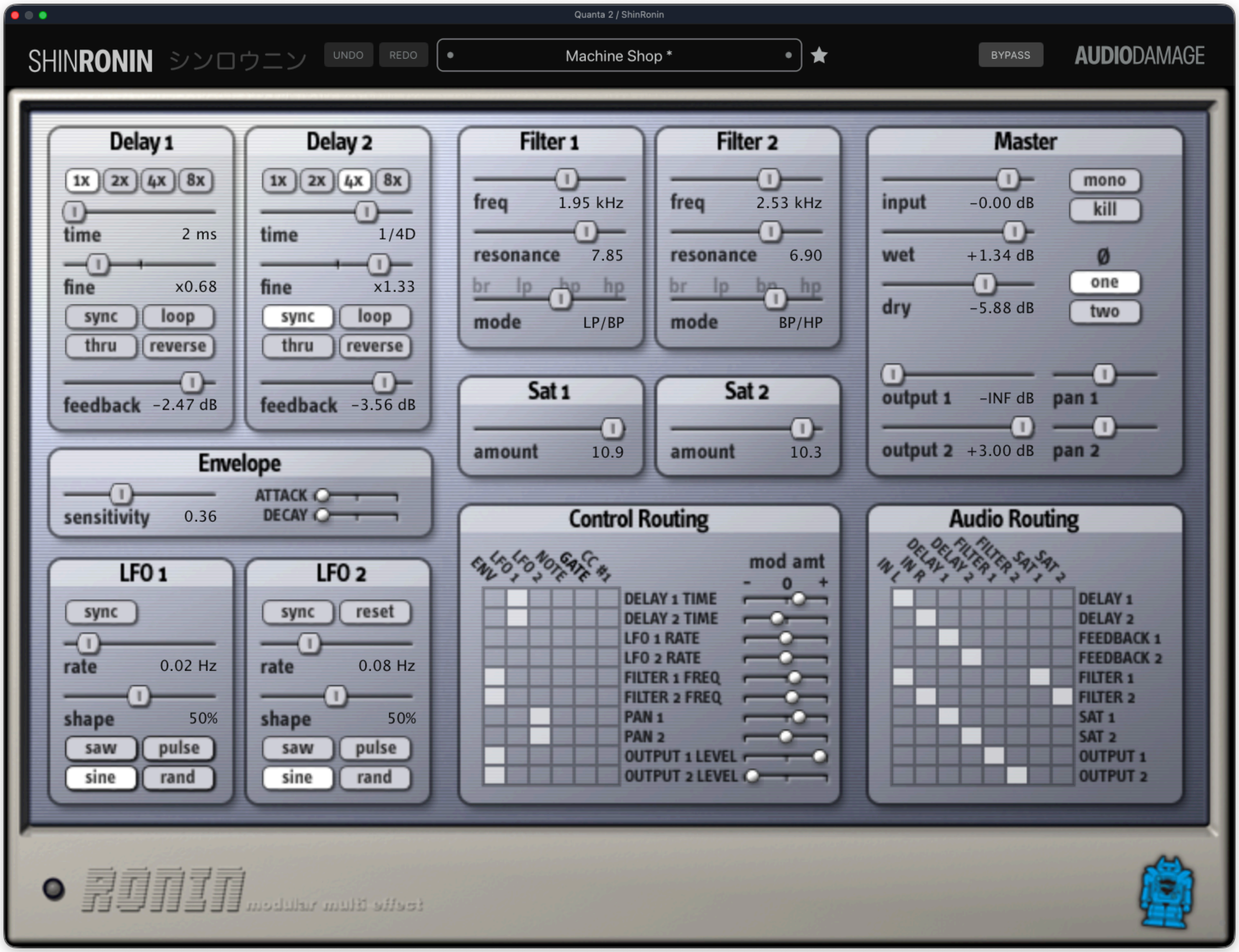
Shin-Ronin Manual

Audio Damage, Inc.

Release 1.0



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Credits

Software Design and Construction, Documentation

Chris Randall
Adam Schabtach

Field Testing

Jens Ågren
Wade Alin
Carl Downing
Mike Fisher
Chris Hahn
Steve Hamman
Jeff Laity
Christian Puffer

Made Possible By

Elle
Tracie

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System Requirements

The following table summarizes the operating system requirements and formats provided for Shin-Ronin. Shin-Ronin is a 64-bit plugin.

Operating System	**Minimum Version**	**Formats**
macOS	10.13	AAX, AudioUnit, CLAP, VST3; Intel and Apple Silicon
Windows	10.0 version 1607	AAX, CLAP, VST3
Ubuntu	18.0	CLAP, LV2, VST3
iOS (separate purchase)	iOS 12	AUv3

Historical Retrospection

Ronin made its debut over 20 years ago in 2005. It was an ambitous product for its time, and for a young company. We aimed high, putting in just about every idea we had for a delay plugin. Ronin was possibly the first commercial software product to recreate the magical connection between pitch shift and time change inherent in older hardware delays. While it was capable of runaway feedback, we can't say that it was a runaway commercial success. Maybe we were too small to get much attention at the time, maybe the product itself was a little too complex. We reused its core delay code in Dubstation, a much simpler but easier to use delay plugin which went on to be one of our most popular products. Eventually, with regret, we put Ronin out to pasture.

But we never really forgot about Ronin. It was, after all, something we built for the sheer joy of it–for pushing the boundaries of what could be done in a plugin, and what we could do.

Recently, some of our long-time friends started asking about Ronin on our Discord server. With their encouragement, we pulled Ronin out of the archives and rebuilt it with contemporary software frameworks and formats. As much as possible, we left it unchanged. It amused us to notice that its sound has a vintage quality, a vibe harkening back to the early days of the dawn of DAW-based in-the-box music creation of the early 2000s. We named the plugin Shin-Ronin, *shin* being the Japanese word for new. Shin-Ronin presents Ronin in a new incarnation, preserving its original character and sound, warts and all.

In light of its warts and its throwback nature, we're giving it away for free.

The remainder of this manual is largely unchanged from the last Ronin manual we published, back in 2012, with some sections updated as needed.

Introduction

Thank you for downloading Ronin, Audio Damage's modular multi-effects plug-in. While Ronin was built primarily for delay-based effects and looping, its modular architecture and complement of filters and low-frequency oscillators allow it to create a wide range of sounds.

This manual provides a brief overview of Ronin, followed by detailed descriptions of all of its modules. You're entirely welcome to plunge ahead, install Ronin, and flip through its presets without reading any further. However, to make full use of Ronin's extensive capabilities, you will find it helpful to return to this document for a thorough reading. This manual provides detailed descriptions of all of the modules, information about using MIDI hardware to control Ronin, and in-depth explanations of four example presets to illustrate how Ronin patches are created and used.

Warning

Ronin is a powerful and flexible plug-in which allows you to connect signal-processing modules together in almost endless ways. There is nothing that will stop you from creating feedback loops, howling oscillations, and badly distorted signals. This power comes with a price: Ronin can generate extremely loud sounds, suddenly and without warning. Please be cautious and use moderate volume levels with your speakers or headphones. As with any new audio toy, you should be particularly careful when you are first learning to use Ronin.

Throughout this manual, we use this symbol to call attention to feedback configurations that are particularly likely to create loud sounds. Remember, though, that the health and safety of your ears (and your speakers) is your responsibility.

Installation

Shin-Ronin uses our custom plugin manager application for installation. Launch it as usual on your operating system of choice and you'll be presented with a window like this:

Near the top of the window, beneath the name of the plugin, you'll see the version number of the software carried by the installer. This is distinct from the version of the plugin manager itself, which is shown in the upper right and usually not of much interest. (Note that the version numbers in this screenshot are for demonstration purposes only and may not reflect the actual version numbers of the software at the time you read this manual.)

Under the heading FORMATS are large buttons corresponding to the plugin formats which can be installed: AAX, AU, CLAP, LV2 and/or VST3, depending on the operating system. If the plugin is already present on your system in one or more formats (i.e. if you're upgrading from a previous version), the corresponding button is drawn in blue. When possible, the version number of the existing plugin is also shown. If the installer contains a newer version of the plugin than the one(s) already installed, the button is drawn in orange to alert you to the fact that an update is available. If a format is not present on your system, its button is drawn in gray.

Click a button to select the format for installation. A yellow outline appears around the button to indicate that its format will be installed. In the above screenshot, VST3 and CLAP are installed, AAX and AudioUnit are not installed, VST3 is installed but older than the version contained in the installer, and the AudioUnit is selected for installation. Clicking a button a second time removes the yellow outline, and the corresponding format will not be installed. Clicking the SELECT ALL button selects all available formats for installation.

No changes to your system's storage device take place until you click the INSTALL button near the lower-right corner of the window. Click that button and you'll receive visual confirmation that the format(s) you've selected have been installed. (Yes, it happens quickly. Our products are not encumbered by any DRM or anti-piracy baggage and hence install more quickly than many others.) On Windows and Linux, if you hold down the Shift key on your keyboard, the INSTALL button's label switches to UNINSTALL, and clicking it will remove the selected formats from your system[1]. Once you're installed and/or removed the formats you need, simply close the application in the usual manner for your operating system. You're done. There is no license code or other authorization necessary; we'd rather assume we can trust you than burden you with an onerous DRM system.

