

RetroMod

Lead

USER GUIDE

Credits

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1. Introduction

Thank you for purchasing our RetroMod Lead Instrument.

We hope you enjoy it as much as we do, have a lot of fun with it and make great music with it.

RetroMod Lead is a premium quality instrument that combines an advanced synthesis engine with a stunning array of sample sounds from a world famous scandinavian synthesizer. Built on the entirely new Acktion platform, it features the same powerful engine as our Biotek organic synthesizer.

RetroMod Lead's user interface is designed to place a straightforward range of sound sculpting tools right at your fingertips.

The deep sound creation possibilities include two oscillators, followed by a multimode filter and a four-band equalizer with shelf and bell curves.

Each sound is mixed into four effect busses, each featuring a sum overdrive, a compressor, a modulation effect like chorus, flanger, phaser or sum filter, and a space effect like delay or clocked delay, or natural, plate or non-linear reverb.



2. Installation & Authorisation

The installation including all sample and sound data requires approximately 8.2 GB space on your hard drive.

RetroMod Lead can be installed and unlocked on up to 4 different computers with a single license.

2.1 Windows PC



Please observe the system requirements of your host application! RetroMod Lead runs within 32 bit as well as 64 bit host applications.

1. Refer to the folder where the downloaded RetroModLead.zip archive is located.
2. Double click on the archive file to extract it.
3. Double click on the “RetroModLead.exe” file. This launches a special installation program.
4. Follow the on-screen instructions.

Default installation locations on Windows:

VST 32-bit: C:\Program Files
(x86)\Steinberg\VstPlugins\RetroModLead.dll

VST 64-bit: C:\Program Files\Steinberg\VstPlugins\RetroModLead(64-bit).dll

AAX 32-bit: C:\Program Files (x86)\Common Files\Avid\Audio\Plug-Ins\RetroModLead.aaxplugin C:\Program Files\Common

AAX 64-bit: Files\Avid\Audio\Plug-Ins\RetroModLead(64-bit).aaxplugin

2.2 macOS

1. Refer to the folder where the downloaded RetroModLead.zip archive is located.
2. Double click on the archive file to extract it.
3. Double click on the “RetroModLead.pkg” file. This launches a special installation program.
4. Follow the on-screen instructions.

Default installation locations on Mac:

AU: /Library/Audio/Plug-Ins/Components/RetroModLead.component

VST: /Library/Audio/Plug-Ins/VST/RetroModLead.vst

AAX: /Library/Application Support/Avid/Audio/Plug-Ins/
RetroModLead.aaxplugin

2.3 Linux Ubuntu

1. Copy the RetroModLead.so file into your VST plugins directory. Common locations are shown below.

2. Ensure your host is set to scan the location

3. Perform a plugin scan in your host

Common Linux VST directories:

/usr/lib/vst /

usr/local/lib/vst

~/vst

Please follow these instructions to install the RetroMod Lead content

1. Create a RetroMod Lead directory at "~/Tracktion/RetroModLead/" (may already exist)

2. Copy the "Instruments" content to "~/Tracktion/RetroModLead/Instruments"

3. Copy the "Samples" content to "~/Tracktion/RetroModLead/Samples"

4. Restart your host and any open instances of RetroModLead

You should end up with two folders:

"~/Tracktion/RetroMod Lead/Instruments" containing ".RetroMod Lead.instrument" files

"~/Tracktion/RetroMod Lead/Samples" containing ".acktionsamples" files

To authorize the plugin on Linux, you must use the offline authorization method.

2.4 Authorisation

RetroMod Lead uses a software based copy protection and can only run without any restrictions with an activated license.

After installation, please load RetroMod Lead as instrument plug-in within your DAW. The plug-in is currently locked and running in demo mode (a noise occurs periodically).

Click on the Unlock button in the headline of the plug-in window to open the authorisation window.

Click Unlock to start a 30 day trial, if you haven't purchased the plug-in. If you have purchased the plug-in it will become unlocked permanently.

