



**NATIVE INSTRUMENTS**  
SOFTWARE SYNTHESIS

# Classic Instrument Guide

Operation Manual

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User's Guide written by: Brian Tester

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## **Germany**

Native Instruments GmbH  
Schlesische Str. 28  
D-10997 Berlin  
Germany  
[info@native-instruments.de](mailto:info@native-instruments.de)  
[www.native-instruments.de](http://www.native-instruments.de)

## **USA**

Native Instruments USA, Inc.  
5631 A Hollywood Boulevard  
Los Angeles, CA 90028  
USA  
[info@native-instruments.com](mailto:info@native-instruments.com)  
[www.native-instruments.com](http://www.native-instruments.com)

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# Classics Instrument Guide

## Synthesizers

### Carbon



*Instrument and Presets: Mike Daliot; Panel Design: Christophe Stoll*

The Carbon synthesizer is a fantastic-sounding machine with over a hundred sounds from a different planet. At its heart, Carbon is a powerful subtractive synthesizer with four oscillators, copious modulation, and a filter. We even wouldn't hesitate to say that Carbon has one of the best-sounding digital filters you've heard. In fact, Carbon has a choice of eleven different filter types and designs, each painstakingly shaped for silky smoothness across the entire frequency range. Carbon's LFOs include innovative features such as the possibility to vary their frequency slightly in each voice for an organic, living sound. Each one of Carbon's breathtaking pads, cutting leads, ripping basses, and quivering atmospheres will convince you that all life is based on Carbon.

## Quick Start

Even though Carbon's panel may show many controls, its clear layout and function grouping make operation easy. Simply step through some presets at random to get a feel for the type of sounds that Carbon can make. In the filter section, try modifying Cutoff and Resonance. You can even try choosing different filter types in the menu panel to the left of Cutoff to see how the different filters change the sound.

## Signal Flow and Structure

Carbon's sound is routed as follows:

Oscillators ⇒ Filter ⇒ Saturator ⇒ Four-band EQ ⇒ Chorus ⇒ Delay

Carbon features four oscillators plus a noise generator. The level of each oscillator is set with the large knob on top of each oscillator channel, while LFO modulation of the amplitude can be switched on or off directly underneath this knob. Each of Carbon's four oscillators offer two different modes, with each mode making a unique sound. Oscillator 1 is focused on pulse-width modulation, switchable between square or sinusoidal waveforms. Oscillator 2 concentrates on saw waves, switchable between a massive 7-saw (seven parallel saw oscillators, each slightly detuned), or a single sawtooth waveform. Oscillator 3 is based on interrupting the waveform - with a quantized sine wave being "interrupted" in amplitude to create low-fi effects, while a sawtooth oscillator is "interrupted" in time for the familiar sync effect. Oscillator 4 is mainly (but not always) used as a sub oscillator, offering standard pulse and sine waveforms with pulse width modulation. Below each oscillator mode selector, the tuning of each oscillator can be set in semitones and cents in the PitchShift section. The pitch of each oscillator can be modulated by either the second envelope (Env 2), or LFO 1. Oscillators 1 and 4 can have the width of their waveforms also modulated by Env 2 or LFO 1. The additional parameters in the oscillator section, such as "odet" (oscillator detune), "LCo," (loudness correction) etc., are used only for very fine shaping of the amplitude and range of modulation of the oscillators across the frequency spectrum. We warned you that Carbon was from another planet!

All four oscillators plus noise are summed and sent to Carbon's outstanding filter. The filter provides eight different filter modes where nearly every mode uses straight forward technology, mainly to emulate the warm and powerful sound of analogue filters. The modes Va, CLa and J/S offer an additional alternative mode which can be switched on by the Alt control below the main mode selector, increasing the effective number of modes to 11. Many of the filter types are very similar and offer a quick and easy way to modify the sound of each preset - simply select a preset at random, then choose a new filter for it. The main parameters of each filter mode - cutoff frequency, frequency bandwidth, resonance and feedback - can be set at the right of the mode selector. Below each parameter knob there are additional controls to modulate the respective value, including the possibility to map the MIDI velocity and the modulation wheel onto the filter's sound.

The sound comes out of the filter and goes right into a saturator and four-band EQ for even more fine-control over the sound's frequency spectrum. Besides a low shelf EQ, a high shelf EQ, and peak EQ with adjustable center frequency, there is also an additional highpass filter used to increase the sound's brightness.

## **Modulation**

Carbon has two envelopes and two LFOs. Envelope 1 always controls the amplitude, while Envelope 2 can control different parameters, such as oscillator volume, oscillator detune, filter cutoff, filter resonance, filter bandwidth, and filter feedback delay. There is no modulation matrix in Carbon - all modulations are set in their respective sections. For instance, to modulate filter cutoff by Envelope 2, simply look underneath the Cutoff knob in the filter section and you'll see a small knob for E2 and a button to activate the modulation amount. Carbon's LFOs offer valuable sound-shaping functions. Using the FSpr knob in either LFO section adds a frequency spread to the LFO frequency across multiple voices. This means that if you play a chord, each note would have slightly different LFO speeds for a shimmering, organic sound. Also, the modulation of the LFO can be delayed, for instance to program a vibrate which takes a few moments before coming in.

## Global Parameters

Parameters effecting the entire synth are set in the Global section, such as Mono/Poly (set with the Mono button), tuning, unison spread, and glide. Hold lets you specify how long each note will last for, no matter how long the actual MIDI note was held down for (value set in mSec underneath the Hold button). Old controls how much pitch randomness is introduced to emulate vintage hardware.

## Green Matrix



*Instruments: Erik Wiegand; Presets: Howard Scarr; Panel Design: Christophe Stoll*

Green Matrix is the MIDI-playable brother of Blue Matrix. Both use the same analog-style synthesis engine and modulation matrix to generate some amazing and challenging sounds. Its sound engine features two multiple-waveforms oscillators, a multi-mode filter, multi-mode distortion, and finally a second multi-mode filter. A beat-synced delay and a diffusion delay round out the effects. A complete modulation matrix lets you flexibly route the two envelopes, LFO, four channels of sequenced modulation, and the usual MIDI controllers to every important synthesis parameter.



## Quick Start

Plug in a MIDI keyboard and flip through the presets! Go ahead and experiment with the synthesizer controls - try changing LFO speed or cutoff frequency, for example.

## Structure and Signal Flow

Green Matrix's synth engine is based on traditional analog synthesis. It has three oscillators, two filters, distortion, and effects. Both Green Matrix and its sequencer-driven brother Blue Matrix are created from Reaktor 4's Classic Modular macro library which puts the sound quality and capabilities of the classic modular synths into your computer. For more information on building your own instruments with the Classic Modular macro library, please check your Reaktor 4 user's guide. The Classic Modular library is included on the Reaktor 4 CD.

Oscillators 1, 2, and 3 each offer a different waveform selection. Osc 1 offers the traditional sawtooth, pulse, triangle, sine, and impulse waveforms, while Osc 2 lets you choose between bipolar ramp, bipolar pulse, pulse, parabolic, and noise. The bipolar waveforms' shape can be modified in realtime with the small Shape knob. Osc 3

Osc 1 can be frequency modulated by Osc 2 or 3 for FM sounds (with the small FM knob and switch below), while Osc 2 and 3 can be ring modulated by Osc 1 for metallic and "gong" sounds (with the small Ring switch). Osc 1 can also be synced to Osc 2, with the choice between hard, soft, and MIDI gate-activated modes. Since Osc 1, 2, and 3 each offer unique features, they can be switched on or off to save CPU with the small on switch on their left sides. The base pitch and fine tuning of each oscillator is done with their respective Pitch and Fine controls.

The two oscillators feed into Filter 1 which offers a choice of 12 or 24 dB/octave lowpass, highpass, bandpass, peak EQ, and notch modes. Cutoff and Reson adjust the cutoff frequency and resonance, respectively, and the FM knobs controls how much Osc 1 or Osc 2 will frequency-modulate the cutoff frequency (settable with the small switch to the right of the FM knob). Pkey determines how the filter tracks the pitch of the notes from the sequencer. When Pkey is set to 1, then the filter cutoff will exactly follow the pitch of the sequencer notes, meaning that higher notes will be brighter, and lower notes will be duller. If set to

zero, then the cutoff frequency won't change. The other knobs, GainC, Limit, F foll, and Rel all control the fine-tuned shaping of the filter across the frequency and resonance spectrum. GainC controls how much loudness reduction will be applied at high resonance values. Limit, F foll, and Rel control a built-in resonance limiter to avoid unwanted harmonic thumps with high resonance values.

Filter 2 is similar in layout to Filter 1. There are a choice of four lowpass filters, two bandpass filters, and a combined bandpass/lowpass filter (BL4). In addition to the controls you're now familiar with from Filter 1, Filter 2 also offers the additional possibility to trigger self-resonance with a built in click. The Click knob controls how loud the click is, and the Vel knob determines how the click responds to velocity.

After Filter 2, the sound passes through a distortion unit then a chorus/delay effect. Note that an extended version of the chorus/delay effect is available in the Fusion Reflections ensemble. The Delay/Chorus consists of four stereo modulation delays. The main delay time is controlled by the Delay knob. Diffusion sets the inaccuracy of the delay times to "spread" the sound.

## Modulation

Green Matrix features three standard ADSR envelopes and a two LFOs that syncs to tempo. All three envelopes are velocity sensitive: Vel>A sets how much the velocity will affect attack time. The speed of the LFO can be set in musical note values, or freely. When Unit is set to 0, then the Frequency of the LFO is set in Hz. When Unit is set to any other value, the frequency of the LFO takes on the musical units set by Unit. For example, if Unit is 96 and frequency is 1, then the LFO will be set to the very fast speed of 1/96th note per cycle. The display underneath the Freq knob depends on how Unit is set: if Unit is set to 0, then the display automatically shows values in Hz, and if Unit is set to a musical value, then the Freq display will automatically update. Width and Phase control the shape of the LFO, which can be selected by the switch to the right. Key S determines if the LFO will synchronize to note-on events, as delivered by the sequencer.

All of Green Matrix's modulation sources are routed in a comprehensive modulation matrix: The LFO, two envelopes, four modulation channels from the sequencer, and typical MIDI controls like mod/pitchbend/

velocity can be routed to synthesizer destinations using the comprehensive modulation matrix. Switch to the "B" view of the synth to see the modulation matrix. You can easily switch to the B view by clicking the B button in the title bar of the synth. You can route the modulation to the following synth parameters: Oscillator pitch (osc 1, osc 2, or osc 1 + 2), osc 1 FM, osc 2 shape, osc ½ mix, filter cutoff (osc 1, osc 2, or osc 1 + 2), filter 1 FM, filter resonance (1 or 2), envelope parameters, and even distortion, delay time, and delay feedback. In short, nearly everything!

The modulation sources (LFOs, envelopes, etc) are arranged vertically. The modulation destinations (synth parameters such as filter cutoff, oscillator pitch, etc) are arranged vertically. Simply activate the switches for the combination you'd like to modulate. Underneath each modulation column is a switch to turn the modulation on or off, and a knob to set the modulation amount.

## Junatik



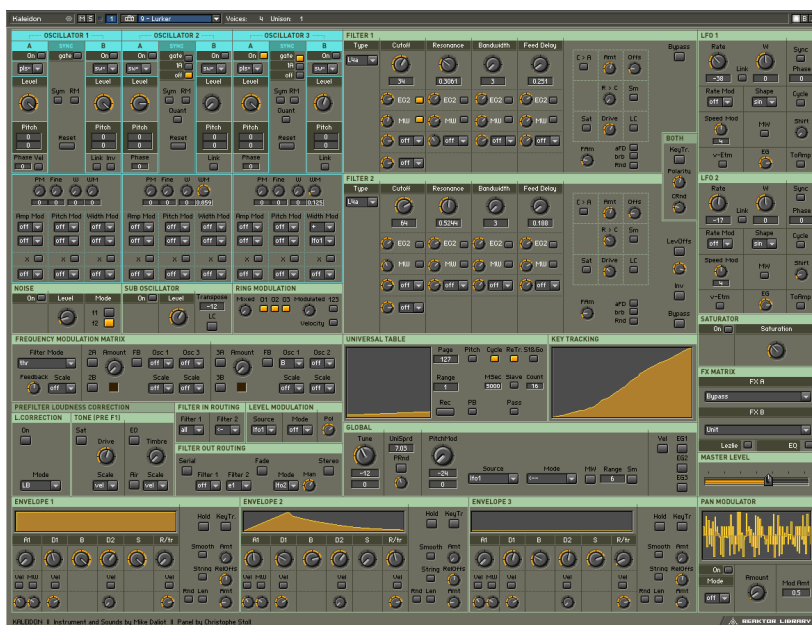
*Instrument: NI, Joerg Holzamer; Sounds: Easy Sounds, Joerg Holzamer, ear2ear, Uwe G. Hoenig; Demo: Joerg Holzamer*

1982 , about one year before the era of FM Synthesis began, polyphonic analogous synthesizers finally became (more) affordable. This was mainly due to a japanese company bringing an attractively priced synthesizer to market which offered six voices and was named after a

very important roman goddess. Although the first version of that synth had no memory locations, it convinced with its charismatic and full sound. In fact, the synthesizer was not complicated at all and sounds could be set up very easily and quickly, therefore the absence of memory locations (which were rather expensive at that time) wasn't very tragic. The fat sound of the synth did not come from its oscillator/sub oscillator combination, but rather from a good sounding lowpass filter and especially an integrated chorus effect. Unfortunately the chorus was very noisy, but it broadened the somehow sterile sound of the oscillator/sub oscillator significantly and made a big contribution to the popularity of the synth.

Junatik offers a surprisingly authentic recreation of the still very popular synth. The sympathically straight sound architecture of Junatik was modelled on the original and carefully supplemented with some important functions. Besides an authentic sounding filter, Junatik therefore offers an improved oscillator section. With its threefold and detunable saw wave it is now even capable of producing detuned and very fat sounds. An optional velocity sensitivity, which was not available in the original, has also been implemented. Additionally, the sound generation has been supplemented by a 3-band EQ with a semiparametric middle band, a very good sounding distortion unit and a tempo based stereo delay with filter – effects which perfectly complement the strong sound of Junatik. It goes without saying that the unrenounceable chorus effect of the original has also been implemented – yet without the noise.

# Kaleidon



*Instrument and Presets: Mike Daliot; Panel Design: Christophe Stoll*

Kaleidon is the biggest synth in the entire Reaktor Library. Its sounds cover an exceptionally wide range, from realistic mallets, organs, reeds, and saxophones to house chords, techno basses, and otherworldly atmospheres. Its specs are formidable: six oscillators plus noise and sub-osc with sync, FM, and ring modulation; two multimode filters; three feature-rich envelopes; three LFOs (one which lets you draw your own waveforms); graphical key-tracking; and a full effects section featuring many unique effects, from rich reverb and chorus to a graphically sequenced delay line and quad ring-modulator.

