Algoritm
FM Synthesizer

Operation Manual

Reason Studios

reasonstudios.com

Algoritm
FM Synthesizer

Introduction

Algoritm FM Synthesizer is an extremely flexible polyphonic, semi-modular hybrid FM synthesizer. It features nine Operator Slots, where each slot can hold either an FM Operator, a wavetable Oscillator, a Shaper or a multi-mode Filter. You can freely interconnect the operators/modules for modulation and audio processing purposes; in effect creating your own custom algorithms!

Parameters can be modulated using a modulation matrix, with modulation sources including envelopes from the operators, three LFOs, two stepped curve generators, and much more.

A unison function multiplies the voices for stereo spread and detuning, and the sound can then be processed by a great-sounding seven-stage multi-effects unit.

In short, welcome to the mother of FM synths!

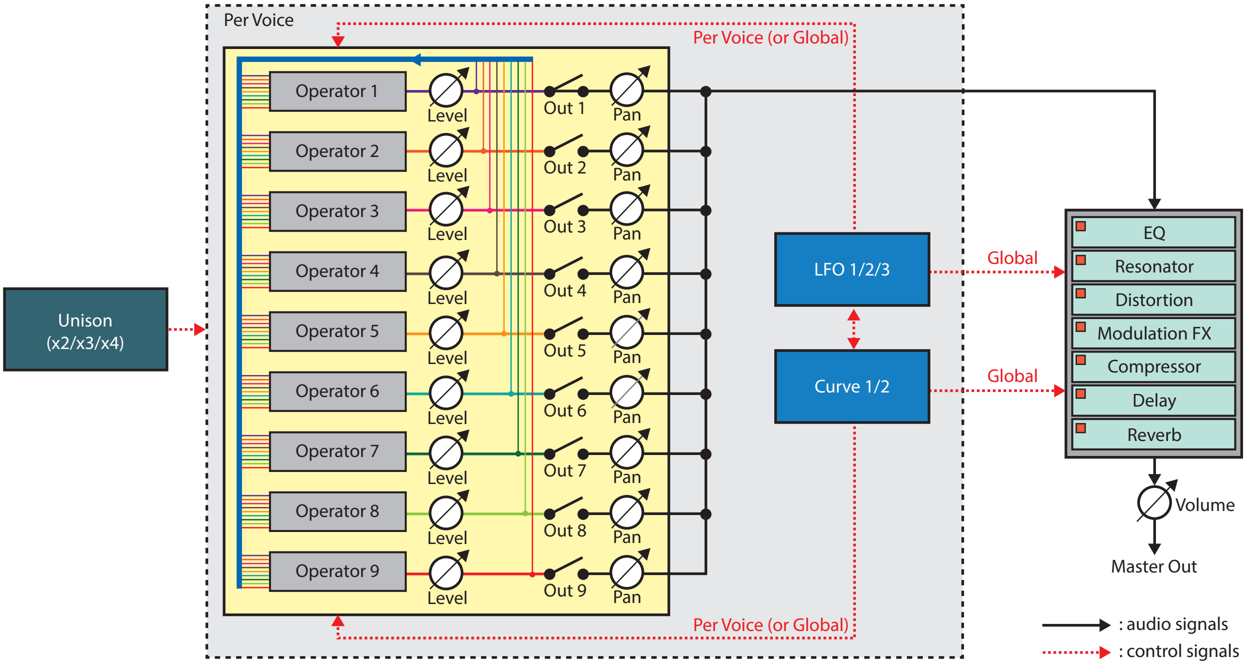
Panel overview

The Algoritm front panel contains the following sections:

- **1. Performance and "play control" section (see "Global performance and "play" controls")**
- **2. Patch Selector (for browsing, loading and saving patches) (see "Loading and saving patches")**
- **3. Randomizer (see "Randomizer")**
- **4. Master Volume control**
- **5. Operator Slots section (see "Operator Slots and Routing section overview")**
- **6. Routing section, where you create/edit the algorithms (see "Operator Slots and Routing section overview")**
- **7. Unison section (see "Unison")**
- **8. LFO & Curve section (to be used as Source in the Modulation Matrix) (see "The LFO & Curves section")**
- **9. Modulation Matrix section (see "Modulation Matrix")**
- **10. Effects section (see "The Effects section")**

Signal flow

Since Algoritm is a semi-modular instrument, the internal signal flow can be much more complex than a traditional pre-patched synthesizer. The picture below shows the main sections and the basic signal flow:



Basic signal flow in Algoritm.

! **The blue thicker cable in the yellow Operator Slots section above holds all available operator cables. In a standard patch, only a few of the operator cables are actually used, though.**

- **The control/modulation signals from the LFO and Curve generators are per voice when controlling operator parameters and global (monophonic) when controlling e.g. the Unison and Effects sections.**
- **In addition to the control signal routings shown in the picture, the Modulation Matrix can be used for routing control signals from the various operator parameters to selectable destinations, see "Modulation Matrix".**

Playing and using Algoritm

Loading and saving patches

Loading and saving patches is done in the same way as with any other internal Reason device. See the “Sounds and Patches” chapter in the Reason/Reason Rack Plugin/Reason Intro/Reason Lite Operation Manual pdf for details.

As with all Rack Extensions, you can find the included patches by clicking "Rack Extensions" in the Reason browser, navigating to the Algoritm folder and opening it.

Global performance and “play” controls

(Pitch) Range

→ **Set the desired Pitch Bend range for the “Pitch” wheel by dragging up/down in the display.**
Range: +/-24 semitones (+/-2 octaves) in steps of +/-1 semitone.

Pitch

The Pitch Bend wheel can be used for bending note pitches up and down. Algoritm also responds to Pitch Bend MIDI data from a connected MIDI master keyboard. You set the desired Pitch Bend range with the “(Pitch) Range” control to the left of the Pitch Bend wheel.

Mod Wheel

The Mod Wheel can be used for controlling almost any parameter in Algoritm. Use the Mod Wheel as a Source parameter in the Modulation Matrix section and then route to the desired Destination parameter(s), see “Modulation Matrix”.

Key Mode

Here you choose how Algoritm should respond to MIDI Note data:

- **Poly**
  Select this if you want to play Algoritm polyphonically.
- **Retrig**
  Select this if you want to play Algoritm in monophonic mode and always retrigger the envelopes as soon as you play a new note.
- **Legato**
  The Legato mode is also monophonic. However, if you play a new note without having released the previous one, the envelopes won’t start over.

Portamento

Portamento makes note pitches glide from previous notes to new ones, at the time set with the Time knob. Portamento can be used in all Key Modes (see above).

- **When On in Poly Key Mode (see above), the pitches will glide from any of the available voices.**
  The results will be unpredictable since there is no way of controlling from which note(s) the glide(s) will commence. The effect is very nice, though.
- **When On in Retrig or Legato Key Mode (see above), the pitch will glide between consecutive notes.**
- **In Auto mode, the pitch will glide between consecutive monophonic notes only when you play legato.**
  If you have selected Poly Key Mode (see above), Auto will have no effect.

  If you release the previous key before hitting the new key, there will be no portamento effect.

Octave

→ **Click and drag up/down in the display to select the desired Octave transposition.**
Range: +/-2 octaves.

Volume

This is the main stereo output volume control.

Panel reference

Operator Slots and Routing section overview

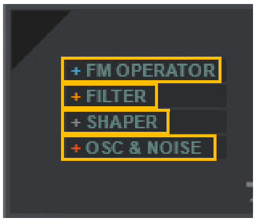
The nine Operator Slots section to the left and the Routing section to the top right

The Operator Slots section is where you select which types of operators (modules) you want to use in your patch (sound). You can then define the signal routing to, from and between the operators in the Routing section.

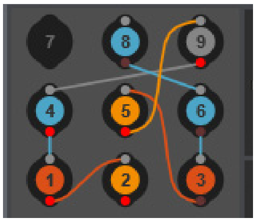
The paragraphs below show how to make the basic creations/connections/reroutings of operators. For more advanced routings (including FM and AM), please refer to "The Routing section".

Adding operators

→ **Click the desired label for the operator type you want to add:**



As the operator is added in the Operator Slot, the corresponding operator is also represented by a colored filled circle in the Routing section:

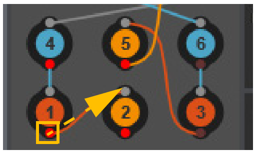


The color of the circle represents the added operator type:

- **Blue color represents an FM Operator.**
- **Light orange color represents a Filter.**
- **Grey color represents a Shaper.**
- **Dark orange color represents an Oscillator.**
- **A red dot below the operator indicates that its audio output is active.**
  To deactivate the audio output, just click the red dot.