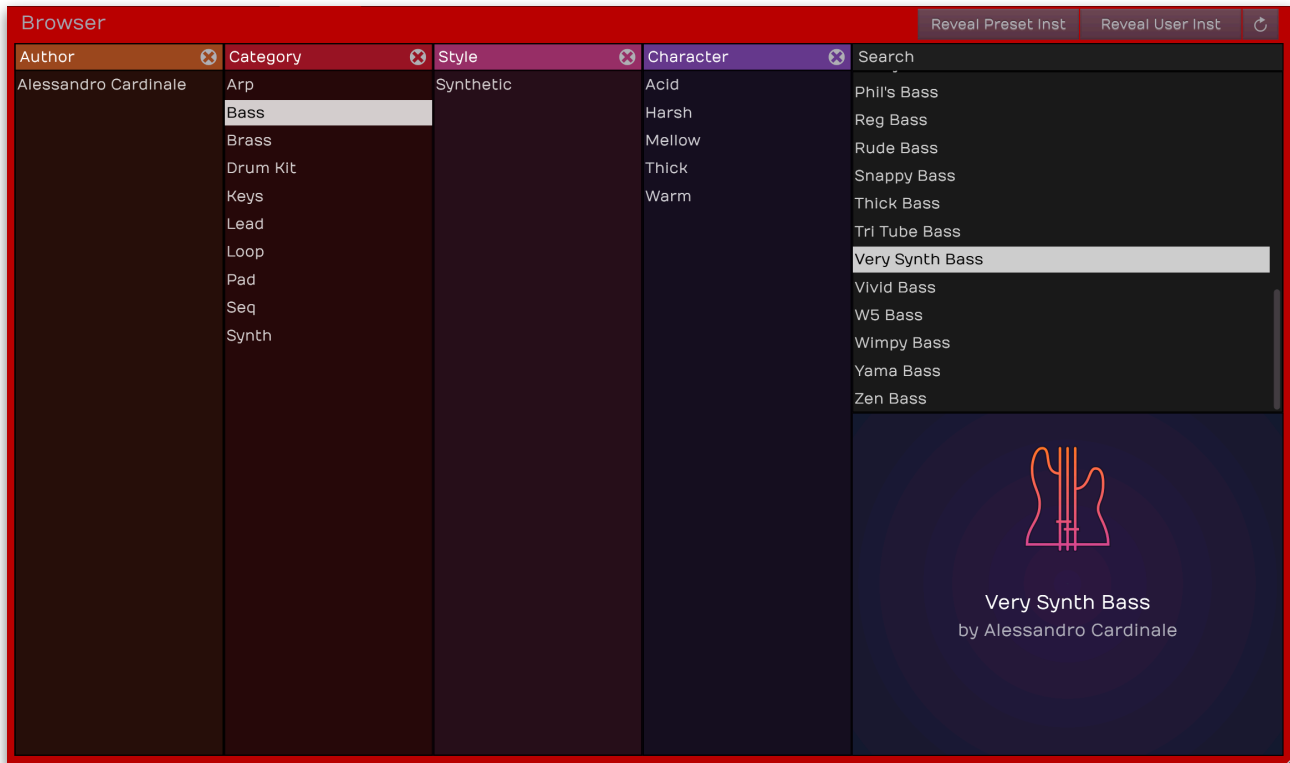
Please enter your email address and password from Tracktion Marketplace and click on "Register" to unlock RetroMod Lead permanently. You need an active internet connection for this procedure.

You can alternatively click on the “Offline Unlock” button, if your computer has no internet connection. Please follow the onscreen instructions.

After a successful authorisation, you can use RetroMod Lead without limitations and full featured forever.

3. Basic Operation

3.1 Loading and Saving Instruments



Click on the folder button in the main area to open the **Browser** section. Here you can quickly locate e.g. categories or styles of Instruments by selecting one or more tag. Standard selection is cumulative unless the “command” (Mac) or “Ctrl” (Windows) key is used in conjunction with selection – this will select individual tags.

Use the **Search** panel in the upper right corner to perform a text search of the Instrument library.

To load a selected Instrument in the instrument result list, simply double click on its name.