## Quick Start

Kaleidon is Carbon's big brother, so you may want to take a few moments to get familiar with the Carbon synth as well. Instead of Carbon's four oscillators plus noise, Kaleidon offers six plus sub and noise. Instead of

Carbon's single filter-design, Kaleidon features two filters with flexible routing. Instead of Carbon's integrated effects, Kaleidon offers full-featured effects in rack format.

Due to its complexity, Kaleidon takes advantage of both view "A" and view "B" - view "B" is designed to be the day-to-day interface of the synth, with control over the most important synth parameters. View "A" is a tweak-head's delight, as it shows every single parameter directly. A monitor with 1280x1024 resolution is recommended to use the full view "A." The effects are laid out in rack-style underneath the synth. You can maximize or minimize each effect, as needed.

Note that all controls in the reduced "B" view are also duplicated in the full "A" view.

## Signal Flow and Structure

Kaleidon's sound begins with three oscillator sections, with each section consisting of two independent oscillators. Each oscillator has a choice of pulse, sawtooth, triangle, sine, bi-saw, parabolic with PWM, and noise. All three oscillator sections are nearly identical, with oscillator sections 2 and 3 giving enhanced sync and waveform options. When the Qnt button is activated in Osc section 2 or 3, additional harmonics are added to the waveform by reducing the bit depth. Osc sections 2 and 3 also offer a three-saw waveform where three sawtooth waveforms are slightly detuned to create a rich sound. The amplitude, pitch, and width of each oscillator section (acting on both of the oscillators in the section) can be modulated with the pull-down menus at the bottom of each oscillator. An additional noise generator and sub oscillator are also available underneath the oscillators. The A and B oscillators of each oscillator section can be ring modulated with the /RM button.

The oscillators can be flexibly routed to the two filters with the Filter In Routing and Filter Out Routing controls. Here you can set which oscillators go to which filters, and you can set serial or parallel filter routing. The Prefilter loudness correction fine-tunes the amplitude of the oscillators before they hit the filters, to avoid unwanted resonant peaks.

The two filters are nearly identical, with each offering multiple modes, including various flavors of lowpass, bandpass, and highpass. Each filter has a slightly different selection of filter types. Four large knobs control the important parameters of each filter: Cutoff, resonance, bandwidth,

and feedback delay. Feedback delay is only applicable to the Grobian and V4 filter types. Underneath each of the large knobs are the modulation possibilities for that knob. The small knob adjusts the amount of modulation, and the switch turns the modulation on or off for its labeled parameter.

After the filters, the sound passes through Kaleidon's innovative Pan modulator. Here, the sound is spread in stereo across the frequency range. For FX Matrix A, you can choose between chorus, the Vierring sequenced ring modulator, (also available separately in the library), and diffusion delays (also available as the Fusion Reflections effect). After FX Matrix A, the sound passes through a saturator, and then through FX Matrix B. For the B effect, you have a choice of a tempo delay, an innovative sequenced "unit" delay, and a reverb. The reverb core is taken from the SpaceMaster reverb, available in the Library.

The Sequenced Unit Delay effect is unique to Kaleidon. It consists of four delays whose delay time can be sequenced graphically. An additional filter rounds out the effect to create dynamically shifting atmospheres. The "Lurker" patch is an excellent demonstration of what the Unit Delay can do.

Three envelopes can control many of Kaleidon's parameters, and their shape can also be modulated by velocity or the mod wheel. The modulation is controlled in the same manner as the filters.

## **Global Parameters**

Kaleidon's global parameters are nearly identical to Carbon's. Parameters effecting the entire synth are set in the Global section, such as Mono/Poly (set with the Mono button), tuning, unison spread, and glide. Hold lets you specify how long each note will last for, no matter how long the actual MIDI note was held down for (value set in mSec underneath the Hold button). Old controls how much pitch randomness is introduced to emulate vintage hardware.

# Nanowave



*Instrument: Uwe G. Hoenig; Sounds: Sound Burst; Demo: Sound Burst*

Nanowave is an homage to the legendary WaveTable synthesizers made by PPG and Waldorf. The sound generation – called "WaveSet" synthesis here – is structured similarly to the classic subtractive synthesis used in analog synthesizers, but it has some special features on the oscillator level which result in a significantly wider variety of sounds.

The WaveSet oscillators of Nanowave are not based on the common waveforms like sawtooth or square. They use so-called WaveSets which contain many different waveforms. 43 of these WaveSets are included with Nanowave. Within a WaveSet, a particular waveform can be chosen manually but the really crucial feature in WaveSet Synthesis is the option to "travel" in the WaveSet, i.e. switching dynamically between the waveforms of a WaveSet.

Both of the Nanowave oscillators have an identical structure: Each gives you the option to select the WaveSet, the octave and the pitch tuning in semitones. With the slide switch you can manually move through the WaveSet, or alternatively the waveform position can be modulated by a separate wave envelope, by attack, keytracking or by the LFO. Additionally frequency modulation is possible. A couple of other sound sources are also available: a simple sine oscillator, a noise generator with selectable tone colour and the ring-modulation signal produced from waves 1 and 2.



Nanowave's multi-mode filter works with a slope of 12 dB/oct and can be modulated in various ways.

The LFO starts with note onset, and consequently allows envelope-like modulations. It also has a separate envelope for the LFO amplitude.

The three envelopes in Nanowave can be modulated by the keyboard velocity and have a special parameter which enhances the exponential characteristic of this process – very good for extremely percussive sounds.

## SteamPipe



*Instruments: Martijn Zwartjes; Presets: Martijn Zwartjes and Paul Swenenhuis;  
Panel Design: Leonhard Lass*

SteamPipe is a physical-modeling synthesizer that uses a tuned resonator to create bowed, blown, and plucked sounds, as well as many strange new and hybrid sounds. SteamPipe effectively models air, or steam, being blown through a tuneable pipe. In addition to a large number of controls for the "shape" of the pipe, and a tuned all pass filter, there is a mod wheel-controlled filter to achieve damping and breath noise effects. Giving dimension to all of this is an excellent-sounding reverb unit.

## Quick Start

Since SteamPipe is such a potentially expressive instrument, you'll want to plug in your favorite MIDI keyboard to try it out. Step through the presets to get an idea of the different kinds of sounds this thing can make. Give the mod wheel a twirl or two and hear how this can add expression to the sound.

## Structure and Signal Flow

SteamPipe simulates the passing of air through a pipe where the size and resonance can be modified. It uses a type of synthesis called physical modeling, which uses contoured noise passing through feedback delays that can be tuned and filtered. The ensemble is basically separated into three parts: Steam, Pipe, and R66, the reverb module. The Steam module generates shaped and filtered noise. The Pipe module gives the "wind" pitch and resonance. R66 provides the space.

You can think of the Steam module as SteamPipe's oscillator, or at least part of it. Steam provides the sound energy that will be pitch-formed by the Pipe module. To this end there are an ADSR volume envelope, modulated by velocity and keyboard scaling (meaning that the envelope can be set to be shorter or longer overall as effected by velocity), and low pass filter with key- and velocity-tracking.

The Gen section of the Steam module is where the timbral shaping of the DC/Noise source takes place. The low pass filter works in 1-pole or 2-pole mode, though the resonance control only applies to the 2-pole filter. After the noise is filtered, the signal is fed into the Pipe module.

The Pipe module is made up of a number of submodules for creating pitch and resonance. The noise signal is fed from a single, tuned delay, which provides pitch, into the Allpass module for resonance. A Saturator receives the signal next and applies edge and breakup. The MW Filter finishes off the signal chain, giving one final overall tone shaping stage. The FeedBack and Push-Pull sections act on signals diverted from the main signal chain and passed back into it via feedback loops.

The Del Tune module contains the tuned delay that provides pitch to the Steam. The Tune and Fine knobs allow you to dial in the fundamental pitch of the signal. The A440 oscillator at the bottom of the ensemble is there to give a stable pitch to tune the pipe to. The Err knob introduces

small amounts of detuning to the pitch, to give a more human quaver. The Delay pitch can be swept negatively or positively by the mod wheel, with the amount of modulation set by the MW knob.

The Allpass filter receives the tuned signal from the single delay. It can be turned on and off with the Power button in the Allpass section. The Allpass can be tuned to create resonant effects. You can make glassy, metallic and bell-like sounds by detuning the All pass filter against the Delay. By adjusting the Diffusion knob, you can also create a variety of reverb sounds - the simulation of air echoing along a pipe's hard surface.

The Saturation module morphs between saturation and clipping, overdriving or breaking up the signal before it hits the MW Filter.

The MW Filter, controlled by the mod wheel, features a 1-pole high pass followed by a 1-pole low pass filter. Each filter allows you to set a wheel-down and a wheel-up setting, making it possible to set up complex timbre changes and damping effects. Each filter can have its own key track setting.

The lone Polarity switch inverts the pipe polarity, changing the timbre of the sound, often turning high frequency tones to deep ones and vice versa.

The FeedBack module processes the feedback in the signal chain. The RT knob extends or shortens the reverb generated from the feedback signal. The reverb signal can be muffled with the Damp control. Damping can be modified by K-Track amount knob. High K-Track values result in more damping on higher pitches. This allows SteamPipe to emulate struck or plucked instruments like pianos, harps, and acoustic guitars.

## **R66**

The final step in SteamPipe's processing chain is the R66 reverb module. R66 is a full featured and rich-sounding reverb with lots of parameters to adjust. First, the Del module uses predelay time and left/right panning offset to position the signal in time and space.

The Diffusion module has controls for size and reverb time and has a low pass filter and low and high frequencies damping in order to model a convincing space. There are also controls to modulate reverb time and

pan position with a sine wave or random LFO, or both. The Pos knob lets you set the reverb reflections early or late, further coloring the sound of your virtual space.

## Sum Synth



*Instruments and Presets: Lazyfish; Panel Design: Ian Warner*

Sum Synth is an additive synthesizer with a few tricks up its sleeve. A unique oscillator section makes it easy to control the frequencies of the oscillators and shape of the sound, while an informative graphical display shows the current pitch of each oscillator. The number of oscillators can easily be varied simply by changing the polyphony of the instrument in the instrument title bar.

### Quick Start

Get your MIDI keyboard warmed up and check out the presets. Pay special attention to the Tones, Step, Detune, and Harmonics controls since they have a big effect on Sum Synth's sound. See how the four controls interact to change the width and quality of the sound.

## Structure and Signal Flow

Sum Synth uses additive synthesis techniques to generate its deep tones. It has sine or square oscillator waveshapes with the capability to produce up to 128 additive waves, and a sound-shaping section consisting of a stereo 4-pole lowpass filter, an overdrive effect, and the Diffuser Chorus. Two envelopes further shape the sound -- one for volume, and one for filter frequency, which can also modulate oscillator pitch.

The oscillator can be switched between sine and square (with adjustable pulse width) waveforms. The Pitch Control module contains controls to divide the oscillator pitch into detunable tones, select the initial octave, and add glide between notes. The Step knob lets you separate the tones from each other by musical intervals. Detune, will, as you can guess, detune the intervals from each other to give chorusing or atonal sounds. The Harmonics knob generates ring mod-like effects between the tones. Along with a pulse width control for the pulse oscillator, there is an Unphase knob, which causes the oscillator tones to trigger successively out of phase with each other, thickening the sound.

The oscillator feeds into the Pan/Overdrive module. The Drive knob allows you to overdrive the stereo amplifier that the oscillator is fed through. Use the Pan control to widen the stereo soundfield.

After the Overdrive circuit, the signal reaches the stereo lowpass filter. The Filter module gives control over cutoff frequency, resonance, and envelope contour amount. The filter modules are taken from the Pro-52 and give out a rich, creamy sound.

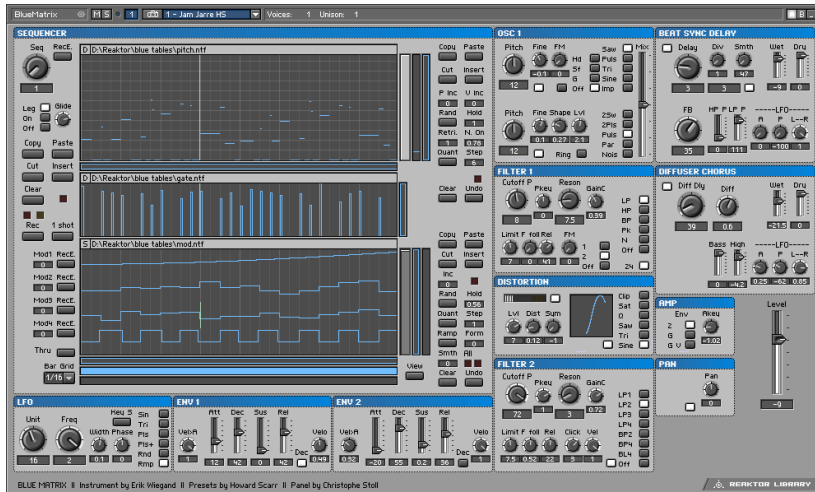
The Filter/Freq AHD envelope works the same way as the Amplitude envelope. Hold/sustain can be activated by velocity, allowing you create dynamic bass drum patterns and expressive lead lines. Velocity can also be routed to filter contour amount. The AmP knob routes the Filter envelope signal to the oscillator pitch. This can be useful for making the little blip at the beginning of those classic bass drums.

The Amplitude ADHSR envelope has a hold stage that can be activated by velocity - click the bottom button to the left of the amplitude envelope display to turn it on. Less velocity will result in more hold time, which can create interesting gating effects in your playing. Velocity can also be routed to volume by clicking the topmost of the two buttons.

The Diffuser Chorus consists of four stereo modulation delays. The main delay time is controlled by the Delay knob. Diffusion sets the inaccuracy of the delay times to "spread" the sound. The Dly Mod control adds an LFO to modulate delay time. The Speed knob sets the rate of modulation. The Stereo knob offers still one more method to spread the sound across the stereo field. A two-band equalizer lets you shape the chorus tone by attenuating the bass and treble frequencies. Clicking the buttons above the Bass and Treble faders engages the EQ bands.

