parameter per tab in the macro section).

Position: The position knob is linked via midi CC data to the Circuit while the Circuit is connected. This means that if you turn the macro knob on the Circuit, the position will be reflected on the screen. The position circle has 3 position markers. The outer position marker illustrates the Start and Stop position, the inner position marker illustrates the Depth position, and the middle position market shows the current position of the macro knob.

Destination: This allows you to select the parameter that is changed when the macro knob is adjusted.

Copy: Allows the settings of the Macro to by pasted to other macro sections

Paste: Pastes the copied macro information

Paste to all: Pastes the copied macro information to all of the same macro positions in the loaded bank.

Start Position: The knob position in which the parameter being changed begins modulation

End Position: The knob position in which the parameter being changed stops modulation

Depth: The magnitude of the parameter change

Mod Matrix: The Mod Matrix contains 20 slots for modulating parameters in the synthesizer. In the free edition, these are 5 tabs with each containing 4 mod matrix options. In the Pro edition, all 20 are presented at the bottom of the synth editor. The Modulation Matrix is both variable and additive. By 'variable', it means that it is not just the routing of a controlling source to a controlled parameter which is defined in each slot, but also the "magnitude" of the control. Thus the 'amount' of control – or control 'range' – used is up to you. By 'additive' it means that a parameter may be varied by more than one source, if wished. Each slot allows for two sources to be routed to a parameter, and their effects are multiplied together. This means that if either of them is at zero, there will be no modulation. However, there is no reason why you can't have further slots routing these or other sources to the same parameter. In this case, the control signals from different slots "add" to produce the overall effect.

Mod Depth: The Depth control sets the level of the control being applied to the Destination – i.e., the parameter being modulated. If both Source1 and Source2 are active in the current slot, Depth controls their combined effect. Depth effectively defines the "amount" by which the controlled parameter varies when under modulation control. Think of it as the "range" of control. It also determines the "sense" or polarity of the control – positive Depth will increase the value of the controlled parameter and negative Depth will decrease it, for the same control input. Note that having defined source and destination in a patch, no modulation will occur until the Depth control is set to something other than zero.

Source 1/ 2: This selects a control source (modulator), which will be routed to the destination set by Destin. Setting both Source1 and Source2 to Direct means no modulation is defined.
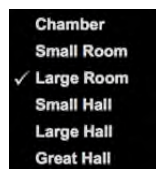
Destination: This sets which Circuit parameter is to be controlled by the selected source (or sources) in the current matrix configuration.

Session View:

Reverb:

Type Dropdown: Circuit provides six different reverb algorithms, designed to simulate the reflections which occur in rooms and halls of various sizes.

Synth One through Drum Four: Reflects the amount of reverb applied to each area of the Circuit

Decay: The Reverb Decay parameter sets the basic reverb time of the selected space. It can be thought of as setting the size of the room.

Damping: Applies a damping effect to reduce the echo level

Mixer: Controls the volume of synth 1 and synth 2 at the session level (this is not the same as the mixer in the editor main window, that mixer is applied to the individual synth setting)

Filter:

Freq: Modifies the Global Filter parameter

Res: Modifies the Global Resonance parameter


Synth1/2 SideChain:

Drum Dropdown: Select the sample track that ducks the synth track (typically this is the sample track that hosts the kick drum)

Attack: The rate at which the synth track is ducked out

Hold: The time between the attack drop out and the Decay ramp up

Decay: The rate at which the synth track is brought back to normal volume

Depth: The magnitude of the volume that is ducked
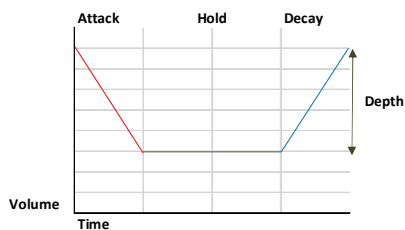
Delay:

Synth One through Drum Four: Reflects the amount of delay applied to each area of the Circuit

Time: This parameter sets the basic delay time. With Sync (see below) set to off, the note played will be repeated after a fixed time. Higher values correspond to a longer delay, with the maximum value of 127 equating to approximately 700 ms. If Delay Time is varied (either manually or via modulation) while a note is being played, pitch shifting will result. See also Delay Slew.

Sync: Delay time may synchronize to the internal or external MIDI clock, using a wide variety of tempo dividers/multipliers to produce delays from about 5 ms to 1 second.

F/back: The output of the delay processor is connected back to the input, at a reduced level; Feedback sets the level. This results in multiple echoes, as the delayed signal is further repeated. With feedback set to zero, no delayed signal at all is fed back, so only a single echo results. As you increase the value, you will hear more echoes for each note, though they still die away in volume. Setting the control in the center of its range (64) results in about 5 or 6 audible echoes; at the maximum setting, the repetitions will still be audible after a minute or more.

Width: The Delay Stereo Image Width parameter is only really relevant to settings of Delay Left-Right Ratio which result in the echoes being split across the stereo image. With its default value of 127, any stereo placement of delayed signals will be fully left and fully right. Decreasing the value of Dly1Wdth reduces the width of the stereo image and panned echoes are at an intermediate position between center and fully left or right.

