

R E S O N A N S 

User Guide

Version 1.1.0

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Introduction

Resonans is a physical modelling synthesizer.

In physical modelling synthesis, you have some kind of **physical model** that simulates the properties of strings, sheets and tubes of different materials such as wood, glass or metal. And just like you have to pluck a string or blow into a tube to make them produce a sound, you have to give the physical model some kind of **impulse** to make it resonate. The contour and timbre of the impulse will greatly affect the sound of the physical model, not completely unlike how you can choose whether to pluck a string or scrape it with a bow.

The impulse is a sound source, usually **white noise** which is shaped using a **filter** and an **envelope** before going into the model. In Resonans, you can also use **sampled sounds** or **external audio signals** to excite the model, giving you near endless sonic possibilities.

Physical modelling can produce realistic sounds if you want, but it can also make some really weird noises that still retain a somewhat familiar character.

The Front Panel



The front panel is divided into three rows: The top row contains the controls to alter the sound. The second row consist of different modulation sources that modulate the sound settings over time. The bottom row consists of performance settings, modulation bus and effects.

The audio signal flow in Resonans is as follows:

Impulse → Filter → Model → Mixer → Amp → Reverb → Delay → Limiter → Volume.

Impulse



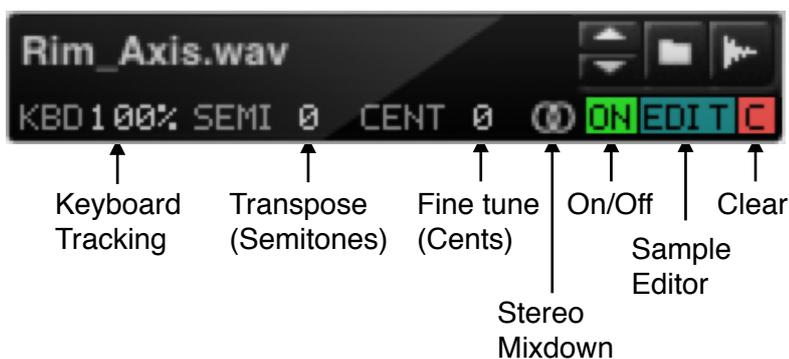
In physics, an impulse is a force that's applied during a certain time interval. You could say that the Impulse section in Resonans applies the necessary force to the physical model to make it produce sounds. The impulse makes use of sound sources such as a white noise generator, a sampled sound, or a live audio signal.

Sampler

The sampler allows you to load or sample any sound you want to use as an impulse signal. You load or sample sounds using the standard mechanisms in Reason, including drag and drop from the browser. When no sample is loaded or the sampler is switched off, Resonans will generate white noise instead.

Typically, samples that contain a lot of harmonic content (such as noise) are best suited for exciting the physical model. Tuned samples with low harmonic content may cause standing waves in the model, resulting in hollow and usually less interesting timbres. We recommend using percussive samples (like rimshots or string plucks) or scraping sounds to get the most interesting results.

There are several settings and controls in the sampler's display:

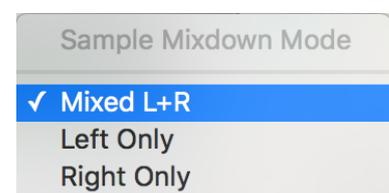


Keyboard Tracking (KBD) controls how much the played note will affect the speed of the sample playback. It ranges from Off to 200%. When set to Off, the pitch of the sample will always be the same, regardless of which key is pressed. At 100%, the sample will track normally across the keyboard, based on the root key configured in the sample editor.

Semi will transpose the pitch of the sample up to +/- 48 semitones.

Cent will fine tune the pitch of the sample in the range +/- 50 cents.

Stereo Mixdown (⊗) affects the way stereo channels are mixed down to mono. (The Impulse section is in mono!) Clicking on the stereo symbol (⊗) will reveal a popup menu with three mixdown options: "Mixed L+R", "Left Only" and "Right Only".



On will turn the sampler on and off. When turned off, the impulse will use white noise instead. This setting can also be automated, or switched via the modulation bus.

Edit will open Reason's sample editor window when clicked. Here you can edit the start, end and loop points of the sample as well as the root key.

Clear will unload the sample from Resonans (but it won't physically delete the sample).