# Synths - Sequenced

## Blue Matrix



*Instrument: Erik Wiegand; Presets: Howard Scarr; Panel Design: Christophe Stoll*

Blue Matrix is an incredible-sounding sequenced synthesizer with a classic analog-style sound engine and an integrated pitch, gate, and modulation sequencer whose capabilities rival those of many stand-alone programs. Its sound engine features two multiple-waveform oscillators, a multi-mode filter, multi-mode distortion, and finally a second multi-mode filter. A beat-synced delay and a diffusion delay round out the effects. The full-featured sequencer offers independent control over gate and pitch for smooth pitch glide, and also offers four channels of graphical modulation sequencing. A complete modulation matrix lets you flexibly route the two envelopes, LFO, four channels of sequenced modulation, and the usual MIDI controllers to every important synth engine parameter.

## Quick Start

Start the system clock and step through the presets! Feel free to experiment with the synth controls - try changing oscillator pitch or cutoff frequency, for instance. If you draw in the tables, you will permanently change that preset if you save the ensemble, so it's probably better to read the Sequencer section of this guide first.

## Structure and Signal Flow

Blue Matrix's synth engine is based on traditional analog-style synthesis. It has two unique oscillators, and a sound-shaping section consisting of Filter 1 ⇒ Distortion ⇒ Filter 2. Both Blue Matrix and its polyphonic, MIDI-playable brother Green Matrix are created from Reaktor 4's Classic Modular macro library which puts the sound quality and capabilities of the classic modular synths into your computer. For more information on building your own instruments with the Classic Modular macro library, please check your Reaktor 4 user's guide. The Classic Modular library is included on the Reaktor 4 CD.

Osc1 and Osc2 each offer a different waveform selection. Osc 1 offers the traditional sawtooth, pulse, triangle, sine, and impulse waveforms, while Osc 2 lets you choose between bipolar ramp, bipolar pulse, pulse, parabolic, and noise. The bipolar waveforms' shape can be modified in realtime with the small Shape knob. Osc 1 can be frequency modulated by Osc 2 for FM sounds (with the small FM knob and switch above), while Osc 2 can be ring modulated by Osc 1 for metallic and "gong" sounds (with the small Ring switch). Osc 1 can also be synced to Osc 2, with the choice between hard, soft, and MIDI gate-activated modes. Since Osc 1 and 2 each offer unique features, they can both be switch on or off to save CPU with the small switch on the right. The base pitch and fine tuning of each oscillator is done with their respective Pitch and Fine knobs.

The two oscillators feed into Filter 1 which offers a choice of 12 or 24 dB/octave lowpass, highpass, bandpass, peak EQ, and notch modes. Cutoff and Reson adjust the cutoff frequency and resonance, respectively, and the FM knobs controls how much Osc 1 or Osc 2 will frequency-modulate the cutoff frequency (settable with the small switch to the right of the FM knob). Pkey determines how the filter tracks the pitch of the notes from the sequencer. When Pkey is set to 1, then the



filter cutoff will exactly follow the pitch of the sequencer notes, meaning that higher notes will be brighter, and lower notes will be duller. If set to zero, then the cutoff frequency won't change. The other knobs, GainC, Limit, F foll, and Rel all control the fine-tuned shaping of the filter across the frequency and resonance spectrum. GainC controls how much loudness reduction will be applied at high resonance values. Limit, F foll, and Rel control a built-in resonance limiter to avoid unwanted harmonic thumps with high resonance values.

The Distortion section between the two multimode filters also features multiple modes of operation. While the clipper mode provides a relatively harsh distortion sound and the saturator mode results in warm overdrive, the several wrapping modes (marked by the name of the waveform used for wrapping) produce unique sounds from subtle to extreme.

A visual display of the distortion function helps to see what's going on inside. Note that the same Filter 1 ⇌ Distortion ⇌ Filter 2 section of Blue Matrix is available as an effect in the Analogic Filter Box.

Filter 2 is similar in layout to Filter 1, but its primary function is to further shape the sound after the Distortion unit. There are a choice of four lowpass filters, two bandpass filters, and a combined bandpass/lowpass filter (BL4). In addition to the controls you're now familiar with from Filter 1, Filter 2 also offers the additional possibility to trigger self-resonance with a built in click. The Click knob controls how loud the click is, and the Vel knob determines how the click responds to velocity.

After Filter 2, the sound passes serially through two delay-based effects: The Beat Sync delay and the Diffuser Chorus. Note that extended versions of these excellent effects are available in the Fusion Reflections and Echomania ensembles. The Beat Sync delay's large Delay knob sets the delay time in sixteenth notes. The Div knob to its right divides the delay time to create dotted and triplet times, while the display above the Div knob shows the actual delay time. For instance, a Delay of seven sixteenths and a Div of two will produce a delay time of 3.5 sixteenths. The small Q switch turns quantization on and off in respect to modulation coming from the sequencer - yes, it is possible to sequence the delay time! You can find more information on the sequencer below. Smth sets an internal smoother on the delay time. Naturally, feedback sets the feedback level, while the HP P and LP P set the

frequency of a highpass and lowpass filter in the feedback path. The delay time can be modulated with an internal LFO, with A (amplitude) controlling the amount of modulation, P (pitch) controlling the frequency, and L-R controlling the stereo spread.

The Diffuser Chorus consists of four stereo modulation delay. The main delay time is controlled by the Diff Dly knob. Diff sets the inaccuracy of the delay times to "spread" the sound. Like the Beat Sync delay, an LFO with similar controls modulates the delay time.

## The Sequencer

Blue Matrix's sequencer consists of three areas where sequencer data can be drawn in - the uppermost handles the pitch of the notes in the sequence, the field below displays the trigger and velocity information of those notes, and the last one contains four independent tracks where additional modulation data can be stored and edited. There are a number of horizontal and vertical control bars for zooming, selecting the length of the sequence, and selecting the "edit" range for commands such as randomize, quantize, and copy/paste/insert.

Important Note!! As the complete sequencer is based on Reaktor's event tables, please note that the table data and snapshot data are independent of each other. The snapshots correctly store the position of all knobs relating to the sequencer, but they don't store any of the table data. Naturally, it is possible for each snapshot to have its own sequence, but you must take care that the global Seq knob is set to the same number as the current snapshot. Otherwise, if different presets share the same table Seq, you could end up changing a table sequence and inadvertently changing the sound of all other snapshots that use the same table. If each snapshot has its unique Seq number, you will avoid this problem. Please note, however, that table data is excluded from the snapshot functions such as morph, randomize, and compare. To be really safe, if you spent a lot of time working on a sequence that you love, save the entire ensemble using the Save As... command.

Pitch-table: To the right of pitch edit-table are three vertical bars. The first is the vertical zoom/scroll bar. Clicking and dragging near the top or bottom of this bar will act as a zoom, while dragging in the middle of the bar scrolls the display. The next vertical bar sets the trigger threshold. Any notes below this threshold won't be triggered. The third bar

sets the edit range. The edit range can be extended by clicking and dragging near the top or bottom, or the edit range can be dragged up or down by clicking and dragging in the middle of it. An additional, horizontal editor bar is provided to select notes in time. The copy/paste/cut/insert functions to the right of the vertical bars do the usual functions (for the area defined by the horizontal and vertical edit bars), with a couple neat tricks. P Inc sets how much the pitch will be incremented when data is inserted or pasted, in semitones. If you wanted to transpose a figure by five semitones, for instance, simply arrange the horizontal and vertical edit bars to cover the length of that figure, hit copy, then insert. Likewise, V inc sets the velocity increment of pasted or inserted data.

**Randomizing and Quantizing data:** In addition to copying and pasting the range denoted by the edit bars, you can also randomize data within this range. Simply click on Rand, and the edit areas will be filled with random pitches whose properties are determined by the Hold, Retrig, and N. On functions. Data in the edit range can also be quantized with the Quant button, with resolution set by Step. An undo function is available for copy/paste, randomizing, quantization, clear, and recording data into the table.

**Gate table and Glide:** Underneath the Pitch table is a table where you can draw in gate (note-on) events. The height of the event corresponds to velocity. Note that in Blue Matrix's sequencer, pitch and gate are separate. The sequencer still sends out pitch information even if there is no gate signal. The means that you can create custom glides. The Glide time is set with the small Glide knob to the left of the pitch table. Small values of glide mean the oscillator pitch reflects what's shown in the pitch table. Large glide values create a ramp, or smoothing, from one pitch to the next. In Leg (Legato) mode, then glide is only active for overlapping notes.

**Modulation Table:** The third table in the sequencer is a four channel modulation sequencer, where you can independently draw and edit four different modulations. You can route the modulations to synth parameters in the matrix modulation section. Similar to the pitch sequencer, to the right of the mod sequencer there are three vertical bars, each with a different function. The first one is a zoom/scroll bar. The second selects the vertical edit range, while the third bar selects which channel of modulation is shown in the display. You can either view and edit a single channel, or you can view (but not edit) all four channels simulta-

neously. To switch between these two modes, simply click on the View button underneath the three vertical control bars. When there is a vertical grid shown, then you can draw in your modulation data. After drawing in a shape, click on View again and you will see the two-dimensional representation, where blue represents minimum values and red represents high values. In two-dimensional mode, you can view all four modulation channels simultaneously.

Global loop length and zoom controls: Underneath the modulation table are three horizontal global control bars that affect all tables. The first one is the edit-range bar, that is also linked to the edit range bar for the pitch table. The middle horizontal bar is the sequence loop length bar. Similar to the other controls you're now familiar with, this bar can be resized by dragging near the beginning or end, and can be moved by dragging in the middle. You can set the quantize value for the loop bar with the Bar control to its left. The bottom-most bar is the horizontal zoom/range bar, which operates the same as all the other zoom/range bars: dragging the beginning or end acts as zoom, and dragging the middle scrolls.

Recording data into the sequencer: It's possible to save MIDI data sent to the instrument, e. g. by a keyboard. The Rec> buttons at the left of the data fields select and activate the different sequencer parts for recording. Note that both upper data fields can only be recorded together (as MIDI events always contain both pitch and trigger information) while each modulation track can be recorded separately. The actual recording starts when the Rec button is pushed. If at this moment another region of the sequence is playing than the region selected by the horizontal edit bar, the W lamp will light up to mark the sequencer's waiting status; when the global read-out pointer enters the edit region, the R lamp is will light to show that the instrument is recording.

### **Two hints for using the tables:**

- 1) If you accidentally draw into a table, you can always use the table's Undo command.
- 2) Use the global copy and paste commands to work on a copy of a sequence, or to build a new sequence up from an existing one. Set the horizontal edit bar (either the one under the pitch table or the modulation table, they're always linked) to the length of the entire sequence.

You may need to zoom out (bottommost bar underneath the modulation sequencer). Use the global copy and paste command (large buttons to the left of the trigger sequencer): Select the sequence that you want to copy using the Seq knob, press Copy, then use the Seq knob to go to the empty or destination sequence. Press Paste and viola!

## Modulation

Blue Matrix features two identical standard ADSR envelopes and a single LFO that syncs to tempo. Both envelopes are velocity sensitive: Vel>A sets how much the velocity will affect attack time, while Velo sets how much velocity will control the amplitude of the entire envelope. The speed of the LFO can be set in musical note values, or freely. When Unit is set to 0, then the Frequency of the LFO is set in Hz. When Unit is set to any other value, the frequency of the LFO takes on the musical units set by Unit. For example, if Unit is 96 and frequency is 1, then the LFO will be set to the very fast speed of 1/96th note per cycle. The display underneath the Freq knob depends on how Unit is set: if Unit is set to 0, then the display automatically shows values in Hz, and if Unit is set to a musical value, then the Freq display will automatically update. Width and Phase control the shape of the LFO, which can be selected by the switch to the right. Key S determines if the LFO will synchronize to note-on events, as delivered by the sequencer.

All of Blue Matrix's modulation sources are routed in a comprehensive modulation matrix: The LFO, two envelopes, four modulation channels from the sequencer, and typical MIDI controls like mod/pitchbend/velocity can be routed to synthesizer destinations using the comprehensive modulation matrix. Switch to the "B" view of the synth to see the modulation matrix. You can easily switch to the B view by clicking the B button in the title bar of the synth. You can route the modulation to the following synth parameters: Oscillator pitch (osc 1, osc 2, or osc 1 + 2), osc 1 FM, osc 2 shape, osc ½ mix, filter cutoff (osc 1, osc 2, or osc 1 + 2), filter 1 FM, filter resonance (1 or 2), envelope parameters, and even distortion, delay time, and delay feedback. In short, nearly everything!

## Vierring



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## Quick Start

Play some audio through the effect\* and jump through the presets. Note that vierring works best with harmonically rich material, such as bright sustained pads. For an interesting result, try running a complete classical music CD or record through!

## Structure and Signal Flow

Before sound enters the Vierring ring modulation engine, the filters in the Freq Shift section split up the sound into four bands. Hicut and locut will remove high and low frequencies respectively, while the three band pass (BP) knobs set the crossover points to create four frequency ranges.

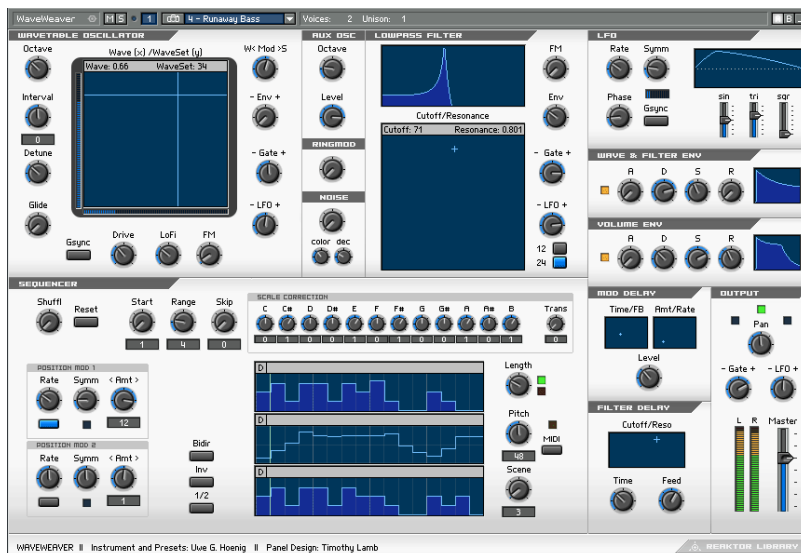
The controls for each of the four bands are identical. Lev sets the volume of the band, and Pan controls the position in the stereo field. The BW and Res controls set the parameters for the band pass filter used to feed audio into the band. The RM button turns ring modulation on, and the RM knobs adjust the amount of ring modulation. You can adjust the attack and decay characteristics of the trigger envelope with the att and dec knobs, respectively. The Length-64 parameter sets the pattern length of that band. Note that it's possible to have an 8-step lowband, and a 7-step midband, for instance!

