MASSIVE X

Manual

NATIVE INSTRUMENTS

THE FUTURE OF SOUND

Table of Contents

1. DISCLAIMER

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2. WELCOME TO MASSIVE X

MASSIVE X is the successor to MASSIVE, the iconic synth that helped build musical genres. Expanding on this legacy, MASSIVE X provides you with all the features you need to create any sound imaginable.

By combining innovative sound generators and processors with modular routing and expressive modulation, MASSIVE X not only facilitates common synthesis techniques but also invites you to experiment and bring new ideas to life.

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

> To quickly learn how to create sounds using MASSIVE X and make yourself familiar with some of the instrument's key features and workflows, the MASSIVE X Quickstart Guide is available on the Native Instruments website here: https://www.native-instruments.com/massive-x-quickstart/

2.1. Document Conventions

This document uses particular formatting to point out special facts and to warn you of potential issues. The icons introducing the following notes let you see what kind of information can be expected:

> The speech bubble icon indicates a useful tip that may help you to solve a task more efficiently.

> The exclamation mark icon highlights important information that is essential for the given context.

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

Furthermore, the following formatting is used:

- Paths to locations on your hard disk or other storage devices are printed in *italics*.

- Important names and concepts are printed in **bold**.
- Square brackets are used to reference keys on a computer's keyboard, e.g., Press [Shift] + [Enter].

2.2. New Features in MASSIVE X 1.2

The following new features have been added to MASSIVE X:

Importing Presets by Drag-and-Drop

- Presets can be added to the User Presets folder by dropping files on the MASSIVE X user interface. For more information, see Importing a Preset.

Remote Octave and KOMPLETE KONTROL S-Series Integration

- The Lightguide on KOMPLETE KONTROL S-Series keyboards indicate the Remote Octave of the Performer. For more information on the Remote Octave, refer to Remote Octave.

Anonymous Data Tracking

- Feature based data tracking has been added to facilitate further improvement of the user experience. For more information on Usage Data Tracking, refer to Settings Menu.

Demo Time Increase

- Demo time has been increased from 30 to 60 minutes.

New Presets

- 100 new presets have been added to the Factory Library. For more information, see Loading a Preset.

2.3. New Features in MASSIVE X 1.1

The following new features have been added to MASSIVE X:

Animated Envelopes

- The Exciter, Amp and Modulation Envelope displays now reflect their actual state and respond to user input. Changes to the parameters will be immediately reflected. For more information on Envelopes, refer to Modulators.

Dark, Light and Flat Modes

- Dark, Light, Flat Default, Flat Dark, and Flat Light themes have been added. The Flat themes increase compatibility with older graphic cards. For more information on themes, refer to Settings Menu.

Rise/Fall Parameter Display

- The Rise/Fall parameter in both the LFO Switcher and LFO Random Envelope now visually displays the actual parameter state and responds to user input.

Category for New Factory Presets

- A separate category type has been added in the Browser for new Factory presets. This provides quick access to any new presets.

Tracker Grid Labels

- Grid labels have been added to the Tracker.

New Presets

- 60 new presets have been added to the Factory Library. For more information, see Loading a Preset.

3. OVERVIEW OF MASSIVE X

Massive X makes its deep functionality accessible through a clear layout that consolidates related modules and functions in horizontal sections. Below you can find an overview of these sections:

(**1**) **Header**: Find, save, and load presets, check output volume, and set the view size in the MASSIVE X drop-down menu. Additionally, the header includes the MIDI controller and Macros that can be assigned to control your sound.

- For more information about global controls and the header, see Header Overview.
- For more information about browsing and presets, see Browsing and Presets.
- For more information about Macros and MIDI control, see Macros and MIDI Control.

(**2**) **Module panels**: Control the generators and processors that you use to create your sound. You can freely connect them in the editor's Routing page to achieve a wide range of different synthesis techniques, including subtractive synthesis, wavetable synthesis, FM synthesis, and physical modeling.

- For more information about Tune, see Voice Page.
- For more information about Oscillators, see Wavetable Oscillators.
- For more information about Noise, see Noise.
- For more information about the Filter, see Filter.
- For more information about the Insert Effects, see Insert Effects.
- For more information about the Amplifier, see Amplifier.
- For more information about the Stereo Effects, see Stereo Effects.

