GUITAR RIG 6

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1. Disclaimer

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Software version: 6.2 (04/2021)

2. Welcome to GUITAR RIG

GUITAR RIG 6 is a multi-effects rack and amp simulator made for creating and experimenting with audio in a way that is fast and direct. Think of it as your own studio, only with more space, less heavy amp heads, and way more flexibility. Design unique processing chains to customize your tones, adding space, warmth and character to everything from guitar and bass, to strings, drums, synths and more.

GUITAR RIG makes its wealth of effects accessible through a clear layout that focuses on the Rack. In the Rack you can combine and tweak the individual effects, called Components, to create any multi-effect imaginable. The Rack is accompanied by the Browser, which enables you to find presets and load Components from GUITAR RIG's extensive library.

It does not stop here, however. GUITAR RIG will grow, adapt, and evolve with regular updates – both inspired by, and to inspire its users. We hope you enjoy this fantastic multi-effects rack as much as we do, and are looking forward to hearing your creations!

3. Document Conventions

In this document the following formatting is used to highlight useful information:

Italics	Indicates paths to locations on your hard disk or other storage devices
Bold	Highlights important names, concepts, and software interface elements.
[Brackets]	References keys on a computer's keyboard

The following three icons represent different types of information:

The **light bulb** icon indicates a useful tip, suggestion, or interesting fact.

The **information** icon highlights important information that is essential for the given context.

The **warning** icon alerts you of serious issues and potential risks that require your full attention.

4. Getting Started

GUITAR RIG is used to process audio from sources like a guitar, a bass, or any other instrument that you want to experiment with, be it acoustic or electronic. For this purpose, GUITAR RIG needs to receive the audio signal from the source, and make the processed signal available for recording, playback, and listening at its output.

Setting up GUITAR RIG accordingly differs between the stand-alone application and the plug-in. The stand-alone application runs on its own and can be opened from the Native Instruments folder in your applications directory. The plug-in can be loaded in a compatible host software, most commonly a DAW.

Using the stand-alone application provides the most efficient way of playing an instrument through GUITAR RIG. If the computer is not used for recording, editing, and arranging the music, the stand-alone application turns it into a powerful, dedicated effects rack to process your instrument. To setup the stand-alone application, you need to configure its Audio settings in the Preferences. For more information, see Setting up GUITAR RIG as Stand-Alone Application.

Using the plug-in fully integrates GUITAR RIG into your DAW, making it an integral part of recording, editing, and arranging your music. Synchronization and saving are handled by the DAW, and you benefit from your DAW's automation functionality. To setup the plug-in, you need to make it available to your DAW, load it into a track, and route your instrument to the track. For more information, see Setting up GUITAR RIG as Plug-in in a DAW and Using GUITAR RIG as Plug-in in a DAW.

4.1. Setting up GUITAR RIG as Stand-Alone Application

The GUITAR RIG stand-alone application turns your computer into a powerful, dedicated effects rack to process your instrument. To setup the stand-alone application, you need to configure its Audio settings in the Preferences:

1. Click the Main menu in the Header and open **Preferences...** in the **File** sub-menu.
2. In the Preferences, go to the **Audio** tab and click on **Inputs** in the lower section.
3. Choose the inputs of the audio interface that you have used to connect your instrument in the **Guitar Rig 6 In L** and **Guitar Rig 6 In R** drop-down menus.

   If you are using a mono source, like a guitar or a bass, you can use either the left or right input.

4. Click on **Outputs** in the lower section of the **Audio** tab.
5. Choose the outputs of your audio interface that are connected to your speaker system or headphones in the **Guitar Rig 6 Out L** and **Guitar Rig 6 Out R** drop-down menus.
6. Close the Preferences by clicking on the **X** icon in the upper-right corner.

→ The GUITAR RIG stand-alone application is setup and ready to process your instrument.

The Input selector in the GUITAR RIG Header needs to be set accordingly in order to feed the audio signals received at the inputs into the Rack. For more information, see Signal Flow.

4.2. Setting up GUITAR RIG as Plug-in in a DAW

The GUITAR RIG plug-in fully integrates into your DAW, making it an integral part of recording, editing, and arranging your music. It can be loaded in any DAW supporting the VST, AU, and AAX plug-in formats. All three formats are automatically included when installing GUITAR RIG using Native Access.

> On Windows computers, the correct path for VST plug-ins needs to be selected in Native Access prior to installation. Learn more in this support video on our website: VST Plug-in Administration on Windows Computers

To setup the plug-in for use in your DAW, you need to load it on an audio track and configure the track's routing:

1. Open your DAW after installing GUITAR RIG using Native Access.
2. Create a new audio track.
3. Choose the inputs of your audio interface that you have used to connect your instrument in the audio track's input routing.

   > Alternatively, you can load audio recordings or software instruments and use the GUITAR RIG plug-in to process them.

4. Load the GUITAR RIG plug-in in the audio track.
5. Activate the record-ready or monitoring state of the audio track.

→ The GUITAR RIG plug-in is setup and ready to process your instrument.

> The Input selector in the GUITAR RIG Header needs to be set accordingly in order to feed the audio signals received at the inputs into the Rack. For more information, see Signal Flow.

4.3. Using GUITAR RIG as Plug-in in a DAW

When using the GUITAR RIG plug-in, keep in mind the following notes about plug-in behavior:

- The plug-in and the stand-alone application share the same presets. When you save a user preset, it will be available in either version of the software. For more information, see Saving User Presets.
- Any changes made in the plug-in are saved as part of the DAW's project file. When you change a preset, you only need to save your project in the DAW in order to restore the changes later.
- The Metronome in the plug-in, and therefore all tempo-related functions, are synchronized to the tempo of your DAW. You can change the Sync mode in the Metronome. For more information, see Metronome.
- The Macros can be used for MIDI control and automation independently of individual presets. When you use Macros to write automation or assign a MIDI controller, the changes will be effective for any preset you load. For more information, see Automation and MIDI Control.

5. Overview of GUITAR RIG

GUITAR RIG makes its wealth of effects accessible through a clear layout that focuses on the Rack. In the Rack you can combine and tweak the individual effects, called Components, to create any multi-effect imaginable. The Rack is accompanied by the Browser, which enables you to find presets and load Components from GUITAR RIG's extensive library. The Header provides global controls and settings.

The following overview highlights where to find the Rack, Browser, and Header in the user interface:

1. **Header**: Provides global controls and settings related to preset management and editing, basic preferences, the magnification of the user interface, the audio inputs and outputs, and the audio engine. For more information, see Header.
2. **Rack**: Facilitates the creation of custom multi-effects based on Components. The required infrastructure is provided using dedicated Rack Tools, which are accessible through the Rack's Toolbar. For more information, see Overview of the Rack.
3. **Browser**: Gives access to GUITAR RIG's extensive library, including presets and Components. Sophisticated browser functionality like Favorites, Filters, and the Search field enable you to quickly find the right content. For more information, see Overview of the Browser.

6. Global Controls

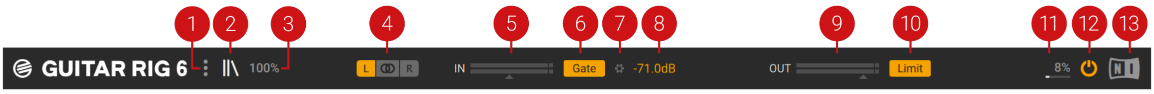
The Header at the top of GUITAR RIG provides you with functions related to preset management and editing, software behavior, as well as the audio input and output. You can access these functions using dedicated controls and via the Main menu on the left.

6.1. Header

The Header at the top of GUITAR RIG contains the Main menu, audio input and output controls, and functions related to the audio engine. Below you can find an overview of the controls and settings.

> The GUITAR RIG stand-alone application retains settings made in the Header from the last session. The plug-in uses default settings, which can be overwritten by using the option in the Preferences. For more information, see General.

The following overview shows you the controls and settings in the Header:



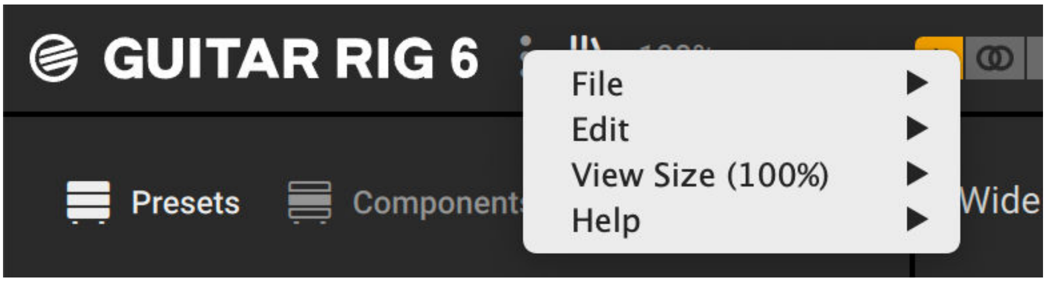
1. **Main menu**: Opens the Main menu that contains global options and preferences. For more information, see Main Menu.
2. **Browser**: Shows or hides the Browser on the left side of the user interface. For more information, see Overview of the Browser.
3. **View size**: Sets the magnification of the user interface, and gives the option to save the current size as default. For more information, see View and Rack Size.
4. **Input selector**: Selects the input configuration for the Rack. You can select select either the left or right input in a mono configuration, or a stereo configuration using both inputs. For more information, see Signal Flow.
5. **Input Level**: Adjusts the input level of the Rack in the range of -20 dB to +20 dB, and shows the current peak input level. Clipping is indicated by the red dot at the right of the level meter.
6. **Gate**: Activates the noise gate that is used to suppress unwanted background noise. The input signal is gated when it falls below the threshold as set with the Gate Threshold control.
7. **Threshold Learn**: Adjusts the Gate Threshold automatically by analyzing the input signal and determining the optimal threshold for separating the direct instrument signal from the background noise. For best results, play softly while the analyzation is in progress.
8. **Gate Threshold**: Adjusts the threshold level of the gate. When the input signal falls below threshold, the level is fully attenuated, or gated. Increasing the threshold results in a higher amount of gating by making the gate less sensitive to low input levels.
9. **Output Level**: Adjusts the input level of the Rack in the range of -70 dB to +6 dB, and shows the current peak output level. Clipping is indicated by the red dot at the right of the level meter.
10. **Limit**: Activates the output limiter that is used to prevent digital clipping. The output signal is limited when it rises above the maximum level of 0 dBfs.
11. **CPU Load**: Shows the current CPU load used by GUITAR RIG's audio engine.
12. **Audio Engine On/Off**: Switches the audio engine on or off.

13. **NI Logo**: Opens the About screen that shows the **Credits** and **User Details** including the version, license type, and serial number of the installed software.

6.2. Main Menu

The Main menu in the Header contains options for managing and editing presets, setting preferences, adjusting the magnification of the user interface, and accessing external resources.

The following overview shows you the sub-menus of the Main menu:



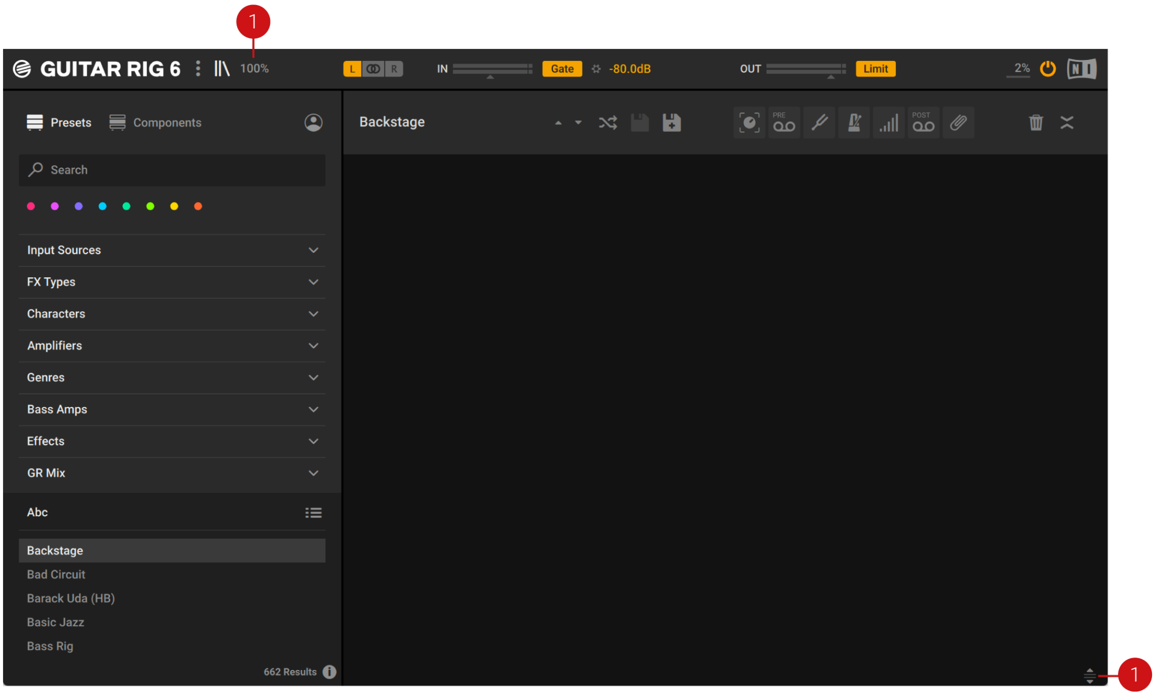
- **File**: Contains options for creating, importing, and saving preset files. Additionally, you can open the Preferences from this sub-menu. For more information, see Using Presets and Preferences.
- **Edit**: Contains options for editing the Rack. You can cut, copy, paste, and delete Components, select them all, or undo and redo your edits. For more information, see Editing the Rack.
- **View Size**: Contains options to set the magnification of the user interface, and gives the option to save the current size as default. For more information, see View and Rack Size.
- **Help**: Contains options to access external resources including Native Access, the latest version of this manual, the NI Knowledge Base, and the product pages.

6.3. View and Rack Size

You can customize GUITAR RIG's visual presentation by adjusting the magnification of the user interface and the vertical size of the Rack.

> The GUITAR RIG stand-alone application retains the View size setting from the last session. The plug-in uses the default setting, which can be overwritten by using the option in the View size menu.

The following overview shows you the available options:



1. **View size**: Sets the magnification of the user interface, and gives the option to save the current size as default. The magnification can be set in 9 steps from 75% to 200%.
2. **Rack size**: Adjusts the vertical size of the Rack by dragging the icon up and down. When used in conjunction with the View size, you can optimize the visual presentation for visibility (large View size, small Rack size) or Rack space (small View size, large Rack size).

7. Signal Flow

GUITAR RIG is a modular effects processor that combines individual effects, called Components, in an effects chain. This effects chain is hosted in the Rack, where you can stack components on top of each other. The signals are passed in the Rack from top to bottom.

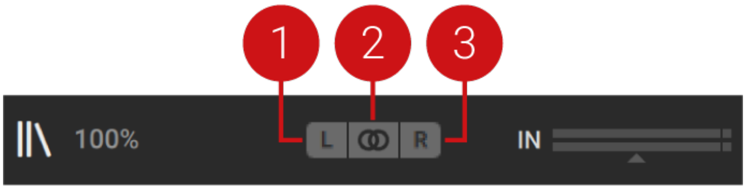
The inputs are sent to the first Component at the top, and the outputs are taken from the last Component at the bottom. The Rack's outputs are sent to the Global FX before being passed on to the final outputs of GUITAR RIG. For more information, see Global FX.

The Rack has a stereo input and a stereo output. In the stand-alone application, you can assign any of the inputs and outputs of your audio interface to the Rack's stereo input and output. In the plug-in, the inputs and outputs depend on where it is inserted in the DAW.

For more information about assigning inputs and outputs in the stand-alone application, see Setting up GUITAR RIG as Stand-Alone Application.

7.1. Input Selector

The Input selector in the Header enables you to select the input configuration for the Rack. The following options are available:



1. **Left Input**: Only the left stereo input is used in a mono configuration. Components using stereo processing receive the signal from the left stereo input on both stereo channels.
2. **Stereo Input**: Both the left and right stereo input are used in a stereo configuration. Components using stereo processing receive the signals from the left and right stereo inputs on the respective stereo channel.

   When Stereo Input is selected, the Stereo option in the Component Controls is activated automatically for all relevant Components. For more information, see Component Controls.

3. **Right Input**: Only the left stereo input is used in a mono configuration. Components using stereo processing receive the signal from the right stereo input on both stereo channels.

7.2. Mono and Stereo Processing

Most Components use stereo processing from input to output, which means that generally stereo signals are preserved throughout the Rack. Certain Components provide an option in the Component Controls that switches between mono and stereo processing. You can use this to reduce the CPU load by deactivating stereo processing. For more information, see Component Controls.

However, there are also Components that always sum the left and right stereo inputs to mono. Nevertheless, their output can be stereo, depending on the the processing applied by the Component.

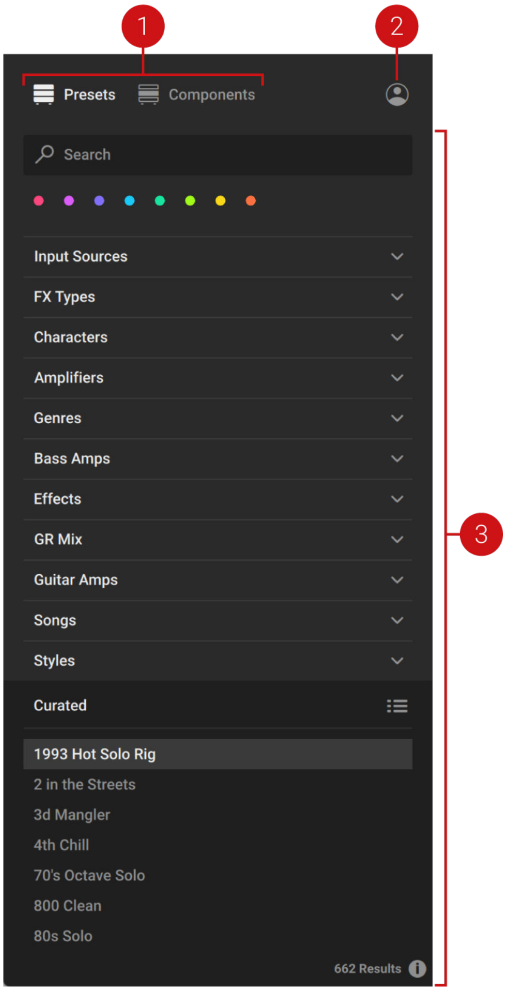
The following Components sum the stereo inputs to mono:

- Cabinets & Mics (mono output)
- Psyche Delay (stereo output)
- Quad Delay (stereo output)
- Flanger Chorus (stereo output)
- Rotator (stereo output)
- Stereo Tune (stereo output)
- Octaverb (stereo output)
- Litte REFLEKTOR (stereo output)
- RAUM (stereo output)
- REFLEKTOR (stereo output)
- TRAKTOR's Reverb (stereo output)

8. Overview of the Browser

The Browser gives access to GUITAR RIG's extensive library, including presets and Components. Sophisticated browser functionality like Favorites, Filters, and the Search field enable you to quickly find the right content.

The following overview shows you the basic structure of the Browser:

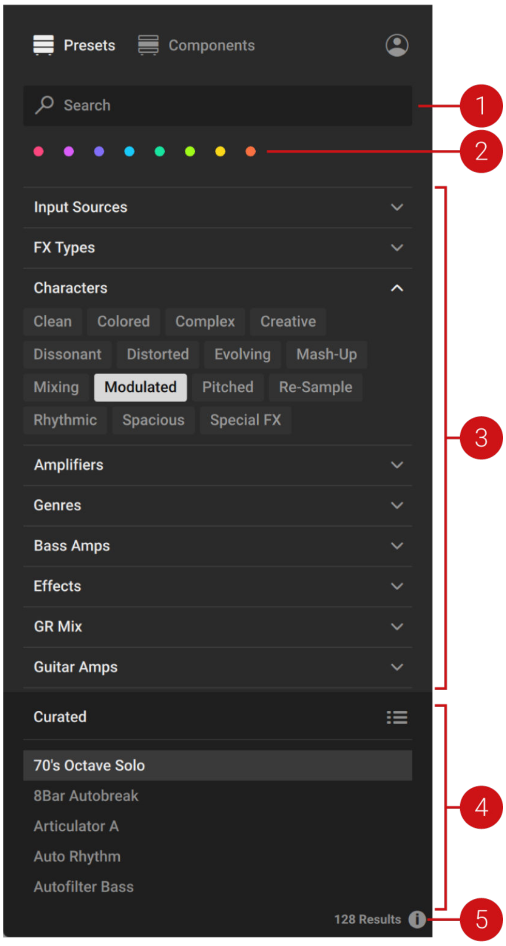


1. **Content selector**: Selects the type of content to be shown in the Browser interface. The following types of content are available in GUITAR RIG:
   - **Presets**: A Preset recalls a previously saved Rack including its Components, all individual parameter settings, and the options set in the Toolbar. For more information about the Browser interface for presets, see Presets in the Browser.
   - **Components**: A Component represents an individual effect that can be added to a Rack. Component presets recall previously saved parameter settings for specific Components. For more information about the Browser interface for Components, see Components in the Browser.
2. **User Content**: Filters by user content. When activated, the Browser interface only shows content created by the user. If the Content selector is set to Presets, only User presets are shown in the Results list. If the Content selector is set to Components, only Components that have associated User Component presets are shown in the Component Tiles. When deactivated, the Browser interface shows both library and user content.

3. **Browser interface**: Shows the content selected using the Content selector. The elements in the Browser user interface adapt to the type of content, in order to facilitate the best possible browsing experience:
    - When **Presets** is selected, you can use the Search field, Favorites, Filters, and Results list to filter and browse for presets.
    - When **Components** is selected, you can use the Category Filter, Component Tiles, and the Component preset list to filter and browse for Components and their dedicated Component presets.

8.1. Presets in the Browser

The following overview shows you all the elements in the Browser that you can use to explore GUITAR RIG's extensive library of presets:

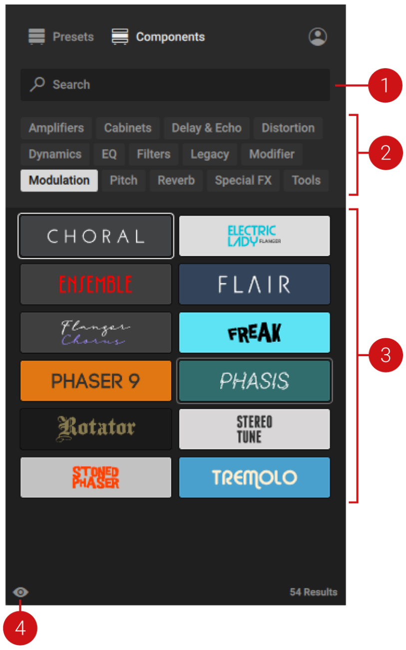


1. **Search field**: Finds presets matching the terms entered in the field and shows them in the Results list. For more information, see Using the Search Field.
2. **Favorites**: Filter the presets in the Results list by colored Favorite tags that you can use to create collections of your favorite presets. You can assign Favorite tags by using the context menu in the Results list. For more information, see Using Favorites.
3. **Filters**: Filter the presets in the Results list by Filter tags that are used to characterize and classify presets. The Filter tags are organized in groups of different Filter types. For more information, see Using Filters.

4. **Results list**: Shows presets according to the options set in the Search field, Favorites, and Filters. The Sorting options at the top provide different options for changing the order of presets. For more information, see Sorting the Results list.
5. **Info pane**: Opens the Info pane that you can use to view the Filter tags and properties of each preset. Additionally, the Info pane enables you to edit the Filter tags and properties for User presets, and add your own User Filter tags. For more information, see Editing User Presets in the Info Pane.

8.2. Components in the Browser

The following overview shows you all the elements in the Browser that you can use to explore GUITAR RIG's extensive library of Components:



1. **Search field**: Finds Components matching the terms entered in the field and shows them in the Component Tiles. For more information, see Using the Search Field.
2. **Category Filter**: Filters the Component Tiles by Filter tags that are used to categorize the Components. For more information, see Using Filters.
3. **Component Tiles**: Represent the individual effects available in GUITAR RIG, called Components. You can use the Component Tiles to browse for Components and add them to your Rack. For more information, see Adding Components to the Rack.
4. **Show Component presets**: Shows the dedicated presets for the Component selected in the Component Tiles. For more information, see Using Component Presets.

9. Using the Browser

The Browser gives access to GUITAR RIG's extensive library by enabling you to quickly find a specific preset or Component, as well as explore groups of presets or Components based on your aesthetic and stylistic preferences.

Utilizing the concept of progressive disclosure, each element in the Browser interface contributes to a fluent workflow that continuously filters the Results list or Component Tiles in a meaningful way.

When browsing for presets, you can combine the Search field, Favorites, and Filters to narrow down the presets in the Results list. When browsing for Components, you can combine the Search field and Category Filter to narrow down the Component Tiles.

The following sections explain how to use each of the individual elements in the Browser interface, including the Search field, Favorites, Filters, and the Sorting options of the Results list.

9.1. Using the Search Field

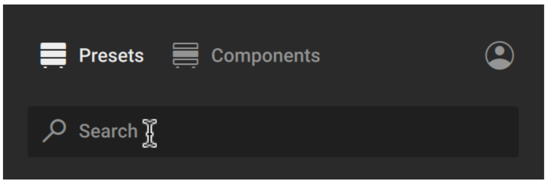
The Search field at the top of the Browser interface enables you to find presets or Components matching the terms entered in the field and shows them in the Results list.

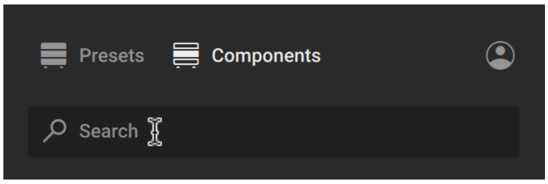
When **Presets** is selected in the Content selector, the search considers preset titles and metadata, including Filter tags and the author.

When **Components** is selected in the Content selector, the search considers Component titles, Category Filters, and the associated Component presets.

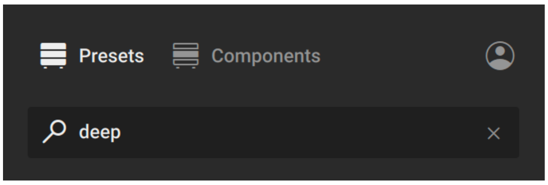
To find presets or Components using the Search field:

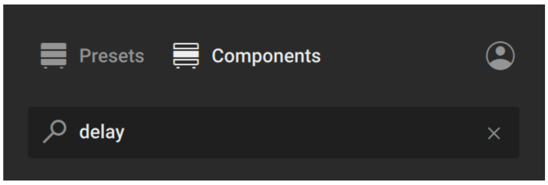
1. Activate the Search field by clicking it.



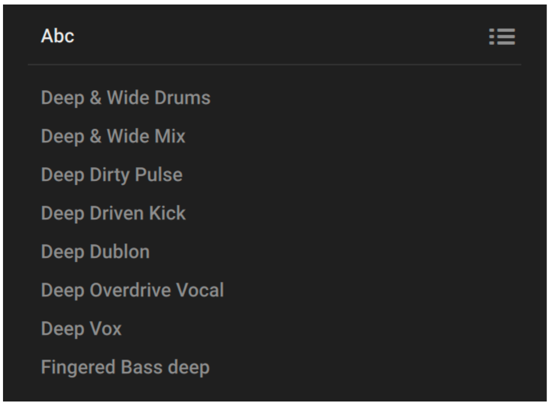


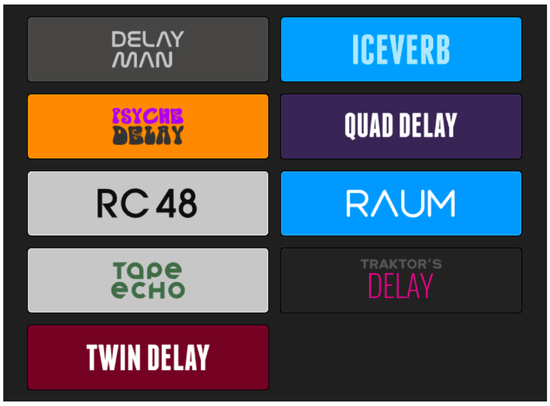
2. Enter your search terms.





→ The Results list or Component Tiles are filtered according to the terms entered in the Search field.





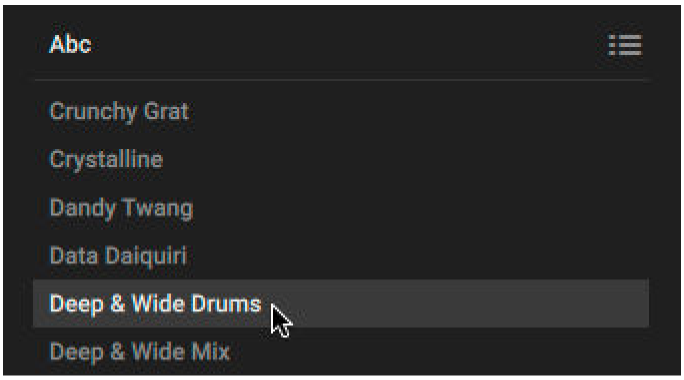
9.2. Using Favorites

The Favorites enable you to filter the presets shown in the Results list by colored Favorite tags. By assigning these tags to your favorite presets, you can create your personal preset collections and quickly access them.

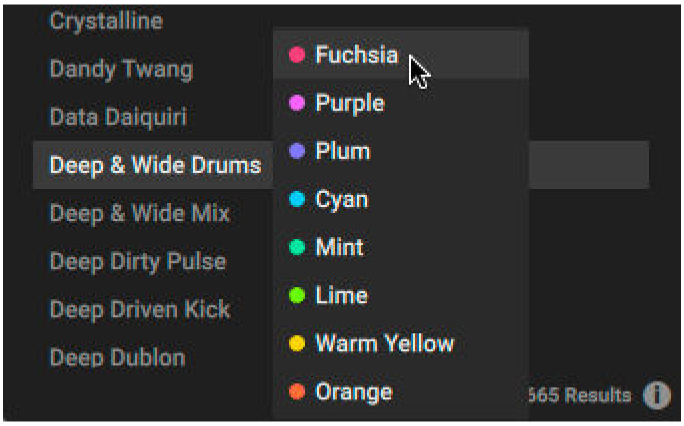
To assign a Favorite tag:

1. Right-click a preset in the Results list to open the context menu. If multiple User presets are selected, the changes will apply to all of them.

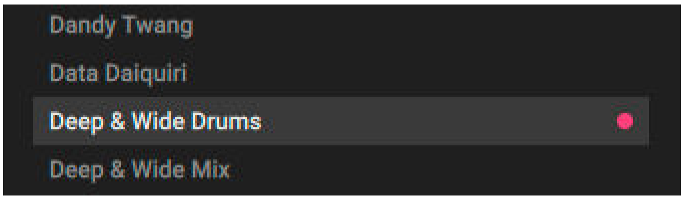
   ⓘ You can select a preset by clicking on it. Holding [SHIFT] on the keyboard allows for selection of multiple presets.



2. Select one of the colored Favorite tags from the context menu.



→ The selected Favorite tag is assigned to the preset and shown as a colored dot next to preset title.



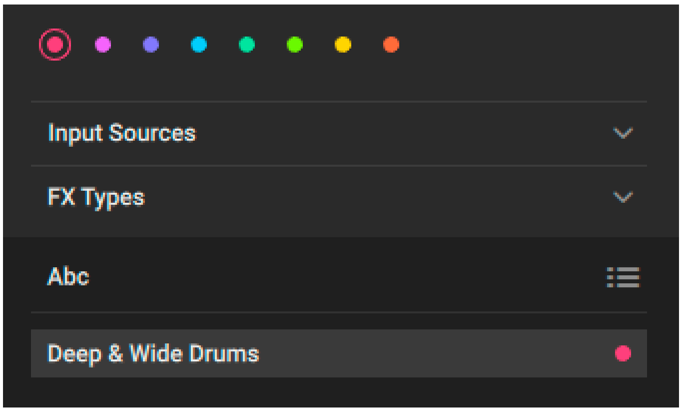
To remove an already assigned Favorite tag from a preset:

▶ Click on the corresponding entry in the context menu. If multiple User presets are selected, the changes will apply to all of them.

To filter the presets shown in the Results list by your previously assigned Favorite tags:

▶ Click a tag in the Favorites to select it.

→ The Results list is filtered according to your selection and only shows presets that have been assigned the respective Favorite tag.



Favorites shown in the Results list can be sorted in a custom order set by the user. For more information, see Sorting Favorites in a Custom Order.

9.3. Using Filters

The Filters enable you to filter presets or Components by Filter tags. These tags are used to characterize and classify presets based on their sound, use case, and other attributes. Components are tagged according to categories of effects. All presets and Components in the GUITAR RIG library are tagged in a meaningful way so you can start browsing right away.

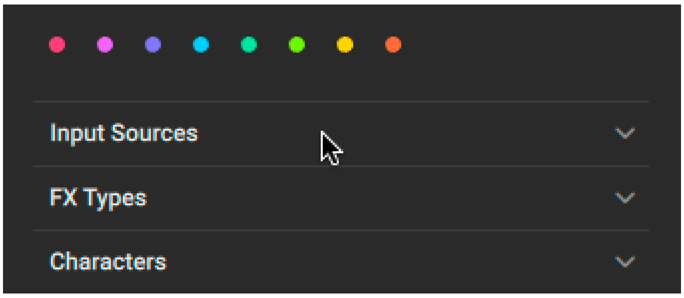
You can also assign Filter tags to your own User presets. For more information, see Editing User Presets in the Info Pane.

When browsing for presets, the Filter tags are organized in groups of different Filter types. When browsing for Components, the Filter tags are organized in the single Category Filter.

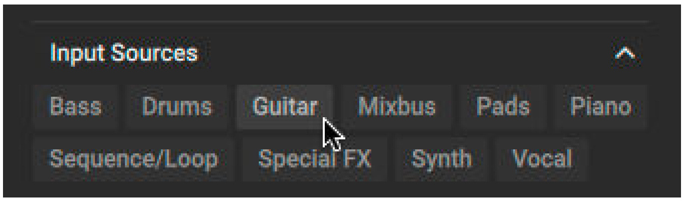
Using Filters for Presets

To filter the presets in the Results list using Filter tags from multiple Filter types:

1. Click a Filter type to open it and show the contained Filter tags.

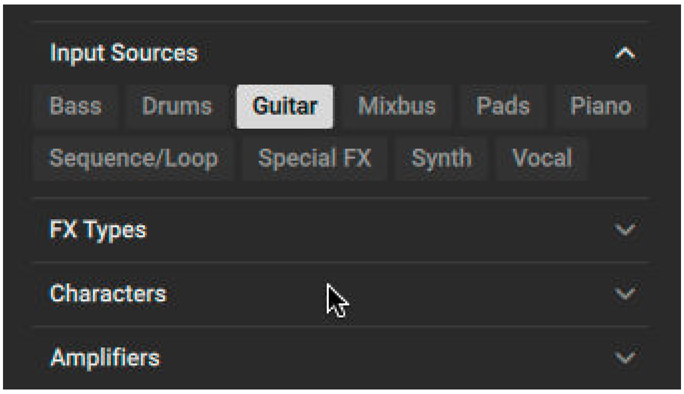


2. Click a Filter tag to start filtering the presets in the Results list.

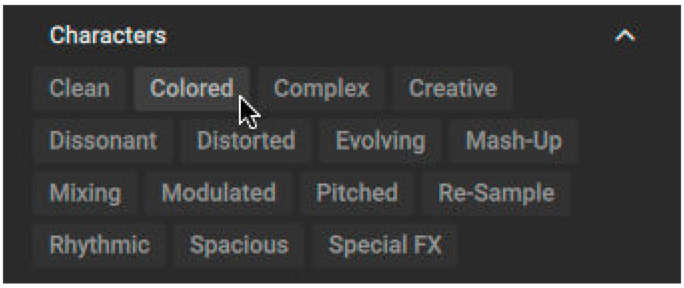


3. Click another Filter type to open it and show the contained Filter tags. The other Filter types will be closed automatically.

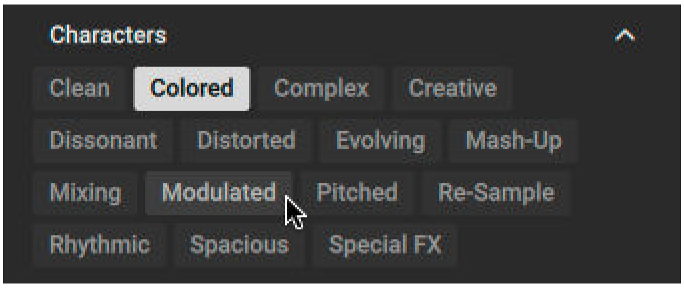
To show Filter tags for multiple Filter types at the same time, press [command] (macOS) or [Ctrl] (Windows) + click to open them.



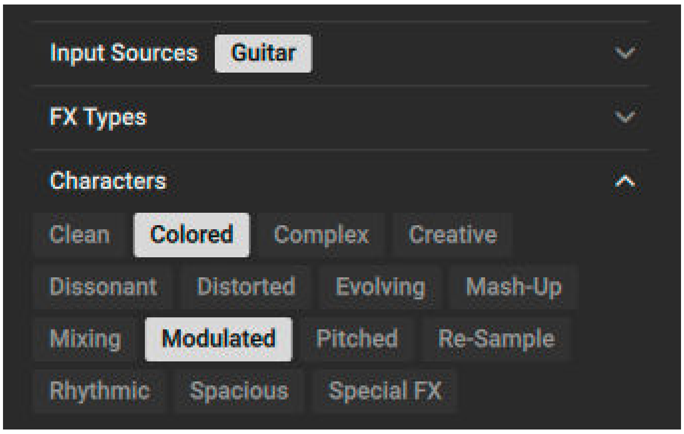
4. Click a Filter tag to continue filtering the presets in the Results list.



5. Press [command] (macOS) or [Ctrl] (Windows) + click to select another Filter tag from the same Filter type and further filter the presets in the Results list.



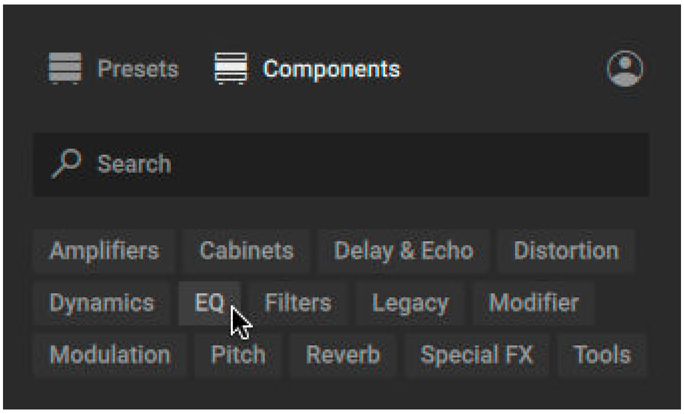
→ The presets in the Results list are filtered by the selected Filter tags. Only presets carrying all of the selected Filter tags are shown.



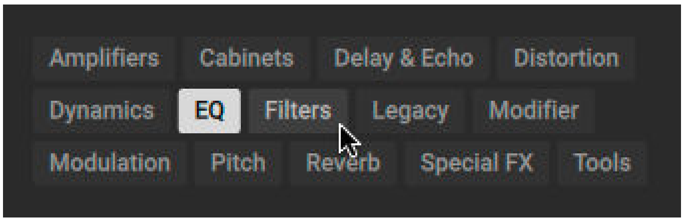
Using the Category Filter for Components

To filter Component Tiles using the Category Filter:

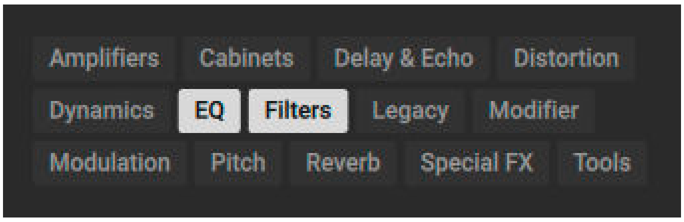
1. Click a Filter tag to start filtering the Components Tiles.



2. Press [command] (macOS) or [Ctrl] (Windows) + click to select another Filter tag and add Component Tiles from the respective category.



→ The Component Tiles are filtered by the selected Filter tags. All Components in any of the selected categories are shown.

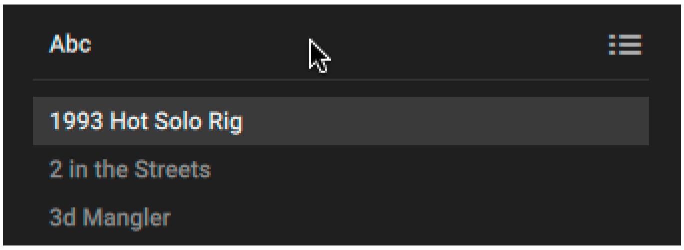


9.4. Sorting the Results list

The Results list provides different Sorting options that you can use to change the order of presets in the Results list.

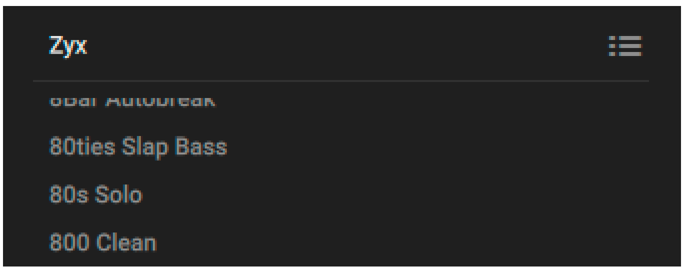
To change the order of presets:

1. Click the area at the top of the Results list to show the Sorting options.



2. Click on one of the options to select it.

→ The presets in the Results list are sorted according to the selected option.



The following Sorting options are available:

- **Curated**: Sorts the Results list in the order curated by the sound designers.
- **Abc**: Sorts the Results list in alphabetical order.
- **Zyx**: Sorts the Results list in reverse alphabetical order.
- **Custom**: Sorts the Results list in the custom order set for Favorites. For more information, see Sorting Favorites in a Custom Order.

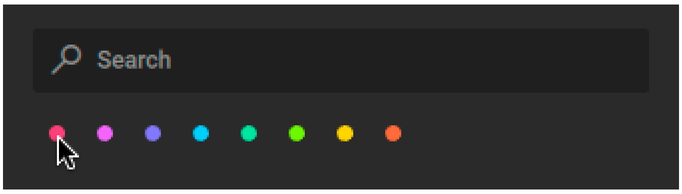
9.5. Sorting Favorites in a Custom Order

Favorites in the Results list can be sorted in a custom order set by the user. You can set a custom order for each of the eight Favorite tags available in the Browser.

> Each Favorite tag retains the last selected Sorting option. When switching between Favorite tags, the Sorting options change accordingly.

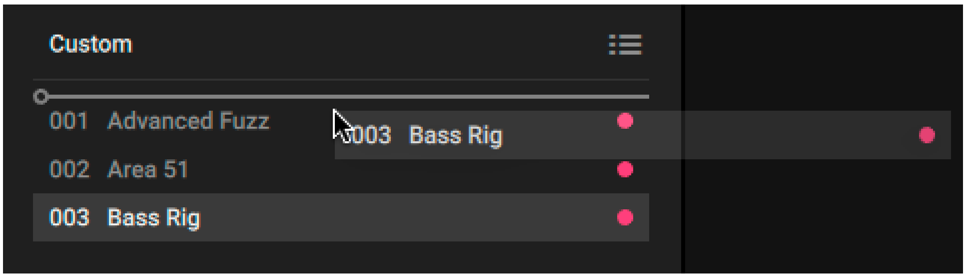
To sort the Favorites in the Results list:

1. Click a Favorites tag to show the corresponding presets in the Results list.

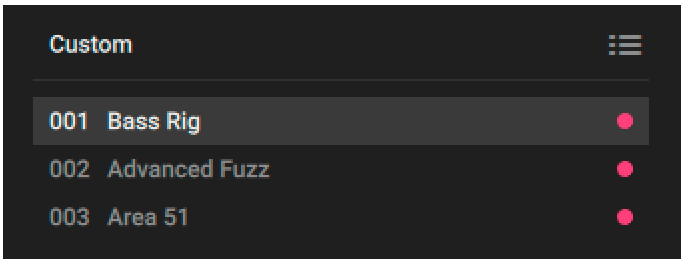


2. Select **Custom** in the Sorting options of the Results list.

3. Click, drag, and drop any of the presets into a new position in the Results list.



→ The Favorites in the Results list are sorted in your new custom order.



You can recall this custom order at any time by selecting **Custom** in the Sorting options when a Favorite tag is selected in the Browser. For more information, see Sorting the Results list.

10. Using Presets

A Preset recalls a previously saved Rack including its Components, all individual parameter settings, and the options set in the Toolbar. In addition to the presets included in GUITAR RIG's extensive library, you can save, load, and import User Presets. Furthermore Component presets enable you to save settings of individual Components. The following sections explain the basic workflows involved in using presets.

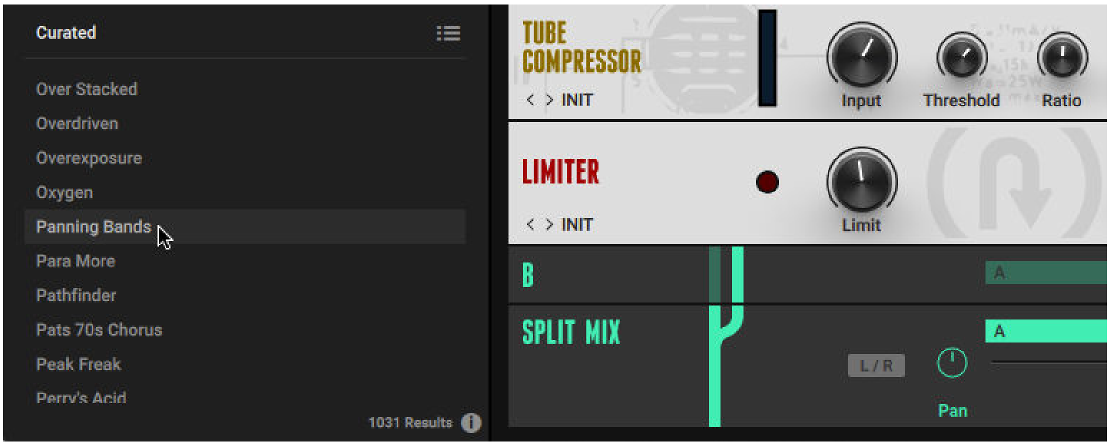
10.1. Loading Presets

You can load presets directly from the Browser's Results list.

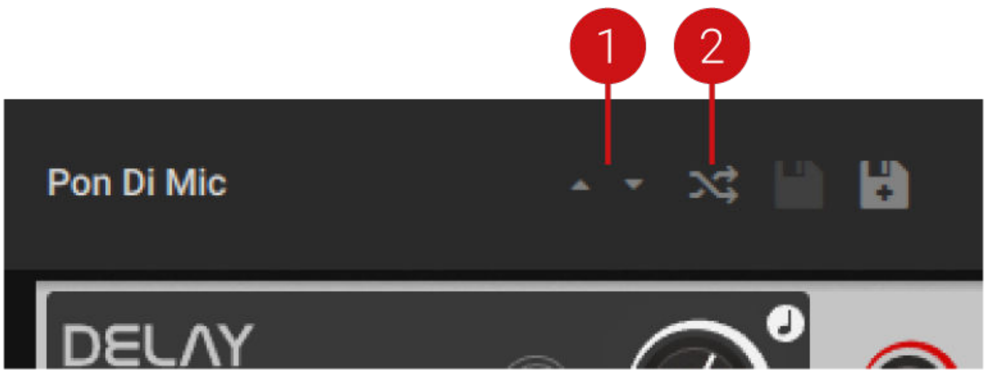
The Search field, Favorites, and Filters in the Browser enable you to narrow down the presets in the Results list. For more information, see Using the Browser.

▶ To load a preset, double-click on the corresponding entry in the Results list.

Alternatively, you can select the entry in the Results list using the mouse or the arrow keys on the keyboard and press [command] (macOS) or [Ctrl] (Windows) to load a preset. For more information, see Keyboard Shortcuts.



You can also use the Rack's Toolbar to load presets. The following overview shows you the available controls:



1. **Previous preset / Next preset**: Switches between presets in the Browser's Results list. Clicking on the upwards arrow loads the previous preset. Clicking on the downwards arrow loads the next preset.
2. **Preset Shuffle**: Loads a random preset from the Browser's Results list.

10.2. Saving User Presets

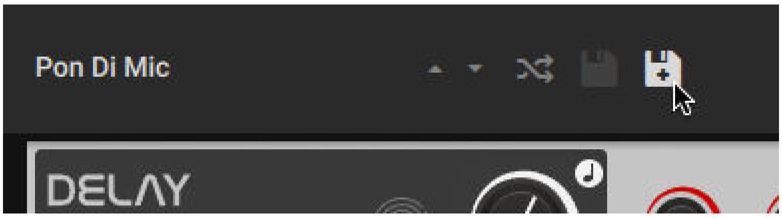
You can save User presets directly in the Rack's Toolbar.

Saving a New User Preset

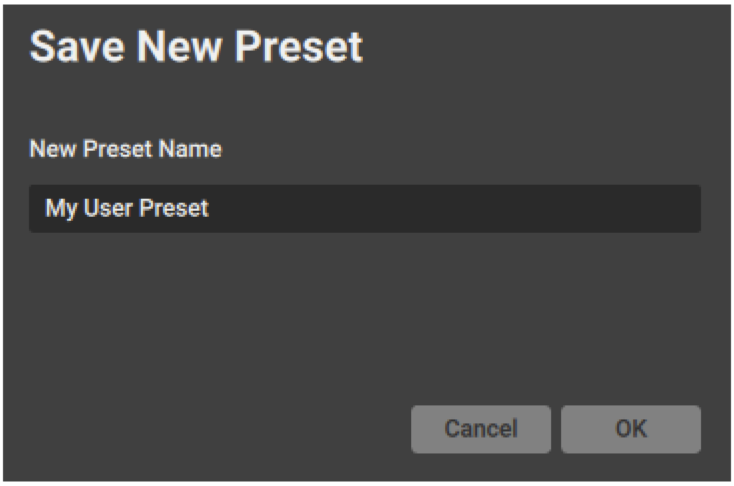
To save a new User preset:

1. Click the Save new preset button in the Toolbar.

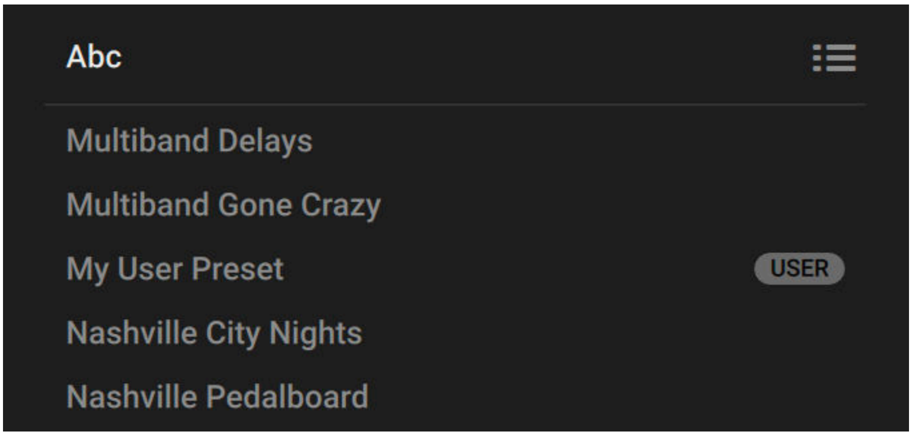
> Alternatively, you can press [command] + [shift] + [S] (macOS) or [Ctrl] + [Shift] + [S] (Windows) on the keyboard, or use the command from the Main menu. For more information, see Managing Presets Using the Main Menu.



2. Type in a name for your User preset and confirm by clicking **OK**.



→ The User preset is saved and a new entry is added to the Results list, highlighted by the **USER** label.



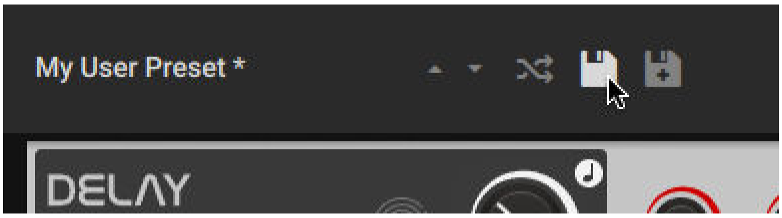
To quickly find your User presets, you can filter the Browser's Results list by User presets. For more information, Overview of the Browser.

Saving Changes in a User Preset

To save changes made in a User preset:

▶ Click the Save preset button in the Toolbar.

> Alternatively, you can press [command] + [S] (macOS) or [Ctrl] + [S] (Windows) on the keyboard, or use the command from the Main menu. For more information, see Managing Presets Using the Main Menu.



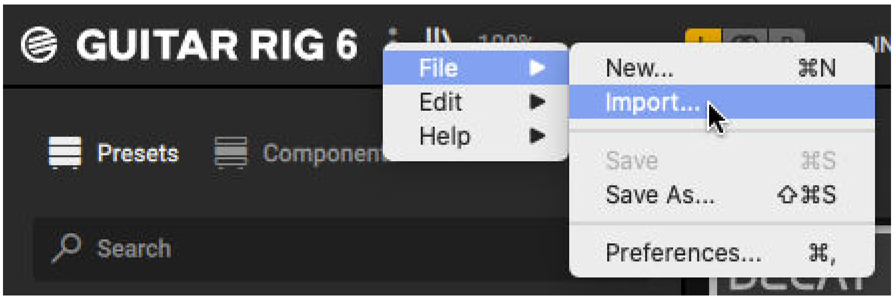
10.3. Importing User Presets

You can import User presets made with both GUITAR RIG 5 and GUITAR RIG 6 from the hard drive. User presets are saved in the following folder:

- macOS: *Macintosh HD/Users/<user name>/Documents/Native Instruments/User Content/Guitar Rig 6/Rack Presets*
- Windows: *C:\Users\<user name>\Documents\Native Instruments\User Content\Guitar Rig 6\Rack Presets*

To import a User preset:

1. Open the Main menu in the Header and select **Import...** in the File sub-menu.



2. Navigate to the preset file you want to import, select it, and confirm by clicking **Open**.

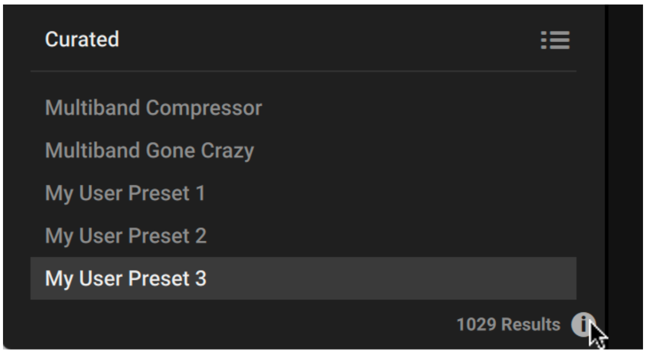
→ The User preset is imported and can be accessed in the Browser.

> You can also import all User presets from an existing GUITAR RIG 5 installation at the same time using the option in the Preferences. For more information, see Library.