Connecting operators

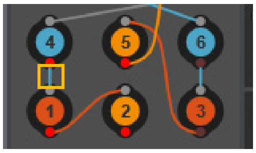
→ **Connect the operators by click-holding a connection point, then drag to the desired destination and release the mouse button:**



- **Inputs can only be connected to outputs, i.e. input-input or output-output connections are not possible.**
- **You could also connect the output of an operator to its own input, resulting in a feedback loop. See “Internal FM feedback” for more details.**

Disconnecting and rerouting operators

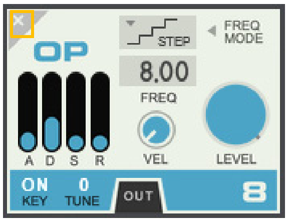
→ **To disconnect an operator, click the desired cable between the operators:**



→ **To reroute a connection, simply delete the existing connection and create a new one, as described in “Connecting operators” above.**

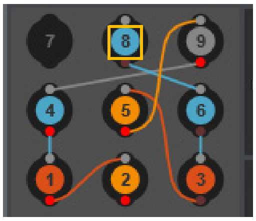
Deleting operators

→ **Click the X at the top left of the operator to delete it:**



! **Any connections will be preserved "hidden" in the background. If you then create another operator in the same operator slot, it will be automatically connected like the previous operator.**

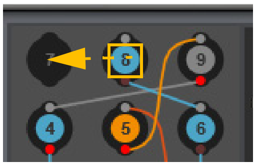
• **You can also delete an operator in the Routing section, by holding down [Cmd](Mac) or [Ctrl](Win) and clicking the operator:**



! **Deleting an operator in the Routing section will also automatically delete all its connections. If you then create another operator in the same operator slot, you have to manually connect it as described in "Connecting operators".**

Moving operators

→ **To move an operator to another position, drag the operator and drop at the new position in the Routing section.**
All connections will be preserved.



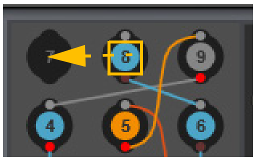
This will automatically move the operator to the corresponding position in the Operator Slots section as well.

! **Note that dropping the operator in an already occupied slot will automatically replace (delete) the operator in that slot!**

! **Note that modulation assignments made in the Modulation Matrix won't be updated to the new slot position, so you will have to update these manually, see "Modulation Matrix".**

Copying operators

→ **To copy an operator (with all its settings) and place in another slot, hold down [Option](Mac) or [Alt](Win) and drag the operator and drop at the new position in the Routing section:**



This will automatically copy the operator to the corresponding position in the Operator Slots section as well.

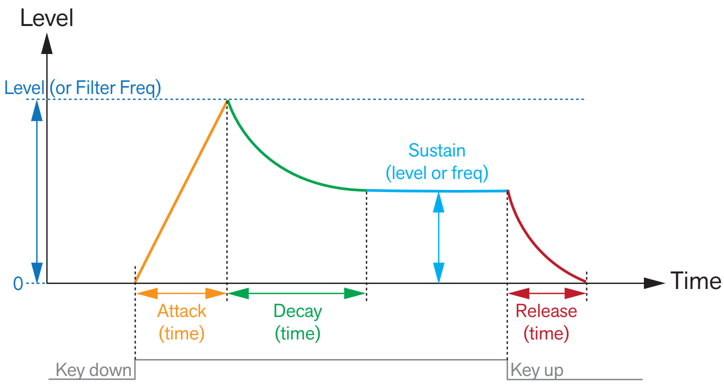
! **Note that dropping the copied operator in an already occupied slot will automatically replace (delete) the operator in that slot!**

Common operator parameters

The four different operator types (FM Operator, Oscillator & Noise, Filter and Shaper) have some parameters in common. These parameters are described in this section.

ADSR Envelope

The ADSR envelope controls the output level (or cutoff/center frequency for the Filter, if the Env knob is up) over time, with the Level parameter determining the maximum output level.



ADSR Envelope stages

- **A(ttack)**
  When you play a note on your keyboard, the envelope is triggered. This means it starts rising from zero to the value set with the Level knob (see "Level"). How long this should take, depends on the Attack setting. If the Attack is set to "0", the Level is reached instantly. If the Attack value is raised, it will take longer time before the Level is reached.
- **D(ecay)**
  After the Level has been reached, it starts to drop. How long this should take is governed by the Decay parameter.

  If you want to emulate the volume envelope of a note played on a piano for example, the Attack should be set to "0", the Decay parameter should be set to a medium value and the Sustain level should be set to "0", so that the volume gradually decreases down to silence, even if you keep holding the key down. Should you want the decay to drop to some other value than zero, you raise the Sustain parameter.
- **S(ustain)**
  The Sustain parameter determines the level the envelope should rest at, after the Decay stage. If you set Sustain to full level, the Decay setting is of no importance since the level is never lowered.

  If you want to emulate the volume envelope of an organ, you theoretically only really need to use the Sustain parameter set to full level, as a basic organ volume envelope instantly goes to the maximum level (Attack "0") and stays there (Decay "0"), until the key is released and the sound instantly stops (Release "0").

  But often a combination of Decay and Sustain is used to generate envelopes that rise up to the Level, then gradually decreases to finally land to rest on a level somewhere in-between zero and the Level. Note that Sustain represents a level, whereas the other envelope parameters represent times.

- **R(elease)**
  The Release parameter works just like the Decay parameter, except it determines the time it takes for the level to drop back to zero after you release the key.

Freq

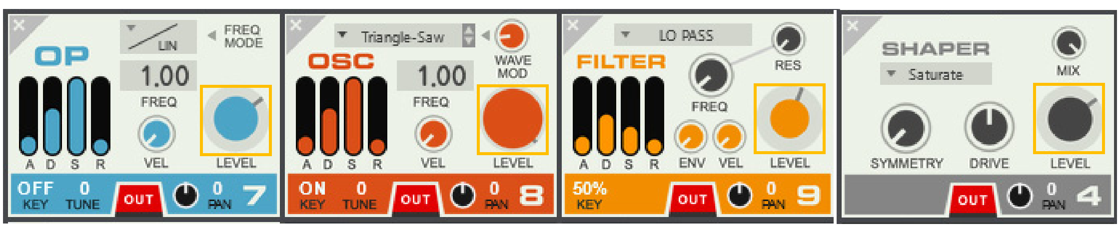
This is the operator frequency expressed as a factor, where "1.00" is the fundamental. "2.00" means the second harmonic (pitched one octave up), "3.00" is the third harmonic (pitched one octave plus on fifth up) and so on. "0.50" means half the fundamental frequency, i.e. one octave down, and “0.25” means two octaves down. On the FM Operator and Oscillator you can also set inharmonic ratios by changing the decimal part of this value. The Freq knob on the Filter controls the cutoff frequency (center frequency on the Bandpass filter).

→ **On the FM Operator and Oscillator, change Freq value by dragging either the integer or the decimal part of the value up or down.**
  As with any parameter, you can [Cmd](Mac)/[Ctrl](Win)-click to reset parameters. In this case, reset-clicking the decimal part sets it to “00” without changing the integer ratio, while reset-clicking the integer part resets the whole Freq value to “1.00”.

Vel

Sets how much the Keyboard Velocity (how hard or soft you play your MIDI keyboard) should scale the Envelope, and consequently the Level.

Level



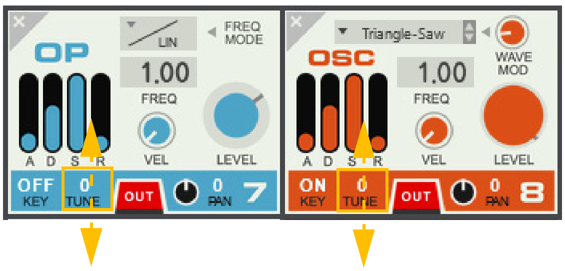
This determines the (maximum) output level from the operator. The level affects both the output volume (if the operator connects directly or indirectly to Out) and the FM amount (if it connects to the input of an FM operator) or AM amount (if it connects to the input of an Oscillator).

Key

Key determines how much the Frequency (pitch) of the operator (or cutoff/center frequency of the Filter) should track the keyboard notes. If you set this to OFF, the operator frequency will be fixed (where FREQ 1.00 matches the pitch/frequency of a C3 note). Set to 50% the pitch (or filter) tracks the keyboard 0.5 semitones per note. Set to ON, the pitch (or filte)r tracks the keyboard 1 semitone per note.

→ **Click and drag the Key "button" up/down to select the desired value.**
Values: OFF, 50% and ON (100%).

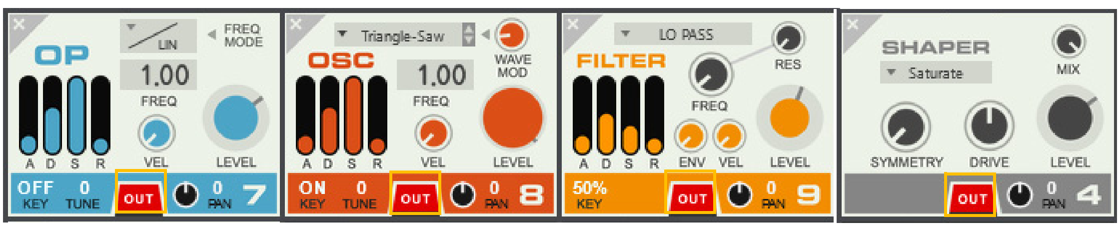
Tune



Tune is for fine-tuning the operator pitch (+/- 50 cents of a semitone).

