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Library credits
Concept + Production: Mate Galic
Technical Assistance: Cornelius Lejeune, Jeremiah Savage
Documentation: Cornelius Lejeune, James Walker-Hall, Thomas Loop, Jace Clayton
Instrument Design: Mike Daliot, Lazyfish, James Walker-Hall, Martijn Zwartjes, Programchild, Tim Exile
Sounddesign: Dennis DeSantis, Junkie XL, AME, Jörg Remmer-Müller, Speedy J, Smyglyssna, Richard Devine, Jam El Mar, Simon Pyke, Tim Exile, Frank Martiniq, Rob Acid, Jake Mandell, Martijn Zwartjes, Jaap Wajer, Telefon Tel Aviv, Mike Dalio, Programchildt
Interface Design: Pfadfinderei, Phillipp Granzin, Ian Warner, Leonard Lass, Phillip Roller, Studiotonne, Johannes Schardt

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Carbon2 is based on Reaktor 4’s well-known workhorse synthesizer, but it has been completely rebuilt. In particular the oscillators and filters are now based on Reaktor Core components developed particularly for this instrument. The panel has been optimized for usability, with a clear structure providing fast access to all parameters while hiding the technical complexity.

Basically, Carbon2 is a classical subtractive synthesizer. The signal of the three oscillator section (left column of the panel) passes through a multi-mode filter (middle column) and is then routed to the effect units (right column). Several modulation sources such as envelope generators and LFOs (placed in a second page in the right panel column) and the global parameters (a third page in the right column) control the sound, adding additional liveliness and movement.

**Oscillators**

The oscillator section produces the instrument’s basic signal. Three oscillator slots provide several different waveforms; along with traditional analogue types like sine and sawtooth there is a digital wavetable oscillator containing a wide array of waveforms that can be crossfaded smoothly. A noise generator and
a ring modulator based on the signal of the three main oscillators are added for a total of five basic sound sources.

Each oscillator slot offers control over volume, pitch, and waveform synchronization. The pitch and sync controls are placed in two pages at the bottom of the panel, grouped with a third page controlling the waveform. This third page is only active if the digital wavetable or the doubled sawtooth is selected.

<p>| Main Routing | Sets the destination of the respective oscillator’s signal. On [F], the sound is sent to the [Filter] section; switching to [D] bypasses the filter and routes the signal directly to the effect units. |
| Noise | Switches the white noise generator on or off. |
| Ring | Selects which oscillator signals are fed into the ring modulator. Switch off to save CPU power if the ring modulator is not used. |
| Osc1/2/3 | Selects the waveform of each oscillator slot. Along with the standard waveforms (sawtooth, pulse, triangular, sine and noise), you will find a doubled sawtooth, a quantized sine, a buzz oscillator based on a noise generator, and a digital wavetable. (See the [Wave] page for additional information on the doubled sawtooth and the digital wavetable.) |
| Level | Sets the slot’s volume level. |
| Level Modulation Source | Selects the slot’s volume level modulation source. |
| Level Modulation Amount | Sets the amount and polarity of modulation applied to the slot’s volume level. Clicking on the control’s title bar restores the value to its default. |
| Pitch A/B Modulation Source | Selects sources to modulate the oscillators’ pitch. The two individual slots ([A] and [B]) can mix up to two sources. |
| Pitch A/B Modulation Amount | Adjusts the amount and polarity of modulation applied to the oscillators’ pitch. The left side of the control adjusts coarse values, the right side is used for fine-tuning; clicking the control’s title bar restores the value to its default. |
| Osc1/2/3 Pitch Shift | Transposes the oscillators’ sound respectively. The left side of the control adjusts coarse values, the right side is used for fine-tuning; by clicking on the controls title bar with the mouse the value is reset to its default. |
| Osc1/2/3 Modulation Switch A/B | Turns modulation of the oscillator’s pitch by modulation slot [A] or [B] on or off. |</p>
<table>
<thead>
<tr>
<th>Wave</th>
<th>A/B Modulation Source</th>
<th>Selects sources to modulate the waveform. The two individual slots ([A] and [B]) can mix up to two sources. This will show no effect until the doubled sawtooth or the wavetable is selected in [Osc1/2/3].</th>
</tr>
</thead>
<tbody>
<tr>
<td>Wave</td>
<td>A/B Modulation Amount</td>
<td>Adjusts the amount and polarity of modulation applied to the waveform. Clicking on the control's title bar resets the value to its default. This will show no effect until the doubled sawtooth or the wavetable is selected in [Osc1/2/3].</td>
</tr>
<tr>
<td>Osc1/2/3 Waveform Control</td>
<td>This either selects a digital waveform from the wavetable, or — if the doubled sawtooth is activated in [Osc1/2/3] — this controls the ratio between the phases of both sawtooth waves.</td>
<td></td>
</tr>
<tr>
<td>Osc1/2/3 Modulation Switch A/B</td>
<td>Turns modulation of the waveform selection by modulation slot [A] or [B] on or off.</td>
<td></td>
</tr>
<tr>
<td>Sync</td>
<td>Gate Sync Switch</td>
<td>Turns synchronization of the oscillators' waveforms to the MIDI gate on or off. If on, all three oscillators are reset to the phase adjusted in [Gate Sync Phase] when a note is pressed.</td>
</tr>
<tr>
<td>Sync</td>
<td>Gate Sync Phase</td>
<td>Controls the phase to which all oscillators are set on MIDI gate events. Clicking on the control's title bar restores the default value.</td>
</tr>
<tr>
<td>Osc2/3 Sync Switch</td>
<td>Switches on or off the synchronization of the oscillators 2 and 3 respectively to the signal of oscillator 1. If on, the oscillator is reset to the phase adjusted in [Osc2/3 Sync Phase] when the signal of oscillator 1 rises above zero. (See also [Osc2/3 Mode Fade].)</td>
<td></td>
</tr>
<tr>
<td>Osc2/3 Sync Phase</td>
<td>Controls the phase to which the oscillators 2 and 3 are reset when the signal of oscillator 1 rises above zero. Clicking on the control's title bar restores the default value. (See also [Osc2/3 Mode Fade].)</td>
<td></td>
</tr>
<tr>
<td>Osc2/3 Mode Fade</td>
<td>Interpolates between hard synchronization (at low values) and soft synchronization (at high values). In hard synchronization mode the oscillator is always reset if the signal of oscillator 1 rises above zero; with soft synchronization this is not always the case, producing a mix between the synced waveform and the non-synced one. Clicking on the control's title bar restores the default value.</td>
<td></td>
</tr>
</tbody>
</table>
**Filter**

The filter section is placed between the oscillators and the effects; it sculpt the oscillators’ basic sounds. Before the signal is routed to the filter it passes two effects that provide saturation and quantization, as well as additional low- and high-shelf equalizers. The filter itself contains several modes, optimized for a warm yet crisp sound. You’ll find standard low-pass and high-pass, bandpass, and band-reject filters, a special feedback filter (called [Zwnl]), and a peak EQ and comb filter. After the main filter comes another effect section, similar to the previous one.

<table>
<thead>
<tr>
<th>Pre-Filter Effects</th>
<th>Mode Select</th>
<th>Effect A/B Mode Select</th>
<th>Amount</th>
<th>Effect A/B Amount</th>
<th>Cutoff</th>
<th>Resonance</th>
<th>Bandwidth</th>
<th>E2</th>
<th>Key</th>
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<td>PreAmp</td>
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</table>

- **Pre-Filter Effects**
  - Mode Select: Selects the effect units applied to the signal before it passes to the filter. There are low and high shelf EQs in the left [A] menu and saturation and quantization in the subsequent right [B] menu.

- **Effect A/B Mode Select**: Sets the parameter of the effect unit selected by [Effect A/B Mode Select]. For the equalizers, this is the amount of damp or boost applied to the signal. For the saturator it’s the amount of drive, and for the quantizer it’s the amount of distortion.

- **Main PreAmp**: Controls the level correction of the signal after it has passed the [Pre-Filter Effects] section and before it enters the main filter.

- **Mode**: Selects the filter mode. There are high-pass, bandpass and band-reject filters, several low-pass modes, a feedback lowpass, a peak equalizer, and a comb filter.

- **Cutoff**: Sets the frequency of the filter.

- **Resonance**: Sets the resonance of the filter.

- **Bandwidth**: Sets the width of the band for the bandpass and band-reject filters. If the peak equalizer is selected, this parameter sets the amount of boost applied.

- **E2**: Controls the amount and polarity of modulation applied to the cutoff control by the second envelope generator. Turn to the left for negative modulation, i.e. low cutoff values at high envelope signals. Turn to the right for normal positive modulation.

- **Key**: Controls the amount and polarity of modulation applied to the cutoff control by the current pitch. Turn to the left for negative modulation, i.e. low cutoff values at high pitches. Turn to the right for normal positive modulation. This modulation is independent of the Key Scaler of the [Modulation] section.
### Cutoff/Resonance/Bandwidth Modulation Source
Selects the sources used to modulate the filter’s cutoff, resonance and bandwidth. Up to two sources can be selected, and their signals are summed together. In case of the cutoff modulation, these signals are added to the hard-wired modulation by the second envelope generator and the MIDI pitch.

### Cutoff/Resonance/Bandwidth Modulation Amount
Adjusts the amount and polarity of modulation applied to the filter’s cutoff, resonance and bandwidth. Clicking on the control’s title bar restores the default value. In case of the cut-off modulation, this amount doesn’t affect the hard-wired modulation by the second envelope generator and the MIDI pitch.

### Post-Filter Effects
Selects the effect units applied to the signal after the filter, before it gets routed to the main effect units. You’ll find saturation and quantization in the left [A] menu, and lowpass and highpass filters in the right [B] menu.

### Effect A/B Amount
Sets the parameter of the effect unit selected by [Effect A/B Mode Select]. For the saturator this is the amount of drive; for the quantizer the amount of distortion; and for both filters the cut-off frequency.

### Effects
The effects additionally enhance the instrument’s sound. There are five units: a pitch shifter, a phaser, a chorus, an equalizer and a delay. These standard effects are engineered for the finest of results.

### Power & Mix
Each effect unit provides a power switch and a mix button. The mix button crossfades between the dry, unprocessed signal (at the left) and the wet effect sound (at the right). To save CPU power, turn the power switch off when the specific effect is not in use.

### Pitch Shifter
Determine the pitch shift of the left and right channel respectively in semitones.

### Grain Size L/R
Adjust the grain size of the pitch shifting algorithm for the left and right channel respectively. Turn to the left for large chunks and echoic sounds, to the right for tiny grains and an accurate pitch shift.

### Feedback
Controls the amount of feedback.

### Reverse
Switches between forward and reverse grain playback.
**Phaser**

- **Center Frequency**
  Sets the center frequency of the filters that produce the phaser signal.

- **Modulation Rate**
  Sets the speed at which the [Center Frequency] is modulated.

- **Phase**
  Sets the phase of the LFO modulating the [Center Frequency].
  (See also [Modulation Rate].)

- **Depth**
  Sets the amount of modulation.

- **Resonance**
  Sets the resonance of the internal filters.

- **Feedback**
  Sets the amount of feedback.

**Chorus**

- **Delay**
  Sets the main delay of the chorus.

- **Depth**
  Sets the amount of modulation applied to the [Delay] time.

- **Rate**
  Sets the speed at which the [Delay] time is modulated.

**Equalizer**

- **Bass Boost**
  Controls the boost (or damping) applied to the bass frequencies below 300 Hz.

- **Mid Frequency**
  Adjusts the frequency of the peak equalizer applied to the middle frequency spectrum.

- **Mid Boost**
  Controls the boost (or damping) applied to the middle frequencies around the [Mid Frequency].

- **Mid Resonance**
  Sets the resonance of the mid equalizer.

- **High Frequency**
  Adjusts the frequency of the high shelf equalizer.

- **High Boost**
  Controls the boost (or damping) applied to the frequencies above the [High Frequency].

**Delay**

- **Delay L / R**
  Sets the delay times of the left and right channel respectively.
  The time is controlled in increments selected by the [Quantize] control.

- **Fine L / R**
  Adds an offset to the values controlled by [Delay L / R] in milliseconds.

- **Quantize**
  Selects the unit by which the delay times are quantized. Sixteenth notes and eighth triplets are available.

- **Feedback**
  Sets the amount of feedback.

- **Wrap**
  Controls the amount of cross-feedback. Turn to the left to route the every channel’s feedback to itself; turn to the right to route it to the other channel.

- **Resonance**
  Sets the amount of resonance applied to the low-pass and high-pass filters within the feedback circuit.

- **Lowpass**
  Controls the frequency of the low-pass filter within the feedback circuit.

- **Highpass**
  Controls the frequency of the high-pass filter within the feedback circuit.
## Modulation Sources

Several modulation sources are available: two ADSR envelope generators, a recordable envelope, and two LFOs combined with a key-scaler that provides four independent control points and four freely assignable MIDI controllers. The envelope generators and LFOs offer several types of MIDI clock interaction for rhythm-based modulation effects.

<table>
<thead>
<tr>
<th>Envelope Generators</th>
<th>Trigger 1/2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Selects the events that re-trigger the envelope generator. [Gate] only activates the MIDI gate signal. [Clock Gate] re-triggers the envelope at each unit selected by [Quantization] as long as the MIDI gate is open. [SP Clock Gate] is similar, but synchronizes the quantization to the global MIDI song position; therefore, the MIDI clock has to be running. (See also [Globals][EG Mode].)</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Envelope Generators</th>
<th>Quantization</th>
</tr>
</thead>
<tbody>
<tr>
<td>Selects the metrical unit used to re-trigger the envelope if [Trigger] is set to [Clock Gate] or [SP Clock Gate].</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Envelope Generators</th>
<th>Key</th>
</tr>
</thead>
<tbody>
<tr>
<td>Controls the amount and polarity of modulation applied to the envelope's transition times by the current pitch. Turn to the left for negative modulation, i.e. shorter attack, decay and release times at low pitches. Turn to the right for normal positive modulation, i.e. longer times at low pitches.</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Envelope Generators</th>
<th>Velocity</th>
</tr>
</thead>
<tbody>
<tr>
<td>Controls the current velocity's influence on the envelope amplitude. At low values the envelope triggers with the same amplitude; at high values the MIDI velocity determines its peak value.</td>
<td></td>
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</tbody>
</table>

<table>
<thead>
<tr>
<th>Envelope Generators</th>
<th>Transition Time Modulation Select</th>
</tr>
</thead>
<tbody>
<tr>
<td>Selects the additional modulation applied to the envelope generator's transition times. The attack phase can be modulated by the MIDI velocity while the decay time can be modulated by the velocity and the four MIDI controllers (see [MIDI Controllers]). The amount and polarity of modulation is controlled by [Transition Time Modulation Amount].</td>
<td></td>
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</tbody>
</table>

<table>
<thead>
<tr>
<th>Envelope Generators</th>
<th>Transition Time Modulation Amount</th>
</tr>
</thead>
<tbody>
<tr>
<td>Controls the amount and polarity of modulation applied to the destination selected by [Transition Time Modulations Select]. Turn to the left for negative modulation, i.e. shorter attack or decay times at low modulation source values. Turn to the right for normal positive modulation, i.e. longer times at low values.</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Envelope Generators</th>
<th>Attack</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sets the attack time of the envelope generator.</td>
<td></td>
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</tbody>
</table>

<table>
<thead>
<tr>
<th>Envelope Generators</th>
<th>Decay</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sets the decay time of the envelope generator.</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Envelope Generators</th>
<th>Sustain</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sets the sustain level of the envelope generator.</td>
<td></td>
</tr>
<tr>
<td>Parameter</td>
<td>Description</td>
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<tr>
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</tr>
<tr>
<td>Release</td>
<td>Sets the release time of the envelope generator.</td>
</tr>
<tr>
<td>Hold</td>
<td>Sets the duration of an additional hold phase between attack and decay.</td>
</tr>
<tr>
<td>Delay</td>
<td>Adds an initial delay period before the trigger signal restarts the envelope.</td>
</tr>
<tr>
<td>R=D</td>
<td>Links the release time to the decay time. If on, the value adjusted by [Decay] is also used to control the release phase.</td>
</tr>
<tr>
<td>Envelope Generator Record 3</td>
<td>Arms the recordable envelope. The recording is started when a MIDI gate is received and ends when the gate closes. All movements of the [Value] knob are stored and can be played back as envelope (see [Play]).</td>
</tr>
<tr>
<td>Play</td>
<td>Enables playback of the recorded movements, triggered like an envelope by MIDI gate signals.</td>
</tr>
<tr>
<td>Loop</td>
<td>Loops the recorded movement on playback.</td>
</tr>
<tr>
<td>Value</td>
<td>When recording (see [Record]), every movement of this knob is stored to the memory. During playback (see [Play]), the knob displays the recorded movements.</td>
</tr>
<tr>
<td>LFO 1/2 Waveform</td>
<td>Selects the waveform of the Low Frequency Oscillator. There are the standard waveforms [Sine], [Triangular], [Pulse], and [Random Steps], and several derivations: [Pulse+] is a pulse waveform with all negative values clipped to 0; [Saw Up+] and [Saw Down+] are triangular forms with only rising resp. falling ramp and only positive values; [Hsin+] is a multiplication of [Pulse+] and [Sine] etc.</td>
</tr>
<tr>
<td>Amplitude Modulation Source</td>
<td>Selects the source used to modulate the LFO’s amplitude. Clicking the control’s title bar restores the value to its default.</td>
</tr>
<tr>
<td>Amplitude Modulation Amount</td>
<td>Adjusts the amount and polarity of modulation applied to the LFO’s amplitude.</td>
</tr>
<tr>
<td>Trigger Mode</td>
<td>Selects the events that re-trigger the LFO. In [Freerun] mode no reset occurs; in [Gate] mode the LFO is set to the phase adjusted by [Reset Phase] on a MIDI gate event. [Clock Gate] is similar to [Gate] mode but also activates a grid for the LFO’s frequency (see [Rate]). [SP Clock Gate] additionally synchronizes the reset to the global MIDI song position.</td>
</tr>
</tbody>
</table>
Reset Phase
Adjust the phase to which the LFO is set on re-triggering events.

Rate
Selects the source used to modulate the LFO's frequency. If [Trigger Mode] is set to [Clock Gate] or [SP Clock Gate], frequency modulation is not available.

Modulation Source
Adjusts the amount and polarity of modulation applied to the LFO's frequency. Clicking the control’s title bar restores the value to its default. If [Trigger Mode] is set to [Clock Gate] or [SP Clock Gate], frequency modulation is not available.

Rate
Sets the frequency of the LFO. If [Trigger Mode] is set to [Clock Gate] or [SP Clock Gate], a grid is applied to this control, quantizing the LFO's rate to the metrical units selected in [Rate Quantization].

Rate Quantization
Selects the metrical unit used as quantization grid for [Rate] when [Trigger Mode] is set to [Clock Gate] or [SP Clock Gate].

KeyScaler Sliders
Provides a signal derived from the current pitch that can be used as modulation source. The four sliders define the function used to map the MIDI pitch onto the modulation signal: At low pitches, the value of the leftmost slider is used as modulation signal; at high pitches the value of the rightmost slider is selected. In between, interpolation occurs, using the two middle sliders as control points. In addition to the normal signal, there is a modulation source that multiplies the key-scalers value by the current MIDI velocity.

MIDI Controllers Faders
The leftmost fader is hard-wired to the MIDI modulation wheel. All others can easily be assigned to any MIDI Continues Controller via MIDI Learn. They are available as modulations sources, named C1, x1, x2 and x3.
**Global Controls**

The global controls access several different functions. First – and most important – the voice allocation of the synthesizer can be controlled, providing polyphonic and monophonic modes; by selecting the unison mode all available voices are set to the same pitch (as in a monophonic synth), but each one is slightly detuned. This results in waveform interference and a thick, chorus-like sound. Monophonic modes also provide portamento.

Parameters determine the master pitch shift and MIDI pitchbend range, and adjust global tremolo and vibrato. Voices’ position within the stereo field can also be adjusted.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
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</thead>
<tbody>
<tr>
<td>Gate Mode</td>
<td>Selects the global operation mode. [Poly] selects the only polyphonic mode; portamento does not work in this mode (see [Glide Speed]). [Mono] results in a monophonic gate signal that is triggered on every MIDI note. [Legato] is similar but generates a new gate trigger signal only when the gate has been closed before, i.e. no note is already pressed. [Uni Mono] and [Uni Legato] activate the unison modes: A monophonic gate signal is used for all voices, but all available voices are used and detuned by the [Unisono] and [Unisemi] controls.</td>
</tr>
<tr>
<td>Envelope Mode</td>
<td>Selects the envelopes’ behavior during the release period if a new attack is triggered. [Re-trigger] starts the attack phase beginning with the current envelope amplitude; [Reset] starts the attack with a value of zero. Thus, [Reset] might lead to unwanted clicks if used without care.</td>
</tr>
<tr>
<td>Unisono</td>
<td>Sets the amount of detuning applied to each voice when [Uni Mono] or [Uni Legato] is selected as [Gate Mode]. Slight detuning results in thick, chorus-like sounds.</td>
</tr>
<tr>
<td>Unisemi</td>
<td>Sets the amount of pitch shifting applied to each voice when [Uni Mono] or [Uni Legato] is selected as [Gate Mode]. This acts like the [Unisono] control but detunes the voices in semitones, e. g. a value of 12 will set all voices one octave apart.</td>
</tr>
<tr>
<td>Drift</td>
<td>Enables a drift mode that slightly detunes higher pitches. This results in a more analogue like sound.</td>
</tr>
<tr>
<td>Key</td>
<td>Activates key-scaling for the unisono control. If pressed, the [Unisono] value is lowered automatically at high pitches for a more constant sound over the complete pitch range of the instrument.</td>
</tr>
<tr>
<td>Velocity</td>
<td>Selects the mapping applied to the MIDI velocity. While [Linear] doesn’t change the velocity, [Log] results in a compressor like effect while [Expo] produces the opposite effect.</td>
</tr>
<tr>
<td>Parameter</td>
<td>Description</td>
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</tr>
<tr>
<td>Coarse</td>
<td>Sets the global tuning of the instrument in semitones, ranging from -63 to +64.</td>
</tr>
<tr>
<td>Fine</td>
<td>Sets the global tuning of the instrument in semitones, ranging from -0.5 to +0.5.</td>
</tr>
<tr>
<td>Glide Speed</td>
<td>Adjusts the speed at which new pitches are reached if they are slurred, i.e. if the previous note was still held when the new one was pressed. This portamento effect only works in monophonic modes (see [gate Mode]).</td>
</tr>
<tr>
<td>Pitchbend Range</td>
<td>Sets the range in semitones by which the MIDI pitchbend wheel transposes the global pitch.</td>
</tr>
<tr>
<td>Vibrato Mode</td>
<td>Selects whether vibrato is off, on, or faded in by the MIDI modulation wheel.</td>
</tr>
<tr>
<td>Vibrato Amount</td>
<td>Sets the amount of vibrato. Clicking the control’s title bar restores the value to its default.</td>
</tr>
<tr>
<td>Vibrato Style</td>
<td>Selects between three different vibrato modes.</td>
</tr>
<tr>
<td>Key</td>
<td>Adjusts the amount of key scaling applied to the vibrato. Turn to the left for no scaling, to the right for less vibrato at low pitches, producing a more musical effect.</td>
</tr>
<tr>
<td>Tremolo Mode</td>
<td>Selects whether tremolo is off, on, or faded in by the MIDI modulation wheel.</td>
</tr>
<tr>
<td>Tremolo Amount</td>
<td>Sets the amount of tremolo. Clicking the control’s title bar restores the value to its default.</td>
</tr>
<tr>
<td>Vibrato &amp; Tremolo Frequency</td>
<td>Sets the speed of both vibrato and tremolo.</td>
</tr>
<tr>
<td>Voice Panning Switch</td>
<td>Selects whether the instrument's voices are placed at different positions within the stereo field. Especially in combination with the [Unisono] control this can produce interesting spatial effects.</td>
</tr>
<tr>
<td>Voice Panning Amount</td>
<td>Sets the amount of voice panning. Clicking the control’s title bar restores the value to its default.</td>
</tr>
<tr>
<td>Master 1/2</td>
<td>Defines the instrument’s output level. Use the large middle knob to adjust the preset’s maximum level; the smaller knob to the right controls the instrument’s output amplitude in all patches.</td>
</tr>
<tr>
<td>Key Amp</td>
<td>Adjusts the amount of automated amplitude correction in respect to the synthesizer's pitch. Turn to the left for no influence of the pitch onto the output level, to the right to damp high pitches. This can be used to simulate the sound of analogue synthesizers.</td>
</tr>
</tbody>
</table>
If you get excited by words such as analog and vintage, look away now. Oki Computer 2 is a compact wavetable synthesizer, a specialist in digital lo-fi sounds that hails back to the era of 8-bit beeps and bleeps... It is also rather capable at creating buzzing leads, rhythmic sequences, and odd tasty bass tones.