(**3**) **Navigation bar**: Select the page you want to view in the editor (**4**). The available pages include Voice, Routing, and modulation sources. From the navigation bar, you can assign modulation sources to controls by using drag and drop.

- For more information about modulation, see Modulation.

(**4**) **Editor**: By selecting an editor page in the navigation bar, you can access all the controls and menus of the respective feature. Set monophonic or polyphonic behavior as well as unison and harmonizing on the Voice page, arrange and connect audio modules on the Routing page, or dive deep into any of the modulation sources.

- For more information about the Voice page, see Voice Page.
- For more information about the Routing page, see Routing.
- For more information about the modulation sources, see Modulation Sources.

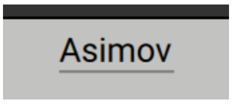
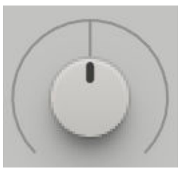
(**5**) **Remote Octave**: Control the contents of the Performers via MIDI. This way you can control the Performers' complex modulation curves in a playable manner and structure your song.

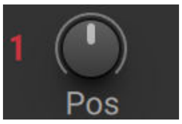
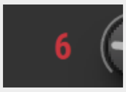
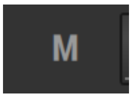
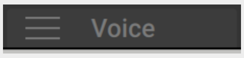
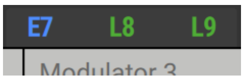
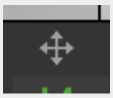
- For more information about the Remote Octave, see Remote Octave.

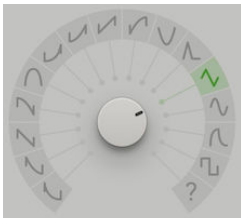
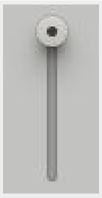
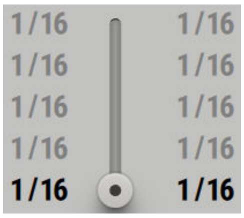
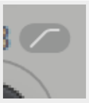
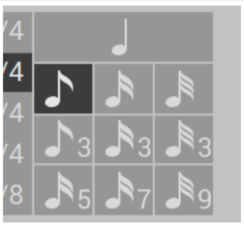
4. CONTROL ELEMENTS

MASSIVE X's user interface features a number of different controls. The following overview explains how to interact with them using a mouse.

Table 1. Control Elements

Name	Image	Description	Key Commands
Dropdown menu		Used to select from a list of options relating to the respective section or function.	• Click opens or closes the dropdown menu. • Click on any of the available entries selects the corresponding option and closes the dropdown menu.
Dropdown menu (with display)		Used to select from a list of options relating to the respective section or function. The selected entry is visualized in the display.	• Click on either the name or the display opens or closes the dropdown menu. • Click on any of the available entries selects the corresponding option and closes the dropdown menu.
Dropdown menu (extended)		Used to select from a list of options relating to the respective section or function.	• Click opens or closes the dropdown menu. • Click on any of the available entries selects the corresponding option and closes the dropdown menu. • Placing the cursor over the dropdown menu shows arrow icons that can be used to select the previous or next entry from the list, respectively.
Unipolar knob (with modulation)		Used to adjust parameters in the range of 0% to 100% from left to right.	• Click + drag up/down changes the control's value. • Double click sets the control to its default value.
Bipolar knob (with modulation)		Used to adjust parameters in the range of -100% to +100% from left to right. The center position is 0%.	• Click + drag up/down changes the control's value. • Double click sets the control to its default value.