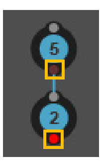
→ **Click and drag the Tune "button" up/down to select the desired value.**

Out



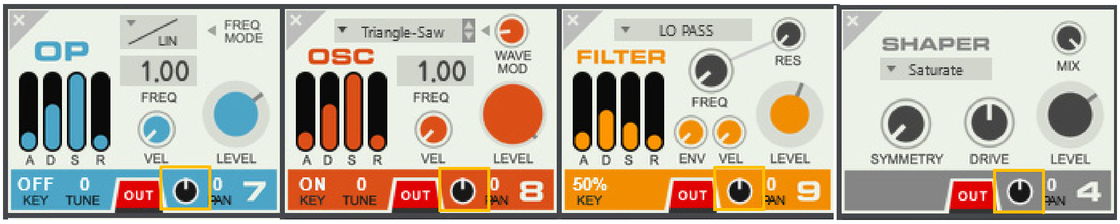
→ **Click the Out button to send the audio from the operator to the audio outputs (via the Effects section).**
If you want to use the operator for modulation purposes only (or route its audio through a Filter or Shaper), deactivate the Out button.

- **You can also activate/deactivate the Out button for an operator by clicking the corresponding output "dot" in the Routing section.**
A red dot indicates that Out is on:



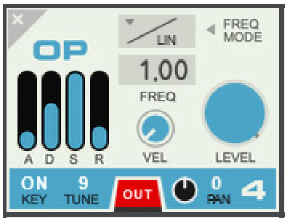
Pan

The Pan knob sets the stereo position of the output signal from the operator.



The Pan parameter is only shown and enabled if the Out button (see "Out" above) is on (red).

FM Operator

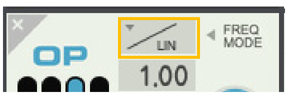


FM Operator

The FM Operator outputs a pure sine wave. It can be frequency-modulated by signals connected to the operator input (or via internal FM Feedback), if desired.

For descriptions of the common operator parameters, please refer to "ADSR Envelope", "Freq", "Vel", "Level", "Key", "Tune", "Out" and "Pan".

Freq Mode



→ **Click the Freq Mode display and select the desired mode from the pop-up.**

Freq Mode affects the results of the Freq setting:

- **In Linear (LIN) mode the output frequency follows the Freq setting exactly, including any inharmonic settings.**
- **In Stepped (STEP) mode the frequency is rounded to the nearest harmonic.**
  If you modulate the Freq parameter, the result will be "stepped harmonics".
- **In Fade mode, a non-integer Freq setting will produce a crossfade between the nearest harmonics above and below.**
  Modulating the Freq parameter in this mode will produce a sound similar to oscillator sync (especially with feedback routings, see "Internal FM feedback").

Oscillator & Noise

Oscillator & Noise

This is a wavetable oscillator that can be used as a sound source (directly or via a Shaper/Filter) or as an FM modulator, by connecting it to the input of an FM Operator.

If a signal is connected to the input of the Oscillator, the result will be AM (amplitude modulation).

For descriptions of the common operator parameters, please refer to "ADSR Envelope", "Freq", "Vel", "Level", "Key", "Tune", "Out" and "Pan".

Waveform selector and Wave Mod knob

The Wave pop-up is where you select the wavetable. The selection includes traditional synth waveforms as well as complex waves.

The Wave Mod knob defines the position in the wavetable, affecting the resulting waveform. In a wavetable like PulseWidth this morphs from a square wave to a narrow pulse; in a more complex wavetable this may morph between widely different waveforms.

The following waveforms are available:

- **Triangle-Saw**
  A triangle wave at Wave Mod=0%, gradually transformed into a negative ramp sawtooth wave at Wave Mod=100%.
- **PulseWidth**
  A 50% duty cycle square wave at Wave Mod=0%, gradually transformed towards a 95% duty cycle pule wave at Wave Mod=100%.

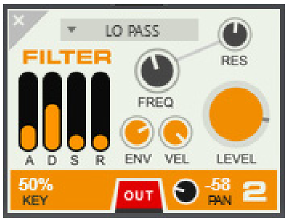
► **Modulate the Wave Mod parameter from an LFO to achieve PWM, see "Modulation Matrix".**

- **Sine-Noise**
  A pure sinewave at Wave Mod=0%, gradually transformed into white noise at Wave Mod=100%.

► **For noise, use the Sine-Noise wavetable and turn the Wave Mod knob all the way up.**

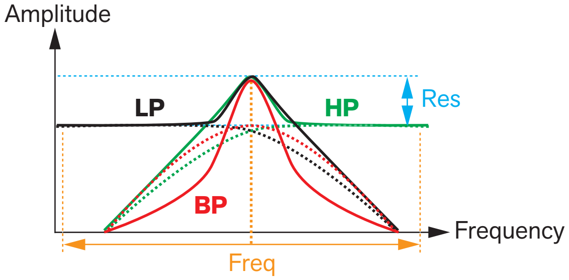
The remaining waveforms consists of various wave tables with all sorts of different characters.

Filter



This is a state variable (SVF) resonant Lowpass/Bandpass/Highpass filter with a 12dB/octave slope. The filter is similar to the State Variable Filter in the Europa synthesizer.

For descriptions of the common operator parameters, please refer to “ADSR Envelope”, “Freq”, “Vel”, “Level”, “Key”, “Tune”, “Out” and “Pan”.



Filter types and characteristics

Filter Type selector

→ **Click the Filter Type selector and select the desired filter type from the pop-up:**

- **LO PASS**
  A lowpass (LP) filter with a 12dB/octave slope.
- **BAND PASS**
  A bandpass (BP) filter with 12dB/octave slopes.
- **HI PASS**
  A highpass (HP) filter with a 12dB/octave slope.

Res

This sets the resonance, i.e. the amplification at, and around, the cutoff/center frequency.

Env

This defines how much the Envelope should affect the Freq parameter.

Shaper

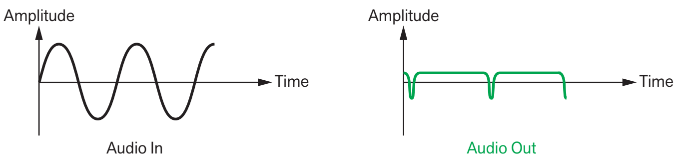
A shaper (wave shaper) distorts the incoming signal in various ways, by multiplying the input signal with a transformation function. It can distort an audio signal routed to its input, or it could change the timbre of an FM modulator, if the Shaper output is routed to the input of an FM Operator.

For descriptions of the common operator parameters, please refer to "Level", "Out" and "Pan".

Shaper Type selector

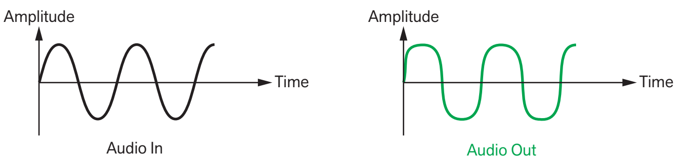
→ **Select the desired filter shaper transformation function (distortion type) from the pop-up:**

- **Power2**



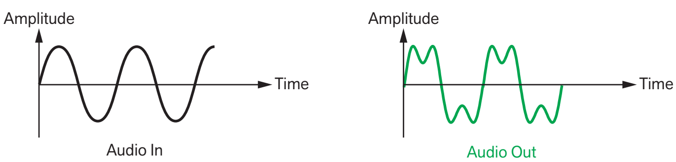
The Power2 shaper type multiplies the input signal by itself (x^2). With increasing Drive amount the fundamental frequency of the signal is gradually attenuated more and more.

- **Saturate**



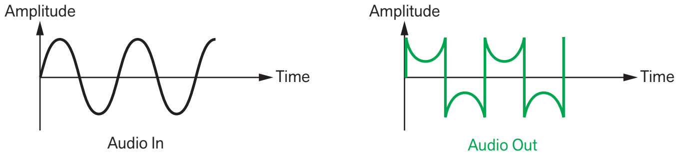
The Saturate shaper type amplifies the signal up to the available headroom and thereby changes the shape of the signal where it hits the headroom "ceiling". For example, a pure triangle wave would transform into more of a square (pulse) wave with increasing Drive amount.

- **Fold**



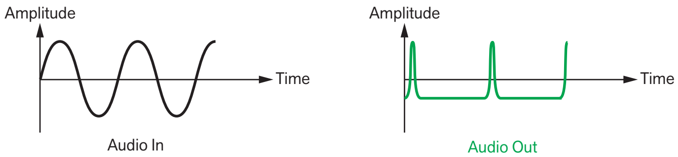
The Fold shaper type amplifies the signal above the available headroom and then "mirrors" the peaks down into the available headroom.

- **Bipulse**



The Bipulse creates a very overtone rich distortion by using a pulse shaped amplification curve (transfer function).

- **Electro**



The Electro shaper type emulates the characteristics of pick-ups in electric pianos, by adding overtones.