Saving patches that use sampled sounds

In Reason, there are three ways to store a sound sample: As a separate file on your hard disk, inside a ReFill, or embedded in the song file itself. When you record your own sample in Reason, or load a sample with the preference "Self-contain samples when loading from disk" enabled, the sample will be self-contained, which means that it will be stored as part of the song file.

You can't save a Resonans patch if it uses a self-contained sample. If you want to save such a patch, you have to perform the following steps to make sure the sample is made external:

1. In Reason's browser, browse to the folder named "Song Samples", "All Assigned Samples" and the subfolder within that corresponds to your instance of Resonans.
2. Right-click the sample name and choose "Export Sample(s)".
3. Choose a destination folder on your hard disk. We recommend that you put it in the same folder (or in a subfolder to) where you are going to save your patch.
4. Browse to the location where you just saved the sample.
5. Drag the sample onto Resonans in the rack.

You should now be able to save the patch.

Note that if you use a sample from a ReFill or the factory sound bank and edit that sample in some way, for instance by changing the loop points or root key, you won't be able to save the device patch! The edited sample will count as self-contained, but unfortunately you can't export it since samples that originate from any type of ReFill (including the factory sound bank) are protected.



Level

The Level knob adjusts the volume of the impulse signal (sample or noise) before it goes into the filter.

Below it, there's a bipolar knob marked VEL that decides how much the note velocity affects the amplitude of the exciter signal. At its 12 o'clock position (0%), velocity has no effect on the amplitude. At 100% (the fully clockwise position), the velocity will fully control the amplitude of the impulse. Negative amounts (counter-clockwise) will inverse the effect, so higher velocity values will instead result in lower amplitudes.

Link

This button connects Envelope 1 to the Impulse level control. When the button is enabled, you can use the controls in Envelope 1 to shape the amplitude contour of the impulse to your liking, for

example to produce bowed and blown sounds. When the button is disabled, the impulse will be played with a very short envelope that's suitable for plucking the physical model.

The link setting will not affect notes that are already playing, so changing the setting will only affect subsequently played notes.

Note: The Link button is available as a destination in the Modulation Bus, so you can for example use one of the key switches to enable/disable the envelope while playing.

Note: When using a sample as sound source with the Link switched off, Resonans will analyze it to find a suitable start point that produces a transient.

Audio In

Determines the level of the sound coming from the external audio input on the back panel. The external audio is mixed together with the impulse sound before going into the filter. The audio input is completely unaffected by the impulse envelopes.

Detect Transients

This button enables a transient detector on the audio input. Every transient in the incoming audio will trigger a new note in Resonans. For example, you can use this to process drum loops through Resonans without having to trigger notes manually.

Filter



Before the impulse and external audio signals go into the physical model, they are mixed together and passed through a resonant multimode filter. It's a standard 2-pole (12dB/octave) state variable filter with lowpass, bandpass and highpass modes. The Cutoff and Resonance knobs probably don't need further introduction.

Below the Cutoff knob there's a knob that sets how much the filter cutoff will track the keyboard, from 0 - 100%.

Physical Model

This is the heart of Resonans, where most of the sound's character is defined. There are three different models to choose from in the current version of Resonans: **Modal Resonator**, **Inharmonic String** and **Frequency Modulation (FM)**. Clicking on the button above the panel will reveal a popup menu where you can choose the model.



- The **Modal Resonator** passes the impulse signal through up to 60 highly resonant bandpass filters per voice, creating partials whose amplitude relationships are constantly being modulated.
- The **Inharmonic String** model uses an extended version of the Karplus-Strong algorithm to simulate vibrations and wave propagations in a string using carefully tuned delay lines.
- The **Frequency Modulation** model does sine FM synthesis. The Brightness control is mapped to the modulation amount (or FM index), the Structure controls the frequency ratio, Damping controls the decay of the internal envelope, and Position controls the feedback amount. Apart from the model's own built-in envelope, the amplitude is also affected by the Impulse via an envelope follower. In this mode, the Filter section goes after the model.
- You can also choose **None**, which means that you will bypass the physical modelling section altogether. While it might not be that great for sound design, it's a useful tool for fine tuning your impulse signal.

Note that the Modal Resonator uses a significant amount of CPU, especially at long decays and when playing notes in the lower register (since more partials will be generated to fill up the audible spectrum). We recommend setting the Polyphony as low as possible when using this model.