Oki Computer 2’s panel is compact but packed with features. Thankfully, most sections should be fairly straightforward to the average synth user. However,
The [Oscillator] section is somewhat unique and users are strongly encouraged to read this part of the manual. Oki Computer 2 features a bank of 50 waves. For every patch you can load any 16 of these waves into the oscillator in any order. This flexibility represents a major improvement over the original ensemble (where the oscillator was permanently ‘hardwired’ to the same 16 waves). What’s more, each wave loaded into the oscillator can be processed in a variety of ways.

**MIDI In**

The drop-down list at the panel’s top-left switches between polyphonic and monophonic operation modes. In polyphonic mode, Oki Computer operates as a standard poly-synth. Monophonic mode does not restrict the number of voices to 1; it enables some very musical features – legato, glide, and unison.

- **Gate Mode** Selects whether the instrument is used as polyphonic or monophonic synthesizer.
- **Unison** Determines the number of simultaneous voices. This is only active if [Gate Mode] is set to [mono].
- **Spread** Defines the amount of voice detuning in semitones. This is only active if [Gate Mode] is set to [mono].
- **Glide** Sets the amount of portamento, i.e. the time used to reach a new MIDI pitch. This is only active if [Gate Mode] is set to [mono].
- **Octave** Transposes the pitch of the entire oscillator in octave steps.
- **Semitone** Transposes the pitch of the entire oscillator in semitone steps.
- **Fine** Fine-tunes the pitch of the entire oscillator's pitch.
- **Pitchbend** Defines the MIDI pitch bend wheel range in semitones.

**Oscillator**

The [Wavetable Position Bar], located beneath the main oscillator window, is perhaps the most difficult part of the synthesizer to understand. This bar has two purposes. Firstly, the square box indicates the current wave slot selected for editing (there are 16 slots). Secondly, the bright green line indicates the current wavetable position. The current wavetable position is set by the [Wavetable Position Knob] (to the left of the drive knob), plus any modulation routed to the wave table position (see [Modulation Matrix]).

The best way to explain how the [Wavetable Position Bar] works is by example: Click on the snapshot menu and recall preset number 1 - ‘Default’.
In this preset, the oscillator is loaded with 16 sine waves (needless to say, this doesn’t sound particularly interesting). Click the leftmost box on the Wavetable Position Bar - this will select the first slot for editing. Note that in the box labeled [Wave] (beneath the [Wavetable Position Bar]) a picture of a sine wave is displayed, with a number zero next to it. This indicates that the sine wave (wave number zero from the master bank) is loaded into the current slot. To load a different wave, click and drag vertically on the Wave Selector. Now click on the second wave slot (i.e. the adjacent dark gray box). The Wave Selector will display a sine wave (remember this patch had 16 identical sine waves loaded). Now try loading a different wave to slot 2, again by clicking and dragging on the Wave Selector.

In the ‘Default’ snapshot, the [Wavetable Position Knob] is set to 1.00. This means that when you play a note, you will hear (and see) the wave loaded into slot 1. Press a note on your keyboard, and slowly move the knob from 1.00 to 2.00. You will hear and see the wave loaded into slot 1 morph into the wave loaded into slot 2. Notice how the wave position indicator moves correspondingly. This is the key to how Oki Computer 2 produces dynamic sounds: by morphing between adjacent waves in the wavetable. While this can be done manually with the wave position knob, things get much more interesting when the various modulators (e.g. envelope, sequencer, LFO) are used to mix between waves.

Apart from the Wave Selector, all the controls underneath the wavetable position bar are used to modify wave shape. When using these controls, it is important to remember that they only affect the wave in the selected slot (i.e. the green box), which is not necessarily the wave shape currently being played (i.e. the green line).

<table>
<thead>
<tr>
<th>Control</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ratio</td>
<td>Sets the number of times the wave shape will repeat in a single oscillator cycle. Note that the integer and decimal values can be set independently, also note that adjusting the ratio will cause pitch shifting.</td>
</tr>
<tr>
<td>Phase</td>
<td>Rotates the wave start position within the oscillator cycle.</td>
</tr>
<tr>
<td>Shape</td>
<td>Skews the wave shape to either the left or right (on the pulse wave this is identical to a pulse width control).</td>
</tr>
<tr>
<td>Digitize</td>
<td>Reduces the wave’s bit depth.</td>
</tr>
<tr>
<td>Amp</td>
<td>Attenuates the wave volume.</td>
</tr>
<tr>
<td>Copy</td>
<td>Stores the current settings into an edit buffer that can be read out again by the [Paste] button.</td>
</tr>
</tbody>
</table>
Paste
Recalls the data from the edit buffer (see [Copy]).

Distortion Amount
Controls the amount of distortion. (See also [Distortion Mode].)

Distortion Mode
Selects the way the signal is distorted. [Saturate] applies a ‘standard’
saturation curve to the signal. [Triangle] and [Sine] both involve wrap-
ing their respective shapes around the input signal. When used on a
sine wave, these two functions can sound somewhat reminiscent of FM.
[Noise] enables a noise generator.

Filter / Out
This section controls the shaping applied to the sound’s frequency spectrum
(filter) and amplitude.

Amplitude
Selects the main output envelope. In most cases, [E1] will be the preferred
choice. Sometimes however, you may want to use Envelope 1 for modula-
tion purposes only. In this case, select either [G] (MIDI gate, ignoring
velocity) or [Vel] (MIDI gate, including velocity).

Damp
Controls the amount of high-frequency damping.

Volume
Sets the main output volume in decibels.

Cut-off
Sets the filter’s cut-off frequency.

Resonance
Adjusts the filter’s resonance amount.

Track
Defines the amount of cut-off pitch tracking. At 100%, cut-off is in-
creased by one semitone for each increment in MIDI pitch. At -100%,
cut-off is decreased by one semitone for each increment in MIDI pitch.
At +/- 200%, cut-off changes by two semitones for every one semitone
change in pitch.

Low-pass, Band-pass, High-pass
Determines the mix ratio of the high-pass, band-pass and low-pass
components of the filter output signal.
**Envelope, CC1, Sequencer and LFO**

Oki Computer 2 features two envelope generators. Both can be used for general modulation via the Modulation Matrix, but envelope 1 can also be routed directly to output volume in the [Filter / Out] section. Otherwise, the envelope generators are identical.

The CC1 section allows you to record modulation wheel movements. To record a sequence, click the [Record] button. The button will flash, indicating that it is armed and waiting. Recording will begin when a MIDI note is pressed, and finish when the note is released (or when all recording memory is used up). You can record movements with the mouse, or the CC1 knob on a MIDI controller. As long as the [Play] button is depressed, recordings will play back each time a note is triggered. The recording is sent to MIDI CC1. Therefore, to use the recording as a modulation source, select CC1 in the Modulation Matrix. Note that even though sequences are recorded monophonically, playback operates in full polyphony.

The sequencer is a highly flexible modulation source. It can operate as an arpeggiator, a custom shape LFO or an additional envelope. You can draw its steps with the mouse.

You’ll find a standard LFO next to the sequencer.

<table>
<thead>
<tr>
<th>Envelope 1/2</th>
<th>Attack</th>
<th>Controls the attack time of the envelope generator.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Hold</td>
<td>Controls the hold time of the envelope generator.</td>
<td></td>
</tr>
<tr>
<td>Decay</td>
<td>Controls the decay time of the envelope generator.</td>
<td></td>
</tr>
<tr>
<td>Sustain</td>
<td>Controls the sustain level of the envelope generator.</td>
<td></td>
</tr>
<tr>
<td>Release</td>
<td>Controls the release time of the envelope generator.</td>
<td></td>
</tr>
<tr>
<td>Speed</td>
<td>Multiplies the overall envelope time.</td>
<td></td>
</tr>
<tr>
<td>Velocity</td>
<td>Determines the extent to which envelope amplitude is linked to velocity.</td>
<td></td>
</tr>
<tr>
<td>Clock Sync</td>
<td>Synchronizes the envelope times to the global MIDI tempo.</td>
<td></td>
</tr>
<tr>
<td>Loop</td>
<td>Activating this button results in the attack, hold and decay stages looping when MIDI notes are depressed.</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>CC1</th>
<th>Record</th>
<th>Armes the CC1 recorder.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Play</td>
<td>Enables playback of the recorded movements, triggered like an envelope by MIDI gate signals.</td>
<td></td>
</tr>
<tr>
<td>Loop</td>
<td>Loops the recorded movement on playback.</td>
<td></td>
</tr>
</tbody>
</table>
Sequencer Clock Sync
Synchronize the sequencer to the MIDI clock. Note that when both [Clock Sync] and [Phase Lock] are activated, the sequencer becomes locked to the MIDI song position.

Phase Lock
Locks the phase of the sequencer. When this is active, MIDI note events will *not* re-trigger the sequencer. Note that when both [Clock Sync] and [Phase Lock] are activated, the sequencer becomes locked to the MIDI song position.

Loop
When enabled, the sequencer will loop indefinitely, otherwise it will play back only once when triggered.

Snap
Activates a vertical grid with a step size of 1/12 of the complete height.

Frequency
Determines the sequencer speed.

Length
Sets the sequencer length in steps.

Smooth
Determines the amount of interpolation between adjacent steps (at hard-right, the sequencer produces smooth envelope-type output).

Variation
Imparts a kind of swing onto the sequencer movement, so steps play alternately faster and slower. Center this control for equal length steps.

LFO Clock Sync
Synchronize the LFO to the MIDI clock. Note that when both [Clock Sync] and [Phase Lock] are activated, the LFO becomes locked to the MIDI song position.

Phase Lock
Locks the phase of the LFO. When this is active, MIDI note events will *not* re-trigger the LFO. Note that when both [Clock Sync] and [Phase Lock] are activated, the LFO becomes locked to the MIDI song position.

Frequency
Sets the LFO speed.

Phase
Determines the point in the LFO wave where oscillation begins when a note is triggered. This only function when [Phase Lock] is off.

Fade
Sets the fade-in time of the LFO (i.e. the time taken to reach full amplitude).

Shape
Skews the LFO shape to either the left or right.

FM / AM
Determine the amount which the modulation wheel (including recorded movements) modulatea frequency and amplitude respectively.
**Modulation Matrix**

The modulation matrix enables any four modulation sources to be routed to any four destinations. You can select modulation sources using the upper drop-down menus. Destinations are selected with the lower menus. The sliders in-between these menu set the modulation amount. The full list of modulation sources and destinations is summarized in the following table:

<table>
<thead>
<tr>
<th>Sources</th>
<th>Destinations</th>
</tr>
</thead>
<tbody>
<tr>
<td>Vel</td>
<td>Amp</td>
</tr>
<tr>
<td>PB</td>
<td>Pitch</td>
</tr>
<tr>
<td>CC1</td>
<td>Wave</td>
</tr>
<tr>
<td>E2</td>
<td>Cutoff</td>
</tr>
<tr>
<td>Seq</td>
<td>Chorus</td>
</tr>
<tr>
<td>LFO</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Sources</th>
<th>Destination Description</th>
<th>Range</th>
</tr>
</thead>
<tbody>
<tr>
<td>Vel</td>
<td>MIDI note on velocity</td>
<td>(0 to 1)</td>
</tr>
<tr>
<td>PB</td>
<td>MIDI pitchbend wheel</td>
<td>(-1 to 1)</td>
</tr>
<tr>
<td>CC1</td>
<td>MIDI CC1 - the modulation wheel. Note that recorded CC1 movements (in the recordable envelope section) are routed to this parameter.</td>
<td>(0 to 1)</td>
</tr>
<tr>
<td>E2</td>
<td>Envelope generator 2</td>
<td>(0 to 1)</td>
</tr>
<tr>
<td>Seq</td>
<td>The sequencer</td>
<td>(-1 to 1)</td>
</tr>
<tr>
<td>LFO</td>
<td>The LFO</td>
<td>(-1 to 1)</td>
</tr>
<tr>
<td>Amp</td>
<td>Output volume</td>
<td>(-100% to +100%)</td>
</tr>
<tr>
<td>Pitch</td>
<td>Oscillator pitch</td>
<td>(-12 to +12 semitones)</td>
</tr>
<tr>
<td>Wave</td>
<td>Oscillator wave position</td>
<td>(-16 to +16)</td>
</tr>
<tr>
<td>Cutoff</td>
<td>Filter cutoff</td>
<td>(-60 to +60 semitones)</td>
</tr>
<tr>
<td>Chorus</td>
<td>Chorus frequency</td>
<td>(-100% to 100%)</td>
</tr>
</tbody>
</table>
SteamPipe 2 is a physical-modeling synthesizer that effectively models air being blown through a tunable pipe. It uses a tuned resonator to create bowed, blown, and plucked sounds, as well as strange new hybrid sounds. In addition to a tuned all-pass filter and many controls for the “shape” of the pipe, there is a mod wheel-controlled filter to achieve damping and breath noise effects. The excellent-sounding SpaceMaster Deluxe reverb unit adds dimension to the overall signal. You can find it on panel B.

SteamPipe 2 simulates air passing through a pipe of variable size and resonance. It’s physical-modeling techniques use contoured noise signals passing through tuned and filtered feedback delays. The ensemble is basically split into two parts: Steam and Pipe. The Steam module generates shaped and filtered noise. Think of the Steam module as SteamPipe 2’s oscillator. Steam provides the sound energy that will be pitch-formed by the Pipe. The Pipe module gives the “wind” pitch and resonance. The patch also has an ADSR volume envelope and a low pass filter. Both can be modulated by key- and velocity-tracking.

Steam Pipe 2 can be a very expressive synthesizer, so make sure that you plug in your MIDI keyboard and check out the presets with the mod wheel in action.

**Steam**

The timbral shaping of the DC/Noise source occurs in the Steam section. 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.

**Envelope**
- **Attack**: Sets the attack time of an ADSR envelope triggered by MIDI gate events and used to generate a short initial steam signal; logarithmic control.
- **Decay**: Sets the decay time of an ADSR envelope triggered by MIDI gate events and used to generate a short initial steam signal; logarithmic control.
- **Sustain**: Sets the maximum level the envelope will reach. This gets modulated by velocity if [VelSns] is on.
- **Release**: Sets the time that passes until the envelope is completely faded out after the note-off signal.
- **Velocity**: Controls the velocity sensitivity of the envelope. The higher this value is, the higher the peak value of the envelope will be.
- **Scaling**: This scales the envelope times depending on the pitch of the incoming MIDI notes. Turn to the left for no keyboard scaling, to the right for shorter envelope times on higher notes.
- **Legato**: Toggles legato mode on or off. If on, the envelope restarts only when the gate changes from zero to positive.

**Generator**
- **DC / Noise**: Crossfades between the DC component at the left and filtered noise at the right. The mixed signal is used as steam input of the resonating pipe.
- **Cutoff**: The cutoff frequency of the low-pass filter.
- **Reso**: Sets the level of resonance of the filter. Only works when the filter is in 2-pole mode.
- **Poles**: Toggles between 1-pole and 2-pole low-pass.
- **Key-track**: Controls the key-tracking of the filter. This will scale the cutoff frequency according to keyboard position. The lower the note pitch, the lower the cutoff frequency will be.
- **Vel-Track**: Controls the velocity scaling of the filter. Turn to the right for higher cutoff frequencies at higher MIDI velocities.
- **Env-Amt**: Sets the amount of envelope to the cutoff frequency.
Pipe

The Pipe module is made up of a number of sub-modules for creating pitch and resonance. The noise signal is fed from a single tuned delay providing pitch, into the [Allpass] module for generating resonance. Next, a [Saturator] receives the signal and applies edge and break-up. The [MW Filter] completes the signal chain with an 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. Unlike the [Feedback] section, which simulates the pipe itself, the Push-Pull section controls the air and its oscillations within the pipe.

The [Delay Tune] module contains the tuned delay that provides pitch to the Steam. The [Tune] and [Fine tune] knobs allow you to set the signal’s fundamental pitch. The A440 oscillator at the bottom of the ensemble provides a reference pitch for tuning purposes. 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 resonant delay. It can be turned on and off with the [Power] button in the [Allpass] section. This allpass can be tuned to create resonance effects. You can produce glassy, metallic and bell-like sounds by detuning the allpass 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 [Polarity] switch inverts the pipe polarity, changing the timbre of the sound. This often transforms high frequency tones to deep ones and vice versa.

The [Feedback] module processes the feedback in the signal chain. The [Rev-Time] knob extends or shortens the reverb generated by this feedback signal. The reverb signal can be muffled with the [Damp] control. Damping can be modified by [Key-Track] amount. High [Key-Track] values result in more damping on higher pitches. This allows SteamPipe 2 to emulate struck or plucked instruments like pianos, harps, and acoustic guitars.
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Tuning Tune</td>
<td>Sets the fundamental pitch of the signal. For standard musical tunings set it to the [A440] oscillator at the bottom of the patch.</td>
</tr>
<tr>
<td>Fine Tune</td>
<td>Fine control of signal pitch.</td>
</tr>
<tr>
<td>SREC</td>
<td>Sample rate error correction. Adjusts the tuning correction of the pipe. When the signal modifications in this patch and Reaktor’s sampling rate interfere with the physical model of SteamPipe, this extra-fine pitch tuning becomes necessary. Tune against the A440 section.</td>
</tr>
<tr>
<td>Mod-Whl</td>
<td>Sets the amount of pitch modification by the MIDI modulation wheel. This simulates pitch changes of pipes getting blown softer or harder.</td>
</tr>
<tr>
<td>Feedback</td>
<td>Adjusts the time of the pipe’s reverberation, i.e. the amount of damping applied to the feedback’s signal before it is mixed again with the new incoming signal. The longer the reverb time, the more the incoming noise steam signal becomes a tone with recognizable pitch.</td>
</tr>
<tr>
<td>Damp</td>
<td>Sets the amount of high frequency damping of the pipe at key-up.</td>
</tr>
<tr>
<td>Key-track</td>
<td>Controls the feedback’s key-tracking. Turn to the right for longer reverb times at high MIDI pitches.</td>
</tr>
<tr>
<td>Allpass Tune</td>
<td>Controls the pitch of the allpass resonance. If the allpass filter is switched off, this control shows no effect.</td>
</tr>
<tr>
<td>Fine Tune</td>
<td>Fine tunes the allpass resonance pitch. If the allpass filter is switched off, this control shows no effect.</td>
</tr>
<tr>
<td>SREC</td>
<td>Sample rate error correction. Adjusts the tuning correction of the pipe. When the signal modifications in this patch and Reaktor’s sampling rate interfere with the physical model of SteamPipe, this extra fine-tuning of pitch becomes necessary. Tune against the A440 section. If the allpass filter is switched off, this control shows no effect.</td>
</tr>
<tr>
<td>Mod-Whl</td>
<td>Sets the amount of pitch modification by the MIDI modulation wheel. This simulates pitch changes of pipes getting blown softer or harder.</td>
</tr>
</tbody>
</table>
**Allpass**

On / Off  
Turns the allpass module on or off. Switch on for additional attack effects of the pipe’s sound.

**Diffusion**

Sets the diffusion of the resonances generated by the allpass module. Turn to the left for additional attack effects of the pipe’s sound. It also enhances the sound of harmonic frequencies which are not multiples of the main pitch, like e.g. in bells.

**Push-Pull**

Offset  
Sets the offset amount added to the reverberating steam signal. This parameter influences the incoming steam and its reverberation in the pipe. It interacts tonally with the Polarity button.

Push  
Sets the amount of reverberating steam.

**Saturation**

Soft / Hard  
Controls the balance between soft saturation and hard clipping.

**Symmetry**

This parameter introduces level asymmetry into the signal. With increasing asymmetry the positive part of the signal is reduced.

**Polarity**

This control inverts the polarity of the pipe, thereby changing the timbre of the sound. It interacts tonally with the [Push-Pull] section.
Hi Pass 0  Sets the cutoff frequency of an additional highpass filter within the pipe to enhance formant frequencies. The formants are modified by the pressure in the pipe (not by the pipe’s pitch). The pressure can be controlled by the mod wheel. At low modulation wheel values this knob is used to determine the formant frequency.

Hi Pass 1  Sets the cutoff frequency of an additional highpass filter within the pipe to enhance formant frequencies. The formants are modified by the pressure in the pipe (not by the pipe’s pitch). The pressure can be controlled by the mod wheel. At high modulation wheel values this knob is used to determine the formant frequency.

Key-track High  Controls the amount of key-tracking applied to the highpass filter’s cutoff frequency. Turn to the right for higher cutoff frequencies at high MIDI pitches.

Lo Pass 0  Sets the cutoff frequency of an additional lowpass filter within the pipe to enhance formant frequencies. The formants are modified by the pressure in the pipe (not by the pipe’s pitch). The pressure can be controlled by the MOD wheel. At low modulation wheel values this knob is used to determine the formant frequency.

Lo Pass 1  Sets the cutoff frequency of an additional lowpass filter within the pipe to enhance formant frequencies. The formants are modified by the pressure in the pipe (not by the pipe’s pitch). The pressure can be controlled by the mod wheel. At low modulation wheel values this knob is used to determine the formant frequency.

Key-Track Low  Controls the amount of key-tracking applied to the lowpass filter’s cutoff frequency. Turn to the right for higher cutoff frequencies at high MIDI pitches.

**Global Controls**

The last section of SteamPipe 2 consists of global controls over pitch, polyphony, glide, and an output stage. You also get an Arpeggiator and a test tone generator.
### Voice Mode

- **Pitch Bend**: Sets the range of the pitch bend wheel.
- **Detune**: Introduces a slight detune into the signal for a livelier sound.
- **Mode**: Menu for the different polyphony modes. Choose between poly, mono, unison, and three arpeggiator modes.
- **Glide on / off**: Toggles glide on or off.
- **Glide Time**: This sets the time the pitch of SteamPipe takes to follow incoming MIDI pitches, when [Glide] is on.
- **Mod-Whl**: This knob follows an incoming mod wheel signal. Use it when you have no hardware controller available.
- **Arp Mode**: This menu offers different arpeggiation modes. Choose between up (>>), down (<<), up and down (>><<), and random.
- **Arp Speed**: Menu for choice between different speeds relative to the global tempo.

### Output

- **Spread**: Introduces a stereo spread into the main output.
- **Gain**: Main output volume control.
- **A440 Tuning tone on / off**: If on, a sine oscillator’s signal is mixed into the main output. Use it to tune the pipe. The frequency is 440Hz.
- **Gain**: Controls the volume of the 440Hz tuning tone.

---

**Space Master Deluxe**

You can find this remarkable reverb module on panel B of the SteamPipe 2. Based on several diffusion delays, Space Master can produce a wide array of high-quality natural or experimental ambiences. The patch’s useful set of reverb parameters include an early reflections section, a late reflections module, and a post EQ. Dials for main reverb time, control of balance between the two reflection stages, and between dry and wet signal round off the controls.

### Input and output stage

You can put an initial delay into the reverb signal with the predelay [Time] dial and control the predelay’s stereo position with the [Symmetry] knob. The [Early / Late Balance] slider can be used to move the source in space – more early reflections bring the signal to the front and more late reflections make...
it appear further back in space. At the end of the signal chain the [Dry / Wet] slider crossfades between original signal and reverb.

**Reflections**

Use the two [Size] and [Diffusion] parameters to control two stages of variable density diffused reflections. The early stage commonly represents the direct response of the virtual space, whereas the late reflections define the sound when the early reflections have died away.

For dynamic reverb effects you can use the Modulation section. It offers an LFO routed to the delay times with [Rate] and [Depth] control. The LFO can enhance your reverb signal by adding liveliness.

