GUITAR RIG 6

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Software version: 6.0.2 (10/2020)

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 UI elements.
[Brackets]	References keys on a computer's keyboard

The following three icons represent different types of information:

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

The **exclamation mark** 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 your 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 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 shows you 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 enable you to quickly find the right content. For more information, see Overview of the Browser.

5.1. 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.

5.2. 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:
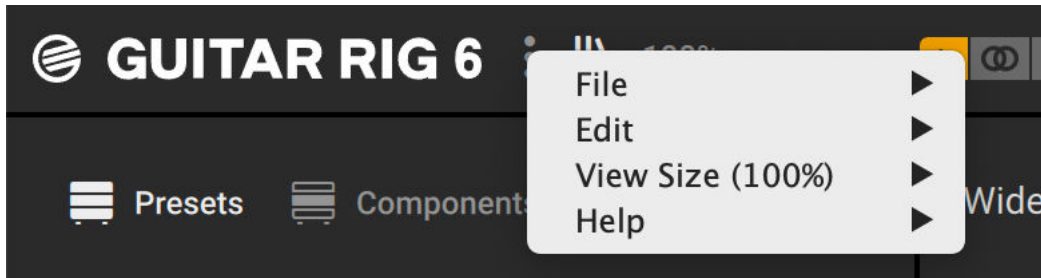
- **File**: Contains options for creating, importing, and saving preset files. Additionally, you can open the Preferences from this sub-menu. For more information, see Using Presets and Preferences.
- **Edit**: Contains options for editing the Rack. You can cut, copy, paste, and delete Components, select them all, or undo and redo your edits. For more information, see Editing the Rack Using the Main Menu.
- **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.

5.3. 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:
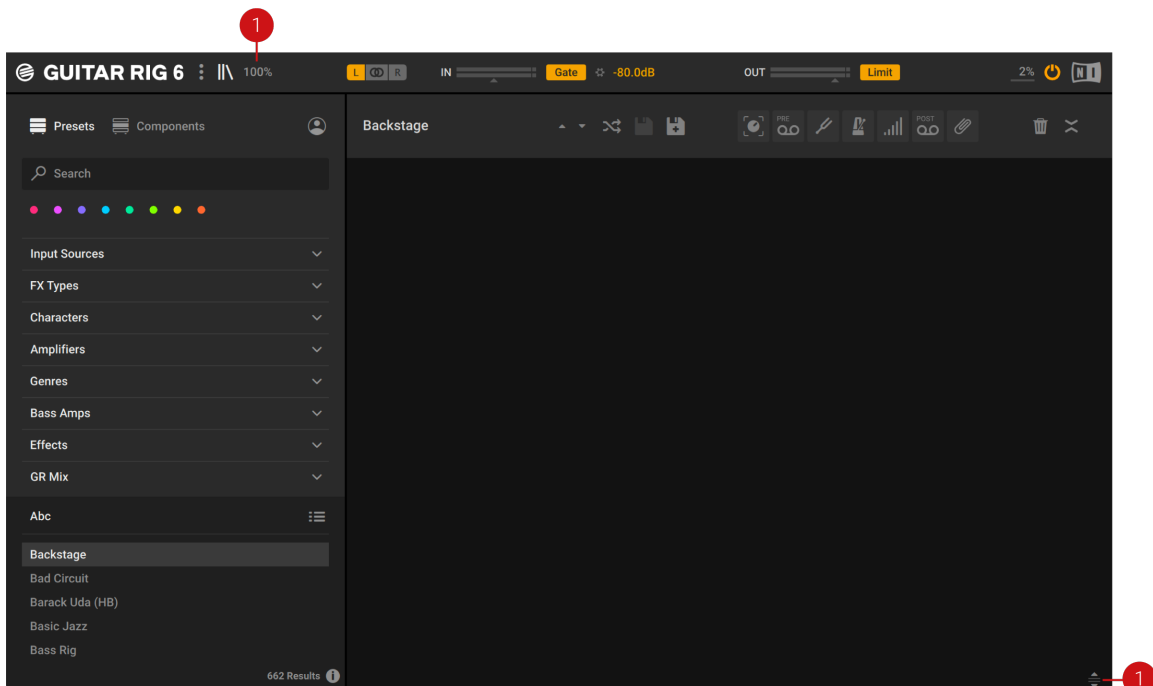
1. **View size**: Sets the magnification of the user interface, and gives the option to save the current size as default. The magnification can be set in 9 steps from 75% to 200%.
2. **Rack size**: Adjusts the vertical size of the Rack by dragging the icon up and down. When used in conjunction with the 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).

5.4. 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 topmost component, and the outputs are taken from the bottommost component.

> **!** 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.

You can use the Input selector in the Header to select the input configuration for the Rack. The following options are available:
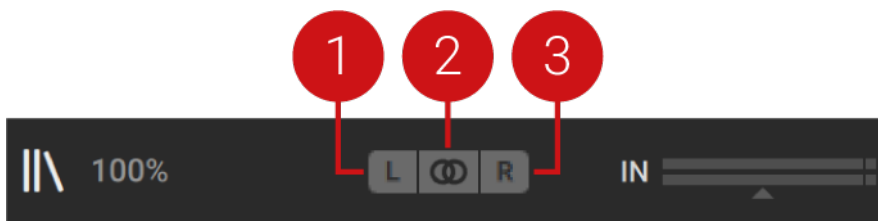
1. **Left Input**: Only the left stereo input is used in a mono configuration. Components using stereo processing receive the signal from the left stereo input on both stereo channels.
2. **Stereo Input**: Both the left and right stereo input are used in a stereo configuration. Components using stereo 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.

Most Components use stereo processing from input 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. For more information, see Component Controls.

However, some Components 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 (for example reverb effects).

The following Components sum the stereo inputs to mono:

• Cabinets & Mics
• Psyche Delay
• Quad Delay
• Flanger Chorus
• Rotator
• Stereo Tune
• Octaverb
• Raum
• Reflektor
• Traktor's Reverb

6. 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 enable you to quickly find the right content.

The following overview shows you the basic structure of the Browser:
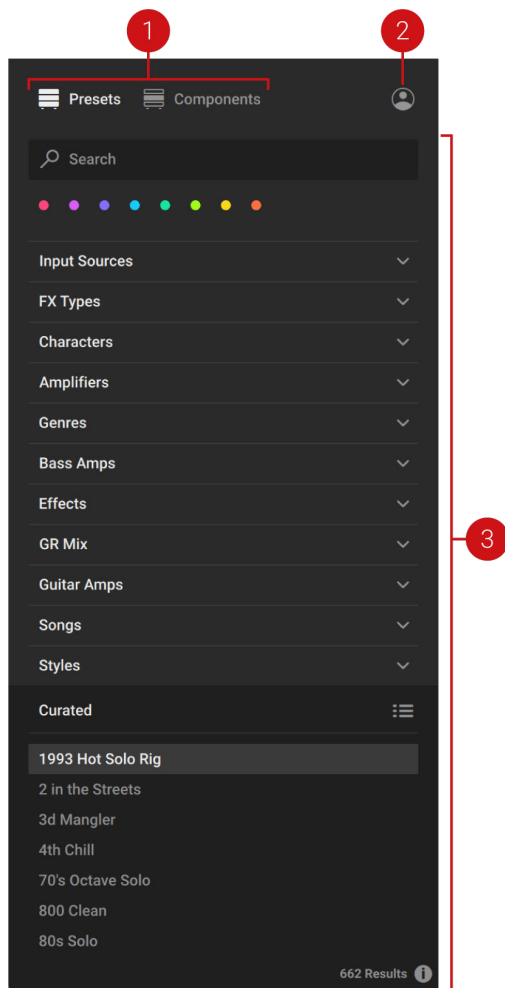
1. **Content selector**: Selects the type of content to be shown in the Browser interface. The following types of content are available in GUITAR RIG:
   - **Presets**: A Preset recalls a previously saved Rack including its Components, all individual parameter settings, and the options set in the Toolbar. 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.

6.1. 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:
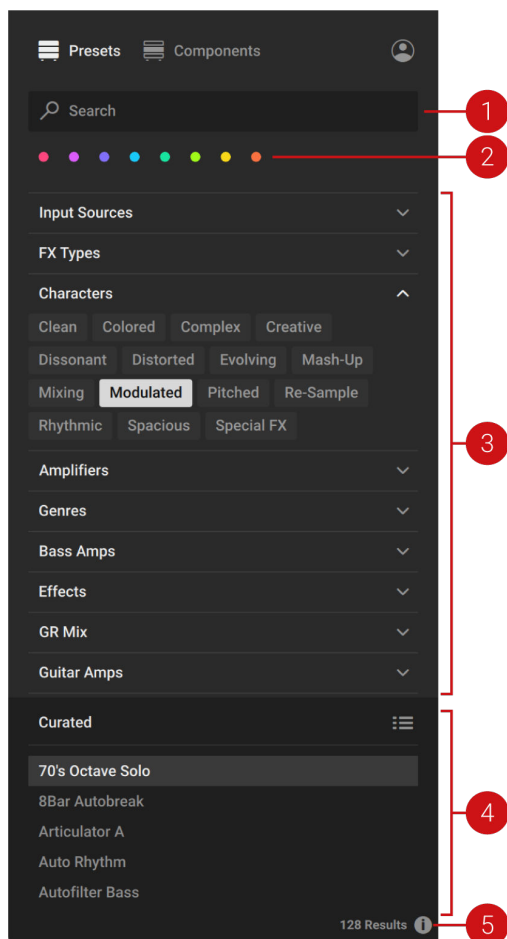
1. **Search field**: Finds presets matching the terms entered in the field and shows them in the Results list. For more information, see Using the Search Field.

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 Using 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. For more information, see Managing User Presets.

6.2. 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:
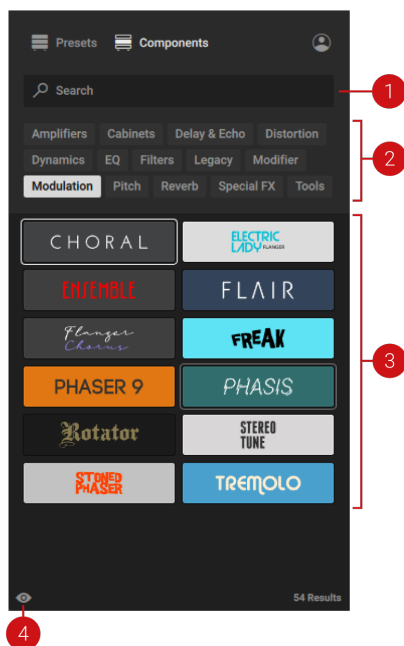
1. **Search field**: Finds Components matching the terms entered in the field and shows them in the Component Tiles. For more information, see Using the Search Field.

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.

7. 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.

7.1. 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.
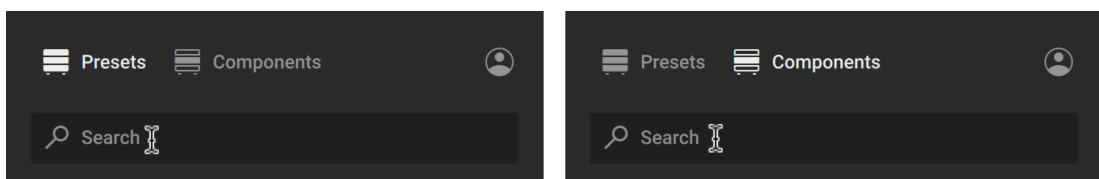
2. Enter your search terms.
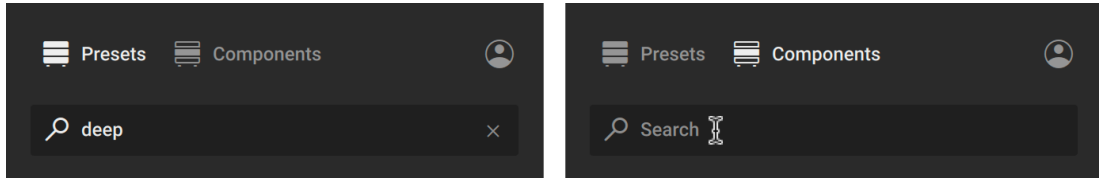
→ The Results list or Component Tiles are filtered according to the terms entered in the Search field.
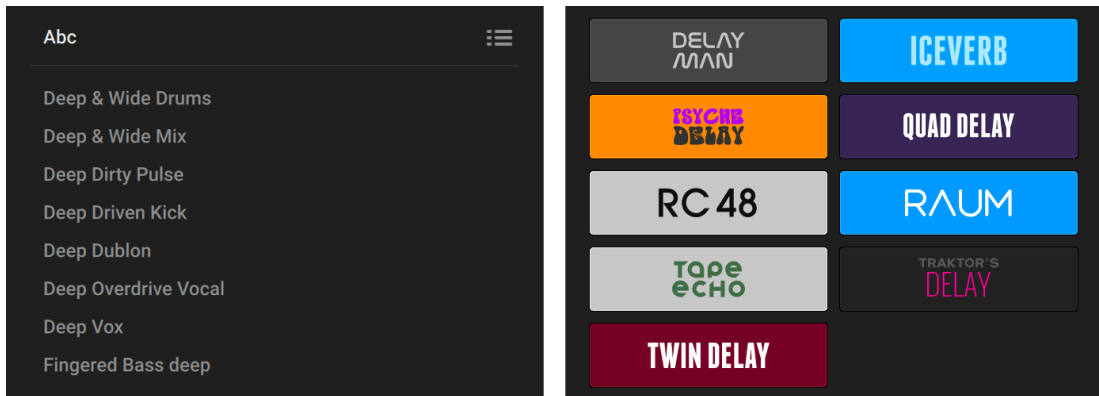
7.2. Using Favorites

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 to a preset:

1. Right-click a preset in the Results list to open the context menu.
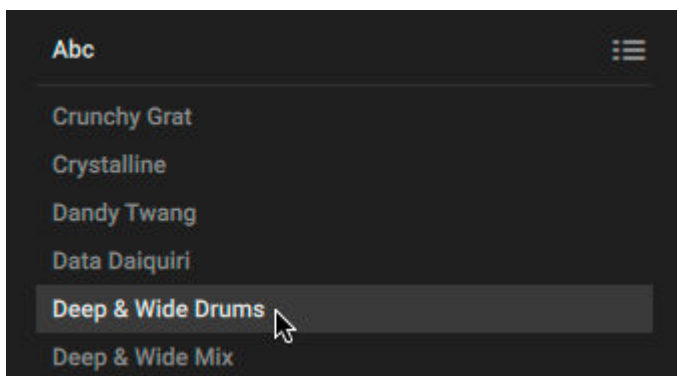
2.   Select one of the colored Favorite tags from the context menu.
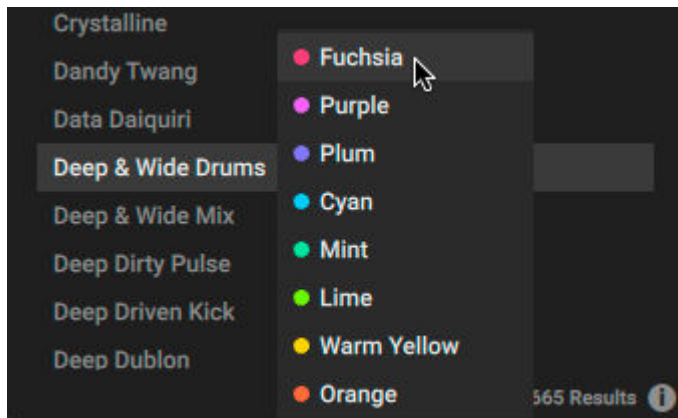
→   The selected Favorite tag is assigned to the preset and shown as a colored dot next to preset title.
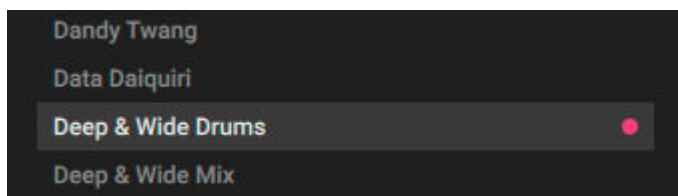
To filter the presets shown in the Results list by your previously assigned Favorite tags:

▶   Click a tag in the Favorites to select it.
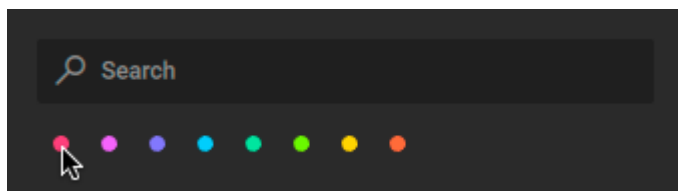
→   The Results list is filtered according to your selection and only shows presets that have been assigned the respective Favorite tag.
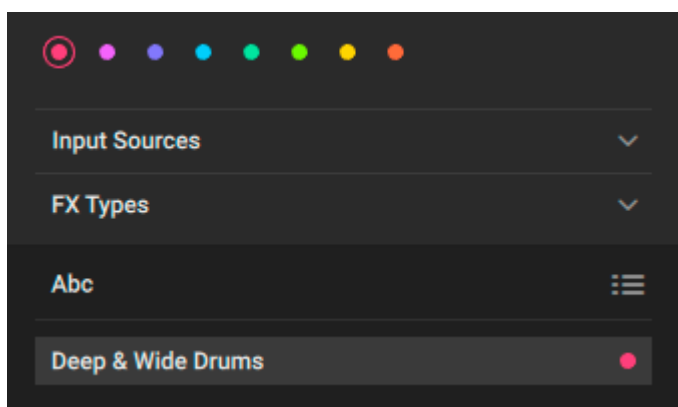
✔ Favorites shown in the Results list can be sorted in a custom order set by the user. For more information, see Sorting Favorites in a Custom Order.

7.3. 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 Managing User Presets.

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.

7.3.1. 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.
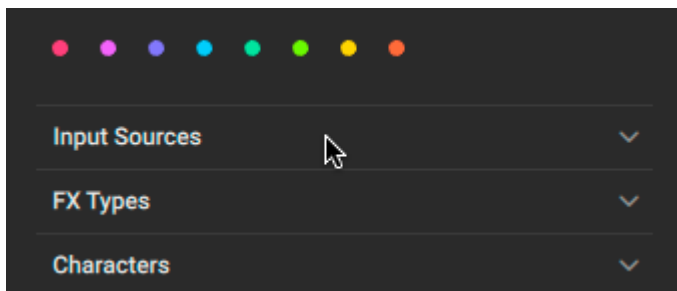
2. Click a Filter tag to start filtering the presets in the Results list.
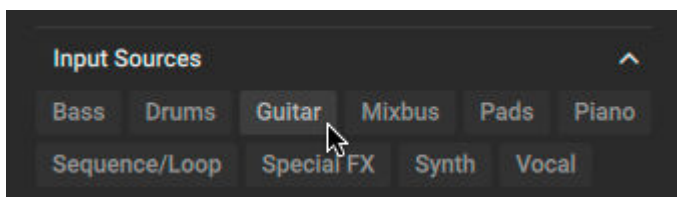
3. Click another Filter type to open it and show the contained Filter tags. The other Filter types will be closed automatically.