Structure

The Structure knob affects the harmonic structure of the sound. The higher the setting, the more dissonant and complex the sound becomes. At lower settings it may sound more like a regular string, and gradually morphs into what sounds more like glass or metal.

Many settings will result in unstable tuning and/or pitch tracking. The most stable pitches are found when Structure is set to 25. This is the default value (that can be set by Cmd/Ctrl-clicking on the knob) and is also marked on the front panel.

At the more dissonant settings, we recommend bringing in the sub oscillator (described in the Pitch section) to help strengthen the fundamental frequency of the sound.

Brightness

Controls the brightness of the sound, using different techniques depending on the synthesis model. At settings below noon (50.0), additional low pass filtering is applied to the impulse signal, subduing it further. At higher settings, frequencies and resonances in the physical models will be affected, resulting in a brighter sound, and it may also affect the decay to a certain extent.

Below the Brightness knob there's a bipolar knob marked **Velocity** that sets the influence of note velocity on the brightness setting, ranging from -100% to 100%. When set to noon (0%), the velocity will have no effect on the brightness. If set to a positive value, it means that higher velocities will make the sound brighter, and if set to a negative value, higher velocities will make the sound duller.

Damping

Controls the decay of the sound. The higher the setting, the longer the decay.

Below the Damping knob there's a bipolar knob that sets the influence of the **Sustain Pedal** on the damping setting, ranging from -100% to 100%. At positive values, the sustain pedal will cause the sound to sustain. At negative values, it will act as a damping pedal instead.

Continuous sustain values are supported.

Position

This decides where on the material/string the excitation is applied, resulting in slightly different harmonic content and stereo images. If you move the position setting around it sounds kind of similar to a PWM control on an analog synthesizer.

Below the Position knob there's a knob marked **Random** that determines the randomness of the position (offset from the knob setting) for each new note, from 0 to 100%. You can use this to make every note sound a little different, giving a slightly more organic quality to your patches.

Pitch



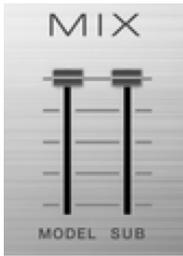
The Pitch section has controls for offsetting the pitch of Resonans. The **Oct** (± 4), **Semi** (± 12) and **Cent** (± 50) knobs will offset the pitch of both the physical model and the sub oscillator.

The **Sub Octave** switch sets the octave offset (0, -1, -2) for the built-in sub oscillator. The sub oscillator is a sine wave oscillator that adds more body to the sound. At 0 octave offset, the pitch will correspond to the fundamental frequency of the physical model. At the lower octave settings, more low frequency content is added to the sound.

The sub oscillator will always track the pitch and amplitude of the physical model.

Note that the pitch controls will not affect the sampler, which has its own offset controls.

Mixer



The mixer controls set the balance between the physical model and the sub oscillator.

Amp



Gain

Attenuates the overall output signal of each voice. Beside the Gain knob there's a **Link** button that enables control over the gain via Envelope 3, allowing you to tame the amplitude shape of the sound. When the link is disabled, the gain is fixed at the knob setting. When the link is enabled, the knob sets the maximum gain, scaling the envelope signal.

Spread

Sets the amount of stereo spread of the signal from 0% (mono) to 100% (channels panned hard left and right). Sometimes the left and right outputs from the physical model can sound quite different from each other. In those cases, a more moderate Spread setting can help the stereo image sound more coherent.

Pan

Sets the stereo balance of the voice. Pan can be modulated per-voice using for example the note number.

Loudness and gain reduction meters

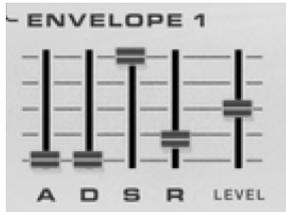
Above the amp section, inside the patch browser display, there is a loudness meter that indicates the overall loudness. The loudness is measured after the effects section and output limiter, but before the master volume knob is applied. There's also a gain reduction meter that indicates how



hard the output limiter is being pushed. When the gain reduction meter appears, it means that you may be introducing additional distortion (soft clipping) to the signal that you may or may not want.

The meters in conjunction with the Amp Gain knob give you more precise control over how the limiter is used in your patch. If you're hearing unwanted distortion, you can look at the gain reduction meter and lower the level of the Amp Gain accordingly. To compensate for any loss of loudness you can turn up the Volume knob instead.