With the event table to the right of the band controls, you can sequence the amplitude and ring modulation amount (top sequence) and the effect send (bottom sequencer). The 1<>2 and wetL knobs determine to which delay the signal to, and the volume of the wet signal. The snd and viaT2 knobs adjust the amount of delay send, and the amount of delay send as specified by the second table. Underneath the two tables is a small horizontal bar that shows the frequency range that that band represents.

In the Global section, you can set the active set of tables with the Pattern knob. The attack, dec, res, and BW knobs perform the same function as the similarly-labeled knobs in each band, but they act globally. Note that the filter bandwidth and the global filter frequency can be modulated by the graphical mod sequencer.

\* You can play a sound through the built-in loop-player, through the realtime audio inputs, or you can process audio in realtime by using Reaktor as an effect plugin. Please check your Reaktor or Reaktor Session user's guide for helpful information.

# WaveWeaver



*Instrument and Presets: Uwe G. Hoenig; Panel Design: Timothy Lamb*

The WaveWeaver ensemble uses wavetable synthesis to produce a range of subtle or harsh tones and rhythmic pads. The oscillator in WaveWeaver is actually several sets of short, looping soundfiles, each used as an oscillator waveform. The soundfiles in each wavetable can be blended into each other, and you can use an XY pad to simultaneously select among them and modify the loop start point. A built-in sequencer with scale correction allows you to create some unique, shifting grooves. MoDelay and FilterDelay let you add some effects to your sounds.

## Quick Start

Although you can play WaveWeaver from a keyboard, it's a good idea to start the system clock and run through some of the snapshots. This way you can get a feel for what WaveWeaver's controls do while sound is being made.



## Structure and Signal Flow

WaveWeaver's wavetable, which can be extensively modulated and morphed in tone, feeds through a distortion circuit and a bit reducer before joining an auxiliary sine wave oscillator and a noise source at the 4-pole low pass filter. After the filter, the signal travels through the ModDelay and the FilterDelay. Along the way, an LFO, two ADSR envelopes, and the sequencer modulate the wavetable and filter.

The Wavetable Oscillator section, in the upper left corner, is where the controls for sound generation are located. There are actually two wavetables, which can be detuned from one another for a wider sound. You can crossfade between the various soundfiles of the waveset and set loop start points by moving the cursor in the XY pad. Soundfiles are located on the vertical, or Y, axis and the start is on the horizontal, or X, axis. The Octave knob to the left of the Wavetable pad lets you set the octave shift for both the main oscillator pair and the auxiliary oscillator. The Interval knob, immediately below this, controls the pitch shift, in semitones, of the second wavetable. The Detune knob allows you to fine tune the interval of the two oscillators. You can also set the amount of pitch glide for the oscillators using the Glide knob.

The four knobs to the right of the XY pad control the intensity and polarity of different modulation sources that can have an effect on the wavetable. At the top, the W< Mod >S knob sets the amount of position modulation within the waveset when turned to the left, and modulates the selection of the soundfile when turned to the right. Below this knob, the - Envelope +, - Gate +, - and LFO + knobs control the mixing of the various modulation sources. You could, for example, mix the negative output of the filter envelope with the positive output of the LFO to modulate soundfile selection. The signal from the wavetable passes through a distortion circuit and a bit reducer. The Drive and LoFi knobs control the amount of processing by these effects. In addition, the wavetables can be frequency modulated by the auxiliary sine wave oscillator, for extreme FM sounds. You can set the amount with the FM knob. Pressing the Gsync switch causes the two wavetable oscillators to trigger in sync with one another on gate signals. Otherwise the oscillators start at their current free-running position. Playing with the Drive, LoFi, and FM controls can yield a variety of sounds ranging from spiky to metallic to liquid.

Before reaching the filter, the wavetables can be mixed with the aux sine wave, which features tuning relative to the main oscillators in octaves, and a noise source with adjustable filter color and envelope decay. The sine oscillator can ring modulate the wavetables, and the resulting signal can also be mixed into the filter input with the Ringmod level knob.

All of these sources are fed into a 4-pole low pass filter, whose cutoff frequency and resonance can be set with the Filter XY pad. Much like in the Wavetable section, the knobs along the right edge of the Filter XY pad control the amount of positive or negative filter modulation from the envelope, LFO, and gate signals.

These modulation sources reside on the right side of the ensemble. The LFO allows you to mix sine, triangle, and square wave shapes to achieve complex modulations. LFO frequency and waveform symmetry can be controlled with the Rate and Symm knobs, respectively. You can also set the start point of the LFO wave when it is retriggered using the Phase knob. When the Gysync button is switched on, the LFO is retriggered with each gate signal; it restarts at the point in the waveform set by the Phase knob.

An ADSR envelope modulates the wavetable and/or the filter frequency. Another ADSR controls the amplitude contour.

The final elements in the signal chain are the MoDelay and the FilterDelay. The MoDelay produces dimensional chorus and flanging sounds with its delay time LFO. The Time/FB XY pad allows you to smoothly adjust delay time and feedback. The Amt/Rate XY pad lets you set the amount and frequency of the LFO modulating delay time.

The FilterDelay is a tempo-synced delay with a resonant low pass filter to alter the sound output. You can change the filter cutoff and resonance with the Cut/Res XY pad. Since the delay time is automatically synced to the system clock, you can use the Time knob to change delay time by beat divisions.

## **The Sequencer**

The three-part sequencer is started and tempo-controlled by the global MIDI clock. Swing can be added with the Shuffle control. You can adjust the sequence start point and loop length with the Start and range knobs. Pressing the Reset button in the Clock section forces the sequence to

reset itself to the position determined by the Start knob. The Skip knob sets the number of steps in the sequencer's event table left out after each time the sequence advances.

The Bidir and Inv buttons change the order in which the sequence steps play back. Bidir causes the sequence to run forward, then backward, while Inv inverts the playback order. These controls are extremely useful for nondestructively altering the character of a sequence in performance, for example. You can also modulate the current playback position of the sequencer over time with the two Position Mod LFOs. Each can have the modulation amount inverted and have control over LFO waveform symmetry with the Symm knobs.

The top sequencer row controls the length of each note. The middle row lets you program a pitch sequence, while the bottom row sets gate value, which is wired to effect velocity, or volume, for each step. The pitch can be offset to transpose the sequence using the Pitch knob. Engaging the MIDI switch next to Pitch allows you to transpose the base pitch of the sequence and play chords with a MIDI keyboard. You can use these controls to create complex, modulating arpeggiations. The Scale Correction section gives you the option of mapping pitch data onto a specific scale, for example a c major or pentatonic scale, or a Dorian mode. This can create subtle and moving effects when you transpose the sequence pitch using a keyboard.

# Samplers and Transformers

## BeatSlicer 2



*Instrument and Documentation: James Walker-Hall; Panel Design; Christophe Stoll*

BeatSlicer will separate any waveform into smaller component 'slices', which can then be individually tailored by adjusting pitch, envelope and a range of FX parameters. BeatSlicer is primarily designed for drum-loop manipulation, but the extensive range of parameters offer many creative options with any material.

### Quick start

Right-click (PC) / Ctrl-click (Mac) on the large central window, select 'File', 'Load data into table ...' and then choose an audio loop from your sound library. Now, press the 'slice' button at the top of the instrument panel. The loop will be scanned and MIDI notes from C-2 (by default) will be assigned to the detected slices.

## Global controls

Tune allows you to transpose the pitch of the entire drum kit in semitones, and Gain controls the overall output level.

Pb determines the amount that the pitchbend wheel affects the pitch of the entire loop, and Vel specifies the amount to which note velocity determines amplitude, again, for all slices. Note, however that both pitchbend and velocity can be assigned to slices individually using the modulation matrix, in which case, it is probably best to set these two global knobs to zero.

Root selects the root key for the loop - i.e. the first slice will be assigned to this key. (60=Middle C, 48 C-1 etc.)

Rst! (reset) sets all per-slice settings to default.

BeatSlicer can slice loops by employing either a transient detection algorithm, or by dividing the loop into equal-length sections.

To scan a loop for transients press Detect!. If more or less slices are required, adjust the Sens (sensitivity) knob (higher values result in more slices) and press Detect! again. Thr (threshold) specifies the lowest permitted volume (in decibels) of slices. You will not normally need to edit this parameter.

To cut a loop into equal length slices, set the desired number of slices in the drop down menu, and then press Slice!. For example, to slice a 4 bar loop (4/4 time) into 16th's, select 64 slices (as  $4 * 16 = 64$ ). Good results will only usually be attained if the loop is an exact number of bars in length.

Note that if you keep the mouse key depressed after pressing either Detect! or Slice!, detected slices will be revealed graphically.

After using either method, additional slices can be added manually with the Add! button. Additional slices are placed at the end of the loop, so the position will require adjustment using the start and length controls (see below). Add can be used to compensate for slices missed by the detection algorithm, and also for creating multiple versions of the same slice, but with different FX parameters.

## Loop section

The large window displays the waveform of the current loop. To load a loop, Right-click (PC) / Ctrl-click (Mac) on this window and select 'File', 'Load data into table ...'.

The large slider bar at the top of the section indicates the current slice selected for editing. Clicking and dragging on this bar selects the slice you wish to edit (and also auditions slices). Slices can also be selected with the Sel (select) knob.

When Trk (track) is enabled, the current slice is determined by the most recent MIDI note input. Disabling Trk is often preferable when BeatSlicer is being 'played' by a sequencer, and you want to select and tweak one slice in particular. Enabling the Solo button filters all MIDI input except for the current slice.

The Zoom button is useful for making fine adjustments to the slice position and length.

Usually, changes to any parameter apply to the current slice only. However, when the All button is enabled, you can apply changes to the entire loop simultaneously.

## Slice tweaking

BeatSlicer enables you to customise the sound of each slice in your loop, as summarised in the following table:

**Start and Length:** This section allows you to manually 'nudge' the position and length of slices. The top row buttons are for coarse adjustments, whereas the second row buttons are for fine adjustments. The fine nudge distance can be specified in milliseconds using the knob at the bottom of each section. The coarse nudge distance is 10x this amount.

**Pitch:** P transposes pitch in semitones, and the < button reverses playback.

**Envelope:** A specifies the envelope attack time, as a proportion of the length of the slice. Thus, if A is set to 0.5, the envelope will reach peak value halfway through the slice. Similarly, D determines decay time as a proportion of slice length, so a setting of 0.5 will start the decay period (the descent to zero amplitude) half way through the slice. Therefore, the overall shape of the envelope is not only determined by the attack and

decay times, but also the slice length. Enabling the s button switches to key sustain mode. If the MIDI note is released before the end of the slice has been reached, the envelope will terminate prematurely.

**Compress:** A digital compressor / wave-shaper that reduces variations in amplitude in order to obtain a louder, denser sound. Thr (threshold) determines the level (db) at which compression begins, and R (ratio) determines the extent of compression. Output gain is automatically increased so that peak input and output remain the same.

**Drive:** Overdrive distortion with a smooth-curved saturation characteristic. Db sets the input drive amount in decibels. Output gain is automatically reduced so that peak input and output remain the same.

**Filter:** A LP / HP resonant filter. Cut determines the filter cutoff point, and Res determines the amount of gain at that point. HP toggles between high-pass and low-pass mode. Hq employs an advanced filter algorithm that may sound better with certain settings (most typically high resonance). In HQ mode the 24 switch changes both the LP and the HP filter to 24db mode, in standard mode the 24 switch changes the LP filter only.

**Aux:** The 2 aux sends enable you to connect external effects to BeatSlicer, and set the send levels individually for each slice. For example, you could route some hi-hats to a tempo delay unit, but not the kick drum. Normally, the aux send level is post-fader (i.e. relative to the gain slider in the Mix section.) However, when Pr (Pre-fader) is enabled, it is independent of the gain slider. Note that setting the send level to minimum (-40db) disables the aux send.

**Out:** Gain determines the slice output level. Setting the gain to minimum (-40db) mutes the output.

## Modulation

BeatSlicer allows various slice parameters to be modulated by several sources.

In both modulation sections (A and B) the left text box displays the current source, and the adjacent button cycles through other available sources. The right text box displays the current destination, and the

adjacent button cycles through other destinations. The slider bar specifies the amount (and direction) to which the source modulates the destination.

For example, to assign velocity to amplitude, select 'Velocity' as the source, 'Amp' as the destination, and then set the slider bar to the full right hand position.

### **Modulation sources**

- Velocity: MIDI note on velocity (range 0 to 1).
- PB: MIDI pitchbend wheel (range -1 to 1)
- PB/H: As above, but held for the duration of the slice.
- CC1: MIDI CC1 - the modulation wheel (range 0 to 1)
- CC1/H: As above, but held for the duration of the slice.
- Random: The randomiser generates a value between -1 and 1 every time a slice is triggered.

### **Modulation destinations**

- Gain: Slice amplitude(-100% to 100%)
- Pitch: Slice pitch(-12 to +12 semitones)
- Attack: Envelope attack time(-1 to +1)
- Decay: Envelope hold time(-1 to +1)
- Ratio: Compression ratio(-20 to +20)
- Drive: Overdrive amount(-60 to +60 decibels)
- Cutoff: Filter cutoff(-60 to +60 semitones)
- Aux 1: Aux 1 send level(-100% to 100%)
- Aux 2: Aux 2 send level(-100% to 100%)

## **Sync Delay and Drum16**

The BeatSlicer2 ensemble also features a tempo sync delay fx unit, connected to BeatSlicers auxiliary outputs. To use it, ensure that the aux section is enabled in BeatSlicer, and the delay module is switched on.

Additionally, a 16-channel sequencer is included, which is useful for driving BeatSlicer in Reaktor standalone mode (but will also sync to the host clock in plug-in mode).



## Delay module

**Delay:** The upper menu specifies the delay time in musical-time. Medium times (e.g. 3/16) are useful for rhythmical delay, whereas short times (e.g. 1/96) can obtain electronic feedback sounds. The lower menu specifies the filter mode - for standard delay-type sounds, either Off or LP is recommended. The other filter modes can obtain interesting sounds, but should be used with caution as they can lead to feedback loops with high resonance. Input specifies the initial feed level into the delay unit. The output is then re-fed into the unit at the level specified by FB (feedback), which thus determines the rate of decay and the length of the tail. Cut and Res control the cutoff and resonance for the feedback filter. Be careful with high resonance!

**Mix:** Gain specifies post-delay output level, Pan the stereo output position of the channel, and Wet mixes between the dry and wet signal. To use this module as a stereo effect, pan channel 1 hard-left and channel 2 hard-right.

**LFO / Route:** The two tempo-synced LFO's can modulate various parameters via the Route section. Most typically perhaps, stereo pan to create ping-pong delay.