<table>
<thead>
<tr>
<th>Reflections</th>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Early/Late Reflections</strong></td>
<td>Size</td>
<td>Determines the amount of space generated by the early or late reflections modules by adjusting delay time of the underlying diffusion delays. Higher values give the impression of larger spaces.</td>
</tr>
<tr>
<td></td>
<td>Symmetry</td>
<td>Introduces a stereo shift into the generated reflections.</td>
</tr>
<tr>
<td></td>
<td>Diffusion</td>
<td>Adjusts the perceived density of the reflections generated. Use for a sparser or fuller reverb sound.</td>
</tr>
<tr>
<td></td>
<td>Reverberation Time RT60</td>
<td>This control alters the decay time of the reverb response.</td>
</tr>
<tr>
<td><strong>Modulation</strong></td>
<td>Rate</td>
<td>Control of LFO frequency modulating the delay times.</td>
</tr>
<tr>
<td></td>
<td>Depth</td>
<td>Control for the LFO’s modulation depth. Higher values yield increased amplitude modulation.</td>
</tr>
</tbody>
</table>
**Frequency response**

The two EQ sections serve slightly different needs. The Damping EQs are integrated into the reflection stages and influence their frequency responses. The Post EQ acts on the patch’s main output and should be used to finalize the overall sound.

<table>
<thead>
<tr>
<th>Damping</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Low Freq</td>
<td>Cutoff for the low shelving filter that cuts into diffusion delay frequency response of both early and late reflections.</td>
</tr>
<tr>
<td>High Freq</td>
<td>Cutoff for the high shelving filter that cuts into diffusion delay frequency response of both early and late reflections.</td>
</tr>
<tr>
<td>Lo Damp</td>
<td>Amount of cut for the low shelving filter.</td>
</tr>
<tr>
<td>Hi Damp</td>
<td>Amount of cut for the high shelving filter.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Equalizer</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Low Freq</td>
<td>Cutoff for the low EQ that acts on the main output of the reverb.</td>
</tr>
<tr>
<td>High Freq</td>
<td>Cutoff for the high EQ that acts on the main output of the reverb.</td>
</tr>
<tr>
<td>Lo Boost</td>
<td>Cut or boost amount for the low Equalizer.</td>
</tr>
<tr>
<td>Hi Boost</td>
<td>Cut or boost amount for the high Equalizer.</td>
</tr>
</tbody>
</table>
SubHarmonic

SubHarmonic generates pad-like, atmospheric sounds and – at the same time – thick, monophonic lead patches. It consists of two independent sound generators: The [Sub Oscillator] is based on additive synthesis, and the [Formant] section performs like an oscillator with a constant frequency vowel filter. Internally, those oscillators are quite complex; the [Sub Oscillator], for example, does not use normal harmonics of the main frequency to perform additive synthesis, but produces subharmonics. However, those technical details remain hidden below the simple user interface.

Voice

This section deals with the voice assignment. You have the normal monophonic and polyphonic modes, and an additional unison mode that uses all available voices (like polyphonic) but sets them to the same pitch, only slightly detuned. The resulting phase interferences produce a chorus-like effect.

The amount of portamento and the influence of the MIDI pitchbend wheel onto the instrument can also be adjusted here.

Voice Mode  
Controls the voice assignment of the instrument. [Poly] selects polyphonic, [Mono] switches to monophonic behavior; [Uni] is also monophonic, but uses all available voices which are slightly detuned to each other.

Detune  
Sets the amount of detuning in [Uni] mode. Turn to the left for larger intervals between the voices.

Glide  
Switches portamento on or off (see also [Speed]).

Speed  
Adjusts the amount of portamento, i.e. the time that passes until a newly received MIDI note’s pitch is gradually reached.

Pitchbend  
Sets the range of the MIDI pitchbend wheel in semitones.
**Vibrato**

Crucial to this instrument’s sound is the vibrato effect. It is produced by mapping an LFO’s signal onto the instrument’s pitch. The LFO waveform, its frequency, and the amount of pitch modulation can be controlled here, providing settings that differ from the normal idea of musical vibrato but generate impressive sounds nonetheless.

- **Shape** Selects the waveform of the LFO whose signal is used to modulate the instrument’s pitch for vibrato effects.
- **Rate** Sets the frequency at which the vibrato LFO oscillates.
- **Width** Sets the pulse width of the vibrato LFO; turn to mid position for a symmetric waveform.
- **Amount** Sets the amount of vibrato. Turn to the left for no vibrato, to the right for a pitch modulation of the range adjusted by [Range].
- **Range** Controls the absolute vibrato amount in semitones (see also [Amount]).

**Amplitude and Modulation Envelope**

These two envelope generators, placed to the left (modulation envelope) and the right (amplitude envelope) of the [Voice] and [Vibrato] sections, shape the instrument’s amplitude and modulate the feedback amount of the [Sub Oscillator] as well as the formant frequency of the [Formant Oscillator]. They operate as normal ADSR envelopes, but offer additional re-triggering options, key-scaling, and adjustable MIDI velocity sensitivity.

- **Mode** Selects the way the envelope generator reacts to a new gate signal when the previous gate is not yet closed. In [Leg] mode the envelope generator doesn’t react to the new gate signal; in [Ret] mode it is re-triggered, using the current envelope level as a starting point; in [Rst] mode the generator is also re-triggered, but reset to its initial level. This control shows no effect if [Voice][Voice Mode] is set to [Poly].
- **Attack** Sets the attack time of the envelope generator.
- **Decay** Sets the decay time of the envelope generator.
- **Sustain** Sets the sustain level of the envelope generator.
- **Release** Sets the release time of the envelope generator.
- **Key** Sets the amount and polarity of key scaling applied to the envelope generator’s transition times. Turn to the left for faster transitions at low pitches, to the right for slower transitions at low pitches.
Velocity          Sets the amount of MIDI velocity influence on the envelope’s amplitude. Turn to the left for constant amplitudes that are independent of velocity, and to the right for full velocity sensitivity.

Sub Oscillator

This oscillator creates the fundamental MIDI pitch which generates four sub-harmonics below this frequency. The ratio of the harmonics to the main pitch and their amplitudes can be controlled individually, similar to a standard additive synthesizer. A feedback feature provides basic waveform modulation, crossfading smoothly from a sine wave to one that sounds like a sawtooth waveform.

Pitch          Controls the pitch shift of the sub oscillator’s main frequency; there is a coarse (top) and a fine (bottom) control. As the sub oscillator generates harmonics below the main frequency, the pitch control shifts those sub tones to a usable frequency.

Harmonic       Select four sub harmonics of the main frequency. Their volume is controlled by the respective [Amplitude] control.
A/B/C/D

Amplitude      Select the amplitude of the corresponding sub harmonic adjusted by the [Harmonic] control.
A/B/C/D

Feedback Amount  Sets the amount of feedback applied to the sub oscillator internally. Turn to the left for a undistorted sine-like sound, to the right for a saw-like signal.

Envelope       Controls the modulation amount applied to the [Feedback amount] by the [Modulation Envelope].
Modulation Amount
Formant Oscillator

This particular oscillator is made up by a simple sine waveform; however, the adjustable frequency band doesn't move with the oscillator's pitch, but remains stable as a formant of the sound. By moving this frequency band you can achieve vowel filter-like effects.

Pitch Controls the pitch of the main frequency; there is a coarse (top) and a fine (bottom) control.

Formant Frequency Adjusts the frequency of the oscillator's formant; this formant is not changed by the main pitch.

Envelope Controls the amount and polarity of modulation applied to the [Formant Frequency] by the [Modulation Envelope]. Turn to the left for inverse modulation, i.e. low formant frequencies at high modulation signals, and to the right for normal modulation, i.e. high formant frequencies at high envelope levels.

Mix and Output

In this section the signals of both oscillators can be mixed, positioned in the stereo field, and leveled.

Mix Crossfades between the sound of the sub oscillator (at the left) and the signal of the formant oscillator (at the right).

Spread Sets the amount of displacement within the stereo field applied to the instruments' voices. Turn to the left for a mono signal, and to the right to pan each voice individually; this is particularly impressive if [Voice][Voice Mode] is set to [Uni].

Gain Sets the output gain.
Reverb

The high-quality reverb unit is contained within the panel's B view. It can further enhance the sound's spatial characteristics. When not in use it should be turned off by the [Power] control to save CPU power.

- **Size**: Sets the size of the virtual reverberation room.
- **Symmetry**: Places the signal in the virtual reverberation room. Turn to the left or right to move the signal away from the center.
- **Diffusion**: Sets the amount of reverb signal diffusion. Turn to the right for a less echoic sound.
- **Release**: Adjusts the time that passes before the reverberation sound has decayed.
- **Spin**: Sets the amount of modulation applied to the reverb. Technically, the modulation affects the delay time of the delay modules on which the reverb is built.
- **Frequency**: Sets the rate of the LFO used as modulation source (see [Spin]).
- **High Cutoff**: Sets the cutoff frequency of the lowpass filter that is damping the high frequencies.
- **High Damp**: Sets the amount of damping applied to the frequencies above the [High Cutoff] frequency.
- **Low Cutoff**: Sets the cutoff frequency of the highpass filter that is damping the low frequencies.
- **Low Damp**: Sets the amount of damping applied to the frequencies below the [Low Cutoff] frequency.
- **Mix**: Crossfades between the unprocessed, dry signal (at the left) and the reverberated, wet sound (at the right).
- **Power**: Switches the reverb unit on or off. Turn off to save CPU power when the reverb is not in use.
Aerobic is a step sequencer that controls a virtual analogue drum synthesizer. The instrument produces tight, innovative sounds far beyond the range of traditional drum computers. This, combined with the sequencer's capacities and the mixer's flexible routing options, makes Aerobic a versatile beat production environment that can be used in live performances.
The drum synthesizer contains six similar, independent units (selectable by the tabs at the top of the panel). Each unit combines an oscillator and a noise section into one signal that can be equalized before it is sent to the master mixer. The sequencer (in the middle of the panel) contains two tracks for each sound unit, selectable via the same tabs as the units themselves. The filled rectangles within the sequencer’s display represent the unit’s trigger signals and their velocity; the unfilled rectangles form a modulation track whose signal can be used to change nearly every parameter of the sound engine and mixer over time: Use the [Modulation] switch in the unit’s master section to select the destination of the modulation. The master mixer provides classical mixing parameters for each unit (solo/mute, pan, and, of course, level), along with controllers to adjust the complete ensemble’s reaction on MIDI messages. Each unit can be triggered by a selectable MIDI note; on a more complex level, note messages can recall complete ensemble snapshots.

**Sound Engine**

The drum synthesizer is built by two sound generators, an equalizer, and a master section that also controls modulation routing. While the oscillator part (on the left side) is based on sine waveforms with frequency modulation capacities, the noise part (on the right side) contains a white noise generator with a multi-mode filter. The mixed signal is sent through an EQ and (within the master section) a final saturator unit before it is passed on to the mixer.

<table>
<thead>
<tr>
<th>Oscillator</th>
<th>Envelope</th>
<th>Selects the operation mode of the envelope shaping the unit’s amplitude. [Lin] activates a standard AD envelope whose transition times are controlled by the [Attack] and [Decay] knobs. When in [Roll] mode, this envelope is re-triggered fast until the next beat; the [Attack] knob in this case also controls the re-triggering frequency. [Roll+Lin] adds both signals of the modes described above. [Noise Env] uses the envelope of the Noise section (see below).</th>
</tr>
</thead>
<tbody>
<tr>
<td>Attack</td>
<td></td>
<td>Sets the time that passes until the amplitude envelope reaches its peak. In [Roll] mode (see [Envelope]) the knob also controls the rate at which the envelope is re-triggered.</td>
</tr>
<tr>
<td>Decay</td>
<td></td>
<td>Sets the time that passes after the amplitude envelope has reached its peak before it decays to silence.</td>
</tr>
</tbody>
</table>
**Oscillator** Selects the operation mode of the oscillator. While [Sin] represents a standard sine wave, [Sin2] activates a squared sine wave with a different frequency spectrum. Similarly, [FM2] selects the squared signal of [FM] which is generated by a sine oscillator modulating the frequency of another one. (This frequency modulation does not interfere with the modulation controlled by [F-Mod], [F] and [Fmod].) [Phase] uses the output of a phase oscillator.

**F-Mod** Selects the source signal used to modulate the main oscillator's frequency. While [Osc Env] and [Noise Env] select the respective amplitude envelopes, the [Sine], [Tri] and [Random] entries use independent oscillators whose frequency can be adjusted with [Rate].

**F** Sets the base frequency of the main oscillator.

**FMod** Sets the amount of frequency modulation applied to the main frequency by the selected source signal.

**Rate** Sets the frequency of the independent oscillator modulating the main oscillator’s frequency.

**Mix** Sets the ratio of the oscillator section’s output and the noise section’s sound in the signal that is passed on to the equalizer.

**Envelope** Similar to [Oscillator][Envelope], applied to the noise generator’s filter.

**Attack** Similar to [Oscillator][Attack], applied to the noise generator section.

**Decay** Similar to [Oscillator][Decay], applied to the noise generator section.

**Noise** Selects the operation mode of the noise section. [White] uses unfiltered noise, [White Mod] modulates the noise generator’s algorithm by the noise section’s envelope signal.

**Filter** Selects the type of 2-pole filter applied to the noise. Highpass, bandpass, and lowpass filters are available, providing 24 dB damping per octave.

**Freq** Sets the center frequency of the filter.

**Peak** Sets the amount of modulation applied to the filter’s center frequency by the envelope.

**Res** Sets the amount of filter resonance.

**EQ**

**Hz** Sets the frequency of the equalizer.

**dB** Sets the amount of volume boost (or cut) applied to the adjusted frequency.
Master Modulation

Selects the target of the sequencer’s modulation track. The modulation shows no effect until the [Track] button is pushed.

Track

Activates the modulation of the target selected by the sequencer’s modulations track.

Amp

Sets the amplitude of the signal before it is routed to the final shaper unit (see [Shape]).

Shape

Selects the operation mode of the shaper unit. [Polysat], [Sinesat], and [Hypersat] saturate the signal with tube-like effects; the effect increases the more the signal is amplified before (see [Amp]). [Clean] doesn’t perform any compression; [Amp] simply controls the amount of amplification before the signal is routed to the master mixer.

Sequencer

The sequencer provides two tracks for each of the six drum synthesizer units: a gate pattern and a modulation track. The gate pattern determines the trigger signals and their velocity. The modulation track signal can be routed to any parameter of the sound engine (see [Sound Engine][Master][Modulation]). A roll mode bar provides three different roll modes for fast re-triggering of a drum sound.

Tempo

Selects the tempo of the track: Each step of the sequence can be interpreted as sixteenth note, etc. Thus, the sequencer is always synchronized to the master MIDI clock; use the host sequencer or Reaktor’s internal MIDI clock to start the sequencer. (See also [Global Tempo].)

Global Tempo

Sets the [Tempo] value of all six tracks.

Swing

Sets the amount of swing, i.e. the amount by which every second step of the sequence is delayed to shuffle the strict MIDI rhythm.

Roll Factors

Sets the number of times the trigger signal is repeated if the [Roll Mode] is set to the respective colors.

Init All

Deletes all sequence patterns and modulation tracks and sets [Swing] to its default values.

Track

Selects the track that can be edited within the [Edit Display]

Step Count

Displays the number of the current step (1 to 16). If the gate is off, the number is dark; if the gate is on, the number is light. This can be helpful when editing the modulation track.
**Edit Display**
Displays the trigger pattern (filled rectangles) as well as the modulation track (unfilled rectangles), depending on the [Track] setting. Clicking within the display allows the patterns to be edited. High values of the trigger pattern represent high velocity; in the modulation track they cause the modulated knob to turn to the right, and low values turn the knob to the left.

**Roll Mode**
Selects how often the trigger signal is sent. Normally, it is only sent once per beat; by clicking with the mouse one can step through three differently colored modes where the trigger signal is sent more often (see [Roll Factors]).

**Loop**
Controls the length and position of the played sequence: Only those steps within the rectangle are used. Drag the ends of the rectangle to adjust the loop’s start and end points. A second, smaller bar represents the current read out’s position.

**Master / Mixer**
This section has two functions. First, it mixes the six drum synthesizers down to a single signal – or to four signals if [Single Outs] is activated. Second, it controls the snapshots of the complete ensemble as the sound engine and the sequencer are slaved to this part of the instrument. An advanced recall system allows for fast changes of sound/pattern settings via a single MIDI note, making the complex drums computer-controllable from a keyboard in live stage use.

**Mixer**
- **Level**
  Sets the volume of the sound unit.
- **Solo**
  Switches the sound unit to solo playing, i.e. mutes all other units.
- **Mute**
  Mutes the sound unit.
- **Pan**
  Positions the sound unit’s mono signal within the stereo field.
- **Wide**
  Enhances the spatial appearance of the sound unit.
- **Ext. Learn**
  Activates the learn feature. When pressed, the next MIDI note will be assigned to this track and can be used as external trigger signal, in addition to the internal gate signals of the sequencer. (See also [External].)
- **Output**
  Selects to which of the four stereo outputs the sound unit is routed. This shows no effect until [Single Outs] is activated.
Snapshots

Power
Switches the snapshot handling on or off.

Key
Turns on or off snapshot recall by external MIDI note messages. See [Root Note] and [Root Snap] for details.

Quantize
Turns quantization of external MIDI notes on or off. When on, incoming MIDI messages will be synchronized to a pattern that is selected by [Quantization Select].

Quantization Select
Selects the quantization pattern to which external MIDI messages can be synchronized.

 Snapshot
Recalls a snapshot of the master mixer. Since all other components are slaved to this one, storing or recalling a snapshot here affects all other instruments, i.e. the sound units and sequencer.

Root Snap
Sets the snapshot number recalled when the MIDI note adjusted by [Root Note] is received; [Snap Via Key] has to be activated. The note above [Root Note] recalls the snapshot that follows on the [Root Snap] etc.

Root Snap Learn
The first snapshot recalled after pressing this button will be used as new [Root Snap].

Root Note
Sets the external MIDI note that, if [Snap Via Key] is on, recalls the snapshot adjusted by [Root Snap].

Root Note Learn
The first MIDI note received after pressing this button will be used as new [Root Note].

Store
Stores the current settings of the complete ensemble to the current snapshot number (see [Snap]). If [Store To Next Snap] is activated, the subsequent snapshot number will be used to store the data. Any data previously stored there will be overwritten. Therefore, one should start a completely new bank of snapshots when working on a project.

Store +1
If activated, upon pressing [Store] the settings of the complete ensemble will not be saved to the current snapshot as displayed by [Snap] but to the next one.

Level

Master
Sets the master volume.

Velocity
Slaves the [Master] control to the velocity of incoming MIDI notes used to recall snapshots.

Single Outs
Switches on or off the sound units’ routing to different outputs. When off, all sound units are mixed to one stereo signal; when on, there are four stereo outputs to which the six sound units can be individually routed (see [Output]).

External
Switches on or off the triggering of the sound units by external MIDI notes. When off, the sound units are only triggered by the internal sequencer. (See also [Ext. Learn].)
This drum computer is “massive” in at least two ways. First, it contains a vast range of signal-shaping capacities: samples in the six drum tracks don't determine the instrument's sound (like in a standard drum machine), but only provide the material from which the beats can be sculpted. Envelopes, filters, and a potent grain re-synthesis algorithm mangle the fundamental sound until it is completely different, but still musical. Second, these versatile sound editing features are combined with an advanced step sequencer system offering copy and (looped) paste functionality, three different roll modes, a triplet mode, independent loop length for each of the six drum tracks, three modulation tracks whose signal can be routed to nearly every parameter of the sound engine – the list of features could be continued.

Yet, those capacities are not hidden behind an endless array of knobs and faders that prevent productive working. The panel is optimized for usability and fast access to all controllers, making Massive to a powerful sound design workstation. And at the same time – thanks to a complex and glitch-free snapshot recall system – Massive can be used in live performances, or as a slave to a master song sequencer that changes the snapshots automatically.
Control

At the top of the panel you’ll find the instrument’s control section. On the left, an edit mode section defines how the various step sequencer displays react to mouse actions. The copy and paste controls are here too. Next to it is the snapshot management system, followed by the quantization and timing controls, and the parameters for external send effects. Finally, there is the master output knob to adjust overall volume.

There are two external effects: a delay unit, and a lo-fi reverb to enhance the sound spatially. Both effects have three parameters that can be controlled from Massive’s main panel. Additional parameters can be edited in the effects’ own panel, which also contains a normalizer and equalizer. Press {Ctrl}+{2} to go to a second panel set where the effects are displayed; press {Ctrl}+{1} to return to the main panel.

**Edit**

**Edit Mode**

Selects the way the various step sequencer displays react to mouse actions. When [Draw] is selected, the mouse can set each step value (see also [Lock] and [Sequencer][Value Display]). When [Copy] is activated, an area of steps can be selected with the mouse that is automatically copied to the [Edit Buffer]. In [Paste] mode the buffer’s data is copied back to any area selected with the mouse; if the paste area is longer than the buffer’s content, the material to be pasted is looped. [Remote] enables the separate [Copy!] and [Paste!] buttons.

**Copy!**

If [Remote] is selected as [Edit Mode], pressing this button activates the same behavior of the step sequencer displays as the separate [Copy] mode of the [Edit Mode]. This button can easily be activated by pressing the (C) key on the computer keyboard (i.e. MIDI note 52). Thus, one can quickly edit the sequencers’ data with one hand on the keyboard and one on the mouse. (See also [Paste!] and [Lock].)

**Paste!**

If [Remote] is selected as [Edit Mode], pressing this button activates the same behavior of the sequencer displays as the separate [Paste] mode of the [Edit Mode]. This button can easily be activated by pressing the (V) key on the computer keyboard (i.e. MIDI note 53). One can quickly edit the sequencers’ data with one hand on the keyboard and the other on the mouse. (See also [Copy!] and [Lock].)

**Lock**

Keeps the mouse locked on the selected sequencer step in [Draw] mode (see [Edit Mode]). This can also be activated by pressing the (Z) key on the computer keyboard (i.e., MIDI note 48).

**Edit Buffer**

Displays the content of the buffer into which data is copied in [Copy] mode and that is used in [Paste] mode (see [Edit Mode]).
With the left mouse button, a snapshot slot number can be selected; by pressing the right mouse button, the current instrument settings (including all sequencer data) is stored into this slot.

Displays a list of the available snapshots; selecting a snapshot with the mouse results in recalling its data, including all sequences, but playback is not interrupted.

Selects whether the snapshots are only recalled via internal signals or if external control signals received at the instrument’s [Snap] port are recognized, too. This allows you to connect to a master song sequencer.

Switches between the quantization controls and the timing parameters to be displayed. One page has the [Quantization Select], [Shuffle] and [Grid] parameters, the other contains the [Timing] controls to adjust a micro delay for each track.

Selects one of twelve quantization presets. Each preset ranges over sixteen steps; the higher the displayed value, the more delay is applied to this step. The first preset, for example, alternates between low and high values, so every second step will be delayed, resulting in a standard off-beat shuffle. The presets only define relative times; the effective delay time at maximum values is set by the [Shuffle] control.

Scales the preset of the [Quantization Select] control. Turn to the left for no quantization (independently of the selected preset), to the right for full delay times. See also the [Sequencer] section for interaction with triplet modes.

Selects the grid of the sequencer displays; it does not affect the sound.

Sets the micro delay for each track.

Controls the cut-off frequency of the low-pass filter placed within the feedback circuit of the external delay unit.

Crossfades between pre-delay gating (at the left) and post-delay gating (at the right).

Gates the signal of the external delay effect. Turn to the left to close the gate, turn to the right to open it. Depending on the [A1] setting, the audio signal will be gated before it is sent into the delay effect or after the effect unit. This parameter can be modulated (see [A/B 2 Modulation Select]).

Controls the internal cutoff of the spatializer. Use this to alter the color of the effect sound.
B2 | Controls the internal resonance of the spatializer. Use this to color the effect sound.
B3 | Gates the signal of the second external effect after the unit. Turn to the left to close the gate, turn to the right to open it.
A/B 1/2/3 | Selects the modulation track whose signal modulates the value of the respective controller.
Return A/B | Sets the level of the signal that is returned from the first / second external effect; there is an additional [Mute] button that switches the sound completely off.

Output
Master Volume | Controls the master volume of the instrument. An additional [Mute] button switches the sound completely off.

### Modulation

The three step sequencers of this section don’t trigger samples but function as modulation sources to change a sound engine parameter synchronized to the six sample sequencers. Each of the modulation tracks is identified with a color that can be selected within the various modulation source selection controls (e.g. below the Transpose control of the Master section). Normally, you adjust the modulation amount below the source selection control of the modulated parameter.

| Track Select | Switches between modulation track 1 (blue), 2 (green) and 3 (orange). All three tracks can be used to modulate various parameters of the instrument, selectable in the various modulation source controls.
| Sequencer Display | The modulation tracks’ step sequencers act like the one described in the [Sequencer] section; the only difference is the absence of a roll mode.
| Half Tempo | Switches between normal and half speed of the resp. track read out: when pressed, each step is interpreted as a eight note; otherwise each step is interpreted as a sixteenth note. (See also [Sequencer][Triplet Display].)