To save an Instrument, click on the **Save** button in the Head section. User Instruments are added to the Instrument library. A newly saved preset is appended to the end of the file browse list. When launching RetroMod Lead, the files are scanned starting with your user's instruments folder, followed by the public instruments folder (the one with the factory instruments). This results in the order that your creations come first followed by the ones from the factory sound bank.



3.2 Control Elements

Dials

To set a value, click on the dial, hold down the mouse button and move the mouse up and down. A double click resets the parameter value to its default.



Buttons

A simple click on a button activates or deactivates its function. Buttons light up when they have been used.



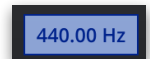
Pop-up Menus

Click on the corresponding slot to open a pop up menu where you can choose the desired parameter.



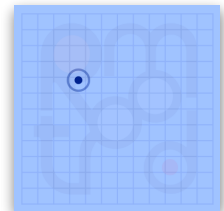
Value Selection Fields

Click in a Value Selection Field and move the mouse up or down to easily change the parameter value. Double click in a Value Selection Field to set the parameter value to its default value.



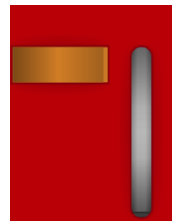
Pad

Click on the circle button within the X-Y pad and move the mouse. A single click on any position within the pad will set the button to this position.



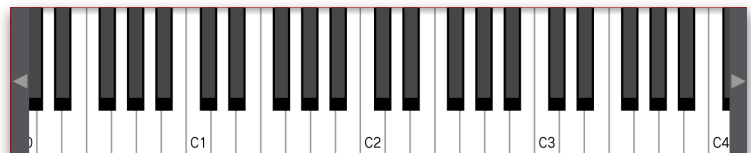
Wheels

To change the value, click on the wheel and drag it up or down (Mod Wheel) or left and right (Pitch Bend). The Pitch Bend wheel snaps automatically back into its center position as soon as you release the mouse button.



The Virtual Keyboard

RetroMod Lead provides a virtual keyboard with 128 keys. Click on a key to play the corresponding note. The vertical position of the initial click of the key determines the velocity.



4. The Basic User Interface

RetroMod Lead is divided into different sections which are described in detail within the next chapters.



The **Header Section** includes basic functions.

The **Browser Section** offers an easy-to-edit realtime modulation section.

The **Pad Section** on the right offers realtime performance parameters.

The **Virtual Keyboard** provides a virtual keyboard with 128 keys

The **Macro Controls** lets you control the basic sound parameters.

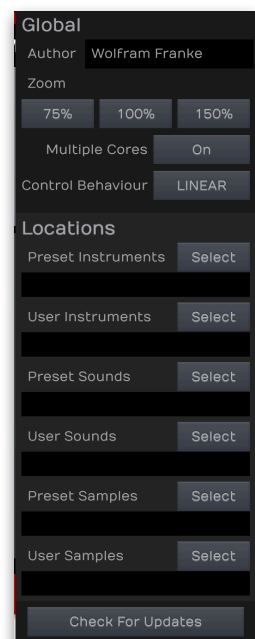
The **Effect Section** includes a four-band equalizer and four effects for each sound.

4.1 Header Section

The Header section offers global and basic settings.



- Click on the **RetroMod Lead** logo to open a pop-up window with additional information about the plug-in version number as well as the development team.
- Click on the **Folder** button to open the **Browser** section. Read more in the chapter “3.1 Loading and Saving Instruments” on page 8 of this manual.
- Click on the **Effects** button to open the **Effects** section. Read more in the chapter “4.5 The effects Section” on page 16 of this manual.
- The **Instrument name field** displays the name of the actual Instrument. Double click on the name for renaming the Instrument.
- The **Instrument information field** opens a pop-up window where you can enter several information regarding the actual sound. These information will be saved with the sound.
 - **Name:** Here you can enter a name for the actual instrument.
 - **Category:** Here you can enter a instrument category, e.g. Lead
 - **Author:** Here you can enter the name of the sound designer.
 - **Style:** Here you can enter a sound or music style of the actual instrument.
 - **Character:** Here you can enter a sound character of the actual instrument.
 - **Description:** here you can enter a further description regarding the instrument.
- The **Save** button lets you save the actual Instrument. You can also change the name of the stored Instrument.
- The **Master Volume** dial allows you to adjust overall volume of RetroMod Lead. The meter left besides the Volume dial displays the outgoing audio signal.
- Click on the **Global** button (gear symbol) to open a pop-up window with additional parameters. Click the **Global** button again to close the Global menu.
 - **Author:** Here you can enter the name of the sound designer.
 - **Zoom** - click on the corresponding zoom button to change the size of the Collection plug-in user interface. The regular size is 100%. You can also drag the handle in the lower right corner of RetroMod Lead to change the size of the user interface to your needs.
 - **Multiple Cores** - when running multiple instances of RetroMod Lead, turning off the multi core processing can improve CPU performance on certain systems – typically older computers. We strongly recommend to activate this option.