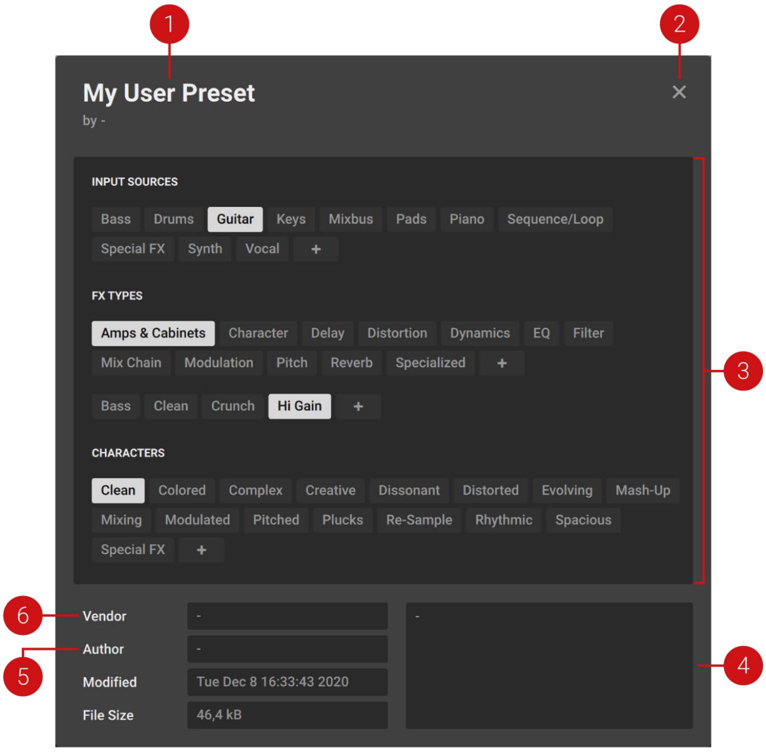
10.4. Editing User Presets in the Info Pane

You can use the Browser's Info pane to edit the Filter tags and properties for User presets.

▶ To open the Info pane for a User preset, select the User preset and click the Info Pane button in the Browser.



The following overview shows the Filter tags and properties you can edit in the Info pane:



1. **Preset name**: Displays the name of the User preset. When placing the mouse over the name, a pen icon appears next to it that you can click to change the name. Alternatively, you can double-click the name to change it.
2. **Close Info pane**: Closes the Info pane.
3. **Filter tags**: Displays all available Filter tags. You can assign Filter tags to your User preset by selecting them. Furthermore, you can add and assign your own User Filter tags. For more information, see Managing User Filter Tags.
4. **Comment**: Displays the comment associated with the User preset. You can click in the field to change the comment.
5. **Author**: Displays the name of the author associated with the User preset. You can click in the field to change the name.
6. **Vendor**: Displays the name of the vendor associated with the User preset. You can click in the field to change the name.

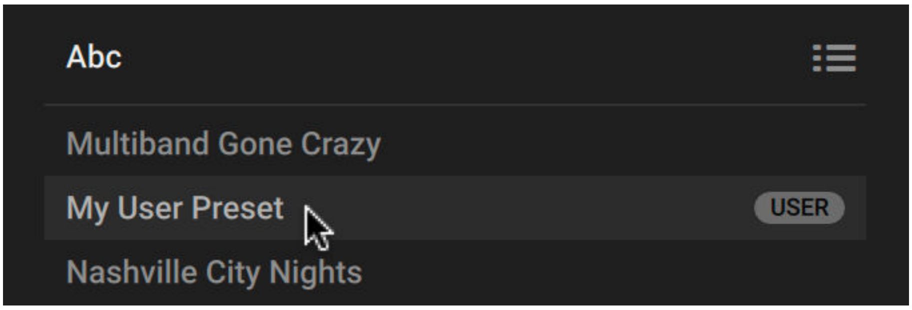
10.5. Managing User Filter Tags

You can add your own User Filter tags and assign them to a User preset in the Info Pane of the Browser.

Adding a User Filter Tag

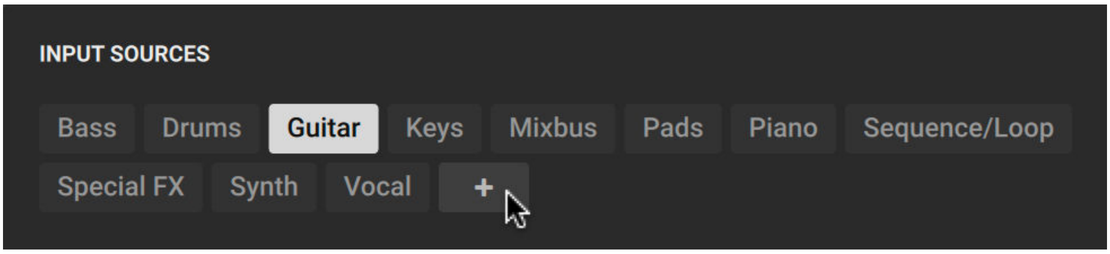
To add a User Filter tag:

1. Select the User preset you want to assign your own User Filter tag to in the Results list by clicking on the corresponding entry.

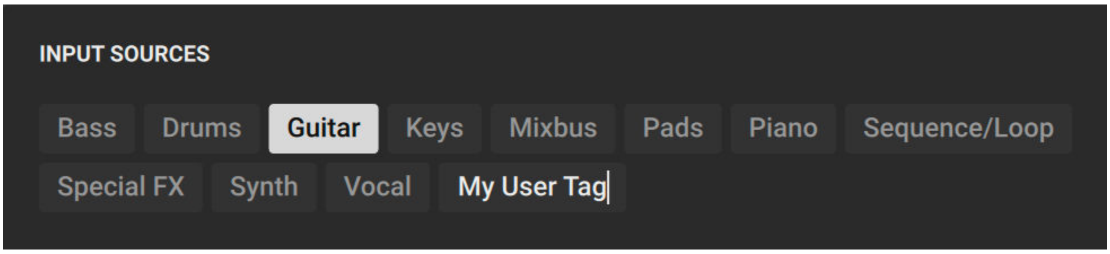


2. Click the Info Pane button in the bottom right corner of the Results list to open the Info pane.

3. Click on the **+** icon under the Filter type you want to add your User Filter tag to.

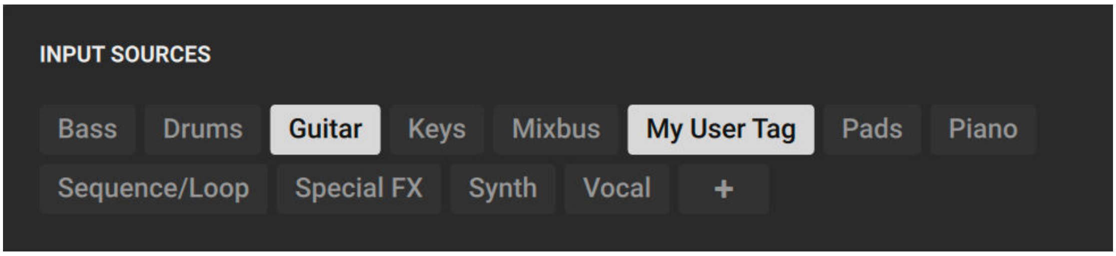


4. Type in a name for your User Filter tag.



5. Press [Enter] or click anywhere in the Info pane to confirm the name and save your User Filter tag.

→ Your new User Filter tag is added and assigned to the User preset.

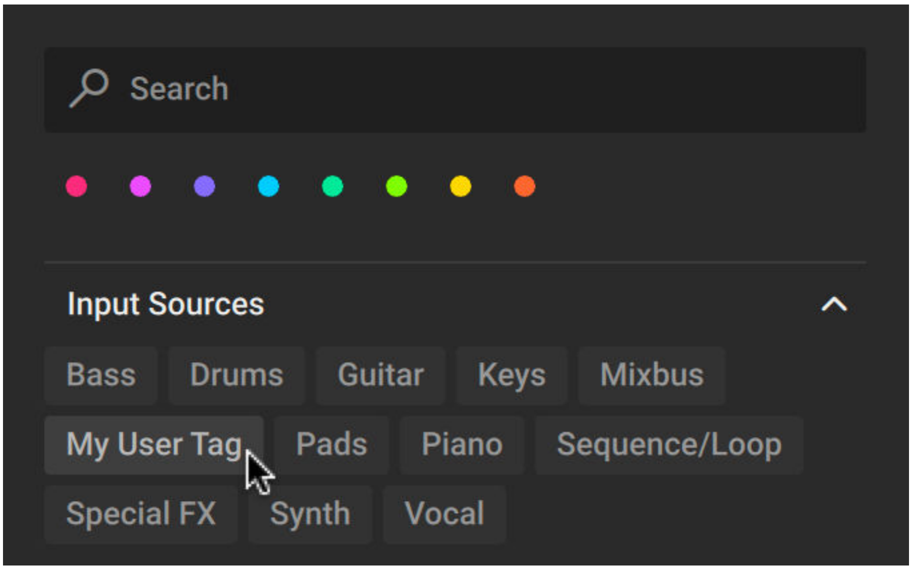


Once added, your User Filter tag shows up in the Browser and can be used to filter the presets in the Results list. For more information, see Using Filters.

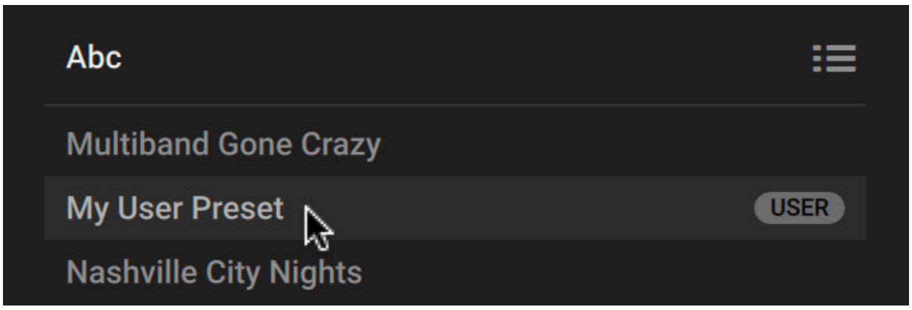
Deleting a User Filter Tag

To delete a User Filter tag from the Browser:

1. Filter the presets in the Results list by clicking on the User Filter tag you want to delete.

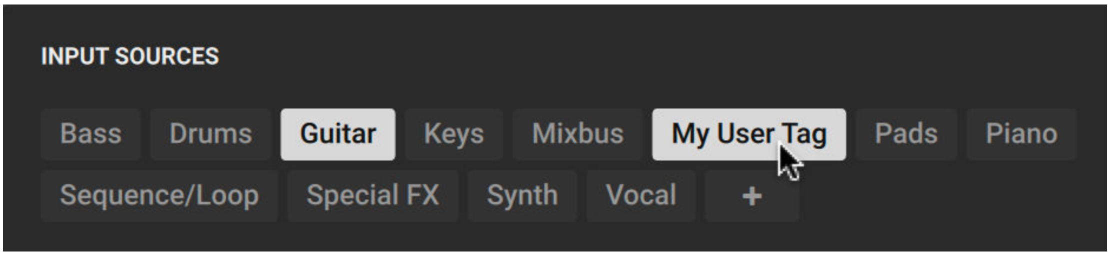


2. Select a User preset in the Results list by clicking on the corresponding entry.



3. Click the Info Pane button in the bottom right corner of the Results list to open the Info pane.

4. In the Info pane, click on the User Filter tag you want to delete in order to remove it from the selected User preset.



5. Repeat steps 2 and 4 for all User presets the User Filter tag is assigned to.

→ The User Filter tag is deleted and does not show up in the Browser any more.

10.6. Deleting User Presets

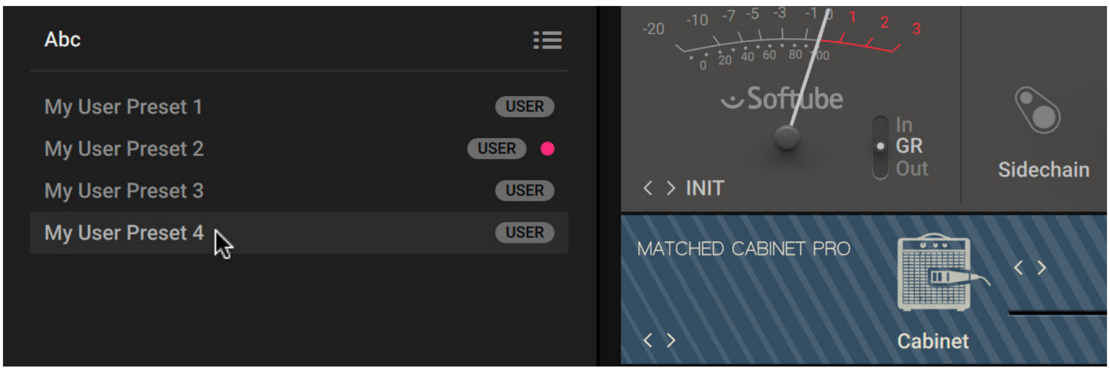
You can delete User presets directly in the Browser. The entries will be removed from the Results list, and the corresponding files will be deleted from the User folder.

For more information about the User folder, see Importing User Presets.

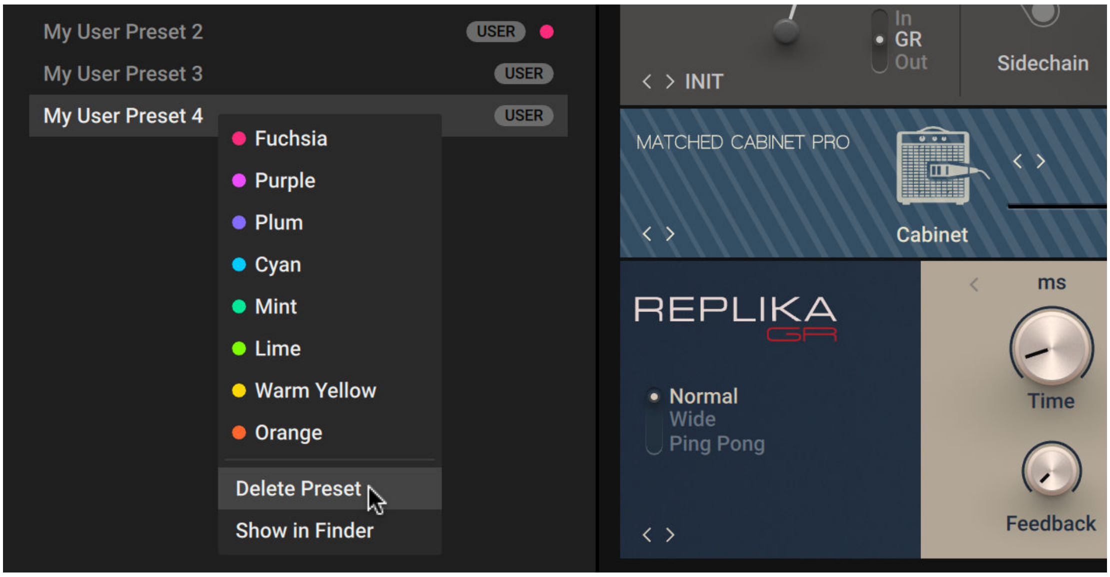
To delete User presets:

1. Right-click a User preset in the Results list to open the context menu. If multiple User presets are selected, the changes will apply to all of them.

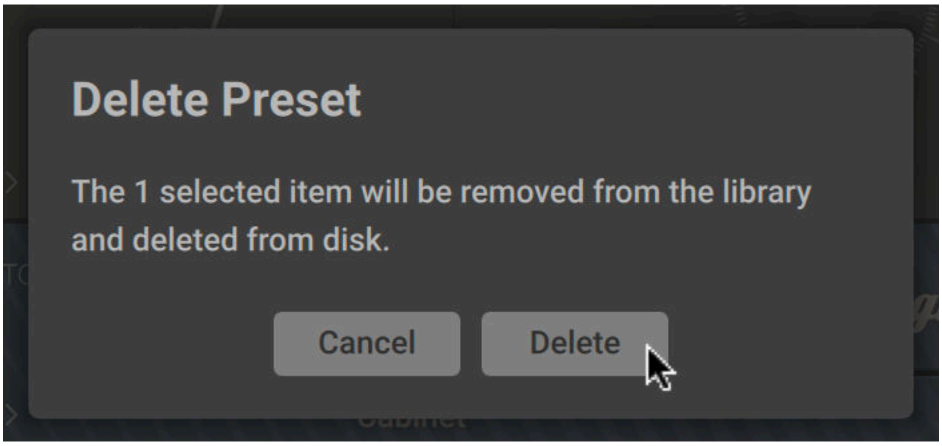
   You can select a preset by clicking on it. Holding [SHIFT] on the keyboard allows for selection of multiple presets.



2. Select **Delete Preset** from the context menu.



3. Confirm that you want to delete your User preset by clicking **Delete** in the dialog box.



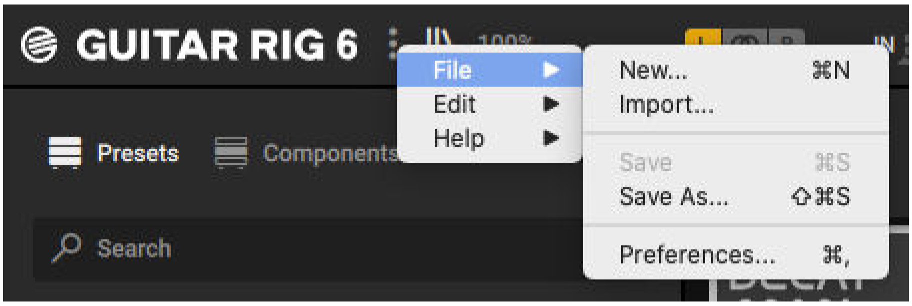
→ The User preset is deleted from the hard drive and does not show up in the Results list any more.

10.7. Managing Presets Using the Main Menu

You can use the Main menu in the Header to manage Presets using common file commands. The commands can be accessed in the **File** sub-menu.

Keyboard shortcuts are available for most of the commands contained in the **File** sub-menu. For more information, see Keyboard Shortcuts.

The following overview shows you the available commands:



- **New...**: Clears the Rack and creates a new User preset.
- **Import...**: Imports User presets from the hard drive. For more information, see Importing User Presets.
- **Save**: Saves the loaded User preset. When no User preset is loaded, the command is deactivated and grayed out.
- **Save As...**: Saves the loaded preset as a new User preset.

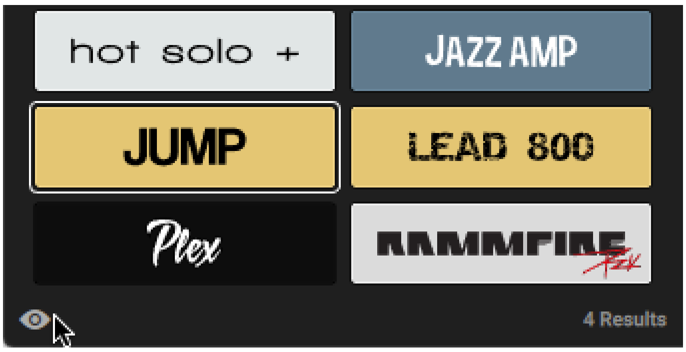
10.8. Using Component Presets

Component presets contain all settings of a Component, enabling you to save and recall the state of a Component independently of the Rack. You can load Component presets from the GUITAR RIG library to quickly try new settings, or save your personal settings for later use. You can access Component presets in the Browser or in the Component itself.

Component Presets in the Browser

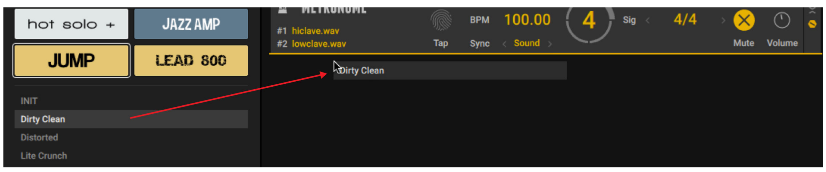
You can access Component presets in the Browser when the Component selector at the top is set to **Components**.

▶ To show the Component presets, click the View Component Presets button in the bottom left corner of the Browser.



You can load a Component preset using drag and drop. The associated Component is added automatically.

▶ Drag and drop the Component preset from the Browser into the Rack. The expected position of the added Component is indicated by a yellow line.



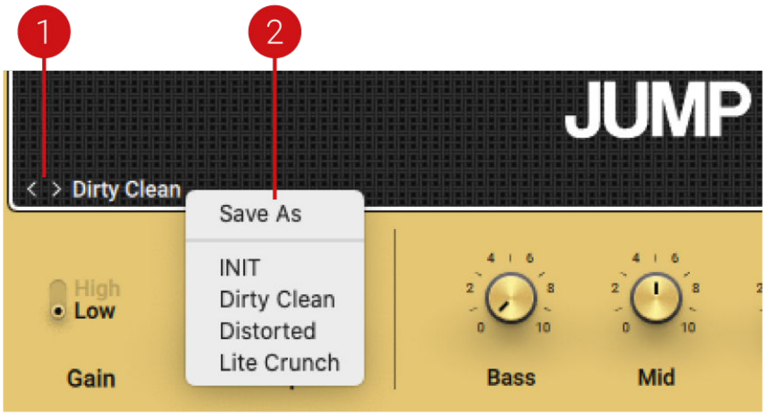
You can also replace one or multiple Components by loading a Component preset in the same way Components can be replaced with a Component. For more information, see Replacing Components in the Rack.

Component Presets in the Component

You can access Component presets in the Component by using the Preset menu.

▶ To show the Preset menu in a Component, click on the preset name shown in the interface.

The following overview shows the available options in the Preset menu:



1. **Previous preset / Next preset**: Switches between Component presets of the respective Component. Clicking on the leftward arrow loads the previous preset. Clicking on the rightward arrow loads the next preset.
2. **Drop-down menu**: Provides the **Save As** option to save a new User Component preset, and shows a list of all available Component presets. Clicking on an entry in the list loads the corresponding Component preset.

11. Overview of the Rack

The Rack is at the heart of GUITAR RIG. Here you can combine and tweak the individual effects, called Components, to create any multi-effect imaginable. The required infrastructure is provided using dedicated Rack Tools, which are accessible through the Rack's Toolbar.

The following overview shows you the basic structure of the Rack:

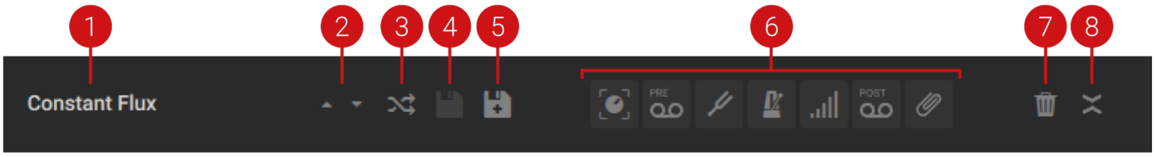
1. **Components**: In the main area of the Rack you can add Components, arrange them in elaborate effects chains, and tweak all of their controls. For more information, see Using the Rack.
2. **Toolbar**: The Toolbar gives access to important Rack controls and the Rack Tools, which provide the Rack's infrastructure. For more information, see Toolbar.
3. **Component controls**: The Component controls provide common functions that are found on most Components. For more information, seeComponent Controls .
4. **Macros**: The Macros are a special Rack Tool providing global controls that can be assigned to any parameter in the Rack. For more information, see Macros.

11.1. Toolbar

The Toolbar enables you to switch between and save presets, clear the Rack, collapse or expand all Components, and show or hide the Rack Tools, which provide the Rack's basic infrastructure.

The GUITAR RIG stand-alone application retains settings made in the Toolbar from the last session. The plug-in uses default settings (all Rack Tools hidden).

The following overview shows you the controls and settings in the Toolbar:

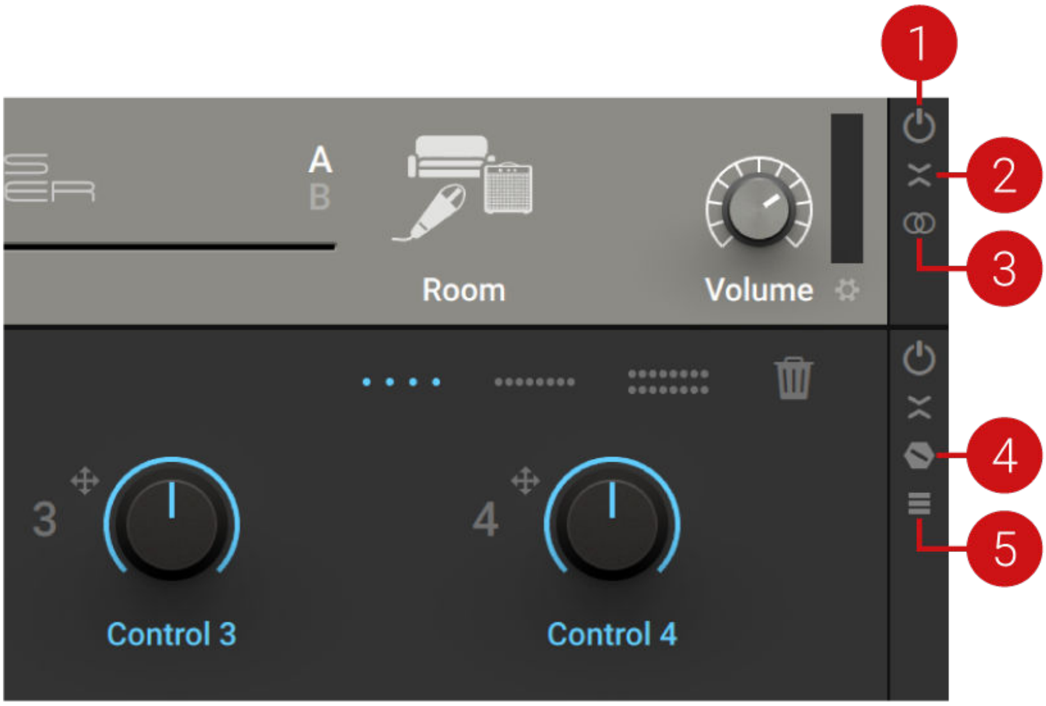


1. **Preset display**: Shows the loaded preset. When the preset has been changed, this is indicated by a * following the preset name.
2. **Preset selection**: Switches between presets in the Browser's Results list. For more information, see Loading Presets.
3. **Preset Shuffle**: Loads a random preset from the Browser's Results list. For more information, see Loading Presets.
4. **Save preset**: Saves the loaded User preset. When no User preset is loaded, the button is deactivated and grayed out. For more information, see Saving User Presets.
5. **Save new preset**: Saves the loaded preset as a new User preset. For more information, see Saving User Presets.
6. **Show Rack Tools**: Show the individual Rack Tools in the Rack. For more information, see Rack Tools.
7. **Clear Rack**: Clears the contents of the Rack and all Macro assignments.
8. **Collapse/Expand All**: Expands or collapses the view of all Components in the Rack. When expanded, the Components are shown in full, including all of their controls. When collapsed, the controls are hidden and the Components are represented by narrow bars showing only their name and artwork.

11.2. Component Controls

The Component controls are located on the right side of each Component and enable you to switch individual Components on or off, collapse or expand their view, and show or hide additional views like the Expert panel and nested Component lists.

The following overview shows you the available Component controls:



1. **Component On/Off**: Switches the Component on or off. When on, the input signal is processed and passed on to the next Component. When off, the input signal bypasses the Component and processing is deactivated, removing the Component's load from the CPU.
2. **Collapse/Expand**: Collapses or expands the view of the Component. When expanded, the Component is shown in full, including all of its controls. When collapsed, the controls are hidden and the Component is represented by a narrow bar showing only its name and artwork.
3. **Stereo**: Activates stereo processing for the Component. When activated, the left and right stereo channels are processed separately. When deactivated, the left and right stereo channels are summed to mono. This control is only available for certain Components.

   You can reduce the CPU load of a Component by deactivating stereo processing.

4. **Show Expert Panel**: Shows the Expert panel containing additional controls that you can use to change the character or behavior of the Component. This control is only available for certain Components, including most of the amplifiers.

   For information about the individual parameters and controls of Components, including the Expert panel, see Components Reference.

5. **Show Content**: Shows a list of nested Components. This control is only available for the Container Component and the Master FX Rack Tool.

12. Using the Rack

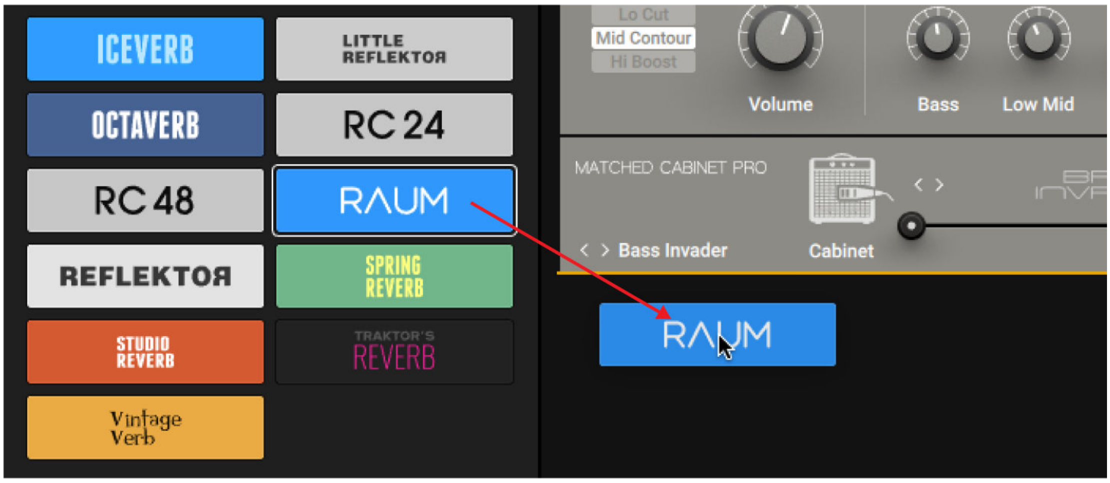
In the Rack you can combine and tweak the individual effects, called Components, to create any multi-effect imaginable. The Rack facilitates this by providing workflows for adding, removing, replacing, and moving Components. The following sections explain these basic workflows.

ⓘ The same workflows also apply for the lists of Components in the Master FX Rack Tool and the Container Component.

12.1. Adding Components to the Rack

You can add Components to the Rack by using drag and drop.

▶ To add a Component, drag and drop the respective Component Tile from the Browser into the Rack. The expected position in the Rack is indicated by a yellow line.

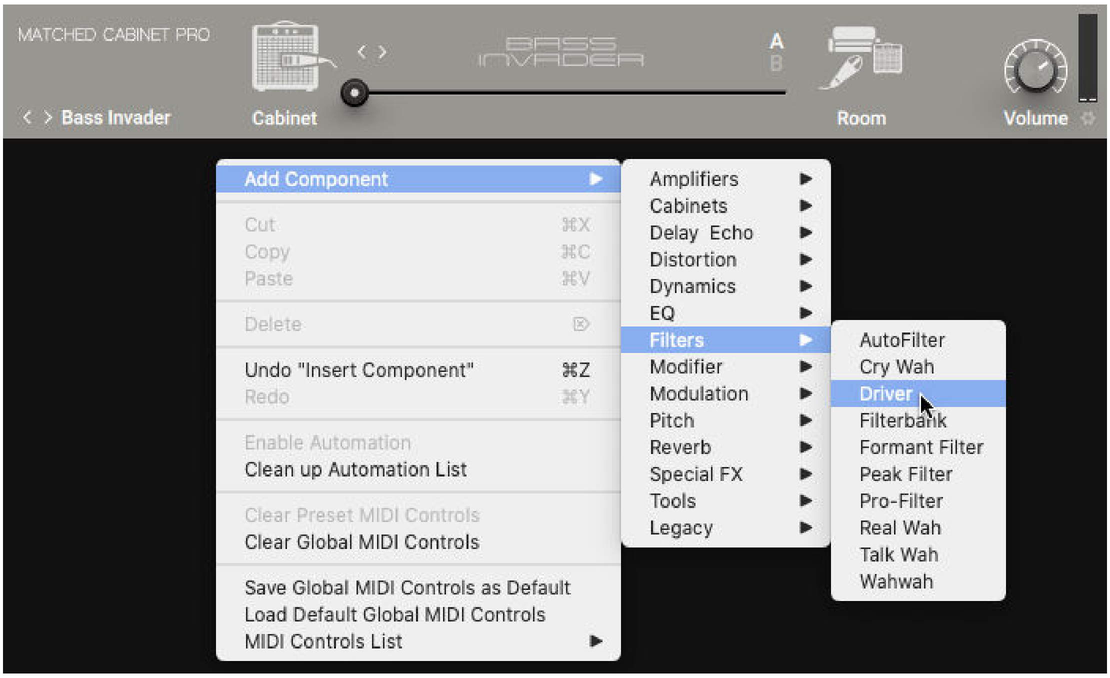


When no Component is selected in the Rack, you can also add Components by double-clicking the respective Component Tile in the Browser. The Component will be added at the bottom of the Rack.

You can also use the context menu in the Rack to add Components. If no Component is selected, the new Component is added at the bottom of the Rack. If a Component is selected, the new Component is added underneath the selected Component. Opening the context menu automatically selects the Component at the mouse position.

ⓘ You can select a Component by clicking on it. Holding [SHIFT] on the keyboard allows for selection of multiple Components.

▶ To add a Component, right-click in the Rack to open the context menu and select the Component from the **Add Component** menu.



12.2. Deleting Components from the Rack

You can delete Components from the Rack by using drag and drop.

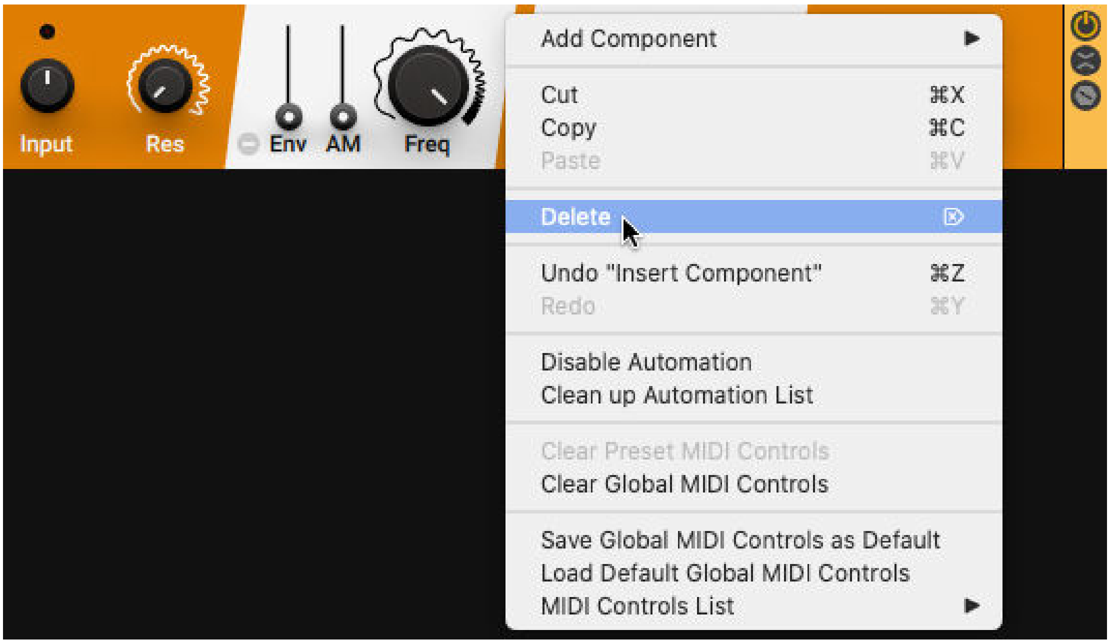
▶ To delete a Component, drag and drop the Component to the outside of the Rack. If multiple Components are selected, they are deleted together.

You can also use the context menu to delete Components. If no Component is selected, the command in the context menu is not available. If one or more Components are selected, they will be deleted. Opening the context menu automatically selects the Component at the mouse position.

> ⓘ You can select a Component by clicking on it. Holding [SHIFT] on the keyboard allows for selection of multiple Components.

▶ To delete all selected Components, right-click in the Rack to open the context menu and select **Delete**.

Alternatively, you can press [delete] (macOS) or [Del] (Windows) on the keyboard, or use the command from the Main menu. For more information, see Editing the Rack.



12.3. Replacing Components in the Rack

You can replace Components in the Rack by using drag and drop.

▶ To replace a Component, drag and drop the respective Component Tile from the Browser onto the Component you want to replace. The Component that is expected to be replaced is highlighted in yellow.

You can also replace Components by double-clicking Component Tiles. All Components selected in the Rack will be replaced.

You can select a Component by clicking on it. Holding [SHIFT] on the keyboard allows for selection of multiple Components.

▶ To replace all selected Components, double-click a Component Tile in the Browser.

12.4. Moving Components in the Rack

You can move Components in the Rack by using drag and drop.

▶ To move a Component, drag and drop the Component to a new position in the Rack. The expected position in the Rack is indicated by a yellow line. If multiple Components are selected, they are moved together.

ⓘ You can select a Component by clicking on it. Holding [SHIFT] on the keyboard allows for selection of multiple Components.

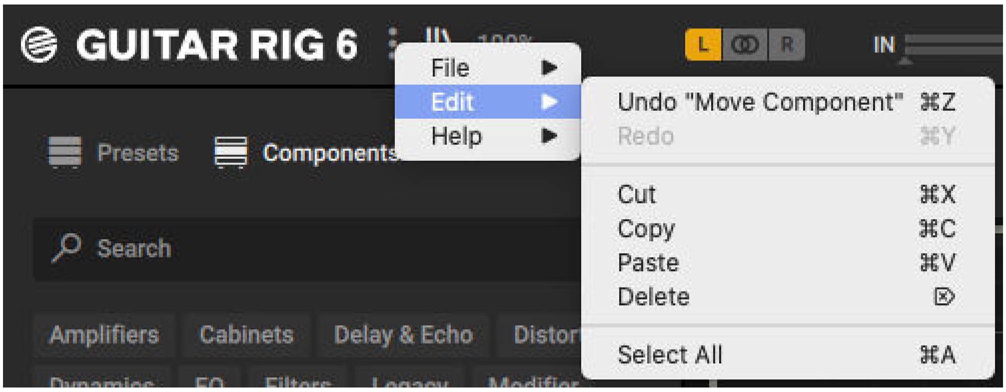
12.5. Editing the Rack

You can edit the Rack by using common editing commands, including undo and redo as well as cut, copy, and paste. The commands can be accessed in the **Edit** sub-menu of the Main menu. Additionally, most of the commands are also available in the Rack's context menu.

Editing the Rack Using the Main Menu

You can access common editing commands for the Rack in the **Edit** sub-menu of the Main menu. The following overview shows you the available commands:

Keyboard shortcuts are available for all commands contained in the **Edit** sub-menu. For more information, see Keyboard Shortcuts.



- **Undo**: Reverts the last change made to the Rack.
- **Redo**: Repeats the last change from the undo history.
- **Cut**: Cuts all selected Components and copies them to the clipboard.
- **Copy**: Copies all selected Components to the clipboard.
- **Paste**: Pastes the Components from the clipboard.
- **Delete**: Deletes all selected Components. For more information, see Deleting Components from the Rack.
- **Select All**: Selects all Components in the Rack.

You can select a Component by clicking on it. Holding [SHIFT] on the keyboard allows for selection of multiple Components.

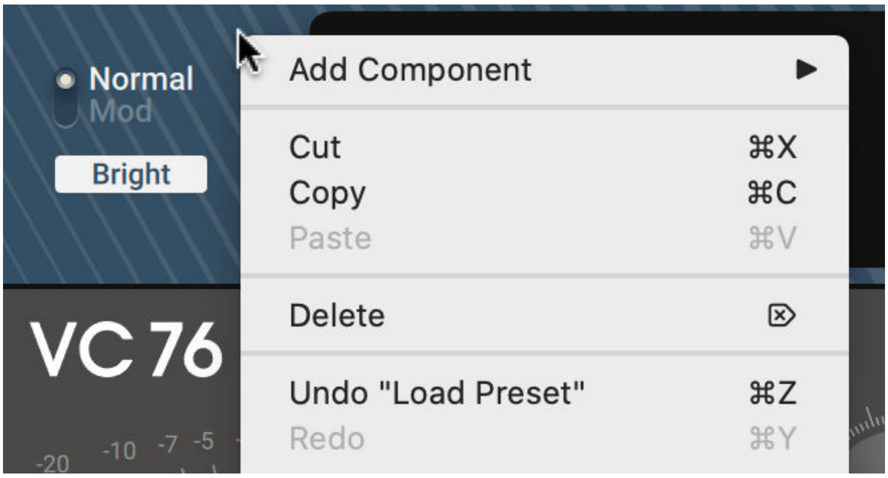
Editing the Rack Using the Context Menu

You can access common editing commands for the Rack in its context menu.

▶ To open the context menu, right-click on the background of a Component or the Rack.

The following overview shows you the available commands:

Keyboard shortcuts are available for all editing commands contained in context menu of the Rack. For more information, see Keyboard Shortcuts.



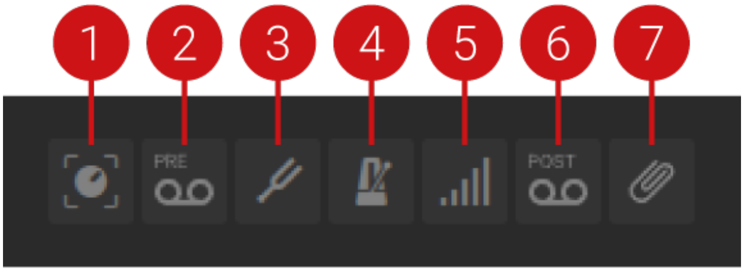
- **Add Component**: Opens sub-menus showing all available Components, and allows to add Components to the Rack by selecting them. For more information, see Adding Components to the Rack.
- **Cut**: Cuts all selected Components and copies them to the clipboard.
- **Copy**: Copies all selected Components to the clipboard.
- **Paste**: Pastes the Components from the clipboard.
- **Delete**: Deletes all selected Components. For more information, see Deleting Components from the Rack.
- **Undo**: Reverts the last change made to the Rack.
- **Redo**: Repeats the last change from the undo history.

> ⓘ You can select a Component by clicking on it. Holding [SHIFT] on the keyboard allows for selection of multiple Components.

13. Rack Tools

The Rack Tools provide the Rack's infrastructure, including macro control of parameters, audio recording, a metronome and a tuner, as well as additional volume control and global effects. You can show or hide each Rack Tool individually from the Toolbar.

The following overview shows you the Rack Tools in the Toolbar:

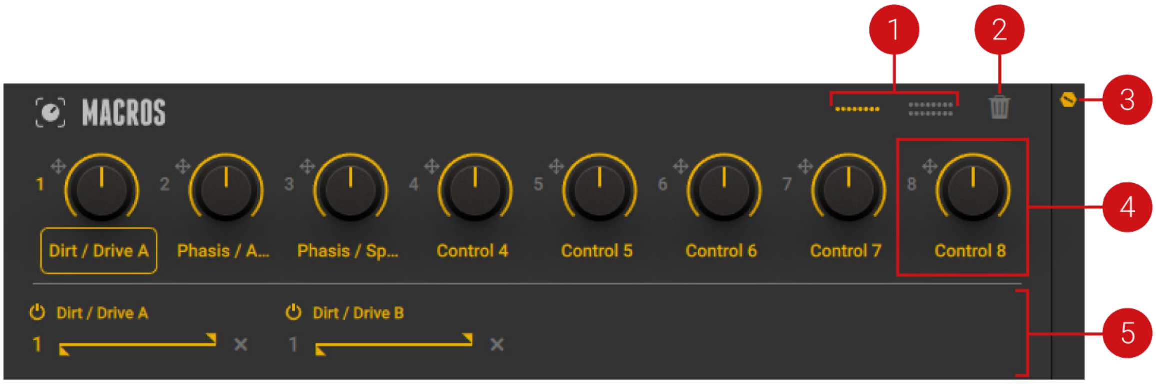


1. **Show Macros**: Shows the Macros Rack Tool in the Rack. Macros are global controls that can be assigned to any parameter in the Rack. For more information, see Macros.
2. **Show Tapedeck Pre**: Shows the Tapedeck Pre Rack Tool in the Rack. The Tapedeck Pre is an audio recorder that can be used for recording and playing back the Rack's input signal. For more information, see Tapedeck Pre.
3. **Show Tuner**: Shows the Tuner Rack Tool in the Rack. The Tuner is an electronic tuner that can be used to detect the pitch of the notes played by an instrument. For more information, see Tuner.
4. **Show Metronome**: Shows the Metronome Rack Tool in the Rack. The Metronome is a metronome and musical clock that can be used to play to a click track. Additionally, it sets the global tempo reference for the Rack. For more information, see Metronome.
5. **Show Preset Volume**: Shows the Preset Volume Rack Tool in the Rack. The Preset Volume is a level control that can be used to compensate for loudness differences between presets. For more information, see Preset Volume.
6. **Show Tapedeck Post**: Shows the Tapedeck Post Rack Tool in the Rack. The Tapedeck Post is an audio recorder that can be used for recording and playing back the Rack's output signal. For more information, see Tapedeck Post.
7. **Show Global FX**: Shows the Global FX Rack Tool in the Rack. The Global FX host Components that are independent of the loaded preset and are applied to the output signal of the Rack. For more information, see Global FX.

13.1. Macros

Macros provide global controls that can be assigned to any parameter in the Rack. They enable you to create a set of controls that bring out the most of your custom multi-effect.

The Macros contain the following parameters and controls:



1. **Macro Layout selector**: Switches the number of available Macros between eight and 16.
2. **Clear Macros**: Clears all Macro assignments.
3. **Show Expert panel**: Shows the Expert panel that you can use to display and edit the assignments of the selected Macro.
4. **Macro control**: Controls parameters of Components in the Rack, allowing you to create custom effect controls. For information, see Assigning Macros.
5. **Expert panel**: Displays the assignments of the selected Macro and provides functions to edit them. For more information, see Editing Macros.

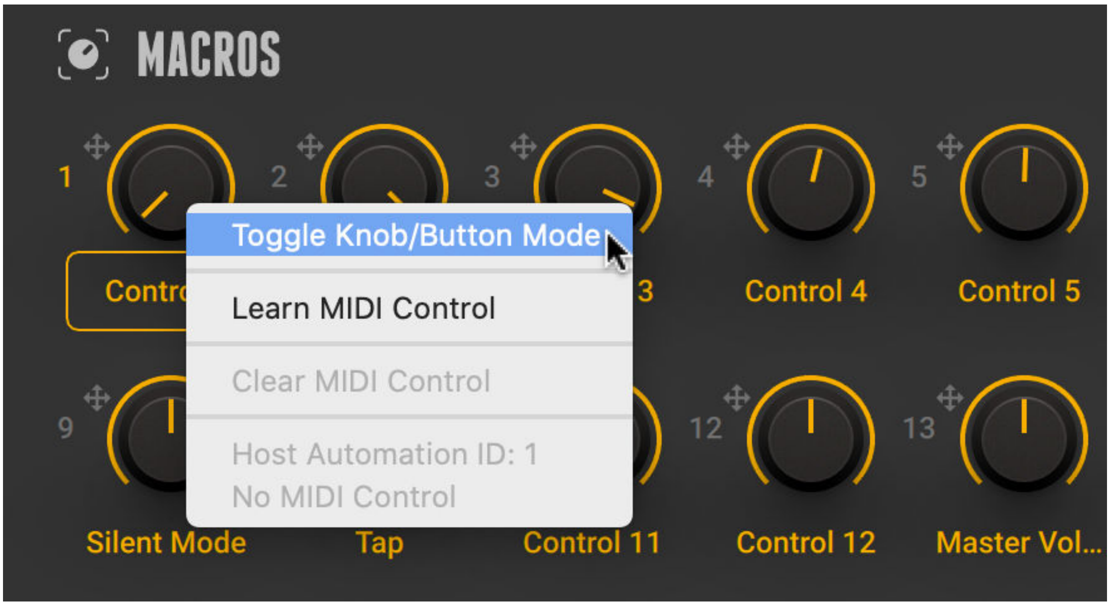
Configuring Macros

Macros can be configured as rotary knobs or buttons. Macro buttons enable you to switch functions on multiple Components at the same time, or adapt the Macros to the layout of your MIDI controller. Additionally, you can use Macro buttons creatively by assigning them to any control on Components and toggle between two specific values.

> ⓘ You can set the specific values controlled by the Macro button for each of the assigned controls in the Expert panel. For more information, see Editing Macros.

To change the configuration of a Macro as either knob or button:

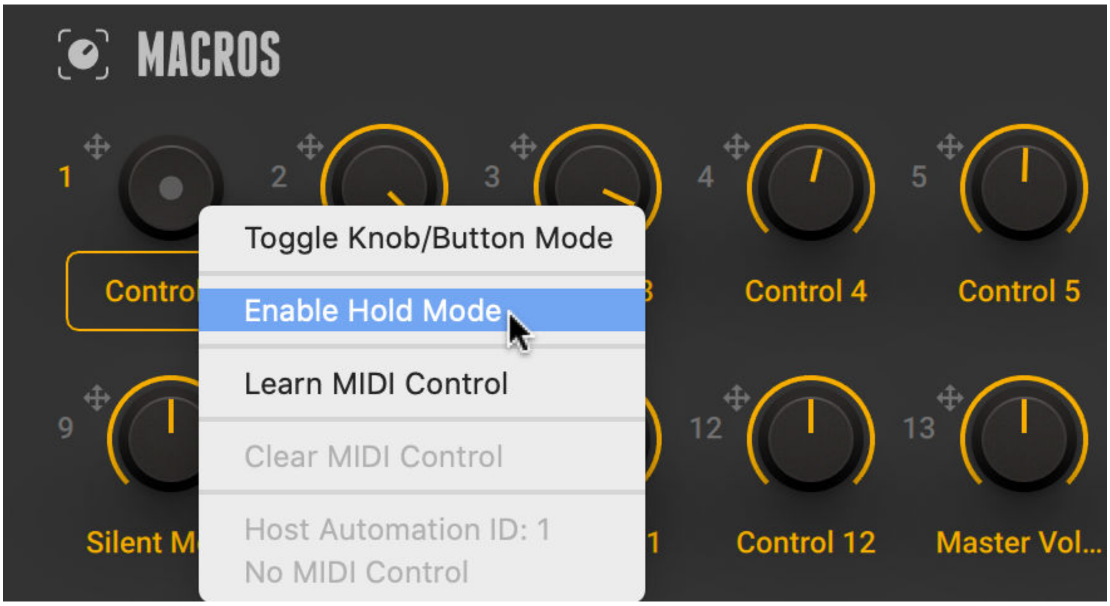
▶ Right-click the Macro and select **Toggle Knob/Button Mode**.



→ The configuration of the the Macro is changed.

When a Macro is configured as a button, it can be set to either Toggle or Hold mode. In Toggle mode, the button changes its state each time it is pressed. In Hold mode, the button holds the On value for as long as the button is pressed, and returns to the Off value as soon as it is released.

▶ To activate Hold mode, right-click the Macro button and select **Enable Hold Mode**.



Assigning Macros

You can assign Macros to parameters using drag and drop. A single Macro can be assigned to up to eight parameters at the same time.

1. Click the Macro number icon or the target icon next to it, and keep the mouse button pressed.

2. Drag and drop the icon onto the parameter you want to assign the Macro to.

→ The Macro is assigned to the parameter, and the Expert Panel displays the assignment accordingly.

Editing Macros

The Macros' Expert panel displays the assignments of the selected Macro and provides functions to deactivate, reassign, delete, and adjust the range of assignments. Additionally, you can rename Macros directly on their label

Deactivating Assignments

▶ To deactivate an assignment, click the Assignment On/Off button.

→ The assignment is deactivated and the assigned parameter will not be controlled by the Macro. You can click the Assignment On/Off button again to activate the assignment.

Reassigning Assignments

▶ To reassign an assignment, drag and drop the Assignment number icon to the new parameter.

→ The assignment is removed from the original parameter and assigned to the new parameter.

Deleting Assignments

▶ To delete an assignment, click on the **x** button next to it.

→ The assignment is removed from the parameter.

Adjusting the Range of Assignments

The range of an assignment determines the response of the assigned parameter to the Macro control. You can adjust the range by setting its minimum and maximum value on the range slider.

By moving the maximum value past the minimum value or vice versa, you can invert the range of the assignment. In this case the response of the parameter to the Macro control is reversed.

▶ To adjust the range of an assignment, move the minimum and maximum values (triangle icons) on the range slider.

The range settings are saved in relation to the assigned parameter. When deleting and reassigning assignments, or when assigning additional Macros to the same parameter, the range settings are preserved.

Renaming Macros

To rename a Macro:

1. Double-click the Macro's label.

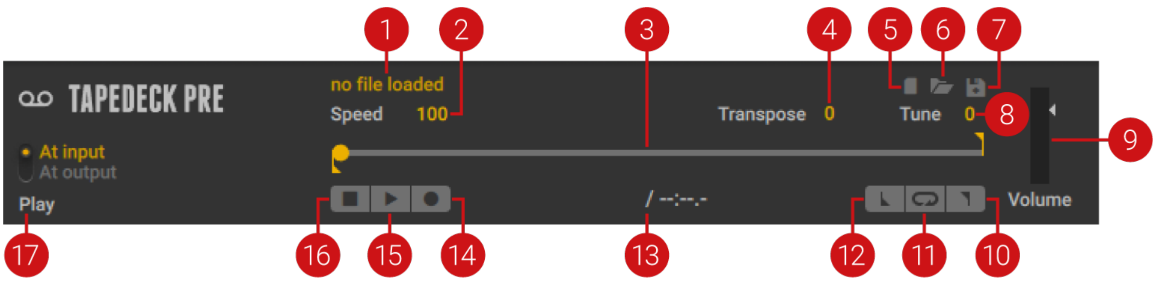
2. Type in the new name for the Macro and press [Enter] to confirm.

→ The new name appears in the Macro's label.

13.2. Tapedeck Pre

Tapedeck Pre is an audio recorder that can be used for recording and playing back the Rack's input signal. You can use it to record the unprocessed sound of your instrument, and play it back through the Rack and when adding and adjusting Components. Additionally, you can change the playback speed of the recording without changing its pitch, and vice versa. For example, this enables you to slow down a musical phrase when playing along in practice.

This Rack Tool contains the following parameters and controls:



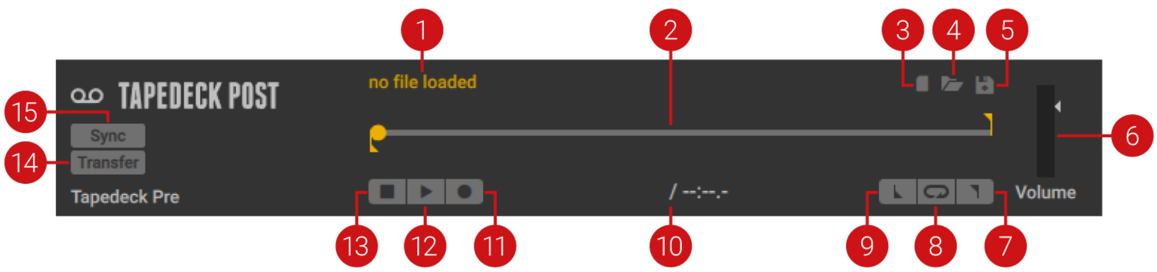
1. **File name**: Displays the file name of the loaded audio file.
2. **Speed**: Adjusts the playback speed of the audio file in the range of 50% to 150%. Time-stretching is used to preserve the original pitch.
3. **Timeline**: Represents the lengths of the audio file. The dot indicates the playback position. You can move the dot along the timeline to adjust the playback position. The downward pointing Start marker indicates the start of the Loop, the upward pointing End marker indicates the end of the Loop. You can move the Loop markers along the timeline to adjust the length and position of the Loop.
4. **Transpose**: Adjusts the pitch of the audio file in the range of -12 to +12 semitones. Pitch-shifting is used to preserve the original playback speed.