## Drum16

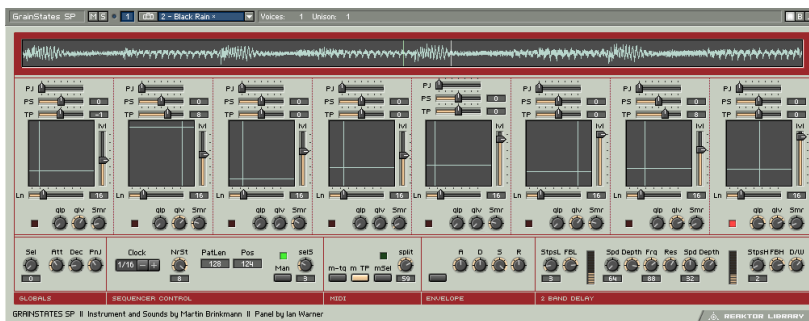
**Arrange:** The pattern length can be specified with the Bars and Beats controls. When Seq is activated, the pattern sequencer controls the current pattern.

**Select / Copy:** When Seq is disabled, the top slider selects the current pattern to play (Pressing Clear will erase this pattern. The lower slider selects the copy destination. Pressing Copy will overwrite this pattern.

**Matrix:** Here, in addition to drawing the actual pattern sequence, you can select the MIDI note of each channel, and Mute and Solo channels with the left and right blue buttons respectively.

**Pitchbend:** By default, values in the Pitchbend grid snap to 1/24 range of the wheel. If BeatSlicers pitchbend range is set to 12 semitones, each step on the grid will correspond to one semitone.

## GrainStates



*Instrument and Presets: Martin Brinkmann; Panel Design: Ian Warner*

GrainStates is a granular texture maker that works wonders for creating dense, breathing atmospheres. Taking advantage of Reaktor 4's grain cloud delay module, GrainStates lets you create granular soundscapes in realtime. You can even freeze the live audio - imagine playing a guitar into Grainstates, freezing the audio, then playing a counterpoint to the granular texture. Eight scenes - each scene storing information about grain size, density, pitch, pitch spread, and more - are sequentially recalled in sync with the master tempo. A dual-frequency delay adds depth to the sound by letting you specify independent delay and feedback times for the high and low frequencies.

Grainstates consists of two sister ensembles: GrainStatesFX and GrainStatesSP. GrainStatesFX is an effect using the grain cloud delay that works on live input, while GrainStatesSP is centered around the grain cloud sampler module. The FX ensemble has the advantages that you can easily process audio without loading anything into a sampler and you can freeze the incoming audio stream - great for live performance! Since the sound passes through the ensemble, however, there's no way to save the sound data with the preset. GrainStatesSP stores the sample with the preset so it can easily be recalled at a later time, but you must first load your sound into the grain cloud sampler.

## Quick Start

GrainStatesFX: Start the system clock and run some audio through the ensemble.\* You'll notice a graphical representation of the sound "marching through" the granular buffer. To freeze the buffer, press the Freeze button to the left of this graphical display. If you stop the system clock, the sound will continue, but the scenes won't advance.

GrainStatesSP: Start the system clock and step through the presets. If you stop the system clock, the sound will continue, but the scenes won't advance.

## Structure and Signal Flow

At the heart of GrainStates is the granular grain cloud module. GrainStatesFX is based on two (for true stereo operation) grain cloud delay modules, while GrainStatesSP is based on a single grain cloud sampler module. Both the grain cloud delay and grain cloud sampler have identical controls, with the grain cloud delay adding the ability to freeze the sound. All of GrainState's controls (with the exception of the 2Band Delay) are used to control the grain cloud.

A master sequencer runs through eight scenes are run through sequentially, with each scene providing control over various granular parameters. Every scene can have its own length, settable by the Ln slider, whose units are set in the Seq Control macro. You can also set the total number of scenes (NrSt), and if you want to disable the scene sequencer simply click on "man" and you can select a scene manually with the SelS knob.

Each scene provide control over pitch jitter (PJ: amount of pitch randomization, in semitones), pitch shift (PS: in semitones), transposition (TP: in semitones), volume (Lvl), and an XY panel lets you set two parameters graphically at once: The horizontal axis sets the start position of the grain (relative to the graphical display on top of the internal buffer), while the vertical axis sets the length of the grain. Three additional knobs provide control over the smoothness of scene transitions, and the grain density smear (Smr).

GrainStatesFX only: The sound is filtered by the Filtor macro with independently adjustably highpass (HP) and lowpass (LP) frequencies. Res adjusts the resonance of both filters, while Byps disables the filter.

The output of the filters is fed back into the grain cloud delay, with feedback independently adjustable per scene with the FB slider to the right of the XY control. The feedback is only active then the grain cloud delay is not frozen. When the grain cloud delay is frozen, it ignores any signal to its inputs.

The sound then passes through a 2Band Delay, which gives independent control over delay time in sixteenth notes (StepsL and StepsH), feedback (FBL and FBH), and a filter-modulating LFO. The cutoff and resonance of the filter that splits the two bands is determined by the Frq and Res knobs. Finally, a D/W knob sets the mix level. To bypass the filter, simply set D/W to zero.

## **Additional Controls**

Global Params. In the global parameters, you can set the global attack and decay of the grains, and the amount of pan jitter (stereo randomization). In GrainStatesSP, this is also where you select the active sample, with Sel.

GrainStatesFX only: the Move macro controls a built-in ramp oscillator that controls the delay time. The Steady knob is the amount of delay modulation - when at zero, then the ramp oscillator does not change the delay time.

## **MIDI Control**

In addition to automated sequencer control, GrainStates also lets perform with a MIDI keyboard. In the MIDI macro, you have control over the MIDI functionality. If "m TP" is activated, then MIDI notes will pitch the sound, like in a conventional sampler. In GrainStates, however, all notes are the same length, regardless of pitch. When "mSel" is active, each scene is mapped onto a note pitch between 48 and 59 (only the white keys of a keyboard); by pressing one of the notes the respective scene is selected. With the Split knob you can specify another keyrange that recalls scenes. "m-tg" toggles MIDI triggering of sound on and off. GrainStatesFX only: "m-frz" lets you toggle the freeze effect on/off via MIDI.

GrainStatesSP only: If you play GrainStatesSP over MIDI, you can activate the envelope (env macro) to also control the amplitude of the sound with MIDI, according to the settings of the envelope.

\* You can play a sound through the built-in loop-player, through the realtime audio inputs, or you can process audio in realtime by using Reaktor as an effect plugin. Please check your Reaktor or Reaktor Session user's guide for helpful information.

## Travelizer III



*Instruments and Presets: Uwe G. Hoenig; Panel Design: Cornelius Lejeune*

Travelizer III is the latest version of the classic granular texture maker. Travelizer lets you scrub through any sample using the grain cloud module. It can be played over MIDI, allowing you to create ethereal pads and leads. Travelizer III differs from its earlier versions because grain length can be quantized to 16th notes, allowing textures and granular rhythms to be synced to MIDI clock.

## Quick Start

Step through the presets with the system clock on and feel free to experiment! Try using the travel xy pad to control sample playback position and grain length while a sample is playing. If you click on the pad, the volume Attack-Release envelope will trigger and start the sound. Click the Gate button to continuously trigger the volume envelope so you can make adjustments while the sample is looping.

## Structure and signal flow

Travelizer III granulates any sample loaded into it, giving you a variety of controls over the quality of the granulation. The sample grains pass through a 3-voice Resonator with independent controls for tuning and the capability to track midi notes. The Resonator is followed by a stereo delay with paired with a high-pass filter. An attack-release envelope allows you to contour the loudness of the sound.

Travelizer III's grain cloud module allows you to break the sample down into grains, specifying certain qualities of the sound, like grain size and smoothness.

The Pitch module has controls to tune the sample and set whether it will track incoming MIDI notes. The MIDI switch turns note-tracking of sample pitch on. The Jitter control introduces small amounts of pitch modulation. The Slide switch next to this makes the sample pitch glide from note to note. Slide intensity can be positive or negative. An LFO allows you to modulate Travelizer III's pitch - turn it on and off with the small switch above the Shp fader. The LFO can be crossfaded between triangle wave and sine wave shapes with the Shp fader. Clicking and dragging on the Amount and Rate x-y pad lets you adjust modulation depth and speed. The LFO can be set to retrigger on MIDI gate events with the Gate button.

The Position module allows you to modulate the playback position of the sample.

You can smoothly, or roughly, sweep through the sample using a crossfadeable sine/triangle LFO like the one in the Pitch module. The Inertia control allows you to smooth out the LFO movement, in effect limiting its swing. The Jitter control adds small amounts of random position modulation to achieve a glitchy rhythmic edge. Like the Pitch LFO, the Position LFO can be retriggered by MIDI gate events by clicking the Gate button.

The Smoothness control lets you adjust volume crossfading between sample grains. A high Smoothness amount will result in a more defined sound with less obvious granulation. The lower the Smoothness control is set, the more granulated the output will sound.

The sample is now fed into the Resonator. The Resonator uses three voices to create tuned atmospheric and spatial effects. Each of the three voices can be tuned in semitones relative to each other. Clicking the

MIDI switch lets you control the root pitch using MIDI notes. You can control damping and decay of the Resonator with the xy pad. A rhythmic loop can be turned into a harmonic soundscape by manipulating the damping and decay controls and creating chords with the voice tunings.

From the Resonator, the sample passes through the Delay module, which includes a high-pass filter. You can use this to manipulate the sonic character of the sample before it hits the delay by removing low frequencies with the filter. The Delay xy pad lets you set 16th note-quantized delay times for the left and right channels.

Finally, an attack-release envelope allows you to contour the loudness of your sound. Use the sample control to select a sample or double-click on the grain cloud module to load your own.

# Live Tools

## goBox



*Instrument and Presets: Uwe G. Hoenig; Panel Design: Timothy Lamb*

goBox is a monophonic sampler specially designed for live use. It features some hands controls for altering patterns, sounds, and modulations during a performance. A series of event tables form an easy to use interface that make it easy to see what's going on. Apart from sample modulation capabilities, there are a tempo-synced Filter Delay, a Mod Delay, and the Sync-ro-nizer module, which can play patterns of short samples metronomically.

## Quick Start

Start the system clock and check out some of the presets. As the sequence plays, try adjusting some of the sample modulation parameters, like Attack, Loop, or Octave. Try out the tap 'n' drag slider located under the waveform display to manually modulate a sample as it plays.



## Structure and Signal Flow

goBox is a sequenced sampler of an unusual sort. Instead of playing loops it triggers one short sample at a time, monophonically. This can lead to bizarre junky/funky beats, uncertain patterings, and minimalist dance floor workouts. Two event tables sequence the pitch and trigger state of each of four sampler modules.

The Start point, Octave, and Pan position can be set using event tables for each of the samples. Start determines where in the sample the loop will start, while Octave tunes each sample in octave steps. Pan lets you position the sample in the stereo field.

The samples have their amplitudes contoured by four ADR envelopes. The tables for these are located in the second row in the sample modulation area. Overall volume for each of the four sample channels can be set using the Gain control. The Reduktor performs sample rate- and bit depth-reduction for each sample. Use this to add some dirt to your samples or to reduce them to static noise. All of the envelopes can be scaled, that is, generally shortened or lengthened in time, with the Scale button below the Morph XY pad. Use the Scale control to tighten up beats or to make beats seem louder by increasing the overall decay and release times.

The Loop table controls how much of the sample loops as it is triggered - beginning at the point set by the Start control. Odd loop lengths can create stuttering and staccato rhythm effects.

After being contoured, the samples can be fed into the effects modules in differing amounts with the Filter Delay and the Mod Delay tables. The Filter Delay's delay time is automatically synchronized to the system clock in 16th notes. Filter frequency and resonance can be adjusted with the XY pad. Use the Envelope knob to modulate filter cutoff with the sample envelope.

A Mod Delay for chorus and reverb-like effects accompanies the Filter Delay. Two xy pads let you adjust the Time/Feedback balance and the Amount/Rate of delay time modulation.

The Sync-ro-nizer module sits on the end of the signal chain after the effects. It offers two synced sample players for generating simple, metronomic beats to help keep everything in line and lock down your rhythms. Select a sample using the Select knob. The sample pitch can be adjusted using the Tune control. The Style knob selects a timing value

on a preset sample trigger grid (i.e. 1/8th notes, 1/16th notes, 1/32nd notes, etc). You can turn each sample on and off using the mute buttons below the level meters.

## The Sequencer

Event tables are used to sequence the pitch and trigger values for the sample loop module. The Pitch table sets the pitch for each step of a 16th note grid. The Trigger table selects which of the four possible samples plays on each step of the grid. You can place a rest into a sequence by selecting no sample for that point on the grid. As the clock runs, the sample selections play on the steps they are set for. A Transpose fader to the right of the Pitch and Trigger tables shifts the pitch sequence up or down.

You can select a sample for each of the four samplers using the Choose table. Load new sample maps into the sampler module by double clicking on the waveform display. You can label your sample categories by double clicking on the text field to the left of the Choose table and typing in the corresponding areas.

The clock controls for running and modulating the sequence all appear in the upper left hand corner. You can use the Scene knob to store sequencer and sample settings independently of the snapshot menu. You can store up to eight scenes. The Reset selector allows you to determine when the sequencer will reset its start point to the step chosen with the Start selector. From top to bottom, the sequence can be set to: reset every bar, reset every 2nd bar, reset every fourth bar, or no reset at all. The Grid selector, below Reset, is used to set the beat resolution of the sequence grid, i.e. 1/32, 1/16, 1/8, 1/4.

Set the length of the sequencer pattern using the Range selector. A range of "0" will mute the sequencer. The T button will instantly switch the sequencer resolution to triplets of the chosen grid value, introducing interesting rhythmic twists. The 1/2 button will set the sequence playback to half time. Since this is a "nondestructive" event, not changing the timing, goBox will always stay on the beat. The Reverse switch, labeled by an arrow pointing to the left, has the effect of reversing the sequence playback. This is also a "nondestructive" effect, instantly switching on the beat back to forward play. The Bi-Directional sequence

playback button, to the right of Reverse, causes the sequence to alternate forward and reverse playback. Finally, the Shuffle knob allows you to add varying degrees of swing to the sample triggering.

goBox can shift the sequencer patterns automatically to keep your beats fresh by using the Position Mod LFO. Position Mod is a low frequency square wave with adjustable width and rate that modulate the current sequencer step position. You can specify the range of modulation in sequencer steps by using the Steps control.

Along goBox's bottom edge are a variety of hands-on real-time sequence modulation controls. The Skip buttons causes the sequencer to skip 1, 2, or 3 steps - adding dramatic rhythmic variations. The Trig buttons cause the currently playing sample to retrigger, or roll, at 1/8, 1/16, or 1/32 notes. Try these buttons while the sequencer is running to drop rolls and trills into the sequence as fills or turn-arounds.