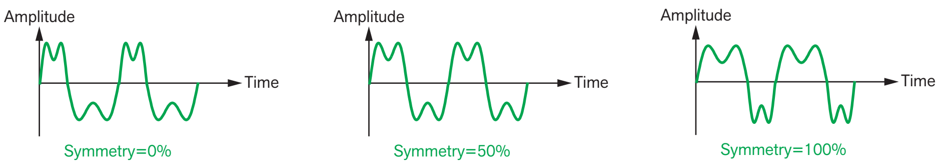
Drive

Drive controls the amount of distortion.

Symmetry

This sets the waveform symmetry.

At 50% the signal is unaffected. At either directions the symmetry of waveform cycle is gradually altered:

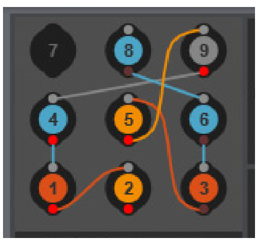


A sinewave signal with some Fold distortion, at 0%, 50% and 100% Symmetry settings.

Mix

Mix blends the direct and processed signals. Note that the Level control affects both the direct and processed signals. This allows you to use the Shaper as an plain gain control that doesn't affect the timbre (with Mix set to 0).

The Routing section



As described in earlier in this manual, the Routing section is tightly associated with the Operator Slots section. The filled circles represent the operators of the Operator Slots section, and the small cables between the circles indicate the operator connections.

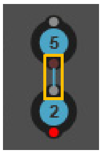
You can manually connect and reroute the operator signals in various ways. You can also move, copy and delete operators here. The basic operations of the Routing section is described in "Operator Slots and Routing section overview", so here we will focus on connecting specific operators for various purposes.

FM modulation from other operator

FM Operators can be frequency-modulated by an external signal connected to its input.

1. **Create one FM Operator and then another one right above it. Make sure the Out button on the lower FM Operator is set to On (red) and on the upper to OFF (black):**

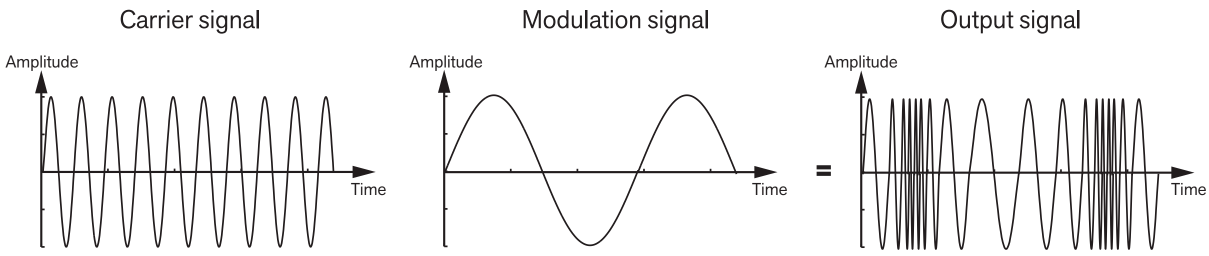
2. **In the Routing section, click and hold the input connector of the lower operator, drag to the output connector of the upper operator and release the mouse button:**



3. **Raise the Level knob on the upper FM Operator to increase the frequency-modulation amount of the lower FM Operator.**
   While the Freq parameter of the carrier (operator 2 in this example) affects the pitch of the signal as usual, the Freq parameter of the modulator (operator 5) will now affect the overtones and timbre of the resulting sound.

► **You could also use an Oscillator & Noise operator as frequency modulation source instead of an FM Operator.**

The picture below shows the frequency modulation principle. The signal to be modulated is called "carrier" in FM language (Operator 2 in the example above) and the modulating signal is called "modulator" (Operator 5 in the example above). In this example the modulation signal has a much lower frequency than the carrier signal, to better visualize the output FM signal:

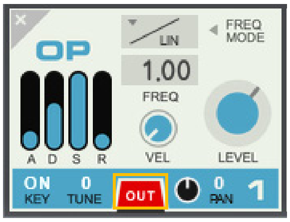


FM modulation principle.

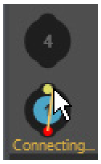
Internal FM feedback

FM Operators can have internal feedback. This means that the output signal is routed back to the input, to frequency-modulate its own signal. Internal FM Feedback results in a sawtooth-like waveform at moderate modulation, and transforms towards noise at higher feedback levels.

1. **Create an FM Operator and make sure the Out button is set to On (red):**



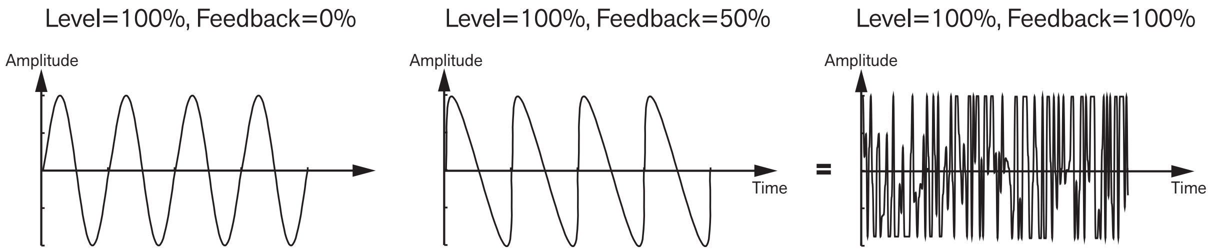
2. **Click and hold the output (or input) connector in the Routing section, drag to the input connector and release the mouse button:**



The internal feedback connection is now indicated by a blue cable:

3. **Raise the Feedback knob to the right of the Routing section to increase the feedback level.**

The picture below shows the internal feedback frequency modulation principle. In the FM feedback configuration the modulation signal has the same frequency as the carrier signal (since they are both from the same original signal):



Internal FM feedback modulation principle.

A note about internal feedback in other operator types

Although it is technically possible to create internal feedback connections in the other operator types as well (Oscillator & Noise, Shaper and Filter), it's not very useful in most situations.

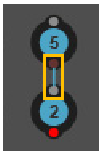
! **Note that having an internal feedback connection in an Oscillator & Noise operator, and using high Level and Feedback settings, might cause unpleasant noises/pops etc. so be careful.**

FM modulation from other operator, plus feedback

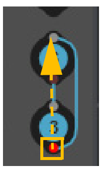
You could also create a feedback loop which includes several FM Operators.

1. **Create one FM Operator and then another one right above it. Make sure the Out button on the lower FM Operator is set to On (red) and on the upper to OFF (black):**

2. **In the Routing section, click and hold the input connector of the lower operator, drag to the output connector of the upper operator and release the mouse button:**



3. **Then, click and hold the output connector of the lower FM Operator, drag to the input connector of the upper FM Operator and release the mouse button.**
   You now get a feedback connection from the lower to the upper FM Operator:



4. **Raise the Feedback knob to the right of the Routing section to increase the feedback level.**

Important note about feedback connections

The sonic result of a feedback connection can differ depending on the numerical order of the operator slots as well as on which operator slots have operators with Out turned on. This is because all operator audio outputs are rendered in order according to the slot numbers - from slot 1 to slot 9.

In the example below we have a simple 2-operator feedback routing of FM Operators in slots 1 and 2. Note that both FM Operators have their Outs set to On - otherwise the slot positions wouldn't affect the sound in this example. We have also set the parameters a little differently, so that there are different sounds in the two FM Operators.

In this slot configuration the FM Operator 1 audio is calculated first (since it's in the first operator slot) and the feedback takes place between the FM Operator 1 output and the FM Operator 2 input (marked with a yellow rectangle in the Routing section):

Now, if we move FM Operator 1 to the operator slot 5 instead, the FM Operator 2 audio is calculated first (since it's now positioned first in the operator slot order. The feedback now takes place between the FM Operator 2 output and FM Operator 5 input instead (marked with a yellow rectangle in the Routing section) and you get a different sonic result:

! **If only one of the FM Operators would have its Out set to On, the slot positions wouldn't matter at all - the sound would be the same.**

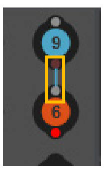
Oscillator AM modulation from other operator

If you connect an output signal (e.g. from an FM Operator or an Oscillator) to the input of an Oscillator, you will get amplitude modulation (AM) of the Oscillator signal. AM typically creates less dramatic timbre changes than FM. Detuning the modulating operator slightly results in subtle fluctuations, while using non-integer Freq settings can give you bell-like tones.