5. **New file**: Creates a new audio file and activates the Record button.
6. **Open file**: Opens a file dialog that loads an audio file from the hard drive. The supported file formats are WAV, AIFF, and MP3.
7. **Save file**: Opens a file dialog that saves the loaded audio file to the hard drive.
8. **Tune**: Adjusts the pitch in the range of -50 to +50 cents. You can use this control to fine-tune the setting made with the Transpose control.
9. **Volume**: Adjust the output level at which the playback is mixed with the input signal. The meter shows the peak level of the output signal. The red indicator at the top indicates clipping. When Record is activated, the peak level of the input signal is shown.
10. **Loop End**: Sets the End marker for the Loop at the current playback position.
11. **Loop**: Activates the Loop. When playback reaches the position of of the End marker, playback resumes at the position of the Start marker.
12. **Loop Start**: Sets the Start marker for the Loop at the current playback position.
13. **Timecode**: Shows the timecode in min:sec:ms. On the left side, the playback position is shown. On the right side, the length of the audio file is shown.
14. **Record**: Arms the recording and shows the peak level of the input signal on the meter of the Volume control. When Record is activated, you can start recording by clicking Play.
15. **Play**: Starts the playback of the audio file. When Record is activated, it starts the recording.
16. **Stop**: Stops the playback. Clicking Stop a second time resets the playback position to the Loop's Start marker. Clicking Stop a third time resets the playback position to the start of the audio file.
17. **Play**: Changes the routing for the output signal. When **At input** is selected, the output signal is sent to the Rack. This setting is used for regular operation of Tapedeck Pre. When **At output** is selected, the output signal bypasses the Rack and is sent to the input of Tapedeck Post. This setting can be used in combination with the Tapedeck Post's **Transfer** and **Sync** functions to create overdub recordings.

13.3. Tapedeck Post

Tapedeck Post is an audio recorder that can be used for recording the Rack's output signal. You can use it to record the processed sound of your instrument, and open the recorded audio file in other applications. Additionally, you can import audio files into the Tapedeck Post and play them back as unprocessed backing tracks.

This Rack Tool contains the following parameters and controls:



1. **File name**: Displays the file name of the loaded audio file.
2. **Timeline**: Represents the lengths of the audio file. The dot indicates the playback position. You can move the dot along the timeline to adjust the playback position. The downward pointing Start marker indicates the start of the Loop, the upward pointing End marker indicates the end of the Loop. You can move the Loop markers along the timeline to adjust the length and position of the Loop.

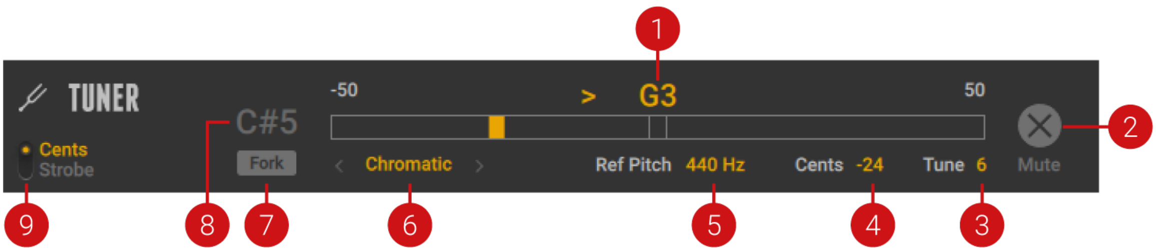
3. **New file**: Creates a new audio file and activates the Record button.
4. **Open file**: Opens a file dialog that loads an audio file from the hard drive. The supported file formats are WAV, AIFF, and MP3.
5. **Save file**: Opens a file dialog that saves the loaded audio file to the hard drive.
6. **Volume**: Adjust the output level at which the playback is mixed with the input signal. The meter shows the peak level of the output signal. The red indicator at the top indicates clipping. When Record is activated, the peak level of the input signal is shown.
7. **Loop End**: Sets the End marker for the Loop at the current playback position.
8. **Loop**: Activates the Loop. When playback reaches the position of of the End marker, playback resumes at the position of the Start marker.
9. **Loop Start**: Sets the Start marker for the Loop at the current playback position.
10. **Timecode**: Shows the timecode in min:sec:ms. On the left side, the playback position is shown. On the right side, the length of the audio file is shown.
11. **Record**: Arms the recording and shows the peak level of the input signal on the meter of the Volume control. When Record is activated, you can start recording by clicking Play.
12. **Play**: Starts the playback of the audio file. When Record is activated, it starts the recording.
13. **Stop**: Stops the playback. Clicking Stop a second time resets the playback position to the Loop's Start marker. Clicking Stop a third time resets the playback position to the start of the audio file.
14. **Transfer**: Sends the loaded audio file to Tapedeck Pre. If an audio file is loaded in Tapedeck Pre, it is replaced by the audio file received from Tapedeck Post. You can use this to create overdubs by playing back a previously recorded file in Tapedeck Pre, playing on top of the playback, and recording the result in Tapedeck Post.
15. **Sync**: Synchronizes the transport of Tapedeck Pre and Tapedeck Pro. When activated, the Play and Stop buttons of either Tapedeck control both Tapedecks simultaneously.

13.4. Tuner

Tuner is an electronic tuner that can be used to detect the pitch of the notes played by an instrument. Additionally, it provides a number of different tuning modes and can also be used to produce a reference tone.

This Rack Tool contains the following parameters and controls:

> Some of the controls are contained in the Expert panel. For information about showing or hiding the Expert panel, see Component Controls.



1. **Tuning display**: Shows the tuning of the input signal in relation to musical pitches. The deviation from the ideal tuning is shown either in cent or as a strobe display, depending on the mode set using **Cents/Strobe**.

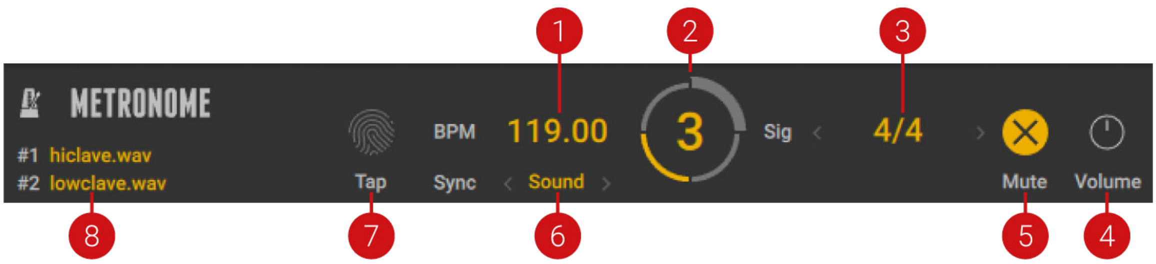
2. **Mute (in Expert panel)**: Mutes the output of the Tuner, but not the reference tune when **Fork** is activated. Muting the output is useful when tuning an instrument in a live situation.
3. **Tune (in Expert panel)**: Transposes the tuning range in the range of -7 to +7 semitones.
4. **Cents (in Expert panel)**: Shows the deviation from the ideal tuning in cents.
5. **Ref Pitch (in Expert panel)**: Sets the tuning reference for the A in the range of 425 to 455 Hz.
6. **Tuning**: Selects between 8 available tuning modes. You can set the tuning mode by clicking on it and selecting an entry from the drop-down menu, or by clicking on the **<** and **>** arrows.
7. **Fork (in Expert panel)**: Activates the reference tone. The pitch of the reference tone can be set using the Fork Pitch control.
8. **Fork Pitch (in Expert panel)**: Sets the pitch of the reference tone that can be activated using **Fork**.
9. **Cents/Strobe**: Switches between two available modes for the visualization in the Tuning display:
   - **Cents**: The display shows the deviation from the ideal tuning to musical pitches in cents. When the yellow indicator is on the left side of the display, the tuning is flat in the range of -50 to -1 cents. When the yellow indicator is on the right side of the display, the tuning is sharp in the range of +1 to +50 cents. When the indicator turns green at the center of the display, the tuning is ideal.
   - **Strobe**: The display shows strobe lights to indicate the relationship between the tuning and the musical pitches. When the strobe lights are moving to the left, the tuning is flat. When the strobe lights are moving to the right, the tuning is sharp. The faster the strobe lights move, the higher is the deviation from the ideal tuning. When the strobe lights stop moving, the tuning is ideal.

13.5. Metronome

Metronome is a metronome and musical clock that can be used to play to a click track. Additionally, it sets the global tempo reference for the Rack, which Components can use to synchronize parameters to the beat of the music.

This Rack Tool contains the following parameters and controls:

> Some of the controls are contained in the Expert panel. For information about showing or hiding the Expert panel, see Component Controls.



1. **BPM**: Sets the tempo of the Metronome in the range of 20 to 400 beats per minute.
2. **Beat display**: Shows the number of the active beat in the bar. Additionally, the time signature as set with **Sig** is displayed as a circular indicator.
3. **Sig**: Sets the time signature of the Metronome. You can set the time signature by clicking on it and selecting an entry from the drop-down menu, or by clicking on the **<** and **>** arrows.

4. **Volume**: Adjusts the output level of the Metronome's audible click track.
5. **Mute**: Mutes the Metronome's audible click track.
6. **Sync**: Selects one of three available synchronization modes for the Metronome:
    - **Extern**: The tempo and time signature are synchronized to the DAW (GUITAR RIG plug-in) or incoming MIDI clock (GUITAR RIG stand-alone application). In this mode the **Tap**, **BPM**, and **Sig** controls are deactivated.
    - **Free**: The tempo and time signature are set independently of the loaded preset using the **Tap**, **BPM**, and **Sig** controls.
    - **Sound**: The tempo and time signature are set according to the values saved in the preset, and can be changed using the **Tap**, **BPM**, and **Sig** controls.
7. **Tap**: Sets the tempo according to the average interval of repeated clicks on the control. **Tap** overwrites the tempo set using the **BPM** control.

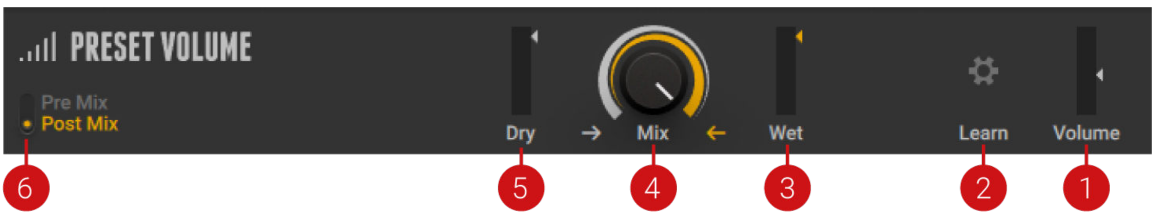
    You can also assign a MIDI controller to **Tap** for hardware control. For more information, see Using MIDI Learn.

8. **#1 / #2 (in Expert panel)**: Displays the audio files used to play the Metronome's click track, and enables you to change them by clicking on the file names. Audio file **#1** is played on downbeats and accentuated beat divisions. Audio file **#2** is played on all other beats.

13.6. Preset Volume

Preset Volume is a utility for adjusting the output level and overall effect mix of a preset. Many of the Components in GUITAR RIG are highly sensitive to level differences, amplifiers and distortion in particular. Therefore, varying the input level and gain settings play an important role in sound design. As a result, presets can have vastly different volume levels. Preset Volume enables you to compensate for this by adjusting the output level, assisted by the automatic **Learn** function. The **Mix** control can be used to blend the original input signal into the processed effect signal from the Rack, for example to achieve parallel compression.

When its view is collapsed, Preset Volume still shows the **Volume** and **Learn** controls, enabling you to apply these key functions while saving space in the Rack.



1. **Volume**: Adjusts the output level of the preset before the Global FX.
2. **Learn**: Adjusts **Volume** automatically by analyzing the input signal and determining the optimal output level. For best results, play loudly while the analyzation is in progress.
3. **Wet**: Adjusts the level of the processed effect signal from the Rack before the **Mix** control is applied.
4. **Mix**: Blends between the input signal of GUITAR RIG and the processed effect signal from the Rack. You can use this to achieve parallel processing, for example parallel compression.

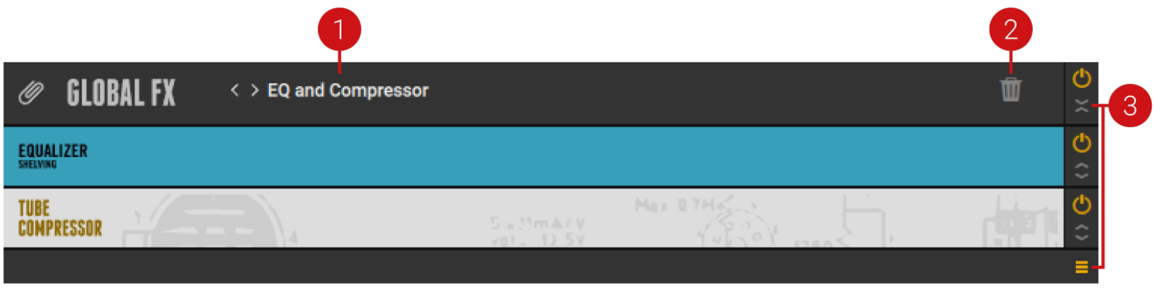
5. **Dry**: Adjusts the level of the input signal of GUITAR RIG that can be blended into the processed effect signal from the Rack using the **Mix** control.
6. **Post Mix / Pre Mix**: Changes the routing and behavior of the **Mix** control. When set to **Post Mix**, the **Mix** control uses its default routing and behavior, blending the input signal of GUITAR RIG into the processed effect signal from the Rack. When set to **Pre Mix**, the **Mix** control splits the input signal before the Rack, adjusting both the level of the Dry signal that bypasses the Rack, and the level of the Wet signal that is processed by the Rack. Since this affects the input level of the Rack, the sound of Components can change, in particular when using amplifiers and distortion.

13.7. Global FX

Global FX host Components that are independent of the loaded preset and are applied to the output signal of the Rack. Therefore, you can use the Global FX to set up an effects chain that finalizes and polishes the sound of any preset you load.

> Since the Global FX are independent of presets, the decay of long echoes and reverbs is not cut off when loading a new preset. You can use this to create smooth transitions when switching between presets.

This Rack Tool contains the following parameters and controls:



1. **Preset menu**: Recalls and saves presets for the Global FX. This menu can be used in the same way as the Preset menu in Components. For more information, see Using Component Presets.
2. **Clear Global FX**: Clears the contents of the Global FX.
3. **Show Content**: Shows or hides the list of Components in the Global FX.

> You can add, remove, replace, and move Components in the list in the same way as in the Rack. For more information, see Using the Rack.

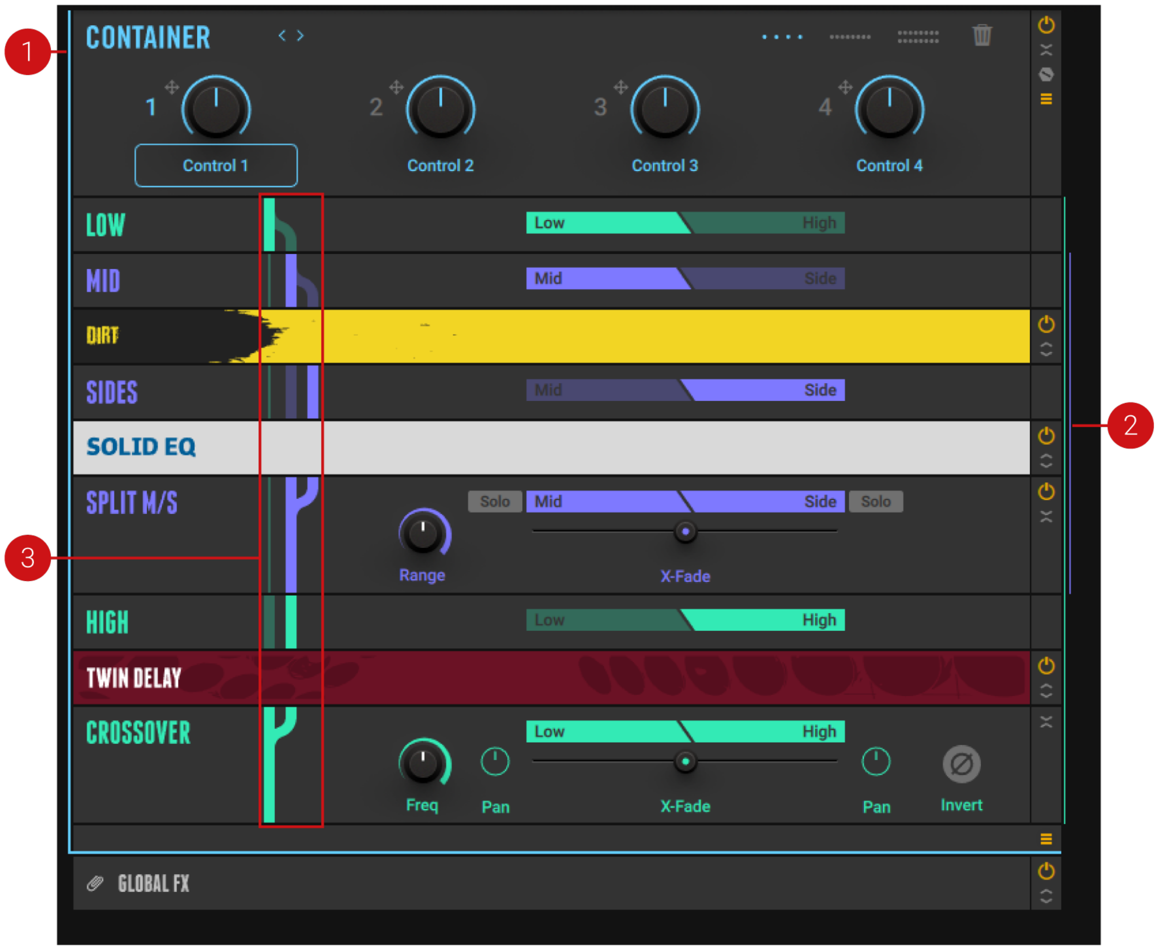
14. Tools

The Components in the Tools category provide auxiliary functions that you can use to change the Rack's signal flow in creative ways. You can use the Container to consolidate multiple effects or create complex effect chains using the Splitters, including Split Mix, Freq Crossover, and M/S Balance.

14.1. Visualization of Tools

Tools feature special visualization in the Rack that helps you maintain an overview of the structure when nesting Components within Containers and Splitters, or even Splitters within Splitters. The visualization dynamically adapts to changes made in the Rack.

The following overview shows you the indicators used in the visualization:



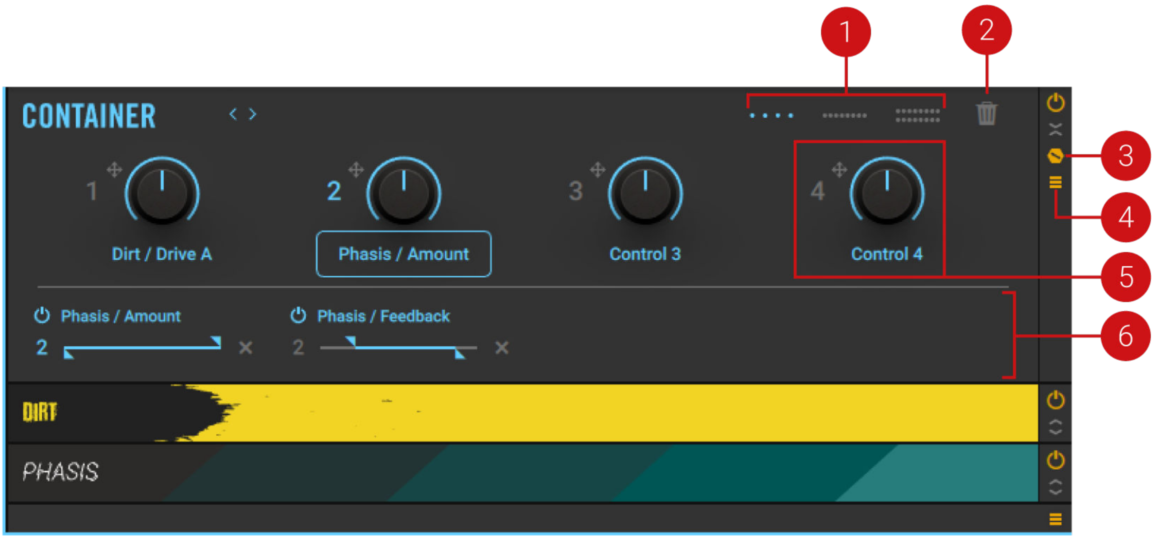
1. **Container indicator**: The thin blue line on the left side of the Rack indicates the extent of the Container and its contents.
2. **Splitter indicator**: The thin colored lines on the right side of the Rack indicate the extent and hierarchical level of Splitters in the Rack, starting with Mint, Plum, Pink, Red. Up to four levels of splitting are shown at the side of the Rack.
3. **Routing indicators**: Show the signal paths of Splitters, including the hierarchy of nested Splitters.

- The thin line represents the signal of a higher-level Splitter that is being passed on down the signal path in the background.
- The thick, solid line represents the signal that is fed into the effects chain nested right below. Additionally, it represents the summed signals in the Splitter's output section.
- The thick, translucent line represents the signal that is fed into other effects chain in the Splitter.

14.2. Container

The Container allows you to create powerful multi effects by combining multiple Components, controlled by up to 16 Container Macros. You can use it to organize your Rack more clearly, or to conveniently reuse your favorite combinations of effects in different Racks.

The Container contains the following parameters and controls:



1. **Macro Layout selector**: Switches the number of available Container Macros between four, eight, and 16.
2. **Clear Container**: Clears the contents of the Container and all Container Macro assignments.
3. **Show Expert panel**: Shows the Expert panel that you can use to display and edit the assignments of the selected Container Macro.
4. **Show Content**: Shows or hides the list of Components in the Container.

> You can add, remove, replace, and move Components in the list in the same way as in the Rack. For more information, see Using the Rack.

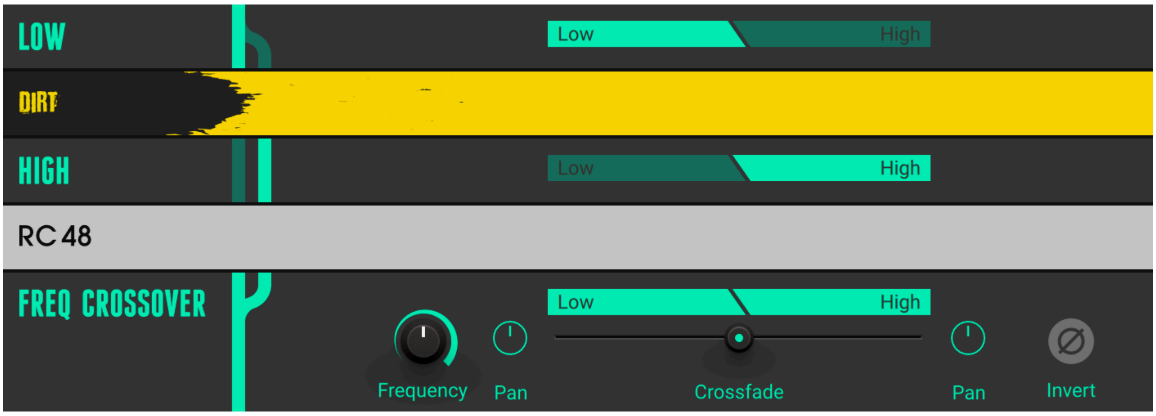
5. **Macro control**: Controls parameters of Components in the Container, allowing you to create custom effect controls. Container Macros can be assigned in the same way global Macros are assigned. For more information, see Assigning Macros.
6. **Expert panel**: Displays the assignments of the selected Container Macro and provides functions to edit them. Container Macros can be edited in the same way global Macros are edited. For more information, see Editing Macros.

14.3. Freq Crossover

Freq Crossover is a Splitter that divides the frequency spectrum into two bands that each feed their own effects chain. You can add Components to the Low and High band independently, and freely adjust the crossover frequency that separates the bands. Additionally, each of the bands can be panned in the stereo image, further increasing the creative potential of the Freq Crossover.

> You can nest multiple Freq Crossover Components to further divide the frequency spectrum and create more complex effects.

The Freq Crossover contains the following parameters and controls:



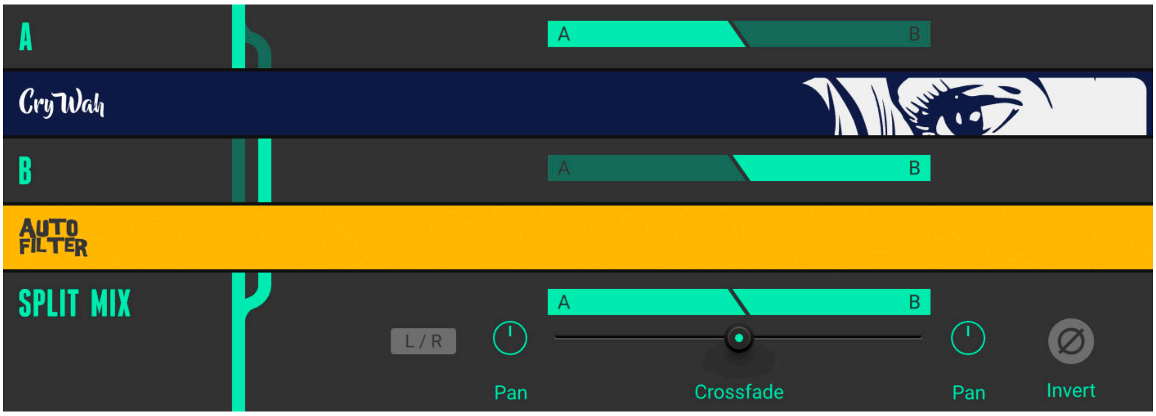
- **Low band**: Feeds the Low band into the Components inserted below.
- **High band**: Feeds the High band into the Components inserted below.
- **Frequency**: Adjusts the crossover frequency at which the frequency spectrum is divided into the Low and High bands.
- **Pan Low**: Adjusts the position of the Low band in the stereo image.
- **Crossfade**: Blends between the Low and High bands. Moving the fader to the left increases the level of the Low band and decreases the level of the High band, and vice versa.
- **Pan High**: Adjusts the position of the High band in the stereo image.
- **Invert**: Inverts the polarity, or phase, of the High band. When mixing the Low and High bands, this creates interesting phase cancellation effects.

14.4. Split Mix

Split Mix is a Splitter that feeds the input into two parallel effects chains. You can add Components to the A and B path independently, and freely adjust the both the panning and mix of the two paths before the output. Additionally, you can split the left and right stereo channels of the input and route them to the A and B path, respectively. This makes it possible to create stereo effects using any combination of Components.

> You can nest multiple Split Mix Components to increase the number of parallel effects chains.

The Split Mix contains the following parameters and controls:



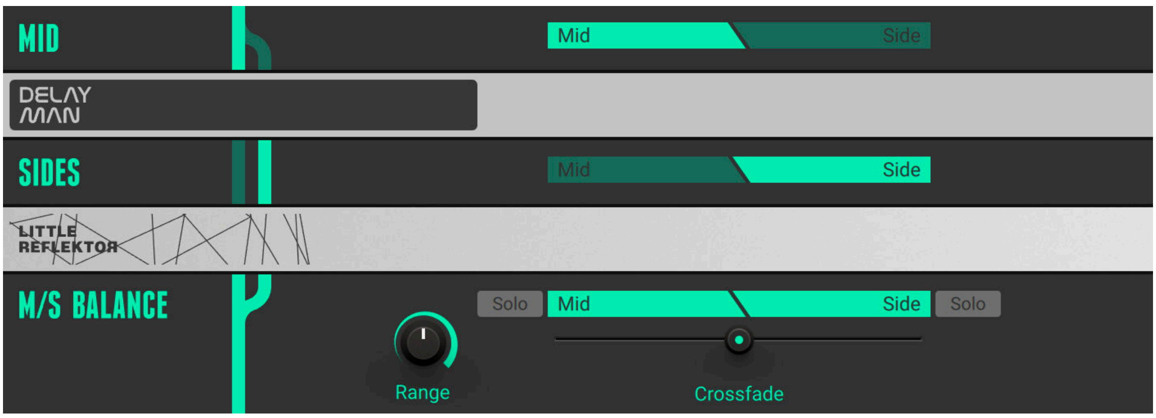
- **A path**: Feeds the A path into the Components inserted below.
- **B path**: Feeds the B path into the Components inserted below.
- **L/R split**: Routes the input's left stereo channel to the A path and the right stereo channel to the B path.
- **Pan A**: Adjusts the position of the A path in the stereo image.
- **Crossfade**: Blends between the A and B paths. Moving the fader to the left increases the level of the A path and decreases the level of the B path, and vice versa.
- **Pan B**: Adjusts the position of the B path in the stereo image.
- **Invert**: Inverts the polarity, or phase, of the B path. When mixing the A and B paths, this creates interesting phase cancellation effects.

14.5. M/S Balance

M/S Balance is a Splitter that separates the mid and side signals of the input, each feeding their own effects chain. The mid signal contains all information that is shared between the stereo channels. The side signal contains all information that is different between the stereo channels. In practical terms, the mid signal contains the direct sound of the instruments, while the side signal contains the reverb and other stereo effects. You can add Components to the Mid and Side band independently, and adjust the mix of the two bands before the output.

You can nest the Freq Crossover in the M/S Balance to process specific frequency bands both in the Mid and Side bands.

The M/S Balance contains the following parameters and controls:



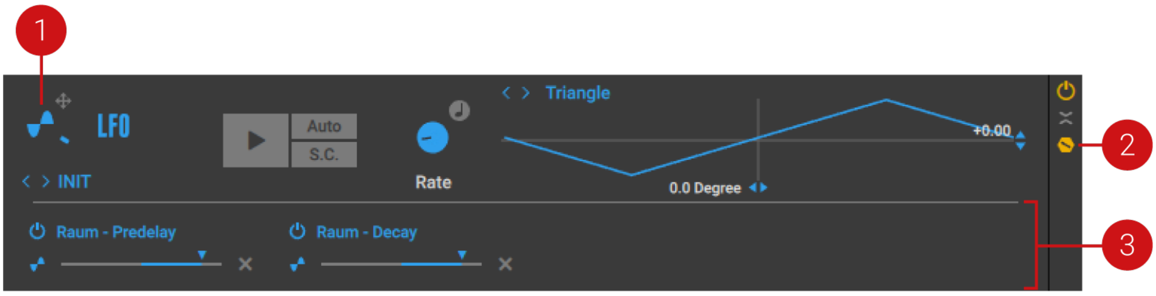
- **Mid band**: Feeds the Mid band into the Components inserted below.
- **Side band**: Feeds the Side band into the Components inserted below.
- **Range**: Adjusts the maximum attenuation of the **Crossfade** on either side of its range. Turning **Range** to the right increases the separation between the Mid and Side bands when using **Crossfade**.
- **Solo Mid**: Replaces the output with the direct signal from the Mid band.
- **Crossfade**: Blends between the Mid and Side bands. Moving the fader to the left increases the level of the Mid band and decreases the level of the Side band, and vice versa.
- **Solo Side**: Replaces the output with the direct signal from the Side band.

15. Modifiers

Modifiers are modulation sources that you can assign to any parameter in the Rack in order to control it. They do not generate or process sound directly, but instead provide a variety of functions to automatically adjust controls over time. Originating from synthesizers, this concept opens up vast possibilities in sound design.

Like on a synthesizer, the available modulation sources include an Envelope, an LFO, and sequencers. Additionally, the Input Level Modifier can be used as an envelope follower that tracks the input signal and transforms it into a modulation source.

The Modifiers share the following controls used for creating and editing assignments:



1. **Modifier icon**: Allows for assignment of the Modifier using drag and drop. For information, see Assigning Modifiers.
2. **Show Expert panel**: Shows the Expert panel that you can use to display and edit the assignments of the Modifier.
3. **Expert panel**: Displays the assignments of the Modifier and provides functions to edit them. For more information, see Editing Modifier Assignments.

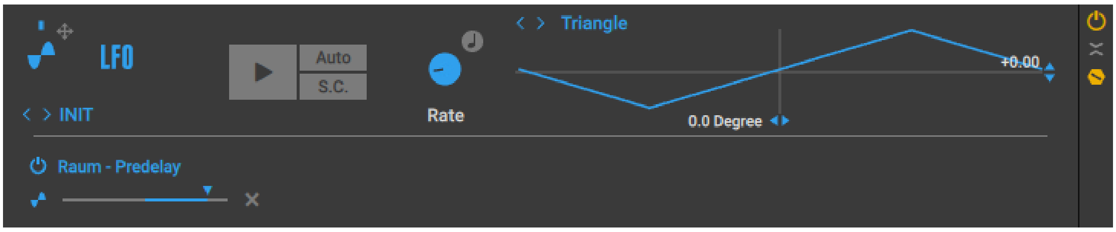
15.1. Assigning Modifiers

You can assign Modifiers to parameters using drag and drop. A single Modifier can be assigned to up to four parameters at the same time.

1. Click the Modifier icon or the target icon next to it, and keep the mouse button pressed.

2. Drag and drop the icon onto the parameter you want to assign the Modifier to.

→ The Modifier is assigned to the parameter, and the Expert Panel displays the assignment accordingly.



15.2. Editing Modifier Assignments

The Modifiers' Expert panel displays the assignments of the selected Macro and provides functions to deactivate, reassign, delete, and adjust the strength of assignments.

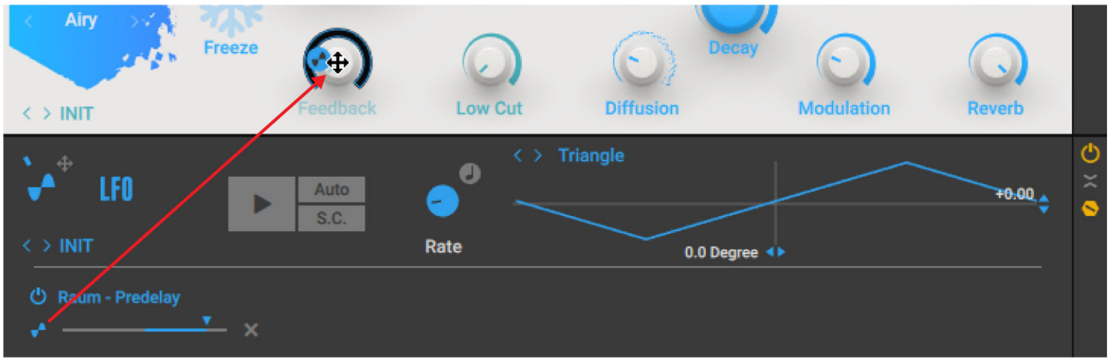
Deactivating Assignments

▶ To deactivate an assignment, click the Assignment On/Off button.

→ The assignment is deactivated and the assigned parameter will not be controlled by the Modifier. You can click the Assignment On/Off button again to activate the assignment.

Reassigning Assignments

▶ To reassign an assignment, drag and drop the Assignment number icon to the new parameter.



→ The assignment is removed from the original parameter and assigned to the new parameter.

Deleting Assignments

▶ To delete an assignment, click on the **x** button next to it.

→ The assignment is removed from the parameter.

Adjusting the Strength of Assignments

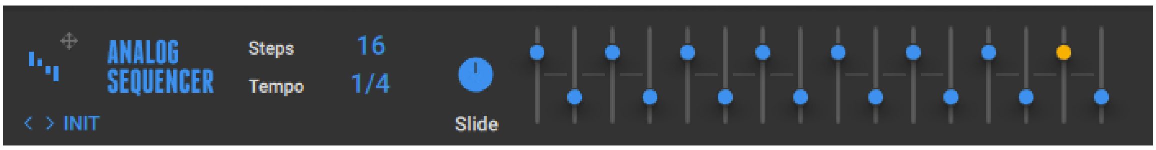
The strength of an assignment determines the response of the assigned parameters to the Modifier. Increasing the strength makes the Modifier control a wider range of the parameters it is assigned to.

▶ To adjust the strength of an assignment, move the value (triangle icon) on the strength slider.

15.3. Analog Sequencer

Analog Sequencer generates a sequence of up to 16 steps as control signal. Each step can have a different value, enabling you to program rhythmic parameter changes. The sequence is played back in a cycle and synchronized to the tempo of the Metronome.

This Component contains the following parameters and controls:



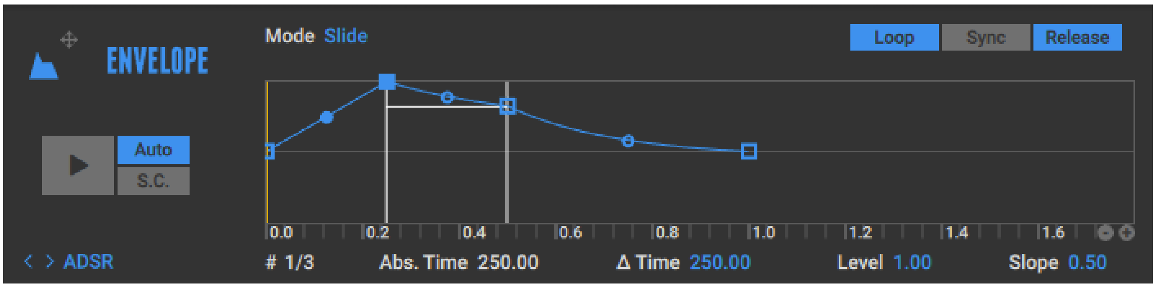
- **Steps**: Sets the number of steps in the sequence and therefore its length.
- **Tempo**: Sets the note length of each of the 16 steps. You can set 1/4, 1/8, 1/16, or 1/32 notes.
- **Slide**: Adjusts the amount of portamento, which is the time it takes the pitch to slide from one sequencer step to the next.

- **Sequence sliders**: Adjust the values of each of the steps in the sequence. In middle position, the step's value is 0. Moving the slider down sets negative values, moving the slider up sets positive values.

15.4. Envelope

The Envelope generates a very flexible control signal that is triggered by musical events. You can accurately edit the envelope's contour using a graphical editor. Once triggered, the envelope changes assigned parameters according to this contour. You can either use the input or the sidechain signal to trigger the envelope, or start it manually using the Play button.

This Component contains the following parameters and controls:



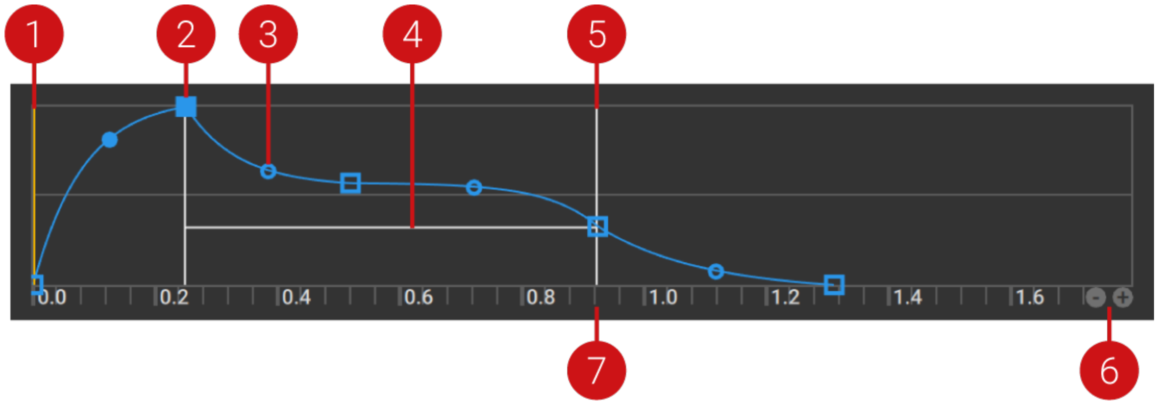
- **Play**: Triggers the envelope, which generates a control signal according to its contour.
- **Auto**: Activates automatic triggering based on the input signal. When you play a note on your instrument, the envelope detects the corresponding peak in the signal and starts playing.
- **S.C.**: Switches the signal used for automatic triggering to the global sidechain signal. **Auto** needs to be activated for this setting to have an effect. For more information about sidechaining, see Sidechaining.
- **Mode**: Switches between two modes for editing the envelope in the Graph, **Slide** and **Fixed**. When **Slide** is selected, moving a node in the Graph also changes the following nodes, therefore extending the total envelope time. When **Fixed** is selected, moving a node in the Graph does not change the other nodes, therefore keeping the total envelope time constant.
- **Loop**: Activates Loop mode. When activated, the sustain phase of the envelope plays in a loop once the envelope is triggered. Deactivating **Loop** stops playback. When both **Loop** and **Release** are activated, the loop plays for as long as the trigger remains active, for example when holding the Play button.
- **Sync**: Activates Sync mode. When activated, the envelope is synchronized to the tempo of the Metronome and the Graph is edited in musical intervals instead of the absolute duration (in seconds).
- **Release**: Activates Release mode. When activated, the release phase of the envelope plays as soon as the trigger is not active any more, for example when releasing the Play button. Additionally, the sustain phase only plays for as long as the trigger remains active, for example when holding the Play button.
- **Graph**: Displays and edits the envelope's contour. For more information, see Envelope Graph.
- **#**: Displays the number of the selected node in the graph, which can be edited using the **Time**, **Level**, and **Slope** controls.
- **Abs. Time**: Displays the duration in seconds from the start of the envelope to the selected node.
- **Time**: Adjusts the duration in seconds from the previous to the selected node.
- **Level**: Adjusts the level of the selected node.

- **Slope**: Adjusts the shape of the envelope's contour to the left of the selected node.

Envelope Graph

The Graph is a visual interface that enables you to view and edit the envelope's contour. The vertical axis represents the strength of the envelope, and the horizontal axis represents the duration of the envelope. The envelope itself is displayed as a blue curve. It consists of multiple nodes which can be moved in the Graph to adjust the envelope's contour.

The following overview highlight the various interface elements of the Graph:



1. **Position marker**: Shows the current playback position of the envelope.
2. **Node**: Divides the envelope into segments that are used to adjust the envelope's contour. You can click and drag them in the Graph to change their level and position in time. When **Sync** is activated, the nodes snap onto the musical grid. You can add new nodes by right-clicking anywhere on the envelope curve, and remove them by right-clicking on a node. You can select a node and change its parameters using the controls beneath the Graph by clicking on it. The selected node is shown as a solid blue square, while the other nodes only show an outline.

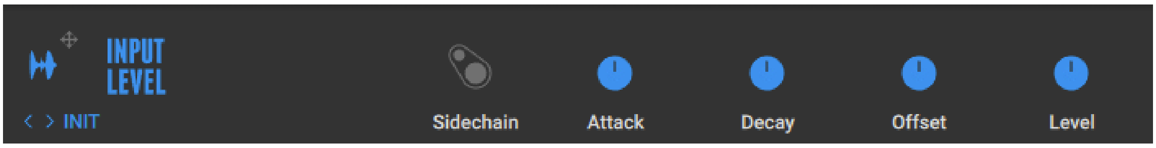
    The first node represents the start of the envelope, and the last node represents the end of the release phase. The first and the last nodes share the same level.

3. **Slope**: Adjusts the shape of the envelope in the respective segment. You can click and drag the circular handle up and down to set logarithmic, linear, and exponential responses.
4. **Sustain phase**: Displays the sustain phase of the envelope. The sustain phase can be played back as a loop when **Loop** is activated, or play for as long as a trigger is active when **Release** is activated. The segment before the sustain phase represents the attack phase. The segment after the sustain phase represents the release phase.
5. **Sustain markers**: Display and edit the start and end points of the sustain phase. You can drag the sustain markers left and right in order to attach them to different nodes. They cannot be attached to the first and the last node.
6. **Zoom**: Sets the time scale of the Graph. You can click on **-** to zoom out, and on **+** to zoom in. When **Sync** is activated, Zoom also sets the resolution of the musical grid that the nodes snap onto.
7. **Time scale**: Shows the time scale of the Graph in seconds. When Sync is activated, it shows musical intervals. You can scroll in the Graph by clicking and dragging the time scale, or fit the envelope in the Graph by double-clicking on the time scale.

15.5. Input Level

The Input Level Modifier generates a control signal based on the incoming input level. This is commonly called an "envelope follower" as the signal tracks changes in amplitude. The most common application for this is to control filters; this is already built in to the AutoFilter component. However, a subtle use of this Modifier can be used to make components like the amplifiers sound much more realistic.

This Component contains the following parameters and controls:

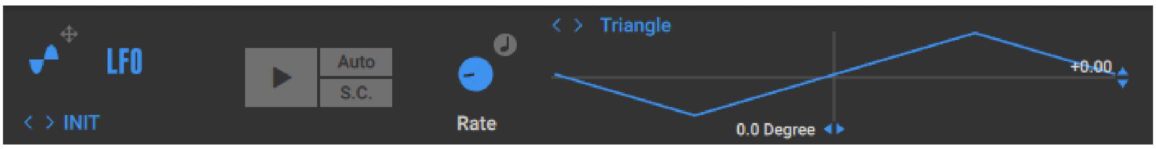


- **Sidechain**: Allows the user to trigger the Envelope with an external source signal. **Auto** must be activated to use **Sidechain**.
- **Attack**: Sets how long the control signal takes to reach its target value (from 1 to 978ms). Setting the attack time too short can create pops when the signal first kicks in; increasing Attack can soften this effect.
- **Decay**: Sets how long the control signal takes to fall back to its initial setting in the absence of an input signal (from 10 to 9863ms).
- **Offset**: Sets the control signal's minimum value. By default, the control signal covers a range from –1 to +1, with 0 as the midpoint. When Offset is at maximum, the baseline is 0.
- **Level**: Sets the overall strength of the control signal.

15.6. LFO

The term LFO (Low Frequency Oscillator) refers to a periodic waveform at a subsonic rate. As a Modifier, this is useful to create constantly changing parameter values.

This Component contains the following parameters and controls:

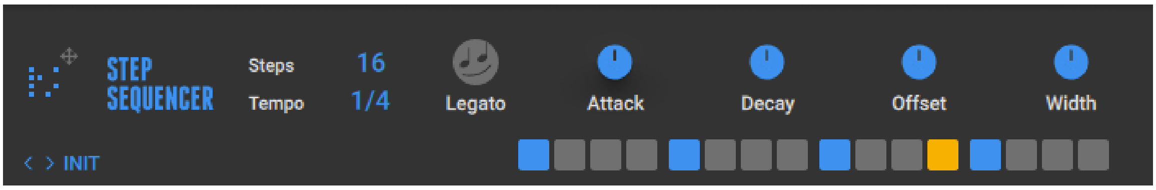


- **Play**: Resets the LFO and starts its cycle from the beginning.
- **Auto**: Activates automatic reset based on the input signal. When you play a note on your instrument, the LFO detects the corresponding peak in the signal and starts its cycle from the beginning.
- **S.C.**: Restarts the Envelope with an external source signal. **Auto** must be activated to use **Sidechain**.
- **Rate**: Sets the LFOs frequency in the range of 0.01 Hz to 10.24 Hz. When Tempo Sync is activated, **Rate** is set in note values.
- **Tempo Sync**: Synchronizes the LFO to the tempo of the Metronome. When activated, **Rate** is set in note values.
- **Waveform selector**: Selects between sine, triangle, square, saw tooth, and random waveforms.
- **Waveform display**: Displays the selected waveform and provides controls for setting the reset phase position (horizontal slider below the display) and the level offset (vertical slider to the right of the display).

15.7. Step Sequencer

The Step Sequencer generates up to 16 sequential control signals, which can have rhythmic values of quarter notes to 1/32nd notes. Note that you cannot set the level of these steps; they are on/off triggers intended to create rhythmic effects. The horizontal bar consisting of 16 numbered buttons is the core of the Step Sequencer. The buttons are addressed from 1 to 16 as shown by an orange outline, moving synchronously to the beat of the Metronome. Active buttons are blue and trigger the control signal for the assigned controls as defined in the Targets list. Clicking on the buttons turns them on and off.

This Component contains the following parameters and controls:



- **Steps**: Adjusts the length of the sequence by reducing the number of steps.
- **Tempo**: Sets the rhythmic value, effectively changing the speed at which the buttons are triggered. Available options are 1/4, 1/8, 1/16, and 1/32 notes.
- **Legato**: Ties adjacent steps to each other, thus creating longer steps.
- **Attack**: Controls how long the control signal takes to reach its maximum value after having been triggered (from 4 to 1233ms).
- **Decay**: Controls how long the control signal takes to reach its minimum value after having been triggered (from 4 to 2197ms)
- **Offset**: Increases all control signal values as the control when turned up.
- **Width**: Sets the length of the control signal, when triggered. When fully clockwise, the width equals the note-value set as Resolution.

16. Sidechaining

Sidechaining enables the use of an external audio signal to control Components independently of the input signal. It is commonly used on compressors, where the sidechain signal is used to control the amount of compression applied to the input signal. This makes it possible to create ducking and gating effects.

GUITAR RIG features a stereo sidechain input that receives the global sidechain signal. Components with sidechaining functionality can use the signal received at this input as the control signal. The following Components include sidechaining functionality:

- **Dynamics**: Fast Comp, Solid Bus Comp, Solid Dynamics, Stomp Compressor, Tube Compressor, VC160, VC2A, VC76
- **Filters**: Auto Filter
- **Modulation**: Freak
- **Modifier**: Envelope, Input Level, LFO

Using Modifiers it is possible to process the sidechain signal and use it to control any parameter in your Rack. For example, by assigning the Input Level Component in sidechain mode to the gain of a low-shelf equalizer, you can use a kick drum signal to cut low-frequency content of your bass guitar when the kick drum hits.

Sidechaining is available in both the GUITAR RIG stand-alone application and the plug-in. While the stand-alone application facilitates routing of the sidechain input in its Preferences, the plug-in relies on the routing functionality of the DAW it is hosted in. For more information, see Sidechaining in the Stand-Alone Application and Sidechaining in the Plug-in.

16.1. Sidechaining in the Stand-Alone Application

To use sidechaining in the GUITAR RIG stand-alone application, you need to configure the sidechain input in the Preferences.

1. Click the Main menu in the Header and open **Preferences...** in the **File** sub-menu.
2. In the Preferences, go to the **Audio** tab and click on **Inputs** in the lower section.
3. Choose the desired inputs of your audio interface in the **Guitar Rig 6 SideChain L** and **Guitar Rig 6 SideChain R** drop-down menus.

→ The signal received from the chosen inputs is now available as global sidechain signal in GUITAR RIG.

Components providing sidechaining functionality feature a special mode that replaces the internal control signal with the global sidechain signal.

▶ To activate sidechaining functionality, click on the Sidechain or S.C. button on the Component.

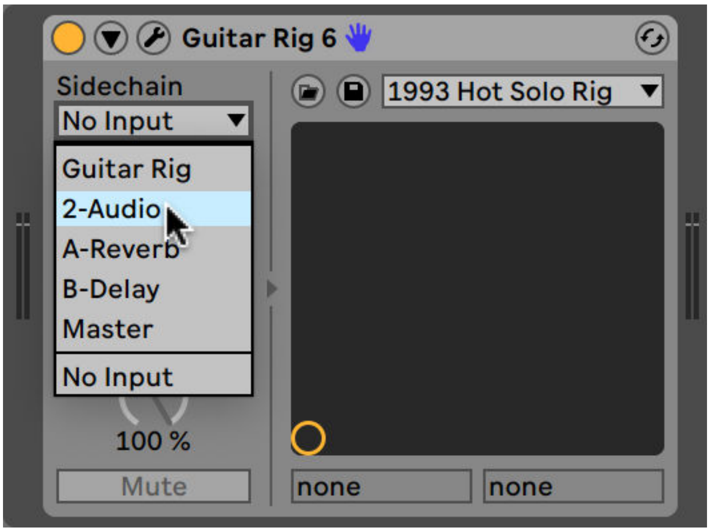
→ The internal control signal of the Component is replaced with the global sidechain signal.

16.2. Sidechaining in the Plug-in

To use sidechaining in the GUITAR RIG plug-in, you need to use your DAW's routing functionality to configure the sidechain input. The plug-in reports the availability of this input to the DAW, which can show it as part of its routing options.

For example, Ableton Live shows sidechain routing as part of the Device in the Detail view:

▶ Choose the desired input track from your Live Set in the **Sidechain** drop-down menu.



→ The signal received from the chosen input track is now available as global sidechain signal in GUITAR RIG.

ⓘ Please refer to the documentation of your DAW to learn more about its routing functionality and how to configure sidechain inputs.

Components providing sidechaining functionality feature a special mode that replaces the internal control signal with the global sidechain signal.

▶ To activate sidechaining functionality, click on the Sidechain or S.C. button on the Component.

→ The internal control signal of the Component is replaced with the global sidechain signal.

17. Automation and MIDI Control

GUITAR RIG can be controlled externally using automation in the DAW and MIDI controllers. Streamlined functionality for managing automation IDs and MIDI assignments allows you to configure external control according to your needs. GUITAR RIG is also fully integrated with KOMPLETE KONTROL and MASCHINE, using NKS (Native Kontrol Standard) to automatically map parameters to the hardware controls.

17.1. Automation

Automation is used to control parameters in GUITAR RIG from the DAW, for example by recording or drawing automation data in the DAW's timeline. The corresponding parameters read the automation data during playback and are adjusted automatically. You can use this to level out the sound according to changes in your recording, or for creative sound design by animating key parameters in real time. The GUITAR RIG plug-in facilitates automation of parameters by using a combination of fixed and dynamically mapped automation IDs in the automation list reported to the DAW.

Fixed automation IDs are always mapped to the same parameters, for example Macros. They are independent of the Rack content and the preset, meaning your automation will still function correctly even if you change the Rack or load a new preset. Dynamically mapped automation IDs are used for parameters of individual Components in the Rack. Since they automatically adapt to changes in the Rack, these automation IDs are specific to each individual preset and the current state of the Rack.

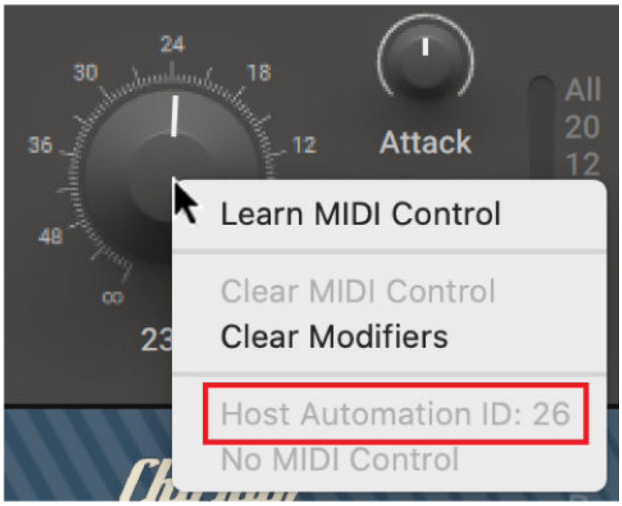
ⓘ When a parameter is assigned to a Macro control, it ignores automation data received on its own automation ID in order to prevent conflicting parameter changes.

The automation IDs in the automation list are grouped according to the function of the respective parameters, and whether their mapping is fixed or dynamic:

- **Automation ID 1-16**: Fixed mapping to the Macro controls 1-16.
- **Automation ID 17-256**: Dynamic mapping to the parameters of Components in the Rack.
- **Automation ID 257-276**: Fixed mapping to the parameters in the Header.
- **Automation ID 277-384**: Fixed mapping to the parameters of the Rack Tools.
- **Automation ID [385-512]**: Dynamic mapping to the parameters of Components in the Global FX.

You can look up the automation ID of a specific parameter in the Rack by using the context menu.

▶ Right-click a parameter to show its automation ID in the context menu.