### Sequencer

Each step sequencer display provides three rows of controllers. At the top there is the main display with 32 steps, read out as sixteenth notes. They represent gate values, i.e. the step’s velocity. When the gate is completely closed (i.e. the bar is a pulled to the bottom), the sampler unit is not triggered at all.

While the normal gate values are controlled with the left mouse button, holding down the right mouse button and moving the step’s value up and down
enables rolls, represented as different colors of the step's bar. In roll mode, the sample is not only triggered once at the beginning of the step, but several times within the step's time.

Below the main step display there is triplet mode control. A yellow bar represents the normal mode, and a shorter red bar appears in triplet mode. Click on the left side of the bar for triplet mode, click on the right for regular mode. In triplet mode, every group of four notes (i.e. steps 1-4, steps 5-8 etc.) can be interpreted as eighth triplets; in this case the last note of the group is not played.

The roll modes depend on the triplet control as seen in the following table:

<table>
<thead>
<tr>
<th>Mode</th>
<th>Triplet</th>
<th>Roll 1</th>
<th>Roll 2</th>
<th>Roll 3</th>
</tr>
</thead>
<tbody>
<tr>
<td>Duolen</td>
<td>No roll</td>
<td>Sixteenth notes</td>
<td>Thirty-second notes; the step is triggered twice</td>
<td>Thirty-second triplets; the step is triggered three times.</td>
</tr>
<tr>
<td>Roll 3</td>
<td>Thirty-second note triplets. Every first and second step are quantized differently; setting two subsequent steps to this roll mode results in the displayed pattern. An alternation of this roll mode and no roll mode for two subsequent steps, combined with a regular shuffle of about 66%, thus leads to sixteenth triplets.</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Triplet</td>
<td>No roll</td>
<td>Eighth triplets.</td>
<td>Sixteenth triplets; the step is triggered twice.</td>
<td>Thirty-second triplets; the step is triggered four times.</td>
</tr>
<tr>
<td>Roll 3</td>
<td>Similar to Roll 2.</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

In the last row each track has a loop control, dictating which sequencer area loops when the instrument plays. Click and drag the mouse to select the loop area. A small marker indicates the current read-out position within the loop.

Value Display Displays the main sequence steps. The height of each step's bar represents the velocity of the gate signal produced when the sequencer is running. Depending on the [Control][Edit][Edit Mode], the values can be drawn manually, copied to the edit buffer, or pasted from that buffer using the left mouse button. Movements with the right mouse button pressed activate the roll mode for each step independently, changing the step's color.
Triplet Display
Switches between normal read-out (the steps are interpreted as sixteenth notes) or triplet read-out (the steps are interpreted as eighth triplets) of the group of four steps above each controller. In triplet mode, the fourth note is not played. See the explanations above.

Loop Display
Controls the length and position of the loop to be played. A moving marker shows the position of the read-out when the sequencer is running.

Sound Engine
The sound engine contains a master section and six independent sampler units. The master section is located to the right of the modulation section; it sets global sample select offset and scales other parameters for all samplers simultaneously. If, for example, [Master][Transpose] is set to 12, all transpose controls of the six drum tracks are scaled from 0 to 12 semitones; if the master control is set to 0, transposition is switched off for all tracks.

The parameters that control the samplers are grouped across four pages. The main page contains the sampler module itself where you load sample files; on the panel it is represented by the sample’s waveform. Further controls select the sample from the map and adjust the pitch shift. The envelope section controls the sample’s amplitude. The parameters of this page can be used to fine-tune the sample, particularly the influence of the gate velocity on attack and decay times. The filter section contains a low-pass and a high-pass filter whose sound can be smoothly crossfaded. The grain section, finally, controls the grain re-synthesis of the sampler; it is only available in the upper three sampler tracks. Here the speed of the sample traversal can be controlled as well as the grain size, which greatly influences the sound of the re-synthesis algorithm.
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Sample Select</strong></td>
<td>Adjusts an offset for all [Sampler][Sample Select] controls of the six independent sampler tracks.</td>
</tr>
<tr>
<td><strong>Sample Select Modulation Source</strong></td>
<td>Selects the modulation track that modulates the [Sampler][Select] parameter. (See also [Sample Select Modulation Amount].)</td>
</tr>
<tr>
<td><strong>Sample Select Modulation Amount</strong></td>
<td>Controls the amount of modulation applied to the [Sampler][Select] parameter by the track selected in [Sample Select Modulation Source].</td>
</tr>
<tr>
<td><strong>Transpose</strong></td>
<td>Adjusts an offset for all [Sampler][Transpose] controls of the six independent sampler tracks. (See also [Transpose Scale].)</td>
</tr>
<tr>
<td><strong>Transpose Modulation Source</strong></td>
<td>Selects the modulation track that modulates the [Transpose] parameter. (See also [Transpose Modulation Amount].)</td>
</tr>
<tr>
<td><strong>Transpose Modulation Amount</strong></td>
<td>Controls the amount of modulation applied to the [Transpose] parameter by the track selected in [Transpose Modulation Source].</td>
</tr>
<tr>
<td><strong>Decay Scale</strong></td>
<td>Scales all decay times adjusted independently for each sampler track with the [Sampler][Decay] control.</td>
</tr>
<tr>
<td><strong>Cutoff Scale</strong></td>
<td>Scales all filters cut-off frequencies adjusted independently for each sampler track with the [Sampler][Cutoff] control.</td>
</tr>
<tr>
<td><strong>Drive Scale</strong></td>
<td>Scales all pre-filter saturation drive amounts adjusted independently for each sampler track with the [Sampler][Drive] control.</td>
</tr>
<tr>
<td><strong>Transpose Scale</strong></td>
<td>Scales all sample transpositions adjusted independently for each sampler track with the [Sampler][Transpose] control. (See also [Transpose].)</td>
</tr>
<tr>
<td><strong>Grain Scale</strong></td>
<td>Scales all grain sizes adjusted independently for each sampler track with the [Sampler][Grain] control. The grain parameter is only available for the upper three sampler tracks.</td>
</tr>
<tr>
<td><strong>Reset</strong></td>
<td>Restores the values of all sampler tracks and the master section to default values.</td>
</tr>
</tbody>
</table>
The controls for [Pan], [Send Level] and [Track Level] are available in every page. Additional modulation controls are available for some of the controls mentioned above.

Sample Map Editor Displays the currently selected sample (see [Sample Select]).

Sample Select Selects the track’s sample played upon a trigger signal of the step sequencer. (See also [Sound Engine][Master][Sample Select].)

Sample Select Modulation Source Selects the source that modulates the [Sample Select] parameter. (See also [Sample Select Modulation Amount].)

Sample Select Modulation Amount Adjusts the amount of modulation applied to the [Sample Select] parameter by the source selected in [Sample Select Modulation Source].

Transpose Sets the amount of transposition applied to the selected sample. For the upper three sampler tracks this transposition does not affect the sample’s playback speed (due to the underlying grain resynthesis algorithm); for the three lower sampler tracks it also changes the playback speed. (See also [Sound Engine][Master][Transpose] and [Sound Engine][Master][Transpose Scale].)

Transpose Modulation Source Selects the source that modulates the [Transpose] parameter. (See also [Transpose Modulation Amount].)

Transpose Modulation Amount Adjust the amount of modulation applied to the [Transpose] parameter by the source selected in [Transpose Modulation Source].

Velocity Adjusts the amount of influence of the sequencer’s gate velocity on the sample’s amplitude. Turn to the left for no influence, i.e. constant maximum amplitude at every gate value; turn to the right for a complete mapping of the gate value onto the sample’s amplitude.

Decay Sets the decay time of the amplitude envelope triggered by a gate event. (See also [Dynamic Decay] and [Sound Engine][Master][Decay Scale].)
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Dynamic Attack</td>
<td>Sets the amount of modulation by the source selected by [Dynamic Source] applied to the attack time of the amplitude envelope. Turn to the left for no modulation; turn to the right for long attack times at high modulation signals.</td>
</tr>
<tr>
<td>Dynamic Decay</td>
<td>Sets the amount of modulation by the source selected by [Dynamic Source] applied to the decay time of the amplitude envelope. Turn to the left for no modulation; turn to the right for long decay times at high modulation signals. (See also [Decay].)</td>
</tr>
<tr>
<td>Dynamic Source</td>
<td>Selects the source track that modulates the amplitude envelope's attack time and the [Decay] parameter.</td>
</tr>
<tr>
<td>Mute By Track</td>
<td>Selects the mute track. If the specified track receives a gate signal, this track's gate is closed. The feature is particularly useful for programming hi-hats, e.g. the first track plays a closed hi-hat and the second one plays an open hi-hat -- since both tracks mute each other, an open hi-hat sound will be muted when the closed hi-hat sample is triggered. There is also a bypass option to exclude the track from muting.</td>
</tr>
<tr>
<td>Drive</td>
<td>Sets the amount of pre-filter saturation drive. (See also [Sound Engine][Master][Drive Scale].)</td>
</tr>
<tr>
<td>Filter Power</td>
<td>Switches the track's filter on or off.</td>
</tr>
<tr>
<td>Cutoff</td>
<td>Controls the cut-off frequency of the track's filter. (See also (See also [Sound Engine][Master][Cutoff Scale].)</td>
</tr>
<tr>
<td>Cutoff Modulation Source</td>
<td>Selects the sequencer track whose signal modulates the [Cutoff] parameter.</td>
</tr>
<tr>
<td>Cutoff Modulation Amount</td>
<td>Sets the amount of modulation applied to the [Cutoff] parameter by the source selected by [Cutoff Modulation Source].</td>
</tr>
<tr>
<td>Resonance</td>
<td>Sets the resonance of the track's filter.</td>
</tr>
<tr>
<td>Lowpass / Highpass Crossfade</td>
<td>Fades between the signal of a low-pass filter (at the left) and the sound of the high-pass filter (at the right). Both use the frequency and resonance adjusted by [Cutoff] and [Resonance].</td>
</tr>
<tr>
<td>Speed</td>
<td>Sets the speed of sample read out. This is only available for the top three sampler tracks, which use grain re-synthesis.</td>
</tr>
<tr>
<td>Speed Modulation Source</td>
<td>Selects the sequencer track whose signal modulates the [Speed] parameter. This is only available for the top three sampler tracks, which use grain resynthesis.</td>
</tr>
</tbody>
</table>
**Speed Modulation Amount**
Sets the amount of modulation applied to the [Speed] parameter by the source selected by [Speed Modulation Source]. This is only available for the top three sampler tracks, which use grain resynthesis.

**Grain**
Sets the grain size of the re-synthesis algorithm. This is only available for the three upper sampler tracks. (See also [Sound Engine][Master][Grain Scale].)

**Reset**
Sets all controls of the respective controller page to their default values.

**Pan**
Controls the position of the track's signal within the stereo panorama.

**Pan Modulation Source**
Selects the sequencer track whose signal modulates the track's [Pan] parameter.

**Send Level**
Sets the volume of the sampler's signal sent to the external effects. (See also [Control][Effect][Send A/B].)

**Track Level**
Sets the volume of the sampler’s signal sent to the main output. (See also [Control][Output][Master Volume].)
Newscool is a Reaktor classic—now it's completely rebuilt, with an innovative sequencer (at the top) and the characteristic sound engine (at the bottom). The engine consists of a tone generator on the left and a multi-effect unit on the right. The signal is produced by eight parallel oscillator units whose parameters are modulated extensively. The effect unit parameters—providing pitch shifting, delay and filter—are similarly modulated.

The sequencer is based on the Life model developed by John Conway in the 1970s. A two-dimensional pattern is processed in steps: An element of the pattern becomes alive (dark in this implementation) in the following step if three of its eight neighbors are alive in this step; it remains alive in the subsequent one if two or three neighbors are alive in the current one—else it dies (and becomes a light square again). Several patterns emerge over time by this
set of rules: Gliders move over the grid, crosses oscillate in several phases, some objects remain stable and don’t change from step to step while others remain unstable forever. These patterns trigger the sound engine, generating “lively” sequences.

**Life Sequencer**

As explained above the sequencer proceeds from one step to the next one by a set of Life rules that translate the current pattern into the following one. The two-dimensional Life pattern is mapped onto the eight channels of the tone generator by the grid of the [Performer Display]: By using the [Wrap X/Y] controllers this mapping can be modified smoothly. The [Sensitivity] knob also interacts with the trigger signals.

Within the [Board Display] Life patterns can be loaded from a bank of factory presets. These patterns can be altered, or you can build completely new ones. The [Board Display]'s content can be copied to the [Performer Display] manually, at the beginning of the Life evolution or at the beginning of each loop.

<table>
<thead>
<tr>
<th>Loop Display</th>
<th>Shows the process of the loop steps. (See also [Run] and [Length].)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Offset</td>
<td>Sets an offset in steps to the sequencer read-out.</td>
</tr>
<tr>
<td>Length</td>
<td>Adjusts the length of the loop in steps. Since the pattern of the [Board Display] can be copied automatically to the [Performer Display] at the beginning of each loop cycle, the loop length controls how often the performer resets to the initial pattern.</td>
</tr>
<tr>
<td>Step</td>
<td>Selects the step length of the life sequencer in MIDI units, e.g. selecting a sixteenth calculates a new pattern life phase each sixteenth of the MIDI clock.</td>
</tr>
<tr>
<td>Run</td>
<td>Switches the life process on or off. When on, each MIDI clock step (see [Step]) generates a new phase of the pattern according to the life rules (see the instrument description); the result is displayed in the [Performer Display]. The MIDI clock has to be running, or else this button shows no effect.</td>
</tr>
<tr>
<td>Next</td>
<td>Calculates the next life sequencer phase independently of the MIDI clock.</td>
</tr>
<tr>
<td>Copy</td>
<td>Selects at which point the pattern of the [Board Display] is copied to the [Performer Display]: manually (by pressing the [To Performer] button), at the start of the sequencer when the [Run] button is pressed, or at the beginning of each loop cycle (see [Length]).</td>
</tr>
<tr>
<td>To Performer</td>
<td>Copies the pattern of the [Board Display] to the [Performer Display].</td>
</tr>
<tr>
<td>To Board</td>
<td>Copies the pattern of the [Performer Display] to the [Board Display].</td>
</tr>
</tbody>
</table>
Board Display  This is a buffer where life patterns can be loaded from the preset list (see [Presets]), edited, or randomly generated. You can draw patterns directly into the display with the mouse.

Presets  Selects a pattern from a list of factory presets, which can then be loaded into the [Board Display] by pushing the [Load] button.

Load  Copies a pattern from the list of factory presets into the [Board Display].

Clear  Deletes the current pattern of the [Board Display].

Random  Randomly generates a pattern within the [Board Display].

Size X/Y  Sets the size of the [Board Display]. When the pattern is copied to the [Performer Display], the size parameters are also adapted to the performer.

Performer Display  Shows the current life phase; its pattern is also used to calculate the next phase. It cannot be edited, patterns can only be copied to it from the [Board Display] (see also [Copy] and [Length]). The grid behind the pattern is used to map the two-dimensional pattern onto a one-dimensional rhythmic sequence (see [Wrap X/Y]).

Wrap X/Y  Controls the projection of the pattern onto the audible sequence; the ratio between horizontal and vertical wrap parameters is visible as a grid within the [Performer Display].

Offset  Adds an offset to the [Wrap X/Y] parameters, thus altering the sequence by shifting it in time.

Sensibility  Determines how many trigger signals are generated from the pattern of the [Performer Board]. Turn to the right for dense trigger sequences, turn to the left for the opposite effect.

**Newscool**

The sound engine consists of a tone generator (in the parameter list below referred to as TG) and a multi-effect unit. Both achieve their characteristic sounds via vast modulation of their parameters by two simple LFOs. Those parameters control eight independent synthesizer tracks that are triggered by the [Life Sequencer]; each of the tracks can be muted. The [Random] button sets all those parameters to random values; within the [TG / Effect] Poly Control areas they can be controlled manually. The parameter shown within these displays is selected using [TG / Effect Parameter Select] controls.

TG Poly Control  Sets the parameters for the tone generator. There are eight bars, one for each track; the value can directly be drawn into the display. The parameter displayed is selectable by [TG Parameter Select].

TG Mute Track  Switches the tracks’ tone generators individually on or off.
**TG Parameter Select**
Selects which parameter of the tone generator is displayed and edited within [TG Poly Control]. There are six parameters available: Pitch, Kick Amount, Frequency Modulation Amount, Ring Modulation Amount, Decay Time and Amplitude.

**TG Parameter Modulation**
Displays the modulation value for each parameter; by clicking into the display the modulation of the respective parameter can be switched on or off. For modulation, a sine LFO is used (see [TG Modulation Rate/Depth/Phase]).

**TG Modulation Rate**
Sets the speed of modulation in sequencer steps.

**TG Modulation Depth**
Sets the amount of modulation.

**TG Modulation Phase**
Sets the phase of the sine LFO.

**Pitch**
Sets the absolute range of the pitch modulation. This is a bipolar control: turn the knob to the left for inverse modulation and to the right for normal modulation. There are individual (relative) values for each track adjustable in the [TG Poly Control].

**FM**
Sets the absolute amount of frequency modulation. There are individual (relative) values for each track adjustable in the [TG Poly Control].

**Decay**
Sets the absolute decay time. There are individual (relative) values for each track adjustable in the [TG Poly Control].

**Drive**
Sets the amount of saturation drive applied to the tone generator’s signal.

**Effect Poly Control**
Sets the parameters for the tone generator. There are eight bars, one for each track; the value can directly be drawn into the display. The parameter displayed is selectable by [Effect Parameter Select].

**Effect Mute Track**
Switches the tracks’ effect units individually on or off.

**Effect Parameter Select**
Selects which parameter of the effect unit is displayed and edited within [Effect Poly Control]. There are six parameters available: pitch shift amount, pitch shift grain size, pitch shift delay time, filter frequency, decay time, and amplitude.

**Effect Parameter Modulation**
Displays the modulation value for each parameter; by clicking on the display the modulation of the respective parameter can be switched on or off. A sine LFO is used for modulation (see [Effect Modulation Rate/Depth/Phase]).

**Effect Modulation Rate**
Sets the speed of modulation in sequencer steps.

**Effect Modulation Depth**
Sets the amount of modulation.
Effect Modulation  Sets the phase of the sine LFO.
Phase
Filter  Sets an absolute offset to the effect’s filter frequency, shifting the individual values of each track that can be edited in the [Effect Poly Display].
Feedback  Sets the level of the signal that is routed from the effect’s output back to its input.
Decay  Sets an absolute offset to the effect’s decay time, shifting the individual values of each track that can be edited in the [Effect Poly Display].
Mix  Controls the ratio between the unprocessed, dry sound (at the left) and the effect’s wet signal (at the right).
Level  Sets the instrument’s master level.
Mute  Mutes the complete instrument.
Random  Randomly sets all parameters of each track within [TG Parameter Display] and [Effect Parameter Display].
The Reaktor library classic Sinebeats has undergone an overhaul for Reaktor 5. Sinebeats is a beatbox based on three sine oscillators and a noise generator. Its synthetic nature in combination with the flexible effects section has made Sinebeats a classic for electronic sequence production. Each of the four instruments features a sequencer and individual sound parameters including distortion and filter. Two flexible filters and two delays which are fed via a send/return feature in the mixer add even more motion to the generated beats.

In its new incarnation, Sinebeats got an enhanced mixer with the possibility of routing the individual sound units to single outputs, a two-band equalizer, and a simple compressor for the sum. The sequencers got updated with individual looping, individual clock settings, and the possibility to introduce rolls for every step. You can also record the pitch information via MIDI input. Modulation of sound parameters has undergone a major overhaul, the sine instruments now have multimode filters, and all the instruments now sport an individual overdrive section and an equalizer. A valuable addition to the effects section are the two modulation sequencers that provide dynamic effects sequencing. Also, there is a new snapshot system that enables you to trigger complete snapshots including the sequencer tracks via MIDI note triggers and in sync with the global tempo.
Sequencer

Each of the four instruments is equipped with its own 16-step sequencer with 2 tracks. The first contains the triggers for the sound units. The second track sends modulation data which can modulate different sound parameters in the instrument. A great addition to your sequencing options is the roll/slide track of the sequencer. You can define 2 different rolls per sound unit that can be assigned to individual steps and you also can introduce pitch slides between steps. The sequencer functionality also allows various direction modes, individual tempo settings and individual loop control. Pitch recording via MIDI input has also been added. If you want to hear Sinebeats without triggering the snapshots via MIDI notes (see [Snapshot system]), you have to switch off [Velocity] in the [Master] section.

Sound units  Switch the view between the [Noise] percussion and the four [Sine] synthesizers and their corresponding sequencers.
Rec  Activates velocity/modulation recording
   Hit [Rec] and let the sequencer run. Then play notes on your keyboard to write triggers and velocity values for the instruments. For the [Sine] units the note pitch information will be written into the [Pitch dials].
   Recording does not delete existing events. They remain untouched as long as no new data is coming in via MIDI.
Run  Starts or stops the sequencer.
Pitch dials  Dial in the desired note pitch per step for the [Sine] units. You can also record the values via midi note input (see [Rec]). These are not available for the [Noise] unit.
Init  Completely initializes the displayed unit’s sequencer. This includes deletion of the modulation and velocity tracks, and, in the case of the [Sine] sequencers, resets the Pitch dials.
Direction  Choose between four different direction modes: forwards (->), backwards (<-), and two ping-pong modes (<>-<>, inverted: -><). Tempo  Choose an individual tempo for the currently displayed unit sequencer. These are clock division settings that always keep the sequencer in sync with the global tempo.
Loop bar  The bar above the sequencer grid is for setting up a loop region for the currently displayed sequencer. Drag the start or the end to change length and drag the bar to change position.
Random  Randomizes the modulation- or velocity track of the displayed unit sequencer, respectively.
Track selector
Switches the view between the modulation- and the velocity track.

Event grid
Click into the grid and drag up or down to create modulation events or velocity triggers. Right-click (ctrl-click if you’re on a Mac) to delete events.

Roll settings
Assign the roll speeds for three freely assignable roll mode colours (yellow, blue and red). You can assign roll overdrive values between 2-times and 16-times. In case of the [Sine] units, the red roll is used for pitch slides only.

Roll and slide modes
The bar below the sequencer grid is for defining roll modes for the individual steps. Left-click to create a roll and repeat the click to change the mode. Right-click (ctrl-click if you’re on a Mac) to delete the roll. You can define three roll modes under [Roll settings]. For the [Sine] unit the red mode always stands for a pitch slide, with the [Nois] instrument you can freely define it as a third roll mode.

**Noise synthesizer**

The noise unit features a simple envelope for controlling the volume. You can stack the different outputs of the multi-mode filter and you can also adjust the parameters cutoff, resonance and envelope modulation intensity. Release, cutoff and resonance can be modulated by the second sequencer track. An overdrive / bit reduction effect and a small equalizer are also at your disposal.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Amp-release</td>
<td>Release time of the amplitude envelope.</td>
</tr>
<tr>
<td>Release mod</td>
<td>Bipolar (-/+ ) amount for the modulation track output of the sequencer, altering release time. This can be initialized with the [init mod] button.</td>
</tr>
<tr>
<td>Init mod</td>
<td>Initializes all [MOD] controls of the [Noise] unit.</td>
</tr>
<tr>
<td>Filter stack</td>
<td>Switches for the output of six different filters. The outputs can be stacked. Since the addition of six signals can produce clipping, the overall level of the instrument will be divided by the number of activated filter signals.</td>
</tr>
<tr>
<td>Drive switch</td>
<td>The [Drive] section reduces bit depth and sample rate of the signal, and includes a saturator. Switch on or off with the power button.</td>
</tr>
<tr>
<td>Drive</td>
<td>Controls overdrive intensity.</td>
</tr>
<tr>
<td>Bit</td>
<td>Controls bit depth reduction.</td>
</tr>
<tr>
<td>EQ switch</td>
<td>Switches the 1-band equalizer on or off.</td>
</tr>
<tr>
<td>Freq</td>
<td>1-band EQ frequency.</td>
</tr>
<tr>
<td>Amt</td>
<td>Bipolar EQ cut / boost. -/+ 24 db.</td>
</tr>
<tr>
<td>Cutoff</td>
<td>Filter frequency of the noise filter. Shown in pitch values.</td>
</tr>
</tbody>
</table>
Cutoff mod  Bipolar (-/+), amount for the modulation track output of the sequencer, altering the cutoff frequency. Can be initialized with the [init mod] button.