Name	Image	Description	Key Commands
Wavetable knob		Used to adjust the wavetable position of the corresponding Wavetable oscillator. The display at the center of the control visualizes the resulting waveform.	• Click + drag up/down changes the control's value. • Double click sets the control to its default value.
Unipolar knob		Used to adjust parameters in the range of 0%-100% from left to right.	• Click + drag up/down changes the control's value. • Double click sets the control to its default value.
Bipolar knob		Used to adjust parameters in the range of -100% to +100% from left to right. The center position is 0%.	• Click + drag up/down changes the control's value. • Double click sets the control to its default value.
Macro knob		Used to adjust parameters assigned to the Macro. For more information, see Macros and MIDI Control.	• Click + drag up/down changes the control's value. • Double click sets the control to its default value.
Macro icon		Used to assign a Macro to parameters. For more information, see Macros and MIDI Control.	• Drag + drop onto a control or a modulation slot assigns the Macro to the corresponding parameter.
Controller icon		Used to assign the respective midi controller to parameters. For more information, see Macros and MIDI Control.	• Drag + drop onto a control or a modulation slot assigns the MIDI controller to the corresponding parameter.
Page tab		Used to show the respective page in the Editor.	• Click shows the page in the Editor.
Modulation source tab		Used to show the respective modulation source in the Editor.	• Click shows the modulation source in the Editor.
Modulation icon		Used to assign a modulation source to parameters. For more information, see Assigning Modulation.	• Drag + drop onto a control or a modulation slot assigns the modulation source to the corresponding parameter.
Pitch control		Used to control the pitch and pitch modulation amounts of components that track MIDI pitch.	• Click + drag on the integer number changes the pitch in semitones. • Click + drag on the decimal places changes the pitch in cent.

Name	Image	Description	Key Commands
Numeric control		Used to adjust parameters in a specific range of values.	• Click + drag up/down changes the control's value. • Double click sets the control to its default value.
Rotary selector		Used to select from a predefined set of different states or values.	• Click + drag up/down changes the control's value. • Double click sets the control to its default value. • Click on icon in circle sets the corresponding value.
Slider		Used to adjust parameters in the range of 0% to 100% from bottom to top.	• Click + drag up/down changes the control's value. • Double click sets the control to its default value.
Quantized slider		Used to select from a user-defined set of different states or values.	• Click + drag up/down changes the control's value. • Double click sets the control to its default value. • Click + drag on any of the adjacent numbers changes the corresponding value.
Function button		Used to switch functions on or off.	• Click toggles between the control's active and inactive state.
Module button		Used to switch modules on or off.	• Click toggles the module's bypass function.
Auxiliary button		Used to switch auxiliary functions related to adjacent controls on or off.	• Click toggles between the control's active and inactive state.
Mode selector		Used to select one of several modes.	• Click on one of the buttons in the mode selector selects the corresponding mode.

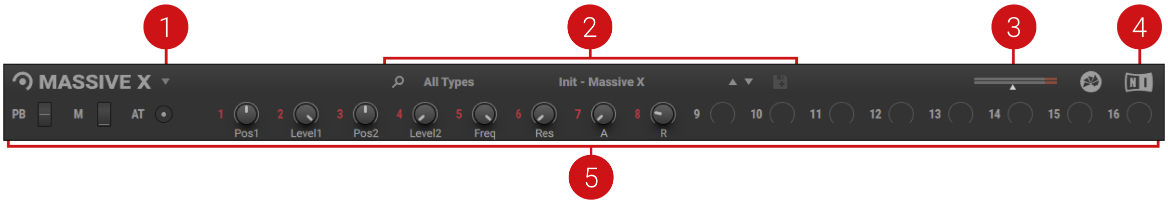
Name	Image	Description	Key Commands
Routing module		Used to make connections between modules on the Routing page.	• Click + drag moves the module to a new position. • Right click toggles the modules bypass function. • Double clock removes all connections from the module. • Click + drag from an output node to an input node creates a connection. Alternatively, successive clicks on the two nodes can be used.

5. GLOBAL CONTROLS

The Header at the top of MASSIVE X provides you with functions related to preset management and plug-in behavior. You can access these functions via the Settings menu on the left as well as the central Browser and preset menu. Additionally, the Header features Macros and MIDI control sources that you can use to play your sound.

5.1. Header Overview

This section provides an overview of the Header's features, including the Settings menu, the Browser, and the Macro controls.



(**1**) **Settings menu**: Provides you with options for managing user presets and adjusting the plug-in's visual appearance. For more information, see Settings Menu.

(**2**) **Browser and presets menu**: Here you can access the browser, switch between presets, and save user presets. For more information, see Browsing and Presets.

(**3**) **Master volume**: Adjusts and displays the volume level of the plug-in's output signal.

(**4**) **Logo**: By clicking on the NI and MASSIVE X logos you can view the version number, license, and credits of the plug-in.