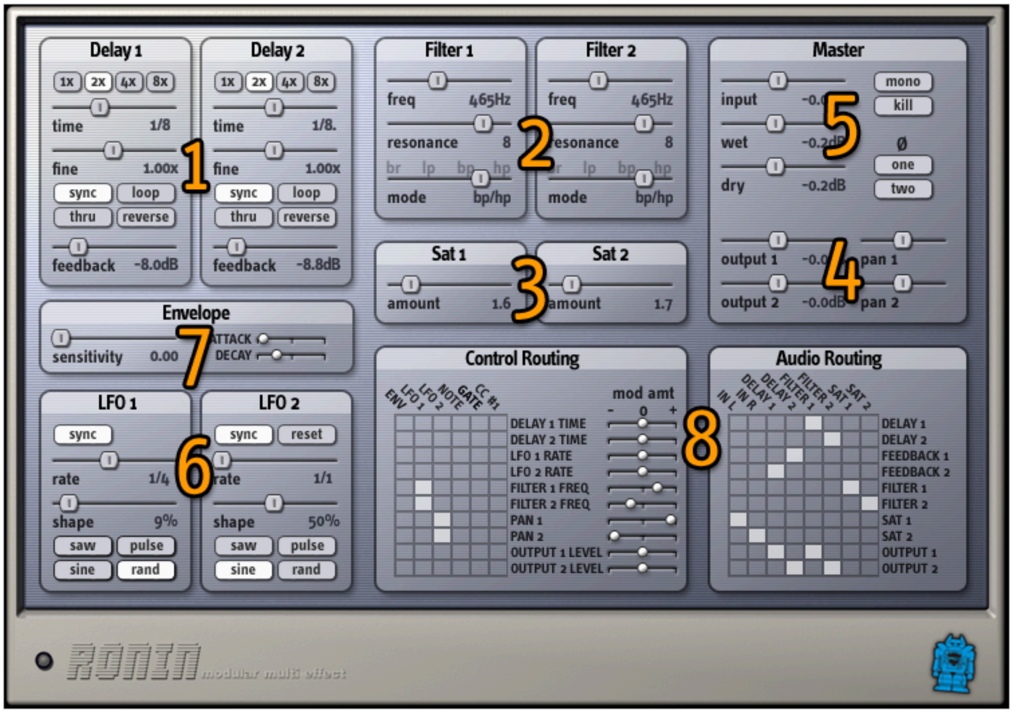
You'll find some handy buttons under the EXTRAS heading, all of which are pretty self-explanatory:

- EULA: presents the End-User License Agreement for our products. By clicking the INSTALL button you're implicitly agreeing to these terms, but we expect that you'll find them reasonable should you take the time to read them.
- MANUAL: opens the current version of this user manual, in PDF form, in your web browser.
- WEBSITE: opens the product's web page in your browser.
- SUPPORT: displays information for contacting us, either via our Discord presence or through email.
- SETTINGS: produces a panel in which you can choose the installation locations for each plugin format. Note that these locations are nominally stipulated by the publishers of the formats and changing them might cause problems for some hosts.
- CHANGES: displays a summary of what we've added, changed or improved since the last version of the product.

[1] Blame Apple, not us, for the lack of this feature on macOS. On macOS just manually delete the plugin(s) from your plugin folder(s).

Overview

Here is a picture of Ronin's editor window, followed by brief descriptions of its modules and their controls:



- Delays — two delay modules, each up to 12 seconds long, with coarse and fine time controls, looping, reverse playback, time sync to your host's current tempo, and bypass switches.
- Filters — two resonant multi-mode filters, with morphing between low-pass, band-pass, high-pass, and notch frequency-response modes.
- Saturators — two saturation stages for creating tube-like distortion.
- Output and Panning — two sets of output level and stereo panning controls.
- Master Level Controls — gain controls for the input signal and the processed and unprocessed output signals, switches to mix the input signals to mono and to silence the input signal altogether.
- LFOs — two low-frequency oscillators for modulating the signal-processing modules, or each other, with four basic wave shapes, a shape control which warps the wave shapes in different ways, and a sync button which causes the LFO to follow the tempo of your host.
- Envelope Follower — a modulator that measures the loudness of the incoming sound and generates a modulation signal for creating effects such as automatic filter sweeps, triggered flanging, and ducked delays.
- Routing Matrix — two sets of switching matrixes for connecting the inputs and outputs of all signal-processing modules, and for connecting the modulators to parameters of the signal-processing modules.

Each of these modules operates completely independently. For example, you can use one delay module looping several seconds of audio while using the second delay module to add a slapback echo.

Routing Matrix

Ronin has two arrays of switches, together called the routing matrix, for making connections between audio modules and modulators. There are two arrays, one for connecting modulators to the audio modules that they control, and one for connecting audio modules to each other. This is because Ronin treats audio data and modulation data differently. You cannot use audio data as a source of modulation data, and you cannot listen to modulation data.

The switch arrays allow the audio modules to be connected to each other in any order or combination. Signals from one module can be connected to several other modules, and the signals from several modules can be mixed together and fed to one or more other modules.

The audio routing matrix is the most dangerous part of Ronin. Because you may patch the audio inputs or outputs of any module to any other, you can create signal chains that would not be useful under normal circumstances, such as routing filters back into themselves, a practice that will almost certainly result in banshee wails of feedback.

Amount Controls

Each parameter in Ronin that can be modulated (that is, controlled by a modulator) has a small slider next to its name in the control routing matrix. This slider is the modulation amount control. It determines how much the modulator affects the parameter. Thus you can make a modulator affect an audio module only slightly, such as to create a subtle vibrato by modulating the delay time of one of the delay lines. Or a modulator can vary a parameter over a wide range, such as creating dramatic timbral sweeps by moving the frequency of a filter up and down.

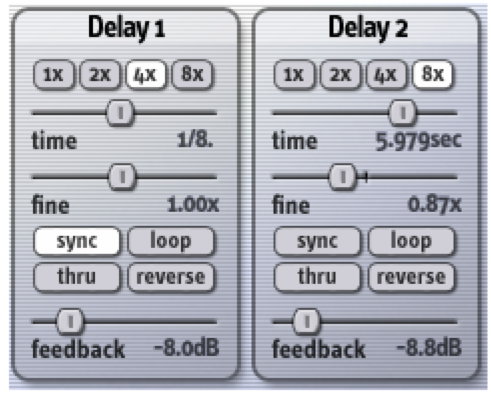
The amount controls are also bidirectional, in that moving them to the right makes the modulator increase the value of the parameter it controls, and moving them to the left makes the modulator decrease the value of the controlled parameter. This affects different parameters in different ways; the detailed descriptions of each module found later in this manual explain specifically how their amount controls work.

Audio Modules

Ronin is made up of several independent audio-processing modules. Like a modular synthesizer, the connections between these modules are not permanently fixed. You can connect the modules in any order you desire. This flexibility presents a wide range of signal-processing configurations that aren't available with other delay-based plug-ins. The connections between modules are made with a switching matrix similar to the ones found on EMS analogue synthesizers.

Delays

The two delay modules in Ronin are the main audio processors. The delay modules accurately emulate the behavior of older digital delays and tape delays, recreating their characteristic interdependence of pitch and time. If you lengthen the delay time while an audio signal re-circulates in the delay, you will hear its pitch drop. If you shorten the delay time, the pitch goes up. Most contemporary digital delays—both hardware and software—do not exhibit this behavior and cannot create the range of weird and wonderful sounds of their older counterparts. On the other hand, older digital delays usually had limited dynamic range and undesirable distortion. Ronin's delays use 32-bit floating-point samples to create a full-bandwidth, noise-free delay.



Each delay has a maximum delay time of 12 seconds. The delay times are freely adjustable across their entire range, and can be locked to the tempo of your host. The delays can be used as loop recorders, letting you record new audio over a repeating phrase. The loops can also play backwards, even during recording. With just one of the two delay modules, you can record a phrase as a loop, transpose it down an octave by slowing it down to half its original speed, record a new phrase over the top, turn the whole loop around and play it backwards, transpose it up by speeding it up, record another phrase on top, and so on—all in real time, without stopping. Of course you can also use the delay modules for all standard delay-based effects such as echo, ping-pong delay, flanging, chorusing, etc.

There are three main controls in the delay module: TIME, FINE, and the multiplier switches. These three controls together determine the delay time of the module. The TIME slider and the multiplier switches can be thought of as the coarse controls. The range of the time slider is zero[^2] to a number of seconds equal to the value of the multiplier switch, which can be one, two, four, or eight. (Because of the emulation of older delay lines, the minimum delay time is actually not quite zero. It's a few samples more than zero, but less than one millisecond.) For example, if you set the time slider to its halfway point and the multiplier switch to 4x, the coarse delay time is two seconds, because half of four is two seconds.

The FINE slider multiplies the coarse time determined by the settings of the time slider and multiplier switches. The range of the fine slider is 0.5 to 1.5. At its halfway setting, its value is 1.0. The total delay time of the delay module is determined by multiplying the coarse delay time by the value of the fine slider. Continuing our example, if you set the fine slider to its maximum value (1.5), the total delay time would be three seconds, because two seconds times 1.5 is three seconds.

Usually we don't think about delay times in terms of seconds and milliseconds when we're writing music. We think in terms of rhythmic units, like an eighth note. Ronin can follow the tempo of your host and automatically calculate delay times to match rhythmic units, thus saving you from having to pull out your calculator in the middle of a recording session. Simply turn on the SYNC switch in the delay module. Now, as you adjust the time slider, you will see that the delay time is shown in metrical units. Delay times are expressed as fractions of a beat, with the assumption that a beat is one quarter of a measure, at the current tempo of your host. Dotted times are represented with the addition of a period (.) in the display, and triplets are represented by a capital T. For example, a sixteenth-note triplet is shown as 1/16 T in the display.