Envelope 1

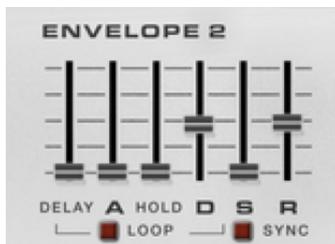


Envelope 1 is a regular ADSR envelope with an additional Level control. It can be linked to the Impulse section, and it can also be freely routed in the Modulation Bus.

The Attack, Decay and Release time all range from 0 to 10 seconds.

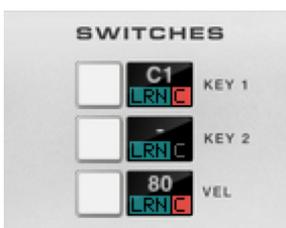
The Level control only affects the envelope signal that goes into the Impulse section. The purpose is to fine tune the level balance between plucked and bowed sounds when toggling the link while playing.

Envelope 2



A slightly more advanced DAHDSR envelope, with controls for Delay, Attack, Hold, Decay, Sustain and Release. This envelope can be tempo synced by enabling the Sync button. It can also loop the Delay-Attack-Hold-Decay stages by enabling the Loop button. The signal from this envelope is only available via the Modulation Bus section.

Switches



Resonans has three switches that can be freely assigned in the Modulation Bus. They are two key switches and one velocity switch. Each has a manual trigger button as well as a display with Learn and Clear buttons.

Key Switches 1 and 2

A key switch will be active while the configured key is being pressed. By default, the key switches are global modulation sources that affect all voices simultaneously. You can change this behaviour by using the latched modulation sources instead, see the Modulation Bus section for more details.

Velocity Switch

The velocity switch will be active if the current note's velocity has the same or higher value than the configured velocity threshold. The velocity switch is a per-voice modulation source.

Using the MIDI Learn function

The easiest way to configure a switch is by using its LRN button. The LRN button can be used in one of the two following ways:

- Hold the MIDI key you want to learn and then click the LRN button
- Or hold the LRN button, and then press the desired MIDI key

When you release the LRN button, the new setting is saved.

You can also configure the switch manually by clicking on the value in the display and dragging up and down. You can set more precise values by holding Shift while dragging. If you hold Cmd (Mac) or Ctrl (Windows) while clicking, the default value for the switch will be set.

To disable a switch altogether, click on its red C button to clear it.

LFO



The LFO, or Low Frequency Oscillator, is a freely assignable modulation source in the Modulation Bus section. It is global and has two outputs: The "regular" output and a 90 degrees phase shifted version of the same waveform.

Waveform display

Shows the currently selected waveform. Available waveforms are **Sine**, **Triangle**, **Square**, **Sawtooth** and **Random**. You can select waveform in two ways: You can click on the name to reveal a popup menu in which you select the waveform. Or you can click on the waveform itself and drag up or down to browse the waveforms. Holding Shift while dragging will give finer control. Holding Cmd (Mac) or Ctrl (Windows) while clicking anywhere in the display will reset it to the default waveform, which is Sine.

Phase

This knob controls the phase offset of the LFO, from 0 - 360 degrees. You will see the effects of the phase offset in the waveform display.

Lag

This knob adds lag to the LFO signal, which will make the curve smoother. The effect will mostly be noticeable on more jagged waveforms such as square. You can see the effects of the lag setting in the waveform display.

Rate

Sets the rate, or frequency, of the LFO. It ranges between 0.05 Hz (20 seconds per cycle) and 100 Hz (100 cycles per second), or between 4 bars to 1/32 note in tempo synced mode.

Sync

Enables/disables tempo sync mode for the LFO. When Sync is enabled, Retrig is disabled and Reason's main transport is running, the LFO will be phase locked to the song position according to the Phase setting.

Retrig

Decides whether the LFO should restart each time a new note is triggered or whether it should be free running.

Vibrato



The vibrato is a specialized LFO that outputs a sine wave between 0.1 and 10 Hz. It can also be tempo synced. The vibrato is per-voice and always restarts when the note is triggered.

Depth

Sets how much the vibrato affects the pitch. The range is 0.0 - 2.0 semitones.

MW

Sets how much the modulation wheel affects the vibrato depth. The MW amount is added on top of the Depth setting.

Rate

Sets the vibrato rate (frequency) from 0.1 - 10 Hz, or 4 bars - 1/32 note in tempo synced mode.