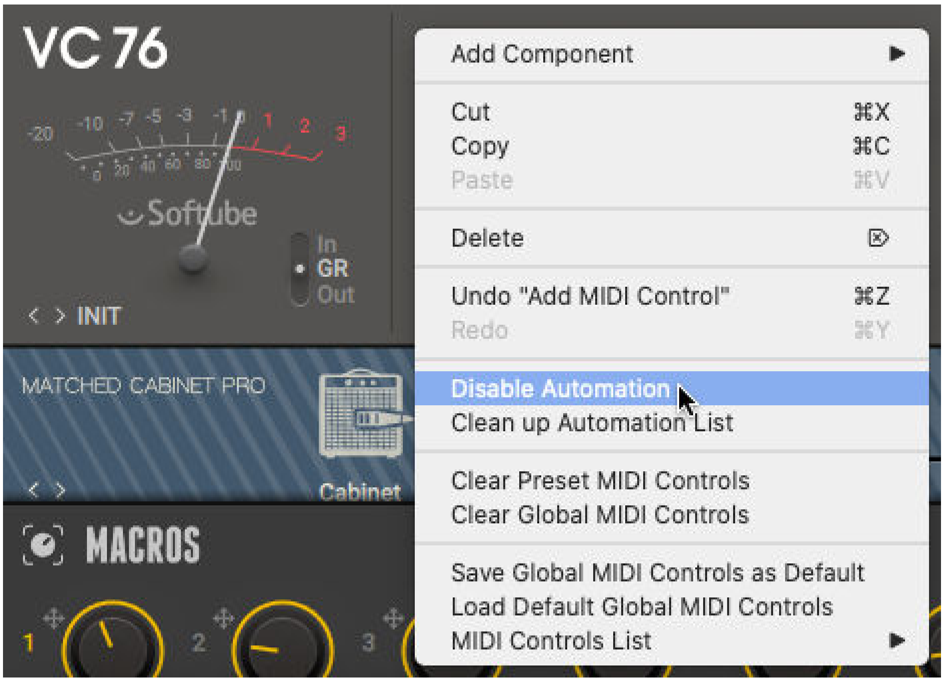


Deactivating Automation for Components

You can deactivate automation for individual Components in the Rack. When automation is deactivated for a Component, all of its parameters are removed from the automation list reported to the DAW and stop responding to incoming automation data.

> You can use this function before cleaning up the automation list in order to reduce the amount of automation IDs and focus on the parameters you require in your project. For more information, see Cleaning up the Automation List.

▶ To deactivate automation for a Component, right-click the Component and select **Disable Automation** from the context menu.



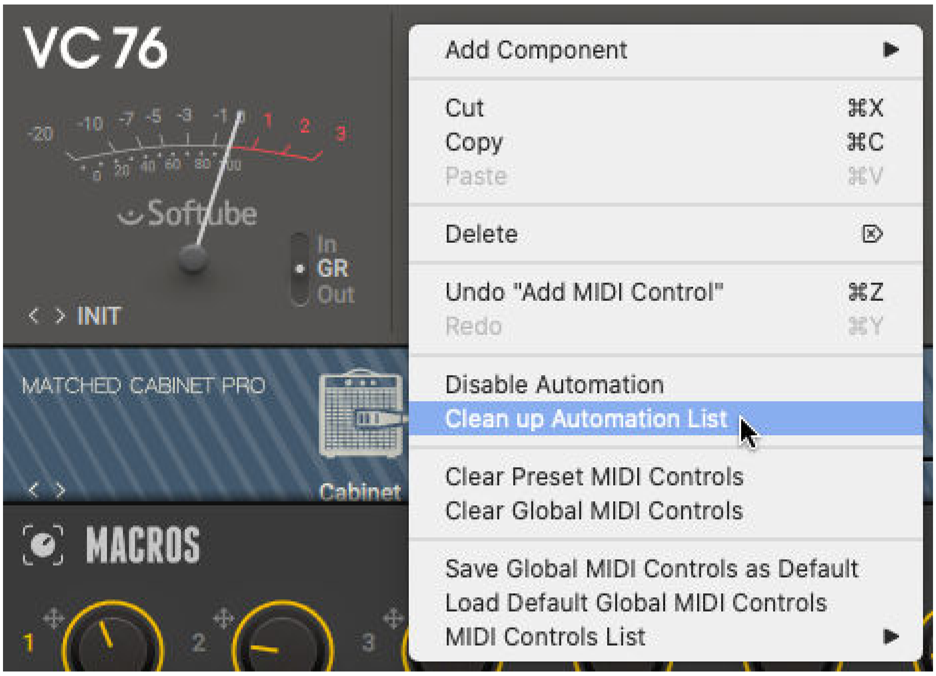
Cleaning up the Automation List

You can consolidate all dynamically mapped automation IDs to clean up the automation list reported to the DAW. The corresponding function maps the automation IDs to the parameters in the Rack from top to bottom. The Macros, Header, and Rack Tools use fixed automation IDs for their parameters and therefore are not affected by this function.

> ⓘ You can deactivate automation for individual Components before cleaning up the automation list in order to reduce the amount of automation IDs and focus on the parameters you require in your project. For more information, see Deactivating Automation for Components.

▶ To clean up the automation list, right-click the Component and select **Clean up Automation List** from the context menu.

> ⚠ This function maps new automation IDs to the parameters of Components in the Rack. If you have already automated these parameters in your project, the automation will not work correctly any more.



17.2. MIDI Control

Parameters in GUITAR RIG can be controlled using MIDI controllers, for example foot controllers and keyboards. Once a MIDI control is mapped to a parameter, you can use the hardware control to adjust the value of the parameter. This enables you to control the software intuitively and remotely, for example during a performance. GUITAR RIG facilitates mapping of MIDI controls by providing MIDI learn and a streamlined set of functions to manage MIDI mappings directly in the Rack.

MIDI controls can be mapped to individual parameters of Components in the Rack. The mappings are stored in each individual preset. Additionally, MIDI controls can be mapped to global controls, for example preset selection and the Macros. In this case, mappings are stored globally, independently of the loaded preset. Global MIDI mappings are especially beneficial when using Macros. The MIDI mappings will function with any preset you load, and you can still assign the Macros to different parameters in each preset.

MIDI control via Macros enables you to set the exact range of a parameter you want to control. For more information, see Editing Macros.

The following parameters use global MIDI mappings:

- Preset selection in the Toolbar
- Parameters of the Rack Tools including Macros
- Parameters of Components in Global FX

The following parameters use individual MIDI mapping per preset:

- Parameters of Components in Rack
- Parameters of Components in Container

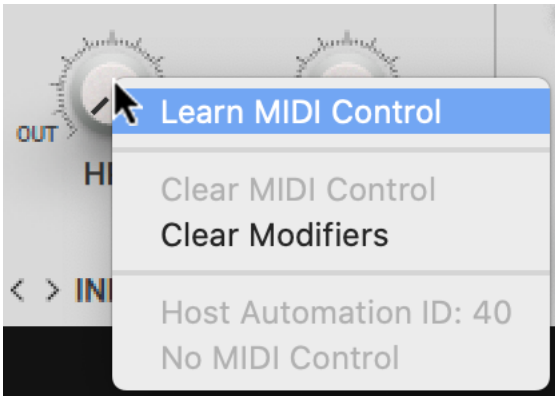
Using MIDI Learn

You can map MIDI controls to parameters using MIDI learn. This function automatically recognizes incoming MIDI data, enabling you to create MIDI mappings simply by turning the hardware controls on your MIDI controller.

Your MIDI controller needs to be activated in the Preferences. There you can also set the MIDI channel for receiving MIDI data. For more information, see MIDI.

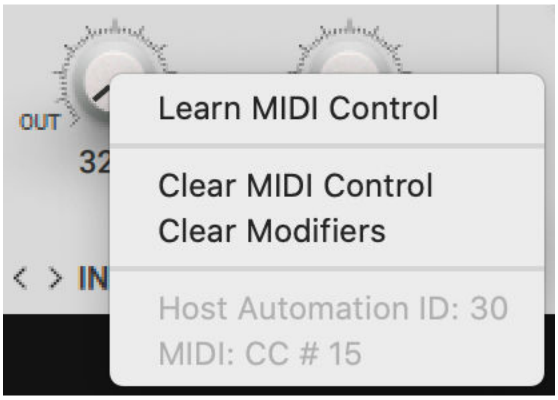
To map a MIDI control to a parameter:

1. Right-click the parameter and select **Learn MIDI Control** in the context menu.



2. Activate the MIDI control you want to map by turning the knob, moving the fader, or pushing the button on your MIDI controller.

→ The MIDI control is mapped and can be used to adjust the parameter. The MIDI CC (Continuous Controller) number of the mapped MIDI control is shown in the parameter's context menu.



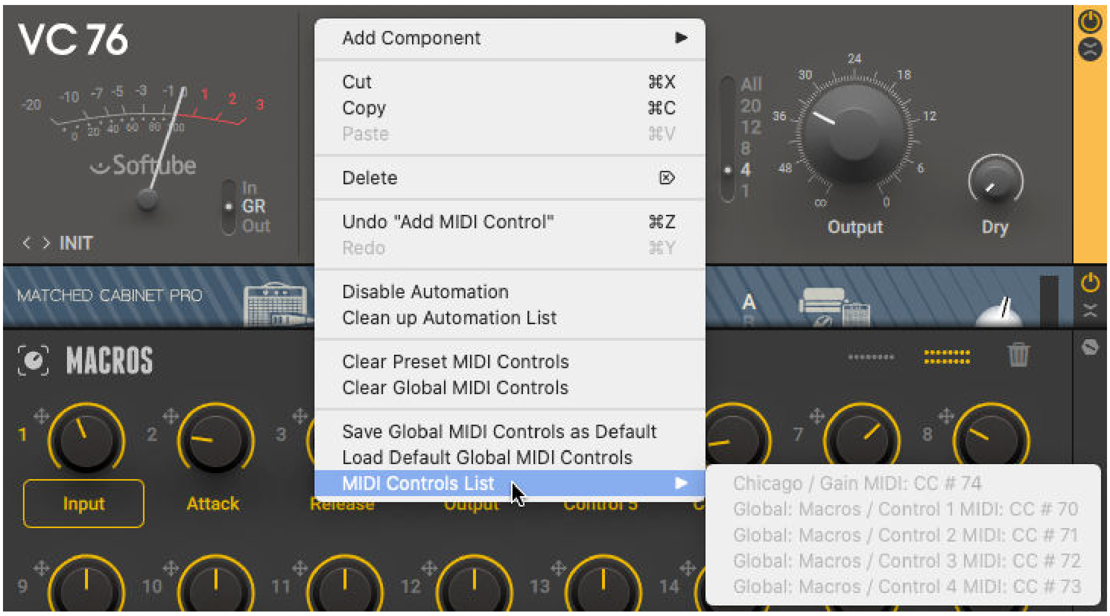
Managing MIDI Mappings

The Rack's context menu provides functions to manage MIDI mappings. You can view all MIDI mappings, save and load default global MIDI mappings, and clear MIDI mappings.

Viewing MIDI Mappings

You can view all existing MIDI mappings using the Rack's context menu.

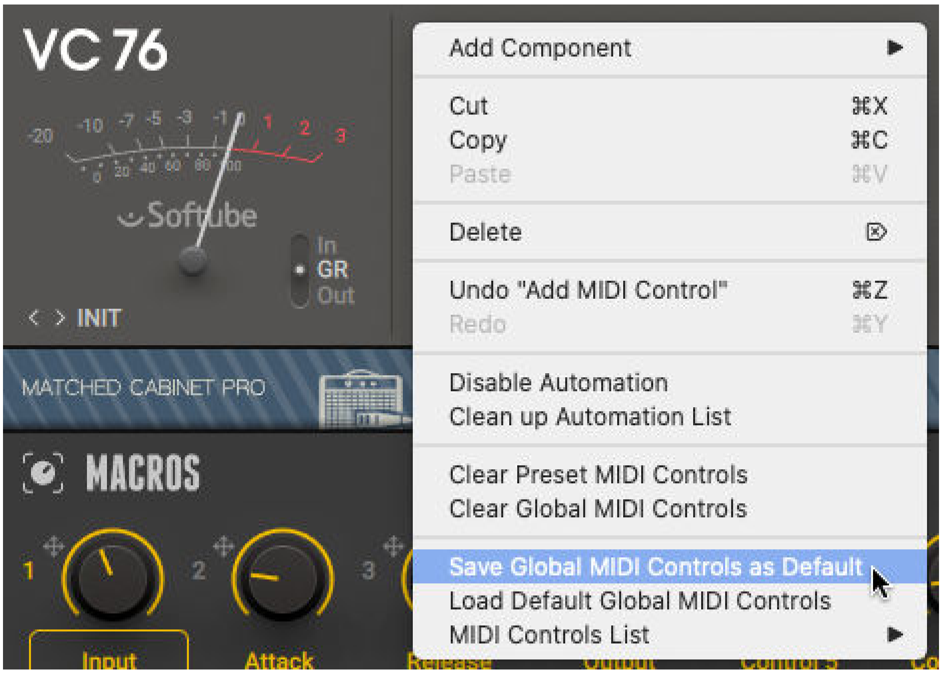
▶ To view all MIDI mappings, right-click on the background of a Component or the Rack and select **MIDI Controls List** in the context menu.



Saving Global MIDI Controls as Default

You can save the global MIDI mappings as default. This enables you to load them at any point and restore your standard MIDI controls that will work with any preset.

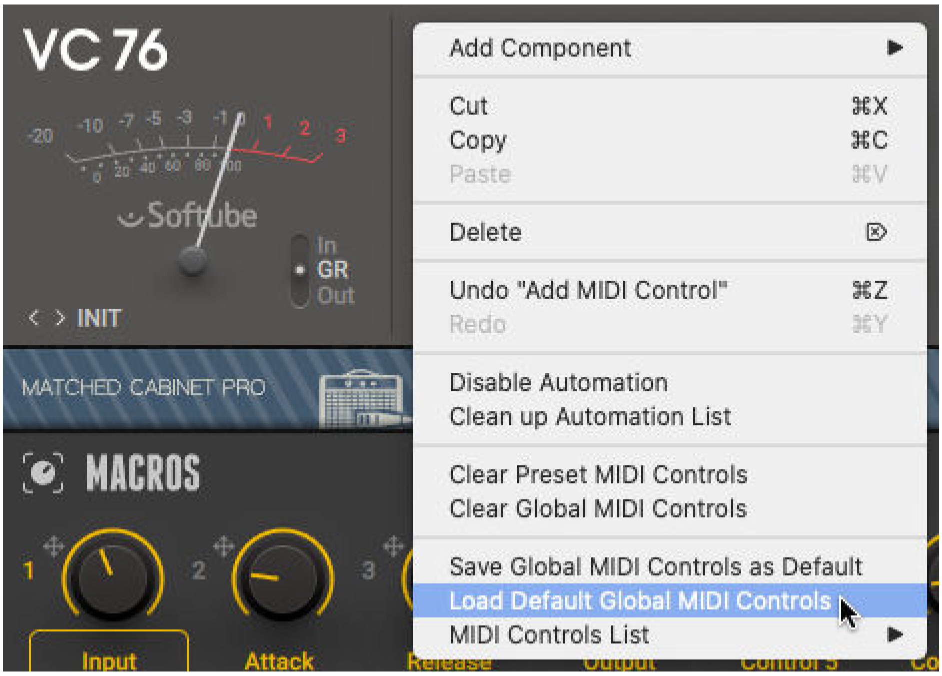
▶ To save the global MIDI mappings as default, right-click on the background of a Component or the Rack and select **Save Global MIDI Controls as Default** in the context menu.



Loading the Default Global MIDI Controls

You can load global MIDI mappings that you have previously saved as default and thus restore your standard MIDI controls that will work with any preset.

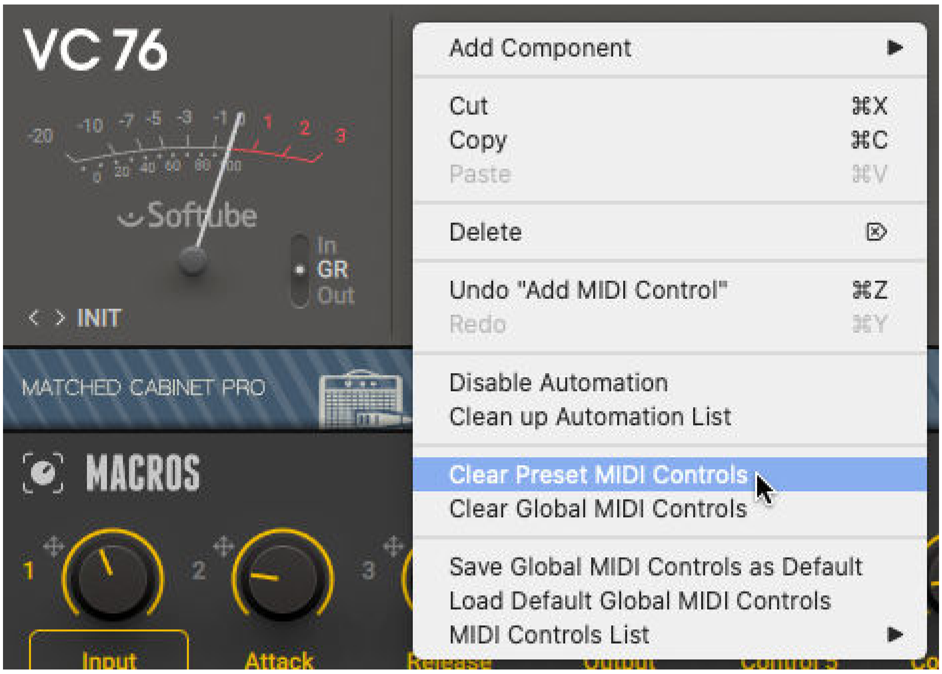
▶ To load the default global MIDI mappings, right-click on the background of a Component or the Rack and select **Load Default Global MIDI Controls** in the context menu.



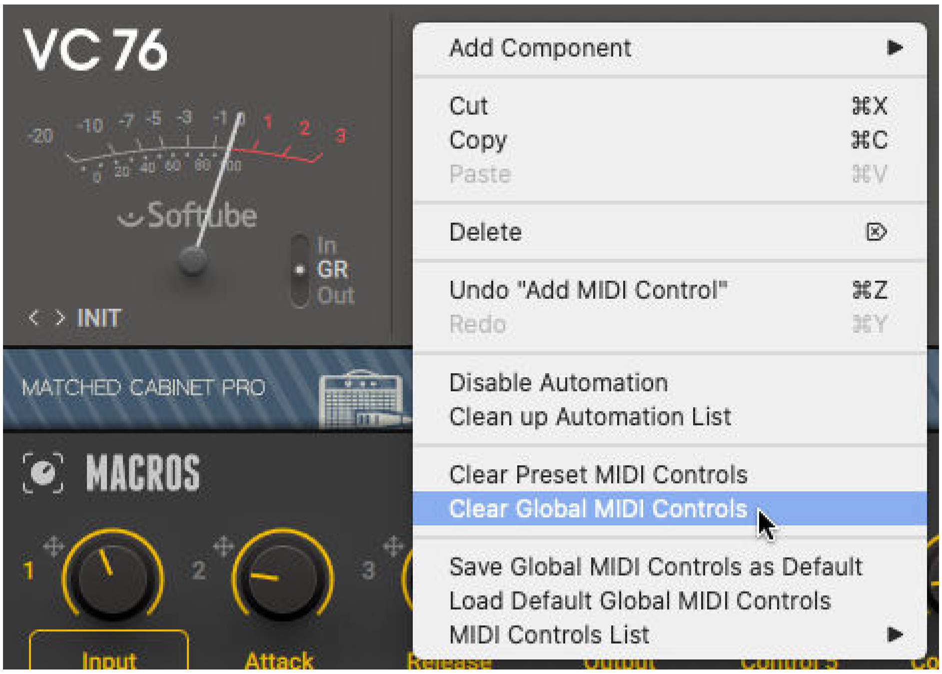
Clearing MIDI Controls

You can use a single command to clear either all individual MIDI mappings of a preset or all global MIDI mappings.

▶ To clear the individual MIDI mappings of the preset, right-click on the background of a Component or the Rack and select **Clear Preset MIDI Controls** in the context menu.



▶ To clear the global MIDI mappings, right-click on the background of a Component or the Rack and select **Clear Global MIDI Controls** in the context menu.

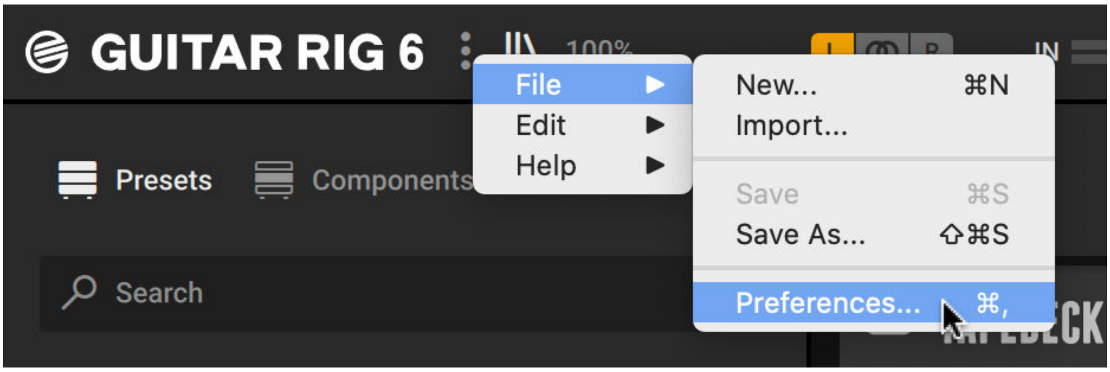


18. Preferences

The Preferences provide settings and options related to the general behavior of the software, audio, MIDI, and the preset library. You can access the Preferences through the Main menu in the Header.

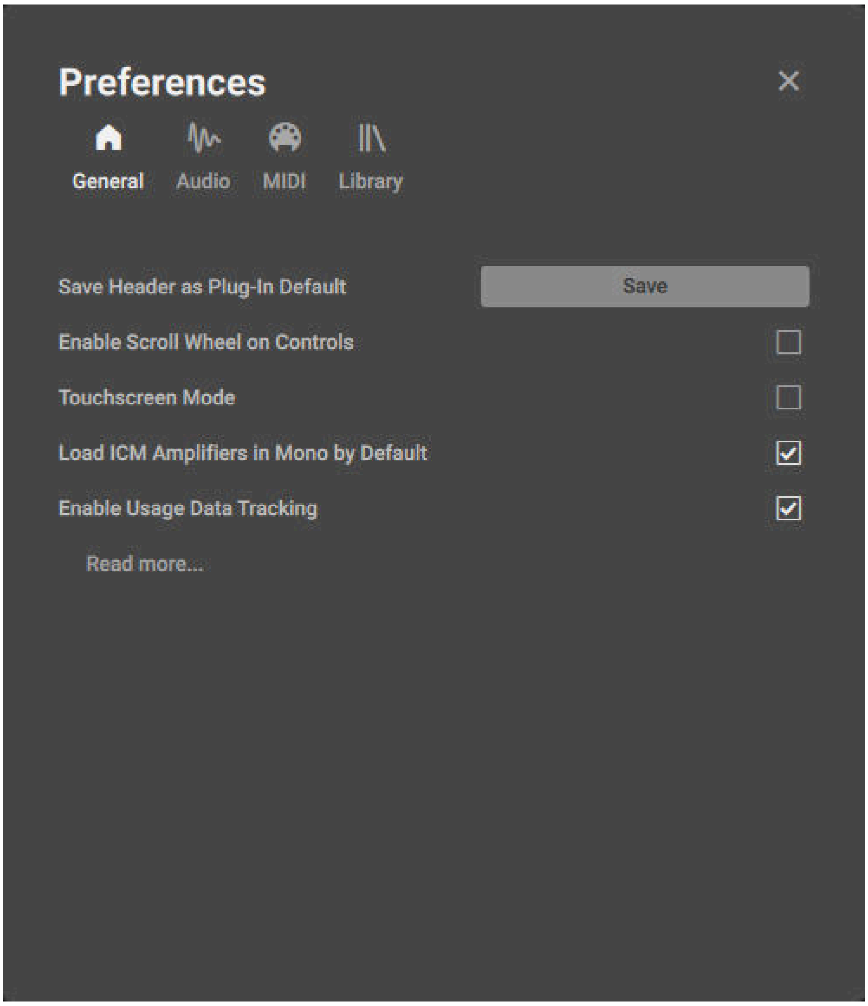
▶ To open the Preferences, click on the Main menu in the Header and select **Preferences...** from the **File** sub-menu.

> Alternatively, you can press [command] + [,] (macOS) or [Ctrl] + [,] (Windows) on the keyboard. For more information, see Keyboard Shortcuts.



18.1. General

This tab in the Preferences contains the following settings and options:

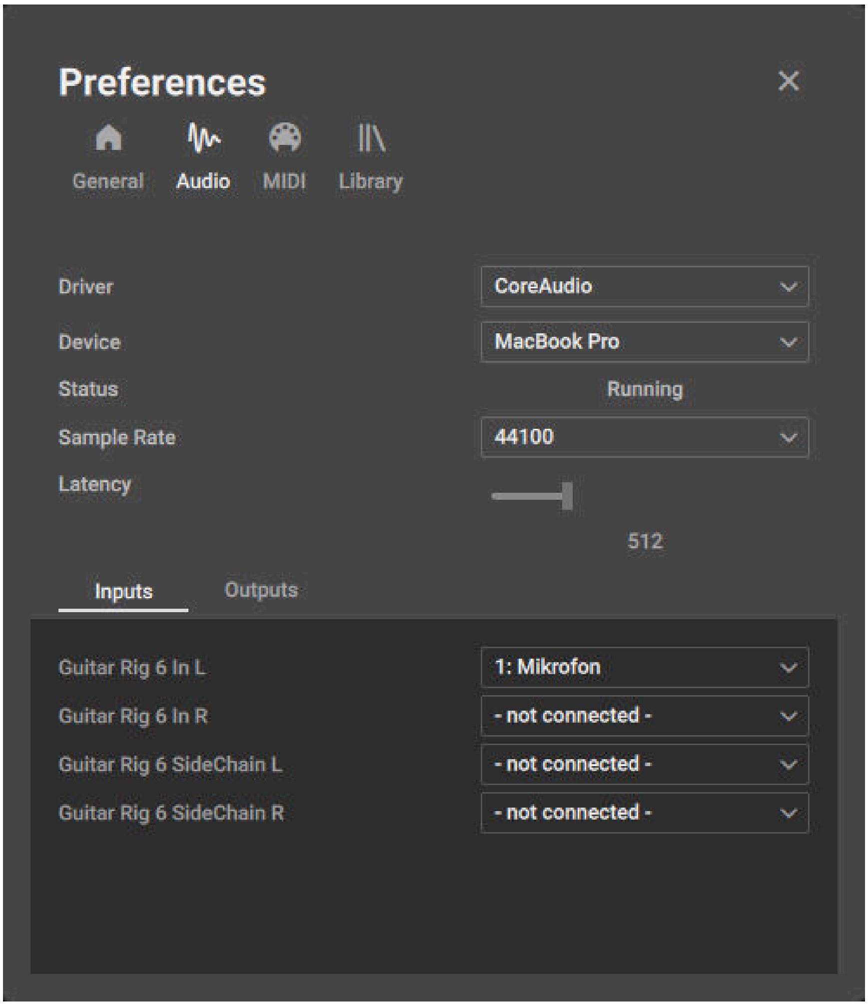


- **Save Header as Plug-in Default**: Stores the current settings in the Header as default. The default settings are restored when opening an instance of the GUITAR RIG plug-in.
- **Enable Scroll Wheel on Controls**: Sets the option to use the scroll wheel for adjusting parameters. When activated, you can place the mouse cursor over a control and turn the scroll wheel to adjust it.
- **Touchscreen Mode**: Sets the option to make the user interface compatible with touch screens and pen tablets.
- **Load ICM Amplifiers in Mono by Default**: Sets the option to load amplifiers based on ICM technology in mono configuration in order to reduce the CPU load. When activated, adding the Component to the Rack automatically deactivates the Stereo option. For more information, see Component Controls.
- **Enable Usage Data Tracking**: Sets the option to track and send anonymous usage data. When activated, this data is sent to Native Instruments, helping us improve our products.

18.2. Audio

This tab in the Preferences contains the following settings and options:

> ⓘ The Audio tab is only available in the GUITAR RIG stand-alone application.



- **Driver**: Selects the type of device driver used to communicate with the audio interface.
- **Device**: Selects the audio interface you want to use for routing audio into and out of GUITAR RIG.
- **Status**: Displays the status of the connection with the audio interface.
- **Sample Rate**: Selects the sample rate of the audio input and output, as well as the audio processing. High sample rate settings improve the audio quality but increase the CPU load.
- **Latency**: Selects the buffer size used for the audio processing. High latency settings reduce the CPU load but increase the time it takes to process the audio input, which can cause a noticeable lag when playing in real-time.

> For information about optimizing the audio settings and your computer system, see the following articles on our website:
> - Mac Tuning Tips for Audio Processing
> - Windows Tuning Tips for Audio Processing

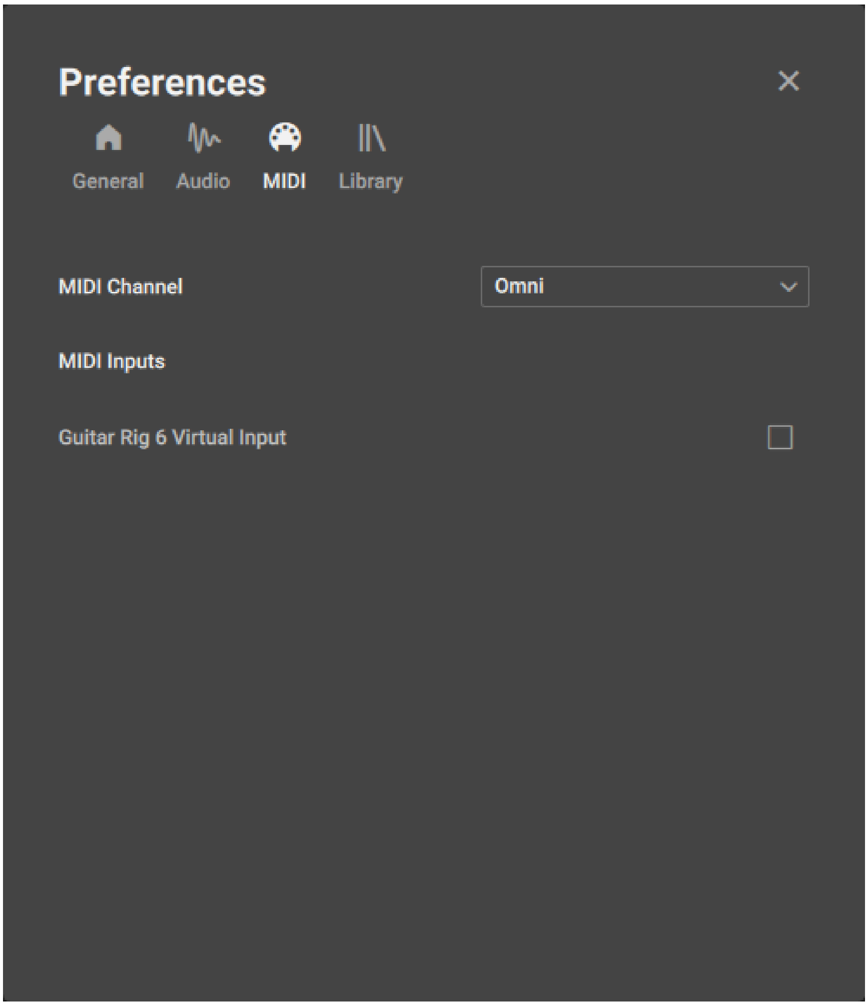
- **Inputs**: Provides menus to select the inputs of your audio interface used in GUITAR RIG.

- **Outputs**: Provides menus to select the outputs of your audio interface used in GUITAR RIG.

> For information about setting the inputs and outputs in the GUITAR RIG stand-alone application, see Setting up GUITAR RIG as Stand-Alone Application.

18.3. MIDI

This tab in the Preferences contains the following settings and options:

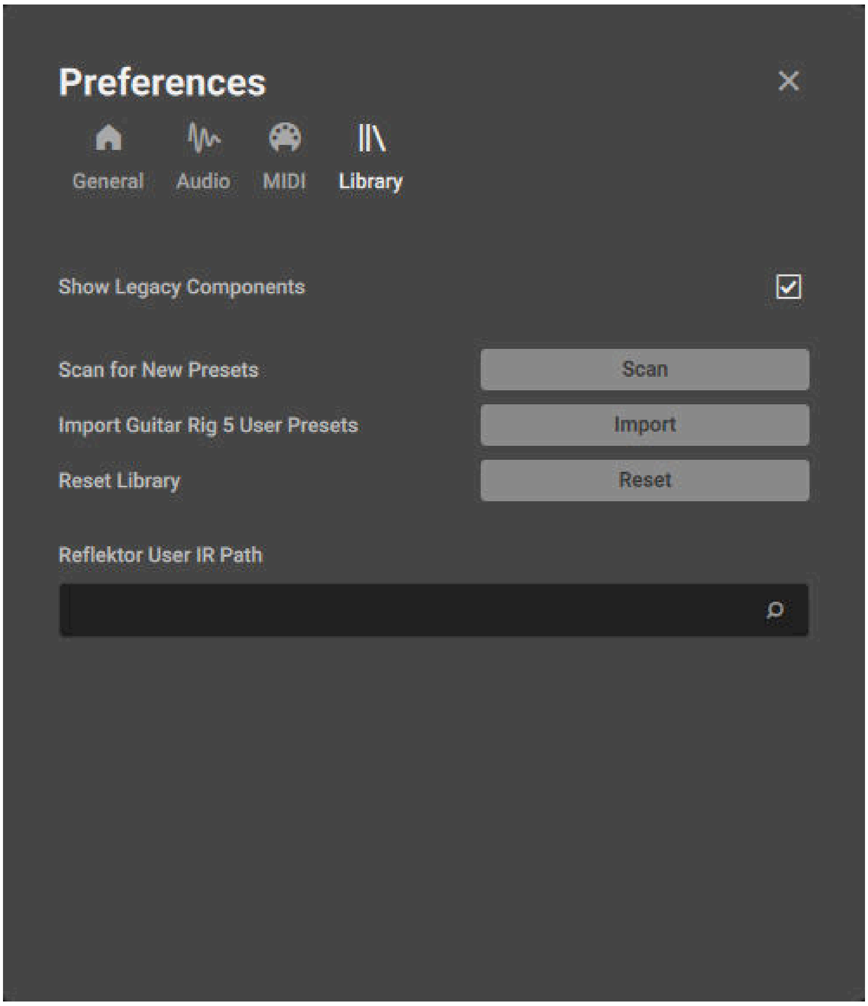


- **MIDI Channel**: Selects the MIDI channel used to receive MIDI data in GUITAR RIG. When set to **Omni**, MIDI data from all MIDI channel is received.
- **MIDI Inputs**: Provides options to activate or deactivate MIDI devices in GUITAR RIG. MIDI data can only be received from MIDI devices that are activated.

> The MIDI inputs are only available in the GUITAR RIG stand-alone application.

18.4. Library

This tab in the Preferences contains the following settings and options:



- **Show Legacy Components**: Sets the option to show Components from the Legacy category in the Browser. When activated, you can view and add Components from this category using the Component Tiles. For more information, see Legacy.
- **Scan for New Presets**: Scans the GUITAR RIG library for any changes made in other running instances of the software.
- **Import Guitar Rig 5 User Presets**: Imports User presets from an existing GUITAR RIG 5 installation on the same system.
- **Reset Library**: Resets the GUITAR RIG library to its default state.
- **Reflektor User IR Path**: Sets the path of the folder on the hard drive containing your user impulse responses for REFLEKTOR. For more information, see REFLEKTOR.

19. Keyboard Shortcuts

The following keyboard shortcuts are available in GUITAR RIG:

Command	macOS	Windows
Create new preset	[command] + [N]	[Ctrl] + [N]
Save User preset	[command] + [S]	[Ctrl] + [S]
Save User preset as	[command] + [shift] + [S]	[Ctrl] + [Shift] + [S]
Quit GUITAR RIG	[command] + [Q]	[Ctrl] + [Q]
Open Preferences	[command] + [,]	[Ctrl] + [,]
Increase View Size	[command] + [+]	[Ctrl] + [+]
Decrease View Size	[command] + [-]	[Ctrl] + [-]
Undo Rack edit	[command] + [Z]	[Ctrl] + [Z]
Redo Rack edit	[command] + [Y]	[Ctrl] + [Y]
Cut selected Components in Rack to clipboard	[command] + [X]	[Ctrl] + [X]
Copy selected Components in Rack to clipboard	[command] + [C]	[Ctrl] + [C]
Paste Components from clipboard into Rack	[command] + [V]	[Ctrl] + [V]
Delete selected Components in Rack	[delete]	[Del]
Select all Components in Rack	[command] + [A]	[Ctrl] + [A]
Show or hide Browser	[command] + [B]	[Ctrl] +[B]
Show or hide Info pane	[I]	[I]
Focus next element in Browser	[tab]	[Tab]
Focus previous element in Browser	[shift] + [tab]	[Shift] + [Tab]
Select focused element in Browser	[space]	[Space]
Select next preset in Results list	[arrow up]	[Arrow up]
Select previous preset in Results list	[arrow down]	[Arrow down]
Load selected preset from Results list	[enter]	[Enter]
Confirm Save New Preset dialog	[enter]	[Enter]
Cancel Save New Preset dialog	[escape]	[Esc]

20. Components Reference

In this chapter you can explore all the Components that come with GUITAR RIG and make yourself familiar with their general functionality and their controls.

20.1. Amplifiers

The amplifier is integral to the tone of an electric guitar or bass. GUITAR RIG provides you with an extensive selection of classic amplifiers spanning decades from the 1950s to the present. The Amplifier Components in GUITAR RIG are meticulously modeled after the sound and behavior of the original devices, including the interaction between their controls. Additionally, Expert panels provide controls that further expand the sonic possibilities.

AC Box

The AC Box models the sound that powered British pop music. This particular model stands out with a unique flavor and the Brilliant channel, famously known as top boost. The Normal channel features a Tone Cut control that reduces high-frequency content, while the Brilliant channel offers Bass and Treble controls. Both channels can be mixed for a great variety of sounds.

This Component contains the following parameters and controls:

- **Normal**: Adjusts the level of the Normal channel. The Bass and Treble controls have no effect in this channel.
- **Brilliant**: Adjusts the level of the Brilliant channel.
- **Bass**: Adjusts the low-frequency response of the Brilliant channel.
- **Treble**: Adjusts the high-frequency response of the Brilliant channel.
- **Speed**: Adjusts the rate of the tremolo effect.
- **Depth**: Adjusts the intensity of the tremolo effect. When turned fully to the left, the effect is switched off.
- **Tone Cut**: Applies a low-pass filter to the Normal channel. Turning the knob to the right reduces high-frequency content.

The Expert panel contains the following parameters and controls:

- **Power Supply**: Switches the frequency of the AC mains voltage powering the amplifier between 50 and 60 Hz. The AC mains voltage introduces ripple to the internal voltages of the amplifier at double this frequency, which imparts a subtle modulation to the sound.

- **Variac**: Emulates the effect of inserting a variable transformer in the AC mains power line, thus reducing the supply voltage for the famous "brown sound", or increasing it to make the sound stronger.
- **Sag**: Emulates the effect of sending excessive signal levels to the amplifier and briefly exceeding its power limit. Turning **Sag** to the right adds a compressed feel to the sound, similar to a a tube-based rectifier circuit. Turning **Sag** to the left reduces the amount of compression, similar to a diode-based rectifier circuit.
- **Response**: Adjusts the power storing capacity of the power supply capacitors. Turning **Response** to the left decelerates the response of the supply voltage to dynamic playing. Turning **Response** to the right accelerates the response of the supply voltage to dynamic playing for a looser feel.
- **Bias**: Adjusts the grid bias of the output tubes. This influences crossover distortion and determines the amplifier class rating of the circuit. Turning **Bias** to the right lets the circuit run hotter for a raw sound character.
- **GR5 Mode**: Activates backwards-compatibility of this Component with presets made in GUITAR RIG 5. Deactivating **GR5 Mode** allows for a deeper tremolo effect.

Bass Invader

Bass Invader models the sound of a versatile amplifier associated with the Rock and Indie sound of the late 1980s and 1990s. It includes extensive tone shaping controls that allow you to precisely tailor the sound. Even though its character can be described as clean and sweet for most its range, it also produces very interesting distortion sounds when the controls are cranked up.

This amp has been modeled using NI's newly developed ICM (Intelligent Circuit Modeling) technology that employs machine learning to reproduce the behavior of hardware devices from the ground up, giving a whole new level of depth and realism to amp emulations. Learn more about ICM in this blog article on our website: Behind the scenes of GUITAR RIG 6's Intelligent Circuit Modeling.

When required, you can reduce the CPU load by deactivating stereo processing for this Component. For more information, see Component Controls. You can also set an option in the Preferences to load Components using ICM in mono by default. For more information, see General.

This Component contains the following parameters and controls:

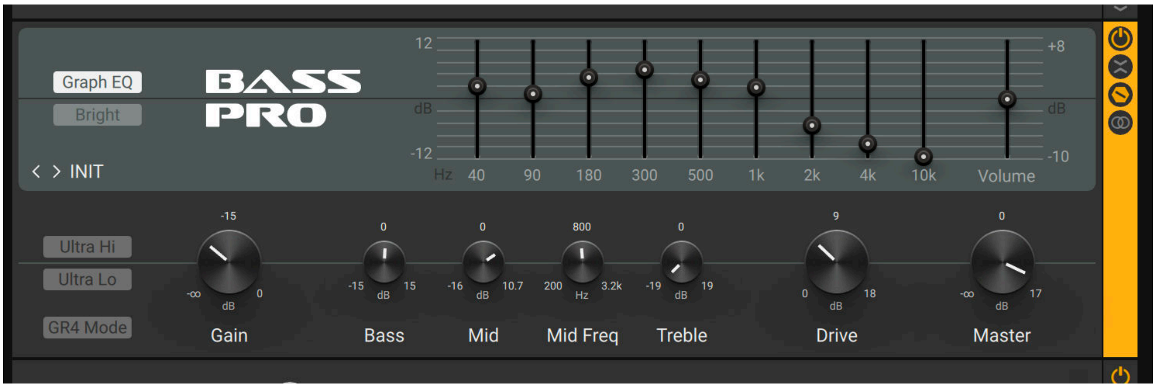
- **Lo Cut**: Cuts low-frequency content, removing rumble from the sound.
- **Mid Contour**: Cuts low mid-frequency content, softening the sound.

- **Hi Boost**: Boosts high-frequency content, adding edge and definition to the sound.
- **Volume**: Adjusts the input level, or gain, of the amplifier. Turning **Volume** to the right adds saturation and distortion to the signal.
- **Bass**: Adjusts the low-frequency response.
- **Low Mid**: Adjusts the low mid-frequency response.
- **Hi Mid**: Adjusts high mid-frequency response.
- **Treble**: Adjusts the high-frequency response.
- **Boost**: Adjusts the amount of extra gain added to the signal.
- **Master**: Adjusts the output level of the Component.

Bass Pro

Bass Pro models a gritty and growling amplifier sound that makes the bass stand out in the mix. It includes a graphic equalizer that allows you to precisely tailor the sound.

This Component contains the following parameters and controls:



- **Graph EQ**: Activates the graphic equalizer in the top panel of the amplifier.
- **Bright**: Boosts high-frequency content.
- **Graphic equalizer**: Boosts or cuts nine specific frequency bands: 40 Hz, 90 Hz, 180 Hz, 300 Hz, 500 Hz, 1 kHz, 2 kHz, 4 kHz, and 10 kHz. Moving a slider up boosts the respective frequency band by up to 12 dB. Moving a slider down cuts the respective frequency band by up to -12 dB. When a slider is centered, the respective frequency band remains unaffected.
- **Volume**: Adjusts the output level of the graphic equalizer. Moving the slider up increases the level by up to 8 dB. Moving the slider down decreases the level by up to -10 dB. You can use this control to compensate for strong boosts or cuts made with the graphic equalizer.
- **Ultra Hi**: Boosts high-frequency content in a wide frequency range. The effect is more pronounced than the one achieved using **Bright**.
- **Ultra Lo**: Boosts low-frequency content and cuts mid-frequency content.
- **Gain**: Adjusts the input level, or gain, of the amplifier. Turning **Gain** to the right adds saturation and distortion to the signal.
- **Bass**: Adjusts the low-frequency response.
- **Mid**: Adjusts mid-frequency content in a frequency band set using the **Mid Freq** control.
- **Mid Freq**: Sets the frequency band adjusted using the **Mid** control from 200 Hz to 3200 Hz.
- **Treble**: Adjusts the high-frequency response.
- **Drive**: Adjusts gain specifically for mid-frequency content, thereby changing the character of the sound.

- **Master**: Adjusts the output level of the Component.

The Expert panel contains the following parameters and controls:

- **GR4 Mode**: Activates backwards-compatibility of this Component with presets made in GUITAR RIG 4.

Chicago

Chicago models the sound of a vintage amp from the 1950s. Its dirty and fuzzy sound is reminiscent of early Rock and Roll guitars, but can also be used to add character to other sounds, for example drums.

This amp has been modeled using NI's newly developed ICM (Intelligent Circuit Modeling) technology that employs machine learning to reproduce the behavior of hardware devices from the ground up, giving a whole new level of depth and realism to amp emulations. Learn more about ICM in this blog article on our website: Behind the scenes of GUITAR RIG 6's Intelligent Circuit Modeling.

When required, you can reduce the CPU load by deactivating stereo processing for this Component. For more information, see Component Controls. You can also set an option in the Preferences to load Components using ICM in mono by default. For more information, see General.

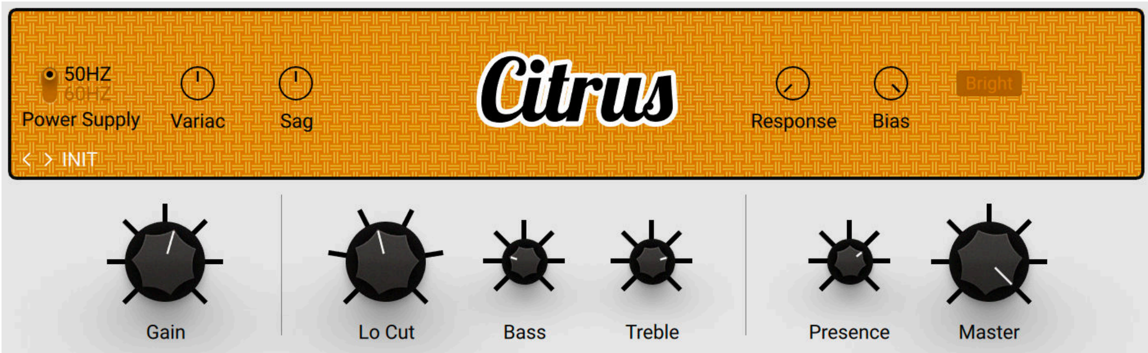
This Component contains the following parameters and controls:

- **Normal/Mod**: Switches between two basic modes of the amplifier. When **Normal** is selected, the original sound character of the amplifier is preserved. When **Mod** is selected, the sound character is tighter with a more controlled low-frequency response.
- **Bright**: Boosts high-frequency content.
- **Tone**: Adjusts the frequency response by changing the balance between low and high frequencies. Turning the control to the left emphasizes low frequencies. Turning the control to the right emphasizes high frequencies.
- **Volume**: Adjusts the gain of the amplifier.

Citrus

Citrus models the sound of a powerful British amp rated at 120W. Its tones range from clean to gritty distortion when **Gain** and **Master** are fully turned to the right.

This Component contains the following parameters and controls:



- **Gain**: Adjusts the input level, or gain, of the amplifier. Turning **Gain** to the right adds saturation and distortion to the signal.
- **Lo Cut**: Applies a low-cut filter. Turning the knob to the right reduces low-frequency content.
- **Bass**: Adjusts the low-frequency response.
- **Treble**: Adjusts the high-frequency response.
- **Presence**: Boosts upper mid-frequency content.
- **Master**: Adjusts the output level of the Component.

The Expert panel contains the following parameters and controls:

- **Power Supply**: Switches the frequency of the AC mains voltage powering the amplifier between 50 and 60 Hz. The AC mains voltage introduces ripple to the internal voltages of the amplifier at double this frequency, which imparts a subtle modulation to the sound.
- **Variac**: Emulates the effect of inserting a variable transformer in the AC mains power line, thus reducing the supply voltage for the famous "brown sound", or increasing it to make the sound stronger.
- **Sag**: Emulates the effect of sending excessive signal levels to the amplifier and briefly exceeding its power limit. Turning **Sag** to the right adds a compressed feel to the sound, similar to a a tube-based rectifier circuit. Turning **Sag** to the left reduces the amount of compression, similar to a diode-based rectifier circuit.
- **Response**: Adjusts the power storing capacity of the power supply capacitors. Turning **Response** to the left decelerates the response of the supply voltage to dynamic playing. Turning **Response** to the right accelerates the response of the supply voltage to dynamic playing for a looser feel.
- **Bias**: Adjusts the grid bias of the output tubes. This influences crossover distortion and determines the amplifier class rating of the circuit. Turning **Bias** to the right lets the circuit run hotter for a raw sound character.
- **Bright**: Extends the high-frequency response for a brighter sound.

Cool Plex

Cool Plex models a clean vintage sound with mild saturation, transparent but charismatic. Two different channels can be used simultaneously and blended using the **Volume I** and **Volume II** controls. The first channel has a brighter sound, while the second channel has a warmer sound.

This Component contains the following parameters and controls:

- **Volume I**: Adjusts the input level, or gain, for the bright channel.
- **Volume I**: Adjusts the input level, or gain, for the warm channel.
- **Bass**: Adjusts the low-frequency response.
- **Middle**: Adjusts the mid-frequency response.
- **Treble**: Adjusts the high-frequency response.
- **Presence**: Boosts upper mid-frequency content.

The Expert panel contains the following parameters and controls:

- **Power Supply**: Switches the frequency of the AC mains voltage powering the amplifier between 50 and 60 Hz. The AC mains voltage introduces ripple to the internal voltages of the amplifier at double this frequency, which imparts a subtle modulation to the sound.
- **Variac**: Emulates the effect of inserting a variable transformer in the AC mains power line, thus reducing the supply voltage for the famous "brown sound", or increasing it to make the sound stronger.
- **Sag**: Emulates the effect of sending excessive signal levels to the amplifier and briefly exceeding its power limit. Turning **Sag** to the right adds a compressed feel to the sound, similar to a a tube-based rectifier circuit. Turning **Sag** to the left reduces the amount of compression, similar to a diode-based rectifier circuit.
- **Response**: Adjusts the power storing capacity of the power supply capacitors. Turning **Response** to the left decelerates the response of the supply voltage to dynamic playing. Turning **Response** to the right accelerates the response of the supply voltage to dynamic playing for a looser feel.
- **Bias**: Adjusts the grid bias of the output tubes. This influences crossover distortion and determines the amplifier class rating of the circuit. Turning **Bias** to the right lets the circuit run hotter for a raw sound character.

Fire Breather

Fire Breather models the sound of a contemporary amplifier that builds on a British legacy. Its clear and detailed character with a tight low-frequency response can be used in a variety of styles, from Blues and classic Rock to Hard Rock and Heavy Metal.

This amp has been modeled using NI's newly developed ICM (Intelligent Circuit Modeling) technology that employs machine learning to reproduce the behavior of hardware devices from the ground up, giving a whole new level of depth and realism to amp emulations. Learn more about ICM in this blog article on our website: Behind the scenes of GUITAR RIG 6's Intelligent Circuit Modeling.

When required, you can reduce the CPU load by deactivating stereo processing for this Component. For more information, see Component Controls. You can also set an option in the Preferences to load Components using ICM in mono by default. For more information, see General.

This Component contains the following parameters and controls:

- **I/II/III**: Switches between three different configurations of the amplifier's internal gain structure.
- **Gain**: Adjusts the input level, or gain, of the amplifier. Turning **Gain** to the right adds saturation and distortion to the signal.
- **Bass**: Adjusts the low-frequency response.
- **Mid**: Adjusts the mid-frequency response.
- **Treble**: Adjusts the high-frequency response.
- **Master**: Adjusts the output level of the Component.

Gratifier

The Gratifier models the sound of a famous American multi-channel solo head. A selection of four different channels provides a tonal range from clean to excessive distortion.

This Component contains the following parameters and controls:

- **Modern/Vintage/Raw/Clean**: Switches between four different channels, each with its own distinct character.
- **Gain**: Adjusts the input level, or gain, of the amplifier. Turning **Gain** to the right adds saturation and distortion to the signal.
- **Bass**: Adjusts the low-frequency response.

- **Mid**: Adjusts the mid-frequency response.
- **Treble**: Adjusts the high-frequency response.
- **Presence**: Boosts upper mid-frequency content.
- **Master**: Adjusts the output level of the Component.

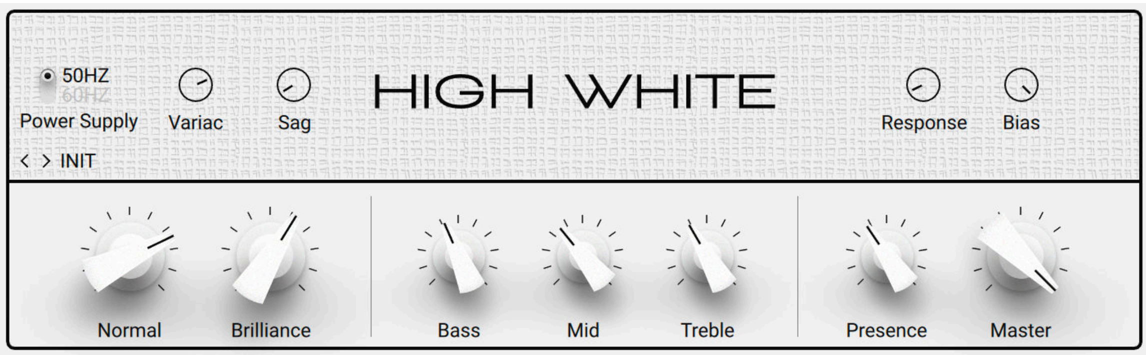
The Expert panel contains the following parameters and controls:

- **Power Supply**: Switches the frequency of the AC mains voltage powering the amplifier between 50 and 60 Hz. The AC mains voltage introduces ripple to the internal voltages of the amplifier at double this frequency, which imparts a subtle modulation to the sound.
- **Variac**: Emulates the effect of inserting a variable transformer in the AC mains power line, thus reducing the supply voltage for the famous "brown sound", or increasing it to make the sound stronger.
- **Sag**: Emulates the effect of sending excessive signal levels to the amplifier and briefly exceeding its power limit. Turning **Sag** to the right adds a compressed feel to the sound, similar to a a tube-based rectifier circuit. Turning **Sag** to the left reduces the amount of compression, similar to a diode-based rectifier circuit.
- **Response**: Adjusts the power storing capacity of the power supply capacitors. Turning **Response** to the left decelerates the response of the supply voltage to dynamic playing. Turning **Response** to the right accelerates the response of the supply voltage to dynamic playing for a looser feel.
- **Bias**: Adjusts the grid bias of the output tubes. This influences crossover distortion and determines the amplifier class rating of the circuit. Turning **Bias** to the right lets the circuit run hotter for a raw sound character.

High White

High White models the British sound made famous by artists like David Gilmour and Pete Townsend. This amplifier features a clean sounding **Normal** channel and a more aggressive sounding **Brilliance** channel. Both channels can be blended and combined for a great variety of sounds.

This Component contains the following parameters and controls:



- **Normal**: Adjusts the level of the Normal channel.
- **Brilliant**: Adjusts the level of the Brilliant channel.
- **Bass**: Adjusts the low-frequency response.
- **Mid**: Adjusts the mid-frequency response.
- **Treble**: Adjusts the high-frequency response.
- **Presence**: Boosts upper mid-frequency content.

- **Master**: Adjusts the output level of the Component.