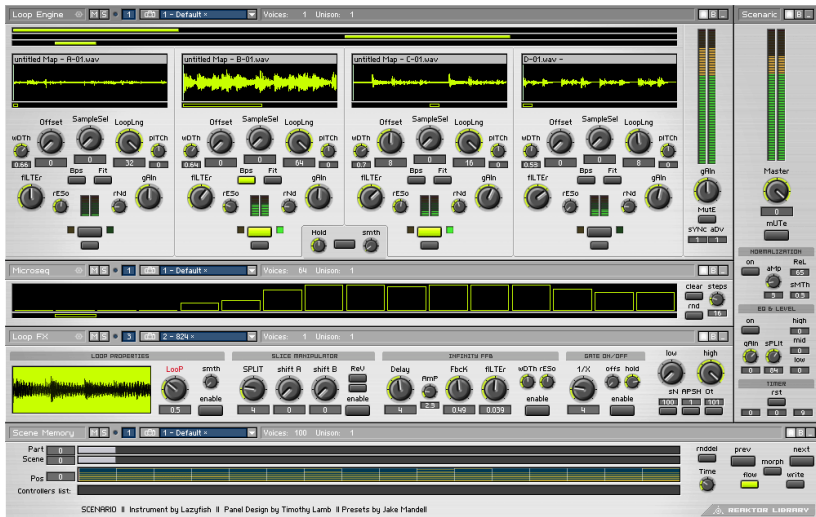
The Hold button repeats the current sequence step when depressed. This is perfect for creating breaks to heighten drama in a performance. Porta introduces a small amount of pitch glide to the samples. The Rev button creates a sort of reverse playback. The Legat button forces consecutive sequencer steps with the same gate value NOT to retrigger the envelope. This causes some samples not to sound on their sequencer steps, stripping back the beat.

## **Morph**

The Morph pad allows you to make overall changes to goBox's settings with one easy to use XY control. It can be used to momentarily transform your sequence and/or sample settings, or to add small amounts of almost random change to your performance.

The Y-axis morphs the sequence parameters, while the X-axis handles the sampler parameters. Clicking on the pad introduces the Morph effect at that position for as long as the mouse button is held down. Settings return to their previous positions when the mouse button is disengaged. Clicking the M On button holds the Morph setting as long as the button is on.

# Scenario



*Instruments: Lazyfish; Presets: Jake Mandell; Panel Design: Timothy Lamb*

Scenario is a complete live-performance environment with realtime timestretching, performance-oriented effects, and memory and instant recall of thousands of scenes. Have you ever played live with a computer and been frustrated that you don't have enough control over the audio? Or - at the opposite end of the spectrum - have you ever been performing and been overwhelmed with the possibilities? Scenario solves both of these seemingly opposing problems in an elegant and ingenious way.

First, let's take a look at what Scenario is, and then we'll see why it's an ideal live-performance tool. Scenario consists of a Loop Engine that contains four identical time-stretching loop players that fits each loop to the system tempo. Each loop player lets you perform in realtime with the loop length, loop start, pitch, and animated filter. A powerful performance-oriented effects block is also included, which lets you do things never possible before since the Scene Effects and Loop Engine are controlled by the same sample-accurate system clock. For instance, you can reshuffle and even reverse the music, rhythmically gate the sound exactly on the beat, create super-tight loops, and more. The Miniseq

instrument can shuffle and reorder each loop. Just draw a pattern into Miniseq, then activate the shuffle function by clicking the small button in the Loop Engine at the bottom of each loop player.

And now, the kicker: All of Scenario's settings can be stored into a "scene". You can store more than sixteen thousand scenes in the built-in Scene Memory. Scenes can be recalled sequentially for "one-touch" performance, but you still have full control over all parameters within each scene. That's what makes Scenario so revolutionary - it gives you the power to lay out an entire live set in advance so you're assured that things go smoothly. Not only you determine when to advance to the next scene (this would be quite boring!), but you also have complete control over every scene. You can adjust the pitch of the loops, the loop length, the filtering, and the effects - every single parameter on the screen. When it's time, switch to the next scene, and - Bam! - it comes in perfectly on the beat, every knob perfectly recalled, including the effects.

## Quick Start

For an easy tour of what Scenario can do, take a look at the demo live-set that is included in it. Start the system clock from the toolbar - you should hear some music now - and then after a few bars, press the Next button on the Scene Memory instrument on the bottom. You just advanced to the next scene. Keep on pressing Next every few bars - notice that every single parameter of Scenario is updated, including all effect settings. Feel free to play with any of the knobs to see how they influence the sound. Notice that each effect must be first enabled in order to hear it - simply click on each effect's Enable button. You can only hear a cell when it's active - to activate or mute a cell, simply click on the button below that cell's level meters.

## Structure and Signal Flow

The Scene Loop Engine makes all the noise. It consists of four identical loop players and a handy bar/beat counter so you can always keep track of where you are without having to count: The three marching bars on top denote the number of 4-bars, bars, and beats. Every sixteen bars it's reset to zero. The summed output of all four loopers goes through the Scene Effects, and then to the output.

Let's take a look at an individual looper in the Scene Loop Engine. Each looper gives precise control over the loop length, pitch, and filter of the loop. The tempo of the loopers are all synchronized to the master system clock, whose tempo is settable in the toolbar. The audio will stretch or compress in realtime to fit at the given tempo. The time stretching algorithm is very high tuned for rhythmical material. SampleSel selects which loop to play. Each looper can hold a Map of 128 different loops, and SampleSel simply selects the desired loops. See the section below, "Loading Samples," for more information on how to easily load your sounds into "Scenario." Offset controls the sample start time, in sixteenth notes - where the sample will start from at the downbeat. LoopLng sets the loop length, also in sixteenth notes. Pitch lets you modify the pitch of the sample without changing its time. Wdth, Filter, Reso, rnd, and Bps all control the looper's internally-animated filter. The filter is a bandpass consisting of two two-pole filters (a high-pass filter that cuts the lows, and a low-pass filter that cuts the highs). The width between the two filters is controlled by Wdth, and the combined resonance of both filters is set by Reso. Filter sets the center frequency of the two filters, and Bps bypasses the filter completely. Rnd controls how animated the filter is. If set to zero, then the filter will stay at the frequency set by Filter. But if Rnd is high, then the filter will slowly move around to create movement. Fit changes the pitch of the loop to "fit" it to the current tempo, just like a turntable changes the pitch when it slows or speeds a record up. Finally, two more controls: Gain controls the volume of the looper, and the unlabelled button underneath the level meters activates (on) or mutes (off) the track. The small button underneath this activation button activates the retrigger sequencer (Miniseq).

The output of the Scene Loop Engine gets fed into the Scene Effects block. Scene Effects consists of four effects (Loop, Slicemanipulator, InfinityFFB, and Gate), routed serially, and a master two-channel crossfader. The first effect is a looper - it simply loops the incoming audio, but it does it very precisely. The Loop knob controls how long the loop will be (in bars), and the Loop button activates and deactivates the loop. Slicemanipulator is an extremely unusual, yet performance-friendly effect. It divides the audio into "slices" then lets you rearrange and even reverse the audio in time. Split controls how many slices there will be per bar. Shift A moves even slices forward in time (in sixteenth notes), while Shift B moves odd slices forward. Rev A reverses the even slices, while Rev B reverses the odd slices. Enable turns the effect on or off.

InfinityFFB is a delay effect with a finely-tuned feedback path. This means that you can freely perform with the Delay, Feedback, and Filter knobs without risk of overloading or running into distortion. Delay sets the delay time in sixteenth notes, while Smth controls how smooth changes in delay time will be. Fback sets the amount of feedback within the delay, while Filter controls the filter frequency of a filter built into the delay feedback path. Wdth and Reso also control that filter - the structure of this filter is also a dual-two pole filter like the filter for each looper channel. Enable turns the effect on or off. The last effect, Gate, rhythmically gates the audio with the frequency set by Freq (in sixteenth notes). Offs sets the latency of the gate - a higher offset value lets more audio through before the gate kicks in. Hold controls how long the gate will be open for - at small values hardly any audio will pass through. The usual Enable button turns the effect on or off. Finally, we have a two band crossfader as a master dry/wet control. We have independent control over the high and low frequency ranges. Low controls the mix of the low frequencies, and High controls the highs. When the crossfader is fully up, the effect is active for that frequency range.

The third instrument is the Miniseq, where you can draw in a retrigger sequencer for the Loop players. All loop players share the same sequence, but the retrigger sequencer can be turned on and off for each Loop player independently. The small bottom-most button underneath each Loop player activates the retrigger sequencer. The bottom-central Hold and Smooth knob adjust the hold time and smoothness (envelope time) of the retrigger sequencer for all loops.

The fourth instrument in Scenario is the Scene Memory, the brain that doesn't make any sound. This brain stores all scenes into a giant table. Don't be frightened by the complex-looking table - it's there just to show if a scene has information in it or not. The current scene is shown in two ways: by the read-only PartN and SceneN numerical displays to the bottom left, and by the two draggable rectangles above the table. You can change the scene in two ways: By pressing Prev/Next to go the previous or next scene, respectively, or by dragging on one of the two rectangles above the table to go directly to a specific scene. There are two settings to adjust the size of the table: Parts and Scenes. The total number of scenes is  $\text{Part} * \text{Scenes}$  - each part can contain from 16 to 128 scenes. Since there can also be from 16 to 128 parts, you can adjust the size of the scene memory from 128 ( $16 * 16$ ) to 16,384 ( $128 * 128$ ). Generally, depending on the style of music, 128 scenes should be sufficient for a

30 to 60 minute live set. Working with high numbers of scenes has the disadvantage that the draggable part and scene-selector rectangles (above the scene memory table) become very small. The Write button (all the way on the right) does exactly what you'd expect - the current positions of every parameter is magically written into the table at the current scene position. The Flow button is an important feature in organizing your scenes - when Flow is on, anytime you go to a new scene (either with the Prev/Next buttons or with the draggable scene and part selectors above the table), then the new scene will be immediately active. This is probably what you want for live performance, but Flow can be turned off for when you're composing your set. With flow off, when you go to a new scene, it does not load. This is very useful for copying and pasting scenes from one location to another.

## **Loading Samples**

The most convenient way to load samples into Scenario is with the Browser. Simply double click on the visual display of the sample to enter the sample map editor. If you want to clear the map to start fresh, simply double twice on the delete button. Using drag and drop, just load your samples into the sample map editor. You probably don't want to use the graphical keyboard sample map editor unless you're very careful to align one sample per key. The text-based sample map editor will take care of this automatically. Please see your user's manual for more helpful information about loading samples and using the sample-map editor.

Because of Scenario's advanced time-stretching algorithm, all samples must first be analyzed before they can be used. When you drag the samples into the sample-map editor, it will first analyze them for you. You will then be asked if you want to save the analysis data into the samples - it's recommended to do so, so you don't have to analyze them again the next time the live set is loaded.

## **Tips and Tricks**

There are many tricks you can use Scenario for. No one every said you had to use it live, either. You can also use it in the studio to create complete tracks with.



Filter-splitting - create one track out of four. You can take advantage of Scenario's filters to combine several tracks into one. Like the bass drum from one track and the synth from another and the high hat from a third? Use Scenario to create track mutations. Simply load a different loop into each of Scenario's loop players in the Loop Engine, and tweak the filter settings to focus on a different frequency range in each loop.

Make good quality loops. Loops have to be cut exactly to 1, 2, 4, 8, 16, etc bars in order for the beat looper to play them correctly. The loops can be at any tempo, however, and Scenario will automatically fit all loops to the same tempo. All common sound editing programs let you bounce audio with exact bar sizes, so check your manual for detailed information.

Make good scenes, then freely improvise. For an effective live performance, it often pays to spend a lot of time making good scenes, and having a good flow between the scenes. You can copy and paste scenes in the Scene Memory simply by activating a scene by going to that scene with Flow enabled, then by turning off Flow and going to the position you want to write the scene to. Press Write, and you just copied and pasted a scene from one part to another. When performing, you can be assured that if any live improvisation gets out of hand, you can just go to the next scene and everything will be on track again.

Use the two-band crossfader in the Scene Effects. The two-band crossfader is a well-kept secret for music played on a powerful PA system. By only effecting the high frequencies, the bass stays clear and doesn't get muddy.

MIDI control. Any of the Scenario's parameters can be MIDI controlled. Due to its complex internal structure (using internal OSC communication between the three instruments), you can't use MIDI learn on the parameters, or they will not be remembered by the Scene Memory any more. If your MIDI controller is programmable, such as a Peavey PC-1600 ([www.peavey.com](http://www.peavey.com)) or Bitsream ([www.wave-idea.com](http://www.wave-idea.com)), then it's better to program your MIDI controller instead of using MIDI learn.

Saving live sets. Each live set should be saved as its own ensemble so it can be recalled with all samples and scene memories intact. Simply use Save As... and give your live set a name.

# Effects

## Analogic Filter Box



Instruments and Presets: Erik Wiegand; Panel Design: Ian Warner

The Analogic Filter Box sandwiches a rich and meaty distortion unit between two hearty analog-style filters to create a sound-shaping tool for every appetite. Juicy modulation is also provided on the side: An envelope, LFO, and envelope follower can be freely routed to the most important filter and distortion parameters. Analogic Filter Box can handle everything from fat disco-loop tweaking to full-on mangling of any sound source imaginable.

### Quick Start

Select Preset 1 (Default) - this resets the Analogic Filter Box to its default settings, with the sound passing through the effect unchanged.\* You must activate either filter 1, 2, or the distortion in order to hear the effect. To turn on filter 2 (the one after the distortion), for instance, simply select a filter type from the pull-down menu in the bottom-right corner of the Filter 2 macro. Now, by moving the Cutoff P and Reson knobs, you should immediately hear your freshly filtered sound. Filter 1 and Distortion are activated in the same way.

To get some motion into the sound, we'd want to take advantage of the Analogic Filter's copious modulation options. To modulate the cutoff of filter 2 by the LFO, for instance, select "Lfo" in the 2 P menu in the filter modulation section. Make sure that Filter 2 is activated - you may want to select Patch 1 (Default) again and activate Filter 2 by hand. Underneath the modulation source menu there's a small number box where you can click and drag to specify how much you want the LFO to modulation the cutoff of the second filter by. For a very obvious effect, just click and drag until the number box shows 100. Note that you can also enter negative numbers in the modulation amount if you want the LFO to decrease the filter frequency. One last step - you must activate the modulation by clicking on the small button next to the modulation amount. Now - assuming filter 2 is activated and audio is running through the effect - you should hear the LFO controlling the filter cutoff.