Delay

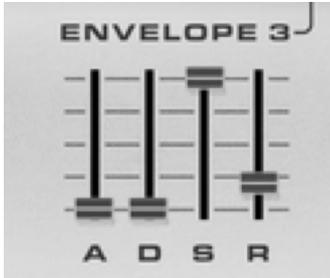
Delays the start of the vibrato, from 0 - 5 seconds, or 0 - 4 bars in tempo synced mode. At the end of the delay the vibrato will be quickly faded in. When the key is released, the vibrato starts slowly fading out.

If the key is released during the delay phase, the vibrato will not be activated.

Sync

Enables/disables tempo sync for the Delay and Rate knobs.

Envelope 3



Envelope 3 is a regular ADSR envelope that can be used as a freely assignable source in the Modulation Bus. It is also prerouted to the Amp Gain using the Link button.

Performance



Here you find settings for pitch bend, polyphony, legato and portamento.

Pitch Bend Range

Sets the range of the pitch wheel, from 0 - 24 semitones. By setting the range to 0, you can use the pitch wheel as a freely assignable source in the modulation bus without simultaneously affecting the pitch.

Polyphony

Sets the number of voices, from 1 - 16. When set to 1 voice, the synth is monophonic and will handle multiple key presses like a mono synth. For example, you can play trills by holding down one key and rapidly pressing and releasing another key multiple times.

Reducing the polyphony to its absolute minimum is a good way to reduce the CPU load when using the Modal Resonator.

Legato

Sets the behavior when pressing more than one key at a time. When legato is disabled, a new note will always be triggered when pressing a key. When enabled, pressing a new key while another is already held will alter the pitch of the existing note.

Legato works in both polyphonic and monophonic modes. In polyphonic mode, the synth will essentially become monophonic (since you can only trigger one note at a time), but the Polyphony

setting will determine how many notes can ring out at the same time. So you can still “strum” chords, for example.

Note: The legato mode is available as a destination in the Modulation Bus, so you can for example use a Key Switch or Velocity Switch to turn legato on and off while playing.

Portamento

The Portamento knob sets the pitch glide time between notes, from 0 to 10 seconds. Portamento works both in polyphonic and monophonic modes.

There are three portamento modes: Off, Auto and On. In Auto mode, portamento will only be active when holding down at least two keys simultaneously.

Reverb



Active

Switches the reverb on and off.

Predelay (PRE)

Sets how long the audio will be delayed before sending it into the reverb, from 0 - 250 ms.

Decay

Affects for how long the reverb will ring. It goes from rather short up to near-infinite sustain.

LPF Cutoff

Cuts higher frequencies inside the reverb’s feedback loop. The higher the setting, the brighter the sound. This setting also affects the decay time of the reverb.

Tilt

The tilt controls a filter that is applied to the wet reverb signal. With the knob at noon (0), the reverb signal is unaffected. At positive settings, the reverb goes through a highpass filter, gradually increasing the cutoff frequency, resulting in a more “shimmering” reverb. At negative settings, the reverb signal is sent through a lowpass filter, gradually decreasing the cutoff frequency, resulting in a more subdued reverb.

Mix

Sets the mix between the dry and wet signals.

Delay



This stereo delay is connected in series after the Reverb.

Active

Switches the delay on and off.

Time

Sets the time delay time, from 0 - 1.0 seconds, or up to 4/4 in tempo synced mode. Changing the time control while a sound is going through the delay will alter the speed and pitch of the delayed signal.

Sync

Enabled tempo sync for the delay.

Feedback (FEED)

Sets how much of the wet delay signal is fed back into itself. At higher settings, more echoes are produced. When set to 0, only one echo will occur.