Slew Rate: Delay Slew Rate only has an effect on the sound when the Delay Time is being modulated. Modulating delay time produces pitch-shifting. With DSP-generated delays, very fast changes of delay time are possible, but these may produce unwelcome effects, including digital glitching and clicks. Delay Slew Rate effectively slows the applied modulation down, so that any such glitches resulting from trying to change the delay time too rapidly may be avoided. The default value of off corresponds to the maximum rate of change, and the delay time will attempt to follow any modulation accurately. Higher values will produce a smoother effect.

L/R Ratio: The value of this parameter is a ratio, and determines how each delayed note is distributed between the left and right outputs. Setting Dly1L/R to the default 1/1 value places all echoes centrally in the stereo image. With other values, the larger number represents the delay time, and an echo will be produced at this time in one channel only, depending whether the larger number is to the left of the slash or the right. It will be accompanied by a faster echo in the other channel, at a time defined by the ratio of the two numbers. Values with OFF to one side of the slash result in all the echoes being in one channel only

```
  1:1
  4:3
  3:4
  3:2
✓ 2:3
  2:1
  1:2
  3:1
  1:3
  4:1
  1:4
  1:Off
  Off:1
```
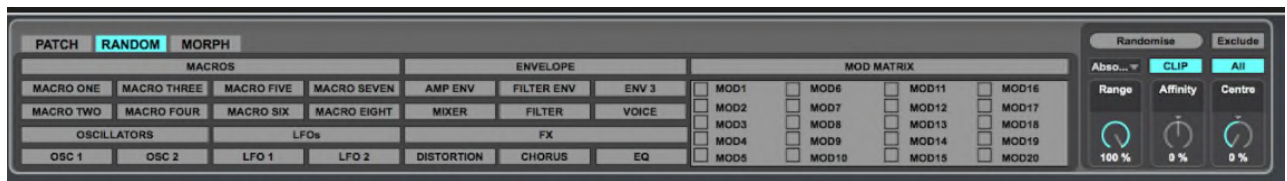
Random:

Randomize: This will execute the randomize function

Exclude: Allows individual controls to be excluded from the randomization

Dials Only/ All: Enabling this will disable all dropdowns from randomization

Algorithm Type:

Absolute: An absolute value will be set for the parameter between its min and max value.

Range: this determines the max upper value – a random value will be chosen between 0 and this %age of its max value

Affinity: whether the value tends to favor min (-ve) or max (+ve) values.

Center: It is unclear what this is, it is not included in any manuals and playing with the value has no meaningful impact on the randomization output.

Relative: This randomizer will generate random values based on the current values in the synth.

Range: the maximum movement as a %age of range (i.e. max – min values) that the value can move

Noise: how often (%age) the range is ignored to pick a random value. Negative values still obey the max of 50% of range, positive values set the range to be 100%

Clip: affects behavior when the new value would be outside of the min/max range. When clipping is on, they are fixed to min/max values. When disabled, they will "bounce back" from the min/max difference by the appropriate amount.

Direction: sets direction of value change to down, up, or both.

Patch: The new value will be picked between min and max values when the minimum is taken from the first patch specific and the maximum is taken from the second.

Patch Select 1: the first patch number to use

Affinity: whether the value tends to favor patch 1 (-ve) or patch 2 (+ve)

Patch Select 2: the second patch number to use

Noise: how often (%age) the affinity is ignored to pick a completely random value. Negative values still obey the bounds of min/max from patch 1 and 2, positive values will pick a value from min to max of the parameter.
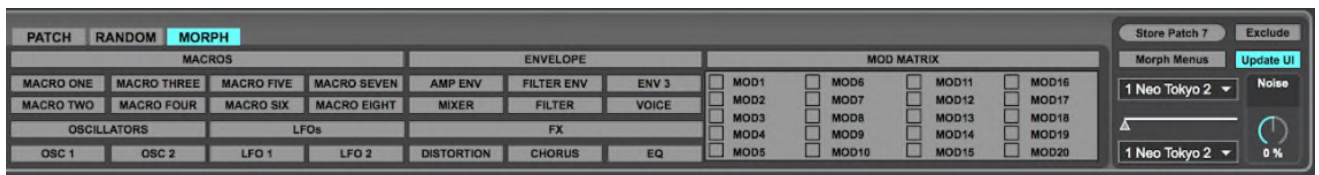
Morph: adjusts the patch settings from one patch to another in a gradual shift

Noise: How often (%age) the min/max values of the patches are ignored to pick a random value. Negative values still obey the bounds of min/max from patch 1 and 2, positive values will pick a value from min to max of the parameter.

Patch Select 1: The first patch number to use

Morph bar: Left is fully patch 1, right is fully patch 2

Patch Select 2: the second patch number to use

Morph Menus: enabling this will enable all dropdowns and parameters from morphing, if disabled only parameters will morph

Exclude: Allows individual controls to be excluded from the morph

Update UI: When turned off, morphing will occur when the morph bar is modulated but UI will not change. When on, the UI will change while sliding the morph bar.