The fine slider is *not* affected by the sync switch and the host's tempo. Its value is still applied after the delay time is determined by the setting of the time slider, the multiplier switches, and the current tempo. This means that you can nudge the delay time a little shorter or a little longer than the precise metrical time shown beneath the slider, so that your delayed audio sits in the groove of your song. This also means that in order for the value shown in the display to be correct, you must set the fine slider to its default center position, so that its value is 1.0. (Most hosts let you set a control to its default value by holding down the CTRL key on your keyboard as you click the control.)

Remember that the two delay modules operate entirely independently, and have independent sync switches. This can be handy if you want to use one delay for tempo-based delay effects while using the other for short delay effects such as doubling, chorusing, or flanging. For these effects you should set the sync switch to off since they depend upon short delay times which have no relationship to your song's tempo. (Of course, sweeping the delay time of a flanger in sync with your song's tempo can be a cool effect, which you can achieve by using Ronin's LFOs to modulate the delay time, with the LFO's sync switch turned on. See the section on LFOs for more information.)

The FEEDBACK slider controls the level of the signal that is fed into the delay line. The delay line has two inputs: a main input, which has no level-control slider, and a feedback input which has the FEEDBACK level-control slider. Unlike most hardware and software delay processors, the feedback input is *not* directly connected to the output of the delay line. You can connect any signal to the feedback input, using Ronin's signal switching matrix (described later in this manual). This means that you can connect other modules in the feedback path of a delay, such as a filter to make the delayed signal change in timbre as it circulates through the delay. You can also connect the output of one delay module to the feedback input of the other, and vice versa, to create "ping-pong delay" effects that bounce the signal back and forth in the stereo field. Note, however, that you must connect the feedback input to something with the signal matrix before you'll hear anything happen when you turn up the feedback level slider.

The LOOP switch makes the delay act as a real-time loop recorder. When you turn on the Loop switch, any signal that is currently in the delay line will play over and over again, indefinitely (or rather, until you turn off the Loop switch, turn off the power to your computer, etc.). Any new signal that enters the input of the delay is overdubbed onto the looped audio, building up layers of sound. You can use the delay time sliders to change the length of the loop and its playback speed, altering the pitch of the looped audio.

When the Loop switch is turned on, the feedback input is automatically turned off. This is to make it easier to switch back and forth between looping and non-looping delay applications. If the feedback input were left on when the loop was turned on, the audio in the delay line would be added to itself over and over again, rapidly creating a thick, distorted mess.

The REVERSE switch causes the delay line to record and play back in the opposite direction. The effect of the Reverse switch is slightly different depending on whether or not the Loop switch is turned on. If the Loop switch is turned on, turning on the Reverse switch causes any audio currently being played in the loop to play backwards, over and over. You can record new audio over the backwards audio, and then turn the Reverse switch off to hear the original audio played forwards again and the newly recorded audio played backwards. If the Loop switch is not turned on, you will hear the audio currently in the delay line played backwards once, and then all subsequent audio will be played forwards, because the new audio coming into the delay line is being recorded in the same direction as the audio coming out of the delay line. Yes, this is a little confusing at first, but it becomes more understandable (and a lot of fun) as you play with it.

The THRU switch bypasses the delay line, sending its input signal directly to its output. The delayed signal is still sent to the output. This enables you to use the Loop switch to record and loop some audio, then play new material mixed with the loop, but not added to the loop itself. For example, you can record a phrase in the loop and then solo over the top of the phrase without adding to the loop by pressing the Thru switch.

Filters

Ronin has two multi-mode filters, modeled after the filters found in analog synthesizers. “Multi-mode” means that the filters have different response modes: low pass, high pass, band pass, and notch (or band reject). An unusual feature of Ronin’s filters allows you to “morph” between the different response modes rather than simply choosing one of the four. The filters have a resonance control and will self-oscillate at high resonance settings. The two filters operate completely independently and, thanks to Ronin’s signal-routing matrix, can be combined to create more complex filtering effects.

The main filter control is the frequency slider (whose name is abbreviated to FREQ in Ronin’s window). This control sets the center frequency, or cutoff frequency, of the filter. The filter passes and attenuates signals depending on their frequency relative to the center frequency, and the mode of the filter, as follows:

- Low pass: signals whose frequency is below the center frequency are passed unmodified, signals whose frequency is above the center frequency are attenuated.
- High pass: signals whose frequency is above the center frequency are passed unmodified, signals whose frequency is below the center frequency are attenuated.
- Band pass: signals whose frequency is near the center frequency are passed unmodified, signals whose frequency is further away from the center frequency are attenuated.
- Band reject: signals whose frequency is near the center frequency are attenuated, signals whose frequency is further away from the center frequency are passed unmodified.

The RESONANCE control causes the response mode of the filter to be sharper or more pronounced. For the low-pass and high-pass modes, turning up the resonance emphasizes signals whose frequency is near the center frequency. In the band-pass mode, turning up the resonance narrows the band of frequencies that are passed without attenuation. In the notch mode, turning up the resonance narrows the band of frequencies that are attenuated. This means, somewhat paradoxically, that the effect of the notch mode is most apparent when the resonance is turned all the way down, because this makes the notch widest.

Note that a resonant filter has greater than unity gain: it amplifies signals near the center frequency. This is what creates its unique sonic character, but it can have a tricky consequence within Ronin. If you set up the signal routing in the signal matrix such that the filter is included in a feedback loop, the entire loop will eventually start to oscillate and generate extremely loud signals. If you connect the output of a filter to its own input, or connect the two filters together so that the outputs of each are sent to the other’s input, you will almost certainly be rewarded by a loud and potentially piercing noise. Don’t say we didn’t warn you.

The MODE slider sets the frequency-response mode of the filter. Rather than clicking between the four modes, the slider moves freely, providing response modes that are in between two modes. When the indicator mark points at one of the reference marks on the panel, the filter’s response corresponds to that mode. The modes are abbreviated on the panel to save space: lp for Low Pass, bp for Band Pass, hp for High Pass, and br for Band Reject. For example, if you set the slider to halfway between LP and BP, the filter’s response will be a blend of the low-pass and band-pass modes, and pass more low-frequency signals than the simple band-pass mode.

Saturators

Ronin’s saturators distort signals that pass through them, somewhat like a guitarist’s distortion pedal or an overdriven tube pre-amp. The saturators distort the signal by amplifying it and flattening the peaks of the loud signals using a process known as **soft clipping**. The saturators have a number of potential uses within Ronin, such as distorting the original signal to give it a brighter, punchier, and more aggressive sound, distorting delayed signals to emulate the less-than-pristine characteristics of older hardware delays, and distorting signals to increase their harmonic content, making subsequent filtering more noticeable.

The saturators each have a single control that adjusts how much they boost and distort the signal. If the slider is all the way at the left, the saturators have almost no effect on the signal passing through them. As you move the slider to the right, the output signal becomes louder and more distorted.

Because the saturators work in part by amplifying the signal, use them with caution, particularly if you place them in a feedback loop with other modules. Turning up the saturator’s slider even a little bit will cause the signals in the feedback loop to grow louder quite rapidly. Also, connecting the output of a saturator directly to its own input is generally not something that you should do. (Yes, we realize that you’re now going to try it just because we said that you shouldn’t. You were probably scolded for running with scissors, weren’t you?)

Output Level and Panning Controls

The PAN and OUTPUT sliders in the Master section control the last processing stage that the signal passes through. Signals connected to OUTPUT 1 and OUTPUT 2 in the audio switching matrix are directed to this stage. The output level and panning modules are the last processing stage that the signals pass through before leaving the plug-in. You can think of this module as a simple two-channel mixer with stereo outputs. The order in which the other processing modules are connected can be changed freely with the signal matrix, but the signal always passes through the output and panning module last.

The pan sliders determine the position of the output signal in the stereo field. Obviously you must use Ronin in a stereo context in your host for these sliders to be useful. Ronin uses a special panning algorithm to create a more interesting sense of spaciousness than a standard equal-power panning algorithm. (What does that mean? It means that it sounds cool.)

The OUTPUT sliders amplify or attenuate the signals. Each has a range of −80dB, which effectively silences all but the loudest signals, to +3dB, which provides a small amount of boost. Often you will simply leave these sliders at their default position of 0dB (unity gain), but you can use them to adjust the relative signal levels of OUTPUT 1 and OUTPUT 2 when using them to process signals in two different ways.

Master Input and Output Controls

The input and output sliders provide adjustable amplification and attenuation of the signals as they enter and leave the plug-in. Each has a range of −80dB, which effectively silences all but the loudest signals, to +3dB, which provides a small amount of boost. Since Ronin is capable of creating effects with a wide dynamic range, the input and output controls are helpful for keeping signals within useful levels.

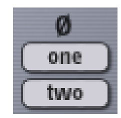
The INPUT slider adjusts the overall level of the signal arriving at Ronin's input from your host. It affects both the left and right channels equally.

The WET slider adjusts the level of the signals leaving Ronin. It affects both the left and right channels equally. This slider is useful for taming Ronin's occasionally high output level, and for adjusting the balance of processed and unprocessed signals if you're using Ronin as an insert effect.