    !   To show Filter tags for multiple Filter types at the same time, press [command] (macOS) or [Ctrl] (Windows) + click to open them.
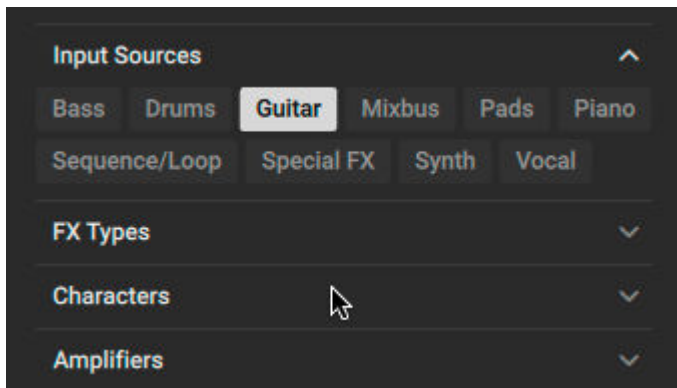
4. Click a Filter tag to continue filtering the presets in the Results list.
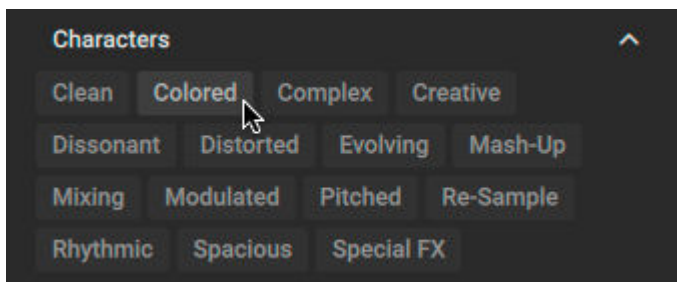
5. Press [command] (macOS) or [Ctrl] (Windows) + click to select another Filter tag from the same Filter type and further filter the presets in the Results list.
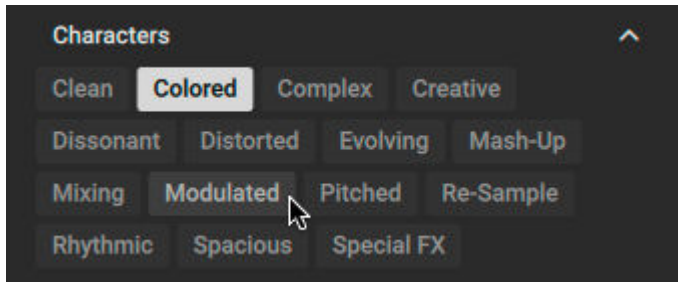
→ The presets in the Results list are filtered by the selected Filter tags. Only presets carrying all of the selected Filter tags are shown.
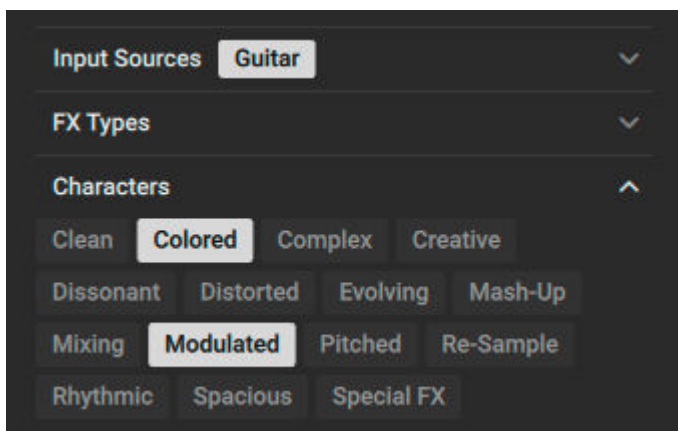
7.3.2. Using Filters for Presets

To filter Component Tiles using the Category Filter:

1. Click a Filter tag to start filtering the Components Tiles.
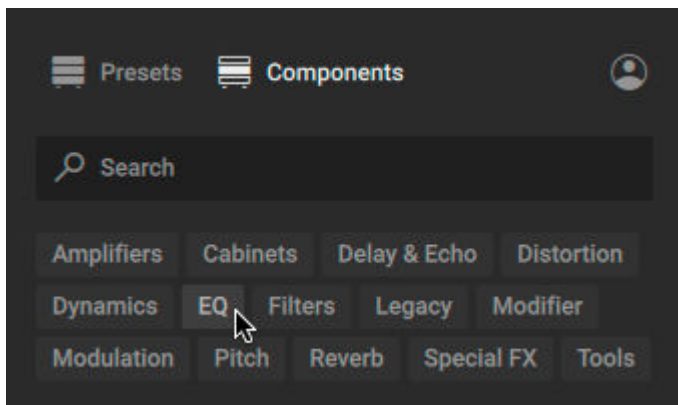
2. Press [command] (macOS) or [Ctrl] (Windows) + click to select another Filter tag and add Component Tiles from the respective category.
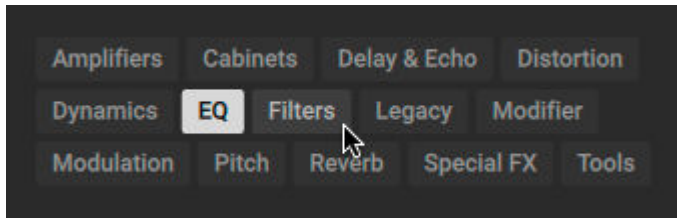
→ The Component Tiles are filtered by the selected Filter tags. All Components in any of the selected categories are shown.
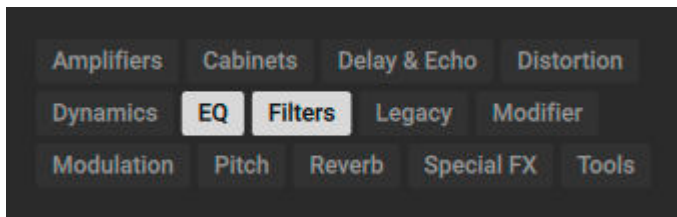
7.4. Using the Results list

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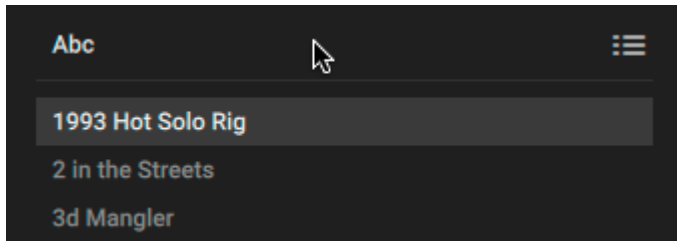
2.   Click on one of the options to select it.

→   The presets in the Results list are sorted according to the selected option.
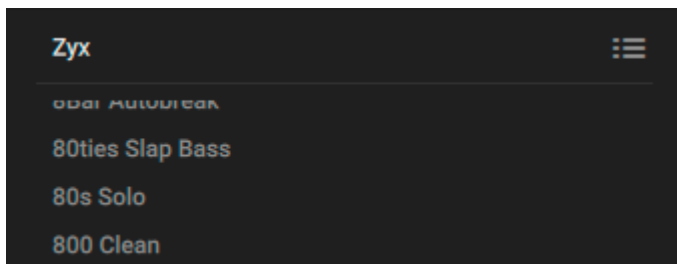
The following Sorting options are available:

- **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.

7.5. 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.
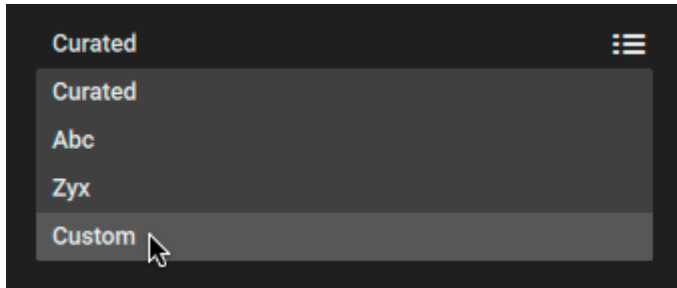
3. Click, drag, and drop any of the presets into a new position in the Results list.
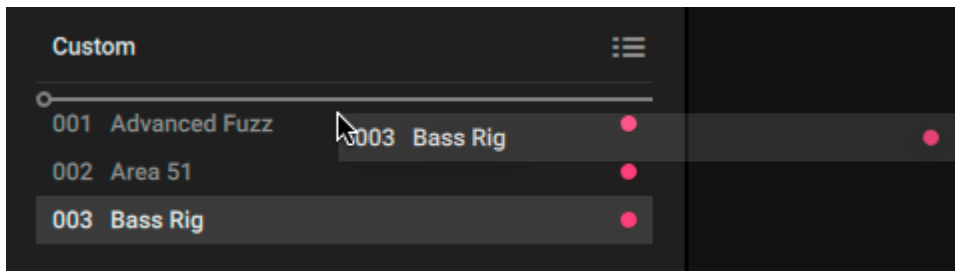
→ The Favorites in the Results list are sorted in your new custom order.
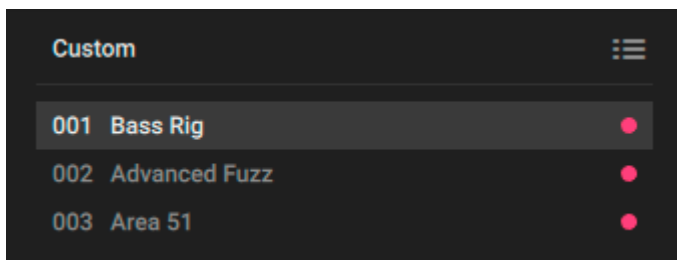
> You can recall this custom order at any time by selecting **Custom** in the Sorting options when a Favorite tag is selected in the Browser. For more information, see Using the Results list.

8. 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.

8.1. 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.
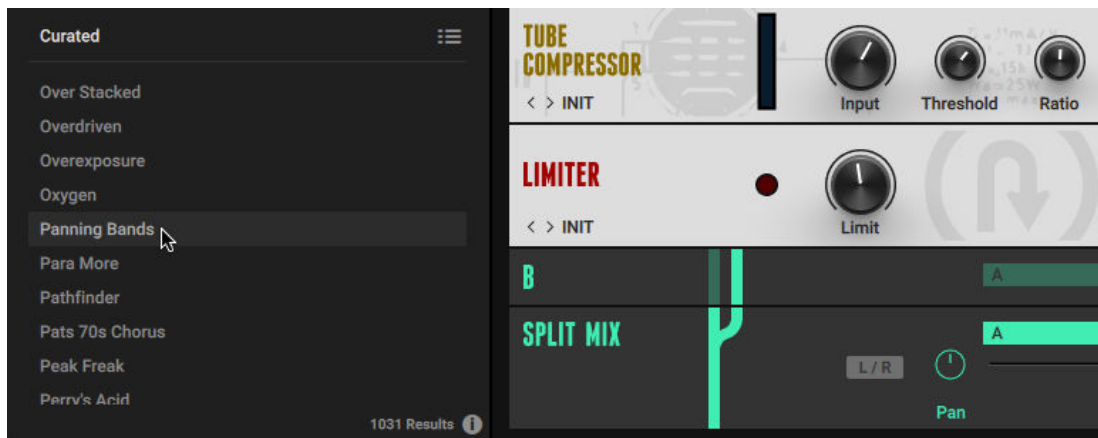
You can also use the Rack's Toolbar to load presets. The following overview shows you the available controls:
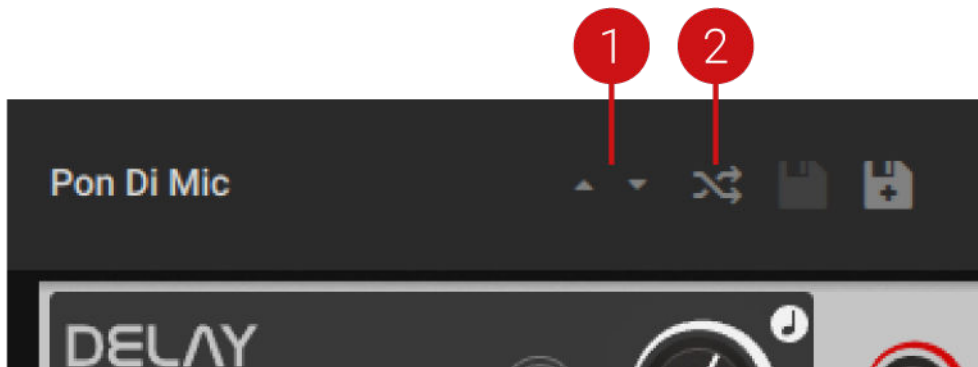
1. **Previous preset / Next preset**: Switches between presets in the Browser's Results list. Clicking on the upwards arrow loads the previous preset. Clicking on the downwards arrow loads the next preset.
2. **Preset Shuffle**: Loads a random preset from the Browser's Results list.

8.2. Saving User Presets

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

8.2.1. 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.
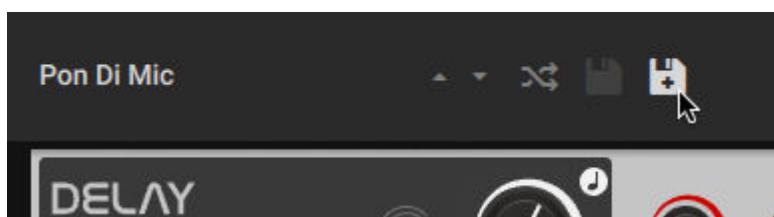
2.   Type in a name for your User preset and confirm by clicking **OK**.
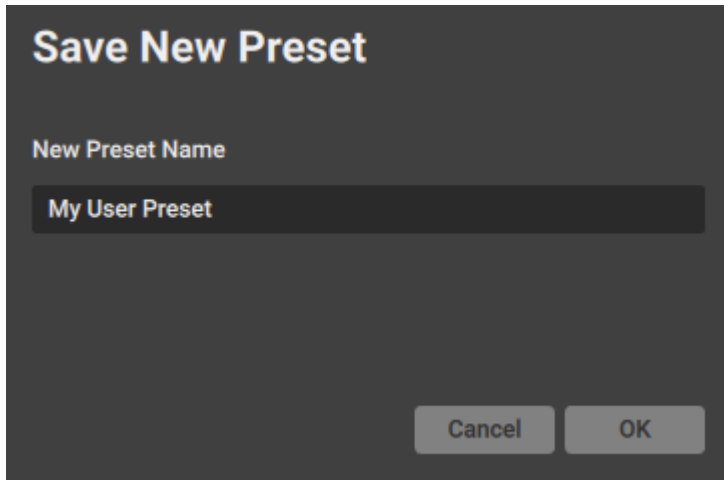
→   The User preset is saved and can be accessed in the Browser.

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

8.2.2. 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.
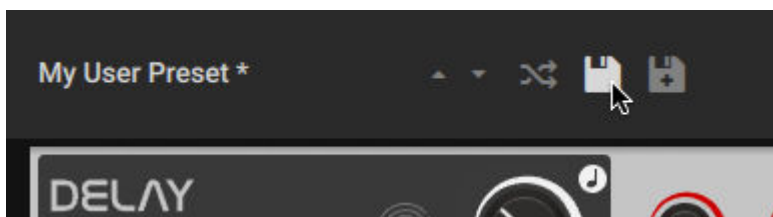
8.3. Importing User Presets

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.
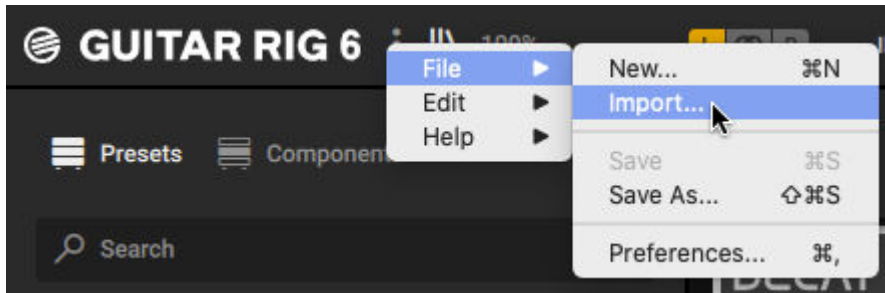
2. Navigate to the preset file you want to import, select it, and confirm by clicking **Open**.
→   The User preset is imported and can be accessed in the Browser.

> **!** 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.

8.4. 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:
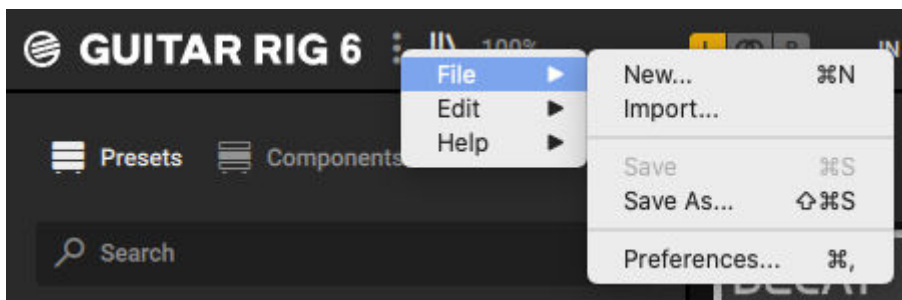
• **New...**: Clears the Rack and creates a new User preset.
• **Import...**: Imports User presets from the hard drive. For more information, see Importing User Presets.
• **Save**: Saves the loaded User preset. When no User preset is loaded, the command is deactivated and grayed out.
• **Save As...**: Saves the loaded preset as a new User preset.

8.5. Managing User Presets

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.
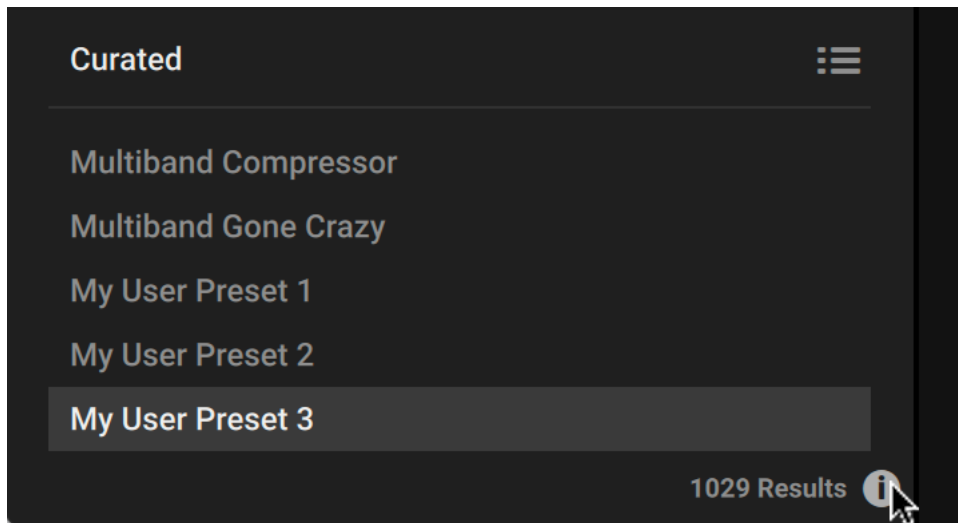
The following overview shows the Filter tags and properties you can edit in the Info pane:
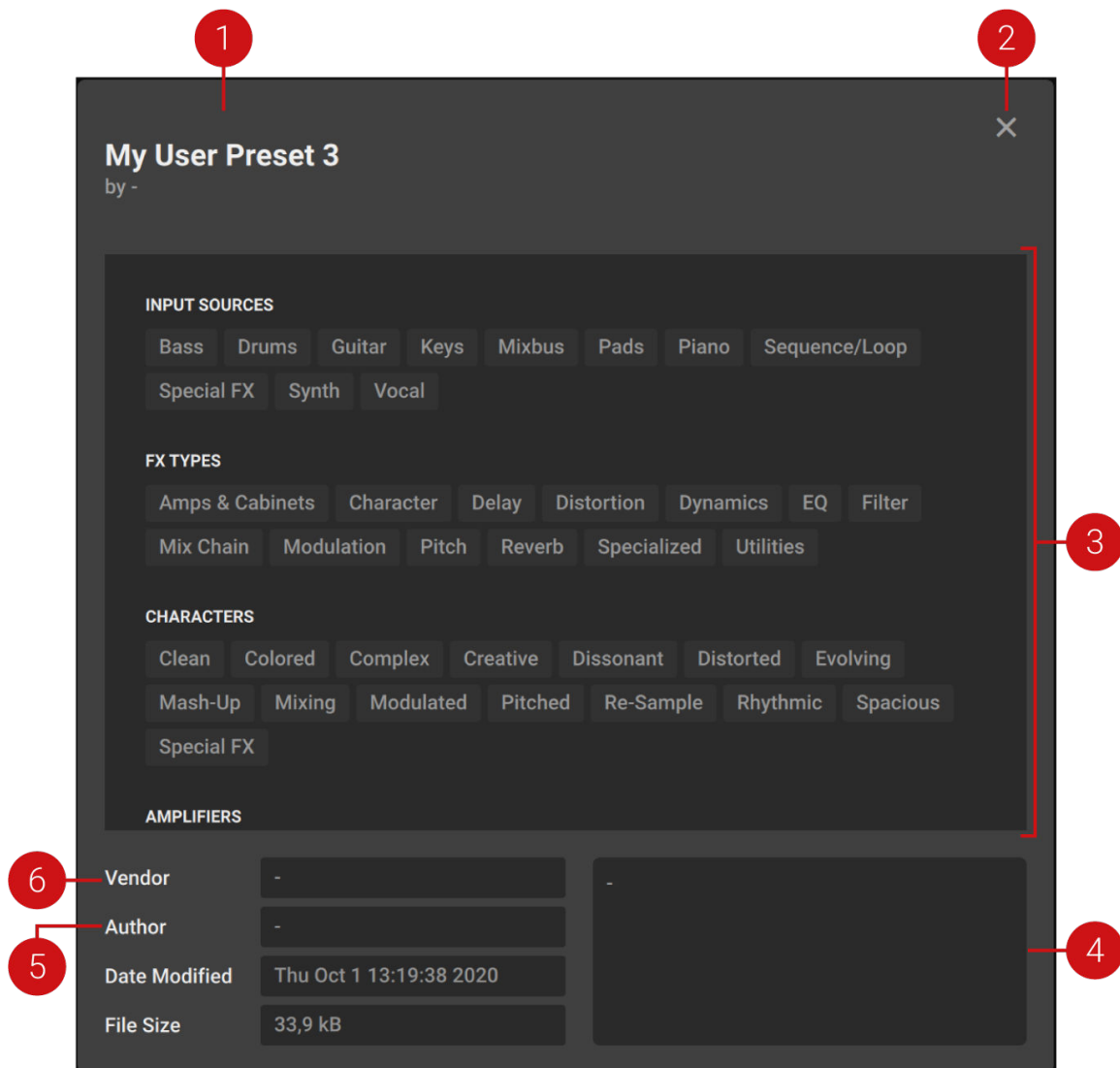
1. **Preset name**: Displays the name of the User preset. You can click the name to change it.
2. **Close Info pane**: Closes the Info pane.
3. **Filter tags**: Displays all available Filter tags. You can add Filter tags to your User preset by selecting them.
4. **Comment**: Displays the comment associated 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.

8.6. 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 access Component presets in the Browser or in the Component itself.