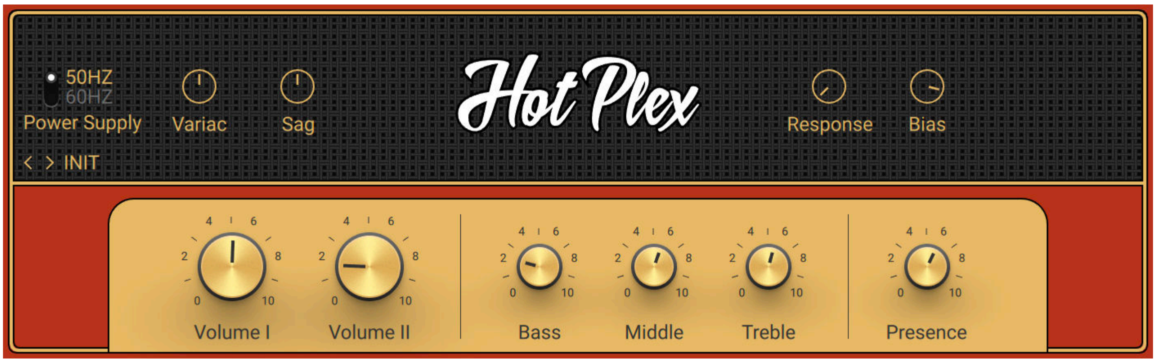
The Expert panel contains the following parameters and controls:

- **Power Supply**: Switches the frequency of the AC mains voltage powering the amplifier between 50 and 60 Hz. The AC mains voltage introduces ripple to the internal voltages of the amplifier at double this frequency, which imparts a subtle modulation to the sound.
- **Variac**: Emulates the effect of inserting a variable transformer in the AC mains power line, thus reducing the supply voltage for the famous "brown sound", or increasing it to make the sound stronger.
- **Sag**: Emulates the effect of sending excessive signal levels to the amplifier and briefly exceeding its power limit. Turning **Sag** to the right adds a compressed feel to the sound, similar to a a tube-based rectifier circuit. Turning **Sag** to the left reduces the amount of compression, similar to a diode-based rectifier circuit.
- **Response**: Adjusts the power storing capacity of the power supply capacitors. Turning **Response** to the left decelerates the response of the supply voltage to dynamic playing. Turning **Response** to the right accelerates the response of the supply voltage to dynamic playing for a looser feel.
- **Bias**: Adjusts the grid bias of the output tubes. This influences crossover distortion and determines the amplifier class rating of the circuit. Turning **Bias** to the right lets the circuit run hotter for a raw sound character.

Hot Plex

Hot Plex models a clean vintage sound, adding extra gain that allows for strong distortion. Two different channels can be used simultaneously and blended using the **Volume I** and **Volume II** controls. The first channel has a brighter sound, while the second channel has a warmer sound.

This Component contains the following parameters and controls:



- **Volume I**: Adjusts the input level, or gain, for the bright channel.
- **Volume I**: Adjusts the input level, or gain, for the warm channel.
- **Bass**: Adjusts the low-frequency response.
- **Middle**: Adjusts the mid-frequency response.
- **Treble**: Adjusts the high-frequency response.
- **Presence**: Boosts upper mid-frequency content.

The Expert panel contains the following parameters and controls:

- **Power Supply**: Switches the frequency of the AC mains voltage powering the amplifier between 50 and 60 Hz. The AC mains voltage introduces ripple to the internal voltages of the amplifier at double this frequency, which imparts a subtle modulation to the sound.

- **Variac**: Emulates the effect of inserting a variable transformer in the AC mains power line, thus reducing the supply voltage for the famous "brown sound", or increasing it to make the sound stronger.
- **Sag**: Emulates the effect of sending excessive signal levels to the amplifier and briefly exceeding its power limit. Turning **Sag** to the right adds a compressed feel to the sound, similar to a a tube-based rectifier circuit. Turning **Sag** to the left reduces the amount of compression, similar to a diode-based rectifier circuit.
- **Response**: Adjusts the power storing capacity of the power supply capacitors. Turning **Response** to the left decelerates the response of the supply voltage to dynamic playing. Turning **Response** to the right accelerates the response of the supply voltage to dynamic playing for a looser feel.
- **Bias**: Adjusts the grid bias of the output tubes. This influences crossover distortion and determines the amplifier class rating of the circuit. Turning **Bias** to the right lets the circuit run hotter for a raw sound character.

Hot Solo+

Hot Solo+ models the sound of a true modern classic. It features two separate channels, the low-gain **Normal** channel and the high-gain **Overdrive** channel, which can be used for a distinctly contemporary rock sound.

This Component contains the following parameters and controls:

- **Overdrive/Normal**: Switches between the **Normal** and the **Overdrive** channel.
- **Overdrive**: Adjusts the input level, or gain, of the Overdrive channel.
- **Normal**: Adjusts the input level, or gain, of the Normal channel.
- **Bass**: Adjusts the low-frequency response.
- **Mid**: Adjusts the mid-frequency response.
- **Treble**: Adjusts the high-frequency response.
- **Presence**: Boosts upper mid-frequency content.
- **Depth**: Boosts low-frequency content.
- **Master**: Adjusts the output level of the Component.

The Expert panel contains the following parameters and controls:

- **Power Supply**: Switches the frequency of the AC mains voltage powering the amplifier between 50 and 60 Hz. The AC mains voltage introduces ripple to the internal voltages of the amplifier at double this frequency, which imparts a subtle modulation to the sound.
- **Variac**: Emulates the effect of inserting a variable transformer in the AC mains power line, thus reducing the supply voltage for the famous "brown sound", or increasing it to make the sound stronger.

- **Sag**: Emulates the effect of sending excessive signal levels to the amplifier and briefly exceeding its power limit. Turning **Sag** to the right adds a compressed feel to the sound, similar to a a tube-based rectifier circuit. Turning **Sag** to the left reduces the amount of compression, similar to a diode-based rectifier circuit.
- **Response**: Adjusts the power storing capacity of the power supply capacitors. Turning **Response** to the left decelerates the response of the supply voltage to dynamic playing. Turning **Response** to the right accelerates the response of the supply voltage to dynamic playing for a looser feel.
- **Bias**: Adjusts the grid bias of the output tubes. This influences crossover distortion and determines the amplifier class rating of the circuit. Turning **Bias** to the right lets the circuit run hotter for a raw sound character.

Jazz Amp

The Jazz Amp models the sound of an amp produced by a renowned synthesizer manufacturer. It produces a warm, clean tone. Its internal vibrato or chorus effect strongly contributes to the amplifier's character.

> If you want to use true stereo vibrato or chorus instead, you can turn off the Jazz Amp's internal effects and insert the **Ensemble** Component from the **Modulation** category in your Rack.

This Component contains the following parameters and controls:

- **Brilliance**: Boosts high-frequency content.
- **Bass**: Adjusts the low-frequency response.
- **Mid**: Adjusts the mid-frequency response.
- **Treble**: Adjusts the high-frequency response.
- **Vibrato/Off/Chorus**: Activates either the **Vibrato** or **Chorus** effect. When set to **Off**, both effects are deactivated.
- **Rate**: Adjusts the rate of the vibrato and chorus effect.
- **Depth**: Adjusts the intensity of the vibrato and chorus effect.
- **Volume**: Adjusts the output level of the Component.

Jump

Jump models the sound of a clean British amplifier that was introduced at the beginning of the 1980s and became a staple of hard rock and heavy metal bands in that era. It is a variant of the Lead 800 amplifier with slightly less gain for a smoother sound.

This Component contains the following parameters and controls:

- **High/Low**: Switches between two modes for the amplifier's gain stage. In **Low** mode, the amplifier's gain is moderate, resulting in a smoother sound. In **High** mode, the gain is significantly increased for a more overdriven sound.
- **Volume**: Adjusts the input level, or gain, of the amplifier. Turning **Volume** to the right adds saturation and distortion to the signal.
- **Bass**: Adjusts the low-frequency response.
- **Mid**: Adjusts the mid-frequency response.
- **Treble**: Adjusts the high-frequency response.
- **Presence**: Boosts upper mid-frequency content.
- **Master Volume**: Adjusts the output level of the Component.

The Expert panel contains the following parameters and controls:

- **Power Supply**: Switches the frequency of the AC mains voltage powering the amplifier between 50 and 60 Hz. The AC mains voltage introduces ripple to the internal voltages of the amplifier at double this frequency, which imparts a subtle modulation to the sound.
- **Variac**: Emulates the effect of inserting a variable transformer in the AC mains power line, thus reducing the supply voltage for the famous "brown sound", or increasing it to make the sound stronger.
- **Sag**: Emulates the effect of sending excessive signal levels to the amplifier and briefly exceeding its power limit. Turning **Sag** to the right adds a compressed feel to the sound, similar to a a tube-based rectifier circuit. Turning **Sag** to the left reduces the amount of compression, similar to a diode-based rectifier circuit.
- **Response**: Adjusts the power storing capacity of the power supply capacitors. Turning **Response** to the left decelerates the response of the supply voltage to dynamic playing. Turning **Response** to the right accelerates the response of the supply voltage to dynamic playing for a looser feel.
- **Bias**: Adjusts the grid bias of the output tubes. This influences crossover distortion and determines the amplifier class rating of the circuit. Turning **Bias** to the right lets the circuit run hotter for a raw sound character.

Lead 800

Lead 800 models the sound of a vintage British amplifier. Other than Plex, which is suitable for both rhythm and lead guitar tones, Lead 800's distinct character leans itself to bright and edgy lead guitars.

This Component contains the following parameters and controls:

- **Boost**: Increases the gain for a more overdriven sound.
- **Pre Amp**: Adjusts the input level, or gain, of the amplifier. Turning **Pre Amp** to the right adds saturation and distortion to the signal.
- **Bass**: Adjusts the low-frequency response.
- **Mid**: Adjusts the mid-frequency response.
- **Treble**: Adjusts the high-frequency response.
- **Presence**: Boosts upper mid-frequency content.
- **Master Volume**: Adjusts the output level of the Component.

The Expert panel contains the following parameters and controls:

- **Power Supply**: Switches the frequency of the AC mains voltage powering the amplifier between 50 and 60 Hz. The AC mains voltage introduces ripple to the internal voltages of the amplifier at double this frequency, which imparts a subtle modulation to the sound.
- **Variac**: Emulates the effect of inserting a variable transformer in the AC mains power line, thus reducing the supply voltage for the famous "brown sound", or increasing it to make the sound stronger.
- **Sag**: Emulates the effect of sending excessive signal levels to the amplifier and briefly exceeding its power limit. Turning **Sag** to the right adds a compressed feel to the sound, similar to a a tube-based rectifier circuit. Turning **Sag** to the left reduces the amount of compression, similar to a diode-based rectifier circuit.
- **Response**: Adjusts the power storing capacity of the power supply capacitors. Turning **Response** to the left decelerates the response of the supply voltage to dynamic playing. Turning **Response** to the right accelerates the response of the supply voltage to dynamic playing for a looser feel.
- **Bias**: Adjusts the grid bias of the output tubes. This influences crossover distortion and determines the amplifier class rating of the circuit. Turning **Bias** to the right lets the circuit run hotter for a raw sound character.

Plex

Plex models a vintage British amplifier with a clean sound. Two different channels can be used simultaneously and blended using the **Volume I** and **Volume II** controls. The first channel has a brighter sound, while the second channel has a warmer sound.

This Component contains the following parameters and controls:

- **Volume I**: Adjusts the input level, or gain, for the bright channel.
- **Volume I**: Adjusts the input level, or gain, for the warm channel.
- **Bass**: Adjusts the low-frequency response.
- **Middle**: Adjusts the mid-frequency response.
- **Treble**: Adjusts the high-frequency response.
- **Presence**: Boosts upper mid-frequency content.

The Expert panel contains the following parameters and controls:

- **Power Supply**: Switches the frequency of the AC mains voltage powering the amplifier between 50 and 60 Hz. The AC mains voltage introduces ripple to the internal voltages of the amplifier at double this frequency, which imparts a subtle modulation to the sound.
- **Variac**: Emulates the effect of inserting a variable transformer in the AC mains power line, thus reducing the supply voltage for the famous "brown sound", or increasing it to make the sound stronger.
- **Sag**: Emulates the effect of sending excessive signal levels to the amplifier and briefly exceeding its power limit. Turning **Sag** to the right adds a compressed feel to the sound, similar to a a tube-based rectifier circuit. Turning **Sag** to the left reduces the amount of compression, similar to a diode-based rectifier circuit.
- **Response**: Adjusts the power storing capacity of the power supply capacitors. Turning **Response** to the left decelerates the response of the supply voltage to dynamic playing. Turning **Response** to the right accelerates the response of the supply voltage to dynamic playing for a looser feel.
- **Bias**: Adjusts the grid bias of the output tubes. This influences crossover distortion and determines the amplifier class rating of the circuit. Turning **Bias** to the right lets the circuit run hotter for a raw sound character.

RAMMFIRE

RAMMFIRE models the sound of a famous multi-channel solo head with a tube power amp, famously used by Rammstein guitarist Richard Z. Kruspe. This emulation is based on his personal rig, giving you the perfect rhythm amp sound that Kruspe has used on every Rammstein album. A fourth channel has been added to further extend the available range of sounds from clean to strong distortion.

This Component contains the following parameters and controls:

- **RZK/Modern/Vintage/Clean**: Switches between four different channels, each with its own distinct character.
- **Gain**: Adjusts the input level, or gain, of the amplifier. Turning **Gain** to the right adds saturation and distortion to the signal.
- **Bass**: Adjusts the low-frequency response.
- **Mid**: Adjusts the mid-frequency response.
- **Treble**: Adjusts the high-frequency response.
- **Presence**: Boosts upper mid-frequency content.
- **Master**: Adjusts the output level of the Component.

The Expert panel contains the following parameters and controls:

- **Power Supply**: Switches the frequency of the AC mains voltage powering the amplifier between 50 and 60 Hz. The AC mains voltage introduces ripple to the internal voltages of the amplifier at double this frequency, which imparts a subtle modulation to the sound.
- **Variac**: Emulates the effect of inserting a variable transformer in the AC mains power line, thus reducing the supply voltage for the famous "brown sound", or increasing it to make the sound stronger.
- **Sag**: Emulates the effect of sending excessive signal levels to the amplifier and briefly exceeding its power limit. Turning **Sag** to the right adds a compressed feel to the sound, similar to a a tube-based rectifier circuit. Turning **Sag** to the left reduces the amount of compression, similar to a diode-based rectifier circuit.
- **Response**: Adjusts the power storing capacity of the power supply capacitors. Turning **Response** to the left decelerates the response of the supply voltage to dynamic playing. Turning **Response** to the right accelerates the response of the supply voltage to dynamic playing for a looser feel.
- **Bias**: Adjusts the grid bias of the output tubes. This influences crossover distortion and determines the amplifier class rating of the circuit. Turning **Bias** to the right lets the circuit run hotter for a raw sound character.

Twang Reverb

The Twang Reverb models the rich tube sound of a classic amp from the 60s. It is ideal for crunchy rhythm guitar and tones that are clean yet full of character.

> To overdrive this amp, it is recommended to first increase the signal level, for example by using the **Volume** or **Gain Booster** Component.

This Component contains the following parameters and controls:

- **Bright**: Boosts high-frequency content.
- **Bass**: Adjusts the low-frequency response.
- **Middle**: Adjusts the mid-frequency response.
- **Treble**: Adjusts the high-frequency response.
- **Reverb on/off**: Switches the reverb effect on or off.
- **Reverb**: Adjust the intensity of the spring spring reverb effect.
- **Vibrato on/off**: Switches the tremolo effect (here called vibrato) on or off.
- **Speed**: Adjusts the rate of the tremolo effect.
- **Vibrato**: Adjusts the intensity of the tremolo effect.
- **Volume**: Adjusts the output level of the Component.

The Expert panel contains the following parameters and controls:

- **Power Supply**: Switches the frequency of the AC mains voltage powering the amplifier between 50 and 60 Hz. The AC mains voltage introduces ripple to the internal voltages of the amplifier at double this frequency, which imparts a subtle modulation to the sound.
- **Variac**: Emulates the effect of inserting a variable transformer in the AC mains power line, thus reducing the supply voltage for the famous "brown sound", or increasing it to make the sound stronger.
- **Sag**: Emulates the effect of sending excessive signal levels to the amplifier and briefly exceeding its power limit. Turning **Sag** to the right adds a compressed feel to the sound, similar to a a tube-based rectifier circuit. Turning **Sag** to the left reduces the amount of compression, similar to a diode-based rectifier circuit.
- **Response**: Adjusts the power storing capacity of the power supply capacitors. Turning **Response** to the left decelerates the response of the supply voltage to dynamic playing. Turning **Response** to the right accelerates the response of the supply voltage to dynamic playing for a looser feel.
- **Bias**: Adjusts the grid bias of the output tubes. This influences crossover distortion and determines the amplifier class rating of the circuit. Turning **Bias** to the right lets the circuit run hotter for a raw sound character.
- **Reverb Time**: Adjusts the length of the reverb effect, or reverb time. Turning the control to the right changes the reverb time from short to long.
- **Reverb Size**: Adjusts the swell and reflection pattern of the reverb effect, giving the impression of differently sized spaces. Turning the control to the right changes the size from small to large.

Tweed Delight

Tweed Delight models the sound of a legendary American amplifier. With only three controls, its sound ranges from thick and clean to roaring blues tones. Tweed Delight features two channels that can be mixed for stronger overdrive and edgy, distorted sounds when the **Bright** and **Normal** controls are turned to the right.

This Component contains the following parameters and controls:

- **Bright**: Adjusts the gain of the Bright channel.
- **Normal**: Adjusts the gain of the Normal channel.
- **Tone**: Adjusts the high-frequency response. Turned fully to the left, high-frequency content is attenuated for a warm sound. Turning **Tone** to the right introduces more high-frequency content for a more defined sound.

The Expert panel contains the following parameters and controls:

- **Power Supply**: Switches the frequency of the AC mains voltage powering the amplifier between 50 and 60 Hz. The AC mains voltage introduces ripple to the internal voltages of the amplifier at double this frequency, which imparts a subtle modulation to the sound.
- **Variac**: Emulates the effect of inserting a variable transformer in the AC mains power line, thus reducing the supply voltage for the famous "brown sound", or increasing it to make the sound stronger.
- **Sag**: Emulates the effect of sending excessive signal levels to the amplifier and briefly exceeding its power limit. Turning **Sag** to the right adds a compressed feel to the sound, similar to a a tube-based rectifier circuit. Turning **Sag** to the left reduces the amount of compression, similar to a diode-based rectifier circuit.
- **Response**: Adjusts the power storing capacity of the power supply capacitors. Turning **Response** to the left decelerates the response of the supply voltage to dynamic playing. Turning **Response** to the right accelerates the response of the supply voltage to dynamic playing for a looser feel.
- **Bias**: Adjusts the grid bias of the output tubes. This influences crossover distortion and determines the amplifier class rating of the circuit. Turning **Bias** to the right lets the circuit run hotter for a raw sound character.

Tweedman

The Tweedman models the sound of an amplifier that was originally designed for bass in the 50s, but is also renowned for its guitar tones. Compared to the Bass Amp Pro, the Tweedman has a more raw and vintage sound. It features two channels that can be mixed for a great variety of sounds.

This Component contains the following parameters and controls:

- **Bright**: Adjusts the level of the Bright channel.
- **Normal**: Adjusts the level of the Normal channel.
- **Bass**: Adjusts the low-frequency response.
- **Middle**: Adjusts the mid-frequency response.
- **Treble**: Adjusts the high-frequency response.
- **Presence**: Boosts upper mid-frequency content.

The Expert panel contains the following parameters and controls:

- **Power Supply**: Switches the frequency of the AC mains voltage powering the amplifier between 50 and 60 Hz. The AC mains voltage introduces ripple to the internal voltages of the amplifier at double this frequency, which imparts a subtle modulation to the sound.
- **Variac**: Emulates the effect of inserting a variable transformer in the AC mains power line, thus reducing the supply voltage for the famous "brown sound", or increasing it to make the sound stronger.
- **Sag**: Emulates the effect of sending excessive signal levels to the amplifier and briefly exceeding its power limit. Turning **Sag** to the right adds a compressed feel to the sound, similar to a a tube-based rectifier circuit. Turning **Sag** to the left reduces the amount of compression, similar to a diode-based rectifier circuit.
- **Response**: Adjusts the power storing capacity of the power supply capacitors. Turning **Response** to the left decelerates the response of the supply voltage to dynamic playing. Turning **Response** to the right accelerates the response of the supply voltage to dynamic playing for a looser feel.
- **Bias**: Adjusts the grid bias of the output tubes. This influences crossover distortion and determines the amplifier class rating of the circuit. Turning **Bias** to the right lets the circuit run hotter for a raw sound character.
- **GR4 Mode**: Activates backwards-compatibility of this Component with presets made in GUITAR RIG 4.

Ultrasonic

Ultrasonic models a contemporary, high-gain sound. This boutique amplifier covers a huge range of modern tones.

This Component contains the following parameters and controls:

- **Overdrive/Clean**: Switches between two modes for the amplifier's gain stage. In **Clean** mode, the amplifier's gain is moderate, resulting in a smoother sound. In **Overdrive** mode, the gain is significantly increased for a more overdriven sound.
- **Gain**: Adjusts the input level, or gain, of the amplifier. Turning **Gain** to the right adds saturation and distortion to the signal.
- **Bass**: Adjusts the low-frequency response.
- **Middle**: Adjusts the mid-frequency response.
- **Treble**: Adjusts the high-frequency response.
- **Presence**: Boosts upper mid-frequency content.
- **Master**: Adjusts the output level of the Component.
- **Volume**: Adjusts the level of the selected channel (**Overdrive** or **Clean**).

The Expert panel contains the following parameters and controls:

- **Power Supply**: Switches the frequency of the AC mains voltage powering the amplifier between 50 and 60 Hz. The AC mains voltage introduces ripple to the internal voltages of the amplifier at double this frequency, which imparts a subtle modulation to the sound.
- **Variac**: Emulates the effect of inserting a variable transformer in the AC mains power line, thus reducing the supply voltage for the famous "brown sound", or increasing it to make the sound stronger.
- **Sag**: Emulates the effect of sending excessive signal levels to the amplifier and briefly exceeding its power limit. Turning **Sag** to the right adds a compressed feel to the sound, similar to a a tube-based rectifier circuit. Turning **Sag** to the left reduces the amount of compression, similar to a diode-based rectifier circuit.
- **Response**: Adjusts the power storing capacity of the power supply capacitors. Turning **Response** to the left decelerates the response of the supply voltage to dynamic playing. Turning **Response** to the right accelerates the response of the supply voltage to dynamic playing for a looser feel.
- **Bias**: Adjusts the grid bias of the output tubes. This influences crossover distortion and determines the amplifier class rating of the circuit. Turning **Bias** to the right lets the circuit run hotter for a raw sound character.

Van 51

Van 51 models the sound of a benchmark high-gain amplifier. It provides raw and edgy guitar tones that are suitable for a wide range of heavy guitar music.

This Component contains the following parameters and controls:

- **Hi Gain**: Increases the gain for a more overdriven sound.
- **Lead/Rhythm**: Switches between the **Lead** and the **Rhythm** channel. The **Rhythm** channel has a clean sound, with the option to add a high amount of distortion using **Crunch** mode. The **Lead** channel offers even more gain.
- **Bright**: Boosts high-frequency content in the Rhythm channel.
- **Crunch**: Adds a high amount of distortion in the Rhythm channel.
- **Rhythm**: Adjusts the input level, or gain, of the Rhythm channel.
- **Lead**: Adjusts the input level, or gain, of the Lead channel.
- **Bass**: Adjusts the low-frequency response.
- **Mid**: Adjusts the mid-frequency response.
- **Treble**: Adjusts the high-frequency response.
- **Resonance**: Adjusts the character of the amplifier's low-frequency response.
- **Presence**: Boosts upper mid-frequency content.
- **Post Gain**: Adjusts the output level of the Component.

The Expert panel contains the following parameters and controls:

- **Power Supply**: Switches the frequency of the AC mains voltage powering the amplifier between 50 and 60 Hz. The AC mains voltage introduces ripple to the internal voltages of the amplifier at double this frequency, which imparts a subtle modulation to the sound.
- **Variac**: Emulates the effect of inserting a variable transformer in the AC mains power line, thus reducing the supply voltage for the famous "brown sound", or increasing it to make the sound stronger.
- **Sag**: Emulates the effect of sending excessive signal levels to the amplifier and briefly exceeding its power limit. Turning **Sag** to the right adds a compressed feel to the sound, similar to a a tube-based rectifier circuit. Turning **Sag** to the left reduces the amount of compression, similar to a diode-based rectifier circuit.
- **Response**: Adjusts the power storing capacity of the power supply capacitors. Turning **Response** to the left decelerates the response of the supply voltage to dynamic playing. Turning **Response** to the right accelerates the response of the supply voltage to dynamic playing for a looser feel.
- **Bias**: Adjusts the grid bias of the output tubes. This influences crossover distortion and determines the amplifier class rating of the circuit. Turning **Bias** to the right lets the circuit run hotter for a raw sound character.

20.2. Cabinets

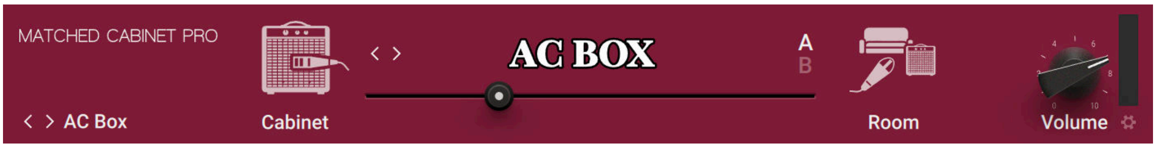
Cabinets play an essential role in shaping the final tone of the amplifier. With the Matched Cabinet, Control Room, and their Pro versions, GUITAR RIG provides both straight-forward and intricate options for finding your perfect sound.

When adding an Amplifier Component to the Rack, the corresponding Matched Cabinet is automatically loaded. It can replaced with any of the other Cabinet Components to change the sound. For more information, see Replacing Components in the Rack.

Matched Cabinet Pro

When adding any amplifier to the Rack, Matched Cabinet Pro is automatically included below it. It provides a cabinet matching the chosen amp as well as a virtual recording room. Matched Cabinet Pro employs convolution processing based on impulse responses that represent real cabinets recorded in real spaces. By blending between **Cabinet** and **Room**, you can adjust the amount of room sound added to the direct signal from the cabinet.

This Component contains the following parameters and controls:



- **Cabinet selector**: Switches between the different models of the matched cabinets, allowing them to be combined with any of the amps.
- **Cabinet/Room slider**: Blends between the direct sound of the cabinet and the virtual recording room, effectively changing the distance of the microphone from the cabinet.
- **A/B**: Switches between two different impulse responses for the virtual recording room.
- **Volume**: Adjusts the output level of the Component.
- **Learn**: Adjusts **Volume** automatically by analyzing the input signal and determining the optimal output level. For best results, play loudly while the analyzation is in progress.

Matched Cabinet

Matched Cabinet serves as an alternative to Matched Cabinet Pro. It provides a cabinet matching any of the available amplifiers. You can refine the sound by blending between two different microphone positions and adjusting the amount of room sound.

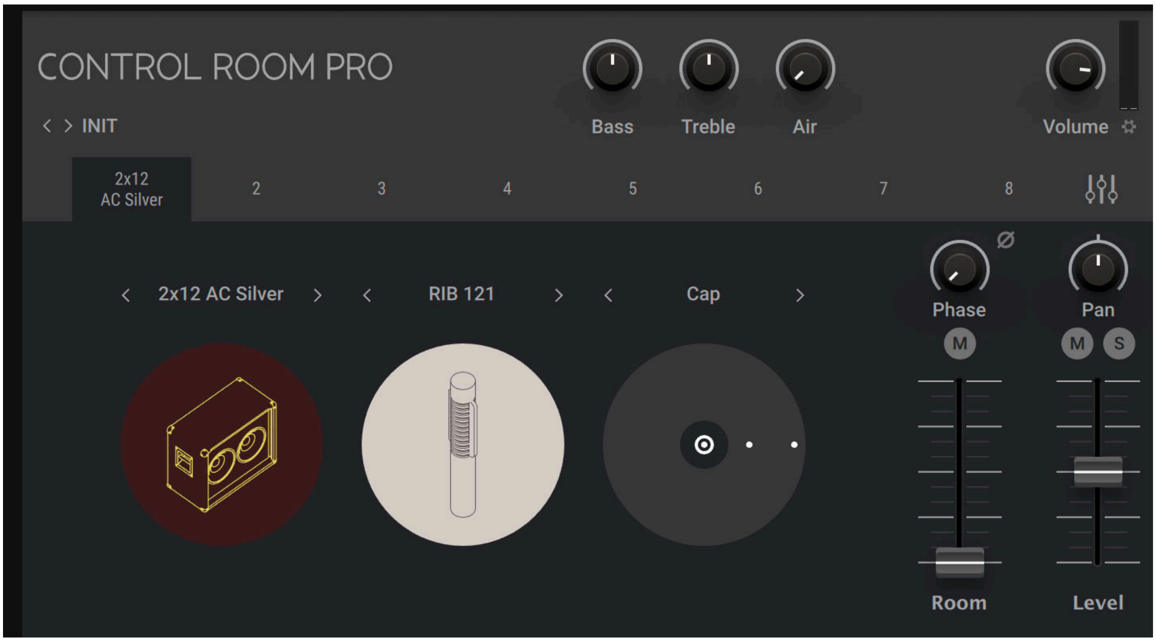
This Component contains the following parameters and controls:

- **Cabinet selector**: Switches between the different models of the matched cabinets, allowing them to be combined with any of the amps.
- **A/B slider**: Blends between two different microphone positions. The general characteristics of both are opposed, giving an edgy and a mellow option.

- **Air**: Adjusts the amount of early reflections picked up by the microphone, simulating the response of the room.
- **Volume**: Adjusts the output level of the Component.
- **Learn**: Adjusts **Volume** automatically by analyzing the input signal and determining the optimal output level. For best results, play loudly while the analyzation is in progress.

Control Room Pro

Control Room Pro enables you to create your own custom recording setup for up to eight different cabinets at the same time. Each channel in Control Room Pro combines a cabinet with a microphone placed at a specific position. By choosing different cabinets, microphones, and microphone positions on each of the channels, you can explore a huge range of different sounds. Further adding to the vast possibilities, the sound of the virtual recording room can be blended in to different degrees for each channel.



- **Bass**: Adjusts the low-frequency response.
- **Treble**: Adjusts the high-frequency response.
- **Air**: Adjusts the amount of early reflections picked up by the microphone, simulating the response of the room.
- **Volume**: Adjusts the output level of the Component.
- **Learn**: Adjusts **Volume** automatically by analyzing the input signal and determining the optimal output level. For best results, play loudly while the analyzation is in progress.

The Channel view in the Expert panel contains the following parameters and controls:

- **Label**: Indicates the type of cabinet used for the respective channel. Clicking on the label opens the Channel view. Clicking on the mixer icon opens the Mixer view.
- **Cabinet selector**: Selects between the available cabinets.
- **Microphone selector**: Selects between the available microphones used to record the cabinet.
- **Microphone Position selector**: Selects between the available microphone positions in front of the cabinet's speaker.

- **Phase**: Adjusts the phase delay relative to the other channels, enabling you to align signals of differently placed microphones. Contrary to a regular delay, the phase delay maintains the phase relationships across the frequency spectrum.
- **Invert**: Inverts the polarity, or phase, of the signal on this channel.
- **M (mute) Room**: Mutes the sound of the virtual recording room on this channel.
- **Room**: Blends between the direct sound of the cabinet and the virtual recording room on this channel, effectively changing the distance of the microphone from the cabinet.
- **Pan**: Adjusts the position of this channel in the stereo image.
- **M (mute)**: Mutes this channel, removing it from the output.
- **S (solo)**: Solos this channel, removing all other channels from the output.
- **Level**: Adjusts the level of this channel.

The Mixer view in the Expert panel contains the following parameters and controls

- **Label**: Indicates the type of cabinet used for the respective channel. Clicking on the label opens the Channel view. Clicking on the mixer icon opens the Mixer view.
- **Pan**: Adjusts the position of the respective channel in the stereo image.
- **M (mute)**: Mutes the respective channel, removing it from the output.
- **S (solo)**: Solos the respective channel, removing all other channels from the output.
- **Level**: Adjusts the level of the respective channel.

Control Room

Expanding on the possibilities of the Matched Cabinet and Matched Cabinet Pro, Control Room enables you to fully adjust a sophisticated recording setup for the chosen cabinet. Eight classic microphones at different positions can be freely mixed and combined. All microphone channels are phase-aligned to ensure a perfect signal at any given combination.

- **Bass**: Adjusts the low-frequency response.
- **Treble**: Adjusts the high-frequency response.
- **Air**: Adjusts the amount of early reflections picked up by the microphone, simulating the response of the room.
- **Volume**: Adjusts the output level of the Component.
- **Learn**: Adjusts **Volume** automatically by analyzing the input signal and determining the optimal output level. For best results, play loudly while the analyzation is in progress.

The Expert panel contains the following parameters and controls:

- **Cabinet selector**: Selects between the available cabinets.
- **Label**: Indicates the type of microphone used for the respective channel.
- **Pan**: Adjusts the position of the respective channel in the stereo image.
- **M (mute)**: Mutes the respective channel, removing it from the output.
- **S (solo)**: Solos the respective channel, removing all other channels from the output.
- **Level**: Adjusts the level of the respective channel.

20.3. Delay / Echo

Delays and Echo are time-based effects that repeat the input signal, recreating natural acoustic reflections or otherworldly types of echoes. If you need complex tempo-synced delays, you'll find them here in the Delay and Echo category!

Delay Man

The Delay Man is an extremely warm and flexible delay unit with built-in chorus and vibrato. It faithfully recreates the sound of esteemed classic hardware modules that are a standard in any advanced effects rig.

Controls

- The **INPUT** knob sets the amplification of the incoming signal. Set this so that the loudest peaks do not engage the overload LED next to it if you want to avoid overdrive.
- **DRY/WET** sets the amount of the signal being fed into the delay section, controlling the intensity of the effect.
- **TIME** sets the amount of delay time, i.e. the time interval between the straight sound and the appearance of the echo. When the component is synchronized (see below), the scale turns from milliseconds into note values.
- **FEEDBACK** determines how much of the output feeds back into the input. Minimum feedback gives a single echo; increasing this parameter produces repeating echoes. Turning the knob fully clockwise will overload the delay and produce a distorted, oscillating sound.
- The **CHOR/VIB** switch sets the modulation section of this component to chorus or vibrato mode.
- **DEPTH** controls the intensity of the modulation section.
- **TAP** allows tapping in the delay time. When you click on the Tap button repeatedly, the average time between the clicks is measured to derive the tempo. When in Sync mode, the tempo will be quantized to the nearest setting.
- **MUTE** only allows the dry input signal through. Nonetheless, delay trails will continue even after the button is pressed. If the **DRY/WET** control is turned up fully and the Mute is activated, you will hear no more sound, because no signal is being allowed into the dry section.

Expert Controls

- **SYNC DELAY** synchronizes the Time control to the metronome or host, depending on the Sync setting.
- **SYNC MOD** synchronizes the modulation speed to the metronome or host, depending on the Sync setting.
- **CHORUS RATE** sets the frequency of modulation of the chorus module. When modulation is synchronized, the scale turns from milliseconds into note values.
- **VIBRATO RATE** sets the frequency of modulation of the vibrato module. When modulation is synchronized, the scale turns from milliseconds into note values.
- **ACCELERATION** controls how fast the delay algorithm adapts to changes of the Time setting.
- **BASS** controls a filter affecting the bass frequencies. Turning it up will boost bass, turning it down will attenuate it.
- **TREBLE** controls a filter affecting the treble frequencies. Turning it up will boost treble, turning it down will attenuate it.

Psychedelay

This stereo delay creates sounds that range from standard echo/ambient sounds, to reversed effects that recall the “backwards tape” sounds of the 1960s.

Controls

- **DRY/WET** sets the amount of the signal being fed into the delay section, controlling the intensity of the effect.
- **TIME** sets the amount of delay time, i.e. the time interval between the straight sound and the appearance of the echo. When the component is synchronized (see below), the scale turns from milliseconds into note values.

- **REVERSE** plays back subsequent echoes in reverse.
- **DETUNE** detunes echoes up to ±50 cents. Combining this with feedback causes successive echoes to have ever-increasing amounts of detuning.
- **FEEDBACK** determines how much of the output feeds back into the input. Minimum feedback gives a single echo; increasing this parameter produces repeating echoes.
- **TAP** allows tapping in the delay time. When you click on the Tap button repeatedly, the average time between the clicks is measured to derive the tempo. When synchronized, the tempo will be quantized to the nearest setting.
- **MUTE** only allows the dry input signal through. Nonetheless, delay trails will continue even after the button is pressed. If the **DRY/WET** control is turned up fully and the Mute is activated, you will hear no more sound, because no signal is being allowed into the dry section.

Expert Controls

- **PITCH** adds a more extreme amount of detuning by transposing the echo in semitones, from –12 to +12. It interacts with the feedback control in the same way as Detune as each successive echo will be transposed an upward or downward.
- **STEREO TIME** controls the time shift between the stereo channels for stereo echo effects. When turned down, the delay time for both channels is equally set by the Time control. Turning it up places echoes in the stereo field by bringing forward the delay for one of the channel: A setting of 0.50 means that the extra echoes will happen at half the time of the main delay setting.
- **REVERSE** causes these additional delays to play back in reverse relative to the setting of the Main Reverse Button. If the latter is turned on, this button will reverse the second delay again, restoring the original signal.
- **DETUNE** allows the main Detune parameter to affect the added stereo echoes as well.
- **CROSS** creates feedback paths that cross between the two channels- right feeds back into the left channel, and left feeds back into the right channel. This creates a more complex, polyrhythmic type of echo effect.
- **TEMPO SYNC** synchronizes the time controls of this module to the metronome or host, depending on the Sync setting.

Quad Delay

The Delay module takes the input signal and plays it back through four delayed stages distributed to the stereo channels, allowing for impressive modulation possibilities. The output can be fed back to the input, producing a series of echoes,

Controls

- **DRY/WET** sets the amount of the signal being fed into the delay section, controlling the intensity of the effect.
- **TIME** sets the amount of delay time, i.e. the time interval between the straight sound and the appearance of the echo. When the component is synchronized (see below), the scale turns from milliseconds into note values.
- **FEEDBACK** determines how much of the output feeds back into the input. Minimum feedback gives a single echo; increasing this parameter produces repeating echoes.

- **RATE** sets the frequency of the four LFOs with which the delay times are modulated. The modulation works much like a chorus or flanger effect: A slower rate produces a slow, gradual detuning while faster rates produce a pulsating effect. When the component is synchronized (see below), the scale turns from milliseconds into note values.
- **DEPTH** determines how much the modulation section varies the delay time. Turning this up will increase the detuning effect of the modulation.
- **TAP** allows tapping in the delay time. When you click on the Tap button repeatedly, the average time between the clicks is measured and used this to derive the tempo. When synchronized, the tempo will be quantized to the nearest setting.
- **MUTE** only allows the dry input signal through. Nonetheless, delay trails will continue even after the button is pressed. If the **DRY/WET** control is turned up fully and the Mute is activated, you will hear no more sound, because no signal is being allowed into the dry section.

Expert Controls

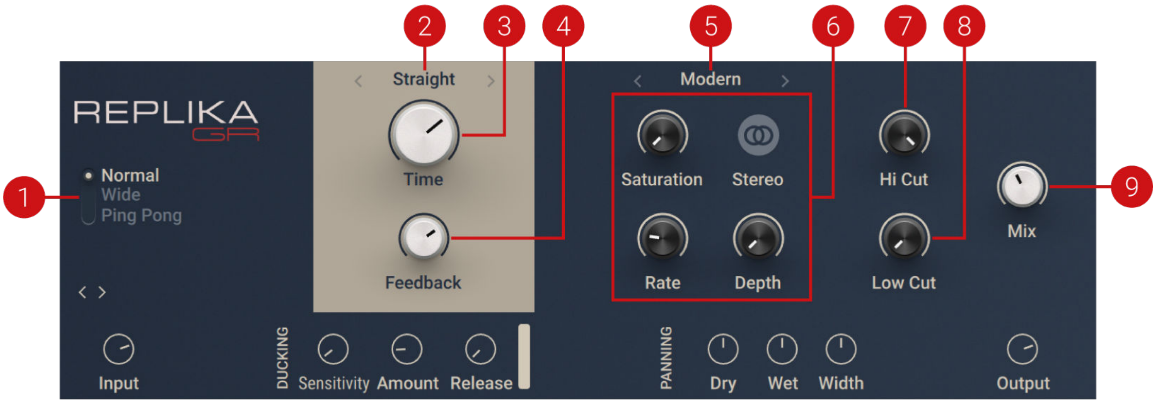
- **TEMPO SYNC** synchronizes the time controls of this module to the metronome or host, depending on the Sync setting.
- **INVERT** changes the phase of the delayed signal, affecting the elimination of frequencies in the mix of dry and processed signal. The result is particularly noticeable with short delays.
- **SYNC DELAY** synchronizes the four LFOs modulating the delays as controlled by the Rate and Depth parameters. When turned off, they are out of phase, causing a shifted modulation of each of the delay times. The result is a more complex effect spread out to both stereo channels.
- **DIFFUSION** controls how much the delay time is spread out between the four stages. Turning it up creates a stereo effect with four distinct delay signals.
- **BASS** adjusts a high-pass filter; turn down to reduce low frequencies in the delay section.
- **TREBLE** adjusts a low-pass filter; turn down to reduce high frequencies in the delay section.

REPLIKA GR

REPLIKA GR is an advanced delay that can be used to create a wide range of different echo and reverb effects. Based on both vintage and contemporary devices it features five unique styles, each with their own distinct sound character: the clean contemporary delay, a Bucket Brigade delay, a vintage digital delay, a tape echo, and a hybrid delay and reverb effect.

You can choose between three basic delay modes that define the stereo behavior, including a basic true-stereo mode, a stereo widening mode, and a ping pong mode. You can further fine-tune the effect using the **Hi Cut** and **Low Cut** filters, as well as the **PANNING** and **DUCKING** controls in the Expert panel.

This Component contains the following parameters and controls:



1. **Delay mode**: Switches between three different delay modes that determine the stereo behavior:
   - **Normal**: A basic echo mode that applies the same delay to both the left and right stereo channel. Therefore, the stereo image if the input signal is preserved.
   - **Wide**: A stereo widening mode that applies a short delay offset between the left and right stereo channel. Therefore, the echo effect appears on the sides of the stereo image.
   - **Ping Pong**: A ping pong mode that alternates the delay repetitions between the left and right stereo channel. Therefore, the echo effect appears to be bouncing back and forth in the stereo image.
2. **Time mode**: Selects between four modes that determine the behavior of the **Time** control. You can use the left and right arrows next to the mode's name to change the mode, or open the drop-down menu by clicking on the mode's name. The following modes are available:
   - **ms**: The delay time can bee freely adjusted in milliseconds.
   - **Straight**: The delay time can be adjusted in even note divisions (1/16, 1/8, 1/4, etc.) relative to the tempo of the Metronome.
   - **Dotted**: The delay time can be adjusted in dotted note divisions (1/16d, 1/8d, 1/4d, etc.) relative to the tempo of the Metronome.
   - **Triplets**: The delay time can be adjusted in triplet note divisions (1/16t, 1/8t, 1/4t, etc.) relative to the tempo of the Metronome.
3. **Time**: Adjusts the delay time. Depending on the selected Time mode, **Time** is either tempo synced and set in note divisions or freely adjusted in milliseconds.
4. **Feedback**: Adjusts the level of the signal that is being fed back to the delay's input. Increasing the feedback level creates more delay repeats that decay over time. Feedback levels of 100% and above are possible, allowing the delay repeats to build up until the point of self-oscillation.
5. **Style selector**: Selects between five different delay styles that determine the basic sound character. The controls in the Style section are specific to each style. The following styles are available:
   - **Modern**: Clean and defined delay with adjustable, tube-like saturation. For more information, see Modern Style.
   - **Analogue**: Dark and hazy emulation of BBD (Bucket Brigade Device) delay. For more information, see Analogue Style.
   - **Vintage Digital**: Warm and crunchy emulation of early digital delay effects. For more information, see Vintage Digital Style.
   - **Tape Echo**: Textured and lively emulation of classic tape delays. For more information, see Tape Echo Style.

- **Diffusion**: Clean delay combined with a unique reverb effect. For more information, see Diffusion Style.

6. **Style section**: Consists of specific controls for each of the five different delay styles. You can select the delay styles using the Style selector at the top of the Style section.
7. **Hi Cut**: Attenuates high-frequency content in the feedback path of the delay using a high-cut filter. Turned all the way to the right, the filter is off. Turning it to the left lowers the cutoff frequency of the filter, resulting in a darker tone of the delay.
8. **Low Cut**: Attenuates low-frequency content in the feedback path of the delay using a low-cut filter. Turned all the way to the left, the filter is off. Turning it to the right raises the cutoff frequency of the filter, resulting in a brighter tone of the delay.
9. **Mix**: Blends between the input signal and the effect signal.

The Expert panel contains the following parameters and controls:

- **Input**: Adjusts the input level of the Component.
- **Sensitivity**: Adjusts the threshold at which the ducking effect kicks in. Signal levels below this threshold will not trigger the ducking effect.
- **Amount**: Adjusts the strength of the ducking effect, which is the amount if gain reduction applied to the delay signal when the ducking effect is triggered.
- **Release**: Adjusts the release time of the ducking effect, which is the time it takes for the gain reduction to return to 0.
- **Dry**: Adjusts the panning of the input signal from left to right.
- **Wet**: Adjusts the panning of the delay signal from left to right.
- **Width**: Adjusts the stereo width of the output signal.
- **Output**: Adjusts the output level of the Component.

Modern Style

The Modern style is a clean and defined delay that can be further shaped with its adjustable, tube-like **Saturation**. The built-in modulation changes the delay time periodically. This shifts the timing and pitch of the delay signal, adding depth and movement to the sound. Unlike the built-in modulation, manual adjustments of the delay time do not alter the pitch of the delay signal.

This Component contains the following parameters and controls:

- **Saturation**: Adds tube-like saturation at the delay input. Turned all the way to the left, the knob bypasses saturation, turning it to the right pushes the sound from subtle warmth to overdrive. The amount of saturation is dependent on the input level.
- **Stereo**: Switches between stereo and mono operation for the built-in modulation. When activated, the modulation between the left and the right stereo channels is offset in time, resulting in a wide stereo effect. When deactivated, the modulation affects both the left and the right stereo channels in the same way.
- **Rate**: Adjusts the speed of the built-in modulation.
- **Depth**: Adjusts the amount of built-in modulation applied to the delay time.

Analogue Style

The Analogue style emulates the dark and hazy sound of BBD (Bucket Brigade Device) delays, which include old analogue studio effects as well as many contemporary guitar pedals. **BBD Type** offers four different models, giving you the full range of warm sounds associated with these delay effects.

The built-in modulation changes the delay time periodically. This shifts the timing and pitch of the delay signal, adding depth and movement to the sound. Manual adjustments of the delay time alter the pitch of the delay signal, which can be used creatively.

This Component contains the following parameters and controls:

- **BBD Type**: Selects one of four BBD delay models: **Clean**, **Warm**, **Dark**, **Grunge**. The character of the four models ranges from subtle filtering and distortion (**Clean**, **Warm**) to a highly degraded sound (**Dark**, **Grunge**).
- **Stereo**: Switches between stereo and mono operation for the built-in modulation. When activated, the modulation between the left and the right stereo channels is offset in time, resulting in a wide stereo effect. When deactivated, the modulation affects both the left and the right stereo channels in the same way.
- **Rate**: Adjusts the speed of the built-in modulation.
- **Depth**: Adjusts the amount of built-in modulation applied to the delay time.

Vintage Digital Style

The Vintage Digital style emulates the warm and crunchy sound of early digital delay effects. Four different **Quality** settings available, ranging from a bright sound with subtle textures to strong digital artifacts. Adjusting the delay time alters the pitch of the delay signal, which can be used creatively.

The built-in modulation changes the delay time periodically. This shifts the timing and pitch of the delay signal, adding depth and movement to the sound. Manual adjustments of the delay time alter the pitch of the delay signal, which can be used creatively.

This Component contains the following parameters and controls:

- **Quality**: Selects one of four quality settings for the delay: **High**, **Medium**, **Low**, **Crunch**. The **High** setting has a bright, subtly textured sound. **Medium** and **Low** sound darker and grittier. **Crunch** has a bright sound but also introduces a lot of digital artifacts to the signal.
- **Stereo**: Switches between stereo and mono operation for the built-in modulation. When activated, the modulation between the left and the right stereo channels is offset in time, resulting in a wide stereo effect. When deactivated, the modulation affects both the left and the right stereo channels in the same way.
- **Rate**: Adjusts the speed of the built-in modulation.
- **Depth**: Adjusts the amount of built-in modulation applied to the delay time.

Tape Echo Style

The Tape style emulates the textured and lively sound of classic tape delays. Replika GR gives you full control over their unique properties, including the amount of **Saturation**, the **Age** of the tape, and the intensity of **Wow & Flutter**. Changing the delay time results in smooth, slow transitions between settings and alters the pitch of the delay signal. This can be used to create interesting sound effects.

This Component contains the following parameters and controls:

- **Saturation**: Adjusts the amount of tape saturation from a clean sound to overdrive.
- **Noise**: Switches the tape hiss on or off. The amount if hiss depends on the **Age** setting.
- **Wow & Flutter**: Increases the effects introduced by mechanical imperfections of the tape delay's motor and tape transport, resulting in pitch variations over time.
- **Age**: Enhances the characteristics of an aging tape, like limited high frequency response and hiss (depending on **Noise** setting).

Diffusion Style

The Diffusion Style links a clean delay to a unique reverb effect. It can be used to create a wide range of different ambiences, from tight and resonant to unnaturally vast spaces. The **Amount** of diffusion applied to the delay signal is freely adjustable, and the **Movement** control adds modulation to the reverb effect.

Changing the delay time does not alter the pitch of the delay signal. However, adjusting the size of the reverb or increasing the amount of modulation can lead to pitch shifting sounds.

This Component contains the following parameters and controls:

- **Amount**: Adjusts the amount of diffusion applied to the delay signal, resulting in a reverb effect. High settings make the delay appear out of sync, so low settings are recommended if the rhythmic timing of the delay is essential.
- **Dense**: Switches between two density settings for the reflection pattern of the reverb effect. When activated, the reflection pattern is dense and washed out. When deactivated, the reflection pattern is sparse with a granular quality.
- **Movement**: Adjusts the depth and speed of modulation applied to the diffusion, shifting the timing and pitch of the reflections for a wide reverb effect.
- **Size**: Adjusts the swell, reflection pattern and decay of the reverb effect, giving the impression of differently sized spaces.

Tape Echo

The Tape Echo recreates the sound of tape-based delays. This component has two tape heads and also includes a spring reverb module.

Controls

- The peak LED in the upper-left corner indicates input overload that can be caused by feeding it with a too strong input signal, or by feedback.
- **INPUT MUTE** shuts off the signal going through the Tape Delay, letting only the dry signal pass through. Nonetheless, delay trails will continue even after the button is pressed. If the **DRY/WET** control is turned up fully and the Mute is activated, you will hear no more sound, because no signal is being allowed into the dry section.
- **DRY MUTE** will mute the dry sound, leaving only the processed sound. If the **Dry/Wet** control is turned fully down, you will hear no more sound, because no signal is allowed into the delay section.
- **TAP** allows tapping in the delay time. When you click on the **TAP** button repeatedly, the average time between the clicks is measured to derive the correct position for the **SPEED** knob and the **HEAD** knobs. When synchronized, these will be quantized to the nearest setting.

- The **HEAD A** and **HEAD B** knobs have five positions for varying delay times for each virtual tape head. They set the relation between both delay stages while the **Speed** setting sets the overall tempo of the virtual tape. Position 0 has no delay while the subsequent head positions will increase delay time by a constant amount, depending on the current Speed setting.
- **BASS** adjusts the low frequency response of the delayed signal.
- **TREBLE** adjusts the high frequency response of the delayed signal.
- **REV VOL** controls the amount of reverb added to dry signal by the component's spring reverb module.
- **SPEED** sets the speed of the virtual tape loop, influencing the delay times of Head A and Head B.
- **FEEDBACK** determines how much of the output feeds back into the input. Minimum feedback gives a single echo; increasing this parameter produces repeating echoes. It will also impart some modulation and eventually distortion to the signal.
- **ECHO VOL** controls the volume of the delayed output added to the dry signal. When turned down completely, the delay will have no audible effect.

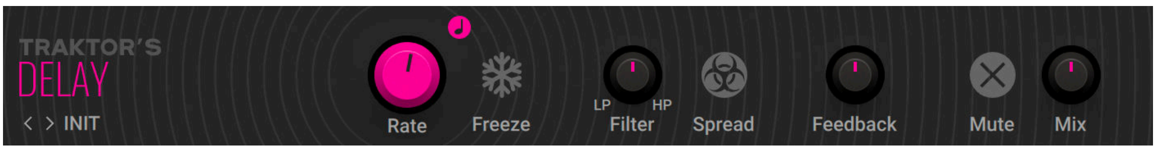
Expert Controls

- **Sync** synchronizes the Speed control to the metronome or host, depending on the Sync setting.
- **Tape Bass** controls the amount of bass response on the virtual tape.
- **Tape Treble** controls the amount of treble response on the virtual tape.
- **Dropouts** controls the simulation of dropouts occurring on a real tape after long periods of use.
- **Noise** controls the amount of tape noise that is added by the virtual tape.
- **Warble** controls the simulation of mechanical problems causing tapes to slip and modulate the signal. This effect is also known as “flutter and wow”.
- **Headroom** controls the amount of saturation the tape can take before distorting.
- **Motor Accel** controls how fast the tape speed adapts to changes of the Speed setting.
- **Head Mix** controls the mix between the outputs of head A and B.
- **STEREO** controls the stereo width of the delayed signal.
- **Revtime** sets the decay time of the reverb added to the dry signal.
- **Spring Length** adjusts the length of the virtual spring.

TRAKTOR’s Delay

TRAKTOR's Delay is a classic delay effect with additional functionality including the Freeze function for infinite delay repetitions, tempo sync, and stereo spread.

This Component contains the following parameters and controls:



- **Rate**: Adjusts the delay time.

- **Sync**: Synchronizes the delay time to the metronome. When **Sync** is active, **Rate** can be set in musical intervals relative to the tempo of the metronome. The interval can be set to 1/32, 1/16, 1/8, 3/16, 1/4, 3/8, and 4/4 notes.
- **Freeze**: Freezes the delay by muting the input and setting maximum feedback, creating infinite delay repetitions. Both **Filter** and **Rate** remain active and can be used to alter the sound of the delay repetitions. To achieve infinite delay repetitions, **Filter** needs to be set to center position.
- **Filter**: Adjusts the cutoff frequencies of the low-pass and high-pass filters in the feedback loop. In center position, no filtering is applied. Turning **Filter** to the left applies the low-pass filter and changes its cutoff frequency. Turning **Filter** to the right applies the high-pass filter and changes its cutoff frequency.
- **Spread**: Produces a wide stereo image by adding an offset between the delay times of the left and right stereo channel.
- **Feedback**: Adjusts the amount of feedback. Turning **Feedback** to the right increases the amount of delay repetitions.
- **Mute**: Mutes the input signal so that only the effect signal can be heard.
- **Mix**: Blends between the input signal and the effect signal.

Twin Delay

The Twin Delay combines two parallel delay modules to provide advanced stereo effects. Each of the delayed signal chains is assigned to one of the stereo channels and features a full set of controls. It works like a charm for bouncing the sound from left to right in any imaginable way.

Controls

- **DRY/WET** sets the amount of the signal being fed into the delay sections, controlling the intensity of the effect.
- **TIME (L/R)** sets the amount of delay time for each channel, i.e. the time interval between the straight sound and the appearance of the echo. When synchronized (see below), the scale turns from milliseconds into note values.
- **FEEDBACK (L/R)** determines how much of the output feeds back into the input for each channel. Minimum feedback gives a single echo; increasing this parameter produces repeating echoes.
- **LEVEL (L/R)** sets the volume for each channel, controlling both the mix and the overall volume of this component.
- **TAP** allows tapping in the delay time for both channels. When you click on the Tap button repeatedly, the average time between the clicks is measured and used this to derive the correct position for the Time knobs.
- **MUTE** only allows the dry input signal through. Nonetheless, delay trails will continue even after the button is pressed. If the **DRY/WET** control is turned up fully and the Mute is activated, you will hear no more sound, because no signal is being allowed into the dry section.