The DRY slider adjusts the amount of unprocessed signal that is sent directly from Ronin's inputs to its outputs. If you're using Ronin as a send effect, you'll usually want to move this slider all the way to the left, since the unprocessed signal is already being sent into your mix via its channel strip. On the other hand, if you're using Ronin as an insert effect, you'll usually set this slider to unity gain (0dB).

If the MONO switch is turned on, the left and right input signals are added together and the summed signal is used for both the left and right input channels within Ronin. You may find this switch useful if you've created a preset intended for stereo processing, and later decide that it would be useful for processing a mono signal.

The KILL switch, when turned on, silences the input signal altogether. This switch can be used for fun effects like dub delays, in which only an occasional snare hit or the last word of a vocal line is passed through the delay. Just turn the Kill switch on, wait for the snare hit, and turn the switch off then back on again just long enough to let the snare through.

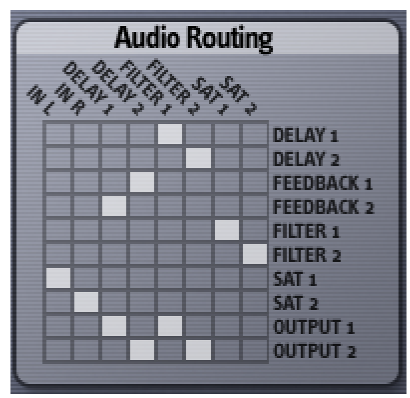


The Kill switch is also useful when using the delays for looping. After turning the loop switch on, turn the Kill switch on. Now your looped audio will play indefinitely, and incoming audio will not be added to the loop.

The two PHASE switches near the lower right corner of Ronin's window affect the plug-in's left and right input signals as they enter the signal switching matrix. If the INVERT switch is turned on, the corresponding signal is inverted, or given opposite polarity. Inverting a signal can cause interesting phase-cancellation effects when you mix the signal with the original, after delaying it or processing it in some other manner.

Audio Routing Matrix

The signal routing matrix, found on the right in the Routing Matrix section of Ronin's window, is how you connect the audio-processing modules to each other, and to the plug-in's inputs and outputs. All of the inputs and outputs for each audio module are represented in the signal routing matrix. The audio signal from any module can be connected to the inputs of one or more other modules. Signals from several modules can be mixed together and connected to one or more other modules.



You can think of signals entering the routing matrix at the top and leaving on the right. The columns are labeled with the **outputs** of audio modules, and the **inputs** to the plug-in itself. This may seem slightly confusing at first, but consider that both of these are **sources** of audio signals within Ronin. The signal sources are labeled across the top of the matrix as follows, in left to right order:

- IN L, IN R: the left and right inputs to the plug-in. Audio signals coming from your host arrive here. You have to connect at least one of these signals to something else in the matrix in order to process audio. (Note that the input signal is also always mixed with Ronin's output, in an amount controlled by the dry level slider in the Master Controls section.)
- DELAY 1, DELAY 2: the outputs of the two delay modules.
- FILTER 1, FILTER 2: the outputs of the two multi-mode filters.
- SAT 1, SAT 2: the outputs of the two saturation modules.

The rows in the matrix are labeled with the **inputs** of audio modules, and the **outputs** of the plug-in itself. Again this may seem slightly confusing, but both of these are **destinations** for audio signals. So by turning on switches in the matrix, you connect audio sources to audio destinations. The signal destinations are labeled down the right side of the matrix as follows, in top to bottom order:

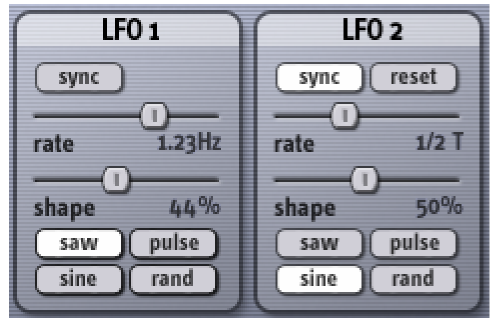
- DELAY 1, DELAY 2: the inputs of the two delay modules.
- FEEDBACK 1, FEEDBACK 2: the feedback inputs of the two delay modules. Signals routed to these destinations are mixed with signals routed to the DELAY 1 and DELAY 2 destinations, and the mixed signals enter the delay lines. The levels of the signals routed through the FEEDBACK inputs are controlled by the Feedback sliders.
- FILTER 1, FILTER 2: the inputs of the two multi-mode filters.
- SAT 1, SAT 2: the inputs of the two saturation modules.
- OUTPUT 1, OUTPUT 2: the outputs of the plug-in. Audio signals routed to these destinations are sent to your host, after passing through the output level and panning controls. You must connect something to these destinations in order to hear the audio processed by Ronin. Note that the OUTPUT 1 and OUTPUT 2 signal destinations are independent and not directly associated with the left and right output channels of the plug-in. If a Ronin patch does true stereo processing, usually the OUTPUT 1 signal will be assigned to the plug-in's left output by moving its pan slider all the way to the left, and the OUTPUT 2 signal will be assigned to the right output by moving its pan slider all the way to the right.

To make a connection in the matrix, click at the intersection of the source and destination that you wish to connect. A white square indicates that a connection is made between the source and destination. Click again on the dot to remove a connection. Hold down the CTRL key on your keyboard and click anywhere in the matrix to remove all of the connections at once.

Modulators

Like a modular synthesizer, Ronin has a number of modulators, that is, modules that affect the behavior of other modules. A modulator alters one or more parameters of the audio-processing modules, or of other modulators. For example, a low-frequency oscillator (LFO) modulator can be used to modulate (change) the delay time of a short delay line to create a chorusing or flanging effect, or to vary the frequency of a filter to create a synthesizer-like timbre sweep. Ronin's modulators are connected to its audio modules with a switching matrix.

LFOs



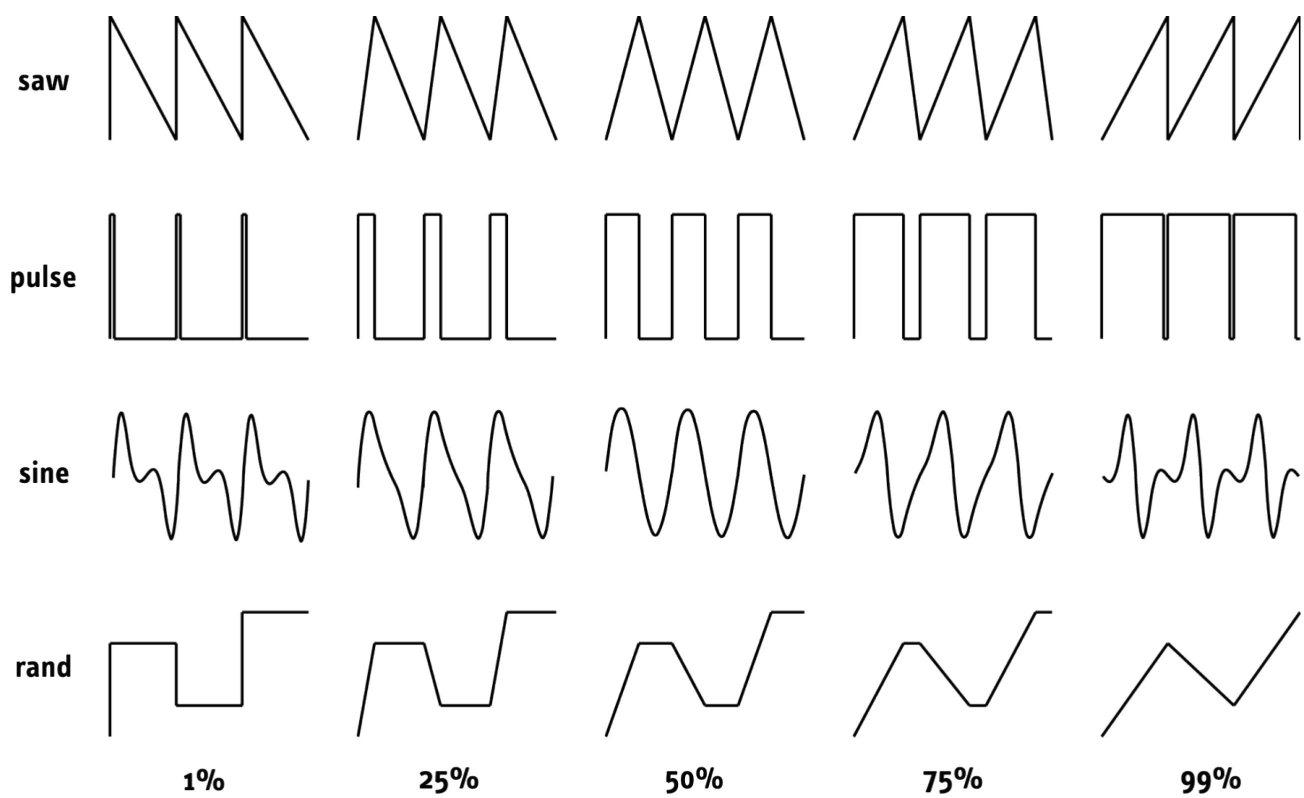
Ronin has two low-frequency oscillators (LFOs) which generate modulation signals that repeat over time. They have a variety of potential uses, such as varying the delay time of a delay processor slightly to create a flanging or chorusing effect, varying the cutoff frequency of a filter to create a wah-wah effect, or controlling the output level and panning parameters to create tremolo, auto-panning, or rhythmic gating effects. The LFOs can be locked to the tempo of your host for creating rhythmic effects that fit with the groove of your music, or can run freely and independently.