8.6.1. Component Presets in the Browser

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

!   The Component selector at the top of the Browser needs to be set to **Components**.
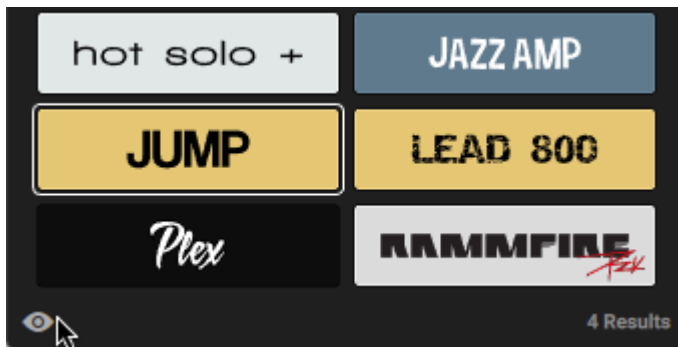
You can load Component presets using drag and drop. The associated Component is loaded automatically.

▶   To load a Component with a specific Component preset, drag and drop the Component preset from the Browser into the Rack. The expected position in the Rack is indicated by a yellow line.

!   You can also replace one or multiple Components by dragging and dropping Component presets, in the same way Components can be replaced with Components. For more information, see Replacing Components in the Rack.
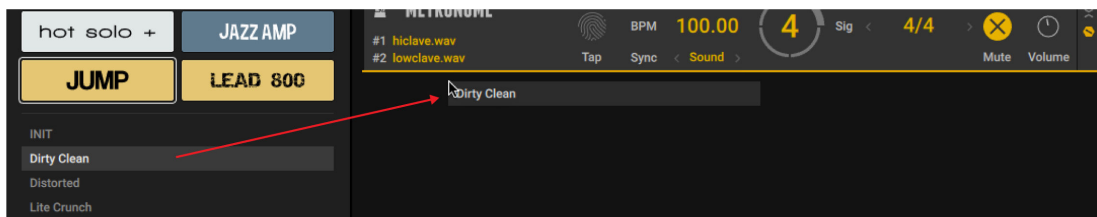
8.6.2. Component Presets in the Component

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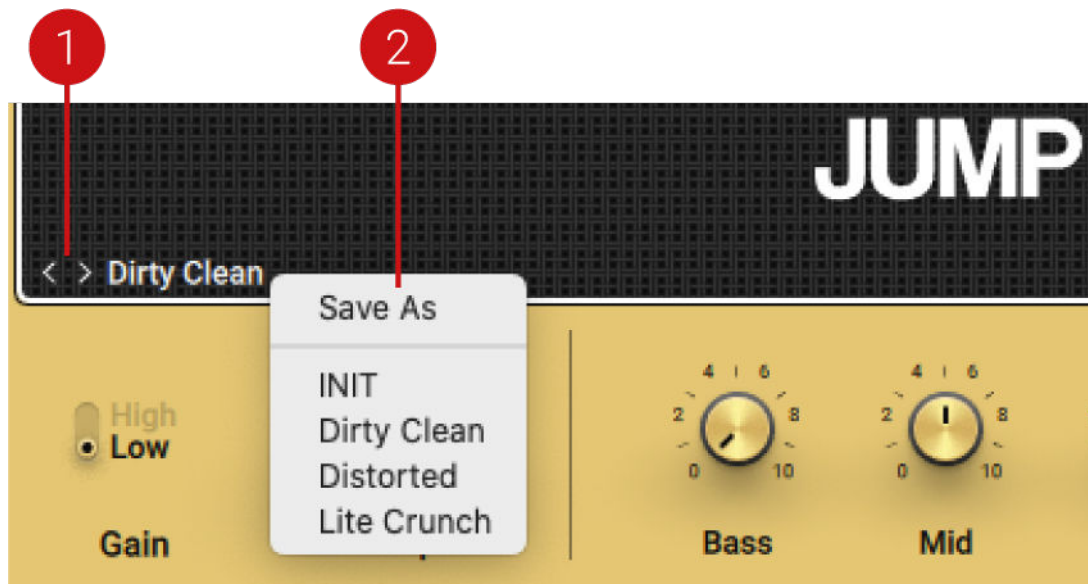
1. **Previous preset / Next preset**: Switches between Component presets of the respective Component. Clicking on the leftward arrow loads the previous preset. Clicking on the rightward arrow loads the next preset.
2. **Drop-down menu**: Provides the **Save As** option to save a new User Component preset,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

9. 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.

9.1. 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:
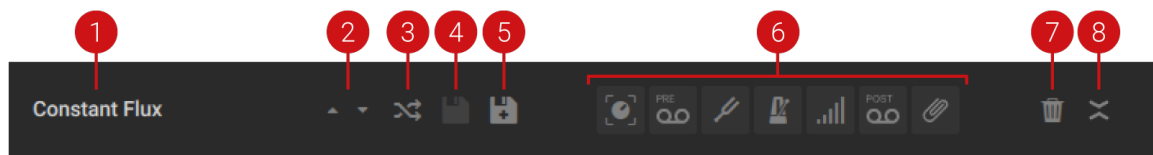
1. **Preset display**: Shows the loaded preset. When the preset has been changed, this is indicated by a * following the preset name.
2. **Preset selection**: Switches between presets in the Browser's Results list. For more information, see Loading Presets.
3. **Preset Shuffle**: Loads a random preset 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.

9.2. 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:
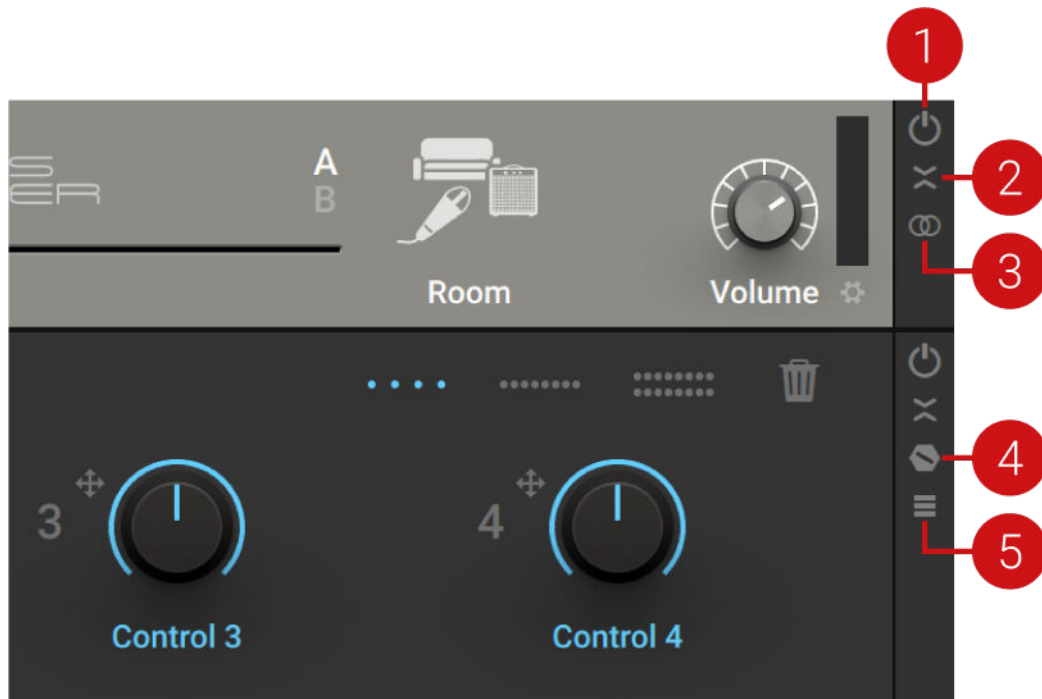
1. **Component On/Off**: Switches the Component on or off. When on, the input signal is processed and passed on to the next Component. When off, the input signal bypasses the Component and processing is deactivated, removing the Component's load from the CPU.

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.

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.

10. 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.

10.1. 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.
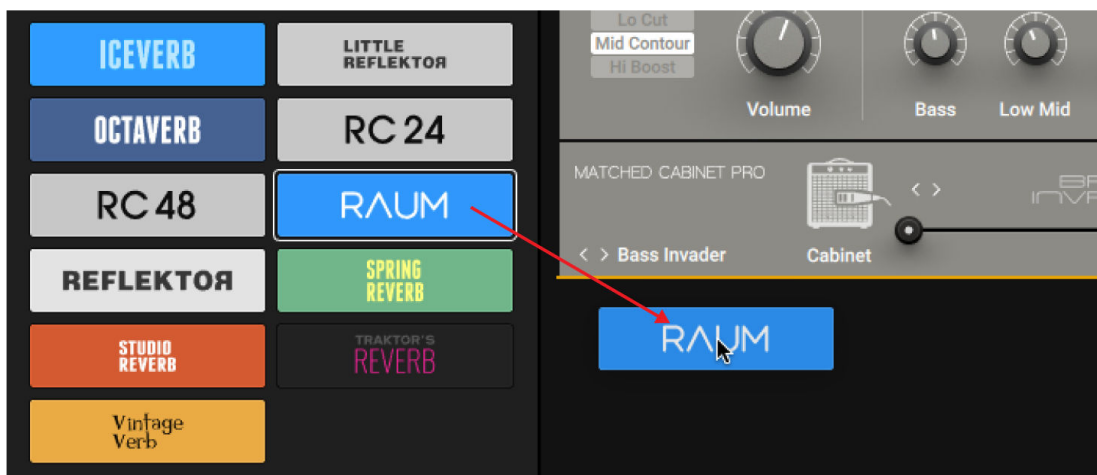
> ✔ When no Component is selected in the Rack, you can also add Components by double-clicking the respective Component Tile in the Browser. The Component will be added at the bottom of the Rack.

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 Components by clicking on them. 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.
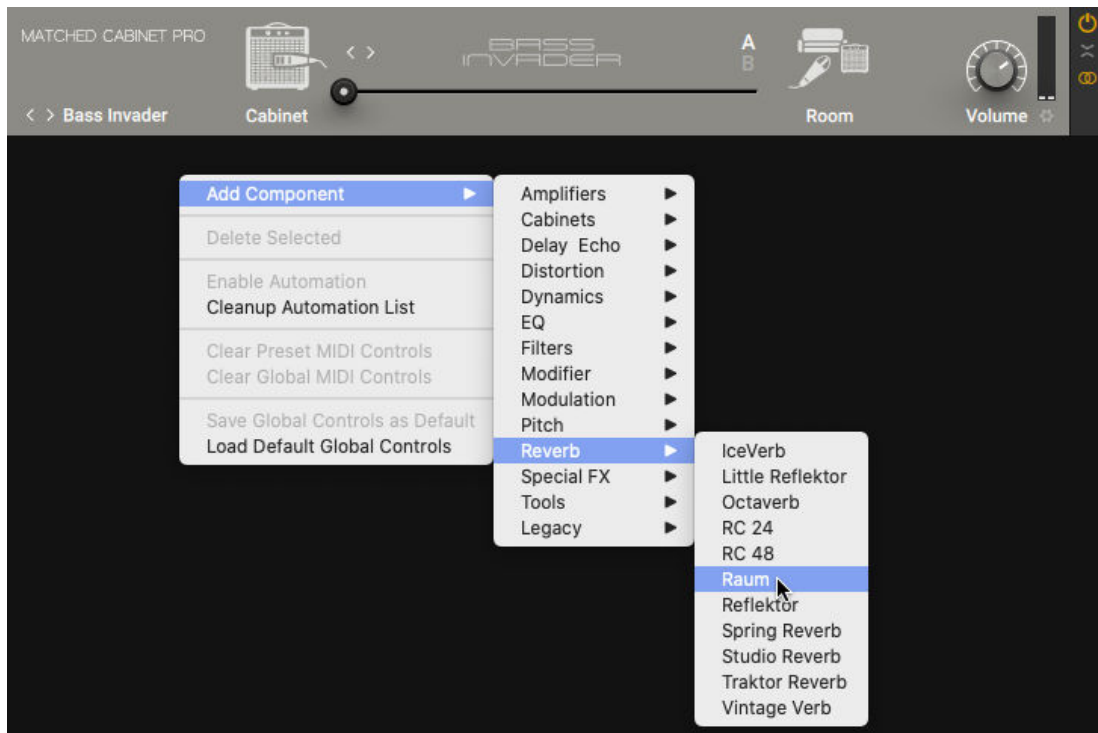
10.2. Deleting Components from the Rack

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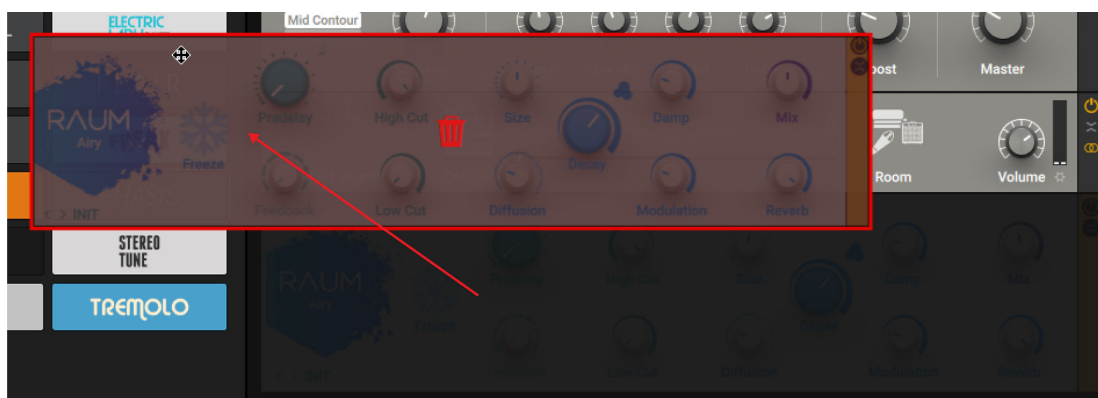
You can also use the context menu to delete Components. If no Component is selected, the command in the context menu is not available. If one or more Components are selected, they will be deleted. Opening the context menu automatically selects the Component at the mouse position.

! You can select Components by clicking on them. 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 Selected**.

✔ 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 Using the Main Menu.
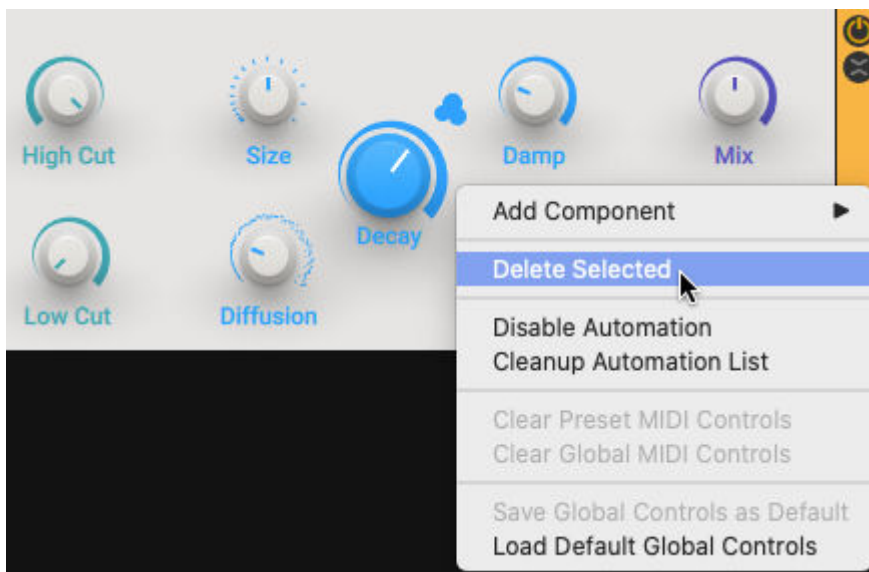
10.3. Replacing Components in the Rack

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 Components by clicking on them. Holding SHIFT on the keyboard allows for selection of multiple Components.

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

10.4. 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.
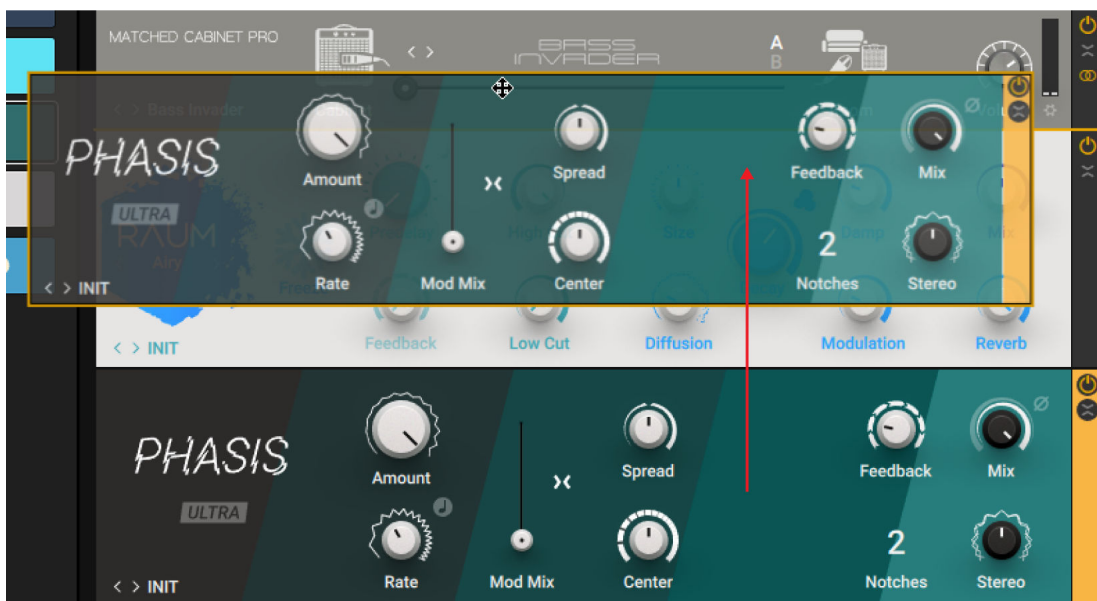
! You can select Components by clicking on them. Holding SHIFT on the keyboard allows for selection of multiple Components.

10.5. Editing the Rack Using the Main Menu

You can use the Main menu in the Header to edit the Rack by using common editing commands. The commands can be accessed in the **Edit** sub-menu.

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

The following overview shows you the available commands:
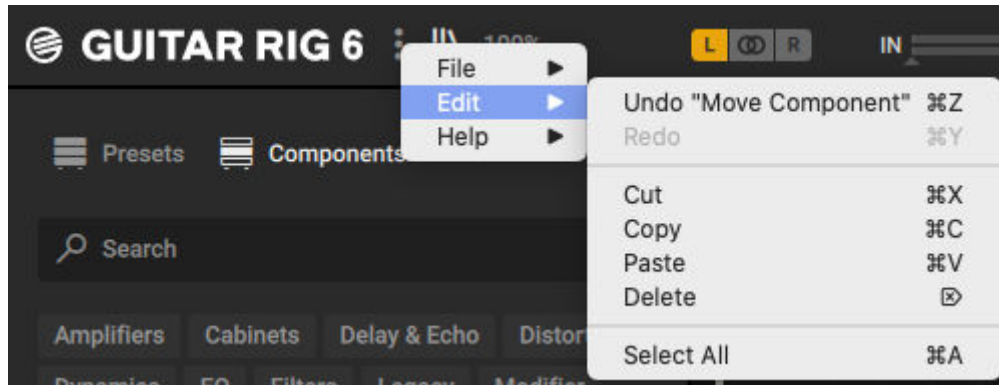
- **Undo**: Reverts the last change made to the Rack.
- **Redo**: Repeats the last change from the undo history.
- **Cut**: Cuts all selected Components and copies them to the clipboard.
- **Copy**: Copies all selected Components to the clipboard.
- **Paste**: Pastes the Components from the clipboard.
- **Delete**: Deletes all selected Components.
- **Select All**: Selects all Components in the Rack.

❗ You can select Components by clicking on them. Holding SHIFT on the keyboard allows for selection of multiple Components.

11. 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.

11.1. 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:
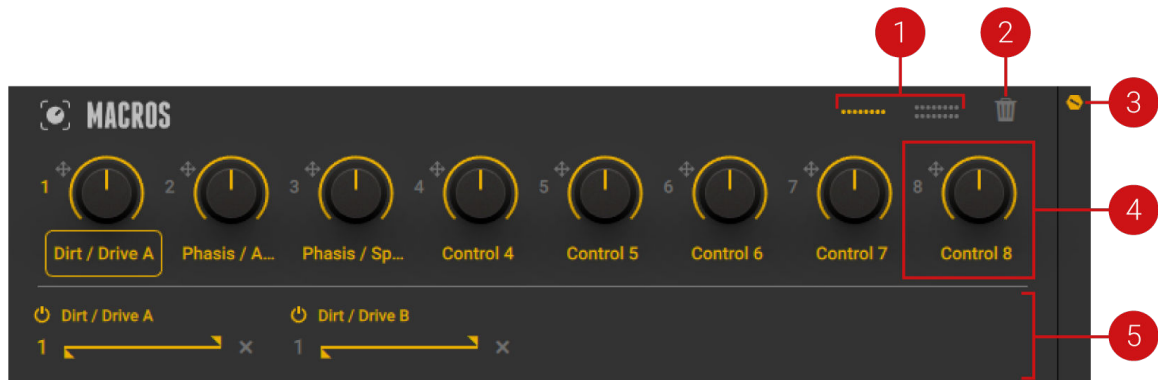
1. **Macro Layout selector**: Switches the number of available Macros between eight and 16.
2. **Clear Macros**: Clears all Macro assignments.
3. **Show Expert panel**: Shows the Expert panel that you can use to display and edit the assignments of the selected Macro.
4. **Macro control**: Controls parameters 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.