## Structure and Signal Flow

The signal is routed from input to Filter 1, to the Distortion, to Filter 2, and then out.

Both filters offer multiple operation modes. For the first filter, you can choose between Lowpass, Highpass, Bandpass, Peak EQ, and Notch filters, with a choice of a 12 or 24 dB/octave slope per filter type. The second filter is designed for shaping the sound after the distortion, so it offers four different lowpass filters, three bandpass filters, and a bandpass/lowpass combo. Both filters were very carefully designed to produce warm, analog sounds even at extreme resonance and cutoff settings.

Beside the normal cutoff frequency control available in both filters, Filter 1 also provides fast modulation of its frequency by an additional oscillator (which can itself be modulated by the LFO, envelope, or envelope follower!).

The Distortion section between the two multimode filters also features multiple modes of operation. While the clipper mode provides a relatively harsh distortion sound and the saturator mode results in warm overdrive, the several wrapping modes (marked by the name of the waveform used for wrapping) produce unique sounds from subtle to extreme.

An additional quantize mode converts the incoming signal into a step waveform, for familiar bit-reduction effects to mimic the character of vintage samplers, for instance. A visual display of the distortion function helps to see what's going on inside.

## Modulation

Analogic Filter offers six modulation sources (A built-in LFO, envelope follower, envelope, MIDI note pitch, modulation wheel, and pitch bend wheel). The modulation sources and the flexible matrix signal routing system at the bottom of this effect transforms it into an incredibly powerful machine.

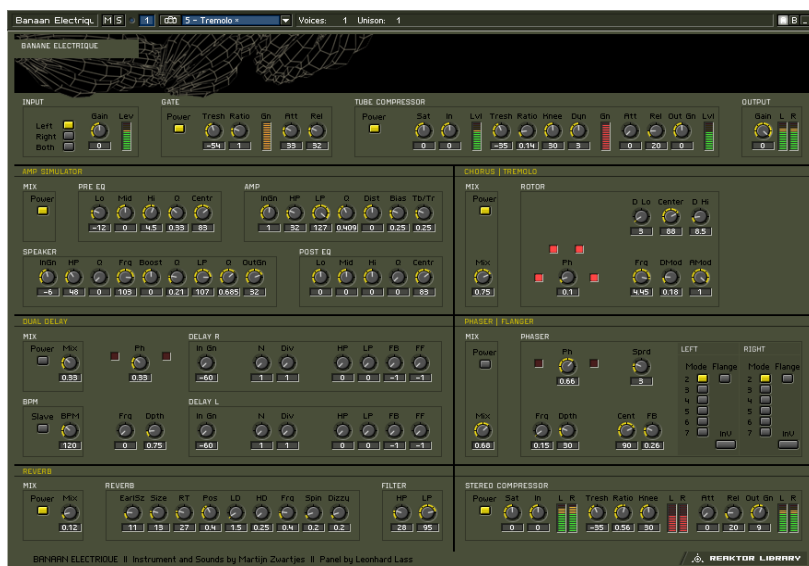
The modulation signal routing system provides a source selector for each parameter to be modified; among those modifiable parameters are the cutoff frequencies and resonance settings of both filters and the distortion amount and symmetry control of the distortion section. In addition to gate information from MIDI note-on events that can even be used to trigger the filter, you can also use the pitch and mod wheels as modulation sources.

An internal LFO, envelope follower, and auto-trigger envelope can add movement to the sound without the need any external MIDI controllers. The LFO offers different waveforms and can also be synchronized to the global tempo or MIDI clock (the small Unit knob syncs the LFO to MIDI clock and sets the musical note-units that are shown under Freq). The Envelope Follower calculates its modulation amount from the incoming signal: At high levels there is a high modulation level, and at low levels it's low. The Interval knob controls the response time to fast level changes. Use the cutoff controls of the internal highpass and lowpass filters to select a specific frequency band of the incoming signal to trigger the envelope follower. The Envelope is a standard attack-decay-sustain-release envelope generator, triggered by MIDI note on events. However, an additional auto-trigger feature allows it to be triggered by the incoming audio, settable with the Tresh slider.

It's even possible to combine any two MIDI controllers to make one dependent on the other - for instance, to have the amount of LFO modulation dependent on MIDI pitch. You can define custom mix modulation combinations in the Define Mix 1 and 2 areas as the bottom of the instrument.

\* You can play a sound through the built-in loop-player, through the realtime audio inputs, or you can process audio in realtime by using Reaktor as an effect plugin. Please check your Reaktor or Reaktor Session user's guide for helpful information.

## Banaan Electricque Amp Simulation



*Instruments and Presets: Martijn Zwartjes; Panel Design: Leonhard Lass*

Banaan Electricque is a sophisticated guitar and bass amp simulator with built in effects. It's like having a pedal board full of vintage pedals and a vintage amp, with the added advantages of not needing to cable anything together or risk electrocuting yourself by running too much current through the amp. You can still get shocked by its lush sound, however. Naturally, Banaan is happy to eat whatever types of sounds you feed it, whether they're vocals, drum loops, synths, or scallops.

## Getting Started

Banaan Electrique was designed to be so easy that even guitar players can use it! Just play some audio through it and let a rip.\* Check out the presets to see the variety of tones that Banaan can achieve.

## Structure and Signal Flow

The audio runs through Banaan Electrique linearly:

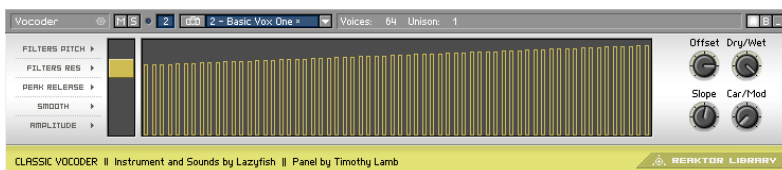
Input ⇒ mono gate ⇒ 3-band EQ ⇒ Compressor 1 ⇒ Amp Simulator ⇒ Phaser ⇒ Rotor ⇒ Dual Delay ⇒ Reverb ⇒ Compressor 2

The incoming signal is converted to mono, amplified, gated, equalized and compressed to control the input's amplitude level and to enhance those sounds wanted to be in the recording, getting rid of any unwanted noise. Then, a guitar amplifier is simulated, including distortion, overdrive and filtering. Its output signal is sent to a rotor module, placing the mono sound in the stereo field dynamically. This signal is then routed to a phaser effect and, afterwards, to a stereo echo unit whose delay times can be synced to master tempo or MIDI clock. A high-quality reverb enhances the spatial sound once more, feeding its output into the final compressor.

Not all modules have to be active at the same time - in fact, it's a good idea to turn off any modules that you're not using to save CPU. You can turn modules on and off with their respective Power buttons.

\* You can play a sound through the built-in loop-player, through the realtime audio inputs, or you can process audio in realtime by using Reaktor as an effect plugin. Please check your Reaktor or Reaktor Session user's guide for helpful information.

## Classic Vocoder



*Instrument and Presets: Lazyfish/Classic 2-VCO Synth: NI*

The Classic Vocoder was designed to faithfully emulate the well-known tones of singing robots made popular in the seventies. The instrument combines a vocoding engine, a vintage-type synthesizer, and a four-band dynamics processor for a warm, smooth sound.

## **Quick Start**

The audio input is vocoded with the built-in synthesizer. Play some audio into the vocoder.\* You don't have to sing into it - the Classic Vocoder also gives great results with drum loops or other sounds.

## **Structure and Signal Flow**

Audio input 1 is the modulator, and is vocoded with the Classic 2-VCO synth, which is the carrier. The entire signal flow is mono. If you are using a stereo signal, only the left channel will be used. The output of the vocoder is fed into the four-band normalizer to smooth-out the sound and remove any uncomfortable signal peaks that could come with vocal sibilants or drum transients.

## **The Vocoding Engine**

The vocoding engine consists only of four controls and a large graphic display of the frequency bands. Using an elegant function selector, you can adjust the important settings for the bandpass filters that divide the carrier and modular into multiple frequency bands. The pitch, resonance, peak release, smoothness, and amplitude of the filters can be set. By dragging the vertical bar on the left side of the graphic display, you can choose which settings to adjust. For instance, to modify filter pitch, bring the bar up to filter pitch, and then the "offset" and "slope" knobs will adjust the filter pitch settings across the bands.

The "dry/wet" and "car/mod" knobs are always active, no matter where the parameter selector is. Dry/wet mixes between the pure input signal and the vocoder output, while car/mod mixes between the audio input and the built-in synth.

## Classic 2-VCO

A simple analog-style synthesizer with two oscillators, one filter and filter envelope, one LFO, and one amplitude envelope. Each oscillator is limited to either a pulse or a sawtooth waveform.

## Four Band Normalizer

A four-band normalizer smooths out the sound before the output. The Split knob controls the filter frequencies that split the incoming audio into four bands - the approximate frequencies splits are shown on top of the instrument with the three horizontal meters. Each band provides four controls and before/after meters. Each band gives you control over: Limiter on/off (Lim), volume of the band (Amp), and the transient response of the limiter (Smth, and Rel).

## What is Vocoding?

Sonically, Vocoding uses the characteristics of one sound to control another. To achieve the popular robotic-singing effect, a voice (technically called the modulator) is vocoded with a constant sound, such as a synth or string sound (the carrier). The frequency content of the voice is split up into many different bands - the number of bands has an obvious impact on the sound, with fewer bands leading to more synthetic voices, and higher bands make the voice easier to understand. You can adjust the number of voices of the vocoder instrument to change how many bands are used. Up to 128 voices (bands) are possible. All changes in number of bands are immediately shown in the graphical display.

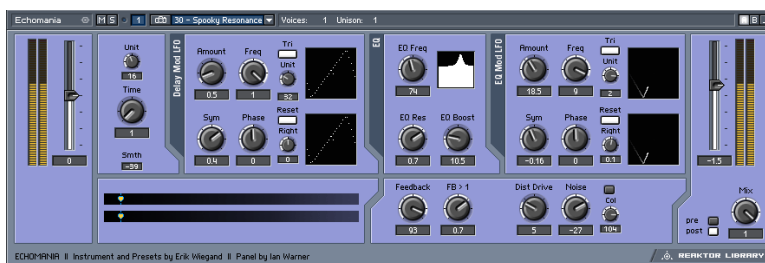
The amplitude of each frequency band of the voice is linked to the frequency bands in the string or synth sound. The re-shaped bands of the carrier signal are mixed together, providing the output signal of the vocoder.

If you're interested in vocoding, you may also want to do check out another NI product - the VOKATOR. The VOKATOR features vocoding up to 1024 bands, a built in synthesizer and granular sampler, and many high-end vocoding features. [www.ni-vokator.com](http://www.ni-vokator.com).



\* You can play a sound through the built-in loop-player, through the realtime audio inputs, or you can process audio in realtime by using Reaktor as an effect plugin. Please check your Reaktor or Reaktor Session user's guide for helpful information.

## Echomania



*Instrument and Presets: Erik Wiegand; Panel Design: Ian Warner*

Echomania is an advanced and spectacular-sounding delay box that excels at tight, tempo-synced rhythms. It includes two LFOs to modulate the delay time and built-in EQ. The delay time is handily synced to the global MIDI clock, for creating stretchy rhythmic effects. A drive and noise circuit combined with a feedback offset control recreate vintage sounds. Get dubbing!

## Quick Start

Play some audio through Echomania.\* Flip through the snapshots and discover the gritty (or crystalline) depths of Tapedelay.

## Structure and Signal Flow

The delay time is set in 16th note increments, synced to the global MIDI clock. The Unit and Divisor windows, to the right of the Time control, allow you to fine tune the beat division and create different rhythmic feels -- to be precise, the numerator divided by the denominator scales the delay time fraction.

There is also an LFO, with blendable sine/triangle/square/slow random waveshapes that is hardwired to modify delay time. You can set the amount of time modulation with the Amount knob. The Freq knob controls the rate of the LFO in 16th notes. The Unit knob divides the speed of the global clock by the Unit amount. For example, if the Unit knob is set to "6", then the LFO Freq amount will be in 16th notes. The Width control morphs between sine and pulse-like, or, if the Tri button is engaged, triangle and saw tooth waveshapes. The starting phase of the LFO can be adjusted positively or negatively using the Phse knob. If you engage the Snc button, the right-side LFO can be phase-offset from the left side by the amount set with the Right knob. This can create a variety of wozy stereo spinning and phasing effects.

The EQ module processes the delayed signal. It is essentially a parametric EQ that contains an LFO identical to the one in the Delay module. You can create synchronized filter sweeps, fizzing hi-frequency delay tails, and all manner of dubby effects by boosting and modulating select frequency bands. The Eq Res control lets you dial in the peak width of the frequency, while EQ Boost lets you crank it up.

Tapedelay's Feedback module provides an offset control, labeled FB > 1, which boosts and shapes the feedback signal, making it seem to get louder and louder (but without degenerating into uncontrollable noise).

The Mixer lets you add a tape saturation-like Dist Drive effect and Noise to give everything that just-pulled-out-of-the-closet feeling.

The Dry Wet section allows you to balance the amount of dry and delayed signal. You can also use the Tap buttons to select whether the delay tap comes before or after the saturation/noise circuit.

\* You can play a sound through the built-in loop-player, through the realtime audio inputs, or you can process audio in realtime by using Reaktor as an effect plugin. Please check your Reaktor or Reaktor Session user's guide for helpful information.

# Flatblaster Multiband Dynamics



*Instrument: Martijn Zwartjes, Presets: Jaap Wajer; Panel Design: Ian Warner*

Flatblaster is a high-end finalizing and multiband dynamic shaping tool. Flatblaster combines four frequency-specific compressors with a full-spectrum peak-limiter. It is an excellent final-step mastering plug-in, but it can also be used while mixing since it doesn't introduce any delay to the signal. Each of the compressors has a saturator, so you could saturate just the upper-mid frequencies, for instance, without muddying the bass. It also makes an excellent de-esser and sibilant reducer.

## Quick Start

Even though Flatblaster gives control over many sound-shaping parameters, there's no need to be intimidated by its complexity. A full range of presets shows off its capabilities and gives good starting points for tweaking the effect for your sound. Simply play some audio through Flatblaster\* and step through the presets. Try experimenting with muting, soloing, and bypassing each individual band so you can carefully hear what the 'blaster's doing. Be careful when adjusting the saturation of each band! The sound can potentially get very loud if you don't first reduce the make-up gain (the knob labeled Gain to the right of the Sat knob).

## Structure and Signal Flow

The input signal is divided into four bands - the three crossover frequencies are adjustable independently. Each frequency band is processed by an independent, identical compressor. Each band can be muted, soloed, and bypassed (no compression). The signal is summed before going through a full-band peak limiter, which can also be independently bypassed. The master bypass for the effect is located to the right of the input meters, above the crossover settings.