The RATE slider determines how fast the output of the LFO varies over time. If the sync button is not turned on, the LFO's rate can be varied from one cycle every 100 seconds (that is 0.01 cycles per second, abbreviated 0.01Hz) to ten cycles every second (10Hz).

The SHAPE slider and buttons work together to control how the LFO's output varies over time. The buttons let you choose one of four waveforms, with triangular, rectangular, sinusoidal, and randomly determined shapes. The Shape slider changes the basic waveform in different ways, depending on which waveform is chosen with the buttons.

- If the saw wave is selected, and the Shape slider is set to the middle of its range, the output of the LFO rises and falls evenly between its lowest and highest values, creating a symmetric triangular wave. If you rotate the Shape slider to the left, the LFO output rises more quickly and falls more slowly, creating what is often called a sawtooth wave. If you rotate the Shape slider to the right, the LFO rises more slowly and falls more quickly, creating what is called a ramp wave.
- If the pulse wave is selected, and the Shape slider is set to the middle of its range, the output of the LFO jumps between its lowest and highest values, staying for an equal period of time at both values. If you rotate the Shape slider to the left, the output stays at its highest value for a shorter period of time. If you rotate the Shape slider to the right, the output stays at its lowest value for a shorter period of time. In engineering terms, the Shape slider varies the duty cycle of the rectangular wave.
- If the sine wave is selected, and the Shape slider is set to the middle of its range, the output of the LFO varies smoothly between its lowest and highest values. The difference between a sine wave and a triangle wave is that the triangle wave abruptly changes direction when it reaches its highest and lowest values, whereas the sine wave gradually slows down, stops, and speeds up again when it changes directions. Moving the Shape slider warps and skews the sine wave without creating any sharp corners in its shape. Its effect is far easier to hear than to describe.
- If the random wave is selected, and the Shape slider is moved all the way to the left, the output of the LFO jumps to a random value, changing at a rate determined by the rate slider. As you rotate the Shape slider to the right, the output moves more slowly from one random value to the next.

The following diagram illustrates the different modulation signals generated by different settings of the shape and wave controls:



Note that the output of the LFO is bipolar, that is, it varies above and below zero. When you use an LFO to modulate a parameter, the parameter will change up and down relative to the value that you set with its slider.

If the SYNC button is turned on, the LFO's rate is determined by the tempo of your music as reported by your host. A value of 1/1 represents a whole note, a value of 1/8 represents an eighth note, and so on. Dotted notes, which have a duration equal to one and a half times a regular note, are shown with a period in the display. For example, "1/4." is displayed to indicate a dotted quarter note, which has a duration of a quarter note plus an eighth note. Triplets, which are groups of three notes that have the same duration as two regular notes, are shown with a T in the display. "1/16T" represents a duration equal to 2/3 that of a sixteenth note.

If LFO 2's RESET switch is turned on, the second LFO always resets to the beginning of its cycle when the first LFO starts its cycle. This switch has no effect if the Sync switch is turned on.

Envelope Follower

The envelope follower generates a modulation signal by measuring the amplitude (or loudness) of the signal arriving at the plug-in's inputs. The envelope follower always measures the two input signals added together, regardless of any connections made in the signal matrix.

The envelope follower has three controls which determine how it responds to the incoming signal:

The Sensitivity slider (labeled S) adjusts the envelope follower's overall sensitivity to the input signal. Essentially it acts like a gain control for the envelope follower. If the modulation signal generated by the envelope follower is too weak, raise the Sensitivity slider.

The Attack slider (labeled A) adjusts how quickly the envelope follower responds to increases in the incoming signal's level. If the Attack slider is at the left of its range, the envelope follower's output jumps almost instantly in response to increases in the input signal's level. As you move the Attack slider to the right, the envelope follower reacts more slowly to signal-level increases. The Attack slider is useful for making the envelope follower's output smoother when the input signal contains sharp transients, such as drum sounds.

The Decay slider (labeled D) adjusts how quickly the envelope follower's output decreases as the incoming signal's level decreases. If the Decay slider is at the left of its range, the envelope follower's output drops almost immediately when the input signal's level decreases. As you move the Decay slider to the right, the envelope follower reacts more slowly to signal-level decreases. The Decay slider is useful for stretching the envelope follower's output, making it fade away more slowly than the input signal.

Note that unlike the LFOs, the envelope follower's output is not bipolar. Its output is never less than zero, so it always acts to increase the parameters that it modulates.

MIDI

Data from your MIDI keyboard or other MIDI controllers can be used as a modulator in several ways, enabling Ronin's effects to respond dynamically as you play. These modulation signals are used within Ronin's modulation routing matrix to modulate the parameters of the audio modules.

One modulation signal is generated from MIDI note numbers received by Ronin. If you play middle C on your keyboard, a modulation value of zero is generated. As you play up the keyboard from middle C, the modulation value increases. If you play C one octave above middle C, the modulation value is +1. If you play below middle C, the modulation value becomes negative. Playing C below middle C generates a modulation value of −1. The modulation values, like all modulation values in Ronin, do not go above or below +1 or −1, even if you play outside of the two-octave range centered on middle C.

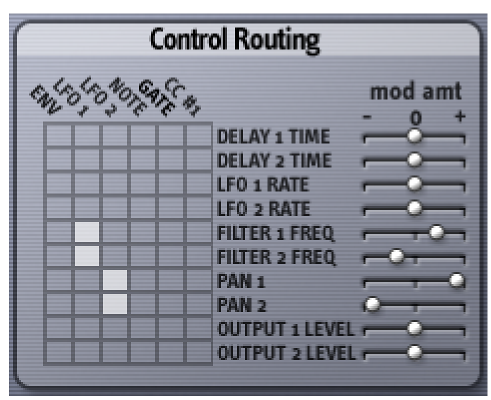
Another modulation signal, similar to the gate signal found in analog synthesizers, is generated from MIDI note messages. Normally this modulation signal has a value of zero. When you press any key on your MIDI keyboard, the gate modulation signal becomes +1. If you play additional keys while still holding the first key, the gate signal stays at +1. After you release all of the keys, the gate signal returns to zero.

A third modulation signal is generated from MIDI continuous controller messages. This modulation signal responds to messages from MIDI continuous controller #1, which is the message usually sent by the modulation control (or "mod wheel") found on most MIDI keyboards. The modulation value is generated directly from the data value sent by the controller. When the value of this controller message is zero, the modulation signal is also zero. As the controller data value increases (e.g., as you push the mod wheel forward), Ronin's modulation value increases. When the controller value reaches 127, the modulation value becomes +1.

Control Routing Matrix

The control routing matrix, found on the left in the Routing Matrix section of Ronin's window, lets you connect the output of the modulators to modulation destinations. Most modulation destinations affect parameters in the audio modules, but the rates of the LFOs can also be modulated. All of the modulation signal sources and all of the modulation destinations are represented in the modulation routing matrix. Any modulation signal can be connected to one or more modulation destinations. Modulation signals from several sources can be mixed together and connected to one or more destinations.

You can think of signals entering the routing matrix at the top and leaving on the right. The columns are labeled with the names of modulation sources, and the rows are labeled with the names of modulation destinations. The sources are labeled across the top of the matrix as follows, in left to right order:



- ENV: the output of the envelope follower.
- LFO1, LFO2: the outputs of the two low-frequency oscillators.

NOTE: the modulation signal generated from MIDI note numbers, in proportion to their pitch.

- GATE: the modulation signal which is +1 when any MIDI notes are played, and zero after all notes are released.
- CC1: the modulation signal generated from MIDI continuous controller messages for controller 1

The rows in the matrix are labeled with modulation destinations, that is, parameters that can be affected by modulators. The modulation destinations are labeled down the right side of the matrix as follows, in top to bottom order:

- DELAY 1 TIME, DELAY 2 TIME: the delay times of the two delay modules. Modulation signals connected to this destination are added to the current position of the delay's Fine slider. Use this destination to create effects such as flanging or chorusing by connecting it to an LFO.
- LFO 1 RATE, LFO 2 RATE: the rates of the two low-frequency oscillators. Modulation signals connected
- to this destination are added to the current position of the LFO's Rate slider. Use this destination to make the LFOs speed up as you move the mod wheel on your MIDI keyboard. Or, borrowing an old trick used on modular analog synthesizers, connect an LFO's output to its own rate modulation destination to create different wave shapes.

- FILTER 1 FREQ, FILTER 2 FREQ: the frequencies of the two filters. Modulation signals connected to this destination are added to the current position of the filter's Freq slider. Use this destination to create synthesizer-like filter-sweep effects.
- PAN 1, PAN 2: the stereo panning position of the two output signals. Modulation signals connected to these destinations are added to the current position of the Pan sliders in the Master Controls section. Use this destination to make Ronin's output move around in the stereo field.
- OUTPUT 1 LEVEL, OUTPUT 2 LEVEL: the loudness level of the two output signals. Modulation signals connected to these destinations are added to the current position of the Output sliders in the Master section. Use this destination to create tremolo effects with an LFO, or ducked-delay effects with the envelope follower. Want to simply gate audio by playing a note on your keyboard to create transformer effects? Use the MIDI gate modulation source and this modulation destination.

To make a connection in the matrix, click at the intersection of the source and destination that you wish to connect. A white square indicates that a connection is made between the source and destination. Click again on the dot to remove a connection. Hold down the CTRL key on your keyboard and click anywhere in the matrix to remove all of the connections at once.