11.1.1. Assigning Macros

Macros can be assigned to parameters by using drag and drop. You can assign a single Macro 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.
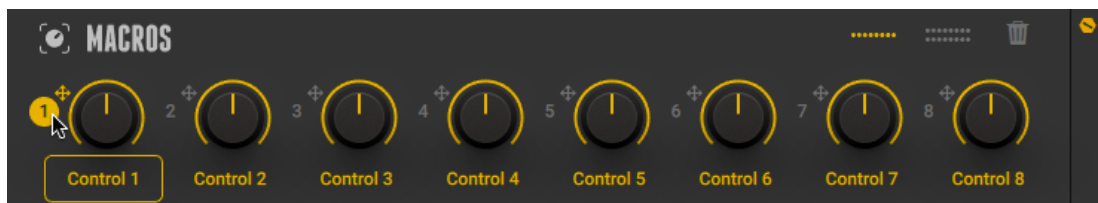
2.   Drag and drop the icon onto the parameter you want to assign the Macro to.

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

11.1.2. 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.

11.2. 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:
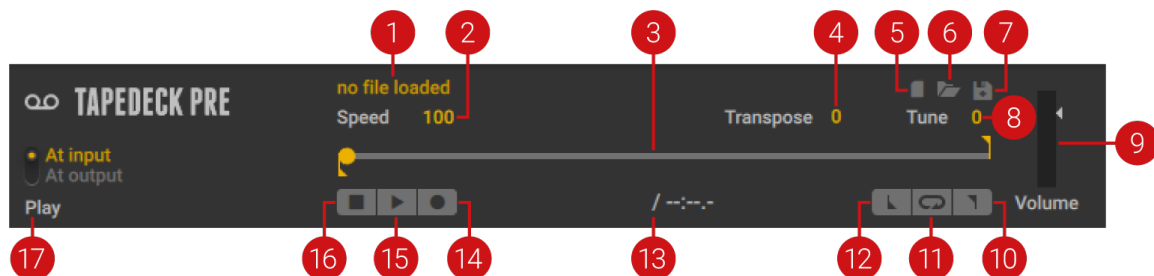
1. **File name**: Displays the file name of the loaded audio file.
2. **Speed**: Adjusts the playback speed of the audio file in the range of 50% to 150%. Time-stretching is used to preserve the original pitch.

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.

11.3. 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:
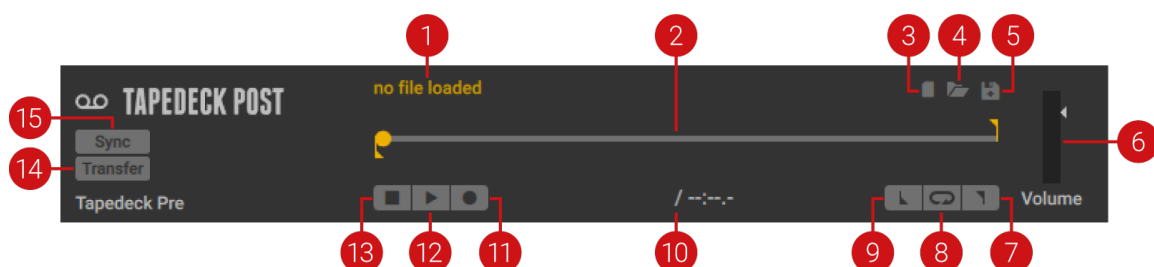
1. **File name**: Displays the file name of the loaded audio file.
2. **Timeline**: Represents the lengths of the audio file. The dot indicates the playback position. You can move the dot along the timeline to adjust the playback position. The downward pointing Start marker indicates the start of the 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.

11.4. 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.
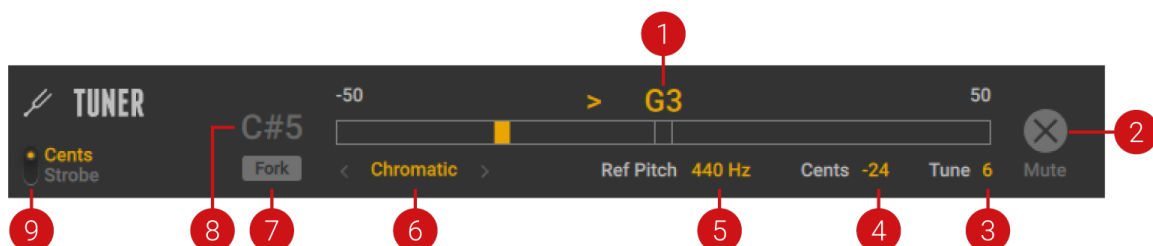
1. **Tuning display**: Shows the tuning of the input signal in relation to musical pitches. The deviation from the ideal tuning is shown either in cent or as a strobe display, depending on the mode set using **Cents/Strobe**.
2. **Mute (in Expert panel)**: Mutes the output of the Tuner, but not the reference 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.

11.5. 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.
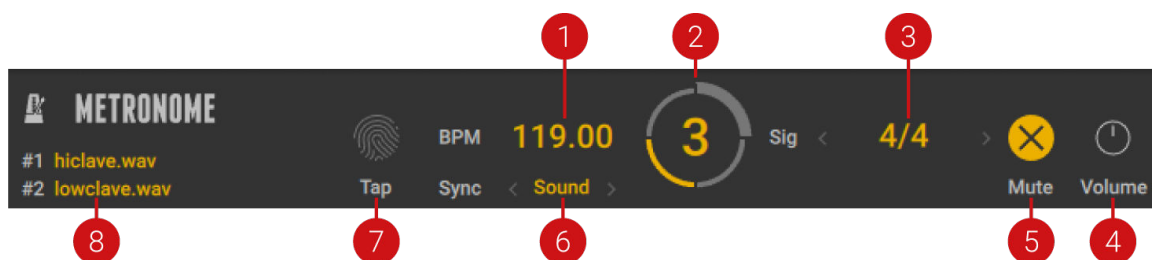
1. **BPM**: Sets the tempo of the Metronome in the range of 20 to 400 beats per minute.
2. **Beat display**: Shows the number of the active beat in the bar. Additionally, the time signature as set with **Sig** is displayed as a circular indicator.

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.

11.6. 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 the 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.

11.7. 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:
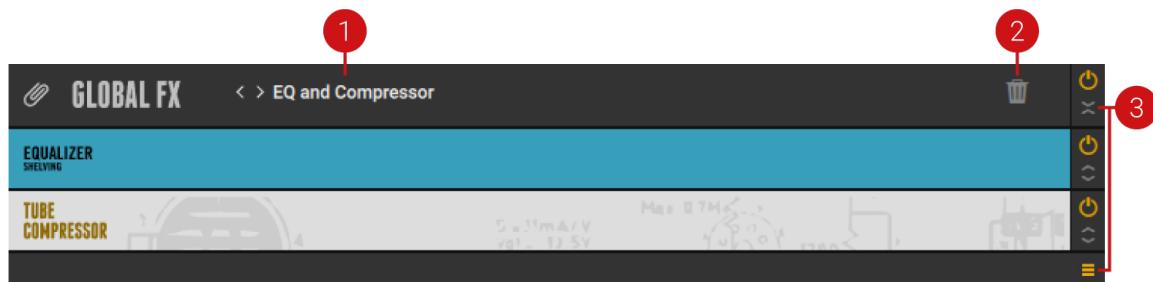
1. **Preset menu**: Recalls and saves presets for the Global FX. This menu can be used in the same way as the Preset menu in Components. For more information, see Using Component Presets.
2. **Clear Global FX**: Clears the contents of the Global FX.
3. **Show Content**: Shows or hides the list of Components 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.

12. 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.

12.1. 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.

12.2. 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:
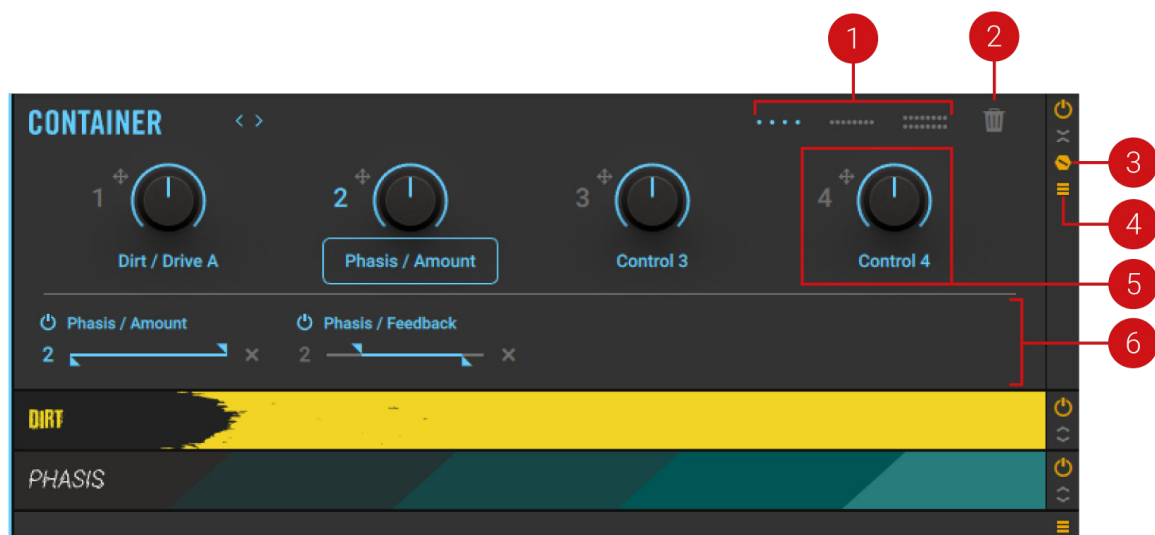
1. **Macro Layout selector**: Switches the number of available Container Macros between four, eight, and 16.
2. **Clear Container**: Clears the contents of the Container and all Container Macro assignments.
3. **Show Expert panel**: Shows the Expert panel that you can use to display and edit the assignments 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.

12.3. 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:
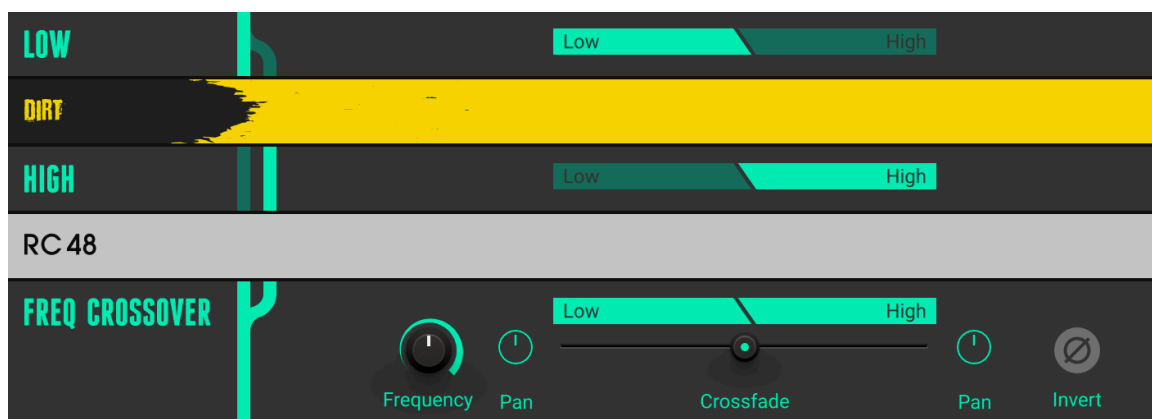
- **Low band**: Feeds the Low band into the Components inserted below.
- **High band**: Feeds the High band into the Components inserted below.
- **Frequency**: Adjusts the crossover frequency at which the frequency spectrum is divided into the Low and High bands.
- **Pan Low**: Adjusts the position 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.

12.4. 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:
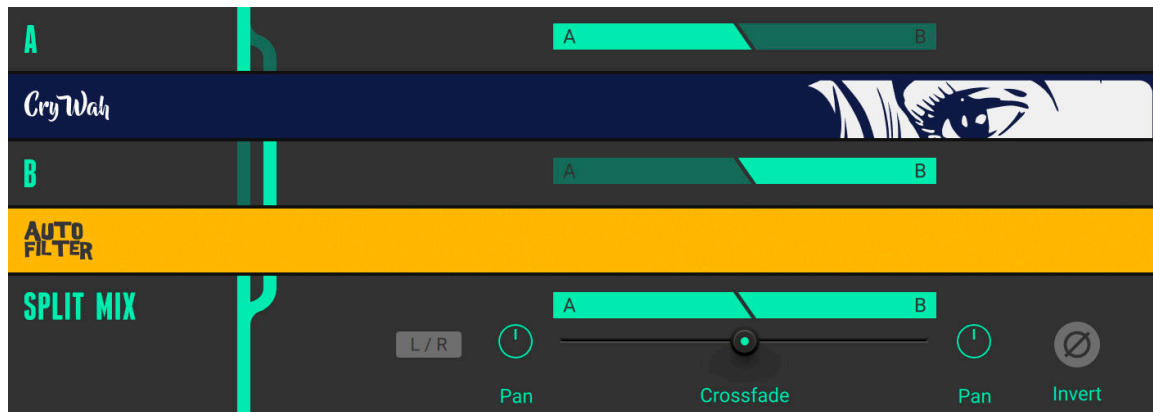
- **A path**: Feeds the A path into the Components inserted below.
- **B path**: Feeds the B path into the Components inserted below.
- **L/R split**: Routes the input's left stereo channel to the A path and the right stereo channel to the B path.
- **Pan A**: Adjusts the position of the A path in the 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.

12.5. 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:
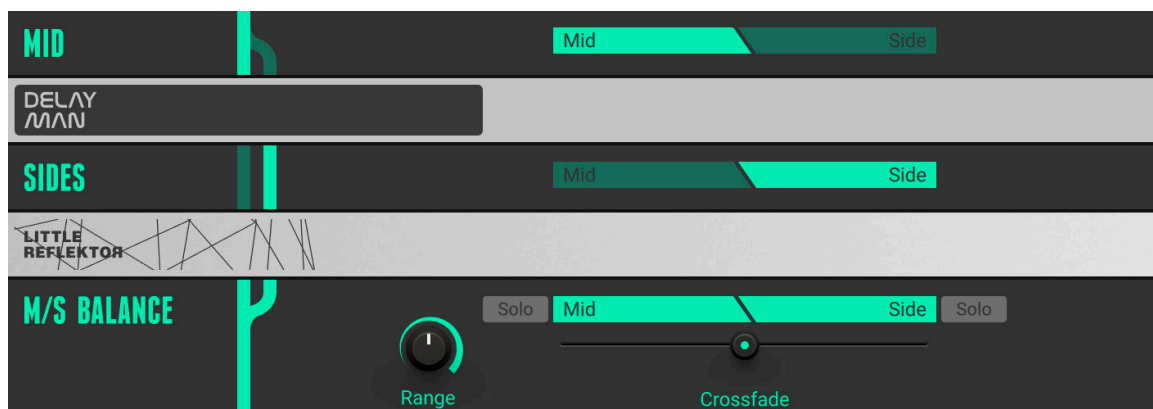
- **Mid band**: Feeds the Mid band into the Components inserted below.
- **Side band**: Feeds the Side band into the Components inserted below.
- **Range**: Adjusts the maximum attenuation of the **Crossfade** on either side of its range. Turning **Range** to the right increases the separation between 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.

13. Modifiers

Modifiers are modulation sources that you can use to control any parameter in the Rack. 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.

13.1. Assigning Modifiers

You can assign Modifiers to any parameter in the Rack.

▶   To assign a Modifier to a parameter, drag and drop the target icon onto the parameter.
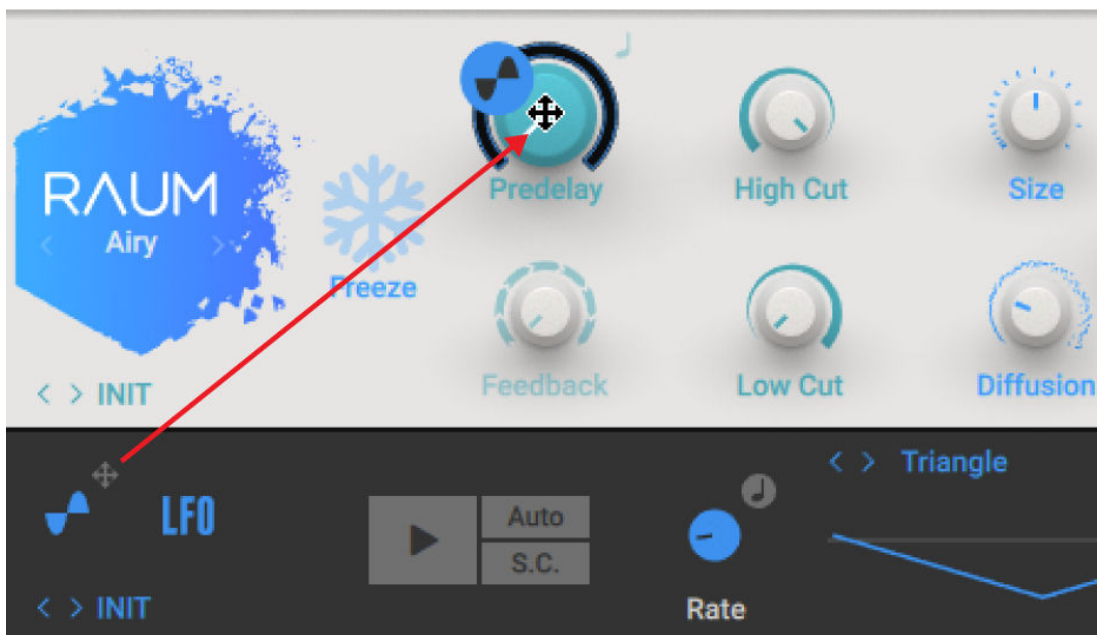
13.2. Analog Sequencer

The Analog Sequencer generates 16 sequential control signals that have rhythmic values of quarter notes to 1/32nd notes. Each control signal´s value can be varied to create a complex rhythmic control signal.

The 16 numbered columns with vertical faders represent steps, or events. These cycle through in sync to the beat of the Metronome. The current position of the sequence is shown by an orange dot moving below the faders. Each step changes the value of the control signal according to its fader's position whereby the midpoint of the fader is neutral, or no control signal is sent. Double-click the fader to return to this position.

This Component contains the following parameters and controls:

- **Resolution** sets the rhythmic value of each of the 16 steps. Available options are 1/4, 1/8, 1/16, and 1/32 notes.
- **Seq Length** adjusts the number of steps.
- **Slide** controls the amount of glissando, i.e. the time the control signal takes to reach the next value.

The Expert panel contains the following parameters and controls:

- Each of the 4 entries in the Expert Panel represents one parameter targeted by the Modifier. You can assign each of the 4 entries to almost any component parameter within the rack.
- The Checkmark preceding the parameter's name means it is set to active. To temporarily deactivate this destination, click on it.
- To permanently remove a destination, click on the **X** (cross) right of the Slider.
- The Slider adjusts the direction and intensity of the effect on the target parameter relative to the control value.

13.3. Envelope

The envelope generates a very flexible control signal, whose shape you can edit accurately using a graphical editor. In most cases (if Auto is activated), the envelope will be triggered with each note you play. Of course, you can also use the Play button to trigger it manually or with a controller.

This Component contains the following parameters and controls:
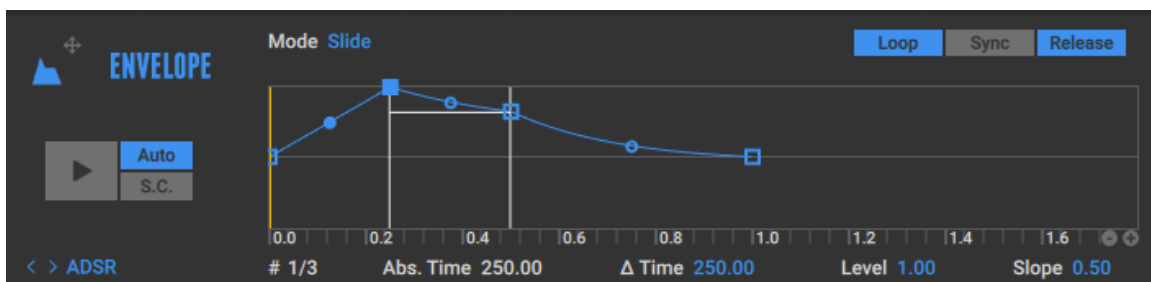
- **Release**
- **Loop** loops the current event
- **Sync** synchronizes the Envelope with the built-in Metronome or external clock.
- **Auto** automatically starts the envelope at the beginning when the input signal is received, this can be manually overridden with the Play button.
- **Sidechain** allows the user to trigger the Envelope with an external source signal. **Auto** must be activated to use **Sidechain**.

In the central graph, you see a blue line representing the development of the control signal (vertical axis) over time (horizontal axis). An orange vertical line indicates the position of the current control signal.

The default envelope consists of four small squares (called breakpoints or nodes) setting the vertices of the control signal at a certain time after the envelope has been triggered. You can move them up or down to increase or decrease the control signal, and left or right to change their point in time.