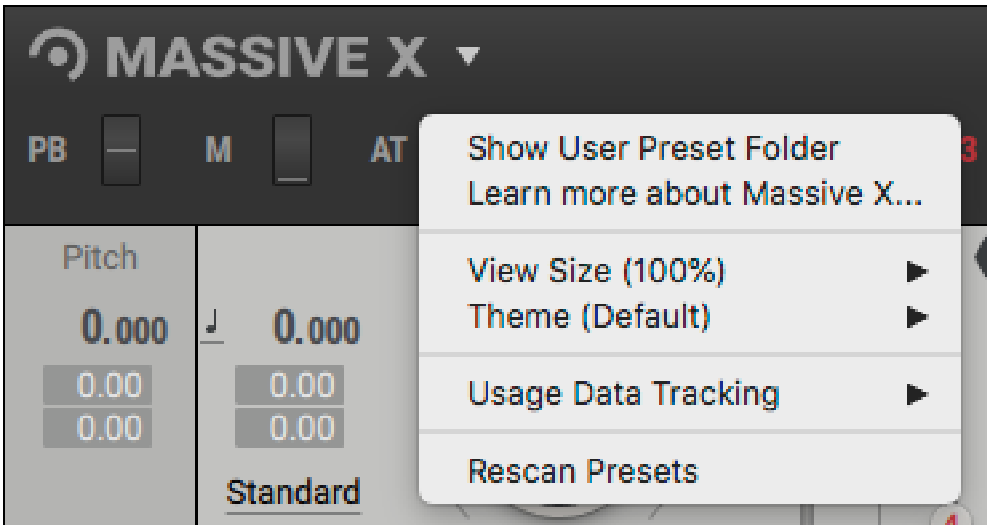
(**5**) **Macros and MIDI control sources**: These controls are used to adjust and assign Macros and MIDI control sources that you can use to play your sound. For more information, see Macros and MIDI Control.

5.2. Settings Menu

The Settings menu in the Header provides you with options for managing user presets and adjusting the plug-in's visual appearance. You can also use it to access learning resources online and enable or disable usage data tracking.

1. Click on the arrow icon next to the MASSIVE X logo in the Header to open the Settings menu.

The following entries are available in the Settings menu:



- **Show User Preset Folder**: Opens the folder on the hard drive containing your user presets. For more information, see Managing User Presets.
- **Learn more about Massive X...**: Opens the Native Instruments website where you can follow the Quick Start Guide, download the Manual, and watch video tutorials.
- **View Size**: Provides eight different sizes for the plug-in and save the current size as default value.
- **Theme**: Provides six different visual appearances for the plug-in. In addition to the default setting, **Dark** and **Light** color schemes are available, as well as **Flat** versions that are optimized for efficiency.
- **Usage Data Tracking**: Here you can learn more about Usage Data Tracking and choose to enable or disable it.
- **Rescan Presets**: Updates the Browser to reflect all changes made to the User Preset folder. For more information, see Managing User Presets.

6. BROWSING AND PRESETS

The Browser in MASSIVE X is used for browsing, loading, and saving sounds.

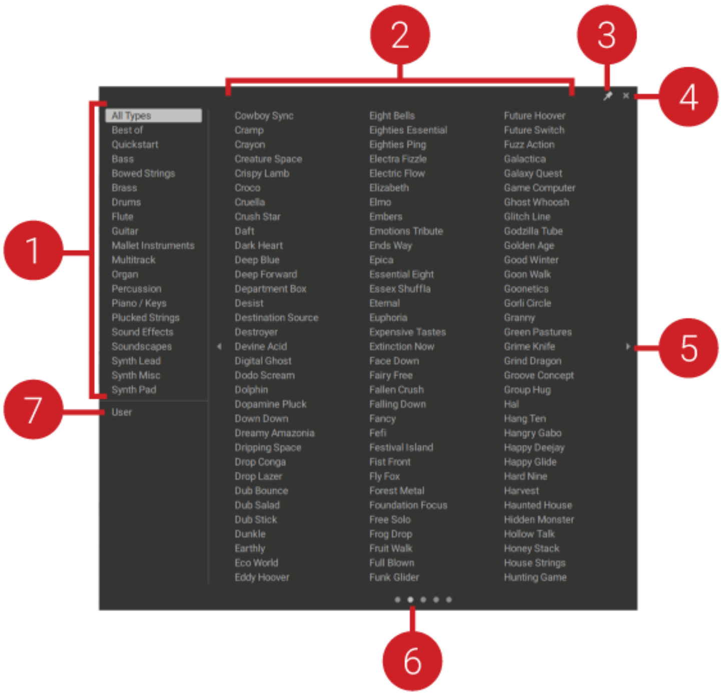
The Browser contains clear categories that provide an intuitive and convenient way to quickly find exactly what you need from a large library of sounds. In addition to the professionally designed sounds, a set of tutorial presets are also provided and have been designed to be used in combination with the MASSIVE X Quick Start Guide.