Depth Controls

Each modulation destination—that is, each parameter that can be modulated—has a modulation depth control. The modulation depth controls are the small sliders to the right of the destination names in the Control Routing matrix. These sliders determine how much the modulation signals affect the parameter. If the depth control is at its center position (which is its default position), the modulation signal has no effect on the parameter. Regardless of which modulation signals you connect to this destination in the modulation switch matrix, the parameter will not respond to these signals if the depth control is at its center position. As you move the depth slider to the right, the modulation signal has an increasing effect on the parameter. If you move the slider only slightly above its center position, you will hear only a small change in the parameter as it responds to the modulation signal. A small amount of modulation is useful for effects that require only small changes in parameter values, such as varying the delay time in a flanger. If you move the slider to the far right of its range, you will hear the parameter change over a wide range in response to the modulation signal. Large amounts of modulation are useful for complete changes in parameter values, such as panning a signal all the way from one side of the stereo field to the other or sweeping a filter's frequency from completely closed to completely open.

If you move a depth control's slider left from its center position, the modulation signal is inverted before affecting the parameter. As you move the slider to the left, the modulation signal has a greater effect on the parameter—but in the opposite direction than if you move the slider to the right. For example, suppose the envelope follower is connected to one of the filter's frequency controls, and the frequency control's depth control is moved upward from its center position. The filter frequency will increase when the incoming signal becomes louder, because the envelope follower generates a positive modulation signal that is proportional to the loudness of the incoming signal. (Controlling a filter with an envelope follower in this manner is often called an "auto-wah" effect.) If you move the depth control to the left of its center position, the filter frequency will *decrease* when the incoming signal becomes louder, because the modulation signal coming from the envelope follower is inverted before it is applied to the frequency parameter.

Things become more complicated when using bipolar modulation sources, such as the LFOs. Since these modulation signals vary above and below zero, they increase and decrease the value of the parameters that they modulate. If you connect an LFO to a filter, and move the depth control slider to the right, you will hear the filter's frequency vary above and below the frequency set with its Freq slider. If you move the depth slider to the left, you will still hear the frequency vary above and below the value set with the Freq slider, because although the modulation signal is now inverted, it is still bipolar. However, if you connect the same LFO to both of Ronin's filters, move one filter's modulation depth slider to the right of its center position, and move the other filter's slider left of the center position, you will hear one filter's frequency go up while the other one goes down, and vice versa. The inverted modulation signal causes the filter's frequency to move in the direction opposite the other filter's frequency.

Automation

All of Ronin's controls, except for the routing matrix switches, can be automated using your host's automation features. Consult your host's documentation for information on how to use these features.

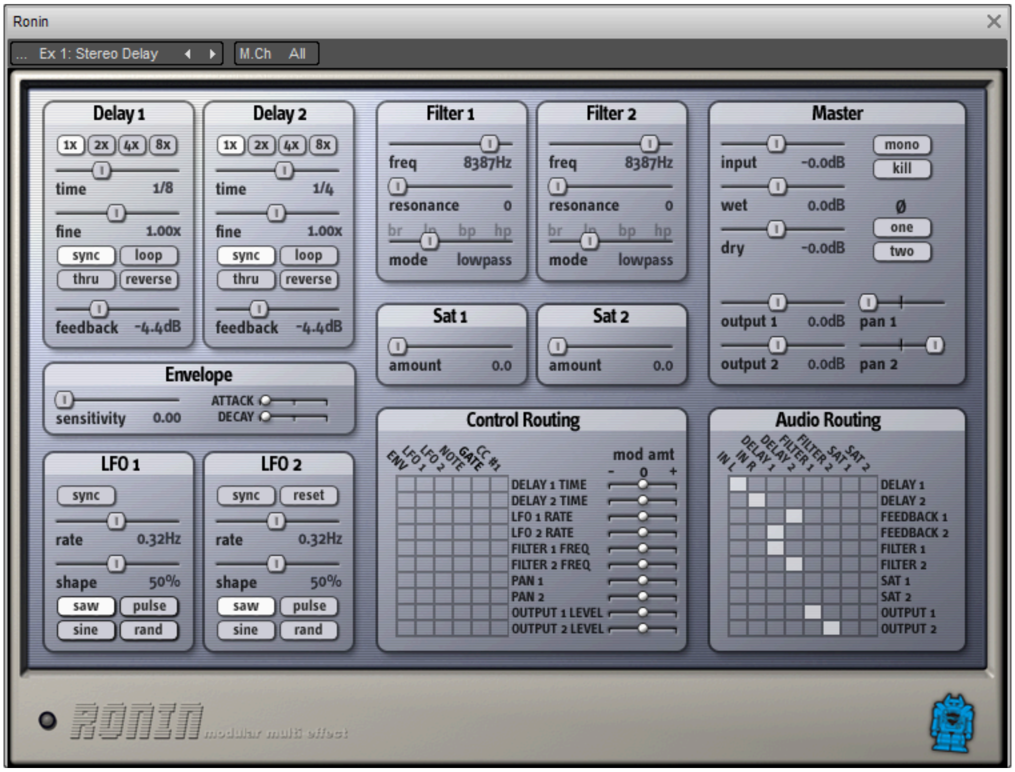
Putting It All Together: Some Examples

Here are detailed descriptions of four of Ronin's presets, which you will find near the bottom of the preset list. These descriptions illustrate some of the techniques you will use to create your own effects with Ronin. Place Ronin as an insert effect on a channel in your host, then load the presets as you read about them here.

Stereo Delay with Cross Feedback and Filtering

The preset "Ex 1: Stereo Delay" demonstrates Ronin's ability to set its delay times based on your host's current tempo, and how to use the signal matrix for creating filtered delay and stereo delay effects.

Look first at the Routing Matrix section. There are no connections in the modulation matrix, so the LFOs, envelope follower, and MIDI have no effect. In the signal matrix, you can see that the left and right inputs are connected to the first and second delay modules, respectively. Each delay's output is connected to a filter's input, and the filters are connected to the plug-in's two output channels. Hence signals pass first through the delays, then the filters, then out of the plug-in. Since we're using separate delays and filters for each channel, this is a "true stereo" processing configuration.



Also notice that we've connected the output of delay 1 to the feedback input of delay 2, and vice versa. This means that signals emerging from one channel's delay will be sent to the input of the other channel's delay, which will cause the delayed signals to bounce back and forth in the stereo field. Of course, in order for this to work, we have to pan the outputs of the plug-in to the left and right sides of the stereo field, which you can see we've done by looking at the Pan controls in the Master Controls section. We've also set the Wet and Dry output level controls to the same value, so that the delayed signal will be as loud as the original signal. (If you want to use this preset as a send effect, move the Dry slider all the way to the left.)

We've turned on both of the Sync switches for the delays, so the delay times will be calculated to match the host's tempo automatically. The delay times are set to an eighth note and a quarter note. The feedback sliders are set to moderate levels so that the delayed signals will fade out after a few repeats.

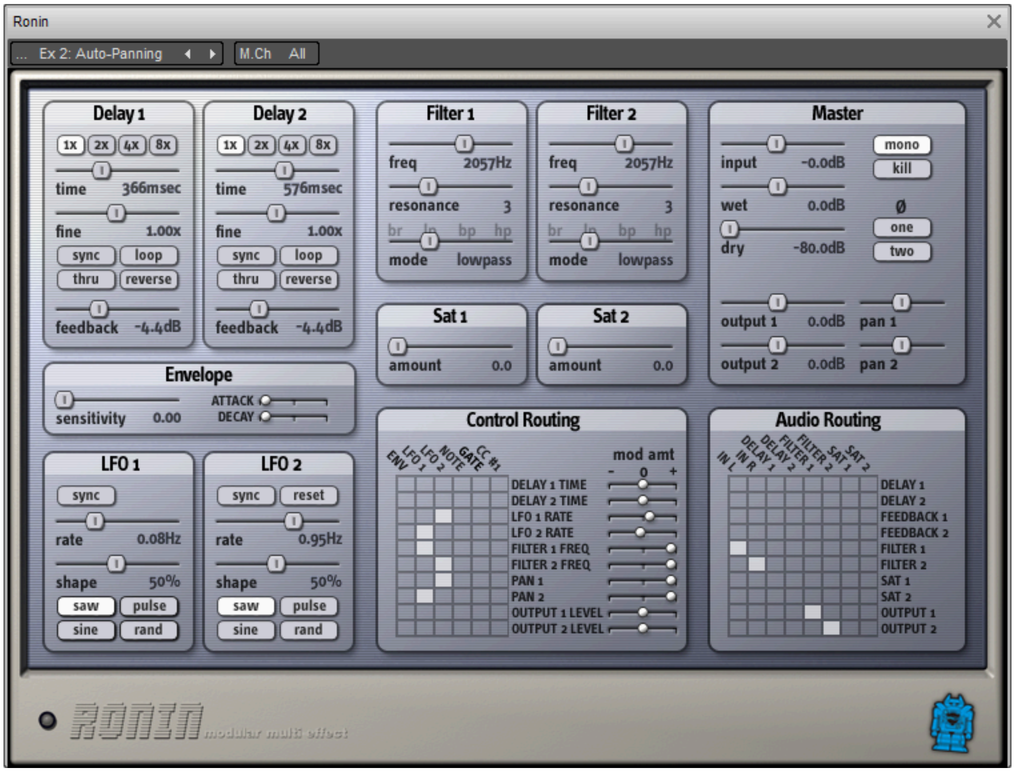
Try playing a drum loop or other strongly rhythmic audio phrase through Ronin. You'll hear the delayed audio playing in time, and bouncing back and forth between the stereo channels as it decays. Also notice that the delayed signals sound a little darker than the original signal. This is because the delayed signals pass through the filters before leaving the plug-in. The filters are set to their low-pass mode, with a cutoff frequency of about 8kHz. This setting filters out the higher frequencies of the signal. Filtering the delayed signal makes it "sit behind" the original signal in the output of the plug-in.