Here is an example where an FM Operator amplitude-modulates an Oscillator signal:

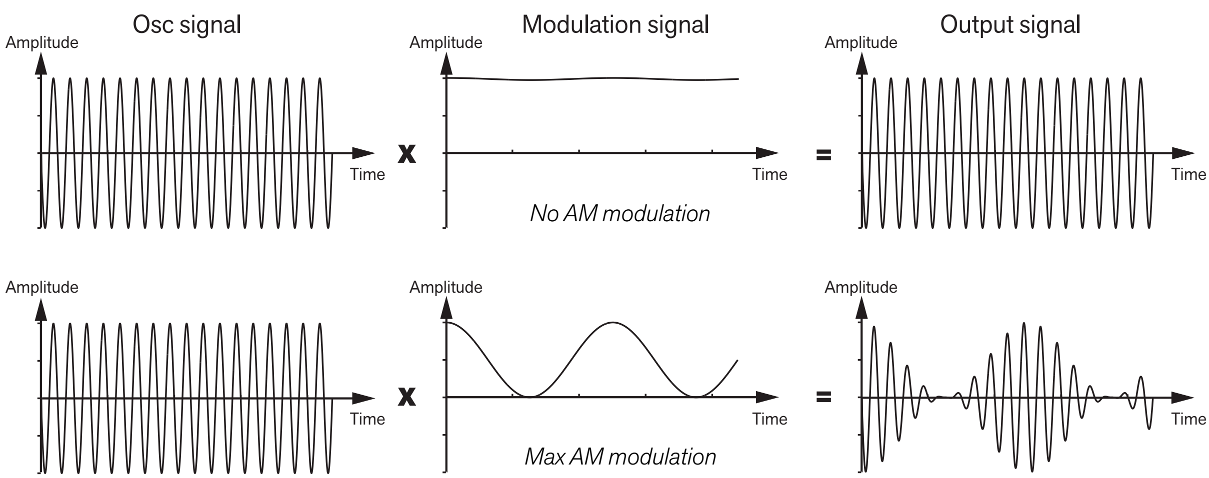
1. **Create an Oscillator and make sure the Out button is set to On (red).**
2. **Create an FM Operator above the Oscillator and make sure the Out button is set to Off (black):**

3. **Click and hold the FM Operator output connector in the Routing section, drag to the input connector of the Oscillator and release the mouse button:**



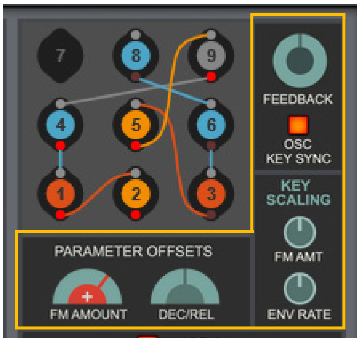
4. **Raise the Level knob on the FM Operator to increase the amplitude-modulation amount of the Oscillator.**

The picture below shows the amplitude modulation principle. In the example the modulation signal has a much lower frequency than the oscillator signal, to better visualize the output signal:



Principle for AM modulation.

Global parameters



Feedback

If you have made any feedback connections (one operator slot routing back to itself, directly or via other modules), this determines the level of the feedback signal.

! **Note that the Feedback affects all feedback signals in the patch.**

Osc Key Sync

When turned on, all FM Operators and Oscillators will be phase synced so that their waveforms start at zero. When turned off, all waveforms will start at a random phase.

! **Note that this can affect the result of Unison (see "Unison")!**

Key Scaling FM Amt

This scales the Frequency Modulation amounts according to the keys being played. This is a bipolar parameter. At positive values, the FM amount will increase for higher notes (from C3 and up) and decrease for lower notes. At negative values, the opposite happens.

Key Scaling Env Rate

This scales the Decay and Release times for all envelopes according to the keys being played. This is a bipolar parameter. At positive values, the decay/release will become longer for higher notes (from C3 and up) and shorter for lower notes. At negative values, the opposite happens.

FM Amount Parameter Offset

This is a macro control, affecting the internal Frequency Modulation amounts for all FM Operators. This is excellent for varying the timbre and aggressiveness of a sound, particularly when assigned to a performance control in the Modulation Matrix. The parameter is bipolar, with "0" at the twelve o'clock position.

Dec/Rel Parameter Offset

This is a macro control that scales the decay and release of all envelopes. This is a quick way to make a sound snappier or more sustaining. The parameter is bipolar, with "0" at the twelve o'clock position.

Unison

The Unison section multiplies complete voices, to create really nice and fat sound. It has the following parameters:

Count

Count determines how many voices will be played for each note: 2, 3 or 4.

→ **Drag up/down in the display to select the desired number.**

Timing

Timing offsets the timing of each voice randomly.

Detune

Detune affects the pitch of the Unison voices, causing a richer sound.

Spread

Spread affects the stereo position of the Unison voices, causing a wider sound.

Blend

Blend is a dry/wet control, mixing the single voice and the Unison voice(s).

The LFO & Curves section

The LFO & Curves section features three separate LFOs and two separate Curve generators for parameter modulation purposes. The modulation destinations can then be freely assigned in the Modulation Matrix, see "Modulation Matrix".

LFO 1/2/3

An LFO (Low Frequency Oscillator) is used for generating cyclic modulation. A typical example is to have an LFO modulate the pitch of a signal to produce vibrato, but there are countless other applications for LFOs.

The LFO section features three separate general purpose LFOs, that can be assigned to control selectable parameter(s) in the Modulation Matrix, see "Modulation Matrix".

→ **Select which of the three LFOs you want to edit by clicking one of the LFO 1, LFO 2 and LFO 3 buttons.**

→ **Select an LFO waveform by clicking the spin controls to the right of the waveform display, or by click-holding in the display and dragging up/down.**
Apart from the standard waveforms (sine, triangle, pulse, etc.) there are random, slope and stepped waveforms. The shape of the waveforms are shown in the display.

→ **Set the LFO frequency with the Rate knob.**
If Beat Sync (see below) is on, the Rate knob controls time divisions instead.

→ **Turn the Delay knob to introduce a delay before the LFO modulation kicks in after a note is played.**
Turn clockwise for longer delay times.

→ **Click the Beat Sync button to sync the LFO to the main sequencer Tempo.**
The Rate parameter now controls the time divisions.

→ **Click the Key Sync button to restart the LFO at every new Note On.**

→ **Click the Global button to make the LFO common for all voices (monophonic).**

Curve 1/2

There are two separate Curve generators available for modulating selectable parameter(s) in the Modulation Matrix, see "Modulation Matrix". The Curve generators can serve as additional LFOs, semi-step sequencers or even additional envelopes.

→ **Select which Curve you want to edit by clicking the Curve 1 or Curve 2 button.**

→ **To create steps or a curve shape, draw by clicking and dragging in the display:**

→ **To switch between steps and a curve shape, click the Stepped button below the display.**

- **Set the number of steps (1-16) by clicking or dragging in the bar at the top of the curve display:**

→ **Set the speed of the Curve with the Rate knob.**
If Beat Sync is on, the Rate value is shown as the length of the entire 16 step cycle. For example, if Rate is 4/4, each of the 16 steps corresponds to a 1/16 note.

→ **Click the One Shot button to trig the curve once from the start when you play a note.**
This makes it work like an envelope.

→ **Click the Bipolar button if you want the values to be bipolar (+/-) instead of unipolar (+ only).**

→ **Click the Key Sync button to restart the Curve at every new Note On.**

→ **Click the Global button to make the Curve common for all voices (monophonic).**

The Effects section

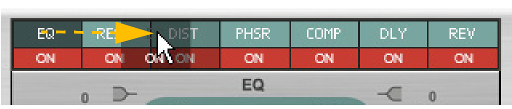
The Effects section features seven different effect modules that can be freely reordered by dragging & dropping. Most of the effect parameters are available as destinations in the Modulation Matrix, see “Modulation Matrix”.

At the top of the Effects section are seven Effect buttons. Click any of these to bring up the control panel for the corresponding effect:

Below the Effect buttons are the On/Off buttons for the individual effects. Click these to activate/deactivate the desired effects:

Reordering the effects

→ **To define the order of the effects in the serial chain, click and hold the desired Effect button, drag sideways to the desired position and release the mouse button:**



Moving the EQ effect to another position in the effects chain.

You can reorder the effects whenever you like.

EQ

The EQ has high and low shelving bands with fixed frequencies, plus a parametric mid band.

- **Lo Gain**
  Specifies how much the level of the low band should be boosted (positive values) or lowered (negative values).
- **Freq**
  This determines the center frequency of the EQ, e.g. at which frequency the level should be decreased or increased.
- **Q**
  This governs the width of the affected area around the set center frequency. The higher the value, the narrower the affected frequency range.
- **Gain**
  Specifies how much the level of the selected frequency range should be boosted (positive values) or lowered (negative values).
- **Hi Gain**
  Specifies how much the level of the high band should be boosted (positive values) or lowered (negative values).

Resonator

The Resonator is a convolution-based resonator/body effect with 21 built-in resonator types (impulse responses). It has a huge effect on the final sound and is great for giving a sound a certain character.

- **Resonator selector**
  Select the desired Resonator type by clicking the spin controls, or by clicking in the display and selecting from the pop-up menu that appears.
- **Pitch**
  The Pitch setting detunes the impulse response, which can generate radically different sounds. The Pitch parameter can also be viewed as “body size”, with lower Pitches simulating larger bodies, and higher Pitches simulating smaller bodies.
- **Decay**
  This sets the decay time of the resonator.
- **Width**
  This adjusts the stereo width of the resonator sound.
- **Amount**
  This controls the mix between the processed signal and the unprocessed sound.