Reso  Resonance of the noise filter.

Reso mod  Bipolar (-/+), amount for the modulation track output of the sequencer, altering the resonance. Can be initialized with the [init mod] button.

**Sine synthesizers**

The three sine instruments are structured identically. Each of them features a release parameter for the decay time, a pitch parameter and a simple pitch envelope with an intensity- and a release parameter. Here you can modulate the intensity of the pitch envelope and the decay time. Pitch can also be slurred with variable glide. As with the noise instrument, you can use an overdrive/bit reduction effect and a small equalizer. You also get a multi-mode filter with variable cutoff and resonance. Both parameters can be targets of the modulation track of the sequencer.

**Amp-release**  Release time of the amplitude envelope.

**Release mod**  Bipolar (-/+), amount for the modulation track output of the sequencer, altering release time. Can be initialized with the [init mod] button.

**Init mod**  Initializes all mode controls of the [Sine] unit.

**Glide**  Controls glide time. This only works when the sequencer reaches a red [Roll] step. Refer to the description of the sequencer.

**Octave**  Master octave of the Sine unit.

**Tune**  Pitch for the sine oscillator of the Sine unit.

**Fine / Integer**  Toggles pitch control of the sine oscillator between fine and integer mode. In fine mode the range of the dial is +/- 100 cents. These get added to or subtracted from the integer value chosen in integer mode.

**Penv**  Controls the amount of a percussive pitch envelope applied to the sine oscillator.

**Penv mod**  Bipolar (-/+), amount for the modulation track output of the sequencer, altering pitch envelope amount. Can be initialized with the [init mod] button.

**Prel**  Controls the release time of the pitch envelope.

**Prel mod**  Bipolar (-/+), amount for the modulation track output of the sequencer, altering pitch envelope release time. Can be initialized with the [init mod] button.

**Drive switch**  The [Drive] section reduces bit depth and sample rate of the signal. Switch on or off with the power button.
Drive Controls overdrive intensity.
Bit Controls bit depth of the sound. Reduce to introduce harsh aliasing.
Eq switch Switches the 1-band equalizer on or off.
Freq Control of EQ frequency.
Amt Bipolar EQ cut / boost. -/+ 24 db.
FilterMode Click multiple times to switch through the available filter modes. You can choose between low-pass high-pass and band-pass.
Cutoff Filter frequency of the noise filter. Depicted in pitch values.
Cutoff mod Bipolar (-/+ ) amount for the modulation track output of the sequencer altering the cutoff frequency. Can be initialized with the [init mod] button.
Reso Resonance of the multi-mode filter.
Reso mod Bipolar (-/+ ) amount for the modulation track output of the sequencer, altering the resonance. Can be initialized with the [init mod] button.

FX 1 & 2

You get 2 effects units that receive their input from the sends of the [Mixer]. The effects units are identical and offer a stereo delay with an integrated resonant multimode filter, feedback and a return level dial. The filter cutoff can be modulated by an integrated three-waveform, tempo-synced LFO. Both effects units have a small step-sequencer including loop- and tempo control that allow for bipolar modulation of the filter parameters, of [Return level], and [Feedback].

DLY / R Delay Time for the right channel. Units are beats per echo.
DLY / L Delay Time for the left channel. Units are beats per echo.
CUT Cutoff frequency of the multimode filter.
CUT MOD Switches on modulation of the cutoff frequency by the effects unit’s modulation sequencer.
RES Resonance control of the multimode filter.
RES MOD Switches on modulation of the resonance by the effects unit’s modulation sequencer.
Filter mode Click multiple times to switch through the available filter modes. You can choose between low-pass high-pass and band-pass. Right-click (ctrl-click if you’re on a Mac) to choose the low-pass mode directly.
TEMPO Dial in the tempo of the LFO. Units are fractions of one bar in quarter notes.

TEMPO MOD Switches on modulation of the LFO tempo by the effects unit's modulation sequencer.

AMT Control of LFO modulation depth.

AMT MOD Switches on modulation of the LFO depth by the effects unit's modulation sequencer.

LFO waveform Choose between sine, pulse and triangle for the LFO waveform.

Modulation sequencer Click into the sequencer and drag the mouse up and down to change the value of the bipolar sequencer steps. The sequencer output can be routed to the filter [Cutoff] frequency, filter [Resonance], LFO [Tempo], LFO [Amount], and [Feedback].

RANDOM Click to randomize the steps of the modulation sequencer.

Loop bar With the loop bar you can define a region in the sequencer that gets repeated. Drag the start or the end to change length and drag the bar to change position.

DIR Choose between four different direction modes: forwards (->), backwards (<-), and two ping-pong modes (<>), inverted: >-<.

TEMPO Via this menu you can choose a tempo for the effects unit's modulation sequencer. These are clock division settings that keep the sequencer always in sync with the global tempo.

Mixer

The four-channel mixer provides control over [Pan], [Volume], and two effects [sends]. It has a routing system, allowing you to send the different channels either to the master stereo bus or into up to 4 individual stereo busses. This output routing system has to be activated in the [Master] section. The four stereo channels are identical in function and carry the signals of the [Noise] synthesizer and the three [Sine] synthesizers.

Power Switches the respective channel on or off. Use to mute single or multiple sound units.

PAN Dial in the position of the respective sound unit in the stereo field.

VOL Volume of the respective sound unit.

FX 1 Send level to effects unit 1.

FX 2 Send level to effects unit 2.
Output busses: Click multiple times to choose the 4 available stereo busses. Works only if [use single outs] is activated in the [Master] section. Also, this is sensible only if your sound hardware has multiple outputs, or if you route the individual outs of Sinebeats into further Reaktor instruments. Right-click (ctrl-click if you’re on a Mac) to reset to outputs 1 / 2.

**EQ and compressor**

With Sinebeats 2 you also get a little effects section that works on the sum. In terms of signal flow it sits between the [Mixer] and the [Master] section. A two-band shelving equalizer and a simple compressor help to spice up the sum of your Sinebeats tracks.

**EQ**

- **EQ Power**: Switches the sum-EQ on or off.
- **F-LOW**: Frequency of the low-band shelving EQ. Units are Hz.
- **LOW AMT**: Cut or boost for the low-band shelving EQ (+/-20 db).
- **F-HIGH**: Frequency of the high-band shelving EQ. Units are Hz.
- **HIGH-AMT**: Cut or boost for the low-band shelving EQ (+/-20 db).

**Compressor**

- **Compressor power**: Switches the sum compressor on or off.
- **comp**: Dial in the threshold and ratio of compression. These two parameters are combined into one.
- **speed**: Control for the release time of the compressor.
- **soft**: If on, the compressor works in soft-knee mode, meaning that the ratio increases gradually to the selected [comp] level. If off, the compression is applied to only to signals above the threshold.

**Master**

The Master section provides control over master volume, a switch for velocity sensitivity of triggering the sequences via MIDI input, and a toggle for activation of the multiple output routing system.

- **Master**: Controls the master volume of the patch.
- **Velocity**: Toggles velocity sensitivity for triggering of snapshots via MIDI. If you want to hear Sinebeats’ output without MIDI triggering switch this control and the [Snap via key] toggle off.
Use single outs: Activates the three additional stereo outputs. You can route sound into them with the [Mixer].

**Snapshot system**

The snapshot system is a new feature of Sinebeats 2, enabling you to store and recall snapshots from within the patch. The most intriguing feature of this module is the [snap via key] function. When this is active you can trigger complete stored sequences including all sound units via incoming MIDI note data. This happens glitch-free and in real-time. Use it to trigger Sinebeats sequences in a live situation via a MIDI controller. You can also trigger sequences from another sequencer.

- **On**: This is the bypass switch for the snapshot calling/storing system.
- **Snap via key**: Enables snapshot calling via pitch input.
- **Start note**: Dial in the note that mapping of snapshots across the keyboard starts with.
- **Learn start note**: The first MIDI note value received after pressing this button will be used as new [Start note].
- **Start snap**: Sets the snapshot number that is recalled when the MIDI note adjusted by [Root Note] is received; [Snap Via Key] has to be activated.
- **Learn start snap**: The first snapshot recalled after pressing this button will be used as new [Start snap].
- **Key-sync**: Incoming MIDI notes are quantized at the given resolution relative to the global tempo.
- **Key sync on / off**: Enables / disables key-sync of snapshot calling. This quantizes the start of the next triggered snapshot to a metric value between 1/16th and whole notes.
- **SnapShot**: Choose a snapshot number to store to.
- **Store**: Stores the current settings of the complete ensemble to the current snapshot number (see [Snap]). If [Store+1] is activated, the subsequent snapshot number will be used to store the data. Any data stored there before will be overwritten. Therefore, one should start a complete new bank of snapshots when working on a project.
- **Store+1**: If activated, upon pressing [Store] the settings of the complete ensemble will not be saved to the current snapshot as displayed by [Snap] but to the subsequent one.
Sound Generators

Skrewell
Skrewell is an intuitive and visual sound design workstation whose soundscapes can range from meditative atmospheres to crackling harshness. Its sound engine uses eight parallel oscillator sections (channels) that blend into a single, complex signal. This unique construction means that its interface is unlike that of a classic additive/subtractive synthesizer. The [Draw] edit mode allows for standard value adjustment for each channel’s parameters. The parameters are represented by eight vertical bars (one bar for each oscillator section), which control the channels’ oscillators, integrated filters and feedback delays. The [Wrap] and [Rand] edit modes provide special ways of altering your chosen parameter across all eight channels simultaneously. The Skrewell unit structure – and therefore the available parameters for each channel – is different in each of the three operation modes. Four main knobs manipulate the sound globally, mainly by mapping the individual channels’ parameters. Additionally, a large display visualizes the audio output as Lissajous figure.

**Operation Modes**

There are three operation modes, each one based on a unique tone generator system. In Implant Quad mode, each channel consists of a pulse oscillator with subsequent feedback delay; within the delay line a normalizer and a filter alter the signal. Xung Tekh is similar, except the filter is placed before the feedback delay. Subtotal uses a parabolic waveform instead of the pulse waveform and omits the filter completely. The parameters of the tone generators are adjusted in the [Sound Engine] section.

- **Operation Mode** selects the main way of operation.
- **Randomize All Ch.** sets all parameters of all channels to random values. The [Output Volume] should be lowered to prevent unexpected bursts of noise.

**Sound Engine**

This section adjusts the parameters of the tone generators. Depending on the [Operation Mode] setting, a list of the currently available parameters is in the [Parameter Select] display. The selected parameter can then be edited within the [Edit Area], where each bar represents one of the eight parallel oscillator units that form the tone generators.

- **Function** switches between the various parameters that control the channels. Depending on the [Operation Mode], there are different sets of available parameters. The values of the selected parameter are displayed for each oscillator section in the [Edit Area].
Edit Mode  Selects in which way the instrument interprets mouse movements within the [Edit Area]. [Draw] allows direct adjustment of each bar. [Wrap] lowers / raises all bars simultaneously, keeping their ratio constant. If a value exceeds the value range it is mirrored. [Rand] performs random changes on all eight bars.

Edit Area  Displays the selected parameter, with one bar representing the parameter’s value for each of the eight channels. Mouse movements within this area alter those values, controlled by the [Edit Mode].

Display Control  Scales the Lissajous display.

**Master Controls**

These master controls either scale the settings of the [Sound Engine] section (e.g. [Delay Time]) or adjust additional tone generator parameters (e.g. [Flow Amount]). As they affect all eight channels of the tone generators simultaneously they can be used to alter the overall sound.

Oscillator Pitch  Modifies the pitch of all channels. Technically, it controls a mapping function that modulates the values adjusted within the [Edit Area] for each channel. Only very high individual settings will result in high pitches if turned to the left, while less high values will be mapped onto low pitches; move to the right for the opposite effect.

Filter Cutoff  Modifies the filter cutoff frequency of all channels; see [Oscillator Pitch] for technical details.

Delay Time  Modifies the delay time of all channels; see [Oscillator Pitch] for technical details. By turning the knob to the left the delay times can be dramatically shortened, resulting in comb filter-like effects.

Flow Amount  Adjusts various amounts of modulation, depending on the selected [Operation Mode], e.g. frequency modulation amount, amplitude modulation amount etc. Like the [Oscillator Pitch], this knob maps the individual channel’s value. Turn to the left for less modulation and more inertia, turn to the right for the opposite effect.

Output Volume  Sets the master output volume. As slight variations of Skrewell’s parameters might result in extreme volume changes, this control should be handled carefully. There is an additional [Mute] button at the knob’s left.
SpaceDrone

SpaceDrone generates atmospheric pads which range from light rain or howling wind noises to deep and uncanny space sounds. Technically, the instrument is based on 96 parallel voices spread across the frequency spectrum. Each voice consists of a noise generator; the signal’s amplitude is shaped by an envelope, its frequency content gets modified by a bandpass filter, and finally it gets positioned in the stereo field.

Sound Engine

The parameters of the sound engine are in the A panel of the instrument. They control the noise generators, their subsequent bandpass filters, the amplitude shaping envelope and corresponding triggering algorithm, and the pan, gain and damping of the signals.

- **Attack**
  Sets the time that passes until the amplitude envelope reaches its peak after triggering. The [Density] knob controls speed at which the envelope is re-triggered.

- **Decay**
  Sets the time that passes until the amplitude envelope completely fades out after it has reached its peak. The [Density] knob controls speed at which the envelope is re-triggered.

- **Pitch**
  Sets the amount by which the amplitude envelope modulates the voice’s pitch, i.e. the bandpass filter’s center frequency. Turn to the left for inverse modulation — the higher the envelope signal, the lower the pitch. Turn to the right for the opposite effect.

- **Resonance**
  Sets the bandpass filter’s resonance.

- **Fundamental**
  Adjusts the fundamental frequency, i.e. the pitch of the lowest voice.
Offset
Sets the offset of the filter harmonics: All voices are harmonics of the fundamental frequency (see [Fundamental]); all harmonics below the one adjusted here are skipped.

Speed
Controls the rate at which a LFO modulates each voice’s frequency randomly.

Amount
Sets the amount by which the voice’s frequency is changed by the random LFO.

Density
Sets the speed at which each voice’s amplitude envelope is re-triggered.

Random
Sets the randomness of the re-triggering events. Turn to the left for completely regular re-triggering; turn to the right to give each voice a slightly varied re-triggering speed.

Dynamic
Sets the dynamic range of the amplitude envelope. Turn to the left to bind every voice to a constant maximum level; turn to the right to allow some (randomly picked) voices to be quieter.

Pan
Sets the rate at which each voice is rotated within the stereo field.

Random
Sets the randomness of the panning speed. At high values each voice has a slightly different pan rate.

Damp
Sets the amount of damping applied to high frequencies.

Gain
Sets the amount of amplification applied to each voice independently.
The reverb unit is contained within the panel’s B view. It can further enhance the spatial character of the atmospheric pads. When not in use it should be turned off by the [Power] control to save CPU power. Although it is built completely within the new and efficient Reaktor core layer, it is designed to produce high-quality reverberation sounds.

- **Size**
  Sets the size of the virtual reverberation room.

- **Symmetry**
  Places the signal in the virtual reverberation room. Turn to the left or right to move the signal away from the center.

- **Diffusion**
  Sets the amount of diffusion of the reverb signal. Turn to the right for a less echoic sound.

- **Release**
  Adjusts the time that passes before the reverberation sound has decayed.

- **Spin**
  Sets the amount of modulation applied to the reverb. Technically, the modulation affects the delay time of the delay modules on which the reverb is build.

- **Frequency**
  Sets the rate of the LFO used as modulation source (see [Spin]).

- **High Cutoff**
  Sets the cutoff frequency of the lowpass filter that is damping the high frequencies.

- **High Damp**
  Sets the amount of damping applied to the frequencies above the [High Cutoff] frequency.

- **Low Cutoff**
  Sets the cutoff frequency of the highpass filter that is damping the low frequencies.

- **Low Damp**
  Sets the amount of damping applied to the frequencies below the [Low Cutoff] frequency.

- **Mix**
  Crossfades between the unprocessed, dry signal (at the left) and the reverberated, wet sound (at the right).

- **Power**
  Switches the reverb unit on or off. Turn off to save CPU power if the reverb is not used.
Sample Player

BeatSlicer 2
BeatSlicer 2 will separate any waveform into smaller component ‘slices’, which can then be individually tweaked by adjusting pitch, envelope and FX settings. BeatSlicer 2 is designed primarily for drum-loop manipulation, but the extensive range of parameters offers creative possibilities with any material. For a 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. The loop will be scanned and MIDI notes from C-2 (by default) will be assigned to the detected slices.

BeatSlicer 2 is designed to be programmed with MIDI controllers. To assign a MIDI controller to a parameter, use the MIDI learn function on the XY modules on the panel, e.g. the [Pitch] control. You can also assign MIDI controllers to certain other controls that use Multi Picture modules, such as the [Shape] on/off control. To do this, right click on the module and select ‘show in structure’. Then assign a MIDI controller to the hidden button module beneath the Multi Picture module. Be aware that whenever BeatSlicer 2 receives an event from an assigned MIDI controller, it will write to the memory of whatever slice is currently selected. This can be a problem with certain hosts, which send extra MIDI controller data to plug-ins on initialization or when stopping/starting playback. Or it can be a problem if you accidentally move a MIDI controller yourself… The safest thing to do is take regular snapshots of your loop settings.

With the exception of the audio loop itself, all global and per-slice parameters are saved in the host plug-in edit buffer. This means that if you do not change the loaded loop, you do not need to save a new copy of the ensemble. However, as you’ll most likely work with different loops in different songs, you should always use the Reaktor autosave feature. This will create a new copy of the BeatSlicer 2 ensemble and save it with your song.
Global section

The master controls are located at the top of the instrument panel. They control global settings that are applied to the complete sample loop and not to individual slices. Note that both pitchbend and velocity can be assigned to slices individually using the modulation matrix, in which case it is probably best to set the global knobs to zero.

BeatSlicer 2 has four output channels. By default, these are used as two stereo output channels, but can be used as 4 discrete mono channels by enabling the [Mono] button.

<table>
<thead>
<tr>
<th>Power</th>
<th>Switches the complete instrument on or off. Technically, this mutes the output; the instrument is still on and consumes CPU power.</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mono</td>
<td>Switches the mono mode on or off. If on, there are four single outs; if off, there are two stereo outputs. (See also [Out][Pan] and [Out][Out Port].)</td>
</tr>
<tr>
<td>Root</td>
<td>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.).</td>
</tr>
<tr>
<td>Pitchbend</td>
<td>Determines the amount by which the pitchbend wheel affects the pitch of the entire loop.</td>
</tr>
<tr>
<td>Velocity</td>
<td>Specifies the amount by which note-on velocity affects amplitude for all slices.</td>
</tr>
<tr>
<td>Gain</td>
<td>Controls the overall output level in decibels.</td>
</tr>
<tr>
<td>Tune</td>
<td>Transpose the pitch of the entire loop in semitones</td>
</tr>
<tr>
<td>Clear</td>
<td>Resets all settings of the current slice to their default values.</td>
</tr>
<tr>
<td>Copy</td>
<td>Copies all settings of the current slice to an edit buffer (see also [Paste]).</td>
</tr>
<tr>
<td>Paste</td>
<td>Copies all settings from the edit buffer into the current slice (see also [Copy]).</td>
</tr>
</tbody>
</table>
**Loop section**

The large window displays the waveform of the current loop; it is also the place to load loop files. BeatSlicer 2 can slice loops by employing a transient detection algorithm, or by dividing the loop into equal-length sections. With either method, adjusting the Sens (sensitivity) knob will result in more or less slices. To slice the loop into equal length sections, ensure that the BPM button is active, and that the detected tempo (displayed at the bottom of the loop section) is correct. If the tempo isn’t correct, even when adjusted by clicking on the + and - buttons, then the loop will not be an integer number of bars. In this case, you should use the transient detection method (deactivate the BPM button).

Click anywhere on the loop window to select (and audition) selected slices. You can either edit slices individually, or you can choose to edit all slices simultaneously by right clicking on the window. The indicator in the top-left of the Loop section displays the MIDI note assigned to the current slice.

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Waveform Display</td>
<td>Shows the waveform of the current loop. To load a different loop, right-click (PC) / ctrl-click (Mac) on the upper part of the window (where the filename is displayed) and select ‘File’, ‘Load data into table ...’.</td>
</tr>
<tr>
<td>Zoom Bar</td>
<td>Scrolls the viewable area across the entire loop. Click the right mouse button on this scrollbar to zoom in or out.</td>
</tr>
<tr>
<td>MIDI Note</td>
<td>Displays the MIDI note number that is triggering the currently selected slice.</td>
</tr>
<tr>
<td>Track</td>
<td>If activated, the received MIDI notes not only trigger the playback of the slices, but also select them for editing.</td>
</tr>
<tr>
<td>Solo</td>
<td>If activated, only the currently selected slice can be triggered by MIDI notes.</td>
</tr>
<tr>
<td>Sensitivity</td>
<td>Adjusts the amount of slices. If [BPM Switch] is on, this determines whether the loop is sliced into eighth notes, sixteenth notes or thirty-second notes; if the switch is off, the knob controls the sensitivity of the transient detection algorithm.</td>
</tr>
<tr>
<td>BPM Switch</td>
<td>Toggles between automated transient detection (off) and a slicing of the loop into parts of equal length (on).</td>
</tr>
<tr>
<td>BPM Control</td>
<td>Adjusts the tempo of the loop. A tempo is extracted from the length of the sample file; by using the [+] and [-] controls this value can be modified.</td>
</tr>
</tbody>
</table>
Slice Parameters

This section below the waveform display adjusts the parameters of the currently selected slice. These parameters control the slice’s start position and length, its transposition, an amplitude envelope, a hybrid compression / distortion unit entitled [Shape], and a filter. By clicking their headlines, the [Envelope], [Shape] and [Filter] part of the section can be switched on or off. Note that the envelope section, if switched off for each slice, can still be used as a modulation source.

Pitch
- Position
  - Start: Adjusts the start position of the current slice.
  - Length: Adjusts the length of the current slice.

Pitch
- Transpose: Determines the amount of transposition and its direction – up or down.
- Reverse: Switches between forward and reverse playback of the current slice.

Envelope
- Attack: Specifies the time taken to reach full amplitude, as a proportion of slice length. Thus, if set to 50%, the envelope will reach peak value halfway through the slice.
- Decay: Determines decay time and shape as a proportion of the time remaining after the attack phase. At maximum, the envelope will sustain at full amplitude for the entire slice (or the remainder of the slice after the attack phase). When between 50% and maximum, the envelope consists of a sustain phase followed by a decay phase. At less than 50%, there is no sustain period, just a decay stage.

Shape
- Pre: Increases the compressor input level.
- Shape: Determines the compressor gain curve.
- Smooth: Reduces the amount of distortion by smoothing gain changes as it controls the attack and release of the compressor.
- Drive: Saturates the output signal.

Filter
- Mode: Selects the operation mode of the filter unit. Low-pass, band-pass and high-pass modes are available.
- Cut-off: Sets the center frequency of the filter.
- Resonance: Sets the resonance of the filter at the cut-off frequency.

Out
- Gain: Adjusts the output level in decibels
- Aux Send: Sets the level of the auxiliary output port of the instrument.
- Pan: Controls the position of the sound within the stereo field.
- Out Port: Selects the output of the instrument to which the slice’s sound will be routed. Depending on the global [Mono] switch either two stereo ports or four mono ports are available.
Modulation

BeatSlicer 2’s advanced modulation routing allows various parameters to be modulated by a variety of sources (both MIDI and internal). In both modulation sections (A and B) the left-hand box displays the current source, and the right-hand box displays the current destination. Click and drag vertically to change the source or destination. The slider bar in between specifies the amount (and direction) by which the source modulates the destination. For example, to assign velocity to amplitude, select ‘Vel’ as the source, ‘Amp’ as the destination, and set the slider bar to the full right-hand position.

Some modulation sources have a variation denoted by ‘/H’. This option samples the value of the source when triggered. Try assigning the Pitchbend wheel to Pan and repeatedly trigger the sample while modulating the Pitchbend wheel. Change the source to ‘PB/H’ and listen for the difference.