Filtered Auto-Panning

The preset "Ex 2: Auto-Panning" doesn't use Ronin's delays or panning controls at all. Instead it uses filters modulated by the LFOs to create a swirling effect.

As you can see by looking at the signal matrix, the signal connections are simple: the left and right input channels are connected to filters 1 and 2, and the filters are connected to their corresponding output channels. We've turned on the Mono switch in the Master Controls section so that if the input signal is stereo, its two channels are added together. We did this because this patch uses modulated panning to create a sense of stereo movement. Also, we move the Dry input level control all the way to the left, so that only the processed signal is heard. You might want to try turning this control up to mix in some of the original signal.

The connections in the modulation matrix are a little more complicated. LFOs 1 and 2 are connected so that they modulate the frequencies of the filters. Notice that the depth controls for both filters are moved to their far right position, so that the LFOs create large changes in the frequencies of the filters. The LFOs are also connected to the two Pan modulation destinations. The Pan controls are at their center positions, but their depth controls are also at the far right. Since the LFOs are bipolar, their modulation signals will cause the plug-in's output signals to move back and forth in the stereo field.



Finally, the output of LFO 1 is connected to the rate modulation destination of LFO 2, and vice versa. The depth controls for these two parameters are turned up only slightly. Connecting the LFOs to each other in this manner causes both of their outputs to vary in an unpredictable manner, making the effects of their modulation more interesting (or at least less predictable).

Try running a signal with a wide range of frequencies, such as a bright synthesizer pad or a guitar sound, through Ronin. You'll hear the output signal get brighter and darker, and move around, as the LFOs change the frequencies of the filters and the pan positions of the outputs.

The filters are set to their low-pass modes, with the resonance controls turned up somewhat. This gives them a familiar "synthesizer-like" sound. Try moving the Mode sliders to their other positions to hear the effects of the different response modes. The band-pass mode can be particularly effective in this preset.

If you want to make the panning and filtering move in time with your song's tempo, turn on the Sync switches for both LFOs, and turn off the connections to the LFO rate modulation destinations in the modulation matrix.

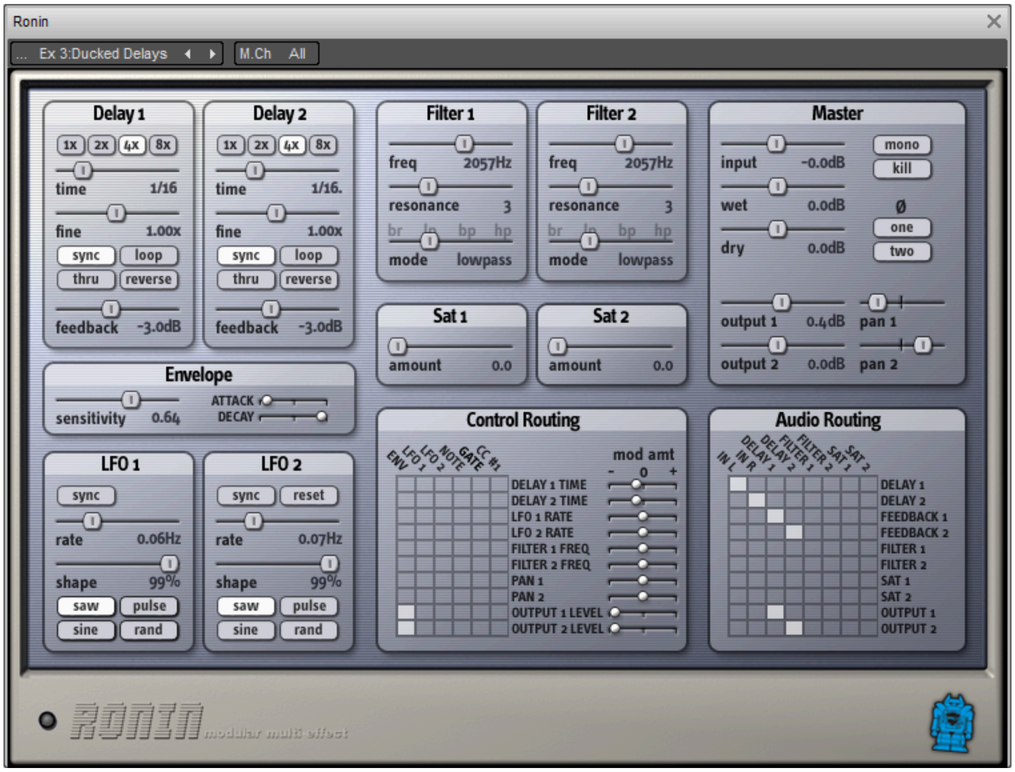
Ducked Delays

"Ducking" is the process of lowering the loudness of one signal in response to the loudness of another signal. For example, ducking is used in radio broadcasts to lower the volume of the music while the DJ talks. In the preset "Ex 3:Ducked Delays", we use ducking to lower the volume of the output of delays so that the delay effect is not heard until after you stop playing.

The signal matrix has a straightforward stereo-delay patch: the left and right inputs are connected to delays 1 and 2, and the delays are connected to their corresponding outputs. The outputs of the delays are also fed back into their inputs. The feedback levels are set high enough so that the delayed signals will repeat several times as they fade out.

The Wet and Dry level controls are set to the same value so that the delayed signals are as loud as the original signal. The Pan positions are set to the left and right to give the delayed signals balanced stereo placement without extreme panning.

Here's the tricky part: the output of the envelope follower is connected to both output Level modulation destinations. The depth controls for both of these destinations are at their lowermost positions. This means that the modulation signal from the envelope generator is inverted before being applied to the output levels, which means that as the input signal gets louder, the output levels will be decreased.

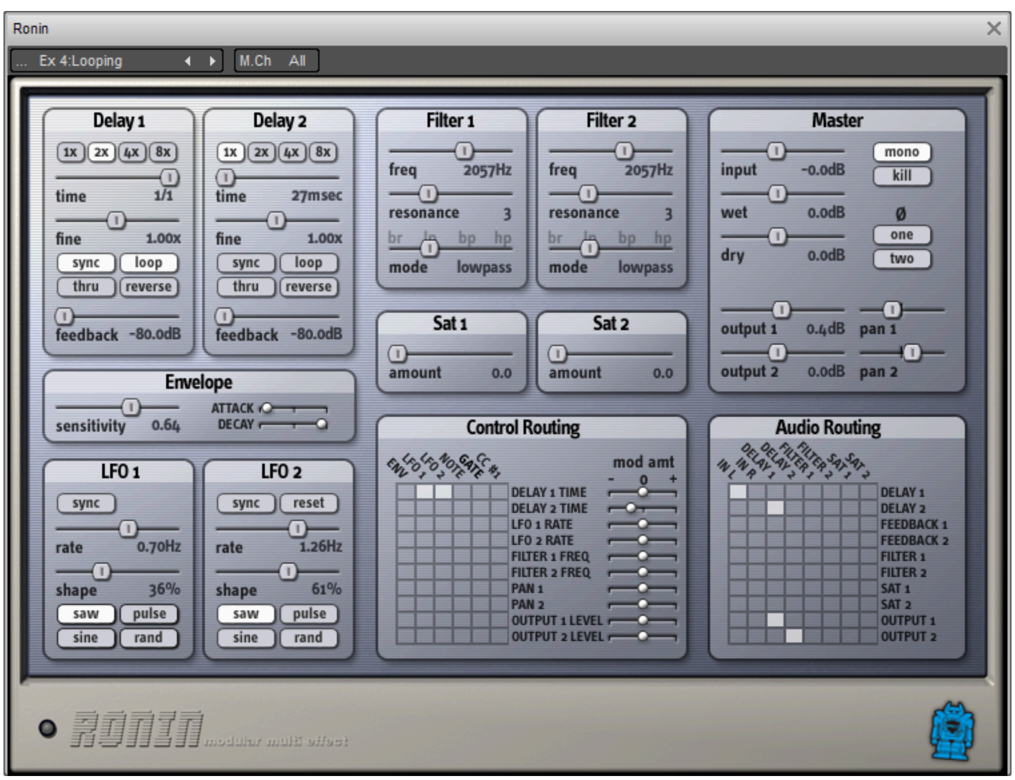


Set up your host to play a few bars of a drum pattern or other fairly dense musical passage through Ronin. As the music plays you will hear little if any delayed signal. When the input signal stops, you'll hear the delayed signal fade in. The length of time it takes for the delayed signal to fade in is determined by the envelope follower's decay slider. Try lowering the slider to hear the delayed signal fade in more rapidly.

Fun With Looping

The preset "Ex 4:Looping" will introduce you to some of the fun things you can do with Ronin's looping delays. You'll need a synthesizer, guitar, microphone, or some other way of getting audio into your host in real time to follow our explanation.

Look at the signal matrix. The left input is connected to delay 1, but we've turned on the Mono switch in the Master section so both input channels are sent to delay 1. (This is simply to make it easier for you to connect a signal to the plug-in within your host.) The output of delay 1 is connected to the plug-in's first output channel, and also to delay 2. Delay 2 is connected to the second output channel.