Distortion

The Distortion effect offers six types of distortion with drive, tone and amount (dry/wet).

- **Distortion type selector**
  Select the desired distortion type with the selector.

  "Dist" produces a dense, rich analog type of distortion.
  "Scream" produces a less bright type of distortion.
  "Tube" emulates a tube type of distortion.
  "Sine" is a sine shaping distortion.
  "S/H" gives the effect of sample rate reduction.
  "Ring" is a ring modulator effect.

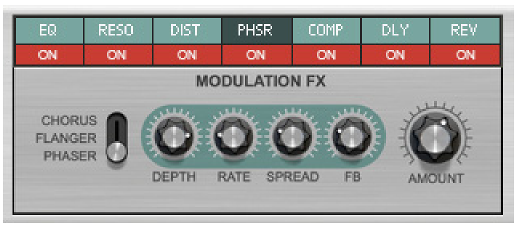
- **Drive**
  Sets the overdrive/feedback level of the selected distortion.

  For the "Ring" (ring modulator) and "S/H" (sample and hold) modes, the Drive adjusts the character of the effect rather than the distortion amount.

- **Tone**
  This is a lowpass filter and sets the tone of the selected distortion.

- **Amount**
  Sets the Dry/Wet amount of the distortion.

Phaser/Flanger/Chorus



This is a stereo Phaser/Flanger/Chorus.

→ **Select effect type with the Phaser/Flanger/Chorus switch.**
  The selected effect type is displayed on the Effect button.

- **Depth**
  Sets the depth of the selected effect. To get a static sound, set Depth to zero.

  It's possible to turn down the Depth completely and instead modulate the Frequency of the effect in the Mod Matrix, from an LFO, Curve or any other source.

- **Rate**
  Sets the rate/speed of the modulation.

- **Spread**
  Sets the stereo width of the effect.

- **FB**
  Sets the feedback of the selected effect.
- **Amount**
  Sets the Dry/Wet amount of the effect.

Compressor

This is a stereo compressor with attack, release, threshold and ratio controls. To adjust the level after compression, use the Master Volume of Algoritm.

- **Attack**
  This governs how quickly the compressor will apply its effect when signals rise above the set threshold. If you raise this value, the response will be slower, allowing more of the signal to pass through the compressor unaffected. Typically, this is used for preserving the attacks of the sounds.
- **Release**
  When the signal level drops below the set threshold, this determines how long it takes before the compressor lets the sound through unaffected. Set this to short values for intense, "pumping" compressor effects, or to longer values for a smoother change of the dynamics.
- **Thres**
  This is the threshold level above which the compression sets in. Signals with levels above the threshold will be affected, signals below it will not. In practice, this means that the lower the Threshold setting, the more the compression effect.
- **Ratio**
  This specifies the amount of gain reduction applied to the signals above the set threshold.

Delay

This is a stereo delay that can be synced to tempo or not. For stereo movement, turn on Ping-Pong or modulate Delay Pan in the Modulation Matrix. The Amount parameter works as a Send level and can also be modulated.

- **Sync**
  Activate Sync to sync the delay time to the main sequencer Tempo.
- **Ping Pong**
  Activate Ping Pong to have the delay repeats alternating between left and right in the stereo panorama. The effect is also dependent on the Pan parameter (see below).

- **Time**
  This sets the time between the delay repeats. If Sync is active (see above), the Time parameter now controls the time divisions.
- **Feedback**
  The Feedback parameter determines the number of delay repeats.
- **Pan**
  Sets the panning of the delay repeats in the stereo panorama. If Ping Pong is active (see above) the Pan knob controls the panning of the initial delay repeat as well as the total stereo spread of the remaining repeats.
- **Amount**
  Use this parameter to adjust the send level to the Delay effect.

► **If you play a note, have a long delay feedback and turn down Amount, the echoes will continue. This allows for automated "triggered delay" fx.**

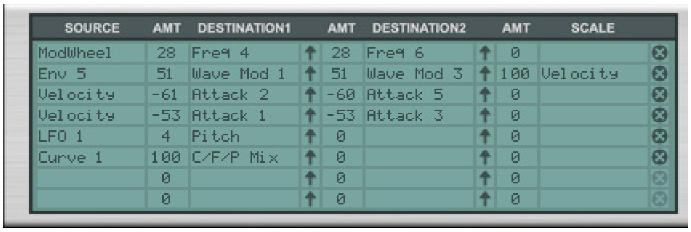
Reverb

This is a stereo algorithmic reverb with control over high frequency damping and Early Reflections level. The Amount parameter works as a Send level and can be modulated in the Modulation Matrix.

- **Decay**
  This governs the length of the reverb effect.
- **Size**
  Sets the emulated room size, from small room to large hall. Middle position is the default room size.
  Lowering this parameter results in a closer and gradually more "canned" sound. Raising the parameter results in a more spacey sound, with longer pre-delay.
- **Damp**
  Raising the Damp value cuts off the high frequencies of the reverb, thereby creating a smoother, warmer effect.
- **ER Level**
  This controls the amount of early reflections in the reverb sound.
- **Amount**
  Use this parameter to adjust the send level to the Reverb effect.

  If you play a note, have a long delay Decay time and turn down Amount, the reverberation will continue.

Modulation Matrix



The Modulation Matrix is used for routing a modulation Source to one or two modulation Destinations each. This creates a very flexible routing system.

The Modulation Matrix in Algoritm is derived from the ones in the Europa and Grain devices, so if you are familiar with these, you will quickly find your way around in Algoritm.

There are eight "Source –> Destination 1 –> Destination 2 –> Scale" busses, which all can be freely assigned.

A Source parameter can modulate two different Destination parameters per bus (with variable Amount settings). Each bus also has a Scale parameter that affects the relative modulation Amount for both Destinations.

► **Note that it is possible to assign the same source parameter as Source in several busses. This allows you to control more than two Destination parameters from the same Source.**

1. Select the desired Source parameter by clicking in the corresponding Source box and selecting from the list.

The following parameters can be used as modulation Sources:

Parameter	Description
Velocity	This applies modulation according to the Keyboard Velocity values (how hard or soft you strike the MIDI keyboard keys).
Key	This is the currently played key (i.e. keyboard tracking). If a positive Amount value is used and the destination is filter frequency, the filter frequency will track the keyboard, i.e. increase with higher notes.
Key In Octave	The current key played, but 12 different values, one for each note in an octave (in other words, this disregards in which octave you play).
Random	This sends out a random bipolar value each time a new note is played.
Noise	This allows you to modulate parameters from white noise.
Unison Index	This sends out different unipolar values to the different voices when Unison is used. For example, you could set the Destination to the Freq parameter of an operator to have the Unison voices tuned in intervals.
LFO & Curves (LFO 1, LFO 2, LFO 3, Curve 1 and Curve 2)	This allows you to modulate parameters from LFO 1, LFO 2, LFO 3, Curve 1 and Curve 2 respectively.
Envelopes (1-9)	This allows you to modulate parameters from any of the Operator Envelopes in Slot 1-9 (provided that there are operators that features Envelopes in the respective Slot).
Mod Wheel	This allows you to modulate parameters from the Mod Wheel.
MW Latched	This allows you to modulate parameters based on the current Mod Wheel value at a given Note On.
Pitch Wheel	This allows you to modulate parameters from the Pitch Bend control.
Breath	This allows you to modulate parameters from the Breath performance controller
Expression	This allows you to modulate parameters from the Expression performance controller
Aftertouch	This allows you to modulate parameters from Keyboard Aftertouch (channel aftertouch)
Sustain	This allows you to modulate parameters from a connected sustain pedal. Note that continuous sustain data (0-127) is supported - not just on/off.
CV Input 1/2/3/4	This takes the current value on the CV 1/CV 2/CV 3/CV 4 inputs on the rear panel and sends to the desired destination.

Modulation Matrix Source parameters.

2. Set the Amount for the first Destination by turning the corresponding Amount knob, or by clicking and dragging vertically in the corresponding Amount box.

! Note that the Amount range is +/-100. This means that the Amount value can exceed the modulated parameter's range. When this happens, the modulated parameter simply stays at its extreme value until the Amount value gets within the parameter's range again.

3. Select the first Destination parameter by click-holding the grey arrow symbol to the right of the corresponding Destination box.

4. While click-holding, drag to the desired destination parameter on the panel:

Assigning LFO 2 Rate as Destination for the Envelope of the operator in Slot 1.

As you hover over a valid destination control on the panel, the parameter name is automatically displayed in the Destination box in the Modulation Matrix.

5. To assign the currently selected Destination control, release the mouse button.

→ Alternatively, click the desired Destination box and select the Destination parameter from the list.