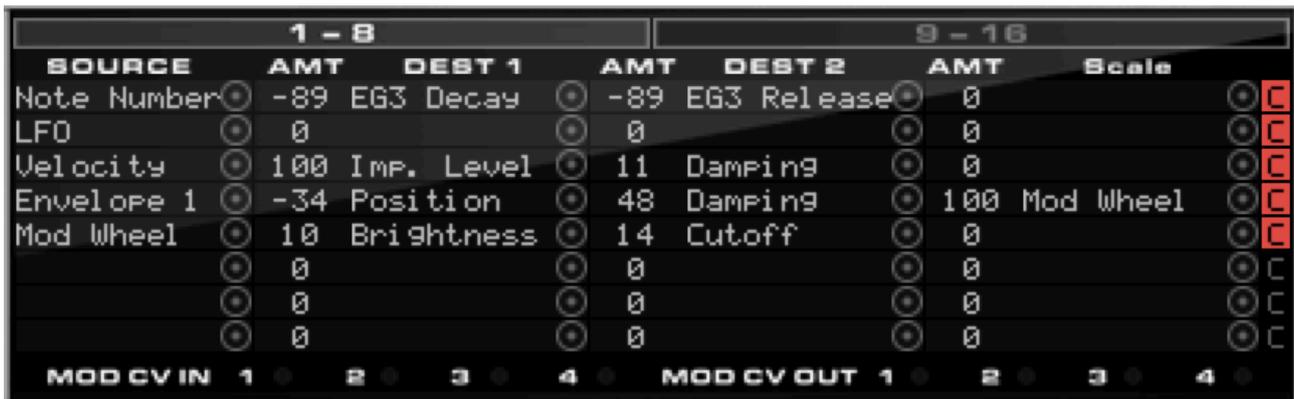
Tilt

This is very similar to the Tilt control on the Reverb in that it uses the exact same filter configuration. But there's one important difference: In the Delay, the filter is inside the feedback loop of the delay itself, resulting in a more heavily filtered sound for each echo.

Mix

Sets the mix between the dry and wet signals

Modulation Bus



The Modulation Bus is your standard **Source** → **Destination 1** → **Destination 2** → **Scale** type modulation bus. It has 2 pages with 8 rows on each page, i.e. 16 rows in total. For each row you can select a modulation source, route it to up to two destinations with separate amount controls (-100 to 100), and finally scale the total amount of modulation using another modulation source.

The modulation bus supports drag and drop routing for each source and destination slot, and it also has buttons to clear entire rows of settings.

At the bottom of the modulation bus there are indicator LEDs for the freely assignable modulation CV inputs and outputs.

Modulation sources

Many modulation sources can be selected using drag and drop. Click on the target symbol next to the modulation slot and drag the mouse pointer to the modulation source on the front panel. The source display will show the currently selected modulation source. Holding **Shift** or **Alt** may reveal an alternative version of the modulation source.

All modulation sources can be selected by clicking in the source display and selecting the source from the popup menu.

Source	Description
Envelope 1, 2 and 3	Output signals from each of the three envelopes. The level setting of Envelope 1 is ignored. Per-voice and unipolar. <i>Drag anywhere inside the corresponding Envelope section on the front panel.</i>
LFO and LFO Shifted	The output signals from the LFO, normal and shifted 90 degrees. Global and bipolar. <i>Drag anywhere inside the LFO section. Hold Shift or Alt to select the shifted version.</i>
Vibrato	The output from the vibrato LFO. Per-voice and bipolar. <i>Drag anywhere inside the Vibrato section.</i>
Key Switch 1 and 2, Normal and Latched versions	Output signals from each of the two key switches. When the key switch is active the output is 1, otherwise it is 0. The regular versions (K Switch #) are global. The latched versions (KS Latched #) are per-voice. <i>Drag anywhere inside each Key Switch. Hold Shift or Alt to select the latched versions.</i>

Source	Description
Velocity Switch	The output signal from the velocity switch. If the note velocity reaches or exceeds the velocity threshold, the switch outputs 1, otherwise it outputs 0. Per-voice. <i>Drag anywhere inside the Velocity Switch.</i>
Note Number	The MIDI note number of the current note scaled between -1 and +1. Per-voice and bipolar.
Velocity	The velocity value of the current note. Per-voice and unipolar.
Polyphony	Produces a value between 0 and 1 depending on the number of held notes at the time the note is pressed. Per-voice and unipolar.
Mod Wheel, Normal and Latched	The current mod wheel value scaled between 0 and 1. Global and unipolar. The latched version is per-voice and will sample and hold the position of the mod wheel when the voice is triggered. <i>Drag to the modulation wheel. Hold Shift or Alt for the latched version.</i>
Pitch Bend	The current pitch bend value scaled between -1 and +1. Global and bipolar. <i>Drag to the pitch wheel.</i>
Aftertouch, Breath Ctrl and Expression	The value of performance MIDI CC's scaled between 0 and 1. Global and unipolar.
Sustain Pedal	The value of the Sustain pedal MIDI CC, scaled between 0 and 1, so continuous values from the pedal can be processed. Global and unipolar.
CV Input 1 - 4, Normal and latched	The current value of the corresponding CV input. The normal versions are global and bipolar. The latched versions are per-voice, and will sample and hold the CV input when the voice is triggered. <i>Drag to the corresponding CV input below the modulation bus display. Hold Shift/Alt for the latched versions.</i>
Random	A random value between -1 and 1 generated on each note on. Per-voice and bipolar.
Noise	White noise. Global and bipolar.
Envelope Follower	The output signal from an envelope follower that's connected to the output from the physical model. Generates a unipolar signal between 0 and 1. Per-voice. <i>Drag anywhere inside the physical model section.</i>
Constant	A constant value of 1. Useful as a scaling source if you want more precision when setting a modulation amount.