To open the MASSIVE X Browser:

- Click either the magnifying glass icon, the category, or the current Preset name in the Header.

Overview of the Browser

The Browser contains the following controls:



(**1**) **Categories**: An alphabetically ordered list of preset categories based on instrument types, a best-of selection, and the Quickstart Guide. The selected category is highlighted. Click another category to display its presets. Your saved presets are not included in this list, but in a separate category named User (**7**)

(**2**) **Presets**: A list of presets belonging to the selected category (**1**). Presets are displayed in alphabetical order. Click the name of a preset to load it. The selected preset is highlighted.

(**3**) **Pin**: By default, the Browser closes automatically after a preset is selected. To keep the Browser open when selecting presets, click the pin icon. When the pin icon is highlighted the Browser is pinned. However, you can still close the pinned Browser by double-clicking a preset.

(**4**) **X**: Click the **X** symbol to exit the Browser.

(**5**) **Next/previous pages**: When a category has more than one page of presets to display, the next and previous symbols appear. Click the next page symbol on the right to display the next page of presets, or click the previous page symbol on the left to see the previous page of presets. You can also select Browser pages using the page symbols (**6**).

(**6**) **Page symbols**: When a category has more than one page of presets, dots appear at the bottom of the Browser to represent the number of pages available for the selected category. Click a dot to display another page of presets. You can also step back and forth through the available pages using the next/previous page symbols (**5**).

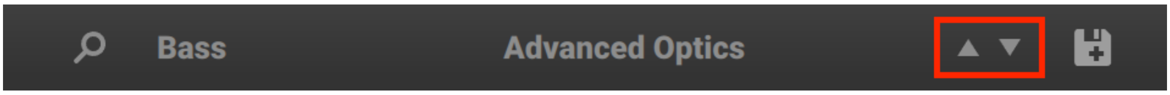
(**7**) **User**: This category includes an alphabetically ordered list of all your saved presets.

6.1. Loading a Preset

Presets can be loaded one by one directly from the Header or selected from within the Browser.

Loading a Preset from the Header

Loading presets from the header is quick and easy. Using the up/down arrows, you can quickly step through the presets from within the selected category.



To load a preset from the Header:

1. Click the arrow up to step upwards through the select category of presets.
2. Click the arrow down to step downwards through the selected category of presets.

Loading a Preset from the Browser

If you require an overview of all categories and presets, then it's best to load presets from the Browser. When the Browser is open you can change the category and select presets.

To load a preset from the Browser:

1. Click either the magnifying glass icon, the category, or the current preset name in the Header.
2. When the Browser opens, select a category name from the list on the left. If you require access to your saved presets, select the User category.
3. Select a preset name from the list on the right.

6.2. Saving a Preset

If you've created a sound that you like, you'll probably want to save it. When a Preset is saved, it is added to the User Preset folder on your computer and becomes visible in the User category of the MASSIVE X Browser.

To save a preset to the User folder:

1. Click the **Save icon** (the disk icon) in the Header.
2. In the dialog that appears, enter a name for your preset.
3. Optionally, enter the name of the author who created the preset. The default name is Native Instruments.
4. Click **Save** to store the preset. If the preset name you provided already exists in the User category of the browser, you will be prompted with the option to replace your existing preset. Click **Replace** if you want to replace it, or provide a new name for your preset and then click **Save**. You can also click **Cancel** at any time to exit the Preset Save procedure.

You can not overwrite presets in the MASSIVE X Factory Library. If you decide to edit a Factory Library preset and save it, it will be stored as a separate preset in the User category.