The following parameters can be used as modulation Destinations:

Parameter	Description
Pitch	This affects the (full range) pitch of all operators.
Feedback	This affects the Feedback parameter in the Global parameters section.
FM Amount Offset	This affects the FM Amount Offset parameter in the Global parameters section.
Decay/Release Offset	This affects the Dec/Rel Offset parameter in the Global parameters section.
Portamento Time	This affects the global Portamento Time parameter.
Tune (1-9)	This affects the operator Tune parameter in Operator Slots 1-9 (provided an operator is available).
Freq (1-9)	This affects the operator Freq parameter in Operator Slots 1-9 (provided an operator is available).
Level (1-9)	This affects the operator Level parameter in Operator Slots 1-9 (provided an operator is available).
Shaper Drive (1-9)	This affects the Shaper Drive parameter in Operator Slots 1-9 (provided a Shaper is available).
Shaper Symmetry (1-9)	This affects the Shaper Symmetry parameter in Operator Slots 1-9 (provided a Shaper is available).
Shaper Mix (1-9)	This affects the Shaper Mix parameter in Operator Slots 1-9 (provided a Shaper is available).
Wave Modulation (1-9)	This affects the Wave Mod parameter in Operator Slots 1-9 (provided an Oscillator is available)
LFO & Curves (LFO 1, LFO 2, LFO 3, Curve 1 and Curve 2)	This affects the Rate parameter of the selected LFO/Curve.
Reverb Decay	This affects the Decay parameter in the Reverb effect.
Reverb Amount	This affects the Amount parameter in the Reverb effect.
Delay Time	This affects the Time parameter in the Delay effect.
Delay Feedback	This affects the FB parameter in the Delay effect.
Delay Amount	This affects the Amount parameter in the Delay effect.

Parameter	Description
Delay Pan	This affects the Pan parameter in the Delay effect.
Dist Drive	This affects the Drive parameter in the Dist effect.
Dist Tone	This affects the Tone parameter in the Dist effect.
Dist Amount	This affects the Amount parameter in the Dist effect.
Comp Release	This affects the Release parameter in the Compressor effect.
Comp Ratio	This affects the Ratio parameter in the Compressor effect.
Mod Effect Frequency	This affects the center frequency of the Chorus/Flanger/Phaser effects.
Mod Effect Mix	This affects the Amount parameter of the Chorus/Flanger/Phaser effects.
EQ Frequency	This affects the Freq parameter in the EQ effect.
EQ Gain	This affects the Gain parameter in the EQ effect.
Reso Amount	This affects the Amount parameter in the Resonator effect.
Unison Detune	This affects the Detune parameter in the Unison section.
Unison Spread	This affects the Spread parameter in the Unison section.
Unison Blend	This affects the Blend parameter in the Unison section.
Attack (1-9)	This affects the Envelope Attack time in Operator Slots 1-9 (provided an operator is available).
Decay (1-9)	This affects the Envelope Decay time in Operator Slots 1-9 (provided an operator is available).
Sustain (1-9)	This affects the Envelope Sustain level in Operator Slots 1-9 (provided an operator is available).
Release (1-9)	This affects the Envelope Release time in Operator Slots 1-9 (provided an operator is available).
Envelope Gates (1-9)	This affects the Envelope Gate in Operator Slots 1-9 (provided an operator is available). Note that the regular gating from played notes will be deactivated
Pan (Voice, 1-9)	This affects the Pan parameter in Operator Slots 1-9 (provided an operator is available). "Voice Pan" pans the whole sound at the voice mix stage.
CV Outputs	This sends out the source modulation value(s) on the CV1/2/3/4 Output on the rear panel.

Modulation Matrix Destination parameters.

6. **Set the Amount for the second Destination (if desired) by turning the corresponding Amount knob, or by clicking and dragging vertically in the Amount box for the second destination.**
7. **If desired, select a second Destination parameter by click-holding the grey arrow symbol to the right of the corresponding Destination box, and dragging to the desired control on the panel.**
8. **If desired, click the Scale box and select a Scale parameter.**
   The available Scale parameters are the same as the Source parameters, see "Modulation Matrix Source parameters.".
9. **Turn the Scale Amount knob, or click the Amount box to the left of the Scale box and move the mouse pointer up or down to set a Scale Amount value.**
   Both positive and negative Scale Amount values can be set (+/- 100%). If you, for example, are using the Mod Wheel as Scale parameter and don't want any modulation when the Mod Wheel is set to zero, set the Scale Amount parameter to 100%. Then, there will be no effect when the Mod wheel is set to zero, and full modulation when the Mod Wheel is all the way up.

- **How much modulation will be applied when the Scale parameter is set to maximum is governed by the to Destination Amount parameter(s).**
- **How much the Scale parameter controls the modulation is set with the Scale Amount parameter.**

→ **To clear an assigned Source, Destination or Scale parameter, hold down [Ctrl](Win) or [Cmd](Mac) and click the Source/Destination/Scale box. Alternatively, click the Source/Destination/Scale box and select "Off" from the list.**

→ **To reset an Amount value to 0, hold down [Ctrl](Win) or [Cmd](Mac) and click the desired Amount box or knob.**

→ **To clear an entire modulation assignment (a whole row), click the circular X button to the right of the corresponding Scale box.**

Randomizer

The Randomizer lets you randomize various parameters in a patch. This is great for experimenting and a fantastic source for inspiration! The randomization can be controlled to generate both subtle variations and completely new sounds.

1. **Click the desired button(s) to select which properties you want to change:**

Freq, Mode (Harmonic mode, Shaper and Filter Types, Wavetable and Wave Mod settings), Level, Envelopes or Algorithm (which type of operators are selected for the slots and how they are connected).

► **Note that you can select multiple properties (buttons), to be able to randomize all of them simultaneously.**

2. **Click the Randomizer bar and keep the mouse button pressed:**

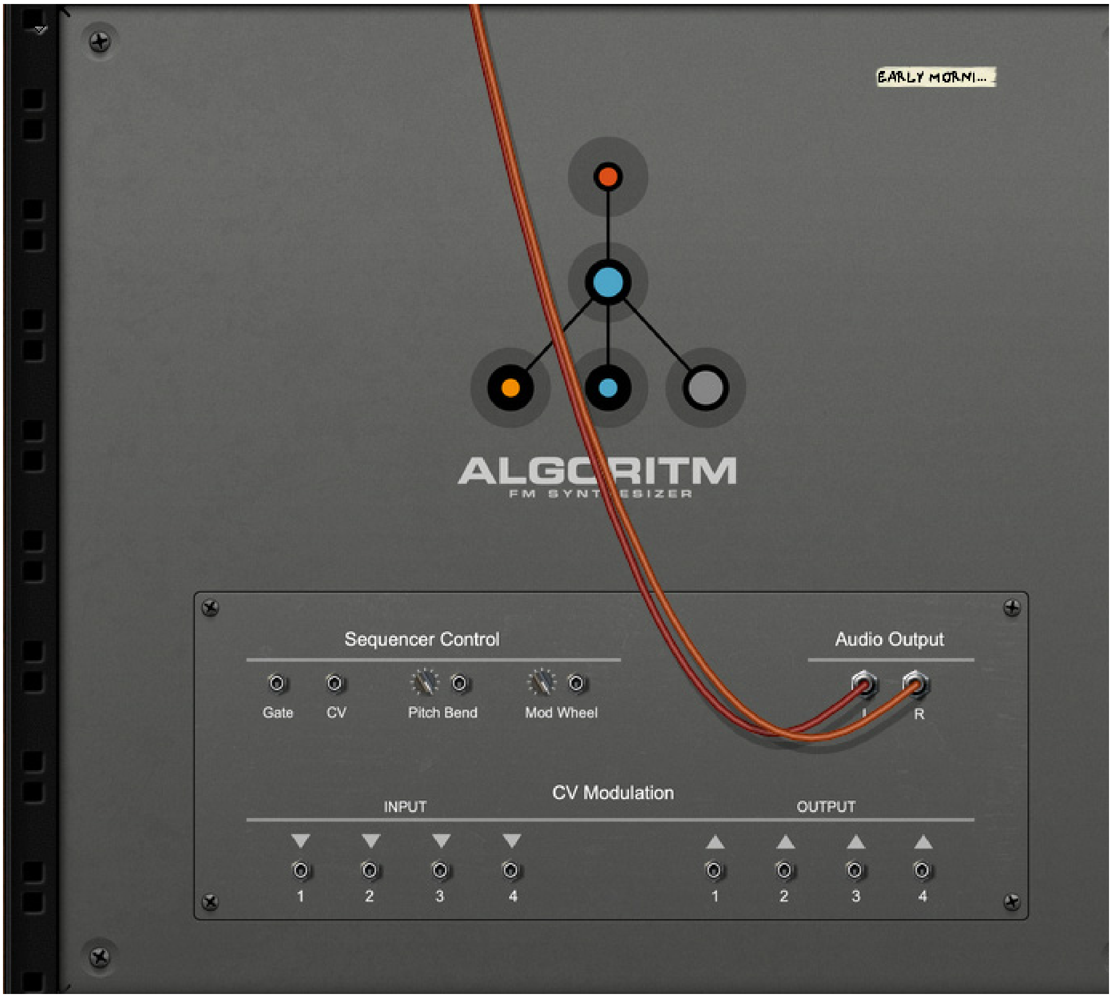
This generates random values for the selected properties.

→ **Drag left or right to set the amount of randomization:**

This morphs between the original settings (left) and the new random values (right). You can play the synth while doing this to check out the results.