Expert Controls

- **X-Feedback** controls the amount of cross feedback, i.e. how much the output of the left channel feeds back into the input of the right channel and vice versa.
- **Stereo Width** controls the stereo panorama: When turned fully clockwise, the channels are completely separated. When centered, the processing is mono. When turned fully down, the channels are inverted, meaning that the left channel of the Dual Delay is routed to the right output channel and vice versa.

- **Pre-Delay** (**LEFT**/**RIGHT**) determines an initial delay for each channel, which is independent of the Time setting. This means you can have a quickly repeating delay that starts up to two seconds after the original signal. To achieve the typical ping-pong effect, set both channels to the same delay time and create an offset by increasing one of the channels' Pre-Delays.
- **Sync** synchronizes the Time controls to the metronome or host, depending on the Sync setting.
- **Pre-Sync** synchronizes the Pre-Delay controls to the metronome or host, depending on the Sync setting.

20.4. Distortion

Distortion has a long history in guitar effects and is an essential ingredient of many contemporary guitar sounds. From overdrive to fuzz, you can choose from a variety of models to create your personal distortion sound. The Distortion Components in GUITAR RIG are meticulously modeled after the sound and behavior of the original devices, including the interaction between their controls.

Big Fuzz

Big Fuzz models the sound of a classic distortion pedal suitable for 1970s Rock guitars. Its sound character is dirty with a lot of grunge.

This Component contains the following parameters and controls:

- **Sustain**: Adjusts the input level, or gain. Turning the control to the right increases the intensity of the distortion.
- **Tone**: Adjusts the frequency response. Turning the control to the left emphasizes high frequencies. Turning the control to the right emphasizes low frequencies.
- **Volume**: Adjusts the output level of the Component.

The Expert panel contains the following parameters and controls:

- **Bass**: Adjusts the low-frequency response.
- **Treble**: Adjusts the high-frequency response.

BITE

BITE combines bit and sample rate reduction effects, making it possible to add complex textures to the signal or completely deteriorate the sound in many different ways. It simulates the audio being sampled and replayed using a low-quality sampler with limited sample rate and bit depth, and can be used to create distortion effects that sound like vintage studio equipment, or inherently lo-fi sound sources, like old video games.

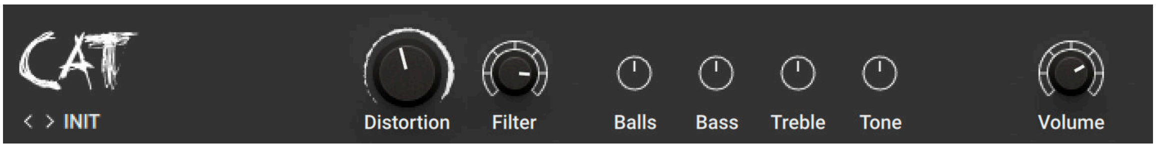
This Component contains the following parameters and controls:

- **Freq**: Adjusts the sampling frequency at which the input signal is resampled in a range of 100 Hz to 44100 Hz.
- **Depth**: Adjusts the amount of available quantization values by setting the bit depth in a range of 2 bits to 16 bits. Each sample of the signal is quantized to the available values. A lesser amount of values results in a more distorted sound.
- **DC**: Toggles between two different modes for the quantization of the input signal according to the set bit depth (**Bits**). When activated, these values do not include the zero level, sustaining the sound with a buzzing square wave. When deactivated, the output signal fades immediately.
- **Jitter**: Adjusts the amount of clock jitter. This adds fluctuations to the sampling rate of the resampling algorithm, effectively making the signal noisier. The jitter is added to the left and right stereo channels independently, resulting in a wide stereo image for the noise component.
- **Crunch**: Provides continuous control over the bit reduction effect, reducing the amount of used quantization values by lowering the signal level before the bit reduction algorithm. This allows you to smoothly control the resolution without stepping effects.
- **Pre Filt**: Adjusts the cutoff frequency of a low-pass filter that is applied to the input signal to remove frequencies that would produce aliasing noise in the resampling algorithm. At center position, the cutoff frequency is half of the sampling frequency. Its range is from 50 Hz to 22050 Hz.
- **Dither**: Adjusts the amount of noise that is added to the resampled signal to reduce distortion caused by quantization errors. In this effect, the noise amount can be increased for creative purposes. Independent noise sources are used for left and right stereo channels, resulting in a wide stereo image for the noise component.
- **Post Filt**: Adjusts the cutoff frequency of a low pass filter that is applied to the output signal to remove aliasing noise. At center position, the cutoff frequency is half of the sampling frequency. Its range is from 50 Hz to 22050 Hz.
- **Expand**: Changes the distribution of the quantization values in the amplitude range. When set fully to the left, low-level signals are quantized at a higher resolution. When set fully to the right, the resolution available for quantizing low-level signals is reduced, effectively turning them into pulse waves.
- **Saturation**: Drives the quantized signal into a saturator and compensates for the loudness increase caused by the saturation.
- **Mix**: Blends between the input signal and the effect signal by means of an equal-power crossfade.
- **HP**: Adjusts the cutoff frequency of a 1-pole high-pass filter that is applied to the output signal. It aims remove low-frequency and DC components from the signal. It provides three cutoffs 5 Hz, 100 Hz and 200 Hz.

Cat

Cat models the sound of a classic distortion pedal commonly used for both lead and rhythm guitar in Blues and Rock. Its sound character is responsive and dynamic.

This Component contains the following parameters and controls:



- **Distortion**: Adjusts the input level, or gain. Turning the control to the right increases the intensity of the distortion.
- **Filter**: Adjusts the frequency response. Turning the control to the left emphasizes high frequencies. Turning the control to the right emphasizes low frequencies.
- **Volume**: Adjusts the output level of the Component.

The Expert panel contains the following parameters and controls:

- **Balls**: Adjusts the amount of low-frequency punch. Turning the control to the right increases the amount of punch. Turning the control to the left produces a flatter, biting sound.
- **Bass**: Adjusts the low-frequency response.
- **Treble**: Adjusts the high-frequency response. Turning the control fully to the left can add a wah-wah-like effect.
- **Tone**: Adjusts the center frequency of an additional mid-frequency boost applied before the distortion stage.

DIRT

DIRT combines two distortion units, which additionally offer control over the type of clipping behavior. The two circuit-modeled diode clipping stages (**A**, **B**) can be configured in series or parallel using the Routing control. There are three modes of distortion (**I**, **II**, **III**) which can be used independently on each stage to set the type of clipping behavior. This allows you to explore different distortion characteristics and tonal qualities from subtle to more extreme.

This Component contains the following parameters and controls:

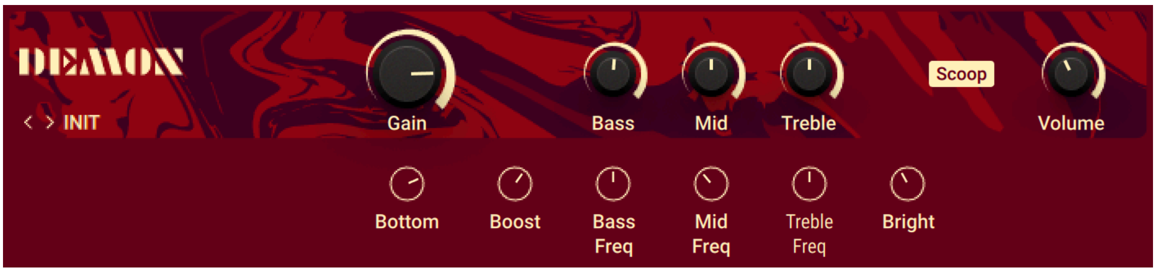
- **Drive**: Adjusts the input level, or gain. Turning **Drive** to the right increases the intensity of the distortion. The extra button at the top right of the control switches off the gain compensation that keeps consistent loudness levels. This can be used, for example, to drive a signal into the second stage at higher distortion levels.
- **Amount**: Adjusts the amount of distortion by introducing saturation in the first half of its range and wave folding in the second half. Instead of clipping the signal, wave folding folds the waveform of the signal back into itself.

- **Mode**: Selects one of three distortion modes to be used independently on stage **A** and stage **B**:
  - **I** is the most subtle mode and adds the least amount of coloring to the audio signal.
  - **II** is the default mode and is a well-balanced type of distortion with the brightest tone.
  - **III** is the most extreme mode which adds a crushed type of distortion with a dark tone.
- **Bias**: Introduces asymmetry into the signal by adding asymmetric behavior to the circuit, which produces even harmonics. This prevents the distorted audio from sounding hollow.
- **Tilt**: Applies filtering to the distorted signal. When turned clockwise, low-frequency content is attenuated and high-frequency content is boosted. When turned counterclockwise, low-frequency content is boosted and high-frequency content is attenuated.
- **Routing**: Determines how the signal is routed between stage **A** and stage **B** of the effect:
  - **A** > **B** serial routing configuration: The signal is fed into stage **A**, then into stage **B**.
  - **A** < **B** serial routing configuration: The signal is fed into stage **B**, then into stage **A**.
  - **A** + **B** parallel routing configuration: The signal is split into both stage **A** and stage **B** before being mixed with the **Blend** control.
- **Mix**: Blends between the dry signal and the wet signal by means of an equal-power crossfade. In **A** + **B** parallel routing configuration **Mix** is replaced with the **Blend** control, allowing you to blend between the output signals of stage **A** and stage **B**.
- **FX Trim**: Adjusts the output level of the wet signal in a range of -18 dB to +6 dB. This can be used to compensate for loudness differences between presets.

Demon Distortion

Demon Distortion models the sound of a distortion pedal commonly used for both lead and rhythm guitar in Hard Rock. Its sound character is sustained and sharp.

This Component contains the following parameters and controls:



- **Gain**: Adjusts the input level, or gain. Turning the control to the right increases the intensity of the distortion.
- **Bass**: Adjusts the low-frequency response.
- **Mid**: Adjusts the mid-frequency response.
- **Treble**: Adjusts the high-frequency response.
- **Scoop**: Attenuates the the mid-frequency range, producing a modern Metal sound. When Scoop is activated, the **Mid** control does not have an effect.
- **Volume**: Adjusts the output level of the Component.

The Expert panel contains the following parameters and controls:

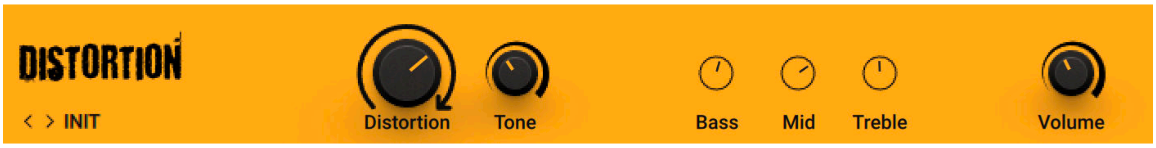
- **Bottom**: Adjusts the amount of low-frequency content passed through the distortion stage. Turning the control to the left produces a more defined, tighter sound.
- **Boost**: Adjusts the center frequency of an additional mid-frequency boost applied before the distortion stage.

- **Bass Freq**: Adjusts the center frequency of the filter applied using the **Bass** control.
- **Mid Freq**: Adjusts the center frequency of the filter applied using the **Mid** control.
- **Treble Freq**: Adjusts the center frequency of the filter applied using the **Treble** control.
- **Bright**: Adjusts the presence of the sound by attenuating or enhancing upper mid-frequency content.

Distortion

Distortion models the sound of a well-known distortion pedal that has been used extensively both live on stage as well as in the studio.

This Component contains the following parameters and controls:



- **Distortion**: Adjusts the input level, or gain. Turning the control to the right increases the intensity of the distortion.
- **Tone**: Adjusts the frequency response. Turning the control to the left attenuates high frequencies and emphasizes low frequencies. Turning the control to the right attenuates low frequencies and emphasizes mid frequencies.
- **Volume**: Adjusts the output level of the Component.

The Expert panel contains the following parameters and controls:

- **Bass**: Adjusts the low-frequency response.
- **Mid**: Adjusts the mid-frequency response.
- **Treble**: Adjusts the high-frequency response.

Fuzz

Fuzz models the sound of a 1960s fuzz pedal suitable for harmonically rich lead guitars that cut through the mix. Additionally, it can be used for buzzing vintage rhythm guitar.

This Component contains the following parameters and controls:

- **Fuzz**: Adjusts the input level, or gain. Turning the control to the right increases the intensity of the distortion.
- **Volume**: Adjusts the output level of the Component.

The Expert panel contains the following parameters and controls:

- **Bass**: Adjusts the low-frequency response.
- **Treble**: Adjusts the high-frequency response.

Gain Booster

Gain Booster enables you to boost the signal level at any point in your Rack's signal chain. You can use it to add a huge amount of overdrive, or to compensate for a low-level signal.

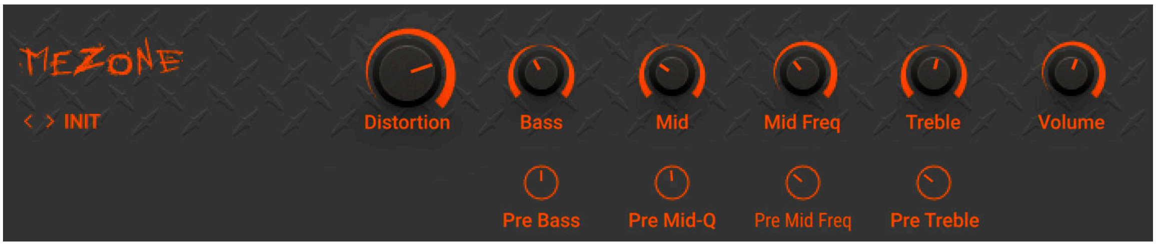
This Component contains the following parameters and controls:

- **Boost**: Adjust the amount of gain added to the signal.

Mezone

Mezone models the sound of a distortion pedal commonly used in Metal. Tone controls both pre and post distortion enable you to fine-tune the sound character.

This Component contains the following parameters and controls:



- **Distortion**: Adjusts the input level, or gain. Turning the control to the right increases the intensity of the distortion.
- **Bass**: Adjusts the low-frequency response.
- **Mid**: Adjusts the mid-frequency response.
- **Mid Freq**: Adjusts the center frequency of the filter applied using the **Mid** control.
- **Treble**: Adjusts the high-frequency response.
- **Volume**: Adjusts the output level of the Component.

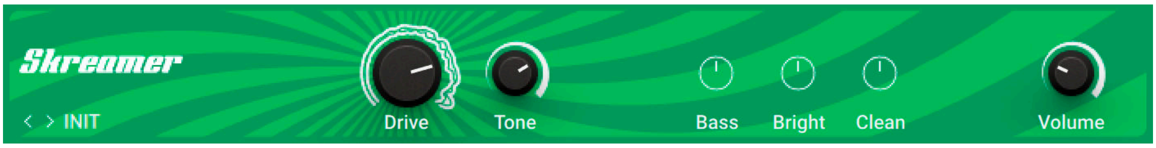
The Expert panel contains the following parameters and controls:

- **Pre Bass**: Adjusts the amount of low-frequency content passed through the distortion stage. Turning the control to the left produces a more defined, tighter sound.
- **Pre Mid Q**: Adjusts the resonance of an additional mid-frequency boost applied before the distortion stage.
- **Pre Mid Freq**: Adjusts the center frequency of an additional mid-frequency boost applied before the distortion stage.
- **Pre Treble**: Adjusts the amount of high-frequency content passed through the distortion stage. Turning the control to the left produces a darker, warmer sound.

Skreamer

Skreamer models the sound of an overdrive pedal suitable for rhythm and smooth lead guitars. Its sound character enhances mid-frequency content.

This Component contains the following parameters and controls:



- **Drive**: Adjusts the input level, or gain. Turning the control to the right increases the intensity of the distortion.
- **Tone**: Adjusts the frequency response. Turning the control to the left emphasizes low frequencies. Turning the control to the right emphasizes high frequencies.
- **Volume**: Adjusts the output level of the Component.

The Expert panel contains the following parameters and controls:

- **Bass**: Adjusts the low-frequency response.
- **Bright**: Adjusts the high-frequency response.
- **Clean**: Blends the input signal with the processed signal. When the control is turned fully to the left, only the processed signal is sent to the output. Turning the control to the right adds the input signal to the output.

Sledgehammer

Sledgehammer models the sound of a distortion pedal suitable for lead and rhythm guitars that cut through the mix. Its intense sound character adds a lot of presence to the sound.

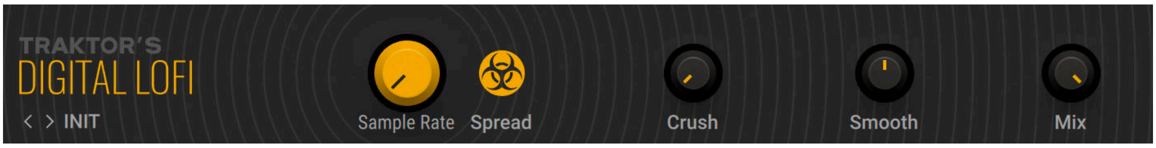
This Component contains the following parameters and controls:

- **Overdrive/Distortion**: Switches between two basic sound characters for the distortion. When set to **Overdrive**, the Component produces soft overdrive. When set to **Distortion**, the Component produces hard and edgy distortion.
- **Gain**: Adjusts the input level, or gain. Turning the control to the right increases the intensity of the distortion.
- **Bass**: Adjusts the low-frequency response.
- **Treble**: Adjusts the high-frequency response.
- **Contour**: Attenuates frequency content around the center frequency as set using the **Freq** control.
- **Freq**: Adjusts the center frequency of the filter applied using the **Contour** control. When **Contour** is turned fully to the left, the **Freq** control has no effect.
- **Volume**: Adjusts the output level of the Component.

TRAKTOR's Digital LoFi

Digital LoFi degrades the sound by reducing bit depth and sample rate of the audio. This effect adds digital artifacts and produces a low-fidelity sound.

This Component contains the following parameters and controls:

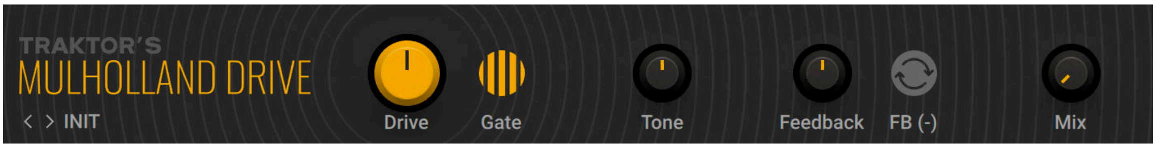


- **Sample Rate**: Reduces the sample rate of the audio. When turned fully to the right, the sample rate is 100 Hz.
- **Spread**: Produces a wide stereo image by adding an offset between the sample rates of the left and right stereo channel.
- **Crush**: Reduces the bit depth of of the audio. When turned fully to the right, the bit depth is slightly above 1 bit.
- **Smooth**: Smoothes the effect by introducing a lag to the sample rate reduction.
- **Mix**: Blends between the input signal and the effect signal.

TRAKTOR's Mulholland Drive

Mulholland Drive combines two overdrive units with internal feedback that can add a tonal character to the distortion sound, or even turn the effect into a sound generator.

This Component contains the following parameters and controls:



- **Drive**: Distributes the input signal to the two internal overdrive units. When turned fully left or right, the input signal is only sent to either unit. In center position, the input signal is equally sent to both units.
- **Gate**: Activates a noise gate that suppresses sustained feedback sounds.
- **Tone**: Adjusts the frequency of a filter applied to the internal feedback signal, effectively changing the pitch of the feedback sound. To create feedback sounds, **Feedback** needs to be turned to the left.
- **Feedback**: Adjusts the amount of the internal feedback. When turned fully to the left, Mulholland Drive behaves like a regular overdrive effect. When **Drive** and **Feedback** are set to high values, Mulholland drives becomes a sound generator.
- **FB (-)**: Inverts the polarity of the feedback signal, producing a different feedback sound character that only consists of uneven harmonics.
- **Mix**: Blends between the input signal and the effect signal.

TransAmp

TransAmp models the sound of a pioneering distortion pedal with included amplifier emulation. Its sound character is highly variable from smooth sounds on the verge of overdrive to raging distortion.

This Component contains the following parameters and controls:

- **Drive**: Adjusts the input level, or gain. Turning the control to the right increases the intensity of the distortion.
- **Amp**: Blends between three basic sound characters for the amplification: **TWEED**, **BRIT** (British), and **CALIF** (Californian).
- **Bass**: Adjusts the low-frequency response.
- **Treble**: Adjusts the high-frequency response.
- **Volume**: Adjusts the output level of the Component.

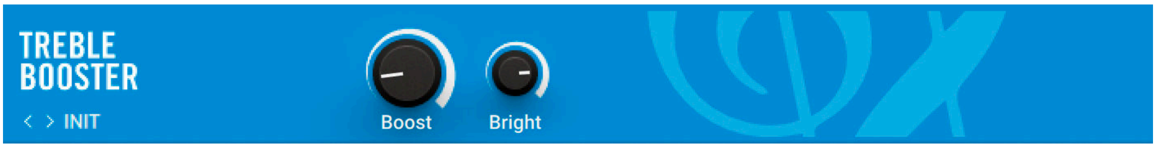
The Expert panel contains the following parameters and controls:

- **Clean**: Reduces the overall gain, producing a variation of the basic sound characters as set using **Amp**.
- **Cab & Mic**: Activates the integrated cabinet and microphone emulation.
- **Mic Pos**: Adjusts the microphone position of the cabinet and microphone emulation. When turning the control to the left, the microphone can be placed off-axis at a variable degree. When turning the control to the right, the microphone can be placed on-axis at a variable distance. This control only has an effect when **Cab & Mic** is activated.
- **Hot**: Adjusts the sound character of the cabinet and microphone emulation. When turning the control to the right, the emulation produces a brighter, hotter sound. This control only has an effect when **Cab & Mic** is activated.

Treble Booster

Treble Booster models the sound of special effect pedals used by artists like Brian May and Eric Clapton to alter the frequency response and distortion behavior of their tube amplifiers. The effect cuts low-frequency content before amplifying the signal, creating a focused sound with emphasis on mid- and high-frequency content.

This Component contains the following parameters and controls:



- **Boost**: Adjusts the input level, or gain. Turning the control to the right increases the intensity of the distortion.
- **Bright**: Adjusts the high-frequency response.

20.5. Dynamics

Dynamic effects include compressors, gates, and more specialized tools for shaping the level of the sound. You can use them to increase the loudness of the signal, balance or feature the attack and sustain of notes, and change the basic character of the sound.

Fast Comp

Fast Comp is a compressor that is tuned to quickly respond to transients in the signal. You can use it to control dynamics of fast playing and percussive sounds.

This Component contains the following parameters and controls:

- **Input**: Adjusts both the input level and the threshold simultaneously. Turning this knob clockwise will result in more compression.
- **Attack**: Adjusts the attack time, which is the time it takes the compressor to apply the full amount of gain reduction after the control signal rises above the threshold.
- **Gain Reduction Meter**: Displays the amount of gain reduction applied to the input signal.
- **Makeup**: Adjusts the amount of gain added to the signal after the compression. You can use this control to compensate for the gain reduction applied by the compressor and thus increase the overall loudness of the signal.

The Expert panel contains the following parameters and controls:

- **Sidechain**: Activates the sidechain input, allowing you to use an external signal as the compressor's control signal.

Limiter

Limiter is a specialized compressor with a high compression ratio and a fast response. It effectively prevents the signal to rise above the specified level. You can use it to protect your outputs and speakers from clipping, however it can also be applied creatively to shape the contour of a sound.

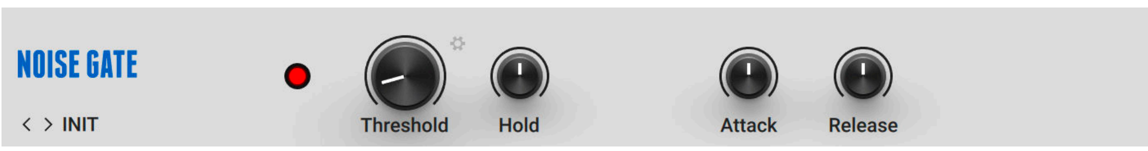
This Component contains the following parameters and controls:

- **Input**: Adjusts the input level of the Component.
- **Limit**: Adjusts the threshold level of the limiter. When the signal rises above the threshold, the maximum amount of gain reduction is applied immediately, effectively limiting the signal at this level.
- **Hold**: Adjusts the hold time, which determines for how long the limiter will apply gain reduction once the signal rises above the threshold (**Limit**).
- **Release**: Adjusts the release time, which is the time it takes the limiter to stop applying gain reduction after the signal falls below the threshold.

Noise Gate

Noise Gate is a specialized dynamics processor used to suppress unwanted noise in the signal. It produces silence when the instrument is not playing, therefore cutting out any noise or atmospheric sounds that are not part of the musical performance. It can also be used creatively by chopping up the input signal when it falls under the specified level.

This Component contains the following parameters and controls:



- **Threshold**: Adjusts the threshold level of the gate. When the signal falls below threshold, gain reduction is applied to the signal. Turning **Threshold** to the right raises the threshold, which increases the amount of gating by making the gate less sensitive to low input levels.
- **Learn**: Adjusts **Threshold** automatically by analyzing the input signal and determining the optimal threshold level. **Learn** needs to be activated when the instrument is not playing. Once **Threshold** is set according to the level of the noise floor, the function will be deactivated automatically.
- **Hold**: Adjusts the hold time, which determines for how long the gate will be open once the signal rises above the threshold (**Limit**). Increasing the hold time prevents stuttering effects.
- **Attack**: Adjusts the attack time, which is the time it takes the gate to reduce gain reduction after the signal rises above the threshold.
- **Release**: Adjusts the release time, which is the time it takes the gate to apply gain reduction after the signal fall below the threshold.

Noise Reduction

Noise Reduction is a specialized dynamics processor and filter used to reduce unwanted noise in the signal. It applies gain reduction and filtering when the instrument is not playing, therefore attenuating any noise or atmospheric sounds that are not part of the musical performance.

This Component contains the following parameters and controls:

- **Noise Reduction Meter**: Displays the amount of noise reduction applied to the input signal.
- **Threshold**: Adjusts the threshold level of the noise reduction. When the signal falls below threshold, gain reduction and filtering is applied to the signal. Turning **Threshold** to the right raises the threshold, which increases the amount of gain reduction and filtering by making the gate less sensitive to low input levels.
- **Learn**: Adjusts **Threshold** automatically by analyzing the input signal and determining the optimal threshold level. **Learn** needs to be activated when the instrument is not playing. Once **Threshold** is set according to the level of the noise floor, the function will be deactivated automatically.

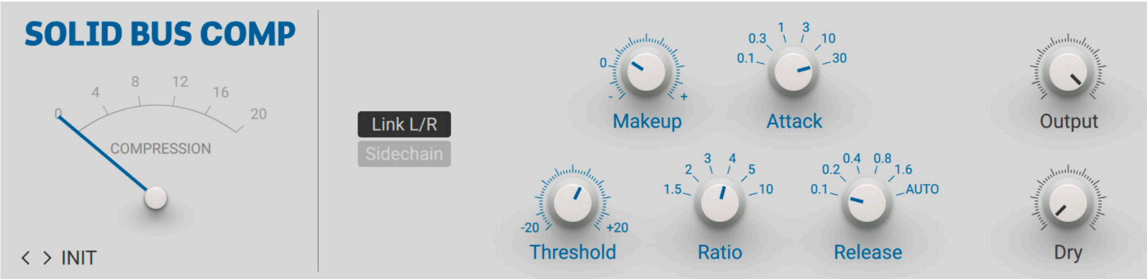
The Expert panel contains the following parameters and controls:

- **Release**: Adjusts the release time, which is the time it takes the noise reduction to apply gain reduction and filtering after the signal fall below the threshold.
- **De-Hiss**: Controls the strength of the filter reducing high frequencies.

SOLID BUS COMP

SOLID BUS COMP models the stereo bus compressor of a highly acclaimed mixing desk from the 70s. This renowned compressor is famed to be one of the most outstanding features of this desk, giving engineers the ability to create a huge sound even before mastering. Adding to the original features, SOLID DYNAMICS allows for parallel processing by mixing the input signal with the effect signal at the output.

This Component contains the following parameters and controls:

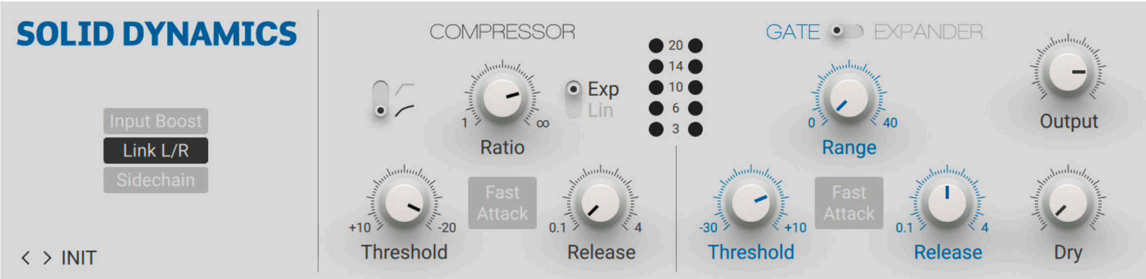


- **Gain Reduction Meter**: Displays the amount of gain reduction applied by the compressor.
- **Link L/R**: Sums of the left and right channel stereo channel for the compressor's control signal, thus applying the same processing to both channels. When deactivated, the left and right stereo channels are processed separately.
- **Sidechain**: Activates the sidechain input, allowing you to use an external signal as the compressor's control signal.
- **Threshold (Compressor)**: Adjusts the threshold level of the compressor. When the control signal rises above the threshold, gain reduction at a ratio set using the **Ratio** control is applied to the signal. Turning **Threshold** to the right lowers the threshold, which increases the amount of gain reduction by making the compressor more sensitive to low input levels.
- **Makeup**: Adjusts the amount of gain added to the signal after the compression. You can use this control to compensate for the gain reduction applied by the compressor and thus increase the overall loudness of the signal.
- **Ratio**: Selects between six different values for the compression ratio, which determines the amount of gain reduction applied to signals rising above the threshold. Ratios from **1.5** to **4** produce soft to moderate compression. The ratio of **5** produces strong compression. When set to a ratio of **10**, the compressor behaves more like a limiter.
- **Attack**: Selects between six different values for the attack time, which is the time it takes the compressor to apply the full amount of gain reduction after the control signal rises above the threshold.
- **Release**: Selects between six different values for the release time, which is the time it takes the compressor to stop applying gain reduction after the control signal falls below the threshold.
- **Output**: Adjusts the output level of the Component.
- **Dry**: Adjusts the level of the input signal that is added to the effect signal at the output.

SOLID DYNAMICS

SOLID DYNAMICS models the channel dynamics section of a highly acclaimed mixing desk from the 70s, including both a compressor and a gate. Its precise and transparent sound is considered a reference among mixing engineers. Adding to the original features, SOLID DYNAMICS allows for parallel processing by mixing the input signal with the effect signal at the output, and boosting the input level to achieve more compression.

This Component contains the following parameters and controls:



- **Input Boost**: Boosts the input level by 10 dB. You can use this to achieve more compression.
- **Link L/R**: Sums of the left and right channel stereo channel for the compressor's control signal, thus applying the same processing to both channels. When deactivated, the left and right stereo channels are processed separately.
- **Sidechain**: Activates the sidechain input, allowing you to use an external signal as the compressor's control signal.
- **Knee**: Selects between two different compression curves, soft knee and hard knee. When hard knee is selected, the full amount of gain reduction is applied immediately when the control signal rises above the threshold. This results in a fast response to transients, but can also lead to distortion. When soft knee is selected, the amount of gain reduction is gradually increased around the threshold level, resulting in a smoother response to the control signal.
- **Threshold (Compressor)**: Adjusts the threshold level of the compressor. When the control signal rises above the threshold, gain reduction at a ratio set using the **Ratio** control is applied to the signal. Turning **Threshold** to the right lowers the threshold, which increases the amount of gain reduction by making the compressor more sensitive to low input levels.
- **Ratio**: Adjust the relative amount of gain reduction applied to signals rising above the threshold. When turned fully to the left, no gain reduction is applied. When turned fully to the right, the compressor acts as a limiter.
- **Fast Attack (Compressor)**: Decreases the attack time from 30 ms to 3ms for a gain reduction of 20 dB. The attack time is the time it takes the compressor to apply the full amount of gain reduction after the control signal rises above the threshold.
- **Exp/Lin**: Switches between two different release behaviors of the compressor. When **Exp** is selected, the gain reduction decreases exponentially during the release time. When Lin is selected, the gain reduction decreases linearly during the release time.
- **Release (Compressor)**: Adjusts the release time, which is the time it takes the compressor to stop applying gain reduction after the control signal falls below the threshold. **Release** can be set in the range from 0.1 s to 4 s.
- **Gain Reduction Meter**: Displays the amount of gain reduction applied by the compressor.
- **Threshold (Gate)**: Adjusts the threshold level of the gate. When the control signal falls below threshold, gain reduction as set using the **Range** control is applied to the signal. Turning **Threshold** to the right raises the threshold, which increases the amount of gating by making the gate less sensitive to low input levels.
- **Range**: Adjust the maximum amount of gain reduction applied to signals falling below the threshold. When turned fully to the left, no gain reduction is applied. When turned fully to the right, 40 dB of gain reduction are applied.
- **Fast Attack (Gate)**: Decreases the attack time from 1.5 ms to 100 µs. The attack time is the time it takes the gate to reduce gain reduction after the control signal rises above the threshold.
- **Release**: Adjusts the release time, which is the time it takes the gate to apply gain reduction after the control signal falls below the threshold. **Release** can be set in the range from 0.1 s to 4 s.

- **Output**: Adjusts the output level of the Component.
- **Dry**: Adjusts the level of the input signal that is added to the effect signal at the output.

Stomp Compressor

Stomp Compressor models the sound of contemporary compressors with a clean and controlled sound. You can use it to tighten up the dynamics of a sound without changing its character.

This Component contains the following parameters and controls:

- **Gain Reduction Meter**: Displays the amount of gain reduction applied to the input signal.
- **Sustain**: Adjusts both the input level and the threshold simultaneously. Turning this knob clockwise will result in more compression.
- **Output**: Adjusts the output level of the Component.

The Expert panel contains the following parameters and controls:

- **Threshold**: Adjusts the threshold level of the compressor. When the control signal rises above the threshold, gain reduction is applied to the signal. Turning **Threshold** to the left lowers the threshold, which increases the amount of gain reduction by making the compressor more sensitive to low input levels.
- **Attack**: Adjusts the attack time, which is the time it takes the compressor to apply the full amount of gain reduction after the control signal rises above the threshold.
- **Release**: Adjusts the release time, which is the time it takes the compressor to stop applying gain reduction after the control signal falls below the threshold.
- **Sidechain**: Activates the sidechain input, allowing you to use an external signal as the compressor's control signal.

SUPERCHARGER

SUPERCHARGER is an advanced compressor with a strong sound character. Its clear interface enables you to quickly find the right settings. The **Saturation** stage creates rich sounds by adding harmonics. The **Attack** and **Release** times can be used to add more bite or sustain to sounds. The **Character** control colors the sound based on three different modes, **Fat**, **Warm**, and **Bright**. Finally, the Expert Panel provides a filter for the sidechain signal and different stereo routing modes including M/S (mid/side) processing.

This Component contains the following parameters and controls:

- **Input Level Indicator**: Displays the input level and indicates the correct setting of the **Input Trim** control. When the level is set correctly, the center indicator is lit green. When the level is too low, the left arrow indicator is lit red. When the level is too high, the right arrow indicator is lit red.
- **Input Trim**: Adjusts the input level. The correct setting is shown by the Input Level indicator.
- **Sidechain**: Activates the sidechain input, allowing you to use an external signal as the compressor's control signal.
- **Gain Reduction Meter**: Displays the amount of gain reduction applied to the input signal.
- **Output Level Meter**: Displays the level of the output signal.
- **Compress**: Adjusts the amount of compression applied to the input signal. Turning the control to the right increases the amount of compression while retaining an even loudness level (if Input Trim is set correctly).
- **Punch**: Increases the attack time of the compressor, which emphasizes transients and performs well on drums.
- **Dirt**: Introduces saturation to the input signal, which adds presence and warmth to the sound.
- **Mix**: Blends between the effect signal (**Wet**) and the input signal (**Dry**). Blending in the input signal can be used to increase dynamics and preserve transients.
- **Output**: Adjusts the output level of the Component.

SUPERCHARGER GT

SUPERCHARGER GT is an advanced compressor with a strong sound character. Its clear but powerful interface enables you to quickly find the right settings while also providing detail parameters for fine-tuning your sound. The **Saturation** stage creates rich sounds by adding harmonics. The **Attack** and **Release** times can be used to add more bite or sustain to sounds. The **Character** control colors the sound based on three different modes, **Fat**, **Warm**, and **Bright**. Finally, the Expert Panel provides a filter for the sidechain signal and different stereo routing modes including M/S (mid/side) processing.

This Component contains the following parameters and controls:

- **Input Level Indicator**: Displays the input level and indicates the correct setting of the **Input Trim** control. When the level is set correctly, the center indicator is lit green. When the level is too low, the left arrow indicator is lit red. When the level is too high, the right arrow indicator is lit red.
- **Input Trim**: Adjusts the input level. The correct setting is shown by the Input Level indicator.

- **Sidechain**: Activates the sidechain input, allowing you to use an external signal as the compressor's control signal.
- **Saturation**: Adjusts the amount of saturation added to the input signal. The type of saturation can be switched between **Mild**, **Moderate**, and **Hot**.
- **Mild/Moderate/Hot**: Switches between three types of saturation. **Mild** subtly colors the sound. **Moderate** produces a pronounced saturation effect. **Hot** adds strong saturation and even distortion to the sound.
- **Gain Reduction Meter**: Displays the amount of gain reduction applied to the input signal.
- **Output Level Meter**: Displays the level of the output signal.
- **Compress**: Adjusts the amount of compression applied to the input signal. Turning the control to the right increases the amount of compression while retaining an even loudness level (if Input Trim is set correctly).
- **Attack**: Adjusts the attack time, which is the time it takes the compressor to apply the full amount of gain reduction after the control signal rises above the threshold.
- **Release**: Adjusts the release time, which is the time it takes the compressor to stop applying gain reduction after the control signal falls below the threshold.
- **Gentle/Punch/Slam**: Switches between three distinct preset settings for the **Attack** and **Release**. **Gentle** produces smooth compression suitable for a wide variety of signals. **Punch** emphasizes transients and performs well on drums. **Slam** produces intense compression that can be used for drastic effects.
- **Character**: Adjusts the color of the signal by applying filtering. The type of filtering can be switched between **Fat**, **Warm**, and **Bright**.
- **Fat/Warm/Bright**: Switches between three different types of filtering. **Fat** emphasizes low- and high-frequency content. **Warm** reduces high-frequency content and enhances low-frequency content. **Bright** enhances high-frequency content and reduces low-frequency content.
- **Mix**: Blends between the effect signal (**Wet**) and the input signal (**Dry**). Blending in the input signal can be used to increase dynamics and preserve transients.
- **Output**: Adjusts the output level of the Component.

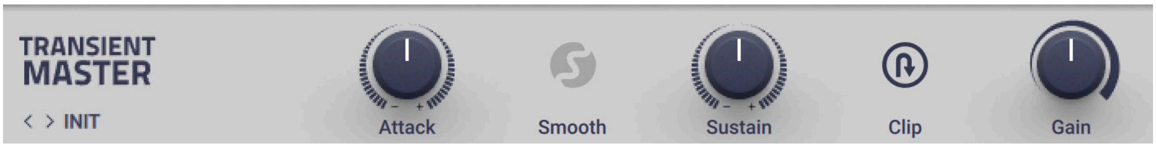
The Expert panel contains the following parameters and controls:

- **Detector HP**: Switches between three different settings for a high-pass filter that is applied to the compressor's control signal. **Off** deactivates the high-pass filter. **100Hz** attenuates frequency content below 100 Hz. **300Hz** attenuates frequency content below 300 Hz.
- **Mode**: Switches between three different stereo routing modes. **Stereo Link** applies the same amount of gain reduction to both the left and right channels, preserving the original stereo image. **Dual Mono** applies individual amounts of gain reduction to the left and right stereo channel, increasing loudness in both channels independently. **MS** applies individual amounts of gain reduction to the mid and the side signal, increasing the width of the stereo image.

TRANSIENT MASTER

TRANSIENT MASTER enables you to emphasize or attenuate the transients of your audio material by manipulating its attack and sustain phases. Unlike compressors and limiters, TRANSIENT MASTER does not use the level of your signal to decide when to come into effect, but rather modifies the envelopes of every attack and sustain phase. A notable benefit of this processing is that it affects all parts of the signal, whatever their level is, therefore retaining the natural character of your sound.

This Component contains the following parameters and controls:



- **Attack**: Sharpens/softens the attack phases in your signal. With the knob at the center position, the attack phases are not altered. From this position, turning the **Attack** knob to the left softens the attack phases, while turning it to the right sharpens them.
- **Smooth**: Activates an operating mode specifically designed for guitar sounds. When **Smooth** is enabled, the attack shaping is slightly smoother. This notably produces less distortion on guitar sounds that already contain a substantial distortion component. When working on other audio material (e.g. acoustic guitar, drums, etc.), you can deactivate the **Smooth** button to achieve faster attacks.
- **Sustain**: Prolongs/shortens the sustain phases in your signal. With the knob at the center position, the sustain phases are not altered. From this position, turning the **Sustain** knob to the left shortens the sustain phases, while turning it to the right prolongs them.
- **Limit**: Activates a hard limiter at the output, preventing the output signal from clipping. This can be useful when the **Attack** knob is set to a high value as this may produce amplified attack phases that become too loud.
- **Gain**: Adjusts the make-up gain. This allows you to offset the overall output level once you have set the desired effect, in order to counterbalance the gain or loss of level that might occur.

Tube Compressor

Tube Compressor models the sound of compressors based on analog tubes, producing a harmonically rich sound. You can use it to bring out the character in your sound.

This Component contains the following parameters and controls:

- **Gain Reduction Meter**: Displays the amount of gain reduction applied to the input signal.
- **Input**: Adjusts the input level of the Component.
- **Threshold**: Adjusts the threshold level of the compressor. When the control signal rises above the threshold, gain reduction at a ratio set using the **Ratio** control is applied to the signal. Turning **Threshold** to the left lowers the threshold, which increases the amount of gain reduction by making the compressor more sensitive to low input levels.
- **Ratio**: Adjust the relative amount of gain reduction applied to signals rising above the threshold. When turned fully to the left, no gain reduction is applied. When turned fully to the right, the compressor acts as a limiter.
- **Attack**: Adjusts the attack time, which is the time it takes the compressor to apply the full amount of gain reduction after the control signal rises above the threshold.
- **Release**: Adjusts the release time, which is the time it takes the compressor to stop applying gain reduction after the control signal falls below the threshold.

- **Output**: Adjusts the output level of the Component.

The Expert panel contains the following parameters and controls:

- **Saturation**: Adjusts the amount of saturation added to the signal.1
- **Knee**: Adjusts the compressor's response from soft knee to hard knee. A hard knee response applies the full amount of gain reduction immediately when the control signal rises above the threshold. This results in a fast response to transients, but can also lead to distortion. A soft knee response increases the gain reduction gradually around the threshold level, resulting in a smoother response to the control signal.
- **Dynamic**: Adjusts the dynamic response of the tube emulation used in the compressor. Turning the control to the right reduces the dynamics slightly.
- **Sidechain**: Activates the sidechain input, allowing you to use an external signal as the compressor's control signal.

VC 160

The VC 160 faithfully reproduces the characteristics and features of a highly popular compressor from the 1970s. Notably, it provides you with the same simple interface, which mainly relies on three knobs. Manual setting of attack and release times is not required since these were automatically determined by the feed-forward gain reduction stage. Adding to the original features, VC 160 offers a sidechain input and parallel compression using the **Dry** control.

This Component contains the following parameters and controls:

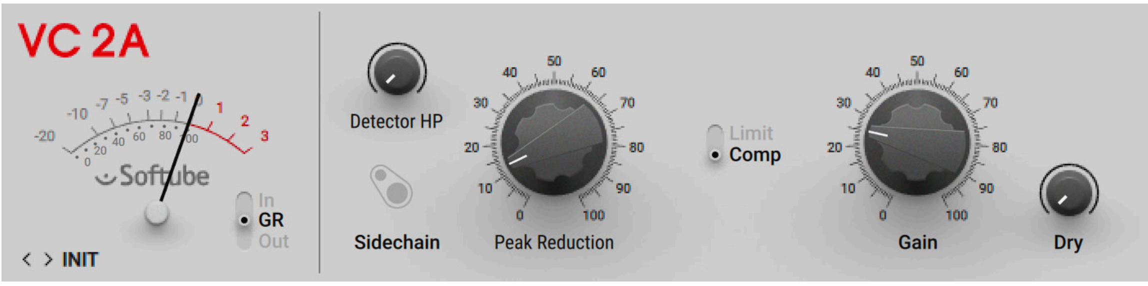
- **VU meter**: Allows you to visually monitor the compression in real-time. This VU meter has three operation modes, which you can select using the Display mode selector (see below).
- **In/GR/Out**: Switches the VU meter between three different modes:
  - **In**: The VU meter displays the level of the input signal. This mode is useful to ensure the optimal input level.
  - **GR** (Gain Reduction): The VU meter displays the amount of gain reduction applied to the input signal. This mode is useful when adjusting the **Compression** control.
  - **Out**: The VU meter displays the level of the output signal. This mode is useful when adjusting the **Output** control.
- **Detector HP**: Applies a high-pass filter to the control signal that is sent to the compressor's detector. When this knob is set fully counter-clockwise, no filtering is applied. Turning the knob clockwise progressively excludes low frequencies from the control signal sent to the detector.
- **Sidechain**: Activates the sidechain input, allowing you to use an external signal as the compressor's control signal.
- **Above/Below**: These two LEDs indicate whether the current level of the control signal is below (**Below** lit) or above (**Above** lit) the defined threshold, thus helping you adjust the **Threshold** value.
- **Threshold**: Adjusts the threshold level above which compression is applied.

- **Compression**: Adjusts the compression ratio. This directly affects how much gain reduction is applied to the input signal. When the control is turned fully to the left, the compressor acts as a limiter by applying a virtually infinite compression ratio.
- **Gain**: Adjusts the make-up gain. This allows you to offset the overall output level once you have set the desired compression.
- **Dry**: Blends the input signal with the compressed signal, facilitating parallel compression. When the control is turned fully to the left, only the compressed signal is sent to the output. Turning the control to the right adds the input signal to the output.

VC 2A

The VC 2A faithfully reproduces the characteristics and features of a highly popular compressor from the 1960s. Notably, it provides you with the same simple interface, which mainly relies on two knobs. Manual setting of attack and release times is not required since they are inherent to the properties of the circuit and dynamically adapt to the input signal. Adding to the original features, VC 160 offers a sidechain input and parallel compression using the **Dry** control.

This Component contains the following parameters and controls:



- **VU meter**: Allows you to visually monitor the compression in real-time. This VU meter has three operation modes, which you can select using the Display mode selector (see below).
- **In/GR/Out**: Switches the VU meter between three different modes:
  - **In**: The VU meter displays the level of the input signal. This mode is useful to ensure the optimal input level.
  - **GR** (Gain Reduction): The VU meter displays the amount of gain reduction applied to the input signal. This mode is useful when adjusting the **Compression** control.
  - **Out**: The VU meter displays the level of the output signal. This mode is useful when adjusting the **Output** control.
- **Detector HP**: Applies a high-pass filter to the control signal that is sent to the compressor's detector. When this knob is set fully counter-clockwise, no filtering is applied. Turning the knob clockwise progressively excludes low frequencies from the control signal sent to the detector.
- **Sidechain**: Activates the sidechain input, allowing you to use an external signal as the compressor's control signal.
- **Peak Reduction**: Adjusts the amount of compression applied to the input signal. This knob affects both the compression ratio and the threshold level simultaneously.
- **Limit/Comp**: Switches between two operation modes. When set to **Comp**, the curve of the gain reduction is softer and the compression ratio is lower. When set to **Limit**, the compression ratio is higher and the VC 2A tends to operate more like a limiter.
- **Gain**: Adjusts the make-up gain. This allows you to offset the overall output level once you have set the desired compression.

- **Dry**: Blends the input signal with the compressed signal, facilitating parallel compression. When the control is turned fully to the left, only the compressed signal is sent to the output. Turning the control to the right adds the input signal to the output.

VC 76

The VC 76 faithfully reproduces the characteristics and features of a highly popular compressor from the 1960s. Notably, it provides you with the distinctive **All** ratio producing a very unusual compression, as well as the **1** ratio coloring the sound without applying any compression. VC 76 unique sound character is emphasized by extremely short attack and release times. Adding to the original features, VC 76 offers a sidechain input and parallel compression using the **Dry** control.

This Component contains the following parameters and controls:

- **VU meter**: Allows you to visually monitor the compression in real-time. This VU meter has three operation modes, which you can select using the Display mode selector (see below).
- **In/GR/Out**: Switches the VU meter between three different modes:
  - **In**: The VU meter displays the level of the input signal. This mode is useful to ensure the optimal input level.
  - **GR** (Gain Reduction): The VU meter displays the amount of gain reduction applied to the input signal. This mode is useful when adjusting the **Compression** control.
  - **Out**: The VU meter displays the level of the output signal. This mode is useful when adjusting the **Output** control.
- **Sidechain**: Activates the sidechain input, allowing you to use an external signal as the compressor's control signal.
- **Input**: Adjusts both the input level and the threshold simultaneously. Turning this knob clockwise will result in more compression.
- **Attack**: Adjusts the attack time of the compressor, i.e. the time it takes for the compressor to come into full effect once the threshold level has been reached. Note that the attack phase, or the time it takes for the compressor to reach full level, gets shorter when the knob is turned clockwise.
- **Release**: Adjusts the release time of the compressor, i.e. the time it takes for the compressor to get back to its standby state after the signal level has fallen below the threshold level. Note that the release phase, or the time it takes for the compressor to reach its standby state, gets shorter when the knob is turned clockwise.

- **Ratio**: Selects the compression ratio. This directly affects how much gain reduction is applied to the input signal. Following ratios are available: 1:1 (**1**), 4:1 (**4**), 8:1 (**8**), 12:1 (**12**), and 20:1 (**20**):
  - The ratio 4:1 (**4**) generates moderate compression.
  - The ratio 8:1 (**8**) generates severe compression.
  - With the ratios 12:1 (**12**) and especially 20:1 (**20**) the compressor tends to behave like a limiter.
  - The position **ALL**, at the top of the selector, was not available as such in VC 76's ancestor. It originates in the so-called "All-Button" mode used by many engineers. Indeed, on the hardware unit the ratios were selected via a set of buttons. By pressing all buttons simultaneously, you could get an extreme (and variable!) form of overdriven compression, which notably became a distinguishing feature of the "British sound" from the 1960s and 1970s. This behavior is made available here via this **ALL** position.
  - On the other hand, with the ratio 1:1 (**1**) no compression is applied. Nevertheless, the input signal still passes through the unit, thus getting its circuitry's signature sound. This is sometimes called the "No-Button" mode (see the "All-Button" mode above for the explanation).
- **Output**: Adjusts the make-up gain. This allows you to offset the overall output level once you have set the desired compression.
- **Dry**: Blends the input signal with the compressed signal, facilitating parallel compression. When the control is turned fully to the left, only the compressed signal is sent to the output. Turning the control to the right adds the input signal to the output.

Volume

Volume is a level control that you can use to boost or attenuate the signal level at any point in your Rack's signal chain.

This Component contains the following parameters and controls:

- **Gain**: Adjusts the signal level. The Expert Panel provides settings that enable you to change the range of the control.
- **Volume**: Adjusts the output level of the Component.

The Expert panel contains the following parameters and controls:

- **Min Volume**: Adjusts the minimum level of the **Gain** control.
- **Half Volume**: Adjusts the response of the **Gain** control by setting how much if its overall range is reached at center position.

20.6. EQ

EQs, or equalizers, shape the frequency content of the signal by boosting or cutting specific frequency bands. You can use them to balance the tonality of the sound, remove resonances or rumble, and make drastic adjustments in the frequency spectrum for creative purposes.

Custom EQ

Custom EQ models the sound of a popular vintage equalizer. Its sound character is colorful and warm.

This Component contains the following parameters and controls:

- **Tone**: Adjusts the balance between low- and high-frequency content. Turning the control to the left boosts low-frequency content and attenuates high-frequency content. Turning the control to the right boosts high-frequency content and attenuates low-frequency content.
- **Freq**: Adjusts the center frequency of the mid-frequency attenuation as set using the **Scoop** control.
- **Scoop**: Adjusts the amount of mid-frequency attenuation at the center frequency set using the **Freq** control.
- **Output**: Adjusts the output level of the Component.

Equalizer Graphic

Equalizer Graphic is a graphic equalizer with eight frequency bands. You can adjust the gain of each band individually.

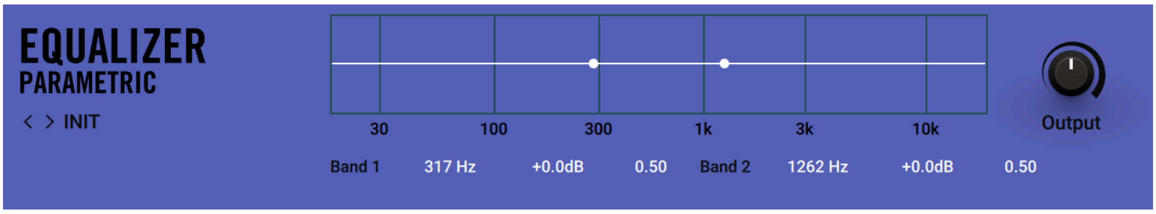
This Component contains the following parameters and controls:

- **Gain Range**: Adjusts the maximum amount of boost and attenuation available in each frequency band.
- **EQ Graph**: Shows and edits the equalizer response. Clicking and dragging the white squares up and down adjusts the gain of each frequency band. Dragging it up increases the gain, dragging it down decreases the gain. Double-clicking a white square resets the gain of the respective frequency band to 0.
- **Minimum Frequency (leftmost frequency value)**: Adjusts the frequency of the lowest frequency band. In combination with the Maximum Frequency, the value determines the relative frequencies of all frequency bands.
- **Maximum Frequency (rightmost frequency value)**: Adjusts the frequency of the highest frequency band. In combination with the Minimum Frequency, the value determines the relative frequencies of all frequency bands.
- **Output**: Adjusts the output level of the Component.

Equalizer Parametric

Equalizer Parametric is a parametric equalizer with two frequency bands. You can adjust the frequency, gain, and bandwidth of each band individually.

This Component contains the following parameters and controls:



- **EQ Graph**: Shows and edits the equalizer response. Clicking and dragging the white squares up and down adjusts the gain of each frequency band. Clicking and dragging left and right adjusts the frequency. Double-clicking a white square resets the respective frequency band to default values.
- **Output**: Adjusts the output level of the Component.

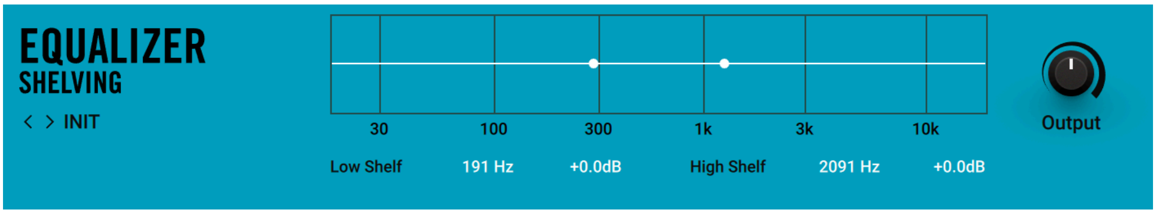
The Expert panel contains the following parameters and controls:

- **Band 1**: Sets the frequency (left value, in **Hz**), gain (middle value, in **dB**), and the bandwidth (right value) of the first frequency band.
- **Band 2**: Sets the frequency (left value, in **Hz**), gain (middle value, in **dB**), and the bandwidth (right value) of the second frequency band.