- The first and last nodes are tied together, because the envelope always starts from where it ended.
- If **Sync** is activated, the nodes will snap to a grid representing the beat.
- To add a node: Right-click (Mac: ctrl-click) anywhere on the envelope curve. To remove a breakpoint: Right-click (Mac: ctrl-click) on it. By dragging the small dot between nodes, you can bend the segment's shape into concave and convex form.
- The left-most node is where the envelope starts when triggered.
- If **Loop** is inactive, the control signal will follow the Envelope once and then reside on the final/ initial value until a new trigger signal is received.
- If **Loop** is active, the sustain/loop segment of the envelope is repeated as long as a triggering signal is present. Vertical lines crossing two of the central nodes define this segment. When repeated, the node on the left is ignored, as the loop starts again from the value of its ending node (shown by a phantom node being the origin of a second blue line). If there are only two nodes in the sustain/loop segment, the left of those lines is grey. Consequently, the control signal then resides on the level reached at the second line for the length of the loop.
- If there are three or more central nodes (other than the start and end), the sustain/loop segment can be shifted by clicking on one of its defining lines and dragging it to another node. When dragged, so that the sustain/loop segment consists of only two nodes, the control signal resides on the level reached at the second line for the length of the loop.
- As soon as the envelope is not triggered anymore, the envelope continues from the second vertical line, if release is active. The control signal then resides on the final/initial value until triggered again. If release is inactive, the envelope will be followed until the right vertical line, no matter how long a triggering signal is present. The part right of the second vertical line is ignored and the control signal resides on the sustain level until the envelope is triggered again.

The Parameter Strip above the envelope provides a variety of data and additional parameters to edit.

- At the left of the strip, the value below **#** shows which of the nodes is being edited.
- Mode offers two options: **Slide** and **Fixed**. If you move a node left or right in Slide mode, the envelope right of it moves as well to maintain the same times and levels. In Fixed mode, the total envelope time doesn't change.
- **Abs. Time** shows the amount of time in seconds from the start of the envelope to the breakpoint being edited.
- **Delta Time** shows the duration in seconds from the node currently edited to the one left of it.
- Level shows the level of the node currently edited in reference to the centerline.
- **Slope** shows the state of the line arriving at the currently edited node. 0.5 indicates a straight line. 0.999 indicates a maximally convex curve. 0.001 indicates a maximally concave curve.

The ruler below the envelope is scaled in seconds if **Sync** is deactivated, in note values if **Sync** is activated. If the envelope extends past the envelope's visible range, click on the ruler and drag to the left or right to scroll. Double-click on the ruler to fit the envelope exactly within the visible window.

To change the range of the ruler by zooming in or out, click on the **(+)** and **(-)** symbols in the lower right. This also sets the quantization time for nodes if **Sync** is activated.

The Expert panel contains the following parameters and controls:

- Each of the 4 entries in the Expert Panel represents one parameter targeted by the Modifier. You can assign each of the 4 entries to almost any component parameter within the rack.
- The Checkmark preceding the parameter's name means it is set to active. To temporarily deactivate this destination, click on it.
- To permanently remove a destination, click on the **X** (cross) right of the Slider.
- The Slider adjusts the direction and intensity of the effect on the target parameter relative to the control value.

13.4. 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:
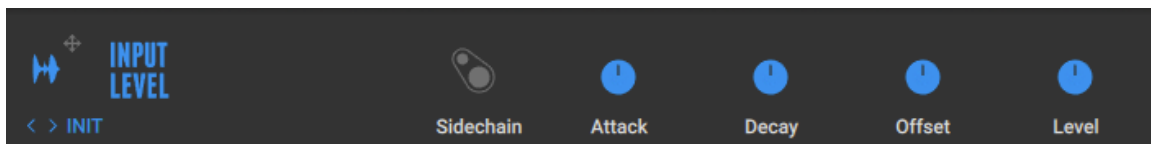
- **Volume** sets the overall strength of the control signal.
- **Offset** sets the control signal's minimum value. By default, the control signal covers a range from −1 to +1, with 0 as the midpoint. When Offset is at maximum, the baseline is 0.
- **Attack** sets how long the control signal takes to 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).
- **Sidechain** allows the user to trigger the Envelope with an external source signal. **Auto** must be activated to use **Sidechain**.

The Expert panel contains the following parameters and controls:

- Each of the 4 entries in the Expert Panel represents one parameter targeted by the Modifier. You can assign each of the 4 entries to almost any component parameter within the rack.
- The Checkmark preceding the parameter's name means it is set to active. To temporarily deactivate this destination, click on it.
- To permanently remove a destination, click on the **X** (cross) right of the Slider.
- The Slider adjusts the direction and intensity of the effect on the target parameter relative to the control value.

13.5. 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:
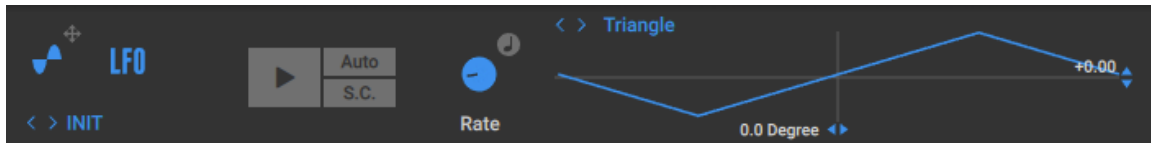
- **Rate** sets the LFOs frequency, from 0.01Hz (oscillations per second) to 10.24Hz. When the component is synchronized (see below), the scale turns from Hz into note values.
- **Tempo Sync** synchronizes the Loop Machine to the metronome or host, depending on the Sync setting.
- **WAVEFORM** provides a graphic display of the control signal. By clicking on the arrows on the right side of the graph you can choose among sine, triangle, square, saw tooth, and random waveforms.
- **Polarity** inverts the waveform's polarity and thereby all its value changes with one click.
- **Start Phase** sets the point within the waveform hitting the Play button jumps to. The actual value depends on the waveform. With a sine wave at 0 degree (Start Phase at center), the value of the control signal starts at 0 (neutral) and increases first. At 180 degrees (Start Phase turned fully clockwise), the value of the control signal starts at 0 and decreases first.
- The Play button retriggers the LFO, which otherwise keeps on running without interrupting. To automatically trigger this button by hitting a note, activate the small **Auto** button above the Play button.
- **Sidechain** allows the user to restart the Envelope with an external source signal. **Auto** must be activated to use **Sidechain**.

The Expert panel contains the following parameters and controls:

- Each of the 4 entries in the Expert Panel represents one parameter targeted by the Modifier. You can assign each of the 4 entries to almost any component parameter within the rack.
- The Checkmark preceding the parameter's name means it is set to active. To temporarily deactivate this destination, click on it.
- To permanently remove a destination, click on the **X** (cross) right of the Slider.
- The Slider adjusts the direction and intensity of the effect on the target parameter relative to the control value.

13.6. 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:
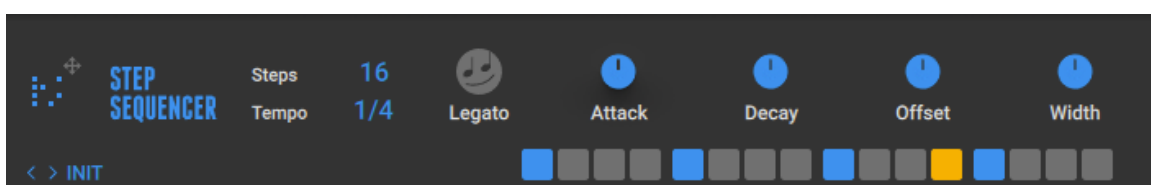
- **Resolution** sets the rhythmic value, effectively changing the speed at which the buttons are triggered. Available options are 1/4, 1/8, 1/16, and 1/32 notes.
- **Seq Length** adjusts the length of the sequence by reducing the number of steps.
- **Legato** ties adjacent steps to each other, thus 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)
- **Width** sets the length of the control signal, when triggered. When fully clockwise, the width equals the note-value set as Resolution.
- **Offset** increases all control signal values as the control when turned up.

The Expert panel contains the following parameters and controls:

- Each of the 4 entries in the Expert Panel represents one parameter targeted by the Modifier. You can assign each of the 4 entries to almost any component parameter within the rack.
- The Checkmark preceding the parameter's name means it is set to active. To temporarily deactivate this destination, click on it.
- To permanently remove a destination, click on the **X** (cross) right of the Slider.
- The Slider adjusts the direction and intensity of the effect on the target parameter relative to the control value.

14. 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 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 both in 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.

14.1. 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.

14.2. 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.
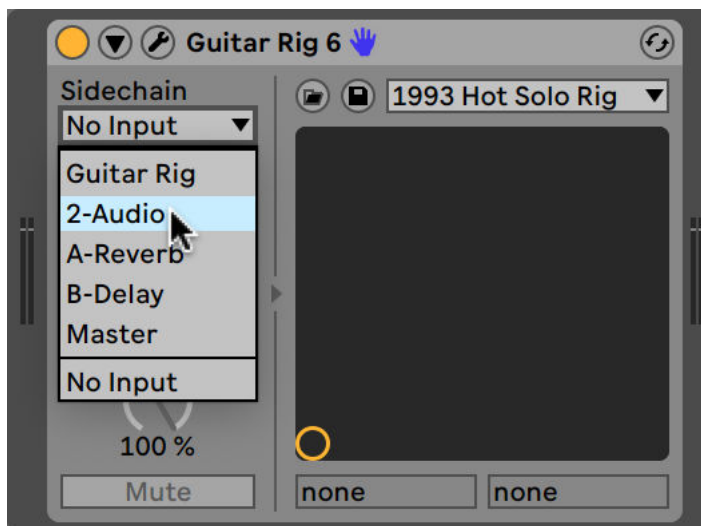
→ The signal received from the chosen input track is now available as global sidechain signal in GUITAR RIG.

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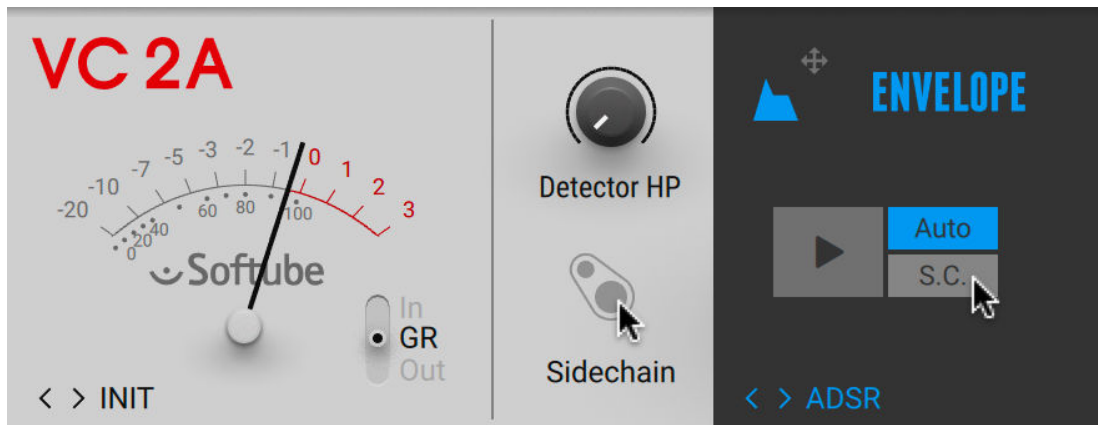
→ The internal control signal of the Component is replaced with the global sidechain signal.

15. 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] + [,]
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 Info pane	[command] + [I]	[Ctrl] +[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]

16. 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.

16.1. 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 amplifiers 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 additional controls that further expand the sonic possibilities.

16.1.1. 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 input level, or gain, of the Normal channel. The Bass and Treble controls have no effect in this channel.
- **Brilliant**: Adjusts the input level, or gain, 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 for presets made in GUITAR RIG 5. Deactivating **GR5 Mode** allows for a deeper tremolo effect.

16.1.2. Bass Pro

Bass Pro models a gritty and growling 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:
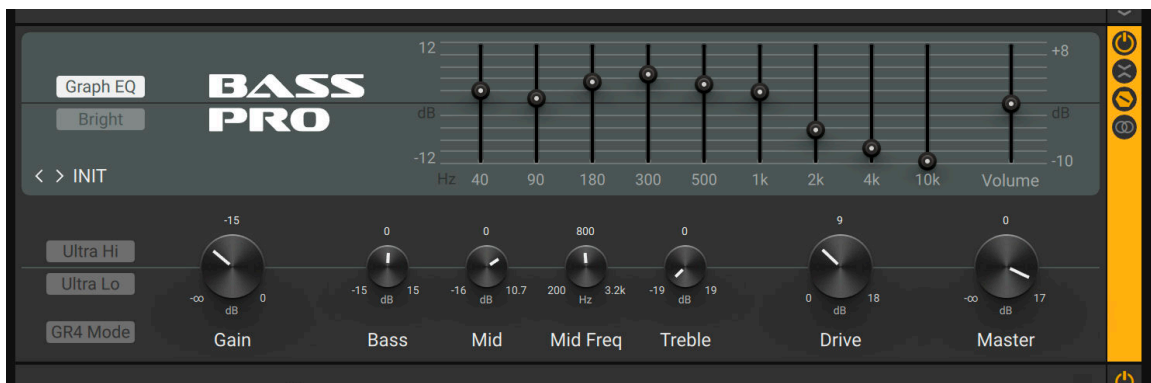
- **Graph EQ**: Activates the graphic equalizer in the top panel of the amplifier.
- **Bright**: Boosts high-frequency content.
- **Graphic equalizer**: Boosts or cuts nine specific frequency bands: 40 Hz, 90 Hz, 180 Hz, 300 Hz, 500 Hz, 1 kHz, 2 kHz, 4 kHz, and 10 kHz. Moving a slider up boosts the respective 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 amplifier.

The Expert panel contains the following parameters and controls:

- **GR4 Mode**: Activates backwards-compatibility for presets made in GUITAR RIG 4.

16.1.3. Citrus

Citrus models the 70s British sound. 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 high-pass 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 amplifier.

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.

16.1.4. 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 mixed for a great variety of sounds.

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.
- **Normal**: Adjusts the input level, or gain, of the Normal channel.
- **Brilliant**: Adjusts the input level, or gain, 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 amplifier.

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.

16.1.5. 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:

- **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 amplifier.

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.

16.1.6. 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.

16.1.7. 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 amplifier.

16.1.8. 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 amplifier.

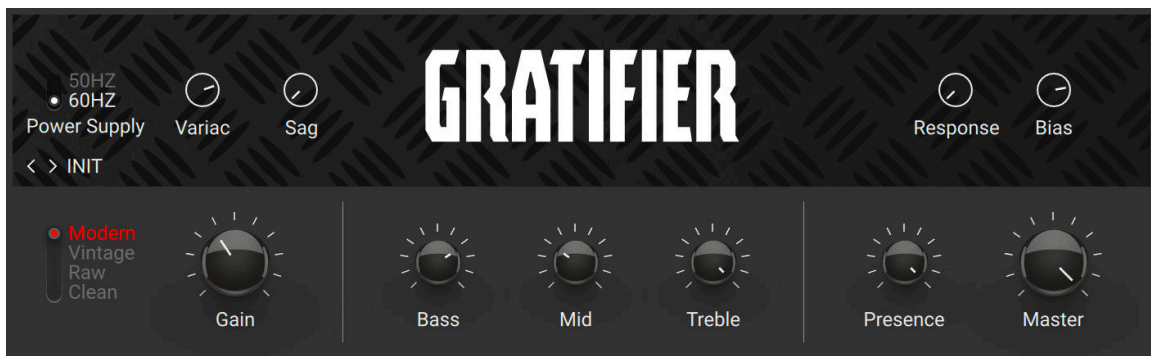The Expert panel contains the following parameters and controls:

- **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.

16.1.9. Lead 800

Jump models the sound of a clean British amplifier. Other than Plex, which is suitable for both rhythm and lead guitar tones, Jump'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 amplifier.

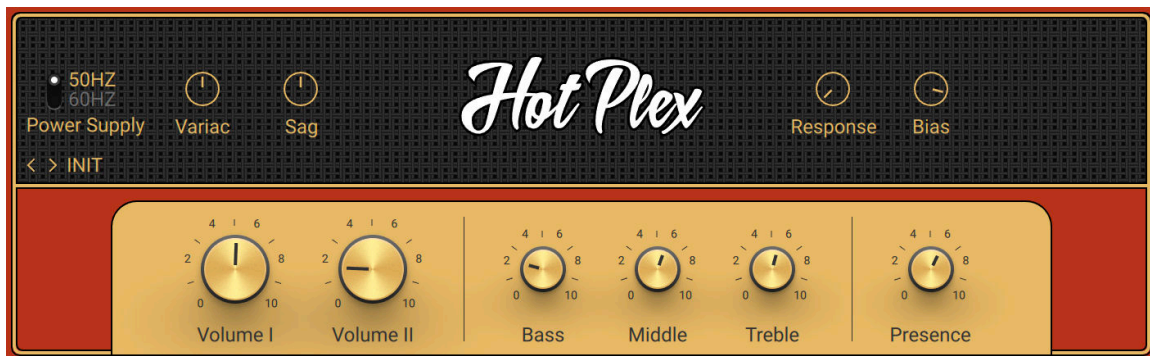The Expert panel contains the following parameters and controls:

- **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.

16.1.10. 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 input level, or gain, of the Bright channel.
- **Normal**: Adjusts the input level, or gain, 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 for presets made in GUITAR RIG 5. Deactivating **GR5 Mode** allows for a deeper tremolo effect.

16.1.11. 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 amplifier.

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.

16.1.12. Jump

Jump models the sound of a clean British amplifier. 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 amplifier.
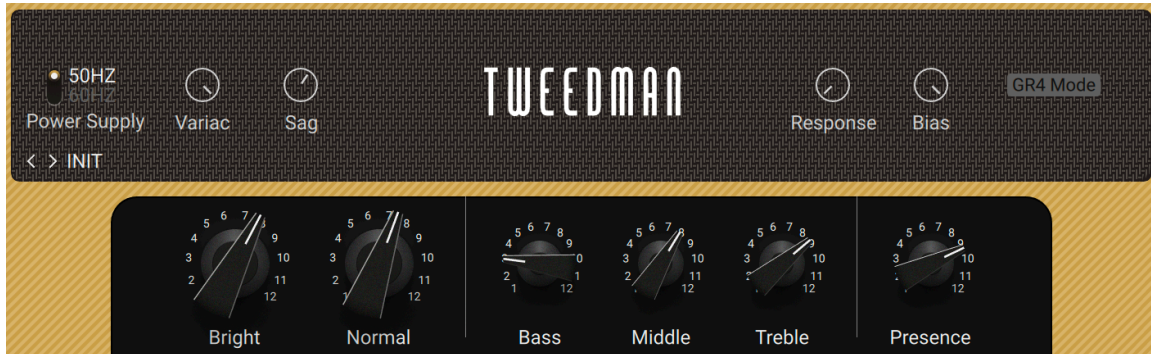
The Expert panel contains the following parameters and controls:

- **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.

16.1.13. 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 input level, or gain, of the Bright channel.
- **Normal**: Adjusts the input level, or 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.

16.1.14. Van 51

Van 51 models the sound of a benchmark high-gain amplifier. It provides a wide range of raw and edgy guitar tones.

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.
- **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 amplifier.

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.

16.1.15. Plex

Plex models a clean vintage 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.

16.1.16. 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 amplifier.
- **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.

16.2. Cabinets

No amp setup is complete without a cabinet—it plays an essential role in the final tone of the amplifier. With GUITAR RIG 5, you have the choice between many different levels of complexity. The sound can be straightforward as in the Matched Cabinet, or involved as in the all-new Control Room Pro.

• When adding any amp to your rack, it will automatically be provided with a Matched Cabinet. This can then be replaced by another Cabinets component at any time.

16.2.1. Control Room

The Control Room

About

Choose among five outstanding guitar cabinets and up to 8 classic microphones to create a unique blend of tonal characteristics. The modeled setup is the outcome of decades of guitar recording experience: each cabinet is paired with perfectly chosen and positioned microphones which are all in phase, so you can mix them as you please. Be sure to check out the Component Presets, as they offer some classic tones that can be used as a starting point.

Controls

The Cabinet Model Selector on the right side of the component shows a small picture of the current cabinet model. Click on the arrows below to switch between them.

The mixing console offers 6-8 channel strips, each consisting of the same set of controls:

• The Headline gives an indication of the type of microphone used.
• The **L/R** knob controls the panning of this microphone's signal, i.e. the position in the stereo panorama.
• The fader controls the volume of this channel; all faders together determine the mix.
• **M**(ute) and **S**(olo) are standard controls of a mixing desk. **M** mutes the channel, S soloes the channel.
• The **CONTROL ROOM** offers some master controls to shape the sum of all the signals:
• **AIR** controls the level of early reflections picked up by the mic, simulating the response of the room.
• **BASS** boosts or cuts the level of the lower frequencies.
• **TREBLE** boosts or cuts the level of the higher frequencies.
• **VOLUME** controls the master volume for all microphones.
• **L**(earn): After pressing the small button right of the **VOLUME** knob, the output volume is "learned" automatically: play as loud as you can for a few seconds. Once the volume has been adjusted to the maximum while avoiding clipping, the learn function will automatically turn off.
• **STEREO** activates true stereo processing for this component.

16.2.2. Control Room Pro

The Control Room Pro

About

The next step in the Control Room concept is here! With 29 cabinets and a Direct Injection (DI) box, Control Room Pro offers an unlimited set of sounds in control of the recording process. A multitude of microphones are also on offer from the old school to high tech modern, while placement, phase, and room can be manipulated.