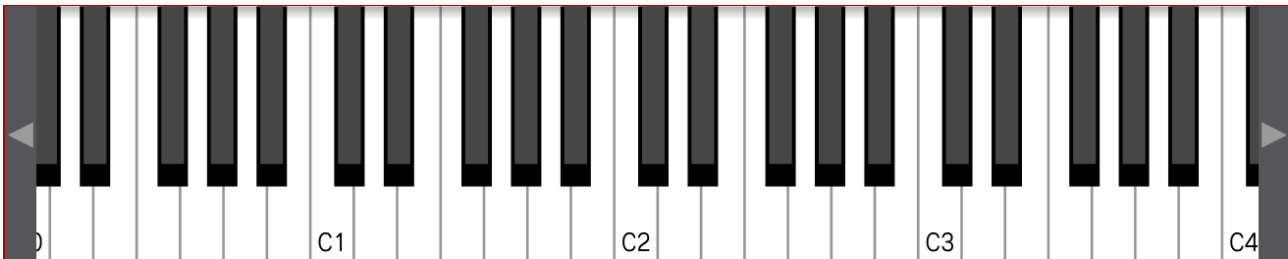


- **Control Behaviour** - when using a mouse to adjust the rotary controls, the input movement can be selected to be linear or circular. If set to “Circular”, you change a value of a rotary control by clicking on it and moving your mouse in a circular motion around the control. You don’t need to stay inside the control bounds, though, you have a wide range to fine-tune the value. If set to “Linear”, rotary controls behave like value display controls. Moving the mouse up or right increases a value, moving it left or down decreases it.
- **Locations** - Click on **Preset Samples** to open the folder where the preset samples are placed. If you want to move the sample content you need to show RetroMod Lead the corresponding data path. Click on **User Samples** to open the folder where the user samples are placed. If you want to move your user sample content you need to show RetroMod Lead the corresponding data path.
- **Question Mark** button - click on this button to open the PDF manual of RetroMod Lead.
- **Panic** button: Sends and executes an “All Notes Off” command to terminate stuck notes. **Panic** immediately sets all envelopes to their release phases. Panic also stops the arpeggiator playing when running in Latch mode.

4.2 Browser Section

For more information please refer to the chapter “3.1 Loading and Saving Instruments” on page 8.

4.3 Keyboard Section



RetroMod Lead provides a virtual keyboard with 128 keys and two wheels. Click on a key to play the corresponding note. The vertical position of the initial click of the key determines the velocity. Click on the left or right arrows to shift the keyboard at your needs.

4.4 Macro Controls Section

This section includes macro controls to edit an Instrument quickly. All controls are patch specific macro controls pre-assigned by the sound design team to numerous features.

Controls can be positive or negative in value – for example the Pitch dials on the right image offers settings for the pitch of each Instrument.

Holding the “Option” (Mac) or “Alt” (Windows) key and clicking on any parameter sets the control to its default location.



The following controls are available:

Global

- **Unison:** Controls the detuning of the unison voices. Each voice is detuned differently.
- **Glide:** “Glide” or “Portamento” describes the continuous gliding from one note to another. This effect can be created on fretless stringed instruments or some brass instruments (e.g. trombone). It is very common on synthesizers and used throughout all music styles. Please note that Glide affects the pitch of all oscillators. This dial determines the glide time. Low values will give a short glide time in a range of milliseconds that gives a special character to the sound. High values will result in a long glide time of up to several seconds which can be useful for solo and effect sounds.
- **Analog Feel:** Allows you to randomly offset the pitch with each note that is played. Higher values cause stronger variations.