Modulation destinations

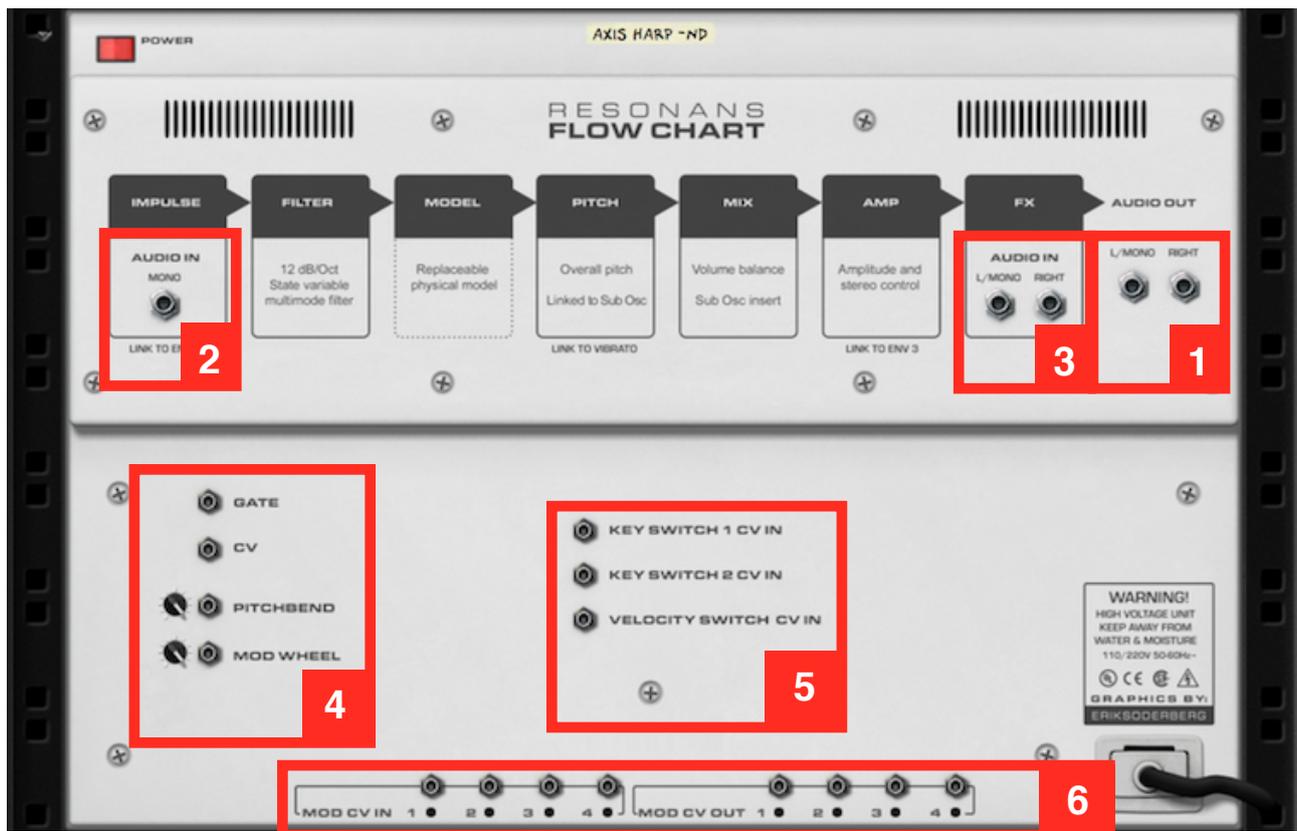
All modulation destinations can be selected by dragging to the related control on the front panel. Just click on the target symbol for the destination slot you want to assign, and then drag the mouse pointer to the control you want to modulate.