Equalizer Shelving

Equalizer Shelving is a shelving equalizer with two frequency bands. You can adjust the frequency and gain of each band individually.

This Component contains the following parameters and controls:



- **EQ Graph**: Shows and edits the equalizer response. Clicking and dragging the white squares up and down adjusts the gain of each frequency band. Clicking and dragging left and right adjusts the frequency. Double-clicking a white square resets the respective frequency band to default values.
- **Output**: Adjusts the output level of the Component.

The Expert panel contains the following parameters and controls:

- **Low Shelf**: Sets the frequency (left value, in **Hz**) and gain (right value, in **dB**) of the low-frequency shelving equalizer.
- **High Shelf**: Sets the frequency (left value, in **Hz**) and gain (right value, in **dB**) of the high-frequency shelving equalizer.

SOLID EQ

SOLID EQ models the channel equalizer section of a highly acclaimed mixing desk from the 70s, including two semi-parametric and two fully parametric equalizer bands: **LF** (low frequency), **HF** (high frequency), **LMF** (low-mid frequency), **HMF** (hi-mid frequency). Its precise and transparent sound is considered a reference among mixing engineers. Adding to the original features, SOLID EQ features the low-pass and high-pass filters from the original channel strip.

This Component contains the following parameters and controls:

- **HPF**: Adjusts the cutoff frequency of the high-pass filter in the range from 30 Hz to 600 Hz. Frequencies below the cutoff frequency are increasingly attenuated. When turned fully to the left, the high-pass filter is switched off.
- **LPF**: Adjusts the cutoff frequency of the low-pass filter in the range from 50 kHz to 3 kHz. Frequencies above the cutoff frequency are increasingly attenuated. When turned fully to the left, the low-pass filter is switched off.
- **LF Bell/Shelf**: Selects between two filter types for the **LF** band. When set to shelf (low position), frequencies below the filter frequency as set with **Freq** are evenly attenuated or boosted, depending on the **Gain** setting. When set to bell (high position), frequencies around the filter frequency are attenuated or boosted.
- **LF Freq**: Adjusts the filter frequency of the **LF** band.
- **LF Gain**: Adjusts the amount of attenuation or boost applied to the **LF** band as set with **Freq**. The **LF** band can either act as a bell or shelf filter (see **LF Bell/Shelf**).
- **LMF Freq**: Adjusts the filter frequency of the **LMF** band.
- **LMF Gain**: Adjusts the amount of attenuation or boost applied to the **LMF** band. The **LMF** band acts as a bell filter that attenuates or boosts frequencies around the filter frequency as set with **Freq**.
- **LMF Q**: Adjusts the Q factor, or resonance, of the **LMF** band. The Q factor determines the width of the frequency band around the filter frequency that is attenuated or boosted.
- **G/E response**: Switches between two different filter characters:
  - **G**: In this setting, the **LF** and **HF** bands have a steeper filter slope. Furthermore, the width of the **LMF** and **HMF** bands vary depending on the **Gain** setting: The lower the attenuation or boost, the wider the frequency band.
  - **E**: In this setting, the **LF** and **HF** bands have a softer filter slope. Furthermore, the width of the **LMF** and **HMF** bands remains constant at all **Gain** settings. You can use this to achieve a high Q factor even when applying low attenuation or boost.
- **HMF Q**: Adjusts the Q factor, or resonance, of the **HMF** band. The Q factor determines the width of the frequency band around the filter frequency that is attenuated or boosted.
- **HMF Gain**: Adjusts the amount of attenuation or boost applied to the **HMF** band. The **HMF** band acts as a bell filter that attenuates or boosts frequencies around the filter frequency as set with **Freq**.
- **HMF Freq**: Adjusts the filter frequency of the **HMF** band.
- **HF Gain**: Adjusts the amount of attenuation or boost applied to the **HF** band as set with **Freq**. The **HF** band can either act as a bell or shelf filter (see **HF Bell/Shelf**).
- **HF Freq**: Adjusts the filter frequency of the **HF** band.
- **HF Bell/Shelf**: Selects between two filter types for the **HF** band. When set to shelf (low position), frequencies above the filter frequency as set with **Freq** are evenly attenuated or boosted, depending on the **Gain** setting. When set to bell (high position), frequencies around the filter frequency are attenuated or boosted.

- **Output**: Adjusts the output level of the equalizer.

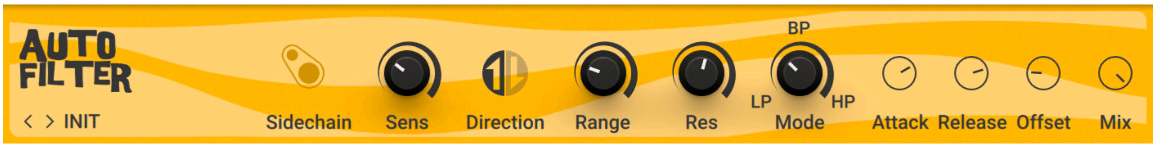
20.7. Filters

Filters shape the frequency content of signals, not unlike equalizers. They are tailored towards creative applications and often have a strong sound character. You can use them to add color, resonant peaks, and distortion to the sound.

Auto Filter

Auto Filter is a multimode filter that responds to the input signal, therefore you can dynamically play it in real-time using your instrument.

This Component contains the following parameters and controls:



- **Sens**: Adjusts the sensitivity of the filter's response to the level of input signal. Turning the control to the right increases the sensitivity, making the filter respond stronger to the articulation of your playing.
- **Direction**: Switches between two different modes that determine the filter's response to the input signal. In Up mode (upwards pointing arrow), the cutoff frequency moves up when the input signal level increases, and down when the input signal level decreases. In Down mode (downwards pointing arrow), the cutoff frequency moves down when the input signal level increases, and up when the input signal level decreases.
- **Range**: Adjusts the range of the cutoff frequency's movement.
- **Res**: Adjusts the resonance amount of the filter. Turning the control to the right makes the frequency content at the cutoff frequency more pronounced.
- **Mode**: Morphs between three different filter modes that determine the basic sound character:
  - **LP**: Resonant low-pass filter In this mode, frequency content above the cutoff frequency is attenuated, creating a darker sound.
  - **BP**: Resonant band-pass filter. In this mode, frequency content above and below the cutoff frequency is attenuated, creating a thinner and more focused sound.
  - **HP**: Resonant high-pass filter. In this mode, frequency content below the cutoff frequency is attenuated, creating a brighter sound.

The Expert panel contains the following parameters and controls:

- **Sidechain**: Activates the sidechain input, allowing you to use an external signal as the filter's control signal.
- **Attack**: Adjusts the time it takes the filter to respond to changes of the input signal level in the range in the range in the range from 5 ms to 80 ms. Increasing the attack time smoothens the response of the filter and makes it less sensitive to small changes of the input signal.
- **Release**: Adjusts the time it takes the filter to return to its initial cutoff frequency in the range from 50 ms to 800 ms. Increasing the release time smoothens the response of the filter and makes it less sensitive to small changes of the input signal.
- **Offset**: Adjusts the basic value of the cutoff frequency, which is the starting point for its movement.

- **Mix**: Blends between the input signal and the effect signal.

Cry Wah

Cry Wah models the sound of the most popular wah-wah pedal of all time. The wah-wah is a classic filter effect used for guitars that moves a resonant filter peak in the frequency spectrum.

This Component contains the following parameters and controls:

- **Wah**: Adjusts the strength of the wah-wah effect.

Doppel-Filter

Doppel-Filter is a flexible and characterful dual stereo filter inspired by both vintage and modern implementations of the Sallen-Key topology. This type of filter can be found in classic monophonic synthesizers from the late 70s as well as contemporary desktop units and Eurorack modules. Its sound can be clean and controlled when set carefully, or dirty and wild when taken to the extremes. Due to the strong character of its resonance, Doppel-Filter can also be used to distort and mangle sounds beyond recognition.

Each of the two filters **A** and **B** has its own set of controls, including common filter parameters like **Cutoff** and **Resonance**. Additionally, both filters offer continuous morphing between three **Modes**, and the **Spread** parameter, which offsets the cutoff frequencies of each filter on the left and right stereo channel. The filter routing can be set to both serial and parallel configurations, enabling you to create complex filter responses and stereo effects.

Doppel-Filter's ability to self-oscillate turns it into an unconventional sound generator. The Safety function in the Expert panel mutes self-oscillation when no input signal is present, making it possible to safely apply high resonance amounts, or control self-oscillation through live playing. The Expert panel also includes **FM** (frequency modulation) controls for each filter, which adjust the amount of audio rate modulation applied to the cutoff frequency from the input signal. Applying FM drastically alters the character of the filter and combines well with self-oscillation. **Drive** adjusts the input level of the filter, which changes the resonance behavior and allows you to dial in the desired amount of distortion for your sound.

This Component contains the following parameters and controls:

The following controls are available for each of the two filters **A** and **B** independently.

- **Cutoff**: Adjusts the cutoff frequency. Depending on the setting of the Mode control, frequency content is attenuated above the cutoff frequency (**LP**), below the cutoff frequency (**HP**), or above and below the cutoff frequency (**BP**).

- **Res**: Adjusts the resonance amount of the filter. Turning the control to the right makes the frequency content at the cutoff frequency more pronounced.
- **Mode**: Morphs between three different filter modes that determine the basic sound character:
  - **LP**: Resonant low-pass filter with a 12 db/Oct response. In this mode, frequency content above the cutoff frequency is attenuated, creating a darker sound.
  - **BP**: Resonant band-pass filter with a 6 db/Oct response. In this mode, frequency content above and below the cutoff frequency is attenuated, creating a thinner and more focused sound.
  - **HP**: Resonant high-pass filter with a 12 db/Oct response. In this mode, frequency content below the cutoff frequency is attenuated, creating a brighter sound.
- **Spread**: Adds an offset to the **Cutoff** parameter on the left and right stereo channel. Turning the control to the left increases the cutoff frequency on the left channel and decreases it on the right channel. Turning the control to the right increases the cutoff frequency on the right channel and decreases it on the left channel. At high resonance amounts it is possible to create two peaks in the frequency spectrum with each of the two filters, with variable distribution in the stereo image.

The following controls act globally and affect both filters at the same time.

- **Filter Routing**: Switches between two different configurations of the filter routing. In serial configuration (downward pointing arrow), the input signal is sent to filter **A**, which sends its output to filter **B**. In parallel configuration (divided, rightwards pointing arrow), the input signal is sent to both filter **A** and filter **B** simultaneously.
- **Mix**: Blends between the output signals of the two filters **A** and **B**.

The Expert panel contains the following parameters and controls:

The following controls are available for each of the two filters **A** and **B** independently.

- **FM**: Adjusts the amount of audio rate modulation applied to the cutoff frequency from the respective filter's input signal. When FM Sidechain is activated, the global sidechain signal is used as modulation source instead.
- **Drive**: Adjusts the input level of the filter, effectively changing the sound character and resonance behavior. When set to high values, the resonance is damped, allowing for more distorted sounds without going into self-oscillation.

The following controls act globally and affect both filters at the same time.

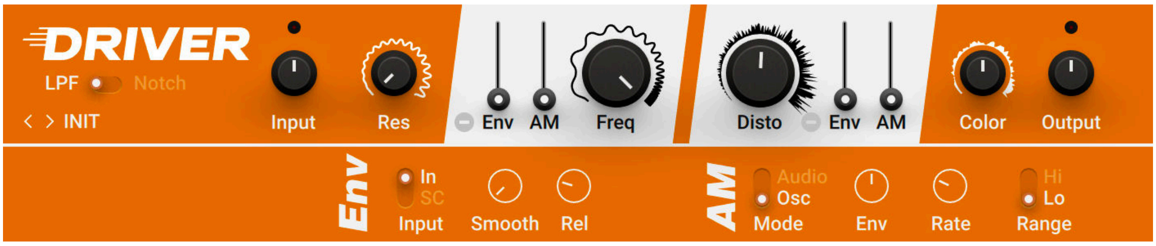
- **FM Sidechain**: Activates the sidechain input for the **FM** controls. When activated, the global sidechain signal is used to apply audio rate modulation to the cutoff frequency.
- **Safety**: Activates Safety mode for both filters **A** and **B**. When activated, the self-oscillation of the filters is muted as soon as no input signal is detected.

DRIVER

DRIVER is a versatile distortion and filter effect that combines a smooth state-variable filter with distortion and creative modulation capabilities. The distortion is integrated in the filter's resonance circuit, resulting in a unique filter response that enables you to explore unusual sounds not possible using common filter and distortion effects. You can use DRIVER to add character and color to a sound, or radically transform a sound's harmonic structure.

Both the **Freq** and **Disto** controls can be modulated using the envelope follower and audio-rate modulator, enabling you to create lively sounds that dynamically respond to the way you play. The envelope follower detects peaks in the input signal and derives an envelope from them. The audio-rate modulator uses either an internal oscillator or the input signal to create sounds similar to FM (Frequency Modulation) synthesis and amplitude modulation.

This Component contains the following parameters and controls:



- **LPF/Notch**: Selects one of two different filter modes that determine the basic sound character:
  - **LPF**: Resonant low-pass filter with a 12 db/Oct response. In this mode, frequency content above the cutoff frequency (**Freq** control) is attenuated, creating a darker sound.
  - **Notch**: Notch filter with a narrow bandwidth. In this mode, frequency content around the center frequency (**Freq** control) is attenuated, creating phasis or vowel-like sounds.
- **Input**: Adjusts the input level of the Component.
- **Res**: Adjusts the resonance amount of the filter. In **LPF** mode, **Res** makes the frequency content at the cutoff frequency more pronounced. In **Notch** mode, **Res** adjusts the width of the attenuated frequency band around the center frequency.
- **Env (Freq)**: Adjusts the amount of modulation applied to **Freq** from the envelope follower.
- **Envelope polarity (Freq)**: Inverts the polarity of the modulation applied to **Freq** from the envelope follower. When inverted, the envelope decreases the cutoff or center frequency, similar to a ducking effect.
- **AM (Freq)**: Adjusts the amount of modulation applied to **Freq** from the audio-rate modulator. Similar to FM (Frequency Modulation) synthesis, this produces additional frequencies in the spectrum, called sidebands. You can use this to create rich harmonics and abrasive timbres.
- **Freq**: Adjusts the cutoff frequency (**LPF** mode) or center frequency (**Notch** mode) of the filter.
- **Disto**: Adjusts the amount of distortion applied in the filter's resonance circuit. The effect of the distortion gets more pronounced as resonance is increased using the **Res** control.
- **Env (Disto)**: Adjusts the amount of modulation applied to **Disto** from the envelope follower.
- **Envelope polarity (Disto)**: Inverts the polarity of the modulation applied to **Disto** from the envelope follower. When inverted, the envelope decreases the amount of distortion, similar to a ducking effect.
- **AM (Disto)**: Adjusts the amount of modulation applied to **Disto** from the audio-rate modulator. Similar to amplitude modulation, this produces additional frequencies in the spectrum, called sidebands. You can use this to create rich harmonics and abrasive timbres.
- **Color**: Adjusts the tone of the distortion. Turning the control to the right changes the tone from dark to bright.
- **Output**: Adjusts the output level of the Component.

The Expert panel contains the following parameters and controls:

- **Input In/SC**: Selects the signal used to trigger the envelope follower that can be applied to **Freq** and **Disto**. When set to **In**, the input signal of the Component is used. When set to **SC**, the global sidechain signal is used.
- **Smooth**: Adjusts the response of the envelope to peaks in the input signal. Turning **Smooth** to the right increases the response time, resulting in a smoother onset of the envelope.
- **Rel**: Adjusts the release of the envelope, which is the time it takes the envelope to return to zero. Turning **Rel** to the right increases the release time, resulting in a longer decay of the modulation produced by the envelope.

- **Mode Audio/Osc**: Selects the signal used to apply audio-rate modulation to **Freq** and **Disto**. When set to **Audio**, the input signal is used. When set to **Osc**, the internal oscillator is used. The frequency of the internal oscillator can be set using the **Rate** control.

The following controls are available in the AM section when **Osc** mode is selected:

- **Env**: Adjust the amount of modulation applied to the frequency of the internal oscillator (**Rate** control) from the envelope follower. At center position, no modulation is applied. Turning the control to the right increases the amount of modulation. Turning the control to the left increases the amount of modulation with inverted polarity.
- **Rate**: Adjusts the frequency of the internal oscillator used for audio-rate modulation.
- **Range Hi/Lo**: Selects the frequency range of the **Rate** control. When **Hi** is selected, the range is approximately 2 Hz to 16 kHz. When **Lo** is selected, the range is approximately 0 Hz to 26 Hz. By selecting **Lo** mode you can use the audio-rate modulator as an LFO (Low Frequency Oscillator).

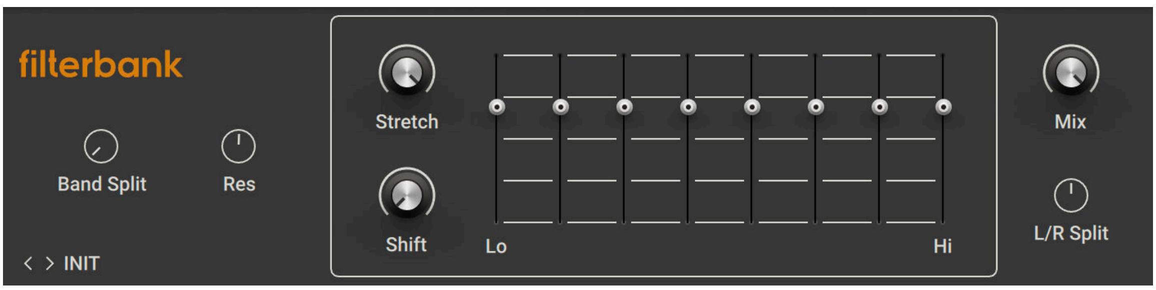
The following controls are available in the AM section when **Audio** mode is selected:

- **Input**: Adjusts the level of the input signal used for audio-rate modulation.
- **Clip Cut**: Adjusts the amount of clipping applied to the input signal used for audio-rate modulation. Turning **Clip Cut** to the right increases the amount of clipping, resulting in a more aggressive sound.
- **Lag**: Adjusts the slew rate of input signal used for audio-rate modulation. The slew rate determines how quickly a signal can change. Turning **Lag** to the right reduces the slew rate, resulting in a smoother sound.

Filterbank

Filterbank is a complex filter that sculpts the sound using eight filter bands, each consisting of a pair of resonant band-pass filters. Global controls enable you to distribute and move the filter bands in the frequency spectrum. Each filter band has a dedicated gain control for fine-tuning the overall response of the filter. The Expert panel includes controls that change the behavior of the pair of band-pass filters in each band, further adding to the vast possibilities of Filterbank.

This Component contains the following parameters and controls:



- **Stretch**: Distributes the individual filter bands in the frequency spectrum. Turning the control to the right increases the spread of the filter bands, covering a wider range at a lower density.
- **Shift**: Adjusts the center frequency around which the individual filter bands are distributed in the frequency spectrum. Turning the control to the right moves the center frequency from low to high frequencies.
- **Filter Bands**: Each fader boosts or cuts frequency content in a specific frequency band. At center position, the frequency band is unaltered. Moving a fader up boosts frequency content. Moving a fader down cuts frequency content.
- **Mix**: Blends between the input signal and the effect signal.

The Expert panel contains the following parameters and controls:

- **Band Split**: Moves the frequencies of the two band-pass filters in each filter band further apart.
- **Res**: Adjusts the resonance amount of the filter bands. Turning the control to the right makes the filter bands sharper and more defined.
- **L/R Split**: Adds an offset to the frequencies of the two band-pass filters in each filter band on the left and right stereo channel. Turning the control to the left increases the frequency on the left channel and decreases it on the right channel. Turning the control to the right increases the frequency on the right channel and decreases it on the left channel.

Pro-Filter

Pro-Filter is based on the filter of Native Instruments' software synthesizer Pro-53. It has a rich and bold sound and can be used both as a character filter and a distinct tone control.

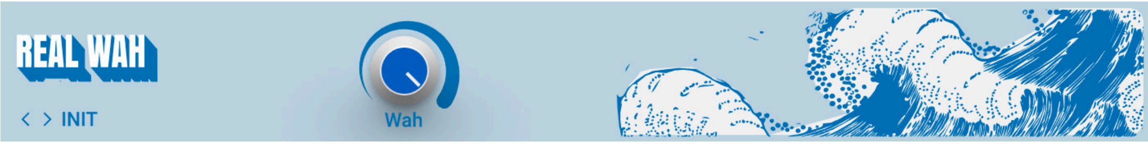
This Component contains the following parameters and controls:

- **Freq**: Adjusts the cutoff frequency. Depending on the setting of the Mode control, frequency content is attenuated above the cutoff frequency (**LP**), below the cutoff frequency (**HP**), or above and below the cutoff frequency (**BP**).
- **Res**: Adjusts the resonance amount of the filter. Turning the control to the right makes the frequency content at the cutoff frequency more pronounced.
- **Slope**: Adjusts the filter response, which determines how strongly frequency content beyond the cutoff frequency is attenuated. When turned fully to the left, the filter has a a 12 db/Oct response. When turned fully to the right, the filter has a a 24 db/Oct response.
- **Mode**: Morphs between three different filter modes that determine the basic sound character:
  - **LP**: Resonant low-pass filter In this mode, frequency content above the cutoff frequency is attenuated, creating a darker sound.
  - **BP**: Resonant band-pass filter. In this mode, frequency content above and below the cutoff frequency is attenuated, creating a thinner and more focused sound.
  - **HP**: Resonant high-pass filter. In this mode, frequency content below the cutoff frequency is attenuated, creating a brighter sound.

Real Wah

Real Wah models the sound of a very successful custom wah-wah pedal from the late 90s. The wah-wah is a classic filter effect used for guitars that moves a resonant filter peak in the frequency spectrum.

This Component contains the following parameters and controls:

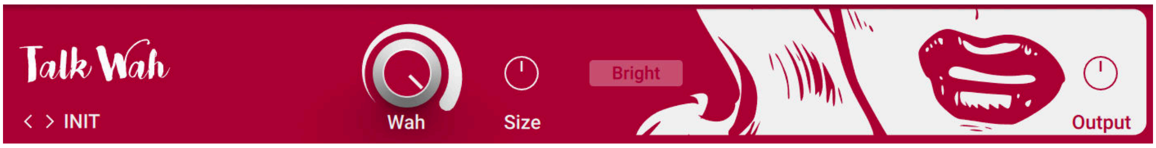


- **Wah**: Adjusts the strength of the wah-wah effect.

Talk Wah

Talk Wah is a special version of the classic wah-wah effect that moves a resonant filter peak in the frequency spectrum. This implementation of the effect mimics the way vowel sounds are produced by the human mouth. The results are similar to the talk box effects of the 70s.

This Component contains the following parameters and controls:



- **Wah**: Adjusts the quality of the wah-wah effect, producing different vowel sounds. When turned fully to the left, the filter produces the vowel sound "o". At center position, the filter produces the vowel sound "a". When turned fully to the right, the filter produces the vowel sound "e".

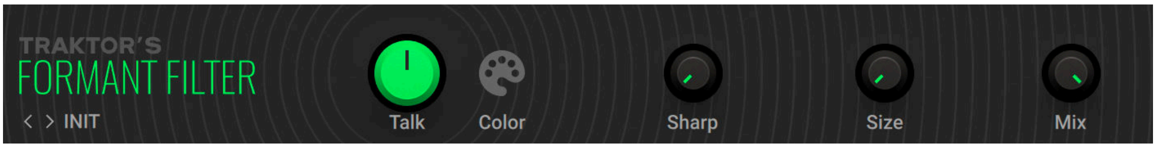
The Expert panel contains the following parameters and controls:

- **Size**: Mimics the effect of differently sized oral cavities, changing the overall character of the sound.
- **Bright**: Boosts high-frequency content.
- **Output**: Adjusts the output level of the Component.

TRAKTOR's Formant Filter

The Formant Filter adds the character of vowel sounds to the input signal. The five vowel sounds, a, e, i, o, u, each feature distinct frequencies in the spectrum.

This Component contains the following parameters and controls:



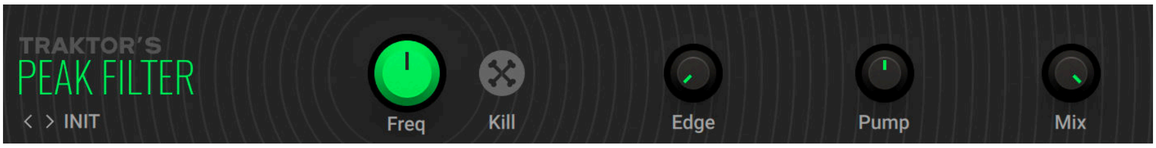
- **Talk**: Morphs between the five vowel sounds (a, e, i, o, u).
- **Color**: Switches between two different characteristics of the vowel sounds.
- **Sharp**: Makes the vowel sounds more sharp and distinct.
- **Size**: Adjusts the size of the modeled oral cavity, shifting the formant frequencies in the vowel sounds.

- **Mix**: Blends between the input signal and the effect signal.

TRAKTOR's Peak Filter

The Peak Filter produces a resonant peak by boosting frequency content at the filter frequency. You can use this effect to make specific frequencies more pronounced.

This Component contains the following parameters and controls:

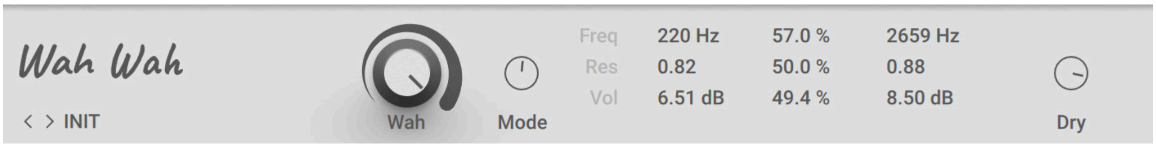


- **Freq**: Adjusts the cutoff frequency. Frequency content at the cutoff frequency is boosted, depending on the amount of resonance as set using **Edge**.
- **Kill**: Inverts the filter curve, creating a notch filter that cuts frequency content at the filter frequency as set using **Freq**.
- **Edge**: Adjusts the amount of resonance. Turning **Edge** to the right boosts frequency content at the cutoff frequency as set using **Freq**.
- **Pump**: Adds brickwall limiting to the signal. You can use this to tame the signal and increase its overall loudness.
- **Mix**: Blends between the input signal and the effect signal.

Wah Wah

Wah Wah is a classic filter effect used for guitars that moves a resonant filter peak in the frequency spectrum. This versatile implementation of the effect offers detailed options to fine-tune the character and movement of the effect.

This Component contains the following parameters and controls:



- **Wah**: Adjusts the strength of the wah-wah effect.

The Expert panel contains the following parameters and controls:

- **Mode**: Morphs between three different filter modes that determine the basic sound character (from left to right):
  - **LP**: Resonant low-pass filter In this mode, frequency content above the cutoff frequency is attenuated, creating a darker sound.
  - **BP**: Resonant band-pass filter. In this mode, frequency content above and below the cutoff frequency is attenuated, creating a thinner and more focused sound. This mode produces the traditional wah-wah effect.
  - **HP**: Resonant high-pass filter. In this mode, frequency content below the cutoff frequency is attenuated, creating a brighter sound.
- **Freq**: Sets the cutoff frequencies of the filter at the minimum position (left value, in **Hz**), center position (middle value, in **%**) and the maximum position (right value, in **Hz**) of the **Wah** control.
- **Res**: Sets the resonance amounts of the filter at the minimum position (left value), center position (middle value, in **%**) and the maximum position (right value) of the **Wah** control.

- **Vol**: Sets the output level of filter at the minimum position (left value, in **dB**), center position (middle value, in **%**) and the maximum position (right value, in **dB**) of the **Wah** control.
- **Dry**: Blends the input signal with the filtered signal. When the control is turned fully to the left, only the filtered signal is sent to the output. Turning the control to the right adds the input signal to the output.

20.8. Legacy

Legacy Components serve the purpose of ensuring backwards compatibility with GUITAR RIG 5.

> Legacy Components are only shown in the Browser if the corresponding option in the Preferences is activated. For more information, see Library.

Cabinets & Mics

Cabinets & Mics is a legacy Component that ensures backwards compatibility with GUITAR RIG 5. It models various cabinets, the type and position of the microphone, and the room response. Except for the **Volume** control and its **Learn** function, all controls are hidden.

> Instead of Cabinets & Mics, you can use the more advanced Components in the Cabinets category. For more information, see Cabinets.

This Component contains the following parameters and controls:

- **Volume**: Adjusts the output level.
- **Learn**: Adjusts **Volume** automatically by analyzing the output signal and determining the optimal output level. For best results, play loudly while the analyzation is in progress.

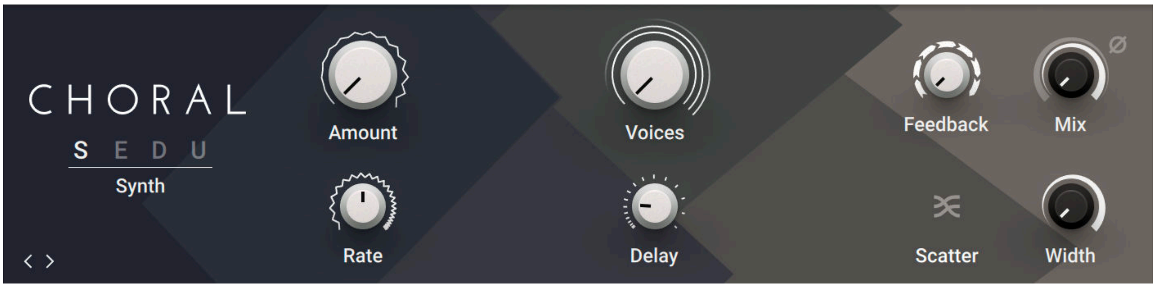
20.9. Modulation

Modulation effects include classics like chorus, phaser, and flanger. Internal modulation changes the sound over time. You can use this to add subtle movement to sounds, or completely transform sounds in creative ways. When syncing the modulation to the tempo of your music, rhythmic effects can be achieved.

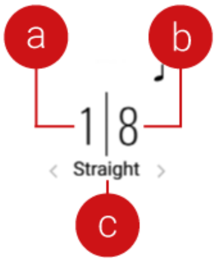
CHORAL

CHORAL is inspired by chorus effects found in synthesizers and studio rack processors from the seventies and early eighties. On these devices, the chorus parameters are hidden. CHORAL gives you enhanced control with parameters that allow you to customize the effect with minimal effort. It features four distinct chorus modes, ranging from the subtle sound of classic studio rack processors to the large ensemble sound of early string synthesizers.

This Component contains the following parameters and controls:



- **Mode**: Switches between four different chorus modes:
  - **Synth**: This mode is inspired by the choruses of polyphonic synthesizers from the late seventies and early eighties. Its sound characteristic is dark and vintage. The modulation behavior is tuned for rich and dispersed sounds.
  - **Ensemble**: This mode is inspired by the choruses of string synthesizers from the seventies. Its sound characteristic is warm and lush. The modulation behavior is tuned for animated and lively sounds.
  - **Dimension**: This mode is inspired by the choruses of studio rack processors from the early eighties. Its sound characteristic is bright and transparent. The modulation behavior is tuned for wide and consistent sounds.
  - **Universal**: This mode is a more generic chorus implementation. Its sound characteristic is clean and modern. The modulation behavior is tuned for a range of sounds from consistent to lively, depending on the number of **Voices**.
- **Amount**: Adjusts the amount of modulation, adding movement to the effect.
- **Rate**: Adjusts the frequency of the modulation. When LFO Sync is enabled, modulation is synchronized to the host and the **Rate** control is replaced by the LFO Sync controls:

  The Numerator (**a**) and Denominator (**b**) set the speed of modulation in musical notes relative to the host tempo. The Numerator sets the number of notes, while the Denominator sets the note value. The Sync Mode (**c**) sets the time value, or subdivision, for the chosen note value.
- **Voices**: Fades from one to three chorus voices. Increasing the number of chorus voices adds a dense and ensemble-like quality to the sound. The modulation affects the second and third chorus voice differently from the first, resulting in a wider and livelier sound.
- **Delay**: Adjusts the delay times of the chorus voices, allowing you to change the spatial depth of the sound. This parameter strongly interacts with **Feedback**.
- **Feedback**: Adjusts the level of the feedback signals from the outputs of the chorus voices to their inputs, creating a more sustained and spacious sound
- **Scatter**: Enables a special feedback routing for the chorus voices that introduces reverb-like behavior.
- **Mix**: Blends between the input signal and the effect signal by means of an equal-power crossfade.
- **Invert**: Changes the sound characteristic of the chorusing effect by inverting the effect signal.

- **Width**: Pans the chorus voices opposite directions, widening the stereo image. When **Width** is set to 0, the input's stereo image is preserved.

Electric Lady

Electric Lady is a versatile flanger that is modeled after a classic effects unit. It produces sounds that range from subtle flanging and chorusing to metallic timbres and extreme jet flanger effects. In Static mode, Electric Lady becomes a complex filter for chime-like tones.

This Component contains the following parameters and controls:

- **Rate**: Adjusts the frequency of the modulation. When Sync is activated, **Rate** is set in musical intervals relative to the tempo of the Metronome.
- **Sync (note symbol)**: Synchronizes **Rate** to the Metronome.
- **Depth**: Adjusts the intensity of the modulation.
- **Color**: Adjusts the overall timbre of the effect.
- **Static**: Activates Static mode, which turns off the modulation. Electric Lady then functions as a filter, the character of which can be adjusted using the **Depth** control.

The Expert panel contains the following parameters and controls:

- **Invert**: Inverts the phase of the effect signal, creating an alternative timbre.
- **Rotate**: Offsets the modulation between the left and right stereo channel, creating a wide stereo effect.
- **Mix**: Blends between the input signal and the effect signal.

Ensemble

Ensemble is a chorus effect that is modeled after a classic effects unit. It adds slightly detuned voices to the signal, creating the illusion of an ensemble. Additionally, it can be used as a vibrato effect.

This Component contains the following parameters and controls:

- **Chorus/Vibrato**: Switches between **Chorus** and **Vibrato** mode.
- **Intensity**: Adjusts the strength of the effect in **Chorus** mode.
- **Rate**: Adjusts the frequency of the modulation in **Vibrato** mode. When Sync is activated, **Rate** is set in musical intervals relative to the tempo of the Metronome.
- **Sync (note symbol)**: Synchronizes **Rate** to the Metronome.
- **Depth**: Adjusts the intensity of the modulation in **Vibrato** mode.
- **Volume**: Adjusts the output level of the Component.

The Expert panel contains the following parameters and controls:

- **Delay**: Adjusts the delay times of the chorus voices in **Chorus** mode.
- **Stereo**: Adjusts the stereo width of the chorus voices in **Chorus** mode. When turned fully to the left, the chorus voices operate in mono. Turning the control to the right distributes the chorus voices on the left and right stereo channel.
- **Mix**: Blends between the input signal and the effect signal.
- **Bass**: Adjusts the low-frequency response.
- **Mid**: Adjusts the mid-frequency response.
- **Treble**: Adjusts the high-frequency response.

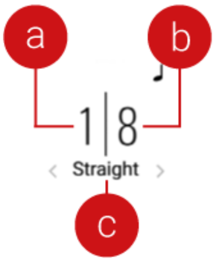
FLAIR

FLAIR is a new take on the concept of the flanger effect with additional features that have been carefully chosen to allow for more sophisticated and extreme sounds than possible with common flangers, while staying true to the ease of use and clarity associated with these devices. FLAIR features three flanger modes that offer different approaches to a range of effects from flanging to harmonization.

This Component contains the following parameters and controls:

- **Mode**: Switches between three different flanger modes:
  - **Standard**: In this mode, each flanger voice behaves like a basic flanger effect, creating harmonically related peaks and notches in the frequency spectrum.
  - **Thru Zero**: In this mode, each flanger voice is duplicated. The duplicated instances of the flanger voices are excluded from the modulation and thus rest at their respective base pitch. When modulation is introduced by increasing **Amount**, the flanger voices shift against the duplicated instances in time. This creates the strong thru zero flanging effect with its characteristic signal cancellation, similar to the flanging effect originally created with two tape machines. The **Offset** slider below the Mode menu allows you to shift the duplicated instances of the flanger voices in the frequency spectrum. This changes their position relative to the center of modulation, which results in rhythmical variations of the thru zero flanging effect. **Offset** also allows you to reduce the amount of signal cancellation when there is no modulation (**Amount** set to 0%).
  - **Scan**: In this mode, instead of adding the flanger voices to form a chord, **Voices** scans through them one after the other. This is similar to how an arpeggiator on a keyboard plays the notes contained in a chord as a sequence.The Scan Mode selector below the Mode menu allows you to choose from three different waveforms for the modulation: Triangle, Sawtooth Up, and Sawtooth Down.
- **Amount**: Adjusts the amount of modulation, adding movement to the effect.

- **Rate**: Adjusts the frequency of the modulation. When LFO Sync is enabled, modulation is synchronized to the host and the **Rate** control is replaced by the LFO Sync controls:

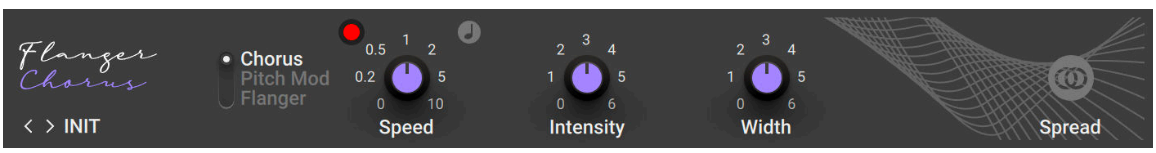


  The Numerator (**a**) and Denominator (**b**) set the speed of modulation in musical notes relative to the host tempo. The Numerator sets the number of notes, while the Denominator sets the note value. The Sync Mode (**c**) sets the time value, or subdivision, for the chosen note value.
- **Pitch**: Adjusts the fundamental frequency of the first flanger voice in semitones, effectively shifting the peaks and notches of all flanger voices in the frequency spectrum.
- **Chord**: Adjusts the level of the feedback signals from the outputs of the flanger voices to their inputs, creating a more resonant and metallic sound.
- **Voices**: When **Mode** is set to **Standard** or **Thru Zero**, **Voices** fades from one to four flanger voices. The additional flanger voices are added in harmonic intervals, forming a chord as set with **Chord**. When **Mode** is set to **Scan**, **Voices** scans through the four flanger voices one after the other by blending between the first and the second flanger voice, then the second and the third flanger voice, and so on.
- **Detune**: Alters the pitch of each individual flanger voice in a range of approximately +/- 60 cent. This creates a rich and lively sound similar to the effect of detuning oscillators on a synthesizer. **Detune** is especially useful when Chord is set to **Unison**.
- **Feedback**: Adjusts the level of the feedback signals from the outputs of the flanger voices to their inputs, creating a more resonant and metallic sound.
- **Damping**: Attenuates the high frequency content of the feedback signals from the outputs of the flanger voices to their inputs, allowing for soft sounds even at high **Feedback** settings.
- **Mix**: Blends between the input signal and the effect signal by means of an equal-power crossfade.
- **Invert**: Swaps the position of the peaks and notches in the frequency spectrum by inverting the effect signal. When enabled, the perceived pitch of the flanger voices is one octave lower. In **Thru Zero** mode, enabling Invert creates strong signal cancellations that can result in interesting rhythmical effects.
- **Width**: Duplicates the flanger voices internally, pans them in opposite directions, and adds a phase offset to the modulation between the left and right stereo channels. Additionally, a special type of cross-feedback is introduced, further animating the stereo image as **Feedback** is increased.

Flanger Chorus

Flanger Chorus is a delay-based Component offering dedicated modes for chorus, pitch modulation, and flanger.

This Component contains the following parameters and controls:



- **Chorus / Pitch Mod / Flanger**: Switches between the chorus, pitch modulation, and flanger modes.
- **Speed**: Adjusts the frequency of the modulation. When Sync is activated, **Speed** is set in musical intervals relative to the tempo of the Metronome.
- **Sync (note symbol)**: Synchronizes **Speed** to the Metronome.
- **Intensity**: Adjusts the intensity of the modulation.
- **Width**: Adjusts the modulation range of the effect.
- **Spread**: Produces an extra wide stereo effect.

FREAK

FREAK combines three amplitude modulation techniques: basic amplitude modulation, ring modulation, and frequency shifting. It is based on a model of an analog diode ring circuit that produces rich harmonic overtones and textures. In addition to a wide range of harmonic transformations, its three FX Modes facilitate special applications like AM radio simulation (**Radio** mode), tremolo and distortion (**Oscillator** mode), as well as gating (**Sidechain** mode).

This Component contains the following parameters and controls:

- **Mode**: Switches between three different modes of operation:
  - **Radio**: Emulates the behavior of so-called demodulation circuits in old AM radios, allowing you to create the effect of dialing in a specific radio station. The emulation complements the amplitude modulation techniques controlled using the **Type** control. This modes uses a sine wave signal as the modulation source.
  - **Oscillator**: Provides the pure sound of the amplitude modulation techniques controlled using the **Type** control. This modes uses a sine wave signal as the modulation source.
  - **Sidechain**: Opens up the amplitude modulation techniques available via the **Type** control for experimentation by modulating the input signal with itself or with an external sidechain signal. Additionally, the modulation signal can be processed with an envelope follower that smoothes out the signal contour.

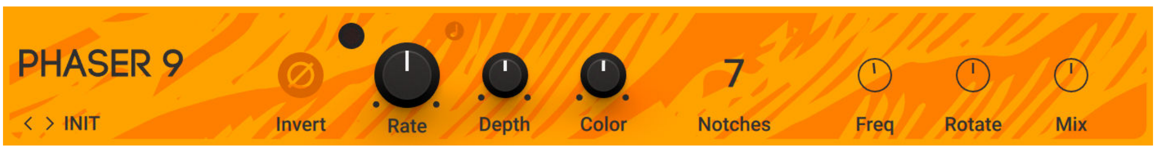
- **Mode control 1**: This control is specific to the selected Mode:
    - **Tuning** (Radio mode): Emulates the effect of tuning an old AM radio. In center position, the best possible tuning is achieved. As you move the control away from center position, the amount of radio interference increases. The extra button at the top right of the control toggles between two different emulations. When activated, a product demodulation circuit is used, producing an aggressive sound. When deactivated, an envelope demodulation circuit is used, recreating the sound of an old AM radio.
    - **Freq** (Oscillator mode): Adjusts the rate of the internal sine wave modulation used by the amplitude modulation techniques controlled using the **Type** control. When **Type** is set to **AMP** and **RING**, this allows you to change the frequency of the sidebands created in the frequency spectrum. When **Type** is set to **FREQ**, the modulation rate equals the amount by which the input signal's frequency content is shifted in the frequency spectrum. The extra button at the top right of the control sets the range of the **Freq** control. When activated, **Freq** has a coarse range of -5000 Hz to +5000 Hz. When deactivated, **Freq** has a fine range of -200 Hz to +200 Hz.
    - **Contour** (Sidechain mode): Blends between the direct signal from the modulation source and the signal processed by the envelope follower. This allows you to adjust how much the envelope follower affects the contour of the modulation signal. The extra button at the top right of the control switches sidechain operation on or off. When activated, the external sidechain input is used. When deactivated, the input signal is used as the modulation source, allowing for self-modulation.
- **Mode control 2**: This control is specific to the selected Mode:
    - **Width** (Radio mode): Adjusts the amount of filtering applied to the signal. Low settings result in a filtered and slightly resonant sound. High settings reduce the filtering effect while adding more noise to the signal.
    - **Stereo** (Oscillator mode): Creates a wide stereo image by adding a phase offset to the modulation applied to the left and right stereo channels.
    - **Release** (Sidechain mode): Adjusts the attack and release times of the envelope follower. At low settings, the envelope follower quickly adapts to the modulation signal's contour. At high settings, it responds slowly and smoothes out the modulation signal's contour.
- **Mode control 3**: This control is specific to the selected Mode:
    - **Carrier** (Radio mode): Adjusts the frequency of the carrier used in the demodulation circuit, controlling the quality of the radio transmission. The extra button at the top right of the control toggles the noise gate on or off. When activated, noise from the demodulation circuit is only passed through if an input signal is present. When deactivated, the noise is constantly passed through, allowing you to use FREAK as a flexible noise source. The amount of noise can be adjusted with the **Feedback** control.
    - **Antifold** (Oscillator mode): Reduces the amount of sidebands folding over 0 Hz, producing a cleaner sound in the low-frequency spectrum. By increasing **Antifold**, thinner sounding distortion effects with a less tonal quality can be achieved.
    - **BP Freq** (Sidechain mode): Adjusts the cutoff frequency of the band-pass filter applied to the modulation signal, making the envelope follower respond to a specific frequency band of the modulation source. The extra button at the top right of the control switches the band-pass filter applied on or off.

- **Type**: Morphs between three different amplitude modulation techniques: **AMP** (basic amplitude modulation, **RING** (ring modulation), and **FREQ** (frequency shifting):
  - **AMP**: Basic amplitude modulation. At modulation rates below the audio spectrum (< 20 Hz), the level of the input signal, or carrier signal, changes slowly. This can be used for tremolo effects. At modulation rates in the audio spectrum (> 20 Hz), new frequencies are added to the carrier signal, called sidebands. The sidebands are the sum and the difference of the frequencies contained in the carrier and modulation signals. This way you can add inharmonic content to the sound while maintaining its basic character.
  - **RING**: Ring modulation, also called balanced modulation. At modulation rates below the audio spectrum (< 20 Hz), the level of the input signal, or carrier signal, changes slowly while also periodically inverting its phase. This can be used for subtle phasing and distortion effects. At modulation rates in the audio spectrum (> 20 Hz), the carrier signal's frequency content is replaced by new frequencies, called sidebands. The sidebands are the sum and the difference of the frequencies contained in the modulation signal and the input signal. This breaks up the harmonic structure of the sound and gives it a metallic sounding character.
  - **FREQ**: Frequency shifting. This complex amplitude modulation technique shifts the input signal's frequency content in the frequency spectrum by an amount that equals the modulation rate. For example, when setting a modulation rate of 100 Hz with the **Freq** control in Oscillator mode, all frequencies contained in the input signal will be shifted up in the frequency spectrum by 100 Hz. This breaks up the harmonic structure of the sound and gives it a metallic sounding yet distinctly tonal character.
- **Harmonics**: Adjusts the amount of harmonic overtones produced by the model of an analog diode ring circuit used to implement the different amplitude modulation techniques in FREAK.
- **Mix**: Blends between the input signal and the effect signal by means of an equal-power crossfade.
- **Feedback**: Adjusts the level of the feedback signal from the effect's output to its input. This increases the amount of frequencies, or sidebands, created by the amplitude modulation. When the FX Mode is set to **Radio**, **Feedback** additionally controls the amount of noise added to the signal.

Phaser 9

Phaser 9 is a phaser effect that is modeled after a classic effects unit. It animates the timbre and adds a shimmering quality to the sound.

This Component contains the following parameters and controls:



- **Rate**: Adjusts the frequency of the modulation. When Sync is activated, **Rate** is set in musical intervals relative to the tempo of the Metronome.
- **Sync (note symbol)**: Synchronizes **Rate** to the Metronome.
- **Depth**: Adjusts the intensity of the modulation.
- **Color**: Adjusts the strength of the phasing effect by increasing the internal feedback level.

The Expert panel contains the following parameters and controls:

- **Invert**: Inverts the phase of the effect signal, creating an alternative timbre.

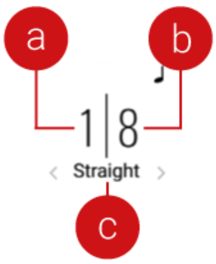
- **Freq**: Adjusts the center frequency of the phasing effect.
- **Notches**: Sets the number of notches in the frequency spectrum produced by the phasing effect.
- **Rotate**: Creates a wide and lively stereo image by adding a phase offset to the modulation between the left and right stereo channels.
- **Mix**: Blends between the input signal and the effect signal.

PHASIS

PHASIS is a new take on the concept of the phaser effect with additional features that have been carefully chosen to allow for more sophisticated and extreme sounds than possible with common phasers, while staying true to the ease of use and clarity associated with these devices. PHASIS features a scalable amount of all-pass filters, producing up to twelve pairs of peaks and notches in the frequency spectrum.

This Component contains the following parameters and controls:

- **Ultra**: Extends the range of **Rate** and **Center**, allowing for more extreme modulation up audio frequencies. This can add new harmonic content to the input signal, similar to the sounds possible with FM synthesis.
- **Amount**: Adjusts the amount of modulation, adding movement to the effect.
- **Rate**: Adjusts the frequency of the modulation. When LFO Sync is enabled, modulation is synchronized to the host and the **Rate** control is replaced by the LFO Sync controls:

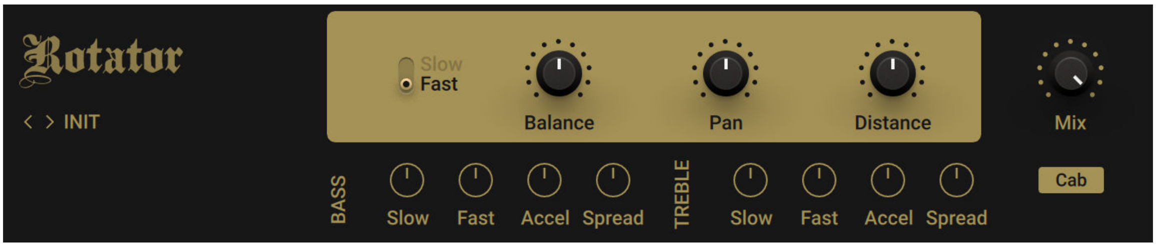
  The Numerator (**a**) and Denominator (**b**) set the speed of modulation in musical notes relative to the host tempo. The Numerator sets the number of notes, while the Denominator sets the note value. The Sync Mode (**c**) sets the time value, or subdivision, for the chosen note value.
- **LFO Sync**: Synchronizes the modulation to the host tempo and replaces **Rate** with the LFO Sync controls.
- **Mod Mix**: Distributes the modulation between **Center** and **Spread**. Moving the slider to the left increases the amount of modulation applied to **Center**, moving the slider to the right increases the amount of modulation applied to **Spread**.
- **Spread Modulation Polarity**: Inverts the polarity of the modulation applied to **Spread**, hence reversing its effect in relation to the modulation applied to **Center**.

- **Spread**: Adjusts the density of the peaks and notches in the frequency spectrum. Turning the knob to the left moves the peaks and notches closer to each other. Turning the knob to the right moves the peaks and notches further apart from each other.
- **Center**: Shifts the peaks and notches in the frequency spectrum by changing the frequencies of the all-pass filters that create the phasing effect (relative to the **Center** frequency).
- **Feedback**: Adjusts the amount of feedback, or resonance, applied to the all-pass filters that create the phasing effect. Turning up **Feedback** makes the peaks and notches in the frequency spectrum more pronounced.
- **Notches**: Sets the number of peaks and notches in the frequency spectrum.
- **Mix**: Blends between the input signal and the effect signal by means of an equal-power crossfade.
- **Invert**: Swaps the position of the peaks and notches in the frequency spectrum by inverting the effect signal.
- **Stereo**: Creates a wide and lively stereo image by adding a phase offset to the modulation between the left and right stereo channels.

Rotator

Rotator simulates the classic Leslie effect produced by two rotating speakers in special cabinets for electric organs. It can be used for unique tremolo and doppler effects.

This Component contains the following parameters and controls:



- **Slow/Fast**: Switches between two speeds for the rotating speakers.
- **Balance**: Blends between the bass and the treble speakers. Turning the control to the right produces a brighter sound, while turning the control to the left accentuates the bass frequencies.
- **Pan**: Adjusts the distribution of the bass and the treble speakers in the stereo image.
- **Distance**: Adjusts the distance between the virtual microphones and the rotating speakers. The Leslie effect is more pronounced at short distances.
- **Mix**: Blends between the input signal and the effect signal.

The Expert panel contains the following parameters and controls:

- **Cab**: Deactivates the internal cabinet emulation to allow for custom configurations in the Rack. You can insert any of the Components from the Cabinets category before Rotator to find a sound you like.

The following controls are individually available for both the **BASS** and the **TREBLE** speaker:

- **Slow**: Adjusts the speed of the respective rotating speaker when the **Slow**/**Fast** switch is set to **Slow**.
- **Fast**: Adjusts the speed of the respective rotating speaker when the **Slow**/**Fast** switch is set to **Fast**.
- **Accel**: Adjusts the acceleration of the respective rotating speaker when switching between the two settings of the **Slow**/**Fast** switch.

- **Spread**: Adjusts the width of the stereo image by changing the distance of the virtual microphones.

Stereo Tune

Stereo Tune is a widening effect. It produces a wide and lively stereo image.

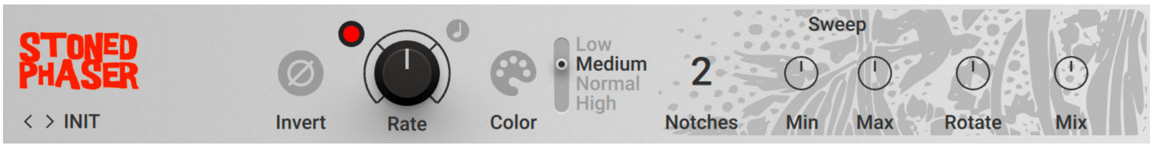
This Component contains the following parameters and controls:

- **Spread**: Adjusts the stereo width from mono to full stereo.
- **Drift**: Adjusts stereo detuning of the input signal. The left and right stereo channels are tuned at different frequencies to create a wide stereo image.
- **Split**: Adjusts the crossover frequency of the effect. Frequency content below this frequency remains unaltered.
- **Mix**: Blends between the input signal and the effect signal.

Stoned Phaser

Stoned Phaser is a phaser effect that is modeled after a popular phaser unit from the 1970s. It adds a swirling quality to the sound that is known from Psychedelic Rock music.

This Component contains the following parameters and controls:



- **Rate**: Adjusts the frequency of the modulation. When Sync is activated, **Rate** is set in musical intervals relative to the tempo of the Metronome.
- **Sync (note symbol)**: Synchronizes **Rate** to the Metronome.
- **Color**: Activates Color mode, which changes the sound of the phasing effect by adjusting the internal feedback level.

The Expert panel contains the following parameters and controls:

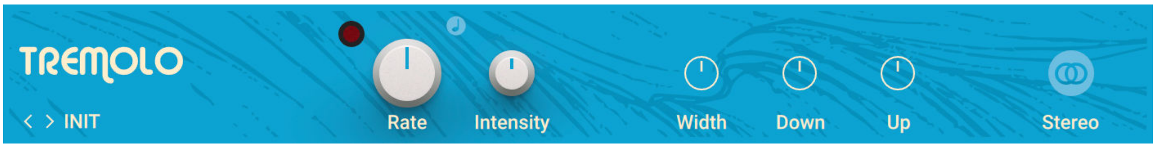
- **Invert**: Inverts the phase of the effect signal, creating an alternative timbre.
- **Low/Medium/Normal/High**: Switches between four strength settings in **Color** mode. Each of the settings corresponds to a different internal feedback level.
- **Notches**: Sets the number of notches in the frequency spectrum produced by the phasing effect.
- **Min**: Adjusts the lower limit of the modulation's frequency range, effectively changing the depth of modulation.
- **May**: Adjusts the upper limit of the modulation's frequency range, effectively changing the depth of modulation.
- **Rotate**: Creates a wide and lively stereo image by adding a phase offset to the modulation between the left and right stereo channels.

- **Mix**: Blends between the input signal and the effect signal.

Tremolo

Tremolo is an amplitude modulation effect, known as tremolo in musical terms. It can add a pulsing movement to the sound and create swirling panning effects.