If you edit or delete any files in the User Preset folder, make sure you rescan your sounds by selecting Rescan Presets in the MASSIVE X drop-down menu in the header. For more information managing presets, see Managing User Presets.

6.3. Importing a Preset

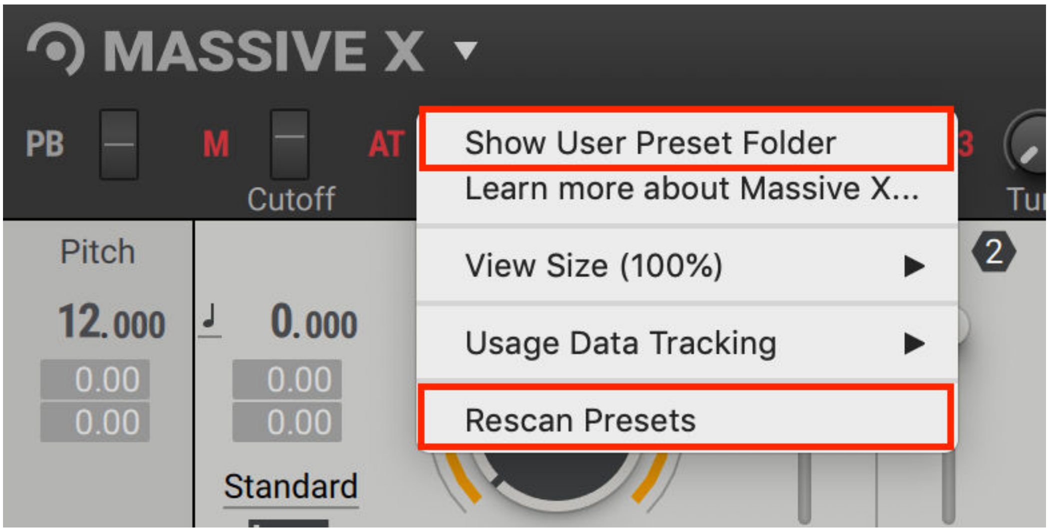
Presets can be imported into the the User Preset folder by dropping the preset files onto the MASSIVE X user interface. A preset file can be dropped anywhere on the user interface and it will automatically be saved to the User Preset folder.

To import a preset file to the User Preset folder:

1. Find the preset file you want to import on your hard drive.
2. Drag-and-drop the file onto the MASSIVE X user interface.
   The preset file is now saved into the User Preset folder.

6.4. Managing User Presets

In the drop-down menu next to the MASSIVE X logo, on the left-hand side of the Header, there are two menu entries provided to help you manage your user presets.



Viewing your Presets

If you want to view or edit the names of your preset files:

1. Click the drop-down menu next to the MASSIVE X logo, on the left-hand side of the Header.
2. In the menu, click **Show User Preset Folder**.
3. In the dialog that opens, you can access your user preset files to sort or rename them.
4. When you have finished, rescan your presets to reflect any changes you made in the MASSIVE X Browser.

Rescanning Presets

When presets have been added, deleted or renamed, it is necessary to perform a rescan. Only by rescanning your User Preset folder will MASSIVE X be able to actively reflect any changes in the Browser.

To rescan your User preset folder:

1. Click the drop-down menu next to the MASSIVE X logo, on the left-hand side of the Header.
2. In the menu, click **Rescan Presets**.
3. The Browser is updated to reflect the files in the User preset folder.