You can also select a destination from the popup menu by clicking in the text field.

Destination	Description
Envelope 1 to Impulse	Controls the link between Envelope 1 and the Impulse Level. Positive modulation values (above 50%) will turn it on, and negative modulation values (below 50%) will turn it off.
Impulse Level and Audio Input Level	Affects the impulse and audio input levels respectively.
Sampler Pitch	Will modulate the pitch on the sampled sound ± 48 semitones (± 4 octaves). <i>Drag to the area with the KBD, SEMI and CENT settings in the sample display.</i>
Sampler On	Will turn the sampler on or off. <i>Drag to the ON button in the sample display.</i>
Cutoff and Resonance	Affects the filter settings.
Structure, Brightness, Damping and Position	Affects the corresponding settings in the physical model.
Octave	Modulates the octave control to its min and max amounts.
Sub Octave	Modulates the octave offset of the sub oscillator.
Pitch	Modulates the pitch in the range of ± 48 semitones (± 4 octaves). You can push the modulation even further if you add together multiple modulation sources.
Model and Sub Osc levels.	Affects the levels of the physical model and sub oscillator in the mixer.
Amp Gain, Spread and Pan	Affect the amp settings.
Envelope 3 to Amp	Controls the link between Envelope 3 and the Amp Gain in a similar way to Envelope 1 and Impulse Level.
Envelope 1 settings	Affects all the fader settings for Envelope 1.
Envelope 2 settings	Affects all the fader settings for Envelope 2.
LFO Rate, Phase and Lag	Affects the corresponding LFO settings.
Vibrato Depth, Rate and Delay	Affect the corresponding Vibrato settings.
Envelope 3 settings	Affects all the settings for Envelope 3
Legato	Lets you toggle the Legato setting in a similar way to the link buttons. Since this is a global setting, it will be affected by the modulation sources of the last triggered voice.
Portamento	Affects the portamento time.

Destination	Description
Reverb settings	Affects all the Reverb settings, including the On/Off switch. Since these are global destinations, they will be affected by the modulation sources of the last triggered voice.
Delay settings	Affects all the Delay settings, including the On/Off switch. Since these are global destinations, they will be affected by the modulation sources of the last triggered voice.
CV Output 1 - 4	<p data-bbox="480 461 1434 555">Routes the modulation source to one of the CV outputs. Since these are global destinations, they will be affected by the modulation source of the last triggered voice.</p> <p data-bbox="480 584 1353 651"><i>Drag to the corresponding CV output at the bottom of the modulation bus display.</i></p>

The Rear Panel



1. Audio Out

Stereo out connection that's automatically routed to your mixer. If you only connect the left channel, it will mix down the output of Resonans to mono.

2. Impulse Audio In

Here you can connect a live audio source to run through the entire signal chain of Resonans. You can control the level of the input signal via the **Audio In** knob on the front panel.

3. FX Audio In

Audio signals connected to this stereo input pair will be processed by the Reverb and Delay in Resonans. If you only connect something to the left input it will be treated as a mono signal.

4. Sequencer Control Inputs

CV inputs for Gate and Note CV, Pitch bend and Mod wheel amounts.

5. Switch CV Inputs

Positive CV pulses on these inputs will trigger the corresponding switch. For example you can use it together with a sequencer device like Euclid Rhythm Generator or Elementary Logic Gates.

5. Modulation CV Inputs/Outputs

Four CV inputs and four CV outputs that are freely assignable in the modulation bus section. The LEDs indicate the signal activity on each socket.

Credits

Concept, design and programming: Niklas Backlund

Graphic design: Erik Söderberg

Early feedback and prototype evaluation: Simon Bader

Resonans is heavily based on code written by Olivier Gillet (see third party licensing information below)

Patch designers

- Navi Retlav (-NR)
- ArxPro (-AP)
- Odarmonix (-ND)
- Benedict Roff-Marsh (-BRM)
- Kristoffer Wallman (-KW)

Beta testers

Simon Bader, Koshdukai, Nicholas Curry, Challism, Tiker01, Benedict Roff-Marsh, Faastwalker, Navi Retlav, ArxPro, Odarmonix, Sabastian Weaver, Erik Söderberg, André Horst

Third Party Licenses

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