Oscillators 1 and 2

- **Octave:** Sets the basic pitch of the Instrument in steps of an octave.
- **Semitone:** Sets the pitch of the Instrument in semitone steps. The standard setting for this parameter is 0.
- **Cent:** Fine-tunes all oscillators in steps of 1/128th of a semitone.
- **Ring Mod** (only available for Oscillator 1): Controls the depth of ring modulation between Oscillator 1 and Oscillator 2.



Ring Modulation is similar in sound to Frequency Modulation, albeit with less complex harmonic content. A Ring Modulator combines two input signals in a way that it outputs all sum and all difference frequencies of all input frequencies. That means, if you have a sine wave oscillator with a frequency of 200 Hz and another one with a 300 Hz frequency, the ring modulator produces a signal consisting of a 100 Hz sine wave ($300 \text{ Hz} - 200 \text{ Hz} = 100 \text{ Hz}$) and a 500 Hz sine wave ($300 \text{ Hz} + 200 \text{ Hz} = 500 \text{ Hz}$). You can easily imagine that more complex waveforms produce far more complex output signals. Note that you shouldn't set the pitches of the two oscillators to similar or identical frequencies, since this might result in signals somewhere in the subsonic range or even 0 Hz (DC), although you can get interesting pumping effects with it.

- **Level** (only available for Oscillator 2): Controls the output level of the Oscillator 2.

FM

- **Pitch:** Sets the pitch for the frequency modulation.
- **Level:** Determines the amount for the FM level.

Amp Section

Basically, an envelope offers three time stages which controls Attack, Decay and Release. Sustain is a time parameter and is defined by the length of the played note that triggers the envelope.

- **Attack:** Determines the attack rate or amount of time it takes for a signal to go from zero to maximum level.
- **Decay:** Determines the decay rate or amount of time it takes for a signal to reach the Sustain level.
- **Release:** Once the note has ended, the release phase begins. During this phase, the envelope fades to zero at the rate determined by the Release value.

Filter Section

A filter is used to attenuate or emphasize certain frequency ranges of a source signal. Most filter types consist of a pass and a stop band, some have two pass or two stop bands. The pass band is the frequency range where the input signal can pass (almost) unaffected, the stop band is the frequency range that is attenuated, sometimes fully.

- **Cutoff:** Controls the cutoff (on low pass or high pass types) or center (on band pass or band stop types) frequency of the filter.
- **Resonance:** Controls the resonance of the filter if set to any of the 12 dB or 24 dB filter types. On 6 dB filter types, this control does

nothing. On band stop types, it narrows the stop band, which is almost inaudible.

- **Env:** Determines the amount of filter envelope modulation that is applied to filter cutoff.

Basically, an envelope offers three time stages which controls Attack, Decay and Release. Sustain is a time parameter and is defined by the length of the played note that triggers the envelope.

- **Attack:** Determines the attack rate or amount of time it takes for a signal to go from zero to maximum level.
- **Decay:** Determines the decay rate or amount of time it takes for a signal to reach the Sustain level.
- **Release:** Once the note has ended, the release phase begins. During this phase, the envelope fades to zero at the rate determined by the Release value.

Master Section

- **Output:** Sets the overall volume for the current instrument.
- **FX Amount:** Controls the mixture between the dry signal output and the wet effect signal output processed by the effect bus. A setting of 0% results in a completely dry signal, 50 % means that the signal is routed with equal volume into the effect unit and forwarded dry and 100 % means that no signal is sent out dry but it is routed into the Effect bus with full level.
- **Master Tune:** 440.00 Hz is widely used as concert pitch in the United Kingdom and the United States. In continental Europe, the frequency of the note A commonly varies between 440 Hz and 444 Hz. In the period instrument movement, a consensus has arisen around a modern baroque pitch of 415 Hz (A_b of A 440), baroque for some special church music (Chorton pitch) at 466 Hz (A[#] of A 440), and classical pitch at 430 Hz.
- **Voices:** Determines the maximum number of voices that the actual instrument produces. Decreasing the number of voices improves efficiency and therefore reduces load on CPU. If voice count is set too low, then “voice stealing” can occur with the possibility of audible artifacts.