3. **When you found a setting you like, release the mouse button.**

Rear panel connections



! **Remember that CV connections are NOT stored in the Algoritm patches! If you want to store CV connections between devices, put them in a Combinator device and save the Combi patch.**

Sequencer Control inputs

The Sequencer Control CV and Gate inputs allow you to play Algoritm from another CV/Gate device (typically a Matrix or an RPG-8). The signal to the CV input controls the note pitch, while the signal to the Gate input delivers note on/off along with velocity. There are also inputs with attenuation knobs for modulating the Pitch Bend and Mod Wheel parameters.

CV Modulation Inputs and Outputs

These control voltage (CV) inputs and outputs can be used for routing external CV signals to and from Algoritm for modulation purposes. The modulation can be configured in the Modulation Matrix (see "Modulation Matrix").

Master Out

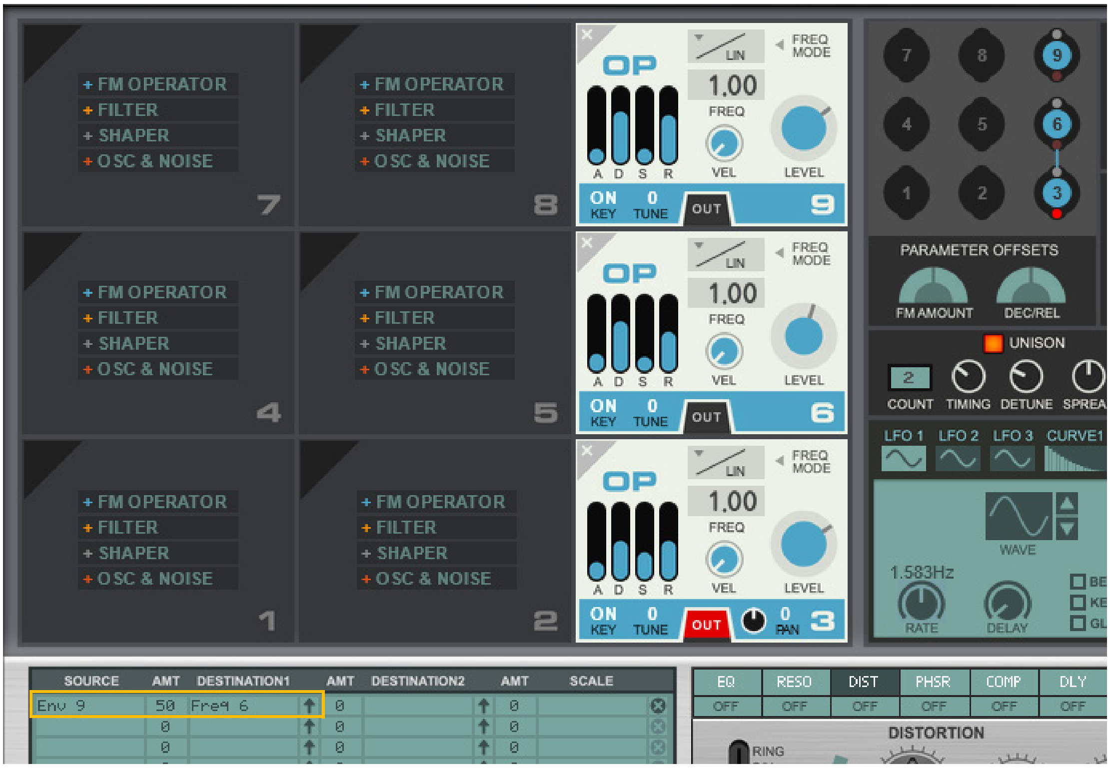
These are the main stereo audio outputs.

Tips & Tricks

Modulating parameters from “unused” operators

Thanks to the flexibility of the Modulation Matrix, you could use non-connected operators (that are not part of the actual sound generation) as modulation sources, to control parameters in your sound.

In the following example we use the Envelope of an unused FM Operator in slot 9 as source for modulating the Freq parameter of the FM Operator in slot 6:

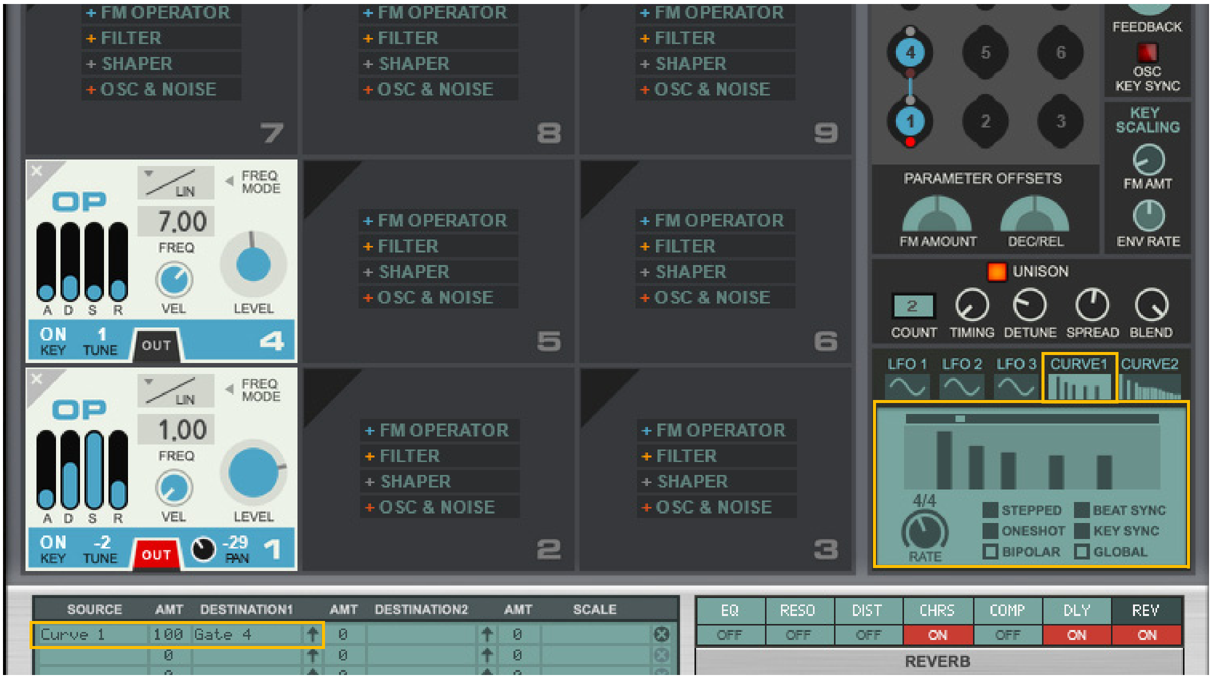


As you can see, the FM Operator in slot 9 doesn’t have any connections to any other operator, but just “sits on its own”.

In the Modulation Matrix we have selected “Env 9” (the Envelope of the operator in slot 9) as Source, and “Freq 6” (the Freq parameter in the operator in slot 6) as Destination. We have also raised the Amount in the Modulation Matrix to hear the effect of the modulation.

Using a Curve for delayed and/or multi-triggered Envelopes

In this example we use Curve 1 in Stepped mode, for gating/triggering an envelope to achieve repeated “delays”.



In the patch above, Operator 4 is frequency-modulating Operator 1. We are using the Curve 1 in Stepped mode, beat-synced to the sequencer tempo, to generate repeated gating of the Envelope of Operator 4. We have also set the Curve to One Shot. As we play a note, the Envelope of Operator 4 is gated five times at 4/4 tempo (once for every “bar” in the curve), creating sort of a “repeated delay” effect.

Since we control the Envelope gate of Operator 4 in the Modulation Matrix, the Envelope doesn’t gate from the played MIDI notes - only from the Curve 1 steps.

Using a Shaper for controlling the FM amount

If you have a patch with an FM Operator that is routed directly to Out, as well as routed for modulating another FM Operator, it can be useful to have separate control over the modulation level. This can be done by placing a Shaper operator between the modulating FM Operator and the FM Operator to be modulated, and use the Shaper purely as an amplifier.

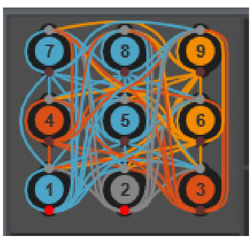
In the patch above, FM Operator 1 is frequency-modulated by FM Operator 3. FM Operator 3 is also routed directly to Out so we can hear its audio as well.

To be able to achieve separate control over the FM modulation amount from FM Operator 3 to FM Operator 1 we have routed the modulation signal via a Shaper. The important thing here is to set the Shaper Mix to “0”. This way the modulation signal through the Shaper won’t be affected by the Shaper distortion algorithm, but throughput completely undistorted.

In the Modulation Matrix we have assigned the Mod Wheel to control the Shaper Level parameter. When we play the sound we can gradually introduce frequency modulation in FM Operator 1 by raising the Mod Wheel.

Creating a rich and full sound (bonus tip for the handy :)

This routing is said to generate a very rich and full sound:



Routing of a rich and full sound.