Controls

The initial screen shows Channel One and includes three selector areas: selector area one is the cabinet; selector area two is the microphone; selector area three is the mic placement on the speaker.

- Control Room Pro presets are available from the top pull-down menu. The procedure to save user presets is explained further in the Application Reference.
- **L**(earn): After pressing the small button right of the **VOLUME** knob, the output volume is "learned" automatically: Just play as loud as you can for a few seconds. Once the volume has been adjusted to the maximum while avoiding clipping, the learn function will automatically turn off.
- The cabinet, microphone, and placement can all be selected by using the arrows to the right of their photo. The upper arrow shows a list of available options, while the up/down arrows allow you to toggle between the assorted elements.
- As well, there are two sliders above the selector areas. The left most slider controls an adjustable phase delay. In this case, phase means that all frequencies are delayed the exact same amount. Use this parameter to correct phase issues when using multiple microphones. The button when depressed inverts the phase.
- The second slider is Room. This function controls the mix between a room mic and the selected close mic in this channel — the M button mutes the Room signal.
- At the right side of the screen is a mixing fader to determine the output level as well as a pan pot for placement in the stereo field and mute and solo buttons.

This scenario is repeated 8 times. Press any of the 8 buttons on top of the windows to add an entirely new cabinet and microphone to the mix.

The final button labeled mixer does just that: similar to the mixer in the original Control Room, it allows the user to manipulate all the levels of the chosen microphones for a master mix.

The Control Room Pro controls

Controls

This is essentially similar to the mixer window in Control Room. All the familiar console controls are present including Mute, Solo and Left/Right pan. As well, a simple EQ is included with Bass and Treble, while Air controls the amount of early reflections picked up by the mic, simulating the response of the room. The button below Bass with two connected circles enables true stereo input for the Control Room Pro.

- **M** mutes the selected channel.
- **S** soloes the selected channel.
- **BASS** adds or decreases low frequency content from the signal.
- **TREBLE** adds or decreases high frequency content from the signal.
- **AIR** adds high-mid frequency presence to the signal.
- **L/R** controls the position in the stereo field.
- **STEREO** allows for the mix to be in true stereo.

16.2.3. Matched Cabinet

The Matched Cabinet

About

When adding any amplifier to the rack, a Matched Cabinet is automatically included below it. It provides a cabinet fitting the amp you chose, as well as two mic settings that can be blended at any ratio. The Dry/Air control allows for a larger room sound.

Controls

- The Cabinet selector switches between the different models of the Matched Cabinets, allowing them to be combined with any of the amps.
- The **A**/**B** Mix slider blends between the two microphones fixed on the cabinet. The general characteristics of both are opposed, giving an edgy and a mellow option. You can mix between the two to achieve your desired blend.
- **VOLUME** sets the volume of the matched cabinets output.
- **VOLUME LEARN**: Automatically set the best output volume by activating this button and playing as loud as you can. GUITAR RIG 5 will select the maximum volume while avoiding clipping. Once the volume has been adjusted, the learn function will automatically turn off.
- **DRY/AIR** controls the amount of early reflections picked up by the microphone, simulating the response of the room.
- **STEREO** activates true stereo processing for this component.

16.3. 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!

16.3.1. Delay Man

The Delay Man

About

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.

16.3.2. Psychedelay

The Psychedelay

About

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.

16.3.3. Quad Delay

The Quad Delay

About

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.

16.3.4. Tape Echo

The Tape Echo

About

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.

16.3.5. 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:
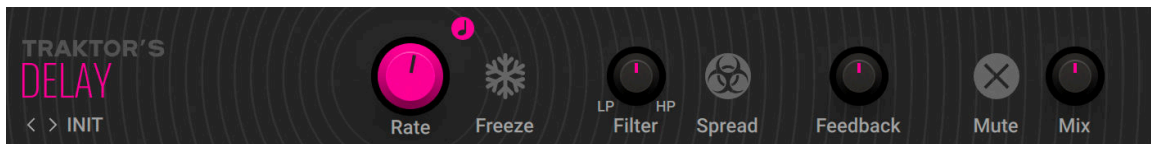
- **Rate**: Adjusts the delay time.
- **Sync**: Synchronizes the delay time to the metronome. When **Sync** is active, **Rate** can be set in musical intervals relative to the tempo of the metronome. The interval can be set to 1/32, 1/16, 1/8, 3/16, 1/4, 3/8, and 4/4 notes.
- **Freeze**: Freezes the 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.

16.3.6. Twin Delay

The Twin Delay

About

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.

16.4. Distortion

Distortion, historically one of the first guitar effects, is still the essential element of many popular guitar sounds today. Generally inserted between guitar and amp, distortion effects offer a wide variety of flavors from simple overdrive to square wave fuzz.

The distortion and overdrive components of GUITAR RIG 5 are accurate reproductions of classic hardware devices. Therefore, just like the originals, these effects can sound somewhat thin and artificial by themselves. Use them with an amp/cabinet combination to create a stronger sound.

16.4.1. Big Fuzz

The Big Fuzz

About

Sustain to the max like a 1970s guitar hero or add some serious grunge to your tone. This is the distortion of choice for some serious rock and roll sounds.

Controls

- **VOLUME** is the master volume control for this component.
- **SUSTAIN** controls the amount of gain.
- Turn **TONE** clockwise to emphasize low frequencies and reduce higher frequencies, or counter-clockwise to attenuate the bass range and enhance the high frequencies.

Expert Controls

- The **BASS** knob adjusts the low frequency response.
- The **TREBLE** knob adjusts the high frequency response.
- **STEREO** activates true stereo processing for this component.

16.4.2. 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.

16.4.3. Cat

The Cat

About

If you want a responsive distortion for blues and rock (rhythm or lead), the Cat is your component of choice.

Controls

- **VOLUME** is the master volume control for this component.
- **FILTER** affects the sound's color. For a darker sound, turn clockwise to enhance the low frequency range; turn counter-clockwise for a brighter, sharper sound.
- **DISTORTION** controls the amount of distortion applied.

Expert Controls

- Turn the **BALLS** control clockwise to add low-end punch. Turning it counter-clockwise creates a flatter, more biting sound.
- The **BASS** knob adjusts the low frequency response.
- The **TREBLE** knob adjusts the high frequency response. In the low end of the range (from 0.00 to 3.00) it can add a wah-wah like effect.
- **TONE** adjusts the frequency range influenced by the built-in pre-distortion midrange booster.
- **STEREO** activates true stereo processing for this component.

16.4.4. 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:
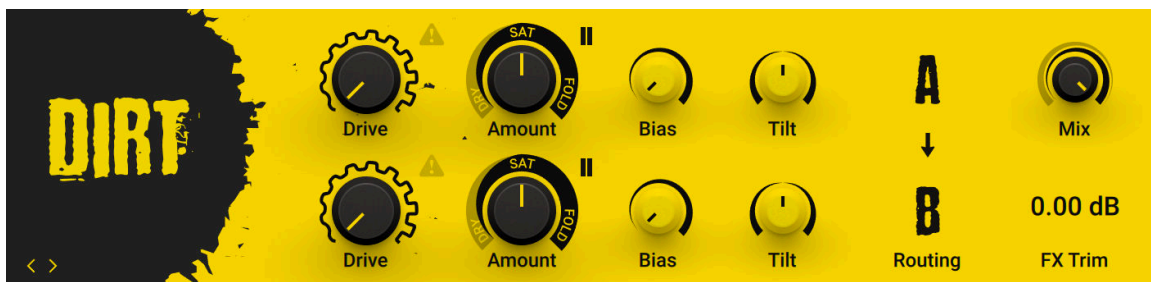
- **Drive**: Controls the amount of input gain, increasing the intensity of the distortion. The extra button at the top right of the control switches off the gain compensation that keeps consistent loudness levels. This can be used, for example, to drive a signal into the second stage at higher distortion 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.

16.4.5. Demon Distortion

The Demon Distortion

About

Definitely a piece of hard rocker's gear, the Demon Distortion provides you with razor-sharp, long-sustaining leads and speaker-blasting rhythm sounds.

Controls

- **VOLUME** is the master volume control for this component.
- The **BASS** knob adjusts the low frequency response.
- The **MID knob** adjusts the midrange frequency response.
- The **TREBLE** knob adjusts the high frequency response.
- **GAIN** controls the amount of distortion.
- The **SCOOP** switch drastically attenuates the midrange frequencies, which produces a typical modern metal sound. With **Scoop** on, the **Mid** control has no effect.
- A red LED will light when a signal has been detected.

Expert Controls

- The **BOTTOM** knob controls how much bass passes through to the distortion section. If the distortion lacks definition, lower this knob to tighten the sound.
- **BOOST** sets the center frequency of the built-in pre-distortion midrange booster.
- **BASS** determines the center frequency of the equalizer's Bass control.
- **MID** determines the center frequency of the equalizer's Mid control.
- **TREBLE** determines the center frequency of the equalizer's treble control.
- **BRIGHT** fine-tunes the signal's presence by attenuating or enhancing the upper midrange frequencies.

- **STEREO** activates true stereo processing for this component.

16.4.6. Distortion

The classic Distortion

About

This one is a classic, instantly recognizable distortion that has featured in countless live shows and recordings.

Controls

- **VOLUME** is the master volume control for this component.
- Turning **TONE** clockwise accents the midrange while dropping the bass. Counterclockwise takes off the highs and boosts the bass for a warmer sound.
- **DISTORTION** adds dirt to the sound.

Expert Controls

- The **BASS** knob adjusts the low frequency response.
- The **MID** knob adjusts the midrange frequency response.
- The **TREBLE** knob adjusts the high frequency response.
- **STEREO** activates true stereo processing for this component.

16.4.7. Fuzz

The Fuzz

About

Looking for that special 60's fuzz sound? The Fuzz's thick and harmonically rich tones will make your leads cut through a mix. As a bonus, you can create buzzing vintage rhythm guitar sounds, too.

Controls

- **VOLUME** is the master volume control for this component.

- A red LED lights when a signal has been detected.
- The **FUZZ** controls the amount of gain into the circuit.

Expert Controls

- The **BASS** knob adjusts the low frequency response.
- The **TREBLE** knob adjusts the high frequency response.
- **STEREO** activates true stereo processing for this component.

16.4.8. Gain Booster
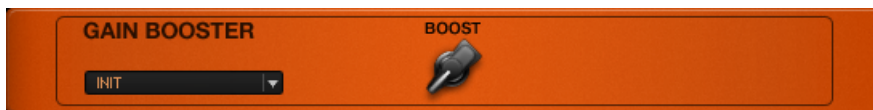
The Gain Booster

About

This component can add gain where you need it- for insane amounts of overdrive, or to compensate for settings that reduce level.

Controls

- **BOOST** sets the degree of gain boosting.

16.4.9. Mezone

The Mezone

About

This component delivers pure Metal. Pre- and post-distortion tone controls let you tailor the sound.

Controls

- **VOLUME** is the master volume control for this component.
- The **BASS** knob adjusts the low frequency response. It is especially powerful in this component.
- **MID** controls the amount of the Mezone's midrange frequency boost.
- **MIDFREQ** sets the center frequency for the Mid boost.
- The **TREBLE** knob adjusts the high frequency response.
- **DISTORTION** controls the amount of gain.

Expert Controls

The Expert controls provide filters applied before the distortion stage, greatly influencing the distortion's character.

- **BASS** controls how much bass is passed through to the distortion.
- **MID-Q** controls the amount of the midrange boost.
- **MIDFREQ** sets the center frequency for the Mid boost.
- Turning **TREBLE** counter-clockwise reduces high frequencies sent to the distortion section.
- **STEREO** activates true stereo processing for this component.

16.4.10. 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.

16.4.11. Sledgehammer

The Sledgehammer

About

If you want your guitar to clear its way through a mix, try the Sledgehammer. This effect is not subtle and will give you as much presence as you need.

Controls

- **VOL** (volume) is the master volume control for this component.

- The **BASS** knob adjusts the low frequency response.
- The **TREBLE** knob adjusts the high frequency response.
- **CONTOUR** attenuates, or scoops, a frequency band determined by the **FREQ** control. When it is completely turned down, the **FREQ** knob has no effect.
- The **FREQ** (frequency) knob sets the center of the frequency range attenuated by the **CONTOUR** knob, from midrange to treble.
- **GAIN** controls the amount of distortion.
- The **DIST** (distortion) / **DRIVE** switch changes the distortion characteristics from softer overdrive to harder, more edgy tones.

Expert Control

- **STEREO** activates true stereo processing for this component.

16.4.12. 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:
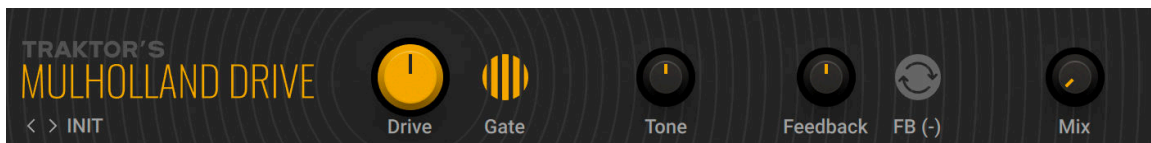
- **Drive**: Distributes the input signal to the two internal overdrive units. When turned fully left or right, the input signal is only sent to either unit. In center position, the input signal is equally sent to both units.
- **Gate**: Activates a noise gate that suppresses sustained feedback sounds.
- **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.

16.4.13. Skreamer

The Skreamer

About

This warm mid-range enhancing overdrive works great for rhythm guitar and smooth leads.

Controls

- **VOLUME** is the master volume control for this component.
- Turning **TONE** clockwise gives bright, screaming leads and biting rhythm tones. Counterclockwise gives a mellower, darker sound.
- **DRIVE** controls the "crunch factor". Turn clockwise for more distortion.
- A red LED lights when a signal has been detected.

Expert Controls

- The **BASS** knob adjusts the low frequency response.
- The **BRIGHT** knob adjusts the high frequency response.
- **CLEAN** mixes the unprocessed sound with the distortion.
- **STEREO** activates true stereo processing for this component.

16.4.14. TransAmp

The TransAmp

About

This versatile distortion set the stage for the emulation of different amplifiers in a stomp box. TransAmp delivers a wide variety of tones- from cool sounds on the verge of distortion to biting, raging heat.

Controls

- **VOLUME** is the master volume control for this component.
- The **BASS** knob adjusts the low frequency response.
- The **TREBLE** knob adjusts the high frequency response.
- **DRIVE** controls the amount of overdrive, from slight crunch to hardcore distortion.
- The **AMP** control blends between three classic amp characters: **TWEED**, **BRIT** (British), and **CALIF** (California).

Expert Controls

- The **CLEAN** switch reduces overall gain, producing interesting variations of the amp types.
- **CAB&MIC** activates speaker cabinet and micing simulation.
- **MICPOS** has an effect only if the **CAB&MIC** function is enabled. It places the mic at an off-axis position of a variable degree when turned counter-clockwise, and at a variable distance from the speaker when turned clockwise.

- **HOT** has an effect only if the **CAB&MIC** function is enabled. Turning the control more clockwise produces a brighter, hotter sound.
- **STEREO** activates true stereo processing for this component.

16.4.15. Treble Booster
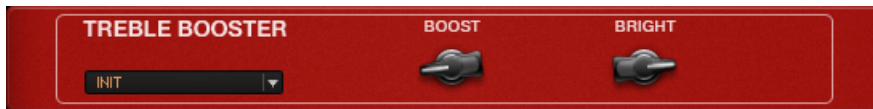
The Treble Booster

About

Treble Boosters were used by Brian May from Queen, as well as Eric Clapton to push their tube amps into creamy distortion. This effect cuts low-end frequencies then amplifies the signal, creating a focused sound with more midrange emphasis.

Controls

- **BOOST** controls the amount of gain.
- The **BRIGHT** control acts as a low-pass filter to attenuate the highest frequency range.

16.5. Dynamics

Dynamic effects are capable of much more than just simple loudness control! In GUITAR RIG 5, you'll find a range of tools to dynamically or interactively adjust levels. Create an incredible range of sounds with practical and even psychoacoustic properties.

16.5.1. Tube Compressor

The Tube Compressor

About

Tubes are always known to give the most harmonically pleasing sound; in a compressor their character is brought to the forefront.

Controls

- **INPUT** sets the level of the signal at the input of the compressor. Turning this control up too much can lead to distortion.
- **THRESHOLD** sets the level above which signals will be compressed.

- **RATIO** controls how much signal is attenuated when the compressor is triggered. The higher the ratio, the greater the amount of compression, and the more dynamics get evened out.
- **ATTACK** sets the time it takes for the compressor to engage. A longer attack time retains more of a signal's original dynamics.
- **RELEASE** sets the time it takes for the compressor to return to its normal state after the signal falls below the threshold. With very short release times, the compressor tracks even slight level changes, possibly producing a choppy sound.
- **GAIN** controls the amount of amplification of the compressed signal.

Expert Controls

- **SATURATION** adds some distortion to the tube characteristics.
- **KNEE** adjusts the compressor response from soft knee (gentler compression effect) to hard knee (more squeezed, harder-sounding compression)
- **DYNAMIC** controls the dynamic response of the virtual tube used in this component. Turning it up flattens dynamics a bit.
- **SIDECHAIN** allows the compressor to be triggered by an external signal. When lit, SideChain is enabled. In stand-alone mode, you can trigger the compressor with an external signal via the Audio and MIDI Settings' routing panel. See the Guitar Rig 5 Application Reference for further instructions. When run as a plug-in, set up the triggering of the compressor via the channel routing function of your DAW / host software. See the relevant documentation for further instructions.

16.5.2. Limiter

The Limiter

About

A limiter works like a compressor with a high compression ratio and a short attack time. Its main use is to prevent subsequent stages from overloading, but it can also be used for tone shaping.

Controls

- **VOLUME** controls the master volume of this component.
- **LIMIT** sets the level at which the limiter is triggered, i.e. the level input signals cannot exceed.
- **HOLD** sets a minimum time that limiting will be applied when the signal exceeds the Limit threshold.
- **RELEASE** sets the time it takes to return to normal state after limiting. Longer release times will sound smoother, but they can cause unwanted volume fluctuations.

16.5.3. Noise Gate

The Noise Gate

About

A noise gate helps remove noise and hiss, but can also be used as a special effect. Although there is a simple gate at the input stage of GUITAR RIG 5, the Noise Gate component can be useful for more sophisticated applications.

To kill hum and buzz, a noise gate mutes all signals below a certain level (threshold). While you're playing, the musical part of your signal is much stronger than the noise and practically renders it inaudible. When you pause, however, the noise will still be there and get all the attention. At that point, a noise gate turns down the volume to create silence.

Controls

- **THRESHOLD** determines the reference level above which the gate opens, and below which it closes. Setting it very low accommodates signals with little noise; higher threshold levels are useful for special effects, such as removing substantial amounts of a string's decay to create a more percussive or gated sound.
- **HOLD** sets a minimum amount of time the gate stays open. With a high threshold, this prevents the gating effect from stuttering.
- **ATTACK** controls the time the gate takes to open up after it has been triggered. The higher it is set, the softer the attack will be.
- **RELEASE** sets the time the noise gate takes to close when triggered.
- **LEARN** automatically sets the threshold. While not playing anything, click on Learn. Your signal is analyzed, the threshold is set just above the residual noise, and the button pops out again.

16.5.4. Volume Pedal

The Volume Pedal

About

This simple volume control can attenuate or boost the level at any point in the signal chain.

Controls

The slider controls volume from the minimum (when fully left) to the maximum level (when fully right). This is perfectly suited to be controlled by the RIG KONTROL or any MIDI foot pedal.

• **VOL** is the master volume control of this component. Use it to adjust the level when the slider is set to the extreme right.

Expert Controls

• **MINVOLUME** sets the volume when the slider is fully left.
• **HALF VOLUME** sets the volume when the slider is in the middle position. You can give pedals a more comfortable feel by adjusting the overall curve of the volume change with this knob.

16.5.5. Noise Reduction

The Noise Reduction component

About

The Noise Reduction component works similarly as the Noise Gate, but with a softer effect. One function is that of an expander, attenuating signals below the threshold without cutting them out completely. At the same time, a low-pass filter is triggered, attenuating the higher frequencies, where noise usually occurs. It is very easy to adjust.

Controls

• **THRESHOLD** determines the level above which the gate opens, and below which it closes.
• **LEARN** automatically sets the threshold. While not playing anything, click on Learn. Your signal is analyzed, the threshold is set just above the residual noise and the button pops out again.

Expert Controls

• **RELEASE** determines how long it takes for the filter to close down after the input signal falls below the threshold.
• **DE-HISS** controls the strength of the filter reducing high frequencies.