Please keep in mind that the chosen number of unison voices will respectively add more CPU usage.

4.5 Effects Section

Here you find all parameters regarding the four effect busses. For each effect bus, one of 9 different effect types can be selected.



Equalizer

RetroMod Lead offers a four-band equalizer. You can edit the equalizer by using either the controls or the response graphic.



- **Low/High Band Selection:** Selects the filter type of the first and last band. A **Shelf** filter type looks like a normal or an inverted “S” and it sounds and works similar to a low or high pass filter with adjustable stop band attenuation or boost. A **Bell** filter looks like a normal or inverted bell and sounds and works similar to a band pass or notch filter, again with variable boost or attenuation and also with variable width. A **Cut** filter is nearly similar to the Shelf filter but with a sharper slew rate.
- **Frequency (Freq):** Controls the cutoff (shelf) or center (bell) frequency of the band. Moving the dial to the left results in a

lower frequency of the band's cutoff or center frequency, to the right increases it.

- **Quality (Q):** Controls the quality or width of the band. When using a bell type, moving the dial upwards results in a narrower band, moving it downwards widens it. With the shelf type, it emphasizes or attenuates the frequency band around the cutoff frequency.
- **Gain:** Controls the gain boost or cut of the band. Clicking and moving the dial upwards increases the gain of the band, moving it downwards decreases it.

To edit the equalizer settings within the graphic, drag the desired button and move it to the left or the right to change the frequency. Move it up or down to change the gain. The small second button is for editing the quality of the desired band.

The Effect Types in Detail

Click on one of the corresponding buttons (**Effect 1** to **Effect 4**) to activate it. The corresponding button will light orange.

Use the pop-up menu to select the desired effect. Based on the selected effect type, different editing parameters are available.

Distortion

This effect offers an overdrive, also known as saturation or distortion. It affects the sum of all voices routed into this effect bus, which creates a different distortion depending on the number of voices you play due to intermodulation.

- **Type:** Selects the **Drive Type**. Light, Medium and Hard are typical diode clipping circuitries in different strengths and symmetries. Clip is a hard clipping similar to that of a transistor distortion, Tube simulates a tube or valve overdrive and Fuzz results in a very strong distortion.
- **Drive:** Controls the amount of overdrive.
- **Post Gain:** Controls the output level of the distorted signal.
- **Tone:** Controls high-frequency attenuation of the drive signal.
- **Emphasis:** Controls the feedback of the **Tone** parameter.



Chorus

The Chorus multiplies the signal by generating delayed copies of it and slightly modulating the speed with which these delayed signals are played back resulting in a detune effect. The signal then sounds like an ensemble or, as the name says, a chorus. **Speed** and **Depth** work hand in hand in: when you increase Speed, RetroMod Lead automatically decreases the depth in the chorus so that the detune effect stays the same. If you want to change the detune, adjust **Depth**.

- **Mix:** Adjusts the relation between the dry signal at the full left to the wet effect signal at the full right.
- **Stages:** RetroMod Lead offers three chorus types with either 2, 4 or 6 independent delay lines, where half of them is routed to

the left input and output and the other to the right input and output.

- **Delay:** Sets the minimum delay length. For a chorus effect, you typically increase it to around a tenth of its value range, for a flanger effect, you would set it to 0. The rest of the value range is just for your pleasure.
- **Feedback:** Controls the level of the signal that is fed back into the chorus. With a small setting of **Delay** and **Depth** and a high positive or negative setting of **Feedback** you can create a typical flanger effect, especially with 2 Stages.
- **Auto Depth:** Adjusts the strength of the modulation automatically to minimize artifacts. **Mod Depth** will be reduced all the more, the higher the settings for **Speed**.
- **Speed:** Controls the basic modulation speed. When using 4 Stage or 6 Stage, the 3rd and all following delays are modulated with a slightly higher or lower speed, depending on the setting of **Spread**.
- **Depth:** Adjusts the strength of the modulation, resulting in the detune effect. When using 4 Stage or 6 Stage, the depths of the 3rd and all following delays are slightly longer or shorter.
- **Spread:** Controls the deviation of the various delays. When **Spread** is set to full left, all modulations happen in sync albeit the 3rd and following stages are offset in phase to let the chorus still sound rich when using 4 Stage or 6 Stage. When you increase Spread, the delay deviations are increasingly stronger.
- **Random:** Adds a random movement to the **Spread** parameter. It produces linear ramps that adapt to **Speed**. This makes sure that the Chorus effect sounds not that static.