### Sources

<table>
<thead>
<tr>
<th>Source</th>
<th>Description</th>
<th>Modulation Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Vel</td>
<td>MIDI note on velocity.</td>
<td>Unipolar</td>
</tr>
<tr>
<td>PB</td>
<td>MIDI pitchbend wheel.</td>
<td>Bipolar</td>
</tr>
<tr>
<td>PB/H</td>
<td>MIDI pitchbend wheel, sampled at note-on.</td>
<td>Bipolar</td>
</tr>
<tr>
<td>CC1</td>
<td>MIDI controller 1 (the modulation wheel).</td>
<td>Unipolar</td>
</tr>
<tr>
<td>CC1/H</td>
<td>MIDI controller 1, sampled at note-on.</td>
<td>Unipolar</td>
</tr>
<tr>
<td>CC7</td>
<td>MIDI controller 7 (the volume slider).</td>
<td>Unipolar</td>
</tr>
<tr>
<td>CC7/H</td>
<td>MIDI controller 7, sampled at note-on.</td>
<td>Unipolar</td>
</tr>
<tr>
<td>Env</td>
<td>Envelope generator.</td>
<td>Unipolar</td>
</tr>
<tr>
<td>Rnd</td>
<td>Random value generator.</td>
<td>Bipolar</td>
</tr>
</tbody>
</table>

### Destinations

<table>
<thead>
<tr>
<th>Destination</th>
<th>Description</th>
<th>Range</th>
</tr>
</thead>
<tbody>
<tr>
<td>Amp</td>
<td>Slice amplitude</td>
<td>(-100% to +100%)</td>
</tr>
<tr>
<td>Pan</td>
<td>Stereo pan</td>
<td>(-100% to +100%)</td>
</tr>
<tr>
<td>P</td>
<td>Slice pitch</td>
<td>(-12 to +12 semitones)</td>
</tr>
<tr>
<td>Len</td>
<td>Slice length</td>
<td>(-100% to +100%)</td>
</tr>
<tr>
<td>Drv</td>
<td>Overdrive amount</td>
<td>(-60 to +60 decibels)</td>
</tr>
<tr>
<td>Cut</td>
<td>Filter cutoff</td>
<td>(-120 to +120 semitones)</td>
</tr>
<tr>
<td>Aux</td>
<td>Aux send level</td>
<td>(-100% to 100%)</td>
</tr>
</tbody>
</table>
Memory Drum 2

Memory Drum 2 is an advanced sampler that enables the independent configuration of up to 128 samples in a compact, easy-to-use interface. Specifically designed for drum sampling, it features an attack-hold-decay envelope, a range of effects, multiple output channels, and complex modulation options. The intuitive interface allows drum kits to be constructed quickly and easily, yet the extensive range of sound-design options offer vast creative possibilities.
for generating new sounds from your existing samples.

For a quick start, double-click on the sampler window, open the Reaktor sample map editor, and load Memory Drum 2 with a few drum samples. As you trigger the samples from your keyboard, notice that the green box in the Edit section moves to indicate the current MIDI note. Any parameter that you adjust will be stored for that MIDI key. For example, press a MIDI note and then adjust the Envelope attack and decay time. Now press another key, and adjust some parameters for that sample, and so on ...

Memory Drum 2 is designed to be programmed with MIDI controllers. To assign a MIDI controller to a parameter, use the MIDI learn function on the XY modules on the panel. You can also assign MIDI controllers to some parameters that use Multi Picture modules, such as the ‘Shape’ on/off button. To do this, right click on the module and select ‘show in structure’. Then assign a MIDI controller to the hidden button module beneath the Multi Picture module. (You can also assign controllers to the Bank and Sample controls in the Sample section in this way, allowing you to browse through the sample map using MIDI controllers.)

Be aware that whenever Memory Drum 2 receives an event from an assigned MIDI controller, it will write to the memory of whatever sample (or samples) are currently selected. This can be a problem with certain hosts, which send extra MIDI controller data to plug-ins on initialization or when stopping/staring playback. Or it can be a problem if you accidentally move a MIDI controller... The safest thing to do is take frequent snapshots of your drum kit configuration.

With the exception of the sampler map configuration, all parameters are saved in the host plug-in edit buffer. This means that if you do not change the sample map, you do not need to save a new copy of the ensemble. But if you make any changes to the sample map, you should use the Reaktor autosave feature. This will create a new copy of the Memory Drum 2 ensemble and save it with your song. When in doubt, use the autosave feature to avoid data loss.

**Global Parameters**

The master controls are located at the top of the instrument panel. They adjust the instrument’s global settings, which affect all loaded samples. Note that both pitchbend and velocity can be assigned to samples individually using the modulation matrix, in which case it is probably best to set the global knobs to zero.
**Power Switch**
Mutes the entire instrument. This does not switch the instrument off to save CPU power.

**Mono**
When activated, the instrument provides four independent mono channels as output ports; otherwise, it offers two stereo ports.

**Bank Number**
Sets the number of sample banks. See the [Edit and Sample] section for details.

**Shift**
Transposes MIDI note input up or down as required.

**Pitchbend**
Determines the amount that the pitchbend wheel affects the pitch of the entire kit.

**Velocity**
Specifies the extent to which note-on velocity affects amplitude for all samples.

**Tune**
Transposes the pitch of the entire drum kit in semitones.

**Gain**
Controls the overall output level in decibels.

**Clear**
Resets all parameters of the current note to their default values.

**Copy**
Copies all parameters of the current note to an internal buffer.

**Paste**
Copies all parameters of the internal buffer to the current note.

---

**Sample & Edit**

The [Edit] section displays the sample map: Each slot represents a MIDI note; if this MIDI note is received, the sample selected within the [Sample] section is triggered. As there is a maximum of 128 different MIDI notes, normally only 128 samples can be loaded into a Reaktor sample map.

However, the two selection controls above the waveform display of the [Sample] section – entitled [Bank Select] and [Sample Select] – override this limitation. The best way to explain this is by example. Imagine you have a total of 512 drum sounds on your hard disk (this is just a hypothetical example!), and you wanted to load them all into Memory Drum so that they are all readily available for selection. Start by setting the number of banks to four (using the bank knob in the global section). Next, load the first 128 samples (using the sample map editor), assigning them to MIDI notes 0 to 127, and from velocity 1 to velocity 31. Then load the next 128 samples to MIDI notes 0 to 127, from velocity 32 to 63. Repeat this process for the remaining two ‘banks’ of 128 samples. You can now select any sample in the map by using the [Bank Select] and [Sample Select] lists on the panel. Although initial map creation is time-consuming, it can be extremely useful once set-up. Imagine having 128 kick drums loaded into the first bank, 128 snares loaded into the second bank, 128 hi hats into the third bank and so on... This can enable quick and easy kit creation, and convenient auditioning of samples on the fly.
<table>
<thead>
<tr>
<th><strong>Edit</strong></th>
<th>Sample Map Display</th>
<th>Selects the current MIDI note slot for editing. You can select a range of notes to edit simultaneously by clicking the right mouse button and dragging the mouse. Double-clicking on this bar automatically selects all notes for simultaneous editing (double-click again to return to the previous selection).</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Zoom Bar</td>
<td>Scrolls the viewable area across the entire MIDI note range. Clicking the right mouse button on this scrollbar cycles between 3 different zoom states.</td>
</tr>
<tr>
<td></td>
<td>MIDI Note</td>
<td>Displays the current MIDI note selected for editing within the [Sample Map Display].</td>
</tr>
<tr>
<td></td>
<td>Track</td>
<td>If activated, the received MIDI notes not only trigger the playback of the sample, but also select them for editing.</td>
</tr>
<tr>
<td></td>
<td>Solo</td>
<td>If activated, only the currently selected sample can be triggered by MIDI notes.</td>
</tr>
<tr>
<td><strong>Sample</strong></td>
<td>Bank Select</td>
<td>Selects the bank from which the [Sample Select] controller picks a sample.</td>
</tr>
<tr>
<td></td>
<td>Sample Select</td>
<td>Selects the sample that is played when the currently active MIDI note is received (see also [MIDI Note] and [Sample Map Display]).</td>
</tr>
<tr>
<td></td>
<td>Sampler</td>
<td>Displays the wave file selected by [Sample Select]. You can also load new files into Reaktor’s internal sample map editor here.</td>
</tr>
<tr>
<td></td>
<td>Start Position</td>
<td>Adjusts the start position within the sample file.</td>
</tr>
<tr>
<td></td>
<td>Reverse</td>
<td>Switches between forward and reverse playback of the sample.</td>
</tr>
<tr>
<td></td>
<td>Pitch</td>
<td>Transposes the sample up or down in semitones.</td>
</tr>
</tbody>
</table>
Sample Parameters

In this section you can adjust the parameters and effect settings of the currently selected sample (see [Edit][Sample map Display]). There is an envelope controlling the sample’s amplitude, a lo-fi distortion effect, a compression / saturation unit labeled [Shape], a multi-mode filter and a final output part. [Lofi], [Shape] and [Filter] can be toggled on and off for the selected sample by clicking the respective section’s title.

**Envelope**

- **Sustain/Release Mode**
  - If activated, the envelope remains at full amplitude after the attack time until the MIDI gate signal is closed; then the decay time is interpreted as release time.
- **Attack**
  - Specifies the time taken to reach full amplitude.
- **Hold**
  - Specifies the time held at full amplitude.
- **Decay**
  - Specifies the time taken for amplitude to fall back to zero.

**Lofi**

- **Hertz**
  - Adjusts the re-sampling frequency in Hertz.
- **Bit**
  - Adjusts the bit depth of the re-sampling algorithm.
- **Mix**
  - Crossfades between the unprocessed, dry signal and the processed, wet sound.
- **Noise**
  - Crossfades between the re-sampled signal and a noise generator to be mixed with the unprocessed sound.

**Shape**

- **Pre**
  - Increases the compressor input level.
- **Shape**
  - Determines the compressor gain curve.
- **Smooth**
  - Reduces the amount of distortion by smoothing gain changes as it controls the attack and release of the compressor.
- **Drive**
  - Saturates the output signal.

**Filter**

- **Mode**
  - Selects the operation mode of the filter unit. You can choose between low-pass, band-pass and high-pass filter modes.
- **Cut-off**
  - Sets the center frequency of the filter.
- **Resonance**
  - Sets the resonance of the filter at the cut-off frequency.
Out Gain Adjusts the output level in decibels.
Aux Send Sets the level of the auxiliary output port of the instrument.
Pan Positions the sound within the stereo field.
Out Port Selects the output of the instrument to which the slice’s sound will be routed. Depending on the global [Mono] switch either two stereo port or four mono ports are available.
Voice Group By default, Reaktor rotates voices to minimize voice-stealing. However in the context of drums, voice-stealing can often be desirable. Consider the example of a pair of open and closed high-hat samples — you may want these two samples to share the same voice, so that the triggering the open-hat sample truncates the closed-hat sample and vice versa. Please note that voice groups will only work effectively if: (1) the highest group number in use does not exceed the number of voices in the instrument properties, (which is four by default); and (2) all samples are manually assigned to a voice group (rather than a mixture of auto and manual voice assignments).

Modulation
Memory Drum 2’s advanced modulation routing allows various parameters to be modulated by a variety of sources. Beside the MIDI sources – like the modulation wheel and the pitchbend control – there are two envelopes and an LFO. ([Envelope A] is hard-wired to the amplitude of the sample playback but can also be used as freely assignable modulation source.) Some modulation sources have a variation denoted by ‘(hold)’. This option samples the value of the source when triggered. Try assigning the Pitchbend wheel to Pan and repeatedly trigger the sample while moving the Pitchbend wheel. Change the source to ‘Pitchbend/H’ and listen to the difference.

Shape Morphs the envelope shape from concave (at low values) to linear (at center) to convex (at high values).
Attack Specifies the time taken to reach full amplitude.
Decay Specifies the time taken for amplitude to fall back to zero.
Waveform Selects the waveform of the low frequency oscillator.
Operation In [Hz] and [Sync] modes, the LFO phase is reset every time the note is triggered, the only difference being that the frequency is quantized to tempo in [Sync] mode. In [Lock] mode, the LFO frequency snaps to the MIDI tempo and the LFO phase gets locked to the MIDI song position.
Speed Sets the rate at which the LFO oscillates.
Phase Sets the phase to which the LFO is reset when triggered by note events.

In each modulation section (A, B and C) the top box displays the current
source, and the bottom box displays the current destination. Click and drag vertically to change the source or destination. 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 set the slider bar to the full right hand position.

**Sources**

<table>
<thead>
<tr>
<th>Source</th>
<th>Description</th>
<th>Modulation Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Velocity</td>
<td>MIDI note on velocity.</td>
<td>Unipolar</td>
</tr>
<tr>
<td>Pitchbend</td>
<td>MIDI pitchbend wheel.</td>
<td>Bipolar</td>
</tr>
<tr>
<td>Pitchbend (hold)</td>
<td>MIDI pitchbend wheel, sampled at note-on.</td>
<td>Bipolar</td>
</tr>
<tr>
<td>CC1</td>
<td>MIDI controller 1 (the modulation wheel).</td>
<td>Unipolar</td>
</tr>
<tr>
<td>CC1 (hold)</td>
<td>MIDI controller 1, sampled at note-on.</td>
<td>Unipolar</td>
</tr>
<tr>
<td>CC7</td>
<td>MIDI controller 7 (the volume slider).</td>
<td>Unipolar</td>
</tr>
<tr>
<td>CC7 (hold)</td>
<td>MIDI controller 7, sampled at note-on.</td>
<td>Unipolar</td>
</tr>
<tr>
<td>Env A</td>
<td>Envelope generator A.</td>
<td>Unipolar</td>
</tr>
<tr>
<td>Env B</td>
<td>Envelope generator B.</td>
<td>Unipolar</td>
</tr>
<tr>
<td>LFO</td>
<td>The LFO.</td>
<td>Bipolar</td>
</tr>
<tr>
<td>LFO (hold)</td>
<td>The LFO, sampled at note-on.</td>
<td>Bipolar</td>
</tr>
<tr>
<td>Rnd</td>
<td>Random value generator.</td>
<td>Bipolar</td>
</tr>
</tbody>
</table>

**Destinations**

<table>
<thead>
<tr>
<th>Destination</th>
<th>Description</th>
<th>Modulation Range</th>
</tr>
</thead>
<tbody>
<tr>
<td>Amp</td>
<td>Slice amplitude</td>
<td>(-100% to +100%)</td>
</tr>
<tr>
<td>Pan</td>
<td>Stereo pan</td>
<td>(-100% to +100%)</td>
</tr>
<tr>
<td>Pitch</td>
<td>Slice pitch</td>
<td>(-12 to +12 semitones)</td>
</tr>
<tr>
<td>Start</td>
<td>Sample start position</td>
<td>(-1 to +1)</td>
</tr>
<tr>
<td>Env A Attack</td>
<td>Envelope A attack time</td>
<td>(approximately -10 to +10 seconds)</td>
</tr>
<tr>
<td>Env A Decay</td>
<td>Envelope A decay and hold time</td>
<td>(-100% to +100%)</td>
</tr>
<tr>
<td>Hz</td>
<td>Lofi resampling frequency</td>
<td>(-100 to +100 semi- tones)</td>
</tr>
<tr>
<td>Drive</td>
<td>Overdrive amount</td>
<td>(-60 to +60 decibels)</td>
</tr>
<tr>
<td>Cut</td>
<td>Filter cutoff</td>
<td>(-120 to +120 semitones)</td>
</tr>
<tr>
<td>Aux</td>
<td>Aux send level</td>
<td>(-100% to 100%)</td>
</tr>
<tr>
<td>Env B Amp</td>
<td>Envelope B amplitude</td>
<td>(-100% to 100%)</td>
</tr>
<tr>
<td>LFO Amp</td>
<td>LFO amplitude</td>
<td>(-100% to 100%)</td>
</tr>
</tbody>
</table>
L3 is a sequenced drum loop recycler: Load a loop, click some stuff, mangle the loop.

The instrument’s panel can be broken down into three sections. The top section contains a pattern sequencer and global controls for pattern length and
tempo swing. The middle section has the main step sequencer where patterns can be edited (up to eight patterns can be programmed and arranged per snapshot). The lower section contains the sample playback engine controls (including the sampler window where loops are loaded).

L3 is driven by the MIDI clock. This means that when used in the stand-alone version of Reaktor, the play button on the Reaktor toolbar must be pressed. When used as a plug-in, L3 will only run when the host sequencer song is playing.

**Pattern sequencer**

Each L3 snapshot consists of up to eight individual patterns labeled A through H. As you will see on the panel, there are 16 [Pattern Selector Boxes] along the top each displaying a letter (A to H). The pattern playback order depends on the arrangement of the letters displayed in these 16 boxes. The leftmost box selects the pattern to be played for the first bar, the second box selects the pattern for the second bar, and so on.

- **Pattern Selector Boxes**: Controls the sequence of the patterns to be played one after another. Click and drag vertically on any of the 16 boxes to select a pattern. Use the [Loop Area Bar] to determine the length and position of the loop.

- **Loop Area Bar**: Defines the area of the [Pattern Selector Boxes] played in a loop. If only one box is selected, only this pattern will be played; this can be useful to edit and audition a pattern.

- **Bars per Pattern**: Adjusts the number of bars in each pattern.

- **Beats per Bar**: Controls the number of beats in each bar. A beat is interpreted as a quarter note; each step represents a sixteenth note.

- **Swing**: Sets the amount of shuffle, i.e. the amount of slight delay on off-beats.

**Step sequencer**

Things get more interesting here. L3 features eight parameters which can be sequenced by programming their value at each 16th. This is what turns old loops into new loops...

The most obvious of these eight parameters is slice order. This is displayed in the lower of the two large windows (the [Slice Position Sequencer]). By clicking the mouse here you can rearrange slices of the original loop. The window is 16 steps high which means you can select the first 16 slices of your loop. If the selected loop has more than 16 steps, use the [Scroll Bar] at the left to see more slices. The right-mouse button has a function here too: it restores any step to its default value (i.e. to the original slice order).
The upper window (the [Parameter Sequencer]) is for editing the remaining seven parameters: gain, pan, pitch, reverse, roll, attack, and decay. Clicking the right-mouse button resets steps to their default value.

**Parameter Sequencer**
Controls the values of the various parameters for each sequencer step. Use the right mouse button to reset a parameter to its default value.

- **Gain**
  Adjusts the gain of each slice.

- **Pan**
  Adjusts the position of each slice within the stereo field.

- **Pitch**
  Adjusts the pitch of each slice, i.e. its transposition in respect to the original pitch of the sample file.

- **Reverse**
  Determines the playback direction. At minimum (the default setting), slices will play forward as normal. At any other value they will play in reverse. At lower values, playback will start from near the end of the slice, whereas with higher values, playback will start from nearer the beginning of the slice.

- **Roll**
  Causes the slice to retrigger repeatedly within each step. With higher values the slice will retrigger more quickly.

- **Attack**
  Causes the loop volume to suddenly cut out and then fade back in. At maximum, the fade in time is exactly 1 beat (i.e. four steps).

- **Decay**
  Modulates the envelope decay time. At center (default) decay time is unaffected. With higher values the decay time is extended, and with lower values the decay time is reduced. (Therefore the effect depends on the envelope decay time setting controlled by the [Decay] control of the [Sampler] section.)

**Slice Position Sequencer**
Controls the order of the slices. Low values represent slices at the beginning of the sample file, high values slices at its end. Thus, a line from the bottom-left to the top-right results in normal playback order as defined by the sample file without any re-arrangement.

**Scroll Bar**
Scrolls the [Slice Position Sequencer] vertically. This can be useful if a long loop with many slices is loaded: As the [Slice Position Sequencer] can only display sixteen vertical values, slices after the sixteenth cannot be controlled. Use this bar to scroll to those higher values.

**Edit Range Bar**
Controls the area of steps within the [Slice Position Sequencer] onto which the edit functions are applied. The edit functions are:

- **Reset Slices**
  Sets each step within the edit range to its default position.

- **Shift Up / Down**
  Shifts each step within the edit range up or down by one position.

- **Shift Left / Right**
  Rotates each step within the edit range to the left or right by one position.

- **Clear**
  Resets all steps within the edit range in both the [Slice Position Sequencer] and the [Parameter Sequencer].
Copy   Copies all steps within the edit range of both the [Slice Position Sequencer] and the [Parameter Sequencer] into an internal buffer.

Paste  Copies all steps from the internal buffer into the edit range of both the [Slice Position Sequencer] and the [Parameter Sequencer].

Sampler

The main window is for loading loops and displays the currently selected waveform. After loading a loop, make sure it is selected using the [Sample Select] knob, and then check that the detected tempo is correct (it’s displayed in the box to the left of the sample window). If incorrect, the tempo can be adjusted using the slider bar beneath. If the correct tempo cannot be selected, then the loop is not an integer number of bars in length, in which case you cannot use it.

All of the sampler controls in this section are stored per-pattern. Clicking a knob with the left mouse-button writes to the current pattern only, whereas clicking with the right-button writes to all eight patterns simultaneously (A to H). Also, double-clicking on a knob resets it to its default position.

Sample Display  Displays the sample currently selected by [Sample Select]. Double-click to open Reaktor’s Sample Map Editor and to load a sample file.

Tempo Control  Displays the automatically extracted tempo of the sample loop in beats per minute. Use the slider to select a different value.

Sample Select  Selects a sample from the map within Reaktor’s Sample Map Editor of the [Sample Display].

Pitch  Transposes the overall pitch of the loop in semitones.

Stretch  Calculates the pitch at which one bar of the audio file will be the same length as one bar of the actual current song tempo, and then transposes the loop accordingly. In other words when ‘stretched’, there will be no gaps between slices (caused by the original loop tempo being slower than the current tempo), nor will slices be prematurely truncated (caused by the original loop tempo being faster than the current tempo). It is still possible to transpose the loop when the stretch button is active, but obviously the loop will no longer be perfectly stretched to tempo. In other words, to be correctly stretched the pitch knob must be set to zero.

Shape  Determines the compressor gain curve (see also [Smooth] and [Damp].)

Smooth  Reduces the amount of distortion by smoothing gain changes; it controls the attack and release of the compressor. (See also [Shape] and [Damp].)
<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Damp</td>
<td>Attenuates high frequencies, reducing ‘grainy’ sounding compression artifacts. (See also [Shape] and [Damp].)</td>
</tr>
<tr>
<td>Length</td>
<td>Sets the hold period (i.e. the length) of the envelope shaping each slice’s amplitude.</td>
</tr>
<tr>
<td>Decay</td>
<td>Sets the decay time of the envelope shaping each slice’s amplitude. This is the master control that can be varied for each step independently.</td>
</tr>
<tr>
<td>Gain</td>
<td>Sets the output level for the current pattern.</td>
</tr>
</tbody>
</table>
Random Step Shifter

Randomstepshifter uses intelligent pseudo-random principles to cut-up and rearrange sample loops, on-beat, in real-time. There’s an intuitive three-part sequencer that triggers sample playback. It also modulates sample selection, positional offset, and playback pitch. In addition, these modulations can be mangled by various pseudo-random sequences. This instrument will create new sample loops for you very easily! You can load in any loop, just keep in mind that you need to cut the loops accurately so that they play correctly when they are looped over their entire length.

SQ2

The Randomstepshifter contains a simple step-sequencer. It consists of three parts, the [Select], [Offset], and [Pitch] sequencers. Each of the parts has two tracks – the trigger track at the bottom of the sequencer, and the modulation track above. The trigger tracks can be used independently from the modulation tracks but you can’t modulate without a trigger. In other words, you can trigger the envelope without sending any modulations, but not vice-versa. In the [Envelope] section you can choose which of the three trigger tracks is used for starting the envelope. The trigger events of the [Offset] track can also be used to reset sample offset. The modulation tracks can be used to modulate
the sample player’s main parameters. These are the [Select] parameter for sample selection, the [Offset] parameter for controlling the start position in the currently selected sample, and the [Pitch] parameter controlling pitch of sample playback.

Loop bar
The bar above the sequencer grid represents the sequencer’s loop region. Right-click (ctrl-click for Mac users) to set length. Left-click and drag to move.