Delay 1 is set up for interactive looping. The Sync switch is on so that the length of the loop is determined by your host's tempo, which should be set to some moderate value like 120bpm. The Loop switch is turned on so that audio recorded in the loop will play indefinitely.

Delay 2 is set to a very short delay time. Outputs 1 and 2 are panned slightly left and right. This creates a stereo chorusing effect, since the left and right outputs of the plug-in are delayed slightly with respect to each other. We use both LFOs to modulate the delay time of delay 2, to add motion to the chorus effect. The LFOs are set to different frequencies and somewhat different wave shapes. Mixing both LFOs together makes the chorus effect less repetitive.

Play a short phrase on your instrument (or say something into your mic). You'll hear the audio playing over and over. If you don't like what you hear, turn the Loop switch off, wait 'til the last repetition of the loop has played, then turn it back on.

Move delay 1's Fine slider all the way to the left. You'll hear the loop slow down to half its original speed, playing one octave lower. Turn on the Reverse switch. You'll hear the loop playing backwards, still one octave lower.

Play another phrase. The new phrase will be overdubbed on the loop.

Hold down the CTRL key on your computer's keyboard and click the Fine slider to return it to its default, center position. Now you'll hear the first phrase playing at its original speed and pitch, but still backwards, and the new phrase playing at twice its original speed and pitch.

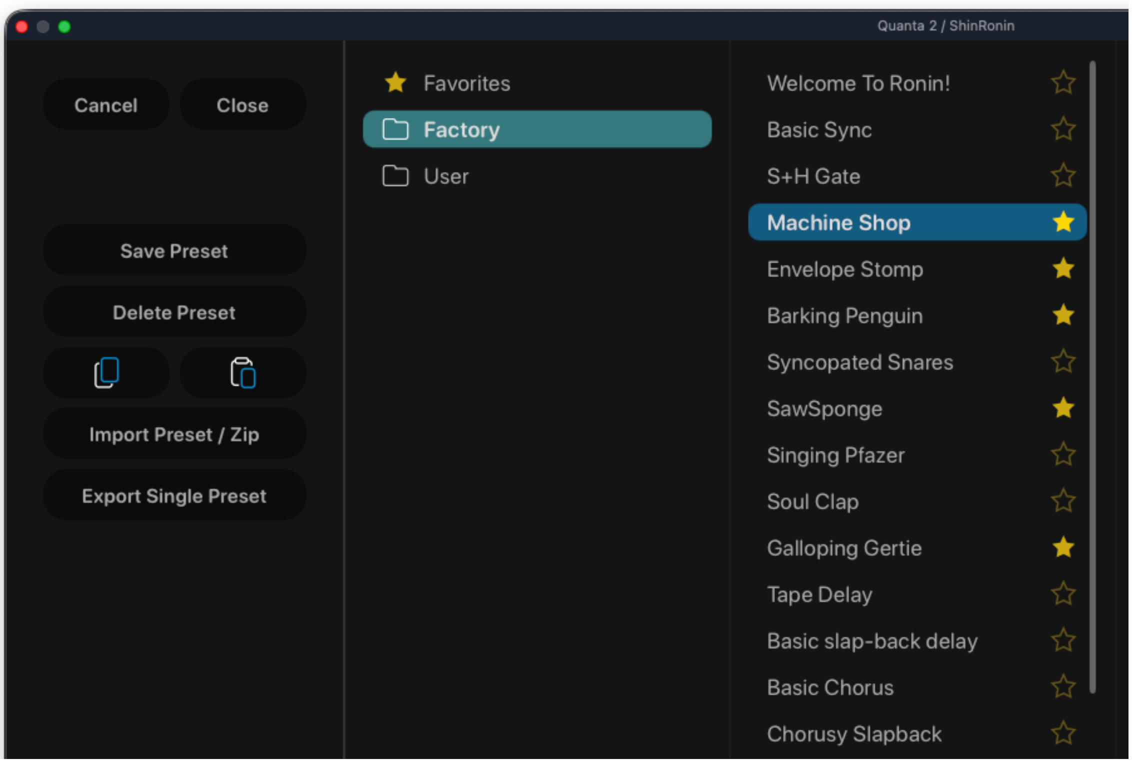
Click the Reverse button. Now you'll hear the first phrase as you originally recorded it, and the new phrase playing backwards at twice its original speed and pitch.

Repeat as desired. We used only one of the two delays for looping, and used the second one to create a chorus effect just for the sake of illustration. Of course you can use both delays for looping, building up a new phrase in one while the other continues to loop.

Presets

Shin-Ronin includes a collection of presets to serve as a demonstration of its capabilities and inspirations for your own creations. There are a few controls at the top of the window associated with presets:

The name of the current preset appears in the center. (You probably figured that out yourself.) Clicking the little dots on the left and right loads presets in alphabetical order. Clicking the star outline marks the preset as a favorite to help you find it again in the future. To examine all of the presets, click the name of the current to open the preset browser.



The browser displays presets and folders in scrollable lists, arranged in columns. The leftmost list shows the folders within Shin-Ronin's preset collection, grouped in two categories: Factory and User. Clicking any of these folders reveals its contents in the next list. Clicking on a preset name loads the settings into Shin-Ronin. Click the CLOSE button in the preset browser to dismiss it. If you click the CANCEL button instead, the browser closes and Shin-Ronin's settings revert to their previous state. Double-clicking a preset name loads the preset and dismisses the preset browser. Loading a preset irretrievably erases the current settings, so if you have created something that you want to use again, save it as a new preset before loading another preset.

Once you have clicked on any item in the panel, you can navigate within the preset browser with the keys on your keyboard. The left and right arrow keys move the selection between columns, and the up and down arrow keys move it within the list. Tapping the ESC key has the same effect as clicking the Cancel button.

Just like the star at the top of the window which we mentioned previously, the stars to the right of preset names mark presets as favorites. Clicking a filled start removes the mark from a favorite. Once you've marked at least one favorite, a correspondingly named category appears in the leftmost column. Clicking it shows you all the presets you've marked, and clicking their names loads them as usual.

The folders and presets in the browser correspond to folders and files within Shin-Ronin's own folder on your storage device (i.e. your computer's hard drive or SSD). This folder is located at `C:\ProgramData\Audio Damage\ShinRonin\` on Windows, and `~/Music/Audio Damage/ShinRonin/` on macOS and Linux. You can store your presets anywhere you like, but for them to show up in the User list they must be placed in the User folder within Shin-Ronin's folder. Also, to avoid possible collisions during future updates, do not store your presets within the Factory folder.

Any folders you create within the User folder will show up as folders in the User list. You can create sub-folders within the User folder, and sub-folders within those folders. You can't nest folders deeper than that because the preset browser has only four lists.

To save your presets, click the SAVE PRESET button at the left edge of the window. This invokes a dialog box with a couple of helpful features. As the text therein describes, you can create a folder within the destination folder (whose name is given at the top of the dialog box) by adding the folder's name to the beginning of the preset's name, separated by a slash mark.

Clicking the RANDOM NAME button replaces the preset's name with a pair of words chosen at random from two lists. While the resulting names won't have any connection with what the plugin is doing, you may find this feature useful for coming up with alternatives to routine names like "My Preset 12".

Potential pitfall: once you've saved a preset, clicking its name in the list loads the preset, overwriting whatever changes you've made since you saved the preset. Hence if you want to save the preset again to preserve the changes you've made, do not click on its name before saving it.

You can delete presets and folders from the lists by clicking their name and then clicking the DELETE PRESET or DELETE FOLDER button. Shin-Ronin will give you a chance to confirm this action or cancel it. If you confirm, the preset/folder will be removed from your storage system and is gone for good.

Importing and Exporting Presets

Preset files are plain-text XML files so that you can exchange them online in forums, copy them between a Windows computer and a Macintosh (and even between an iPad and a regular computer), email them to your friends, etc.

The two buttons with icons representing copying and pasting (copy on the left, paste on the right) copy Shin-Ronin's current settings to the system clipboard and paste settings from the clipboard. You can use the copy and paste commands to transfer settings between two instances of Shin-Ronin or paste the settings into an email message or text editor. When copied to the clipboard, presets are presented in the same XML text as used in preset files.

The IMPORT PRESET/ZIP button provides a way to add presets to Shin-Ronin without manually moving them into the appropriate folders in your file system. Clicking this button produces a file-browser window wherein you can select either a single preset file or a .zip file containing one or more presets. After you select the file, Shin-Ronin copies the preset(s) into whichever folder you have selected in Shin-Ronin's preset list, unzipping the file first if necessary.

Depending on whether you've selected a preset or folder, the EXPORT SINGLE PRESET or EXPORT FOLDER AS ZIP button performs the complementary functions of the Import button. First select either a preset or a folder in Shin-Ronin's list, then click the export button. A file-save window appears; choose a location in your file system, give the file a name, and click SAVE. If you have chosen a folder in Shin-Ronin's preset list, the plugin places it and all of the presets it contains in a .zip file.

Default Preset

If you save a preset with the special name "Default" in the User folder, new instances of Shin-Ronin will load it automatically when you add it to your DAW session. You can use a default preset file to give yourself the same starting point with Shin-Ronin whenever you use it. The plugin installer creates a default preset file for you but feel free to replace it with your own.

Document Revisions

- 2026-06-23:
  - Derived from Ronin 1.6 manual, with up to date material for system requirements, installation, presets, etc.