Phaser

The Phaser creates 2, 4, 6, 8, 10 or 12 phase-shifted signal copies creating equally spaced peaks and dips in the frequency spectrum. These phase-shifters are modulated in a way that a sweeping effect is created.

- **Mix:** Adjusts the relation between the dry signal at the full left to the wet effect signal at the full right.
- **Stages:** Sets the number of stages that are used for creating the phase-shifted signal copies.
- **Frequency:** Sets the center frequency of the phaser. Note that **Depth** and **Spacing** both influence the resulting frequencies of the phase shifters.
- **Spacing:** Controls the relative frequencies of the phase shifters. This spreads the peaks and dips in the frequency spectrum



- **Feedback:** Controls the level of the signal that is fed back into the phaser.
- **Modulation Speed:** Controls the modulation speed
- **Modulation Depth:** Sets the depth of the modulation.
- **Modulation Spread:** Controls the deviation between the left and right stereo side.

Compressor

The Compressor reduces signals which exceed a certain **Threshold** level in volume by an adjustable **Ratio**. The speed of this level reduction is controlled by **Attack**, when the signal starts to exceed the threshold, and **Release**, when it drops below the threshold again. When you set its parameters to extreme values, the signal might start to „pump“, so use your ears to set the values properly. This pumping effect might even be wanted, so use it to your liking. However, extremely short **Attack** or **Release** settings might create audible noise in the signal, which happens because the compressor changes its settings far in the audio range rather than creating a smooth envelope.

- **Gain:** Sets an input gain offset. Sometimes, the input signal is a little bit too low or loud, which you can change with **Gain**.
- **Threshold:** Sets the threshold above which the Compressor starts to reduce the signal level.
- **Make Up Gain:** Sets the output gain offset. A compressor reduces loud signals and **Make-up Gain** allows you to boost up this reduction back to a nice level.
- **Ratio:** Sets the compression ratio. 1:1 means that no compression takes place, while higher values result in an increasingly stronger compression.
- **Attack:** Sets the speed of the level reduction when the signal exceeds the **Threshold** level.
- **Release:** Sets the speed of the return to normal level when the signal drops below the **Threshold** level again.
- **Soft Knee:** With the setting of 0, the signals above the threshold is compressed instantly according to the set ratio (hard knee). Higher values let the onset of compression sound more gradual while producing a less drastic result.
- **Characteristic Diagram:** Displays the characteristic curve of the compression based on **Threshold** and **Ratio**.



Filter

The Filter effect filters the sum of all voices routed into this effect bus. Please refer to the chapter “Filter/Amp” for a thorough description of the various filter types.

- **Type:** Selects the filter type (LP6/12/24, HP 6/12/24, BP12/24, BS 12/24, Comb, Redux)
- **Cutoff:** Controls the cutoff (on low pass or high pass types) or center (on band pass or band stop types) frequency of the filter.
- **Resonance:** Controls the resonance of the filter if set to any of the 12 dB or 24 dB filter types. On 6 dB filter types, this control does nothing. On band stop types, it narrows the stop band, which is almost inaudible.



Delay

RetroMod Lead offers a time-based and a clock-based stereo modulation delay. Stereo means that the left and the right signals are processed independently, while modulation means that the Delay can slightly or heavily detune its output periodically making the signal richer or creating special effects.