16.5.6. 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:
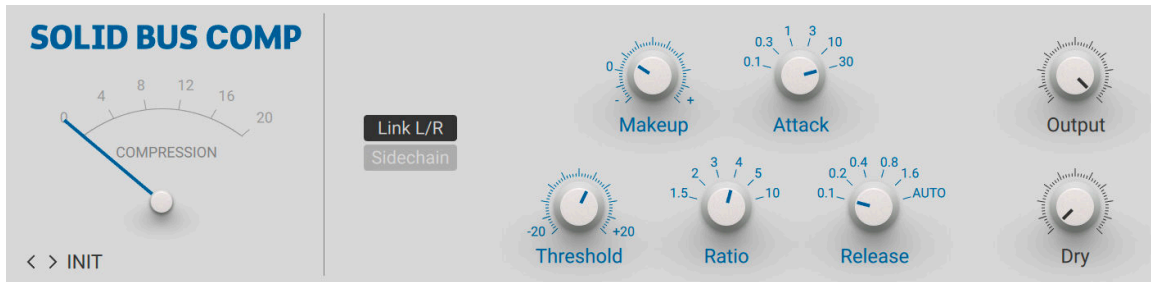
- **Gain Reduction Meter**: Displays the amount of gain reduction applied by the compressor.
- **Link L/R**: Sums of the left and right channel stereo channel for the compressor's control signal, thus applying the same processing to both channels. When deactivated, the left and right stereo channels 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 effect signal.
- **Dry**: Adjusts the level of the input signal that is added to the effect signal at the output.

16.5.7. 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 of the compressor 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 of the compressor, 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 of the gate from 1.5 ms to 100 µs. The attack time is the time it takes the gate to apply the full amount of gain reduction after the control signal falls below the threshold.
- **Release**: Adjusts the release time of the gate, which is the time it takes the gate to stop applying gain reduction after the control signal rises the threshold. **Release** can be set in the range from 0.1 s to 4 s.

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

16.5.8. Stomp Compressor

The Stomp Compressor

About

The Stomp Compressor provides a different type of compression compared to the Tube Compressor; the effect can be characterized as tighter and cleaner.

Controls

- **VOLUME** is the master volume control of this component.
- **SUSTAIN** is simultaneously affecting the threshold and the gain of the compressor. Turning it up will increase compression and boost the compressed signal. Note that other low-level signals like noise and hum will also be boosted.
- The meter shows how much the signal is currently being attenuated.

Expert Controls

- **ATTACK** sets the time it takes for the compressor to come to full effect. A longer attack time retains more of a signal's original dynamics, for example picking.
- **RELEASE** sets the time it takes for the compressor to return to its normal state after the signal falls below the threshold. With very short release times, the compressor tracks even slight level changes, possibly producing a choppy sound.
- **THRESHOLD** sets the level above which signals will be compressed. A lower threshold increases compression leading to greater sustain.
- **SIDECHAIN** allows the compressor to be triggered by an external signal. When lit, SideChain is enabled. In stand-alone mode, you can trigger the compressor with an external signal via the Audio and MIDI Settings' routing panel. See the Guitar Rig 5 Application Reference for further instructions. When run as a plug-in, set up the triggering of the compressor via the channel routing function of your DAW / host software. See the relevant documentation for further instructions.

16.6. EQ

An Equalizer is a way of boosting or attenuating parts of the frequency spectrum. It is one of the most basic ways to shape your sound, and GUITAR RIG 5 offers several great tools to do it.

16.6.1. Custom EQ

The Custom EQ

About

The Custom EQ is the emulation of a "boutique" EQ that is very popular in the studio these days. It has a particularly colorful and warm EQ sound.

Controls

- The **TONE** knob controls the balance between the treble and bass. When turned down, the low frequencies are boosted and the high frequencies are suppressed, when turned up it is the other way round.
- **FREQ** sets the center frequency for the midrange attenuation of this EQ, as controlled by the Scoop knob.
- **SCOOP** controls the amount of midrange attenuation.
- **VOLUME** sets the master volume of this component.

16.6.2. EQ Graphic
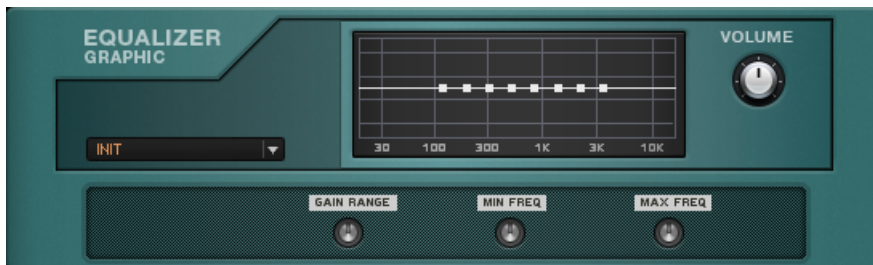
The EQ Graphic

About

This module uses band-pass filters to split the audio spectrum into eight bands providing an individual gain control for each band.

Controls

- The squares set the gain applied to each frequency band. Drag a square up to increase gain, down to decrease it. Double-clicking on a square restores it to zero gain.
- The **VOLUME** is the master level control.

Expert controls

- **GAIN RANGE** sets the maximum amount of boost and cut, influencing the effect of the gain settings. It ranges from ±1dB (very fine adjustment) to ±30dB (very intense).
- **MIN FREQ / MAX FREQ** set the lowest and highest band's frequencies, thereby defining the position of the affected frequency bands.

16.6.3. EQ Parametric
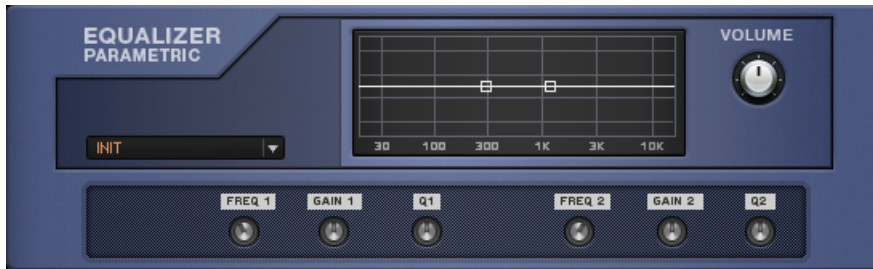
The EQ Parametric

About

A parametric equalizer is a highly sophisticated form of tone control, as it not only allows boosting or attenuating frequencies but as well the Q or width of the boost. The graph illustrates the response created by the controls, while the tool tips for each dot show the exact frequency in Hz and amount of boost/cut in dB.

EQ Parametric includes two complete parametric EQ stages.

Controls

There are two ways to adjust this component:

1. Click on one of the boxes in the graph and drag them: up to increase gain, down to decrease gain, sideways to change the center frequency.
2. Adjust these same parameters using the knobs in expert mode.

- **VOLUME** controls the output level for the device.

Expert Controls

- **FREQ 1** and **FREQ 2** set the center frequencies for each EQ stage.
- **GAIN 1** and **GAIN 2** set the amount of boost or attenuation.
- **Q1** and **Q2** control the width of the affected frequencies.

16.6.4. EQ Shelving
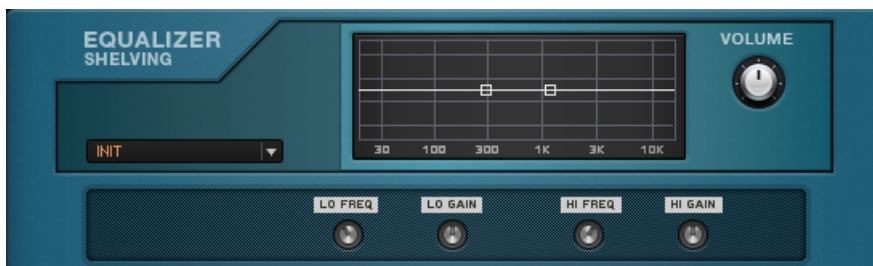
The EQ Shelving

About

A shelving equalizer combines a high-pass and a low-pass filter. A high-pass filter "passes" or allows through high frequencies meaning it attenuates bass. Conversely, a low-pass filter attenuates high frequencies.

This EQ easily corrects issues like a lack of high frequency "sparkle" or a boomy bass. The graph illustrates the response, while the tool tips for each dot show the exact frequency in Hz and amount of boost/cut in dB.

Controls

There are two ways to adjust this component:

1. Click on one of the boxes in the graph and drag them: up to increase gain, down to decrease gain, sideways to change the frequency limit.
2. Adjust these same parameters using the knobs in expert mode.

- The **VOLUME** control sets the output level for the effect.

Expert Controls

- **LO FREQ** sets the upper frequency limit for Gain1 (low shelf).
- **LO GAIN** controls the amount of gain for the low-pass filter.
- **HI FREQ** sets the lower frequency limit for **Gain2** (high shelf).
- **HI GAIN** controls the amount of gain for the high-pass filter.

16.6.5. 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.

16.7. Filters

Filters are essentially Equalizers that are tailored for a specific purpose. This collection of filters and wah-wah pedals will allow you to create sweeps and resonance for great effect.

16.7.1. AutoFilter

The Auto Filter

About

The AutoFilter is a multi-band filter that responds to the dynamics of your playing. This effect is very popular in funk and disco music and is used to add a synth dimension to instruments.

Controls

- **SENS** controls the sensitivity of the AutoFilter and should be tuned to match your guitar's level. If the filter isn't triggered or doesn't sweep sufficiently, increase the sensitivity. If the filter is triggered too easily, turn the sensitivity down.
- With the **UP/DOWN** switch set to "down," the filter sweeps down to a lower frequency; as the string decays, the filter returns to a higher frequency. When set to "up", the filter sweeps up to a higher frequency; as the string decays, the filter returns to a lower frequency.
- **RANGE** sets the width of the filter sweep. Turn it clockwise to sweep over a wider range.
- **RESO** controls the sharpness or resonance of the filter response. Turning it clockwise makes the effect more pronounced.
- **LFP/BPF/HPF** blends continuously between three different filter modes:
  - Low-pass (**LFP**)- the filter passes low frequencies and attenuates high frequencies.
  - Band-pass (**BPF**)- the filter passes a band of frequencies and attenuates frequencies above or below that band.
  - High-pass (**HPF**)- the filter passes high frequencies and attenuates low frequencies.

Expert Controls

- **ATTACK** sets the time the filter takes to complete the sweep, from 5ms to 80ms.
- **RELEASE** sets the time the filter takes to return to its initial setting after the sweep, from 50ms to 800ms. With short release times, the AutoFilter tracks even slight level changes. This can produce a choppy sound; turn up the release time for a smoother decay.
- **OFFSET** adjusts the filter's center frequency, i.e. the starting point for the filter sweep.
- **WET** controls the blend between dry and filtered signal. This is particularly useful with a bass guitar as the filter might thin out the sound. By turning this knob down, you can add some of the dry signal to prevent this from happening.

16.7.2. Real Wah

The Real Wah

About

The Real Wah is based on a custom wah-wah pedal that became an instant success when it appeared in the late 90's.

Controls

- The slider is used to control the wah-wah frequency. Moving the slider towards the left lowers the frequency, moving it towards the right raises the frequency.

Expert Controls

- **STEREO** activates true stereo processing for this module.

16.7.3. Cry Wah

The Cry Wah

About

The Cry Wah is based on the most popular wah-wah of all time. While its use with a foot controller such as that found in the Rig Kontrol is obvious, many famous guitarists have used the wah-wah in "fixed" mode creating a unique resonant peak.

Controls

- The slider is used to control the wah-wah frequency. Moving the slider towards the left lowers the frequency, moving it towards the right raises the frequency.

Expert Controls

- **STEREO** allows for true stereo operation.

16.7.4. 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:
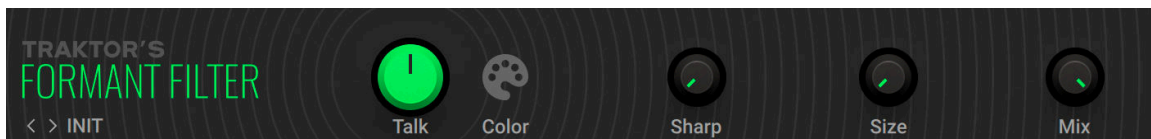
- **Talk**: Morphs between the five vowel sounds (a, e, i, o, u).
- **Color**: Switches between two different characteristics of the vowel sounds.
- **Sharp**: Makes the vowel sounds more sharp and distinct.
- **Size**: Adjusts the size of the modeled oral cavity, shifting the formant frequencies in the vowel sounds.

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

16.7.5. Filterbank

The Filterbank

About

The latest Filter effect in Guitar Rig 5 is FILTERBANK-- a multi-mode resonant filter. While the controls may look like a graphic equalizer, the FILTERBANK can do so much more.

Controls

- **DRY/WET** determines the amount of affected signal heard.
- Each of the 8 sliders represents two band-pass filters. Each filter is able to fully cut its frequency range enabling for drastic filtering and great sound design potential.
- The **STRETCH** control to the right of the sliders increases the frequency spread of the filters. At the leftmost point, the stretch is in a very narrow frequency band, while at the right it is spread out greater.
- The **SHIFT** control sets the midpoint of the frequency range: to the left goes lower, to the right goes higher.

Expert Controls

- **STEREO** switches for stereo operation. A small light will turn on when activated.
- **RESONANCE** makes the filters sharper and more prominent to give the familiar sound from wah-wahs and synthesizers.
- As each slider represents two band-pass filters, the **BAND SPLIT** parameter shifts the frequencies of the filters further apart.
- **LR SPLIT** shifts the frequency range of the left and right filters represented in each slider in opposite directions. For example, moving the knob to the right shifts the filters of the right channel up and the left channel down.

16.7.6. Wah-Wah Pedal

The Wah-Wah Pedal

About

The classic wah-wah pedal is a filter that sweeps a resonant peak over a certain frequency range; the wah-wah in GUITAR RIG 5 is perhaps the most versatile execution of this idea! Independent settings for filter frequency, resonance, and level at the high, mid, and low positions of the pedal allow for meticulous customization of the response and feel of the pedal.

Controls

• The slider controls the peak frequency. Moving the slider toward the left lowers the frequency, and moving it toward the right raises the frequency. The RIG KONTROL expression pedal is ideal for controlling this.

Expert Controls

• **STEREO** activates true stereo processing for this module.
• **LP-BP-HP** controls the filter's characteristics (low-pass, band-pass, and high-pass) and continuously blends between them. LP has the response of a typical low-pass filter, BP works more like a traditional wah-wah, and HP attenuates low frequencies as you move the slider to the right.
• **DRY** controls the blend between dry and filtered signal. By turning this knob up, you add more of the dry signal to the mix.
• **FREQ MIN** sets the filter frequency with the slider set all to the left.
• **FREQ MID** sets the filter frequency with the slider at its middle position.
• **FREQ MAX** sets the filter frequency with the slider set all to the right.
• **RES MIN** sets the amount of filter resonance with the slider set all to the left.
• **RES MID** sets the amount of filter resonance with the slider at its middle position.
• **RES MAX** sets the amount of filter resonance with the slider set all to the right.
• **VOLUME MIN** sets the filter level with the slider set all to the left.
• **VOLUME MID** sets the filter level with the slider at its middle position.
• **VOLUME MAX** sets the filter level with the slider set all to the left.

16.7.7. Pro-Filter

The Pro Filter

About

The Pro-Filter is based on a filter included in Native Instruments' popular software synthesizer Pro-53. It has a fat, rich sound that works very well as a tone control, but tying the Cutoff to a controller pedal also allows for great wah-wah sounds.

For advanced effects, try modulating the cutoff using the various Modifier components. For example, you can produce filtered effects that sync to the beat using the Step Sequencer.

Controls

- **CUTOFF** sets the operation frequency of the filter. Filtering occurs above and/or below it as defined by the **LFP/BPF/HPF** knob.
- **RESO** sets the filter bandwidth- turning it up narrows the frequency range thus accentuating the center frequency. When fully clockwise, the **RESO** control will push the filter into self-oscillation. The resultant frequency can be played by the **CUTOFF** control.
- **SLOPE** controls the rate at which frequencies are attenuated past the cutoff frequency. When turned down, frequencies are attenuated softly, at a rate of 12dB per octave. When turned up, frequencies are attenuated at a rate of 24dB per octave, producing a more intense filter sound.
- **LFP**/**BPF**/**HPF** blends continuously between three different filter modes:
  - Low-pass (**LFP**)- the filter passes low frequencies and attenuates high frequencies.
  - Band-pass (**BPF**)- the filter passes a band of frequencies and attenuates frequencies above or below that band.
  - High-pass (**HPF**)- the filter passes high frequencies and attenuates low frequencies.

16.7.8. 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:
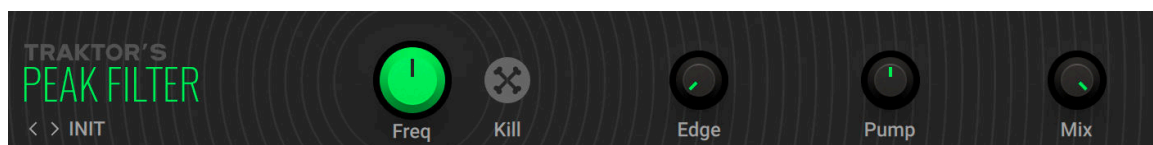
- **Freq**: Adjusts the filter frequency. Frequency content at the filter frequency is boosted, depending on the amount of resonance as set using **Edge**.
- **Kill**: Inverts the filter curve, creating a notch filter that cuts frequency content at the filter frequency as set using **Freq**.
- **Edge**: Adjusts the filter resonance. Turning **Edge** to the right makes the filter frequency as set using **Freq** more pronounced.
- **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.

16.7.9. Talk Wah

The Talk Wah

About

This is similar to a wah-wah, but utilizes a different filter type emulating the way vowels are shaped by the human mouth. The result is similar to the "talk box" effect that was very popular in the 70s.

Controls

- The slider controls the filter's frequency. Moving the slider toward the left produces a sound like the vowel "o"; moving it toward center morphs into more of an "a"; going further to the right creates an "e" sound. Control this with the RIG KONTROL's foot pedal and you'll be amazed at the sounds you are able to coax out of your guitar.

Expert Controls

- **STEREO** activates true stereo processing for this module.
- The **BRIGHT** switch accentuates the high frequencies.
- **VOLUME** controls the overall level of this component.
- **SIZE** adjusts the size of the "virtual mouth", influencing the overall sound of the filter.

16.8. Modulation

A modulator introduces motion into formerly static sounds. Common modulation devices in hardware rigs are chorus, vibrato and flanger. GUITAR RIG 5 provides you with a vast array of modulators than can be inserted anywhere in your rig. The majority of modulation components have the ability to sync either to the host clock or the metronome clock.

16.8.1. 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:
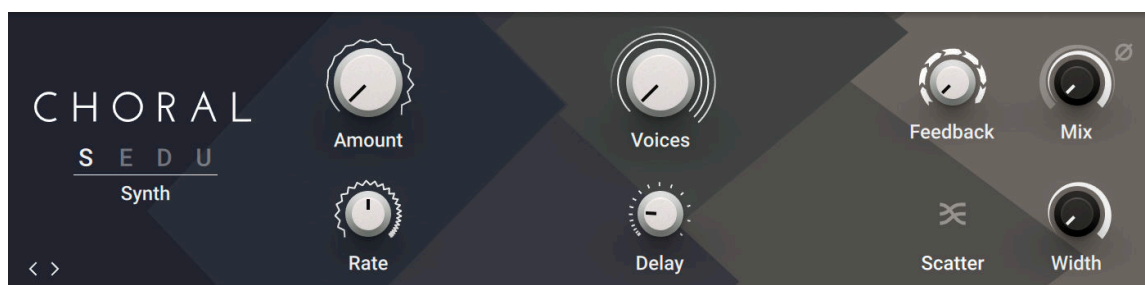
- **Mode**: Switches between four different chorus modes:
  - **Synth**: This mode is inspired by the choruses of polyphonic synthesizers from the late seventies and early eighties. Its sound characteristic is dark and vintage. The modulation behavior is tuned for rich and dispersed sounds.
  - **Ensemble**: 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:
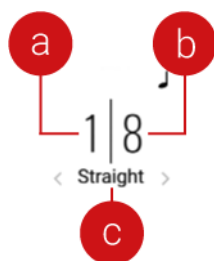
  The Numerator (**a**) and Denominator (**b**) set the speed of modulation in musical notes relative to the host tempo. The Numerator sets the number of notes, while the Denominator sets the note value. The Sync Mode (**c**) sets the time value, or subdivision, for the chosen note value.
- **Voices**: 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.

16.8.2. Stoned Phaser

The Stoned Phaser

About

The Stoned Phaser adds a swirling effect to the sound especially known from Psychedelic Rock music. This component is modeled after a popular phaser from the 1970s.

Controls

- **RATE** controls the speed of the phaser effect.
- **SYNC** synchronizes the phaser rate to the metronome or host, depending on the Sync setting.
- The **COLOR** switch provides a timbre change to the standard phaser sound.

Expert Controls

- **INVERT** changes the phase of the shifted signal, producing a yet another coloration.
- **NOTCHES** determines the number of stages of the phase shifter. Click on the number and drag to select from 1 to 5 notches.
- **COLOR STRENGTH** controls how much the Color Button affects the sound.
- **SWEEP MIN** sets the sweep's lower frequency limit.
- **SWEEP MAX** sets the sweep's upper frequency limit.
- **ROTATE** alters the phase difference between the LFOs feeding the left and right channels for a stereo effect.
- **DRY/WET** adjusts the mix of the dry and processed sound.

16.8.3. Electric Lady

Electric Lady is a versatile flanger effect. Modeled after a classic flanger, 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.

Electric Lady 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. Turning **Depth** to the right increases the frequency range of the effect.
- **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.

16.8.4. Ensemble

The Ensemble

About

This unique effect is based on a popular vintage processor. It adds slightly detuned voices to the signal and thus creates a polyphonic sound. Also, it provides a vibrato effect.