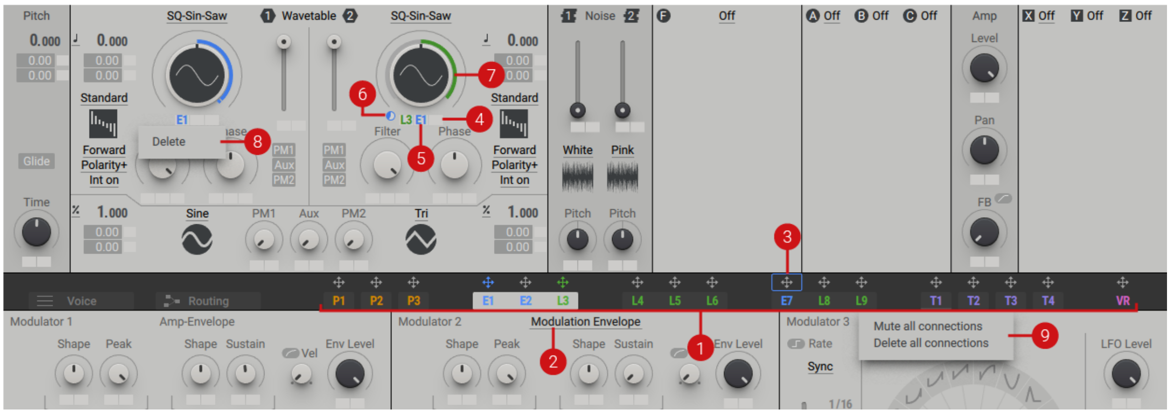
7. MODULATION

Modulation adds movement to your sound by changing controls over time. This movement defines the character and expression of a sound as you play it, from the contour of its amplitude, or volume level, to its timbral qualities. In technical terms, modulation is composed of slowly changing signals that cannot be heard directly. When you apply modulation to controls in your patch, the controls change their value according to the shape of the modulation signal.

MASSIVE X puts a strong focus on the concept of modulation by combining sophisticated modulation sources with an intuitive system for modulation assignment: modulation sources can be applied to any number of parameters by using drag and drop, and the modulation assignments are visualized directly in the user interface. Color-coded rings or lines next to controls show not only which type of modulation source is used, but also the amount of modulation applied to the parameter.

7.1. Modulation Overview

This section provides an overview of the user interface elements used for modulation, including the modulation sources and the modulation slots that are used as destinations.



(**1**) **Modulation sources**: All available modulation sources can be accessed from the navigation bar. You can show them in the Editor in groups of three by clicking on their labels (e.g. **P1**, **E2**, **L8**, **T3**). For an overview of the available modulation sources, see Modulation Sources.

(**2**) **Modulator menu**: Selects one of four available Modulators: **Modulation Envelope**, **Exciter Envelope**, **Switcher LFO**, **Random LFO**. You can choose one of the Modulators for each of the blue and green modulation sources in the navigation bar. For more information about the Modulators, see Modulators.

(**3**) **Modulation assignment**: The arrow icon is used to assign the corresponding modulation source (**1**) to a parameter. To do this, you can either drag and drop the arrow icon onto a modulation slot (**5**), or first click the arrow icon and then the modulation slot. For more information, see Assigning Modulation.

(**4**) **Modulation slot**: Displays and controls the modulation assignment for the corresponding parameter. Two modulation slots are available for each parameter (left and right slot beneath a control). When a modulation source is assigned, clicking and dragging the modulation slot up and down adjusts the modulation amount as displayed by the color-coded ring or line next to the control. For more information, see Assigning Modulation.

(**5**) **Sidechain modulation slot**: Used to assign a modulation source for sidechain modulation. Sidechain modulation controls the modulation amount, or strength, of the modulation slots to the left and to the right (**4**). The sidechain modulation slot can also be used for direct modulation of the parameter like a regular modulation slot. For more information, see Sidechain Modulation.

(**6**) **Sidechain modulation amount**: Adjusts the amount of sidechain modulation applied to the corresponding modulation slot from the sidechain modulation slot (**5**). For more information, see Sidechain Modulation.

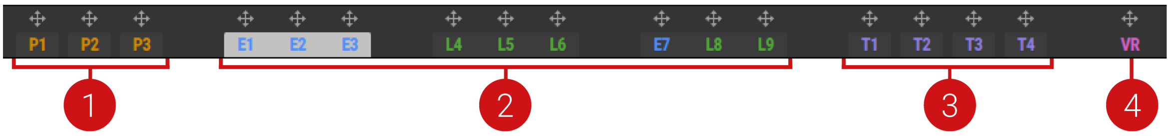
(**7**) **Modulation amount**: A color-coded ring or line next to a control shows the amount of modulation applied to the parameter from the modulation source assigned to the respective modulation slot. For more information, see Assigning Modulation.

(**8**) **Modulation slot menu**: This context menu opens by right-clicking a modulation slot. You can use it to delete the modulation assignment for this modulation slot.

(**9**) **Modulation source menu**: This context menu opens by right-clicking a modulation source in the navigation bar (**1**). You can use it to mute or delete all modulation assignments for this modulation source.