Modulation tracks
Click on the grid to create modulation events. Drag the mouse up or down to set the level. These events are wired to the [Sample select], [Sample offset], and [Pitch] modules, respectively. They can be used to modulate the parameters of the respective modules in a controlled or randomly varied way. These parameters are sample selection, sample offset, and pitch of playback. Right-click (ctrl-click if you’re on a Mac) to delete the event, along with its trigger event. You will automatically create [Trigger] steps associated with the modulation events. See [Trigger tracks] for more information. Drag the modulation bar completely down to have a trigger event without modulation output. The three tracks can be selected with three buttons below the sequencer ([Select], [Offset], and [Pitch]).

Trigger track
Click to create events that trigger the [Envelope]. Drag down the corresponding modulation event to zero if you want a sole trigger without modulation. All three trigger tracks can be used to start the envelope. Use the respective buttons in the [Envelope] section to choose which track will do it. Furthermore, the [Offset] trigger track can reset the sample offset if the [Seq] button in the [Sample offset] section is active.

Select / Offset / Pitch
These buttons switch the view to show the three tracks of the sequencer. The modulation part of the [Select] track is wired to the [Sample Select] module, the modulation part of the [Offset] track is wired to the [Sample Offset] module and the modulation parts of the [Pitch] track is wired to the [Pitch] module.

Copy
Copies the current loop region into the clipboard.

Paste
Pastes the pattern clipboard into the current pattern.

Rand
Randomizes the current loop region.

Clear
Clears the current loop region.

Zoom Level
(16 st, 32 st, 64 st)
Click and drag mouse up or down to zoom in and out of the currently displayed pattern.

Clock divider
(1/6, 1/8, 1/12, 1/16, 1/24, 1/32)
Choose between different clock divisions. This speeds up or slows down the pattern but retains a metric relationship to the original speed. You get original speed with the 1/16 setting.

Run
Starts and stops the sequencer.
Sampler

At the heart of the Randomstepshifter lies the sample player. Just load your pre-cut loops into the sample map and the patch will rearrange them. You are free to use the modulation events from the three [Sequencer] tracks to control the parameters for [Sample select], [Sample offset] and [Pitch], or to let these three parameters be randomly varied. Activate random mode (the [Rnd] buttons) and adjust the three [RAND] knobs to get different pseudo-random results. Try moving the [QNTZ] knob in the [Sample offset] module while the sequencer runs for interesting dynamic sample cut-up in real-time. Please make sure that all samples in the sample map have a transposition value of 0.

Sample select
- **Rand**: This knob selects one of the pseudo-random sequences. Each incoming value from the modulation track is transformed in a pseudo-random way.
- **Seq / Rnd / Off**: These three buttons switch the modulation modes for the [Select] parameter. Choose between direct modulation by the modulation track, random variation based on the modulation track, and no modulation.
- **Select**: Sets the base for sample selection via the modulation track. This is the sample that gets played when [MOD] is inactive.
- **FIRST**: Defines the start point of the range for sample selection in the sample map.
- **LAST**: Sets the end point of the range for sample selection in the sample map.

Sample offset
- **RAND**: This knob selects one of the pseudo-random sequences. Each incoming value from the modulation track is transformed in pseudo-random way.
- **Seq / Rnd / Off**: These three buttons switch the modulation modes for the [Offset] parameter. Choose between direct modulation by the modulation track, random variation based on the modulation track, and no modulation.
- **MOD**: Switches the modulation input from the sequencer on for the [Sample offset] module.
- **Offset**: Sets the base offset in the sample. This is the offset that is applied when [MOD] is inactive.
QNTZ: Controls sample offset quantization. 1 = 1/16th, 2 = 1/8, 4 = 1/4, etc.

Smth: Controls for re-synthesis smoothness of sample playback. This alters the sound when introducing extreme pitch settings.

Pitch:
- RAND: This knob selects one of the pseudo-random sequences. Each incoming value from the modulation track is transformed in a pseudo-random way.
- Seq / Rnd / Off: These three buttons switch the modulation modes for the [Pitch] parameter. Choose between direct modulation by the modulation track, random variation based on the modulation track, and no modulation.

MOD: Switches on the sequencer’s modulation for the [Pitch] module.
- Pitch: Sets the transposition of the sample. This is the transposition applied when [MOD] is inactive. This interacts with the [RANGE] control and is independent from tempo as long as [Fit] is inactive.
- RANGE: Sets the range of bipolar sample transposition in semitones. A value of 12 gives you a transposition range from -12 to +12 semitones.
- Fit: When active, the pitch follows the tempo changes of sample playback, like in a conventional sample player.

Env:
- Attack: Sets the attack time of an ADSR envelope triggered by the sequencer events.
- Decay: Sets the decay time of an ADSR envelope triggered by the sequencer events.
- Sustain: Sets the maximum level the envelope will reach.
- Release: Sets the time that passes until the envelope is completely faded out after it has reached the sustain level.
- Sel / Offs / P on: Selects the trigger input for the envelope. It can be the trigger track from the select-, offset-, or pitch track, respectively.
- Mute: Switches the envelope on or off.
- Gain: Mutes sound output from the sample player.

Output: Controls the main volume of the sample player.
The Splitter is a small but sonically flexible sequenced sample-player. Geared towards granular beat production, it can also be used for melodies or pads. The main idea behind this sequencer/sample-player combo are the 16 sample slots. You can assign different fragments of the selected sample with individual settings for all parameters to the different slots above the waveform display. You can also assign individual MIDI notes.

**Sequencer**

The sequencer delivers classic step-sequencing in a very useable package. It offers 16 notes tracks with velocity control plus an additional modulation track, a song mode, and the ability to record incoming MIDI notes. The 16 sample slots (see description of the Splitter below) are represented by the sequencer’s 16 notes tracks. The leftmost sample slot corresponds to the bottom track, the rightmost sample slot corresponds to the top track.
Mode

Song Seq  Toggles song mode on and off. If on, the pattern sequence defined under [Song Sequence] is played. If off, the currently selected pattern is played and looped.

Zoom Level  Chooses whether 16, 32, or 64 steps are displayed. This has no influence on the notes played.

Notes  Displays the sequencer’s notes track. Click into the notes grid to create notes, right-click to delete notes (ctrl-click for Mac users). Note length depends on the quantization settings on the top right side of the sequencer.

Velocity  Displays the sequencer’s velocity track. Every note in the notes grid has a velocity bar. Drag with the mouse to change the levels.

Modulation  Displays the sequencer’s modulation track. Enter the desired modulation steps by dragging with the mouse. Quantized in 16ths.

Pattern

A/B/C/D  When not in song mode (see [Song Seq]), the selected pattern is played and looped.

Song Edit  The edit button enables you to assign patterns to the [Pattern Slots].

Pattern Slots  When [Song Edit] is active, click on a pattern slot and drag the mouse up / down to select the desired pattern.

Global controls

Loop Bar  The brown bar above the sequencer grid represents the loop region of the sequencer. Right click to set length (ctrl-click for Mac users), left click and drag to move.

Run  Switches sequence playback on or off.

Q’96 / Q’32 / Q’16  Quantization setting for the note-length resolution. Q’96 means 96th resolution, Q’32 is 32th resolution, and Q’16 is 16th resolution.

Copy  Copies the currently selected notes or modulation events, respectively, into the clipboard.

Paste  Pastes the pattern in the clipboard onto the current pattern.

Select  Toggles select mode on or off. When on, you can select multiple notes in the notes track via click or by dragging a square around them. You can also select a range in the modulation track.

Rec !  Switches on note recording via MIDI in.

Init !  Deletes all notes of the pattern and resets the modulation track events to zero. (You need to double click it!)
### Splitter

The granular sample-player enables you to load samples and trigger defined parts of them with individually stored settings for envelope, pitch, speed and grain length via the built-in sequencer or MIDI input. You also have a tempo-syncable LFO and some control over routing modulations and quantizing sample playback parameters.

<table>
<thead>
<tr>
<th>Slots</th>
<th>Edit</th>
<th>If active, you can change the parameters of the selected slot (see [LFO], [Modulation], [Shape], [Envelope] and [Output]). Also, an incoming MIDI note value is assigned to the selected slot. You can also assign notes with the mouse (see [Slots]).</th>
</tr>
</thead>
<tbody>
<tr>
<td>Slots</td>
<td>If [Edit] is active, you can change the slot’s parameters and change the assigned MIDI note by clicking into the slot and dragging up or down. You can also assign MIDI notes via MIDI in (see [Edit])</td>
<td></td>
</tr>
<tr>
<td>Copy</td>
<td></td>
<td>Copies all parameters of the selected slot.</td>
</tr>
<tr>
<td>Paste</td>
<td></td>
<td>Pastes the slot clipboard into the selected slot, overwriting all the slot’s parameters.</td>
</tr>
<tr>
<td>Waveform</td>
<td>Display</td>
<td>Right-click on the title bar of the waveform display to access the sample map menu (ctrl-click for Mac users). When in Edit mode, you can set the red sample-start line by clicking on the waveform.</td>
</tr>
<tr>
<td>Control</td>
<td>Speed Mode</td>
<td>These settings affect the [Speed] parameter of all sample slots. In free mode the speed can be dialed in freely; in grid mode it is quantized to 16th note values.</td>
</tr>
<tr>
<td>Grain Mode</td>
<td>These settings affect the [Grain] parameter of all sample slots. In free mode the grain length can be dialed in freely; in grid mode it is quantized to 16th note values; in note mode the grain length is quantized to steps corresponding to the 127 MIDI notes.</td>
<td></td>
</tr>
<tr>
<td>Length</td>
<td>Shows the length (in 16ths) of the sample currently selected from the sample map.</td>
<td></td>
</tr>
<tr>
<td>Pitch</td>
<td>Shows the sample's pitch deviation from original pitch when played at the current tempo.</td>
<td></td>
</tr>
<tr>
<td>Speed</td>
<td>Shows the sample’s pitch deviation when played at the current tempo. 1 represents original speed.</td>
<td></td>
</tr>
<tr>
<td>LFO</td>
<td>Speed</td>
<td>Adjusts the frequency of the Low Frequency Oscillator.</td>
</tr>
<tr>
<td>snc</td>
<td>Activates synchronization of the LFO to the song tempo.</td>
<td></td>
</tr>
<tr>
<td>Waveform</td>
<td>With this menu you can choose between six different LFO waveforms (sine, sawtooth, reverse sawtooth, pulse, and two random modes).</td>
<td></td>
</tr>
</tbody>
</table>
**LFO**
Routes the LFO to different parameters. Choose which parameter of the sample-player gets modulated by the LFO’s oscillations. Target parameters are [Offset], [Pitch], [Speed], and [Grain].

**LFO Dpth**
This control determines how much the LFO affects the chosen parameter.

**Seq**
Routes the modulation track of the sequencer to different parameters. Choose which parameter of the sample-player is modulated by the Modulation track. Target parameters are [Offset], [Pitch], [Speed], and [Grain].

**Seq Dpth**
This parameter determines how much the modulation track of the sequencer affects the target parameter.
Vectory

Vectory is an aggressive sample destruction unit. It consists of a sampler (on the left side) with vast re-arranging capacities whose signal is fed into a grain multi-effect (on the right side) that re-synthesizes the sound.

This structure is optimized for live use, with low-level Reaktor Core DSP. Complete settings for sample loops, re-arrangements, and grain effects can be recalled by moving the markers within the large square selection displays - changes occur instantly and with no audio drop-out. The effect unit even offers morphing between two settings.

The sample is loaded in a sub-instrument of Vectory called **Sample Loader**. Press Ctrl+2 to open its panel; to return to the main Vectory display press Ctrl+1. Only one sample can be loaded at a time. However, this sample can be quite long and contain several discrete loops.
The second panel set also contains a further sub-instrument labeled **Controllers**. This is designed to automate Vectory’s parameters via MIDI / VST.

**Sample**

This section at the top-right of the panel selects the sample material from the Sample Loader. By using the large square markers one of sixteen slots can be selected; each slot contains the data of the sample loop’s beginning, measured from the start of the sample file in bars and sixteenth.

- **Sample Selection Display**
  - Selects the active sample loop slot; each slot stores independent values for [Bar] and [Offset]. Those two parameters control the starting point within the loaded sample; thus, they define the sample material being played, which is then subject to re-arrangement by the [Sequencer] section. The length of the loop within the sample is controlled by [Sequencer][Meter] and [Sequencer][Unit].
  - **Write**
    - Stores the current values of [Bar] and [Offset] into the current slot.
  - **Slot**
    - Displays the number of the active slot in the [Selection Display].
  - **Bar**
    - Sets the starting point of the sample readout. This control adjusts the number of bars to be skipped in the sample file. (See also [Sample Loader][Bar] and [Sequencer][Position].)
  - **Offset**
    - Sets the starting point of the sample readout. This control adjusts the number of sixteenth steps that are added to the number of bars set by the [Bar] control. (See also [Sample Loader][Tempo] and [Sequencer][Position].)
Sequencer

The sequencer contains two sections: The [Seq Select] part chooses one of several sequencer settings; each setting is defined within the main [Sequencer] part that fills the left side of the instrument’s panel. The sequencer pattern is mapped onto the sample material selected in the [Sample] section.

Sequence Selection

Selects the active sequencer pattern. There are sixteen slots in each bank. (See also [Bank].)

Display

Selects the bank from which the [Selection Display] loads the sequencer pattern. There are eight banks available.

Bank

Defines the rearrangement pattern. The sequence is read in sixteenth steps from left to right; the vertical axis sets the offset from the sample readout starting point for each step in sixteenths; e.g. a scale from the bottom-left to the top-right represents the normal sample readout while a scale from the bottom-right to the top-left results in inverse readout: first the last sixteenth of the sample, then the one before the last etc. The starting point of the sample readout is controlled within the [Sample] section.

Position

Defines whether the sixteenth note selected by the [Position] pattern is played from the end to its beginning or in its normal playback direction.

Reverse

Adjusts the amplitude for each sequencer step.

Amplitude

Adjusts the hold time for each sequencer step. (See also [Release].)

Hold

Lengthens the sample at this sequencer step. The higher the value, the more it is stretched – the first square represents a ratio of 2:1, the next one is 3:1, etc. The sample is stretched by a grain re-synthesis algorithm; therefore, the grain pitch and grain frequency parameters of the [Grain Effect][Parameter Display] will greatly affect the sound of stretched steps. As the sequence moves on without being influenced by the stretch, parts of the stretched sample that do not fit in the step are cut (at a ratio of 2:1 the second half will be cut, etc.) See also [Slur].

Stretch

Ties stretches over consecutive sequencer steps. If slur is off, the stretch will be re-triggered at each sequencer step; if it is switched on, the stretch will be continued. This also affects the [Reverse] function.

Slur

Adjusts the pitch shift for each sequencer step. The values set here are relative ones; the absolute range of shifting is controlled by [Range].

Pitch

Controls the loop length in steps; the step length is adjusted by [Unit].

Meter

Sets the rhythmic unit (fourth, eighth or sixteenth, according to the current MIDI tempo) used as step for the [Meter] control.

Unit
Release  Adjusts the release time after each sequencer step's hold period. (See also [Hold].)

Range  Sets the absolute pitch range available for the sequencer steps. To actually adjust the pitch in each step, use the [Pitch] pattern, which exerts relative control over pitch. With higher [Range] values, identical [Pitch] patterns produce more drastic pitch shifts.

Rotate  Sets an offset to the sequencer readout.

Copy  Copies the current sequencer pattern into a buffer that can be read out by pushing the [Paste] button. A complete pattern can easily be duplicated or moved by selecting another slot with [Sequencer Select][Selection Display] and [Sequencer Select][Bank] before pasting the buffer.

Paste  Pastes the buffer's data into the current sequencer pattern, overwriting the old values. (See also [Copy].)

---

**Grain Effect**

This section controls the multi-effect placed after the sampler and the re-arrangement sequencer. The separation between sound generator and effect unit is only true for the panel; internally, those sections are closely interrelated. For example, both the frequency modulation and the grain resynthesis parameters show no results within the effect unit, but inside the sampler itself; they are placed here because they impact the instrument's sound as much as the other effect parameters.

There are two slots named A and B containing two different sets of effect unit settings. The [Morph] control interpolates between both settings for smooth transitions in live use.

Parameter Display  Shows the currently active effect parameters. If [Morph] is set completely to the left or right – thus selecting either the A settings or the B settings and not an interpolation – the parameters can also be edited. There are eleven parameters: FM Pitch, FM Amount, Bias, Pre-Quantize EQ Frequency, Pre-Quantize EQ Amount, Distortion (overdrive saturation), Sample Rate Reduction (frequency quantization), Post-Quantize EQ Frequency, Post-Quantize EQ Amount, Grain Pitch, Grain Frequency. Their technical meaning cannot be explained here in detail; their influence on the sound, however, can easily be heard when changing the values.

Grain Random  Adjusts the amount of randomness applied to the grain synthesis. The lower the value, the more constant is the frequency at which new grains are generated.
Copy
Copies the current effect parameter setting into a buffer that can be read out with the [Paste] button. The data can easily be moved to another storage position by selecting another parameter slot with [Morph], [A/B Selection Display] and [A/B Bank] before pasting.

Paste
Copies the buffer’s data into the current parameter setting, overwriting old values. (See also [Copy].)

Morph
Interpolates between the parameter settings selected with [A Selection Display] and [B Selection Display]. Move the marker completely to the left to activate and edit preset A; move it completely to the right to activate and edit preset B. Moving the marker in the space between gradually morphs from one preset to the other; during morphing no editing of the preset settings is possible.

Preset A
Selects the slot whose parameter settings are active (and editable if [Morph] is moved completely to the left). Each bank has sixteen slots. (See [A Bank].)

Bank A
Selects the bank from which the [A Selection Display] loads its data. Eight banks are available.

Preset B
Selects the slot whose parameter settings are active (and editable if [Morph] is moved completely to the right). Each bank has sixteen slots. (See [B Bank].)

Bank B
Selects the bank from which the [B Selection Display] loads its data. Eight banks are available.

Drive
Adjusts the amount of compression applied to the final output signal. High values represent high compressor thresholds; all audio data below this threshold will be amplified.

Sample Loader
The [Sample Loader] imports audio material. Only one sample can be loaded, but you can assign playback for different loops and parts.

BPM
Sets the tempo of the loaded sample in beats per minute. This should be done accurately – three small boxes to the right of the main BPM box allow you to set the tempo with three decimals – as this value is used to calculate the positions within the sample file (see [Sample][Bar] and [Sample][Offset]).

Start
Adjusts an offset in milliseconds at the beginning of the sample file that is skipped by all calculations concerning positions within the sample file.

Bar
Sets the number of sixteenth notes (according to the tempo adjusted in [BPM]) within one bar (see [Sample][Bar]).
### MIDI Controller

This sub-instrument of Vectory provides various automation possibilities to control the parameters via MIDI or VST. Five MIDI continuous controllers can be selected as modulation sources [Control A] to [Control E]. Additionally, two two-dimensional sources are available as [XY1] and [XY2]; they are controlled by two MIDI Ccs, one for the horizontal movements, the other one for vertical ones. [XY2] can also be controller via the MIDI pitch. Those modulation sources can be assigned to various parameters of Vectory within the [Assignment] section of the sub-instrument.

<table>
<thead>
<tr>
<th>Control A .. E</th>
<th>Selects the MIDI CC number that is referred to as [Control A] to [Control E] respectively.</th>
</tr>
</thead>
<tbody>
<tr>
<td>XY1 X</td>
<td>Selects the MIDI CC number that controls the horizontal position of the marker.</td>
</tr>
<tr>
<td></td>
<td>Selects the MIDI CC number that controls the vertical position of the marker.</td>
</tr>
<tr>
<td>XY2 X</td>
<td>Selects the MIDI CC number that controls the horizontal position of the marker.</td>
</tr>
<tr>
<td></td>
<td>Selects the MIDI CC number that controls the vertical position of the marker.</td>
</tr>
<tr>
<td>Note</td>
<td>Switches between MIDI CC mode (off) and MIDI note mode (on). In MIDI note mode, the position of the marker is controlled by the pitch of incoming MIDI events. The pitch adjusted by [Origin] selects the first position, the next pitch selects the second position etc.</td>
</tr>
<tr>
<td>Origin</td>
<td>Sets the MIDI pitch that selects the first position of the marker if [Note] is on.</td>
</tr>
<tr>
<td>Assignments</td>
<td>The four square displays [Sample / Sequence / A / B Selection Display] of the main instrument can be controlled by all seven modulation sources; all other parameters only provide the five one-dimensional sources [Control A] to [Control B].</td>
</tr>
</tbody>
</table>
The great final-stage mastering tool FlatBlaster 2 has been rebuilt using the new Reaktor Core features. This patch combines four frequency-specific compressors with a full-spectrum peak-limiter to produce a high-end package for your multiband dynamics shaping needs. As it does not introduce any delay it is not limited to mastering use but can also be applied on a per channel basis. The controls might appear intimidating at first glance, but are actually straightforward when you examine the signal chain. The separately compressed bands get mixed together and then processed by a full-band peak limiter. Please note that the master bypass for the complete patch is situated to the left above the X Over section.

**Multi-band compressor**

After the input stage the signal gets split into four independent frequency bands as defined by the X Over section. Each frequency band gets processed by independent, identical compressors and can be independently muted, soloed and bypassed. Separate saturators for each band make it possible, for example, to add punch and bite to the mids without affecting clarity in the lower registers.
<table>
<thead>
<tr>
<th>Input</th>
<th>Bypass</th>
<th>X Over</th>
<th>High</th>
<th>Mid</th>
<th>Low</th>
<th>Stereo</th>
<th>Tresh</th>
<th>Ratio</th>
<th>Knee</th>
<th>Sat</th>
<th>Link</th>
<th>Att</th>
<th>Rel</th>
<th>Out Gain</th>
<th>Bypass</th>
<th>Mute</th>
<th>Solo</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Trims the input gain to prevent overload.</td>
<td>Bypasses the complete effect. This is the master bypass switching off all compressors and the Limiter.</td>
<td>Sets the crossover frequency between the High and Mid High compressor bands.</td>
<td>Sets the crossover frequency between the Mid High and Mid Low compressor bands.</td>
<td>Sets the crossover frequency between the Mid Low and Low compressor bands.</td>
<td>Sets stereo width of the frequency band. 0 is mono, 1 is original stereo, 2 is extra stereo.</td>
<td>Sets the point at which the compressor will begin to work (in db). Levels below this threshold remain unprocessed.</td>
<td>Adjusts the ratio of the input level to the output level after compression.</td>
<td>This parameter adjusts how gradually the full amount of compression is introduced. Think of it as a slope control for the attack time.</td>
<td>Drives the band into saturation.</td>
<td>Activates stereo linking of the two input channels. When active, the compressor takes the max of the left and right peak levels and uses it for both channels. This preserves a clean stereo image and is lighter on CPU cycles.</td>
<td>This dial adjusts the attack time. It is the time the compressor takes to react to an above-threshold signal.</td>
<td>With this control you set the release time. This is the time the compressor takes to return the signal to normal when it falls below the compression threshold.</td>
<td>Sets the amount of amplification applied to the compressed signal of the specific band before it gets mixed with the other bands.</td>
<td>Bypasses the compressor for the respective band.</td>
<td>Turns the sound of the respective band off.</td>
<td>Turns all other bands off, leaving only the signal of the soloed band. Use it to fine tune single compressor bands.</td>
</tr>
</tbody>
</table>
**Full-band peak limiter**

The peak limiter affects the full band signal. For clean mastering purposes we recommend a limiter threshold setting of about -3 to -4 db and a peak setting at 0db. Should pumping effects be desired, adjust the threshold to more extreme values.

- **Thr** Adjusts the threshold of the limiter. Levels above this value get processed.
- **Peak** Adjusts the hard limit of the signal. No signal will exceed this limit.
- **Rel** This adjusts the release time. It is the time the limiter takes to return the signal to normal when it falls below the limiting threshold.
- **Soft / Hard** Balances between soft saturation and hard clipping of the above peak signal.
- **Compare** Controls the amplification of the uncompressed signal if Bypass is active. If you want compression without amplification set it to 0 and make sure there is no change in level when toggling the Bypass button.
- **Link** Activates stereo linking of the two input channels.
- **Bypass** Bypasses the Full Band Peak Limiter only, leaving the 4 compressors active.
Lurker is a hybrid effect capable of classic phaser sounds, spring reverbs and feedback echoes – but most of all it transforms any incoming signal into stunning rhythmic sequences, mangling pitches and re-arranging the sound. This is technically possible because all those effects are based on a delay unit (and this instrument is an extremely versatile one).