## Frequency-Specific Compressor

Each of the four compressors are absolutely identical. In fact, they have to be! If they weren't, then unwanted phase shifts could creep in. Each compressor gives control over saturation, saturation makeup gain, threshold, compression ratio, adjustable knee, attack, release, and output makeup gain. Note that the Ratio has to be higher than 0 for the compressor to have any effect - at a Ratio of 1 (maximum), the compressor acts as a limiter. The red meters show the amount of peak reduction. The Attack and Release knobs control how the compressor responds to transient signals.

## Full-Band Peak Limiter

The peak limiter effects the full frequency range of the audio, after each of the four frequency bands has been compressed separately. The Threshold slider controls when the peak limiter will start working. With Threshold at 0, the peak limiter will have no effect. For mastering, it's recommended to have the Threshold set to around -3 or -4 dB. Severe Threshold settings will lead to pumping, which may or may not be desirable. The Attack and Release knobs control how the peak limiter responds to transient signals. The final Peak slider sets the output level of the signal. When Peak is set to its maximum 0 dB, then the audio will be as loud as possible. There's really no reason to ever set Peak lower than this, unless you needed to ensure a certain amount of headroom.

\* You can play a sound through the built-in loop-player, through the realtime audio inputs, or you can process audio in realtime by using Reaktor as an effect plugin. Please check your Reaktor or Reaktor Session user's guide for helpful information.

# Fusion Reflections



*Instrument and Presets: Erik Wiegand; Panel Design: Ian Warner*

Fusion Reflections is a delay-based effect that can create early reflections, shimmering choruses, fluttering delays, and even ambient reverbs. Two distinct diffusion engines are chained together to create an extremely wide range of effects. Each finely-tuned diffusion engine consists of four stereo modulation delays and an innovative graphical display that shows the actual delay time for each delay. Just five controls control the core parameters of each diffusion engine.

## Quick Start

Play some audio through the effect\* and select Preset 1 Long Decay Echo. This preset only uses the Chor Fusion diffusion engine. Play with the Diff Dly knob - note how the graphic display constantly updates to show you the current delay times. Stop the incoming audio to listen to the sound's decay. Play with Dly Mod and Speed to see what effect they have on the sound and the graphic. Be sure to check out all the other presets to see how versatile this effect can be!

## Structure and Signal Flow

The sound passes through two diffusion engines serially. It's important to note that the two engines are similar but not identical - they each offer unique sound-shaping capabilities. The first diffuser, Chor Fusion, is the simpler of the two and is designed for early reflections, choruses, and

atmosphere. It offers a high and low shelf EQ, with a graphical display of the EQ curve. The second diffusion engine, Echo Fusion, adds a feedback delay before the diffusion delays. The diffusion delays in the Chor and Echo engines are identical. Echo Fusion is perfectly tailored to late reflections, long delays, and long reverbs. A highpass filter (HP) cuts the low frequencies after the delay and before the diffusors, while a lowpass filter (LP) can reduce the brightness at the last step. The input to the Echo Fusion engine can be switched between the dry signal, and the signal coming out of Chor Fusion.

In the Mixer section the signals of Chor Fusion> and Echo Fusion are combined and mixed with the dry signal. Each section can also be switched on or off to save CPU.

## **Diffusion Delays**

Both the Chor and Echo Fusion engines contain identical diffusion delays. As mentioned above, Echo fusion adds a single feedback delay before the diffusion delays, but the operation of the diffusion delay section is identical per effect. Each diffusion delay consists of four stereo modulation delays and an innovative graphical display that shows the actual delay time of each delay. The main delay time is controlled by the Diff Dly knob. The Dly Mod knob controls the amount of delay time modulation by the internal LFOs. Speed controls the speed of the LFO. Stereo sets the stereo spread of the delays, and Diffusion sets the inaccuracy of the delay times. The sum effect of these five knobs is graphically shown in the display underneath, where each "pendulum" represents the delay time of a single delay.

\* You can play a sound through the built-in loop-player, through the realtime audio inputs, or you can process audio in realtime by using Reaktor as an effect plugin. Please check your Reaktor or Reaktor Session user's guide for helpful information.

# SpaceMaster



*Instrument and Presets: Martijn Zwartjes; Panel Design: Leonhard Hass*

The SpaceMaster series of reverbs breaks new ground in reverb models for Reaktor. Spacemaster uses two different Diffusion modules to achieve stunningly convincing room sounds. And to fully exploit SpaceMaster's lushness, there are three versions - SpaceMaster stereo, SpaceMaster Quad, and SpaceMaster 5.1 Surround.

## Quick Start

To really get a feel for the kinds of lush atmospherics SpaceMaster can provide, it's a good idea to either hook a beat looper or external sound source up to it, or, to utilize it in plugin mode in your favorite audio sequencer. Stepping through the presets should give you a good impression of the kinds of real and imaginary spaces SpaceMaster can emulate. Adjusting the controls of the Early and Late Diffusion modules will have the most effect on the sound, and will give you an idea of how the two main components interact in creating ambiances - especially since they can be arranged in serial or parallel signal paths.

## Structure and Signal Flow

This guide will use the Stereo SpaceMaster ensemble to outline the various controls for shaping the reverb signal, since most of them are the same among the three types of SpaceMasters. Read on below to learn about the specifics of the Quad and Surround flavors of SpaceMaster reverb.

## SpaceMaster Stereo:

SpaceMaster Stereo is composed of a PreDelay module, an Early Diffusion module, a Late Diffusion module, and a post EQ. The Diffusion modules can be combined together to create a complex impression of space. By adjusting the balance of the Early and Late diffusion modules, you can precisely move the origin of the reflections from near to far (or front to back), making SpaceMaster perfect for Surround mixing situations in which a truly room-filling reverb can be created.

The Input section, at the far left, allows you to trim the input gain to avoid overloading the audio signal. The input is next processed by the PreDelay. Use this to add an initial delay to the wet signal. You can also use the L/R offset knob to add some perceived stereo width by altering the delay time of the left-side single delay module. You can bypass the PreDelay with the Byps button.

Next in the signal path, the Early Diffusion module, which is actually a series of up to 12 diffusion delays, provides the near reverb processing. The Size control allows you to determine the range of space that the close reflections will be generated by. It changes delay time in milliseconds. The Diffusion knob lets you adjust the density of the reverb signal. To further adjust the reverb depth of the Early Diffusion module, the Mode switch allows you to select 6 or 12 diffusion modules. Watch your CPU load carefully to make sure your computer can handle the strain of processing with 12 (or 24) diffusion modules. The Damp knob controls the frequency of a 1-pole low pass filter for attenuating high frequencies. As in the PreDelay section, clicking the Byps button will take the Early Diffusion module out of the signal path.

The Routing switch, located between the Early and Late Diffusion modules, lets you determine how the input signal will be routed through the two modules. The Ser button engages Serial mode, where the Early Diffusion module is simply routed directly into the Late module. The Par switch engages Parallel mode, allowing the Early and Late modules to maintain separate signal paths, until they are mixed at the EQ module. You can also use the Early/Late knob to balance the amount of early and late reverb signal. This is a great way to change the perception of location within a space, by shifting between the early and late reverb sounds (to make it seem like a sound is moving around inside of the "room").



The Late Diffusion module provides the capabilities for creating a larger and more richly defined space. The Size and Diffusion knobs work the same way as those in the Early section, but there are a few new options. The RT knob controls a feedback loop, allowing you to stretch out the apparent reverb time, or the time it takes for the echo to return to the point of origin. There are also controls for high and low shelving EQs, called Hi Damp and Lo Damp, respectively, that allow you to shape the frequency range of the reverb signal. You can modulate the delay time and apparent position of the reverb signal by using the modulation controls. The Spin knob adjusts the amplitude of a sine wave LFO, while the Dizzy knob controls amplitude for a Slow Random LFO. You can create complex modulations by adjusting the balance of sine wave and random LFOs. The Freq knob controls the rate of both LFOs together. As in the Early section, 6 or 12 diffusion modules can be selected with the Mode switch. This section can also be bypassed with the Byps button.

Both the Early and Late Diffusion sections feed into the EQ. The EQ consists of stereo low shelf, parametric, and high shelf filters. Starting at the left, the Lo knob lets you control attenuation or boost of the low frequency set with the Frq immediately above. The Mid knob controls the parametric EQ band. The Frq and Q knobs above it let you adjust the frequency and bandwidth of the parametric. The Hi knob controls attenuation of the high shelf filter. Use the Frq immediately above the Hi knob to adjust the frequency. As in the other modules, the EQ can be taken offline with the Byps button.

Finally, the mixture of wet and dry signal can be adjusted to taste with the Mix knob in the output section.

## SpaceMaster Quad



*Instrument and Presets: Martijn Zwartjes; Panel Design: Leonhard Hass*

The SpaceMaster Quad ensemble works in the same way as the stereo version, but it splits the reverb signal into four discrete outputs for using in a four-speaker system. The signal for the forward speakers is referred to as the Front signal, while the signal for the rear speakers is referred to as the Surround, or Sur, signal. In the Input section, you can adjust the gain for Front and Sur signals to determine how much signal sources reaches each signal chain.

The PreDelay section is simplified from its stereo brother by use of an XY pad to finely adjust Front and Surround L/R offset. The overall PreDelay time can be controlled by the Time fader.

The Early Diffusion section is also simplified, offering only Size and high frequency Cut controls for both Front and Surround diffusion signals.

The Late Diffusion module is effectively the same as the one found in SpaceMaster Stereo. Refer to the description above the learn how it works.

The Mix section allows you to balance the Front and Surround signal amount and to adjust the wet/dry mix for each.

## SpaceMaster 5.1 Surround



*Instrument and Presets: Martijn Zwartjes; Panel Design: Leonhard Hass*

The Surround version of SpaceMaster is effectively the same as the Quad version, except that it adds a Center channel. The PreDelay XY pad adds control for the center channel. SpaceMaster Surround also adds an

Output Gain section that allows you to use precise metering to control the relative volumes of the Center, Sub, Front, and Surround reverb signals. The balance of Early/Late signal can also be adjusted for each channel using the Mix module, which has familiar controls for wet/dry levels. Finally, SpaceMaster Surround does not include the EQ module found in the Stereo and Quad versions.

## Spring Tank



*Instrument and Presets: Tim Schwerdtfeger*

Spring Tank is not your typical room simulator - instead, it pays homage to the trashy, unrealistic spring reverbs of the past. While Spring Tank isn't exactly a physical model of a spring, it goes a long way toward recreating the spring reverb characteristics: dull, transducer-saturated, and boingy, with the familiar nonlinear resonating decay.

## Quick Start

Spring Tank is meant to be an experimental effect, rather than a realistic one, play some audio through the effect\* and experiment! It allows control over the elements of the "spring's" morphology, so you can design your own spring.

## Structure and Signal Flow

The input source is amplified or attenuated with the Level knob at the far left of the instrument. High levels here can introduce transducer saturation (which is not such a bad thing). The signal is then fed into the spring tank. Here, you can adjust the spring's physical characteristics. The Damp knob sets high frequency damping for the spring. Turning the knob clockwise increases the damping. Stiffness, which mimics the flexibility of the spring, gives brighter, more resonant sounds at higher

settings. The Shape knob allows you to crossfade between a round spring shape to the left, and a rectangular one to the right. The round shape emphasized a ringing sound, while the rectangular shape creates less ringing but a more diffuse sound.

The Thickness of the spring determines the virtual diameter of spring winding, and therefore the overall length of the spring. Longer spring settings result in reduced brightness. The Length setting reflects global length. Turn this up to increase the shattering decay sounds. The Decay knob changes the length of the decays. It alters the amount of feedback in the system.

Switching the Mono button off engages an additional short delay in one channel, resulting in the simulation of stereo.

The Suspension section allows you to mimic the type of spring suspension in the system. When you turn the knob to the left, you increase the "softness" of the suspension material. This will lengthen the decay rate of lower frequencies. What this is actually doing is adjusting the cutoff of a high pass filter in the feedback loop. Suspension will therefore also have an effect on the sound of the Transducer saturation. The Color knob lets you adjust a global tone control. Turning this clockwise results in a brighter sound, and vice versa.

The Level knob controls the volume of the wet signal. You can adjust the balance of wet signal to dry signal with the Mix knob. Finally, you can bypass the effect all together by clicking the On button.

\* You can play a sound through the built-in loop-player, through the realtime audio inputs, or you can process audio in realtime by using Reaktor as an effect plugin. Please check your Reaktor or Reaktor Session user's guide for helpful information.

## Two Knees Compressor



*Instrument and Presets: Erik Wiegand; Panel Design: Ian Warner*

Two Knees Compressor is a simple compressor with an important quirk - it has two separate adjustable thresholds and ratio controls. 2-Knees Compressor can perform as a precisely accurate compressor/limiter, a distorting sound shaper, a transient modifier, or anything in between. It includes a compression curve display to show the relationship between the amplitude of the input signal and the amplitude of the processed signal.

### Quick Start

Play some audio through Two Knees\*. Try to input a beat and run through the snapshots to hear a clear example of how it can alter transients.

### Structure and Signal Flow

Set the input level with the In fader to achieve 0 dB on the input meters. The top set of Threshold and Ratio knobs refer to the upper threshold of the compressor. The upper compression ration is applied if the amplitude of the input signal exceeds the upper threshold. The ratio determines how many dB the signal must exceed the threshold to result in a 1dB increase of the output signal. The lower threshold and ration controls are directly beneath the upper controls. The lower compression ratio is applied if the amplitude signal is within the range between the upper and lower thresholds.

The Thresh setting shows up on the display just below the compression curve display. The upper threshold setting will be on the right side of the display and the lower will appear on the left. This will help you to dial in your threshold range to get particular hard-knee or soft-knee sounds. Try adjusting the degree of separation between the upper and lower thresholds and watching the display to see where the compression crosses from sharp-angled hard-knee style to rounded soft-knee.

You can adjust the attack time of the compression in milliseconds using the Attack knob. The Attack setting determines how fast the compressor responds to (turns down, basically) a signal that exceeds the threshold limits. Slower attack times tend to let more transients through. The Release knob lets you set how fast the compressor returns to unity gain (or zero gain) after falling below the threshold limits. You can manipulate the apparent sustain time of some sounds with the Threshold adjustment. Long release times tend to sound more "natural".

Finally, you can make up for gain reduction after compression by adjusting the Out level. You can also bypass the compressor by turning off the On switch.

\* You can play a sound through the built-in loop-player, through the realtime audio inputs, or you can process audio in realtime by using Reaktor as an effect plugin. Please check your Reaktor or Reaktor Session user's guide for helpful information.