Controls

- **VOLUME** sets the master level of this component.
- **CHORUS** Intensity sets the effect depth and is only active in Chorus mode
- **MODE** switches between chorus and vibrato mode.
- **DEPTH** sets the amount of pitch deviation and is only active in Vibrato mode.
- **RATE** sets the vibrato speed and is active only in Vibrato mode.

Expert Controls

- **TEMP SYNC** synchronizes the Rate setting to the metronome or host tempo, depending on the Sync setting.
- **DRY/WET** sets the ratio of dry to processed sound.
- **BASS** controls the level of the lower frequencies.
- **MID** controls the level of the midrange frequencies.
- **TREBLE** controls the level of the high frequencies.
- **DELAY** sets the chorus circuit's delay time, i.e. the time until the effect affects the signal.
- **STEREO** adjusts the panning of the chorus voices from mono to full stereo.

16.8.5. 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:
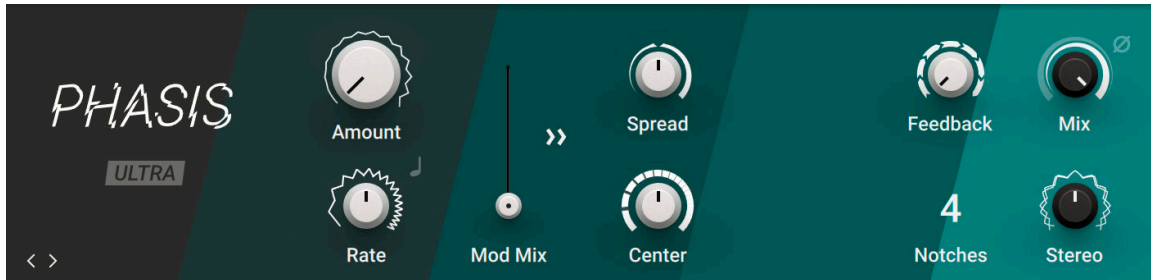
- **Ultra**: Extends the range of **Rate** and **Center**, allowing for more extreme modulation up audio frequencies. This can add new harmonic content to the input signal, similar to the sounds possible with FM synthesis.
- **Amount**: Adjusts the amount of modulation, adding movement to the effect.
- **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.
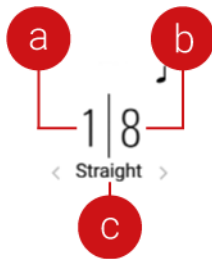- **Mod Mix**: Distributes the modulation between **Center** (**6**) and **Spread** (**9**). Moving the slider to the left increases the amount of modulation applied to **Center**, moving the slider to the right increases the amount of modulation applied to **Spread**.
- **Spread Modulation Polarity**: 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.

16.8.6. 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:
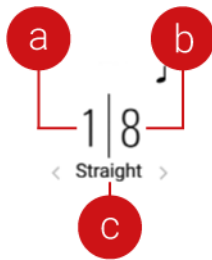
The Numerator (**a**) and Denominator (**b**) set the speed of modulation in musical notes relative to the host tempo. The Numerator sets the number of notes, while the Denominator sets the note value. The Sync Mode (**c**) sets the time value, or subdivision, for the chosen note value.

- **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.

16.8.7. Rotator

The Rotator

About

This effect simulates a rotating speaker. A true rotating speaker has two elements- the lower rotor and upper rotor. This module faithfully emulates both, providing separate controls for each one.

Controls

- **ROTATOR** allows switching the speed of the rotating speaker from slow to fast.
- **BALANCE** sets the ratio between the sound produced by the rotating speaker's high frequency horn and the low frequency woofer. Turning clockwise gives more highs, while turning counterclockwise gives a more pronounced bass sound.
- **PAN** changes the treble and bass rotors' location in the stereo field. Turning clockwise moves treble right while bass moves left and vice versa.
- **DISTANCE** sets the distance between the virtual microphones and the rotating speaker. The closer they are, the stronger the rotating effect is perceived.
- **DRY/WET** controls the effect's strength; turn fully clockwise to hear the rotating speakers only.

Expert Controls

- **CAB TONE** allows deactivating the Rotator's own cabinet simulation for more customization possibilities. For the most authentic sound, insert the Cabinets & Mics component preceding the Rotator with two signal chains. One holds the Rotator Bass cabinet panned to the left, the other the Rotator Horn cabinet panned to the right. Generally, the Rotator routes the left incoming channel to the bass rotor and the right incoming channel to the treble rotor and creates a new stereo field.
- Each rotor (treble and bass) has a set of identical controls.
- **SLOW** sets the rotor speed in slow mode.
- **FAST** sets the rotor speed in fast mode.
- **ACCEL** controls the acceleration of the rotors when switching from slow to fast mode and vice versa. It spans from a very sedate speed change to almost instant switching.
- **SPREAD** controls the width of the stereo image by setting the distance between the virtual microphones.

16.8.8. Flanger

The Flanger

About

This delay-based unit can create chorusing, flanging, and pitch modulation effects, depending on the Mode switch:

- Chorusing adds slightly detuned voices to the signal and thus creates a polyphonic sound.
- Flanging produces a spacey whooshing sound due to a mix of the original signal with one processed with a modulated delay time.

- Pitch modulation is better known as vibrato and causes a cyclic pitch change.

Controls

- **SPEED** varies the modulation rate of the chosen effect mode.
- **INTENSITY** controls the ratio of unprocessed to processed signal and creates a different effect depending on the chosen mode- it's best to try it out!
- **WIDTH** determines the range of each effect, i.e. the difference between both extremes of modulation.
- **MODE** switches between the Chorus, Flanger, and Pitch Modulation mode.

Expert Controls

- **SYNC** synchronizes the Rate setting to the metronome or host tempo, depending on the sync setting.
- **STEREO** creates an extra-wide stereo effect.

16.8.9. 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). For more information, refer to Modulation Types.
- **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.

16.8.10. Tremolo

The Tremolo

About

This component provides a periodic amplitude change creating a pulsing effect.

Controls

- **INTENS** (intensity) controls the tremolo effect depth.
- **RATE** sets the modulation frequency. Fast settings add a shimmering effect to the sound.
- **SYNC** will synchronize the tremolo rate to the host tempo or the Metronome tempo, depending on your sync settings.
- **STEREO Pan** activates a stereo tremolo effect: When the level increases in one channel, it decreases in the other one and vice-versa.

Expert 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.
- **DOWN** controls the decay time of the tremolo, i.e. the time it takes to go from the highest to the lowest volume level.
- **UP** changes the attack time, i.e. the time it takes to go from the lowest to the highest volume level. Turning both down creates an edgy stutter effect.

16.8.11. Phaser Nine

The Phaser Nine

About

This phaser is based on another well-known effect design, adding all kinds of animated shimmering to your guitar sounds.

Controls

- **RATE** controls the speed of the LFO causing the phaser modulation.
- **DEPTH** determines the intensity of the phasing.
- **COLOR** controls the timbre of the phasing as caused by the feedback of the effect. Higher settings put more emphasis on the harmonics.

Expert Controls

- **SYNC** synchronizes the LFO speed to the metronome or host, depending on the Sync setting.
- **INVERT** shifts the phase of the effect signal, creating a different timbre.
- **FREQUENCY** controls the base frequency around which the LFO modulation will take place.
- **NOTCHES** sets the number of notches the effect applies to the frequency range.
- **ROTATE** alters the phase difference between the LFOs feeding the left and right channels for a stereo effect.
- **DRY/WET** adjusts the mix of the dry and processed sound.

16.8.12. Stereo Tune

The Stereo Tune

About

The STEREO TUNE is a stereo widening effect. Using this tool will give a larger soundstage and the impression of a moving stereo image.

Controls

- MIX mixes the dry and wet signals
- SPLIT adjusts the split frequency for the detuned input signal. This allows the user to leave the bass frequencies unaffected.
- DRIFT detunes the input signal by a given percentage. The left and right channels are tuned at different frequencies to create a stereo widening effect.
- SPREAD adjusts the affected signal from mono to stereo.

16.9. Pitch

A collection of Pitch and synthesizer based effects for interesting, modern sounds. It's time to experiment with harmonics!

16.9.1. Harmonic Synthesizer
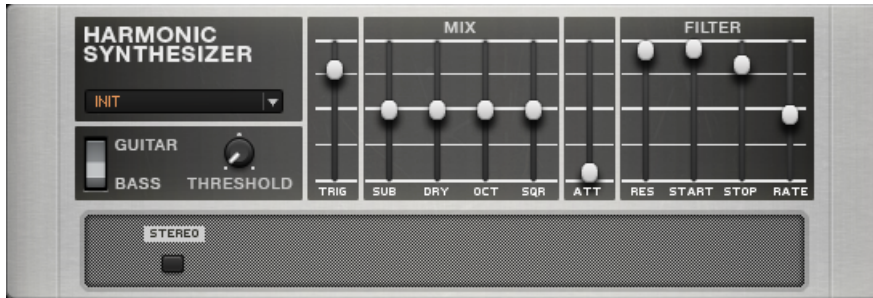
The Harmonic Synthesizer

About

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.

16.9.2. Oktaver

The Oktaver

About

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.

16.9.3. Pitch Pedal

The Pitch Pedal

About

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.

16.9.4. Resochord

The Resochord

About

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.

16.9.5. 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:
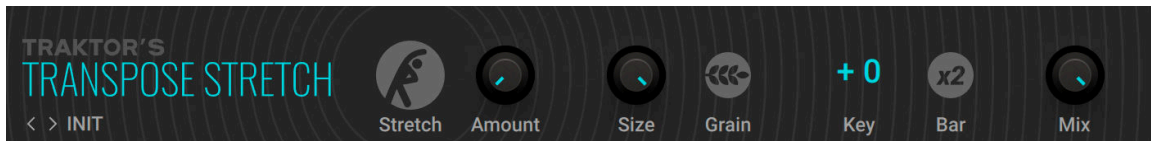
- **Stretch**: Starts the effect by recording the input signal into the internal buffer. When deactivated, the input signal is remains unaltered.
- **Amount**: Adjusts the amount of time-stretching. Turning **Amount** to the right slows the audio from the internal buffer down until it freezes using a 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.

16.10. Reverb

Reverbs offer an electronic interpretation of sound in a particular space. While technically speaking reverbs are essentially short delays, they certainly deserve a separate category of their own!

16.10.1. Iceverb

The Iceverb

About

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.

16.10.2. Spring Reverb

The Spring Reverb

About

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.

16.10.3. Little Reflektor

The Little Reflektor

About

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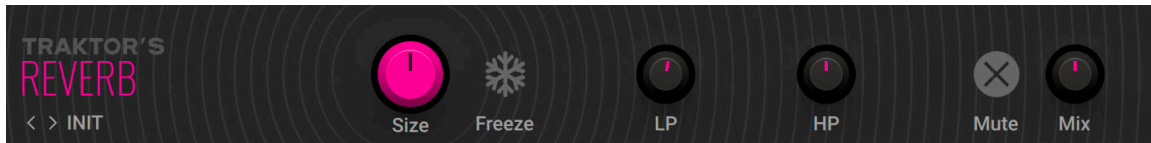
Controls

- **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.

16.10.4. 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.

16.10.5. Octaverb

The Octaverb

About

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.

16.10.6. 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:
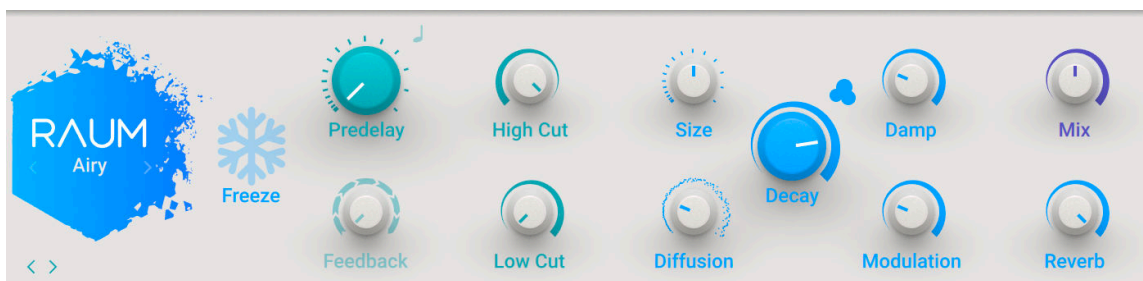
- **Reverb algorithm**: Switches between three different reverb algorithms:
  - **Grounded**: This algorithm is based on the paradigm of a room with strong early reflections and a dense reverb tail. The **Size** ranges from tiny ambiences to large chambers.
  - **Airy**: This algorithm is based on the 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:
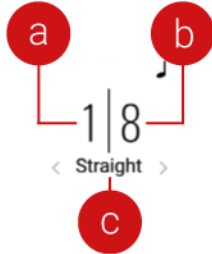
The Numerator (**a**) and Denominator (**b**) set the predelay in musical notes relative to the host tempo. The Numerator sets the number of notes, while the Denominator sets the note value. The Sync mode (**c**) sets the time value, or subdivision, for the chosen note value.

- **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.

16.10.7. Vintage Verb

The Vintage Verb

About

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.

16.10.8. Studio Reverb

The Studio Reverb

About

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.

16.11. Special FX

A collection of Special FX devices for advanced sound design and unusual effects.

16.11.1. Grain Delay

The Grain Delay

About

The Grain Delay is much more than just another delay effect. With its innovative granular approach it allows for a spacious refinement of your tones as well as massive walls of echoing sound- its interdependent controls are an invitation to experiment in order to achieve truly individual effects. The signal is sliced into samples of arbitrary length that are delayed, transformed and repeated in manifold ways, including a powerful stereo feature. With the Grain Delay, it gets just as weird as you like, at the same time the effect is perfectly controllable on all levels.

Controls

- **DRY/WET** sets the amount of the signal being fed into the delay section, controlling the intensity of the effect.
- **MODULATION** controls the amount of periodical pitch shift (vibrato) applied to the delayed signal.
- **PITCH** controls the pitch shift applied to the delayed signal. The amount of shifting is set in chromatic steps, spanning 4 octaves in both directions. Simultaneously, the duration of the sample in each "grain" is time-stretched by the according amount- think of a record player playing at different speeds! This pitch shift is not applied to the feedback channel, so it will not produce ever-ascending scales, but a constantly pitched delay.

- **FINE** allows changing the pitch shift by the amount of cents to refine the setting of the Pitch control.
- **SIZE** sets the duration of the samples that are fed into the delay section. The original signal is sliced into snippets ("grains") of this duration and repeated. If the slices are time-stretched using the Pitch control, the contained sample will be repeated within each grain or cut off, keeping the overall duration of the grain constant.
- **SPACE** controls the amount of time between the repetitions of each grain relative to the Size parameter. When set to 100%, the first repetition will start just when the grain is completely buffered. Reducing Space will start the repetition sooner and vice versa. The former leads to overlapping delays, the latter to gaps between the repetitions.
- **DENSITY** controls how much of the output feeds back into the input. Minimum feedback gives a single echo; increasing this parameter produces repeating echoes that are modulated and distorted, depending on the according expert controls.
- **DRIVE** controls the amount of overdrive created when the signal feeds back repeatedly and causes the component to overload.
- **H-CUT** sets the threshold frequency of a filter applied to the delayed signal. Frequencies above the threshold are strongly attenuated.
- **L-CUT** sets the frequency of a filter applied to the delayed signal. Frequencies below the threshold are strongly attenuated.
- **REVERSE** plays back the delayed signal in reverse.
- **FREEZE** stops sampling new slices of the signal and will repeat the current grain as long as it is activated.
- **MUTE** shuts off the signal going through the delay section, letting only the dry signals pass through. Delays 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

- **Sync** synchronizes the Size control to the metronome or host, depending on the Sync setting.
- **Jitter** varies the durations of each sampled grain by a small amount, creating a more lively texture.
- **Stereo** controls the amount by which the delayed signal is distributed to the stereo channels. Turning this up oscillates the signal between the stereo channels creating a ping-pong effect.

16.11.2. Ring Modulator

The Ring Modulator

About

A ring modulator is one of the seminal effects from the era of analogue noisemaking. The basic idea is very simple, nothing more than taking two signals and mixing the sum and difference of both, but the result is a large variety of unique sounds.

Controls

- The **MOD** section offers the following controls:
- **RING** is a dry/wet control and blends the original and the ring-modulated sound.
- **FM** controls the amount of frequency modulation applied to the original signal.
- The **FREQ** knob controls the frequency of the oscillator's signal being mixed with the original signal.
- **HI/LO** sets the frequency range of the oscillator, influencing the range of the **FM** and **FREQ** knobs.
- The **LFO** adds movement to the sound by periodically affecting the oscillator in the **MOD** section.
- The **AMOUNT** knob controls the amount of modulation the LFO applies to the oscillator.
- The **RATE** knob controls the rate of periodic modulation through the LFO.
- The sine/square switches the LFO's signal between a sine wave or square wave, creating either soft or sudden changes of the oscillator's frequency.
- The red LED blinks in time to the LFO frequency.

Expert Controls

- **SYNC** synchronizes the LFO speed to the metronome or host, depending on the Sync setting.
- **EDGE** modifies the waveform of the oscillator. You can add overtones resulting in a more aggressive sound.

16.11.3. 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.

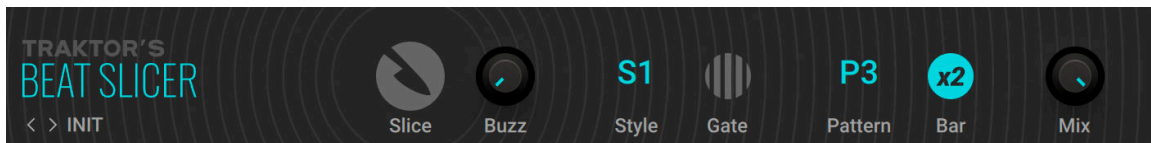This Component contains the following parameters and controls:
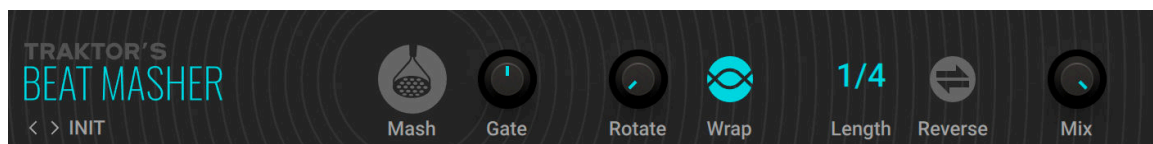
- **Mash**: Starts the effect by recording the input signal into the internal buffer. When deactivated, the input signal is unaltered.
- **Gate**: Grabs and plays slices of audio from the internal buffer. When turned fully to the left, the effect is bypassed. When turned right towards center position, 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.

16.11.4. 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.

16.11.5. 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:
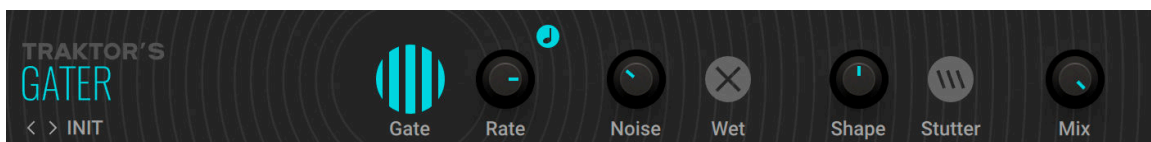
- **Gate**: Switches the effect on or off.
- **Rate**: Adjusts the rate of the gating effect.
- **Sync**: Synchronizes the rate of the gating effect to the metronome. When **Sync** is active, **Rate** can be set in musical intervals relative to the tempo of the metronome. The interval can be set to 1/4, 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.

16.11.6. 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:
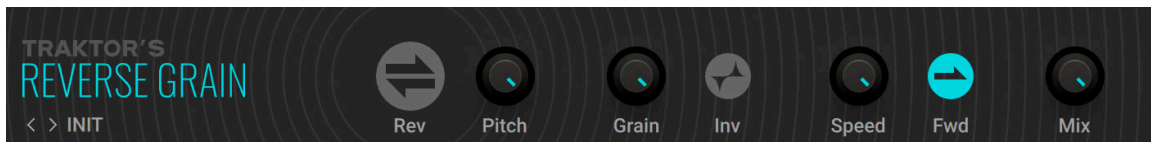
- **Rev**: Starts the effect by recording the input signal into the internal buffer and playing it back in reverse.
- **Pitch**: Adjusts the pitch of the audio from the internal buffer. When turned fully to the right, the pitch is unaltered. Turning **Pitch** to the left pitches the audio down.
- **Grain**: 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.

16.11.7. 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:
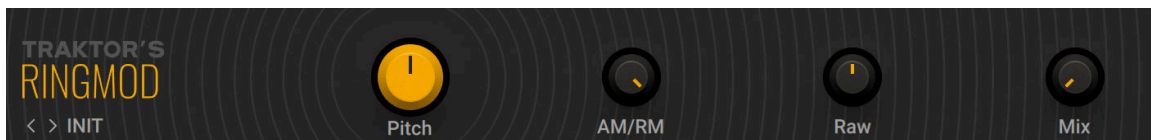
- **Pitch**: Adjusts the frequency of the modulation oscillator in the range of 100 Hz - 8371 Hz.
- **AM/RM**: Blends between AM (amplitude modulation) and RM (ring modulation).
- **Raw**: Adjusts the wave shape of the modulation oscillator from a sine wave to a filtered square wave. Turning **Raw** to the right makes the sound increasingly harsher.

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