This Component contains the following parameters and controls:



- **Intensity**: Adjusts the intensity of the modulation.
- **Rate**: Adjusts the frequency of the modulation. When Sync is activated, **Rate** is set in musical intervals relative to the tempo of the Metronome.
- **Sync (note symbol)**: Synchronizes **Rate** to the Metronome.
- **Stereo**: Creates a panning effect by adding a phase offset to the modulation between the left and right stereo channels.

The Expert panel contains the following parameters and controls:

- **WIDTH** controls the ratio between the phases of high and low amplitude. Turning it up increases the “gaps” between the times the signal is at full amplitude. In stereo mode, turning this knob to the left decreases the time the signal is panned to the left and vice versa. Turning it to the right does the same in the opposite direction.
- **Width**: Adjusts the symmetry of the modulation signal, resulting in rhythmic variations of the tremolo effect. When **Stereo** is activated, **Width** adjusts the stereo balance of the panning effect.
- **DOWN** controls the decay time of the tremolo, i.e. the time it takes to go from the highest to the lowest volume level.
- **Down**: Adjusts the fall time of the modulation signal, which is the time it takes the modulation to go from the highest level to the lowest level. Long fall times result in a smooth tremolo effect, while short fall times result in a stuttering effect.
- **Up**: Adjusts the rise time of the modulation signal, which is the time it takes the modulation to go from the lowest level to the highest level. Long rise times result in a smooth tremolo effect, while short rise times result in a stuttering effect.

20.10. Pitch

Pitch effects alter the pitch and harmonic content of signals. You can use them to add harmonization and virtual voices to your playing, or completely transform the sound in creative ways.

Harmonic Synthesizer

The Harmonic Synthesizer opens up a whole world of synthesizer sounds, controlled simply by playing your guitar. It can produce faithful re-creations of classic sounds but is also well suited for generating effects never heard before.

The effect adds three synthetic voices to the dry signal- sub-octave, octave and square wave- that can be freely mixed. An envelope-controlled filter is also included.

Mind that this component tracks the instrument´s pitch best when playing single notes.

Controls

- **GUITAR/BASS** is used for switching the filter sweep range to be fed by a guitar or a bass.
- **THRESHOLD** acts as a noise gate for the input signal. Only signals above the threshold will get through and trigger the volume envelope and the filter sweep. Dial in a setting according to your individual instrument and playing style.
- The **TRIGGER** fader controls the sensitivity of the filter's trigger, reducing the signal volume needed to activate it. If you set it too high it may result in a stuttering filter due to multiple triggering. Experiment a bit to find the setting that suits your needs.
- **SUB** controls the volume of the sub-octave added by the synthesizer. Only works with single notes.
- **DRY** controls the volume of the original guitar signal.
- **OCT** controls the volume of the upper octave added by the synthesizer.
- **SQR** mixes in a distorted or square-wave version of the original signal that can be controlled by dynamic playing.
- **ATT** shapes the attack of the synthesizer envelope. The higher it is set, the slower the signal will be faded in, attenuating the attack sound of the instrument.
- **RES** controls resonance and width of the filter. With higher values you get a narrower frequency range and a more pronounced peak around the filtered frequencies.
- **STRT** sets the frequency at which the filter sweep starts.
- **STOP** sets the destination and resting frequency of the filter sweep. If **Start** and **Stop** are set to the same value, the filter will emphasize that particular frequency without sweeping.
- **RATE** determines the speed of the filter sweep from the start frequency to the stop frequency.

Oktaver

This component adds two signals to the original pitch that are one and two octaves below. Please mind:

The Oktaver works well only with single notes, not with chords. Insert the Oktaver near the beginning of your rack's signal chain. Do not precede it with reverb, delay, or other modulation effects, as these will confuse its pitch tracking. However, it usually works well after compressors and EQs.

Controls

- **DIRECT** sets the level of the dry signal.
- **OCT 1** sets the level of the signal one octave below.
- **OCT 2** sets the level of the signal two octaves below.

Expert Controls

- **CUTOFF** changes the timbre for **OCT1** and **OCT 2** separately. Turn up to increase brightness.
- **RESO** sets the filter resonance for **OCT1** and **OCT 2** separately.
- **STEREO** activates true stereo processing for this module.

Pitch Pedal

The Pitch Pedal basically has the same effect as a guitar's vibrato tailpiece, except that all the strings stay in tune as you bend up and down. Controlling the pitch shift with a controller pedal is highly recommended for hands-free control over pitch changes.

The expert mode offers many controls to tweak the effect for your particular bending needs. If you don't want to get involved in these, simply choose one of the component presets for common string-bending effects.

Controls

- **DRAG** changes pitch within the range set in the expert controls.
- **DRY/WET** controls the blend of dry and processed sound.

Expert Controls

- **MIN SHIFT** sets amount and direction of the pitch shift when the slider is set to the extreme left. The range is ±24 semitones.
- **MIN DETUNE** allows fine-tuning the pitch shift for the left position of the slider. The range is ±100 cents.
- **MAX SHIFT** sets amount and direction of the pitch shift when the slider is set to the extreme right. The range is ±24 semitones.
- **MAX DETUNE** allows fine-tuning the pitch shift for the right position of the slider. The range is ±100 cents.
- **FEEDBACK** determines the amount of the output signal to be looped back to the input, offering interesting effects. If the Pitch Pedal is set to transpose the signal +1 semitone, that signal is looped to be transposed another semitone, and so on, producing an ascending series of tones.
- **DELAY** controls the amount of delay in the feedback path, from 10 to 50 ms. The longer the delay, the more it creates a discreet series of notes; with shorter delays the result is a smooth reverberation.

Resochord

A new Pitch component for Guitar Rig 5 is the RESOCHORD. The Resochord is a bank of 6 comb filters, each of which is individually tuned according to the selected chord. The results are most effective with non-melodic content (like drums) as the Resochord will print its own harmonic content on to any input material.

Controls

The **MIX** controls the amount of signal being affected. The **MODE** section has three controls and one switch.

- The switch toggles between **CHORD** and **STRING**. In **CHORD** mode, three controls are used:
  - **CHORD** determines the chord overlay of the processed signal.
  - **STYLE** sets the chord type from major, minor, alt, meta and frank.
  - **KEY** allows you to transpose the Resochord in semitones.
- In **STRING** mode, only **SPREAD** and **KEY** controls are used.
  - **SPREAD** sets the range of frequency affected.
  - **KEY** allows you to transpose the Resochord in semitones.

- In **CHORD** mode, the 6 combs are tuned according to various chords. In **STRING** mode, the 6 combs are centered around one frequency and can be spread for an intense chorus-like effect.
- The **DECAY** control determines the time the effect is held after the original signal.

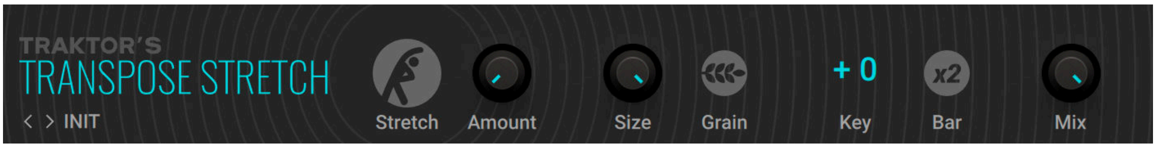
Expert Controls

- **BRIGHTNESS** allows for more high frequency content in the effect.
- **FEEDBACK** intensifies the effect, adding more resonance.
- **INPUT GAIN** increases input level to the effect.
- **MUTE** mutes the effect.

TRAKTOR's Transpose Stretch

Transpose Stretch captures a loop from the incoming audio and manipulates it by means of granular pitch-shifting and time-stretching. You can use it to radically transform the sound.

This Component contains the following parameters and controls:



- **Stretch**: Starts the effect by recording the input signal into the internal buffer. When deactivated, the input signal is remains unaltered.
- **Amount**: Adjusts the amount of time-stretching. Turning **Amount** to the right slows the audio from the internal buffer down until it freezes using a single grain.
- **Size**: Adjusts the size of the grains used to process audio from the internal buffer. When turned fully to the left, large grains of 333 ms length are produced. When turned fully to the right, small grains of 5 ms length are produced. The **Size** control only has an effect when **Grain** is activated.
- **Grain**: Activates grain size adjustment using the **Size** control. When deactivated, the grain size is automatically optimized for best pitch-shifting results
- **Key**: Adjusts the pitch of the grains used to process audio from the internal buffer in semitones. When turned fully to the left, the grains are pitched down by 60 semitones, or 5 octaves. In center position, the original pitch is maintained. When turned fully to the right, the grains are pitched up by 12 semitones, or 1 octave.
- **Bar x2**: Extends the audio used from the buffer to two bars. Otherwise only the first bar of audio is used.
- **Mix**: Blends between the input signal and the effect signal.

20.11. Reverb

Reverb effects simulate the properties of acoustic spaces, both natural and imaginary. You can use them to achieve a wide range of spatial effects, from adding ambience and depth to completely washing out the sound.

Iceverb

The Iceverb is a very colorful reverb that can sound like you're playing in a giant icy cave- or in an igloo! Seriously, it offers a wide range of reverb characteristics and a filter that can even be used like a very special wah-wah effect when controlled with a foot pedal.

Controls

- **DRY/WET** sets the amount of the signal being fed into the reverb section, controlling the intensity of the effect.
- **SIZE** controls the duration of the reverberation, which creates a varying perception of the room size.
- **COLOUR** sets the frequency range that is emphasized in the filter preceding the reverberation. This control resembles a wah-wah, as it allows sliding a strong frequency peak up and down the spectrum.
- **ICE** controls the intensity of the filter by setting the resonance of the filtered frequency band.
- **FREEZE** completely shuts off the dry signal and simultaneously increases the volume of the reverberation. This function can be triggered to create impressive stops that are followed by a majestic, fading reverb sound.
- **MUTE** shuts off the signal going through the reverb section, but lets only the dry signals pass through. Sounds currently being processed will continue ringing out even after the button is pressed. If the **DRY/WET** control is turned fully up, you will hear no more sound, because no signal is allowed into the dry section.

Little Reflektor

Another new reverb in Guitar RIG 5 is LITTLEREFLEKTOR. This versatile reverb is simple to use and can go from subtle to extreme.

Controls

- **DRY/WET** controls the amount of signal being affected by the reverb.
- In the control window there are 8 buttons for **SHORT**, **MED**, **LONG**, **XXL**. These describe the length of the reverb signal.
- **A** and **B** are different styles of reverb and reflections.
- **DECAY** sets the time for the reverb to trail off. Further clockwise makes for a longer reverb.
- **LOW CUT** is high-pass filter to remove bass frequencies that can make a reverb sound muddy.
- **MUTE** shuts off the signal going through the reverb section, but lets only the dry signals pass through. Sounds currently being processed will continue ringing out even after the button is pressed. If the **DRY/WET** control is turned fully up, you will hear no more sound, because no signal is allowed into the dry section.

Octaverb

The Octaverb is a powerful stereo reverb, precisely emulating the early acoustic reflections of eight different rooms. Besides the usual reverb parameters such as room size, it also offers some unique features that can be used both for subtle tone shaping and for extreme effects.

Controls

- **DRY/WET** sets the amount of the signal being fed into the reverb section, controlling the intensity of the effect.
- **ER MODE** selects the room shape, which determines the character of the reverb by emulating different patterns of early acoustic reflections. These are perceived as more or less distinct echoes, before their further reflection in the room creates a diffuse mix. This control offers realistic presets such as "Concrete Room" and completely virtual ones like "Strange Localization"- just try them out and play with the Size parameter to see what they are about.
- **SIZE** sets the amount and duration of diffuse reverberation following the early reflections. This setting mainly influences the perception of room size.
- **HI-DAMP** controls how much high frequencies are attenuated in the process of reverberation.
- **BASSTRAP** controls to what extent low frequencies get "caught" in the reverberation. Turning it up gives the reverb a thicker bottom end.
- **MUTE** shuts off the signal going through the reverb section, but lets only the dry signals pass through. Sounds currently being processed will be ringing out even after the button is pressed. If the **DRY/WET** control is turned fully up, you will hear no more sound, because no signal is allowed into the dry section.

Expert Controls

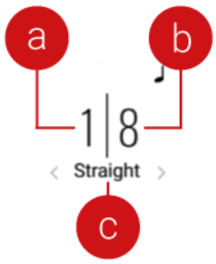
- **EQ** is an additional tone control for the reverberated signal, mainly useful for boosting or attenuating higher frequencies.
- **Wetlevel** controls the level of the processed signal, allowing changing the mix while preserving the volume of the dry signal.
- **Source** controls the position of the dry signal in the stereo panorama. Turning it clockwise brings it to the right channel, turning it counterclockwise brings it to the left channel.
- **Width** adjusts the stereo panorama of the processed signal: When turned clockwise, the effect is fully distributed across both channels. When centered, the processing is mono. When turned fully down, the channels are inverted, meaning that the left part of the reverb signal is routed to the right output channel and vice versa.
- **Freeze** completely shuts off the dry signal and simultaneously increases the volume of the reverberation. This function can be triggered to create impressive stops that are followed by a majestic, fading reverb sound.

RAUM

RAUM features three reverb algorithms that cover vast range of sounds, from tight ambiences to otherworldly soundscapes. It expands on the traditional notion of a reverb effect by adding predelay feedback, the Freeze function, and unusual modulation capabilities. RAUM is designed to respond well to realtime control, whether it be using Macros in a performance situation or when modulated using the Modifiers.

This Component contains the following parameters and controls:

- **Reverb algorithm**: Switches between three different reverb algorithms:
    - **Grounded**: This algorithm is based on the paradigm of a room with strong early reflections and a dense reverb tail. The **Size** ranges from tiny ambiences to large chambers.
    - **Airy**: This algorithm is based on the paradigm of a hall with naturally dispersed early reflections and a smooth reverb tail. The **Size** ranges from medium-sized chambers to large cathedrals and beyond.
    - **Cosmic**: This algorithm is based on a classic digital reverb paradigm and extends RAUM's range of sounds to abstract spaces and ambient soundscapes. Since the algorithm is placed inside the predelay feedback path, it interacts more closely with all of the parameters related to **Predelay**.
- **Freeze**: Holds the reverb's sound content for as long as the function is activated. The Freeze function is available for the Grounded and Airy algorithms. Switching is optimized for smooth transitions without clicks, and you can still use the **Size** control to manipulate the sound.
- **Predelay**: Adjusts the duration of the predelay, which is the time it takes for the reverb effect to set in. By increasing the predelay, you can add separation between the input signal and the reverb signal. When Predelay Sync is activated, the predelay is synchronized to the host and the **Predelay** control is replaced by the Predelay Sync controls:



The Numerator (**a**) and Denominator (**b**) set the predelay in musical notes relative to the host tempo. The Numerator sets the number of notes, while the Denominator sets the note value. The Sync mode (**c**) sets the time value, or subdivision, for the chosen note value.

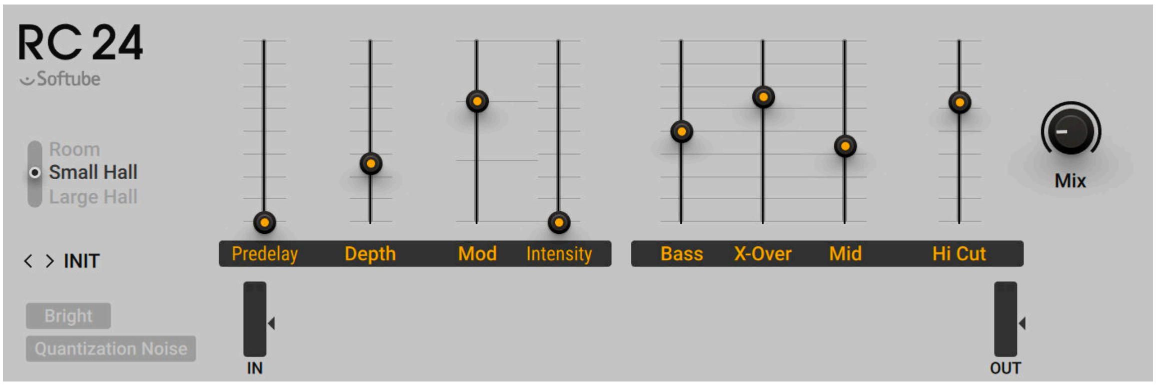
- **Feedback**: Adjusts the level of the feedback signal from the output of the predelay to its input. Turning **Feedback** to the right creates echo effects that can be further processed using the selected reverb algorithm. When **Feedback** is set to 100%, you can achieve infinite delay repeats and therefore use the predelay as a looper.
- **High Cut**: Attenuates high-frequency content of the reverb effect and the predelay's feedback signal. Turning the control to the left increases the attenuation. Turning the control to the right decreases the attenuation. Turning the control fully to the right switches the filter off.
- **Low Cut**: Attenuates low-frequency content of the reverb effect and the predelay's feedback signal. Turning the control to the left decreases the attenuation. Turning the control to the right increases the attenuation. Turning the control fully to the left switches the filter off.
- **Size**: Adjusts the swell and reflection pattern of the reverb effect, giving the impression of differently sized spaces. Turning the control to the right changes the size from small to large.
- **Diffusion**: Adjusts the texture of the reverb reflections. Turning the control to the right softens the early reflections and produces a smooth onset of the reverb.
- **Decay**: Adjusts the length of the reverb, or reverb time. Turning the control to the right changes the reverb time from short to long.
- **Density**: Switches between two basic density modes for the reflection pattern of the reverb effect, Sparse and Dense. Sparse produces dispersed and clearly discernible reflections. Dense produces a more uniform reverb tail.
- **Damp**: Adjusts the tonal quality from bright to dark. Turning the control to the right attenuates the reverb's high-frequency content.

- **Modulation**: Adjusts the amount of movement added to the reverb sound by changing internal parameters of the reverb over time. Turning the control to the right changes the movement of the reverb from lush to strongly detuned sounds.
- **Mix**: Blends between the input signal and the effect signal by means of an equal-power crossfade.
- **Reverb**: Adjusts the amount of reverb added to the effect signal by means of an equal-power crossfade.

RC 24

The RC 24 is a classic reverb with a unique and warm sound. It is inspired by an iconic hardware effect from the 1980s. RC 24 brings this renowned effect to GUITAR RIG, successfully recreating its vintage characteristics while greatly extending its usability.

This Component contains the following parameters and controls:



- **Room / Small Hall / Large Hall**: Selects from three different reverb algorithms, each of them emulating a different type of reverberating room. The Reverb selector is your first stop when searching for a particular reverb sound since it defines the overall type (size and characteristics) of the room to be emulated. Click the desired radio button or the label nearby to select the corresponding room type (the selected radio button lights up while the other two labels are grayed out). Available algorithms are: **Room**, **Small Hall**, and **Large Hall**. These algorithms closely emulate the corresponding programs of a classic hardware unit—including their distinctive (and meanwhile much appreciated) limitations and flaws.
- **Predelay**: Adjusts the initial delay between the original signal and the first reverberant sound. The possible values range from 24 to 152 milliseconds. At higher values this can also be used in many creative ways.
- **Depth**: Adjusts the apparent distance between the listening point and the source signal. The possible values range from 0 to 71. Increasing **Depth** creates a richer, fuller—but also less focused—reverb. High values of **Depth** emphasize the “running reverberation” (the reverberation occurring while the music is playing), whereas lower values bring out the attack of the sound.
- **Modulation**: Allows you to choose from four modes labeled **Mode A–D**. These modes subtly modulate the reverb tail in various ways. This can lead to somewhat “unstable” phases and pitches during the decaying phases while producing a warmer sound. Whatever mode is selected here, the intensity of the modulation is controlled via the **Intensity** fader. From the four modes, mode A has the smallest impact on the sound, while mode B, C, and D modify the sound increasingly (in that order). The resulting changes in the reverberant sound are especially noticeable on very long decaying phases.

- **Intensity**: Adjusts the intensity of the modulation as selected with the **Modulation** knob below. The possible values range from 1.00 to 3.00. At the minimal value, the modulation is rather subtle. Increasing the **Intensity** value accentuates the slight phase and pitch variations in the reverb tail.
- **Bass**: Adjusts the reverb time for the lower frequency band. The possible values range from 0.60 to 70.00 seconds. Lower values can be helpful for example to focus the reverb on higher frequencies while keeping the bottom clean. Higher values of **Bass** can create massive reverb effects, and even reach never-ending reverb sounds.
- **Crossover**: Adjusts the split frequency between the lower and higher frequency band. The possible values range from 100.0 to 10,900.0 hertz.
- **Mid**: Adjusts the reverb time for the higher frequency band. The possible values range from 0.60 to 70.00 seconds. Like **Bass**, higher values of **Mid** can lead to infinite reverb sounds! Of course, this is even more true when increasing both parameters simultaneously.
- **Hi Cut**: Adjusts the cutoff frequency of a low-pass filter that is applied to the processed signal to attenuate its higher frequencies. The possible values range from 100.0 to 10,900.0 hertz. Notably, cutting the high frequencies of the reverberant sound can make it sound more natural.
- **Mix**: Blends between the input signal and the effect signal by means of an equal-power crossfade.

The Expert panel contains the following parameters and controls:

- **Quantization Noise**: Enables sound artifacts that are created as a side effect of the lower bit resolution of classic hardware devices for a more authentic vintage sound.
- **IN**: Adjusts the input level of the Component.
- **OUT**: Adjusts the output level of the Component.

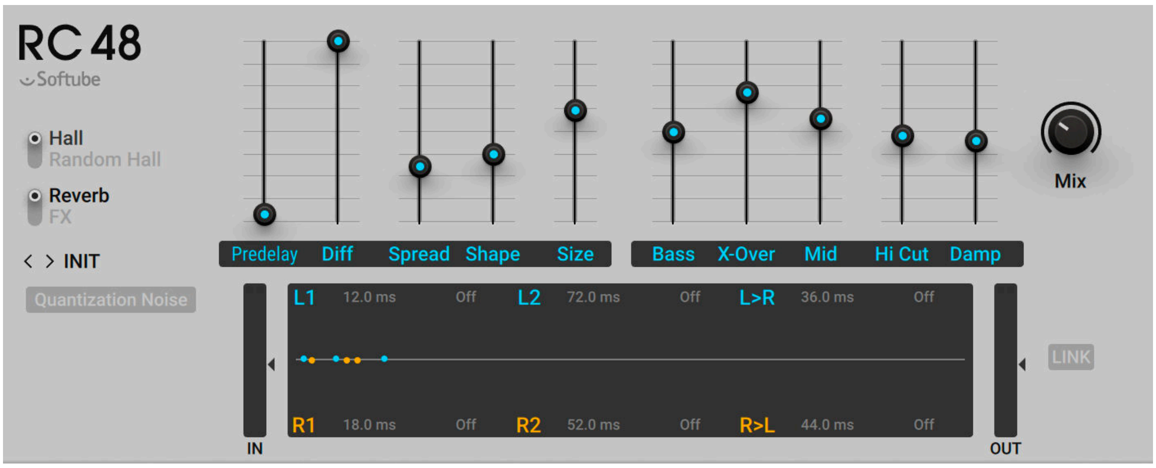
If **Small Hall** or **Large Hall** is selected, the following additional parameter is available in the Expert panel:

- **Bright**: Switches between two basic sound characters for the reverb effect. When deactivated, the sound is gloomy and dark. When activated, the sound is sparkling and bright.

RC 48

The RC 48 is a classic reverb with a unique and warm sound. It is inspired by an iconic hardware effect from the 1980s. RC 48 brings this renowned effect to GUITAR RIG, successfully recreating its vintage characteristics while greatly extending its usability.

This Component contains the following parameters and controls:



- **Hall / Random Hall**: Selects from two different reverb algorithms. The Reverb selector is your first stop when searching for a particular reverb sound since it defines the overall type and characteristics of the reverb. Click the selector to switch between both algorithms (the selected radio button lights up while the other label is grayed out). Available algorithms are **Hall** and **Random Hall**. These algorithms closely emulate the corresponding programs of a classic hardware unit—including their distinctive (and meanwhile much appreciated) limitations and flaws.
- **Reverb/FX**: Selects from two operating modes affecting how the reverb fades out ("decay" phase). Click the control to switch between both modes. Following modes are available:
  - **Reverb**: In this mode, the **Size** fader is additionally used to attenuate the value of the **Spread** and **Mid** parameter. This mode tends to reproduce what happens with a reverberating room in the real world—for example, you couldn't have very long reverb times in a tiny room. As a result, the reverb will sound more natural.
  - **Effect**: In this mode, the two aforementioned **Spread** and **Mid** parameters are not linked to the **Size** parameter. This notably allows you to experiment with unnatural settings, thus greatly expanding the reverb's creative possibilities.
- **Predelay**: Adjusts the initial delay between the original signal and the first reverberant sound. The possible values range from 0 to 510 milliseconds. At higher values this can also be used in many creative ways.
- **Diffusion**: Sets the rate at which the density of early reflections build up in the initial phase of the reverb. Adjust this parameter according to the audio material you want to process:
  - At low **Diffusion** values, the early reflections stay more distinct. This is well suited for keeping sustained, melodic sounds (e.g., vocals) natural and well defined. At very low values, percussive audio material can even sound a bit grainy.
  - At higher **Diffusion** values, the density of early reflections grows faster to a cluster of reflections, quickly thickening the sound. Notably, this can be useful to enrich percussive sounds.
- **Spread**: Adjusts the duration of the build up time and maximum phase ("sustain" phase) of the reverb envelope. Increasing **Spread** creates a fuller, but also less focused reverb. This control is related to the **Shape** fader :
  - If **Shape** is set to zero, the **Spread** knob has almost no effect.
  - The more you increase **Shape**, the more **Spread** will affect the sound.
  - If the Size Mode switch is set to **Reverb**, the **Spread** value is also influenced by the **Size** fader .
- **Shape**: Adjusts the shape of the reverberation envelope over time. This control works together with the **Spread** knob below:
  - With **Shape** at zero (fader all the way down), the reverberation strikes immediately and quickly starts decaying, producing a sharp, dry reverb. The **Spread** knob only has a very small effect and does not affect the sound.
  - When you raise **Shape**, the reverberation takes longer to develop, creating a richer sound. The duration of the reverb's attack and sustain phase can then be further adjusted via the **Spread** knob .

- **Size**: This important parameter adjusts the overall size of the acoustic space to be emulated. It additionally acts as a master control for various other parameters:
  - If the Size Mode switch is set to **Reverb**, the **Size** fader also affects the **Spread** knob and the **Mid** knob in the right part. When the Reverb selector is set to **Random Hall** at the top of the plug-in interface, the **Size** fader also limits the range of the **Wander** parameter in the **Options** page of the display.
  - If the Size Mode switch (**6**) is set to **Reverb**, the **Size** fader also affects the **Spread** knob and the **Mid** knob in the right part.
- **Bass**: Adjusts the reverb time for the lower frequency band. Note that this reverb time isn't defined in seconds but instead as a **multiplier** of the reverb time for the higher band, as set by the **Mid** fader —thus, the **Bass** value is always linked to the **Mid** value. The possible values for **Bass** range from 0.20 (a fifth of the **Mid** value) to 4.00 (four times the **Mid** value). Lower values can be helpful for example to focus the reverb on higher frequencies while keeping the bottom clean. Higher values of **Bass** can create massive reverb effects. The width of the **Bass** frequency band can be adjusted via the **Crossover** knob .
- **Crossover**: Adjusts the split frequency between the lower and higher frequency band.
- **Mid**: This important parameter adjusts the reverb time for the higher frequency band. Since the reverb time for the lower band is defined as a multiplier of this one, the **Mid** fader actually controls the overall reverb time of the unit. The available values and range of **Mid** depend on various other settings:
- **HiCut**: Adjusts the cutoff frequency of a low-pass filter that is applied to the processed signal to attenuate its higher frequencies. Notably, cutting the high frequencies of the reverberant sound can make it sound more natural.
- **RevDamp**: Adjusts the cutoff frequency above which the reverb algorithm is dampened. Whereas the **HiCut** fader described above attenuates the high frequencies in the whole processed signal, the **RevDamp** knob attenuates the high frequencies in the reverberant sound only, leaving the pre echoes untouched.
- **Mix**: Blends between the input signal and the effect signal by means of an equal-power crossfade.

The Expert panel contains the following parameters and controls:

- **Quantization Noise**: Enables sound artifacts that are created as a side effect of the lower bit resolution of classic hardware devices for a more authentic vintage sound.
- **IN**: Adjusts the input level of the Component.
- **Echo display**: Shows and edits the pre-echo configuration. The pre-echos are not fed into the reverb but instead sent directly to the output. You can add up to six echoes with varying levels and delay times. Each stereo channel has two dedicated echoes, and the two additional cross-feed echoes enable you to send an echo from one stereo channel to the other. Each line with a dot represents a particular echo. You can click and drag the dots to change the amplitude of an echo and its position in time. Alternatively, you can edit the level and delay values for each echo directly using the controls next the corresponding labels for the left echos (L1, L2), right echoes (R1, R2), and the cross-feed echoes (L>R, R>L).
- **OUT**: Adjusts the output level of the Component.

If **Random Hall** is selected, the following additional parameters are available in the Expert panel:

- **Spin**: Sets the rate at which many of the delay taps that make up the reverb will be modulated.
- **Wander**: Sets the distance in time by which the delay taps are shifted.
- **Shelf**: Adjusts a high-frequency boost applied to the processed signal. This allows you to add a second knee to the low-pass filter controlled by the **HiCut** fader.

REFLEKTOR

REFLEKTOR is a powerful convolution reverb featuring zero latency processing and smooth operation of controls across their whole range, while optimizing CPU efficiency at the same time. It contains over 350 impulse responses (IR) ranging over a wide spectrum of sources, from hi-end reverb units, to a variety of acoustic spaces, artificially modeled rooms, and special processed impulse responses.

This Component contains the following parameters and controls:

- **Category selector**: Use this drop-down menu to select a category from the factory IR's or to choose your own user selected folder. The list displays all the available categories and folders in the User Directory. When you select a folder from the menu, the first sample in that category is loaded. The first item in this list, “**Select an IR Folder...**”, allows you to point REFLEKTOR to a directory containing IRs you want to use within this particular instance of REFLEKTOR. Once a user directory is selected, it will appear in the list of categories below the factory content. Note that this selection only affects the REFLEKTOR instance it was made in.
- **Impulse Response selector**: Use this drop down menu to choose an IR sample from the list in the current category/folder. To load a user sample directly, use the first item in this list, “**Open IR...**.” Once a user sample is selected, it will appear and will load, its name will appear in the display, and its folder will be used as the User Directory. After a sample is loaded (and after all parameter changes are made to the sample), it is analyzed and the level is auto-normalized. You can load many file formats from the library including mp3. If a sample exceeds the maximum length it will be cut (Note: user selected IR's will be saved with the GUITAR RIG preset. This is a great feature that allows you to trade presets without sending samples, ensuring that presets and songs will always be loaded exactly the same in the future, even if their sample folders change).
- **Predelay**: This knob adjusts the predelay time. It adds a delay to the wet signal to simulate distance and produces rhythmic effects. When the Sync button is pressed, the knob operates in lengths related to the tempo. The values for sync are 1/32, 1/16T, 1/32., 1/16, 1/8T, 1/16., 1/8, 1/4T, 1/8., 1/4, 1/2T, 1/4., 1/2, 1/1T, 1/2., 1/1.
- **Start**: This adjusts the start position of the IR sample. You can use this to remove predelay from an IR, but adjusting it will give you a totally new sample to use. This is useful in the following type of situation: you have a long plate reverb with nice high frequency content in the end but you don't like the beginning. You can easily change it into a short high frequency IR by adjusting the start, and all your other parameters will be applied on top of this new (normalized) sample.

- **Decay**: This affects how long the tail of the signal is heard. At its default, no change occurs. When the knob is turned to less than 100%, an envelope is applied that shortens the sample length. When the knob is turned to more than 100%, an envelope is applied that amplifies the end of sample. When the Sync button is pressed, the knob operates in lengths related to the tempo. The values for sync are 1/32, 1/16T, 1/32., 1/16, 1/8T, 1/16., 1/8, 1/4T, 1/8., 1/4, 1/2T, 1/4., 1/2, 1/1T, 1/2., 1/1. When the **Reverse** button is on, it also truncates the IR (adjusting the time between the dry signal and the 'pumping end' of the reversed reverb).
- **(22) SYNC**: When turned on, the Decay and Predelay parameters are synced to the host tempo. Their scale is switched from a percentage to musical values.
- **Size**: Resizes the IR, affecting both the time and the color of the reverb. If turned to the right, the size becomes larger which makes the length longer and the frequency content lower much like what is experienced in a room when the size is increased. Turn the knob to the left to achieve the opposite effect.
- **Reverse**: This reverses the IR sample. This button also enables the **R Pos** (Reverse Position) knob. Use **Reverse** in conjunction with Sync to get a reversed sound that will always end with its climax on a beat.
- **R Pos**: This is one of REFLEKTOR's special features and is activated by pressing the **Reverse** button. With this knob you can set which part of the IR is reversed. 0% reverses the full sample, 25% keeps the first 25% of the sample unaffected and reverses the last 75%. Use this control to get an early reflection followed by an aspirated reverse climax.
- **Mute**: this will mute the send of the dry signal to the reverb processor. The tail of the reverb is still there after pressing it. This is mostly a performance control and should always be off in the requested rack presets.
- **Wet**: This knob controls the amount of the wet or processed signal that is heard.
- **Dry**: This knob controls the amount of unprocessed signal that is heard.

The Expert panel contains the following parameters and controls:

- **LOW ENV Gain**: This produces an effect similar to low dampening on a reverb, except you can increase the low frequencies present in the IR too. When increased there is an envelope that increases the volume of the content below the Frequency over time and when it is decreased, there is an envelope to lower that volume over time. Only the wet signal is affected.
- **LOW ENV Freq**: sets the cutoff frequency for the Low Frequency envelope.
- **HIGH ENV Gain**: This produces an effect similar to high dampening on a reverb, except you can increase the high frequencies present in the IR too. When increased there is an envelope that increases the volume of the content above the Frequency over time and when it is decreased, there is an envelope to lower that volume over time. Only the wet signal is affected.
- **HIGH ENV Freq**: sets the cutoff frequency for the High Frequency envelope.
- **PEAK Gain**: This is a static peak filter. The bipolar slider adjusts the gain/cut for the post reverb Peak EQ.
- **PEAK Freq**: sets the frequency for the Peak EQ.
- **PEAK Q**: sets the width for the Peak EQ.
- **Depth**: This knob simulates the depth of the room, bringing it closer or further away by increasing or decreasing the early reflections in the IR. Using this control once the **Start** knob is turned produces interesting results. It has little effect in Reverse mode.
- **Width**: This knob adjusts the stereo image. Position 0 sums the input into a mono signal and convolves it with the IR (mostly 2-channel samples). In this case, all stereo information comes from the IR. Position 1 convolves the left side of the input with the left side of the IR and the right side of the input with the right side of the IR. The knob cross-fades between these signal flows.

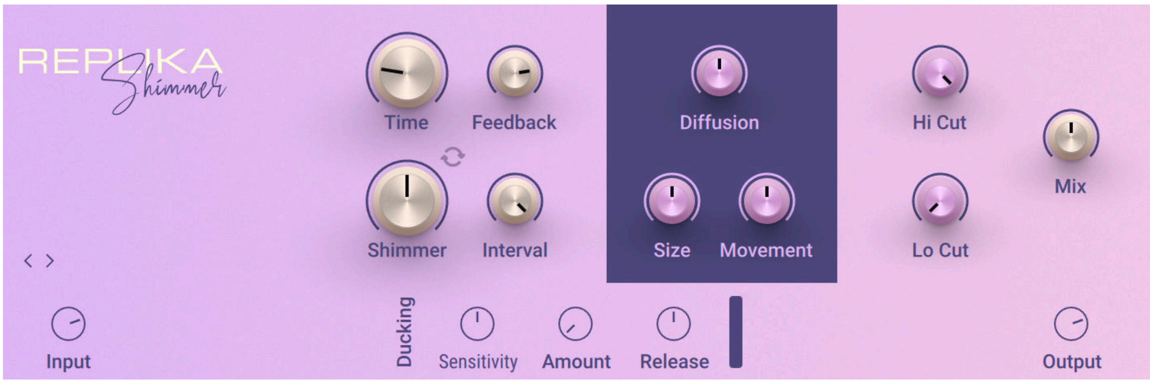
- **Pan**: This does a psychoacoustic panning of the wet signal. Turn this knob to the left to make the wet signal pan left, simulating the acoustics of shifting the reflections to the left. Turn the knob to the right to cause the wet signal to pan right.

> Decay, Size, Start, Reverse Position, LOW ENV, HIGH ENV, Depth and Pan parameters cannot be controlled using Modifiers because the calculations performed on the impulse response when changing parameter values cannot be done in real-time. You can assign MIDI controls to these parameters, however it is not recommended to use them as performance controls.

REPLIKA Shimmer

REPLIKA Shimmer combines delay, pitch shifting, and reverb to create classic shimmer effects and beyond. Pitch shifting can either be applied to the output of the delay for echoes with constant harmonization, or in the delay's feedback loop for echoes with cascading pitches. To round out the classic shimmer effect, **Diffusion** adds a reverb-like quality, which can be used to subtly wash out the delay repetitions, or turn them into clouds of sound.

This Component contains the following parameters and controls:



- **Time**: Adjusts the delay time in milliseconds.
- **Feedback**: Adjusts the level of the signal that is being fed back to the delay's input. Increasing the feedback level creates more delay repeats that decay over time. Feedback levels of 100% and above are possible, allowing the delay repeats to build up until the point of self-oscillation.
- **Shimmer Routing**: Toggles between two routing modes for the pitch shifting applied to the delay. When deactivated, pitch shifting is applied to the output of the delay, resulting in harmonized echoes with a constant pitch. When activated, pitch shifting is applied in the delay's feedback loop, resulting in echo cascades that progressively shift the pitch of each delay repetition.
- **Shimmer**: Blends between the unprocessed and the pitch-shifted delay signal, therefore adjusting the intensity of pitch shifting applied to the delay. Depending on the selected Shimmer Routing mode, pitch shifting can either be applied to the output of the delay, or in its feedback path.
- **Interval**: Adjusts the amount of pitch shifting applied to the delay in the range of -12 to +12 semitones.
- **Diffusion**: Adjusts the amount of diffusion applied to the delay signal, resulting in a reverb effect. High settings make the delay appear out of sync, so low settings are recommended if the rhythmic timing of the delay is essential.

- **Size**: Adjusts the swell, reflection pattern and decay of the reverb effect, giving the impression of differently sized spaces.
- **Movement**: Adjusts the depth and speed of modulation applied to the diffusion, shifting the timing and pitch of the reflections for a wide reverb effect.
- **Hi Cut**: Attenuates high-frequency content in the feedback path of the delay using a high-cut filter. Turned all the way to the right, the filter is off. Turning it to the left lowers the cutoff frequency of the filter, resulting in a darker tone of the delay.
- **Low Cut**: Attenuates low-frequency content in the feedback path of the delay using a low-cut filter. Turned all the way to the left, the filter is off. Turning it to the right raises the cutoff frequency of the filter, resulting in a brighter tone of the delay.
- **Mix**: Blends between the input signal and the effect signal.

The Expert panel contains the following parameters and controls:

- **Input**: Adjusts the input level of the Component.
- **Sensitivity**: Adjusts the threshold at which the ducking effect kicks in. Signal levels below this threshold will not trigger the ducking effect.
- **Amount**: Adjusts the strength of the ducking effect, which is the amount if gain reduction applied to the delay signal when the ducking effect is triggered.
- **Release**: Adjusts the release time of the ducking effect, which is the time it takes for the gain reduction to return to 0.
- **Output**: Adjusts the output level of the Component.

Spring Reverb

This is the classic reverb effect found in older amps, before the advent of solid-state reverb units. But luckily, the classic noise and hum is missing in this component.

Controls

- **REVERB** sets the amount of the signal being fed into the reverb section, controlling the intensity of the effect.
- **TIME** controls the reverb decay time. Turn clockwise to increase decay.
- **BASS** controls the low-frequency response characteristics. Turn up for a more pronounced bass sound.
- **MUTE** shuts off the signal going through the reverb section, letting only the dry signal pass through. This button can be used to trigger the reverb for single “splash” effects. When the **REVERB** control is turned fully up, you will hear no more sound, because no signal is allowed into the dry section.

Expert Controls

- **SPRING LENGTH** adjusts the length of the virtual spring. Shorter springs produce a tighter, more metallic effect, and longer springs create a diffused sound with a longer decay.
- **HIGH DAMP** reduces the decay time for high frequencies in relation to the **Time** setting.
- **LOW DAMP** reduces the decay time for low frequencies in relation to the **Time** setting.

Studio Reverb

The Studio Reverb provides a realistic emulation of halls and rooms.

Controls

- **DRY/WET** sets the amount of the signal being fed into the reverb section, controlling the intensity of the effect.
- **PRE DELAY** sets the delay time before the reverberated signal sets in.
- **ROOM SIZE** sets the cubic volume of the virtual room. Turn clockwise for a large concert hall, counterclockwise for a small auditorium or room.
- **BRIGHT** boosts high frequencies in the reverberated signal.
- **MUTE** shuts off the signal going through the reverb section, letting only the dry signals pass through. This button can be used to trigger the reverb for single “splash” effects. When the **REVERB** control is turned fully up, you will hear no more sound, because no signal is allowed into the dry section.

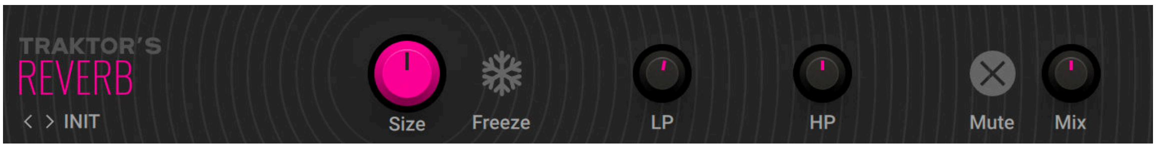
Expert Controls

- **STEREO** controls the stereo width of the reverberated signal.
- **TREBLE** controls the decay time for high frequencies in relation to the Time setting.

TRAKTOR’s Reverb

TRAKTOR's Delay is a classic delay effect with additional functionality including the Freeze function for infinite washes of sound, extended room sizes, and independent low-pass and high-pass filters.

This Component contains the following parameters and controls:



- **Size**: Adjusts the swell and reflection pattern of the reverb effect, giving the impression of differently sized spaces. Turning the control to the right changes the size from small to large.
- **Freeze**: Holds the reverb's sound content for as long as the function is activated, creating and infinite wash of sound.
- **LP**: Adjusts the cutoff frequency of the low-pass filter applied to the effect signal.
- **HP**: Adjusts the cutoff frequency of the high-pass filter applied to the effect signal.
- **Mute**: Mutes the input signal so that only the effect signal can be heard.
- **Mix**: Blends between the input signal and the effect signal.

Vintage Verb

This component offers a selection of the finest reverbs at the flick of a switch! Vintage Verb delivers the sound of selected classic plates and spring reverbs.

Controls

- **MIX** sets the amount of signal being affected by the reverb.
- The second control provides eight distinct choices of reverb effect.
- **GOLD S** creates a short gold foil plate reverb sound.
- **GOLD L** creates a long gold foil plate reverb sound.

- **SILVER S** creates a short plate reverb sound
- **SILVER L** creates a long plate reverb sound
- Next is the **STUDIO** spring reverb. This sound is similar to a plate reverb without some of the boing normally associated with springs.
- The **DUAL** spring uses two springs in series to create a unique long reverb.
- The **AMP** setting is derived from a guitar amplifier´s spring reverb, lots of boing and splash.
- Finally, the **DUB** spring reverb emulates the style of Jamaica´s famous producers—loads of resonance and tunnel sound.
- **MUTE** shuts off the signal going through the reverb section, letting only the dry signal pass through. When the **REVERB** control is turned fully up, you will hear no more sound, because no signal is allowed into the dry section.

20.12. Special FX

Special FX don't fit the other categories and enable you to transform sounds in unusual ways. From rhythmic effects to granular sound design, these Components can spark your creativity.

Grain Delay

Grain Delay is a real-time granular processor and echo effect. It slices the input signal into small segments, called grains. Grains can be manipulated in a number of ways and fed back into the input, producing otherworldly echoes. From adding subtle textures to creating massive walls of sound, Grain Delay enables you to transform your sound in unheard ways.

This Component contains the following parameters and controls:

- **Pitch**: Adjusts the amount of pitch-shifting applied to the delay signal in the range of -4 to +4 octaves (in semitones). The grains are time-stretched accordingly. The feedback signal is not affected by the pitch-shifting.
- **Size**: Adjusts the duration of the individual grains. If the grains are time-stretched according to the Pitch control, the audio contained in each grain will be repeated or cut off, keeping the overall duration of the grain constant.
- **Sync**: Synchronizes the grain duration to the metronome. When Sync is active, **Size** can be set in musical intervals relative to the tempo of the metronome.
- **Fine**: Adjusts the amount of pitch-shifting applied to the delay signal in small increments.
- **Density**: Adjusts the feedback level of the delay effect. When turned fully to the left, the effect produces a single delay repetition. Turning the control to the right increases the amount of delay repetitions.
- **Space**: Adjusts the rate of the grain repetition relative to **Size**. When turned fully to the right, the the grains overlap. Turning the control to the left first reduces the amount of overlap and then introduces a gap between the grains.

- **Drive**: Adjusts the amount of overdrive applied to the feedback signal.
- **Hi Cut**: Adjusts the cutoff frequency of a low-pass filter that is applied to the delay signal. Frequency content above the cutoff frequency is attenuated.
- **Lo Cut**: Adjusts the cutoff frequency of a high-pass filter that is applied to the delay signal. Frequency content below the cutoff frequency is attenuated.
- **Mod**: Adjusts the amount of vibrato applied to the delay signal.
- **Mute**: Mutes the effect signal going into the delay, allowing active delay repetitions to complete their decay.
- **Reverse**: Activates reverse playback of the delay signal.
- **Mix**: Blends between the input signal and the effect signal.
- **Freeze**: Activates freeze mode, in which existing grains are repeated infinitely are no new grains are acquired from the input signal.

The Expert panel contains the following parameters and controls:

- **Stereo**: Adjust the amount of stereo spread applied to the grains. Turning the control fully to the right produces a ping-ping effect.
- **Jitter**: Adds a small amount of variation to the duration of the grains, producing a more textured sound.

Ring Modulator

Ring Modulator is an effect based on the technique of ring modulation, also called balanced modulation. This special type of amplitude modulation produces sidebands in the frequency spectrum. The sidebands are the sum and the difference of the frequencies contained in the input signal and the Ring Modulator's internal modulation signal. This breaks up the harmonic structure of the sound and gives it a metallic sounding character. When using a low-frequency modulation signal, tremolo effects can be achieved. The internal modulation oscillator can also be used to apply phase modulation to the input signal (**FM**).

This Component contains the following parameters and controls:

- **Hi/Lo**: Switches between two frequency ranges for the internal modulation oscillator. When set to **Hi**, the effect produces sidebands in the frequency spectrum. When set to **Lo**, it produces tremolo effects.
- **Ring**: Adjusts the amount of ring modulation applied to the input signal using the internal modulation oscillator.
- **FM**: Adjusts the amount of phase modulation applied to the input signal using the internal modulation oscillator.
- **Freq**: Adjusts the frequency of the internal modulation oscillator, effectively moving the sidebands in the frequency spectrum. The frequency range depends on the setting of the **Hi/Lo** switch.
- **Sin/Sqr**: Switches the LFO's waveform between sine and square, producing either soft or sudden changes of the internal modulation oscillator's frequency.
- **Amount**: Adjusts the amount of modulation applied to the frequency of the internal modulation oscillator using the LFO.

- **Rate**: Adjusts the frequency of the LFO that can be used to apply modulation to the frequency of the internal modulation oscillator.
- **Sync**: Synchronizes **Rate** to the metronome. When **Sync** is active, **Rate** can be set in musical intervals relative to the tempo of the metronome. The interval can be set to 1/4, 1/8, 1/16, and 1/32 notes.

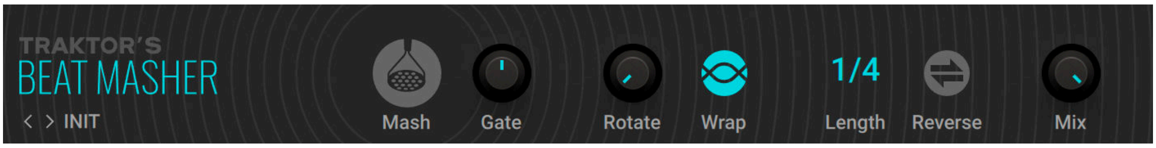
The Expert panel contains the following parameters and controls:

- **Edge**: Changes the waveform of the internal modulation oscillator. Turning the control to the right adds harmonics, resulting in a more aggressive sound of the ring modulation.

TRAKTOR's Beat Masher

Beat Masher captures a loop from the incoming audio and manipulates it by applying rhythmic stutter, gating, repeater, and reverse effects in real time.

This Component contains the following parameters and controls:

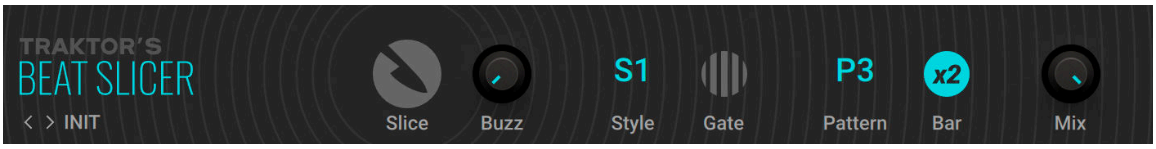


- **Mash**: Starts the effect by recording the input signal into the internal buffer. When deactivated, the input signal is unaltered.
- **Gate**: Grabs and plays slices of audio from the internal buffer. When turned fully to the left, the effect is bypassed. When turned right towards center position, increasingly longer slices are played. When set to center position, the full buffer is played. When turned right from the center position, increasingly longer slices of audio are cut, creating a gating effect.
- **Rotate**: Shifts the audio from the buffer relative to its original position in steps of 1/8 notes. When **Length** is turned fully to the left, **Rotate** continuously rotates the sample.
- **Wrap**: Restarts the effect from the start of each bar independently from the **Length** setting.
- **Length**: Adjusts the length of the audio from the internal buffer.
- **Reverse**: Reverses the playback direction of the audio from the internal buffer.
- **Mix**: Blends between the input signal and the effect signal.

TRAKTOR's Beat Slicer

Beat Slicer captures a loop from the incoming audio and manipulates it by rearranging slices into a variety of rhythmic patterns in real time.

This Component contains the following parameters and controls:



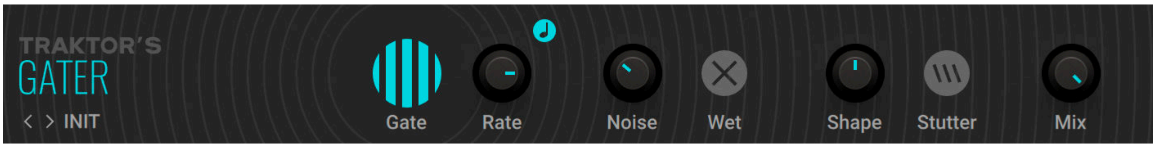
- **Slice**: Starts the effect by recording the input signal into the internal buffer. When deactivated, the input signal is remains unaltered.
- **Buzz**: Creates a rolling beat effect by repeating beats from the current pattern. Turning the control to the right increases the rate of repetition.
- **Style**: Selects one of five groups of patterns.

- **Gate**: Gates the audio based on a rhythm derived from another pattern, creating infinite variations through combination of the playback and gating patterns. When activated, the **Buzz** control is inactive.
- **Pattern**: Selects pattern from the group set using **Style**. The first pattern in each group plays back the unaltered audio from the buffer.
- **Bar x2**: Extends the audio used from the buffer to two bars. Otherwise only the first bar of audio is used.
- **Mix**: Blends between the input signal and the effect signal.

TRAKTOR's Gater

The Gater rhythmically mutes incoming audio to create the classic gating effect. When using the internal noise source, it can be used as a rhythmic sound generator.

This Component contains the following parameters and controls:

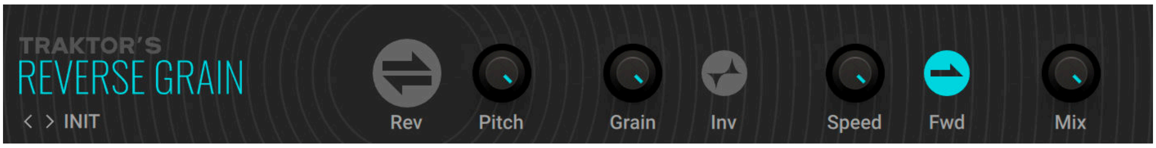


- **Gate**: Switches the effect on or off.
- **Rate**: Adjusts the rate of the gating effect.
- **Sync**: Synchronizes **Rate** to the metronome. When **Sync** is active, **Rate** can be set in musical intervals relative to the tempo of the metronome. The interval can be set to 1/4, 1/8, 1/16, and 1/32 notes.
- **Noise**: Adjusts the amount of hissing noise added to the signal.
- **Wet**: Mutes the input signal so that only the effect signal can be heard. When combined with the **Noise** control, the Gater can be used as a rhythmically gated noise source .
- **Shape**: Adjusts the hold and decay times of the gating effect's contour:
  - Turned fully left: 1% hold, 0% decay
  - Center position: 50% hold, 0% decay
  - Turned fully right: 0% hold, 100% decay
- **Stutter**: Sets the gating time to a 3/16 note, producing a stuttering effect.
- **Mix**: Blends between the input signal and the effect signal.

TRAKTOR's Reverse Grain

Reverse Grain captures a loop from the incoming audio and applies granular processing to it, including control over playback direction, pitch, and grain size.

This Component contains the following parameters and controls:



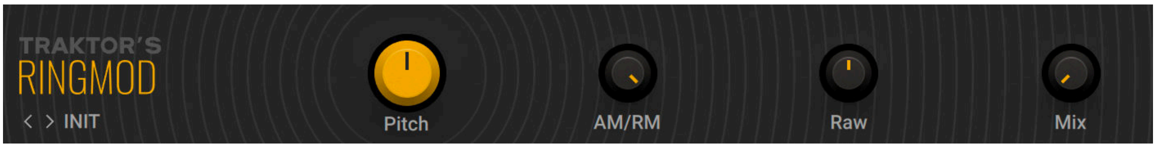
- **Rev**: Starts the effect by recording the input signal into the internal buffer and playing it back in reverse.

- **Pitch**: Adjusts the pitch of the audio from the internal buffer. When turned fully to the right, the pitch is unaltered. Turning **Pitch** to the left pitches the audio down.
- **Grain**: Adjusts the size of the grains used to process audio from the internal buffer. Interesting effects can be achieved when combined with the **Speed** control.
- **Inv**: Plays the grains in reversed order.
- **Speed**: Adjusts the playback speed of the grains used to process audio from the internal buffer. When turned fully to the right, the playback speed is unaltered. Turning **Speed** to the left reduces the playback speed.
- **Fwd**: Inverts the playback direction to forward.
- **Mix**: Blends between the input signal and the effect signal.

TRAKTOR's Ring Modulator

The Ring Modulator transforms sound by modulating the input signal's amplitude using an oscillator. This process adds and shifts the harmonic content, resulting in metallic and bell-like tones. It features both AM (amplitude modulation) for a softer sound, and RM (ring modulation) for a harsher sound.

This Component contains the following parameters and controls:



- **Pitch**: Adjusts the frequency of the modulation oscillator in the range of 100 Hz - 8371 Hz.
- **AM/RM**: Blends between AM (amplitude modulation) and RM (ring modulation).
- **Raw**: Adjusts the wave shape of the modulation oscillator from a sine wave to a filtered square wave. Turning **Raw** to the right makes the sound increasingly harsher.
- **Mix**: Blends between the input signal and the effect signal.