Four internal sequencer tracks are the most prominent feature. They allow for fast, visual creation of musical patterns that you can use to modulate parameters such as delay times of the two independent delay units. Those times can be set in sixteenth note multiples (for tempo-based effects) or in milliseconds (for comb filter-like effects that map a new pitch onto the signal). A filter, a gating envelope generator, and a final delay further enrich the sound.
### Global

This top section of the instrument panel contains three parts: the input control (at the left), the snapshot management (in the middle), and the shuffle control (at the right).

The input control provides a simple sampler to load files and re-trigger their playback synchronized to the sequencers. The level of external signals can be controlled here. The snapshot handling and the shuffle system are identical to those of Massive; see that instrument's manual for further details.

<table>
<thead>
<tr>
<th>Input</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Loop Switch</td>
<td>Controls the events that re-trigger the sampler. If on, the sampler starts</td>
</tr>
<tr>
<td></td>
<td>playback at the file's beginning when the loop controlled by [Length Control]</td>
</tr>
<tr>
<td></td>
<td>and [Unit Select] is returning to its origin; if off, the sampler</td>
</tr>
<tr>
<td></td>
<td>is re-triggered only when the global MIDI clock starts playing.</td>
</tr>
<tr>
<td>Length Control</td>
<td>Sets the length of the loop that controls the sampler’s re-triggering if</td>
</tr>
<tr>
<td></td>
<td>the [Loop Switch] is on. (See also [Unit Select].)</td>
</tr>
<tr>
<td>Unit Select</td>
<td>Selects the rhythmical unit on which [Length Control] is based. This unit</td>
</tr>
<tr>
<td></td>
<td>refers to the global MIDI clock.</td>
</tr>
<tr>
<td>Sampler</td>
<td>Displays the currently active sample (see [Sample Select]). Double-click</td>
</tr>
<tr>
<td></td>
<td>to open the Sample Map Editor where sample files can be loaded and</td>
</tr>
<tr>
<td></td>
<td>organized.</td>
</tr>
<tr>
<td>Sample Select</td>
<td>Selects one of the samples loaded into the [Sampler].</td>
</tr>
<tr>
<td>Sample Pitch</td>
<td>Transposes the selected sample. This also affects the samples playback</td>
</tr>
<tr>
<td></td>
<td>speed. (An octave up or down sets the playback to double or half speed</td>
</tr>
<tr>
<td></td>
<td>respectively.)</td>
</tr>
<tr>
<td>Internal Level</td>
<td>Controls the amplitude of the sampler.</td>
</tr>
<tr>
<td>External Level</td>
<td>Controls the amplitude of the external signal.</td>
</tr>
<tr>
<td>External Mute</td>
<td>Disables the external input.</td>
</tr>
<tr>
<td>External Display</td>
<td>Shows the level of the external signal.</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Snapshot</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Snapshot Store</td>
<td>With the left mouse button, a snapshot slot number can be selected; by</td>
</tr>
<tr>
<td></td>
<td>pressing the right mouse button, the current instrument settings (including</td>
</tr>
<tr>
<td></td>
<td>all sequencer data) are stored into this snapshot slot.</td>
</tr>
<tr>
<td>Snapshot Recall</td>
<td>Displays a list of the available snapshots; selecting a snapshot with the</td>
</tr>
<tr>
<td></td>
<td>mouse recalls all its data, including sequences.</td>
</tr>
<tr>
<td>Snapshot Mode</td>
<td>Selects whether the snapshots are only recalled internally or if external</td>
</tr>
<tr>
<td></td>
<td>control signals received at the instrument’s [Snap] port are recognized,</td>
</tr>
<tr>
<td></td>
<td>too. This allows connection to a master song sequencer.</td>
</tr>
</tbody>
</table>
Shuffle Select

Selects one of twelve quantization presets. Each preset ranges over sixteen steps; the higher the value within the display, the more delay is applied to this step. The first preset, for example, alternates between low and high values, so every second step will be delayed, resulting in a standard off-beat shuffle. The presets only define relative times; the effective delay time at maximum values is set by the [Shuffle] control.

Shuffle

Scales the preset of the [Quantization Select] control. Turn to the left for no quantization — independent of the selected preset —, to the right for full delay times.

Sequencer

There are two step sequencers (tracks [A] and [B]) and two tracks that glide from step to step ([C] and [D]). Each sequencer provides individual control over length and speed.

Length Control

Sets the length of the loop that can be edited within the sequencer display in steps. (See also [Unit Select].)

Unit Select

Selects the rhythmic unit with which each step of the sequencer track is interpreted. This refers to the global MIDI clock.

Sequencer

Defines and displays the track’s rhythmic pattern.

Delay Units

Two identical delay units form Lurker’s core. They can be used either in parallel or serially. Each offers independent delay times for the left and the right audio channels, both defined as multiples of sixteenth notes or in milliseconds. At the left of the controls that adjust the delay times, their modulation can be controlled; even the depth of modulation is subject to modulation, resulting in complex interactions of several modulation patterns. The knobs at the delay time controls’ right define the channel swap, the amount of feedback and the feedback’s filtering.

Depth

Sets the amount of modulation applied to the delay time. This is independent of the static delay time and ranges from no modulation (at the left) to a modulation of about 260 milliseconds (at the right). The modulation signal is selected by the [Modulations Source] control below. (See also [Depth Modulation Amount].)

Modulation Source

Selects the sequencer track that modulates the delay time. The amount of modulation at maximum modulation signals is controlled by [Depth].
**Depth Modulation Amount**
Adjusts the amount of modulation of the [Depth] control. Turn to the left for inverted modulation (i.e. there is much modulation at low modulation signals and vice versa), to a mid position for no modulation and to the right for normal modulation. High values at the right result in very much modulation of the [Depth] control, increasing its maximal modulation amount to approx. 2400 milliseconds. The signal that actually modulates the [Depth] control is selected below.

**Depth Modulation Source**
Selects the sequencer track that modulates the modulation depth. The amount of modulation is controlled by [Depth Modulation Amount].

**Modulation Slur**
Sets the amount of interpolation applied to subsequent steps of the modulation track. Turn to the left for no interpolation and fast delay time changes, to the right for soft and slow ramps between different states.

**Modulation Invert**
Inverts the modulation signal i.e. if on, the modulation signal is not added to the static delay times controlled by [Quantized Delay Time Left / Right] and [Millisecond Delay Time Left / Right], but subtracted.

**Quantized Delay Time Left / Right**
Sets the static delay time of the left channel (upper control) and right channel (lower control) respectively in multiples of sixteenth notes of the global MIDI clock. The actual delay time is calculated from the sum of this value, the delay time adjusted by [Millisecond Delay Time Left / Right] and the modulation signal (see [Depth]),

**Millisecond Delay Time Left / Right**
Sets the static delay time of the left channel (upper control) and right channel (lower control) respectively in milliseconds. The actual delay time is calculated from the sum of this value, the delay time adjusted by [Quantized Delay Time Left / Right] and the modulation signal (see [Depth]),

**Channel Swap Amount**
Controls the modulation applied to the interaction of the left and right feedback signal. At low modulation signals the left channel’s feedback signal is routed again to the left channel; at mid modulation values both channels are mixed to a mono sound that is fed back into both channels identically; at high modulation signals the channels are swapped and the left channel’s signal is routed to the right channel (and vice versa). This control scales the modulation signal, i.e. at mid position high modulation signals are mapped to mid modulation signals; at the complete left there is no modulation and no channel swapping. The modulation signal is selected below.

**Channel Swap Modulation Source**
Selects the sequencer track that modulates the channel swap. The amount of modulation is controlled by [Channel Swap Amount].

**Cutoff**
Sets the cut-off frequency of the low-pass filter within the feedback loop.

**Reset**
Sets all controllers of the delay unit to their default values.
Feedback Amount  Controls the amount of feedback.
Bypass Switch  Toggles between the dry, unprocessed signal (when on) and the wet, delayed signal.
Mode Select  Switches between parallel and serial modes. In parallel mode, both delay units receive the same input signal and the [Crossfade] control can crossfade between their output signals. In serial mode, the signal enters the upper delay unit and is then routed to the lower unit.
Crossfade  Mixes between the sound of the upper and lower delay unit when [Mode Select] is set to parallel.

Filter

The filter is placed after the two delay units. The low-pass filter’s cut-off frequency and resonance can be edited (you can adjust left and right channels independently); the cut-off can also be modulated by one of the four modulation tracks.

Cutoff  Adjusts the cut-off frequency of the filter. The horizontal axis controls the left channel, the vertical axis the right one.
Cutoff Modulation Amount  Sets the amount and polarity of the modulation applied to the low-pass filter’s cut-off frequency.
Cutoff Modulation Source  Selects the modulation track that is used to modulate the filter’s cut-off frequency.
Resonance  Adjusts the resonance of the filter. The horizontal axis controls the left channel, the vertical axis the right one.
Reset  Sets all controllers of the filter to their default values.
## Master and Envelope

The master section simply controls the instrument’s output level before its signal passes to the additional delay. The [Env] control enables an envelope generator that is triggered by one of the two step sequencer tracks. This can be used to gate the instrument’s signal.

<table>
<thead>
<tr>
<th>Master</th>
<th>Envelope</th>
</tr>
</thead>
<tbody>
<tr>
<td>Output</td>
<td>Controls the instrument’s main output level.</td>
</tr>
<tr>
<td>Bypass</td>
<td>Mutes the effect and directly routes the input signal to the output.</td>
</tr>
<tr>
<td>Envelope Amount</td>
<td>Adjusts the influence of the envelope generator on the output amplitude. Turn to the left for complete independence. Turn to the right for full shaping of the amplitude by the envelope.</td>
</tr>
<tr>
<td>Source Select</td>
<td>Selects one of the two step sequencers to be used as trigger signal. (See also [Gate Threshold].)</td>
</tr>
<tr>
<td>Gate Threshold</td>
<td>Controls which steps of the selected modulation track are used as trigger signals. All steps with values below the one adjusted by this control are ignored.</td>
</tr>
<tr>
<td>Velocity Amount</td>
<td>Controls the influence of the trigger gate’s height (i.e. velocity) on the envelope’s amplitude. Turn to the left for full amplitude with every trigger signal, turn to the right to map the step’s value onto the envelope’s amplitude.</td>
</tr>
<tr>
<td>Velocity Attack</td>
<td>Controls the amount of modulation applied to the envelope’s attack time by the triggering step’s velocity. At low velocities (i.e. low step values) the attack time is increased if the knob is turned to the right. At left positions the velocity doesn’t affect the attack time. This is independent of the [Velocity] control.</td>
</tr>
<tr>
<td>Velocity Decay</td>
<td>Controls the amount of modulation applied to the envelope’s decay time by the triggering step’s velocity. At low velocities (i.e. low step values) the decay time is decreased if the knob is turned to the right. At left positions the velocity doesn’t affect the decay time. This is independent of the [Velocity] control. (See also [Decay].)</td>
</tr>
<tr>
<td>Decay</td>
<td>Sets the static decay time of the envelope that can be modulated by the triggering step’s velocity (see [Velocity Decay]).</td>
</tr>
</tbody>
</table>
**Additional Delay**

The delay unit after the output section allows for further manipulation of the signal. It is similar to the main delay units, but the delay times can't be modulated and the channels can't be swapped; instead, there is a high-pass filter within the feedback loop. As a special feature, the ratio between dry, unprocessed signal and wet, delayed sound can be modulated by one of the modulation tracks.

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Quantized Delay Time Left / Right</td>
<td>Sets the static delay time of the left channel (upper control) and right channel (lower control) respectively in sixteenth note multiples of the global MIDI clock. The actual delay time is calculated as the sum of this value and the delay time adjusted by [Millisecond Delay Time Left / Right].</td>
</tr>
<tr>
<td>Millisecond Delay Time Left / Right</td>
<td>Sets the static delay time of the left channel (upper control) and right channel (lower control) respectively in milliseconds. The actual delay time is calculated as the sum of this value and the delay time adjusted by [Quantized Delay Time Left / Right].</td>
</tr>
<tr>
<td>Feedback Amount</td>
<td>Controls the amount of feedback.</td>
</tr>
<tr>
<td>Highpass</td>
<td>Sets the cut-off frequency of the high-pass filter within the feedback loop.</td>
</tr>
<tr>
<td>Lowpass</td>
<td>Sets the cut-off frequency of the low-pass filter within the feedback loop.</td>
</tr>
<tr>
<td>Mix Modulation Source</td>
<td>Selects the sequencer track modulating the ratio between dry, unprocessed signal and wet, delayed sound (see [Mix]).</td>
</tr>
<tr>
<td>Mix</td>
<td>Controls whether the ratio between dry, unprocessed signal (heard at mid position) and wet, delayed sound (at the right). Turn to the left to use the sequencer track selected by [Mix Modulation Source] as control: At high modulation levels the wet sound is passed on; at low values, the dry signal.</td>
</tr>
</tbody>
</table>
The well-known Space Master series of reverb modellers has been updated for Reaktor 5. Based on several diffusion delays, Space Master 2 can produce a wide array of high-quality natural or experimental ambiences. The patch’s efficient set of reverb parameters include an early reflections section, a late reflections module and a post EQ. Dials for main reverb time, control of balance between the two reflection stages, and between dry and wet signal round off the controls.

**Input and output stage**

You can introduce an initial delay into the reverb signal with the predelay [Time] dial and control the predelay’s stereo position with the [Symmetry] knob. The [Early / Late Balance] slider can be used to move the source in space – more early reflections bring the signal to the front and more late reflections make it appear further back in space. At the end of the signal chain, the [Dry / Wet] slider crossfades between the dry original signal and the processed sound.

**Predelay**
- **Time**
  - Sets an initial delay for the wet signal.
- **Symmetry**
  - Introduces a difference into the delay times for the right and left predelay channels. Use this to shift the signal around in the stereo image.

**Mixing**
- **Early/Late Balance**
  - With this parameter you can set how much of the early and late reflections, respectively, can be heard in the output.
- **Dry / Wet**
  - This controls the balance between dry and wet signal.
Reflections

Use the two [Size] and [Diffusion] parameters to dial in the early and late stages of variable density diffused reflections. The early stage commonly represents the direct response of the virtual space, whereas the late reflections define the sound when the early reflections have died away.

For dynamic reverb effects you can use the Modulation section. It offers an LFO routed to the delay times with [Rate] and [Depth] control. The LFO can enhance your reverb signal by adding liveliness.

<table>
<thead>
<tr>
<th>Early / Late Reflections</th>
</tr>
</thead>
<tbody>
<tr>
<td>Size</td>
</tr>
<tr>
<td>Symmetry</td>
</tr>
<tr>
<td>Diffusion</td>
</tr>
<tr>
<td>Reverberation Time</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Modulation</th>
</tr>
</thead>
<tbody>
<tr>
<td>Rate</td>
</tr>
<tr>
<td>Depth</td>
</tr>
</tbody>
</table>

Frequency response

The two EQ sections serve slightly different needs. The Damping EQs are integrated into the reflection stages and influence their frequency responses. The Post EQ acts on the main output of the patch should be used to color the overall sound.

<table>
<thead>
<tr>
<th>Frequency Damping</th>
</tr>
</thead>
<tbody>
<tr>
<td>Low Frequency Damp</td>
</tr>
<tr>
<td>High Frequency Damp</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Post EQ</th>
</tr>
</thead>
<tbody>
<tr>
<td>Low Frequency Boost</td>
</tr>
<tr>
<td>High Frequency Boost</td>
</tr>
</tbody>
</table>
Sequencer

SQ16

Description

The SQ16 sequencer delivers classic step-sequencing in a very usable package. It features 16 notes tracks with velocity control plus additional modulation tracks, a song mode, and the ability to record incoming MIDI notes.

Details

Control

- **Song Seq**: Toggles song mode on and off. When on, the pattern defined under [Song Sequence] is played. When off, the currently selected pattern is played and looped.
- **Zoom Level**: Choose whether 16, 32, or 64 steps are displayed. This has no influence on the notes played.
- **Notes**: Displays the notes track of the sequencer. Click the notes grid to create notes, right-click to delete notes (ctrl-click for Mac users). Note length depends on the quantization settings on the top right side of the sequencer.
- **Velocity**: Displays the sequencer’s velocity track. Each note in the notes grid has a velocity bar in the velocity track. Drag with the mouse to change the levels.
- **Modulation**: Displays the sequencer’s modulation track. Enter the desired modulation steps by dragging with the mouse. Quantized in 16ths.

Pattern

- **A/B/C/D**: When not in song mode (see [Song Seq]), the selected pattern is played and looped.
### Song Edit
The edit button enables you to assign patterns to the [Pattern Slots].

### Pattern Slots
When [Song Edit] is active, click into a pattern slot and drag the mouse up / down to select the desired pattern.

### Global Controls

<table>
<thead>
<tr>
<th>Control</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Loop Bar</strong></td>
<td>The brown bar above the sequencer grid represents the loop region. Right-click (ctrl-click for Mac users) to set length and left-click and drag to move.</td>
</tr>
<tr>
<td><strong>Run</strong></td>
<td>Switches sequence playback on or off.</td>
</tr>
<tr>
<td><strong>Q'96 / Q'32 / Q'16</strong></td>
<td>Quantization setting for the note-length resolution. Q'96 means 96th resolution, Q'32 is 32th resolution, and Q'16 is 16th resolution.</td>
</tr>
<tr>
<td><strong>Copy</strong></td>
<td>Copies the currently selected notes or modulation events into the clipboard.</td>
</tr>
<tr>
<td><strong>Paste</strong></td>
<td>Pastes the pattern clipboard into the current pattern.</td>
</tr>
<tr>
<td><strong>Select</strong></td>
<td>Toggles select mode on or off. When on, you can select multiple notes in the notes track by clicking or by dragging a square around them. You can also select a range of the modulation track.</td>
</tr>
<tr>
<td><strong>Rec !</strong></td>
<td>Switches on note recording via MIDI in.</td>
</tr>
<tr>
<td><strong>Init !</strong></td>
<td>Deletes all pattern notes and resets the modulation track events to zero. (You need to double click on it!)</td>
</tr>
</tbody>
</table>
Description

The SQ8 is your standard building block for rhythmic step-sequencing. It sports a clean interface: 4 patterns with 8 tracks (consisting of 64 steps each). You also get variable looping, shuffle, reverse play, and multiple viewing options. On top of that, you can chain 16 patterns together into a song.

Details

<table>
<thead>
<tr>
<th>Mode</th>
<th>Pattern A/B/C/D</th>
<th>Song Sequence</th>
</tr>
</thead>
<tbody>
<tr>
<td>Song Seq</td>
<td>Toggles the song mode on and off. If on, the pattern sequence defined under [Song Sequence] is played. If off, the currently selected pattern is played and looped.</td>
<td></td>
</tr>
<tr>
<td>A/B/C/D</td>
<td>When not in song mode (see [Song Seq]), the selected pattern is played and looped.</td>
<td></td>
</tr>
<tr>
<td>Song Edit</td>
<td>The edit button enables you to assign patterns to the [Pattern Slots].</td>
<td></td>
</tr>
<tr>
<td>Pattern Slots</td>
<td>When [Song Edit] is active, click into a pattern slot and drag the mouse up / down to select the desired pattern.</td>
<td></td>
</tr>
</tbody>
</table>
Pattern view / Track view

Click on [All] to see the complete pattern with all tracks. Click on any button to the left of a track to view the track exclusively. In track view you can also change the velocity of the individual notes.

Loop Bar

The darkish green bar above the sequencer grid represents the sequencer’s loop region. Right-click (ctrl-click for Mac users) to set length and left-click and drag to move.

Notegrid

Click into the grid to add or delete note events.

Sel

Toggles note select mode on or off. When on, you can select an area of the note grid to be cleared, copied from, or pasted to. This works in all viewing modes.

Copy

Copies the content of the current pattern.

Paste

Pastes the pattern clipboard into the current pattern, overwriting all events.

Rec !

Switches on note recording via MIDI in.

Clr

Deletes all selected notes of the pattern.

Zoom Level

Click and drag mouse up or down to zoom in and out of the currently displayed pattern.

(16 st, 32 st, 64 st)

Clock divider

Chooses between different clock divisions. This speeds up or slows down the pattern but retains a metric relationship to the original speed. You get original speed with the 1/16 setting.

(1/6, 1/8, 1/12, 1/16, 1/24, 1/32)

Rev

Toggles reverse play on and off. The direction is reversed by pattern mirroring.

Stepshifter

This menu determines the playback mode. --- is normal, 1324 and 1432 let the steps swap their positions, <??> plays in random direction, <???> randomly jumps to the previous or next step, ???? jumps to a completely random step.

Shffl

Shuffle function. Click and drag mouse up or down to select the amount of shuffle.

Run

Starts and stops the sequencer.
Description

The SQ 8x8 is a small step-sequencer with a twist. You put events in a grid and drag a rectangle around a group of them by right-clicking and dragging (ctrl-click when you’re on a Mac). This rectangle defines the sequencer’s loop area, controlling what gets played line-wise. You can change this area in realtime. Think of it as a two-dimensional loop-bar. Some nice realtime step-shifting and shuffle features are also part of the package.
Details

Mute
Mutes the sequencer output.

Notegrid
Click the grid to add or delete note events. Click and drag the mouse up or down to change velocity. Right-click (ctrl-click for Mac users) and drag to define the loop-area.

Clock divider
Choose between different clock divisions. This speeds up or slows down the pattern but retains a metric relationship to the original speed. You get original speed with the 1/16 setting.

Shffl
Shuffle function. Click and drag mouse up or down to adjust the amount of shuffle.

X - playback modes
-- - walk normally
X - +/- random step in X direction
XX - +/- random step in whole X row

Y - playback modes
-- - walk normally
Y - +/- random step in Y direction
YY - random in whole column

R!
Randomizes the current loop area.

Clr
Clears the current loop area.

Copy
Copies the content of the current loop area.

Paste
Pastes the pattern clipboard into the current loop area, overwriting all events.
**Description**

The SQP is a piano roll-style sequencer covering a very wide midi note-range. You can enter notes via the mouse or record incoming MIDI notes. If you want to input longer events with the mouse, just click and drag the start or end of an existing note. Move events by clicking and dragging them around. When [Select] is on you can move selected events as a group.
**Details**

<table>
<thead>
<tr>
<th>Control</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Song Seq</td>
<td>Toggles song mode on and off. If on, the pattern sequence defined under [Song Sequence] is played. If off, the currently selected pattern is played and looped.</td>
</tr>
<tr>
<td>Quantization</td>
<td>Controls the quantization of note events. Choose between quantization in 16th, 24th and 32th. You can also switch off quantization.</td>
</tr>
<tr>
<td>Zoom Level</td>
<td>Choose whether 16, 32, or 64 steps are displayed. This has no influence on the notes played.</td>
</tr>
<tr>
<td>Velocity</td>
<td>Displays the sequencer’s velocity track. Each note in the notes grid has a velocity bar in the velocity track. Drag with the mouse to change the levels.</td>
</tr>
<tr>
<td>Select</td>
<td>Toggles select mode on and off. If on, you can select note events by clicking them or by dragging a square around them.</td>
</tr>
<tr>
<td>Copy</td>
<td>Copies the currently selected events.</td>
</tr>
<tr>
<td>Pste</td>
<td>Pastes the pattern clipboard into the current pattern.</td>
</tr>
<tr>
<td>Rec !</td>
<td>Switches on note recording via MIDI in.</td>
</tr>
<tr>
<td>Init !</td>
<td>Deletes all note events of the pattern. (Needs to be double clicked!)</td>
</tr>
<tr>
<td>A/B/C/D</td>
<td>When not in song mode (see [Song Seq]), the selected pattern is played and looped.</td>
</tr>
<tr>
<td>Song Edit</td>
<td>The edit button enables you to assign patterns to the [Pattern Slots].</td>
</tr>
<tr>
<td>Pattern Slots</td>
<td>When [Song Edit] is active, click into a pattern slot and drag the mouse up or down to select the desired pattern.</td>
</tr>
</tbody>
</table>

**Global controls**

<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Loop Bar</td>
<td>The blue bar above the sequencer grid represents the sequencer’s loop region. Right- click (ctrl-click for Mac users) to set length and left-click and drag to move.</td>
</tr>
<tr>
<td>Run</td>
<td>Starts and stops the sequencer.</td>
</tr>
<tr>
<td>Roll bar</td>
<td>To the right of the note grid you will find the roll bar that navigates the MIDI note range. Drag it up or down to see the higher or lower registers, respectively.</td>
</tr>
</tbody>
</table>