- **Mix:** Adjusts the relation between the dry signal at the full left to the wet effect signal at the full right.
- **Time:** Sets the delay time in milliseconds. This parameter is only editable when **Clock** is deactivated.
- **Clock:** Activate it, to set the delay time in clock divisions synchronized to the current tempo.
- **Feedback:** Controls the level of the signal that is fed back into the delay.
- **Spread:** Adjusts the relative time of the first repetition compared to the delay length. A value of 0 % means that the delay of both stereo sides produce their first repetition at the same time while smaller values result in the left side repeating first and larger values result in the right side repeating first. This allows you to create ping-pong delays. E.g., when setting **Clock** to 1/4 and **Spread** to -50 % you get an 1/8 ping pong delay. Other interesting **Spread** percentages are 33 % for triplets or 350 % for dotted notes, provided that **Clock** is set to a straight value. If **Clock** is set to a dotted value, you would, for example, set **Spread** to 333 % to have one of the repetitions straight. Also interesting are only slight deviations from 0 %, creating deeper delays. Sounds complicated, but offers a lot of fun.
- **Mod Speed:** Controls the modulation speed.
- **Low Cut:** Removes low frequencies before the signal enters the delay stage so that the first repetition is already filtered.



- **High Cut:** Removes high frequencies before the signal enters the delay stage so that the first repetition is already filtered.
- **Mod Depth:** Adjusts the strength of the modulation, resulting in a detune effect.
- **Low Damp:** Removes low frequencies in the feedback stage.
- **High Damp:** Removes high frequencies in the feedback stage of the delay meaning that each repetition has less and less high frequencies.

Reverb Natural / Reverb Plate / Reverb Non-Linear

RetroMod Lead features high-quality reverb algorithms in different flavors. They all share common parameters.

The reverb natural algorithm simulates the reverberation that occurs in natural rooms. The names suggest the basic size of the simulated room although you can create artificially sounding rooms by changing the reverb time to unnatural values.

The Reverb Plate algorithm simulates a reverb plate. In the late 1950's, the German company EMT (Elektro-Mess-Technik) introduced huge steel plates fitted with a loudspeaker and a microphone. A signal would be sent to the loudspeaker and recorded by the microphone including the vibrations the steel plate generated resulting in a smooth but slightly artificial sounding reverb. The Plate Reverb and its digital simulation is suited well for drum and percussion sounds due to their dense initial reflections.

The non-linear reverb algorithm generates typical gated reverb effects (although it works completely differently) or inverse reverbs. Since it doesn't have an internal feedback it must produce the whole reverb in one turn to create the gate effect. This means that longer reverb times sound less dense than shorter ones. If the reverb starts to sound not dense enough, you might try to increase the Decay of the sound, maybe a snare drum, by a little to smooth it out.



- **Mix:** Adjusts the relation between the dry signal at the full left to the wet effect signal at the full right.
- **Predelay:** Sets the delay before the signal enters the reverb stage. Typically you set it to around a hundredth to a tenth of the reverb **Decay**. **Predelay** strongly affects the perceived size of the reverb effect since it controls the time until the first reflection is generated.
- **Size:** Sets the length of the room size. Lower settings simulate a small room while higher settings simulate a big hall or church.
- **Decay:** Sets the time the reverb needs to decay.
- **Slope:** Controls the slope of the reverb. Setting it to center results in a typical gated reverb where the reverb sound stays equally loud before it is cut off abruptly. Lower values let the reverb decay slightly while higher values result in an inverse reverb which gets louder over its time.

- **Density:** Controls the initial density of the reverb. Lower values result in audible early reflections while higher values produce a dense reverb start.
- **Diffusion:** Controls the density or diffusion of the reverb the longer it sounds. Lower values result in more audible individual reflections while higher values result in a dense reverb tail. Note that even a setting of 0 will still create a reverb becoming more and more diffuse.
- **Low Cut:** Removes low frequencies before the signal enters the reverb stage so that the reverb sound is filtered.
- **High Cut:** Removes high frequencies before the signal enters the reverb stage so that the reverb sound is filtered.
- **Low Damp:** Removes low frequencies in the feedback stage of the reverb meaning that the low frequencies are dampened over time.
- **High Damp:** Removes high frequencies in the feedback stage of the reverb meaning that the high frequencies are dampened over time.

5. Product Support

If you have any questions, please contact us by email:

support@tracktion.com

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