7.2. Modulation Sources

MASSIVE X features a comprehensive set of modulation sources that you can use to add movement to your sound. All modulation sources can be assigned and accessed from the navigation bar:



(**1**) **Performers**: The three Performers are specialized sequencers for modulation. They provide a flexible way to apply complex, rhythmical movement to parameters. For more information, see Performers.

(**2**) **Modulators**: The Modulators cover all basic modulation needs, but also provide advanced features for special applications. The first Modulator (**5**) is defined as Amp Envelope (**E1**) and hardwired to control the Amp level. Each of the other eight Modulators can either be a **Modulation Envelope**, an **Exciter Envelope** for physical modelling, a **Switcher LFO** for periodic modulation, or a **Random LFO** for random effects. For more information, see Modulators.

(**3**) **Trackers**: The four Trackers provide deep functionality for advanced keyboard tracking. They map incoming MIDI control data to modulation that you can apply to any parameter. This enables you to define exactly how your sound responds to the MIDI input. For more information, see Trackers.

(**4**) **Voice Randomization**: Voice Randomization allows you to add pseudo-random variation to your sound by generating a different modulation value per voice. For more information, see Voice Randomization.

7.3. Assigning Modulation

Before the effect of the envelope or LFO can be heard, it must first be assigned to a control's modulation slot. Modulation sources can be assigned to multiple parameters at once, and the process for assigning each is the same.

To assign a modulation source to a parameter:

1. Drag and drop the arrow icon of a modulation source, for example the first Performer (**P1**), to a modulation slot.

> Alternatively, you can first click the arrow icon and then the modulation slot to make the assignment.

2. Click and drag the modulation slot upwards or downwards to increase the modulation amount applied to the parameter. Dragging the slot upwards adds positive (non-inverted) modulation, while dragging the slot downwards adds negative (inverted) modulation.

> You can double-click the modulation slot to set the maximum modulation amount or reset it to zero.

For information about removing or replacing Modulation assignments, see Removing and Replacing Modulation.

7.4. Removing and Replacing Modulation

You can remove or replace the modulation assignment of any modulation slot while retaining the modulation amount that has been set. When assigning a new modulation source, the original modulation amount is taken over.

To remove a modulation assignment:

- Right-click the modulation slot and select **Delete**.

To replace a modulation assignment with another:

- Drag and drop the new modulation source onto the already assigned modulation slot.

Alternatively, you can first click the arrow icon and then the modulation slot to make assignment.

7.5. Sidechain Modulation

Sidechain modulation allows you to add movement to the modulation amount, or strength, of other modulation assignments. Instead of modulating the parameter itself, it changes the intensity of how another modulation source is applied to the parameter.

Sidechain modulation is facilitated by the middle modulation slot beneath a control:

(**1**) **Sidechain modulation slot**: Used to assign a modulation source for sidechain modulation, which can be applied to the modulation slots to the left and to the right. It can also be used for direct modulation of the parameter like a regular modulation slot.

(**2**) **Sidechain modulation amount**: Adjusts the amount of sidechain modulation applied to the corresponding modulation slot from the sidechain modulation slot (**1**).

In technical terms, the signal produced by the modulation source assigned to a modulation slot is multiplied by the signal produced by the modulation source assigned to the corresponding sidechain modulation slot, by a factor set with the sidechain modulation amount control.

8. MACROS AND MIDI CONTROL

The Header includes a number of control sources that you can use to interact with your sound. Assign the modulation wheel, pitch bend and aftertouch to parameters for instant playability from a keyboard controller, or use Macros for automation and MIDI control in your host.

The following control sources are available:

(**1**) **PB (Pitch Bend)**: A standard MIDI control found on any keyboard controller, often in the form of a spring-loaded wheel.

(**2**) **M (Modulation)**: A standard MIDI control found on most keyboard controllers, often in the form of a freely adjustable wheel.

(**3**) **AT (Aftertouch)**: A special MIDI expression found on many keyboard controllers, transmitting the amount of pressure applied to a pressed key.

(**4**) **1-16 (Macros)**: Global controls that can be used for automation and MIDI control in your host.

All control sources available in the Header can either take over the full range of a single parameter or control any number of parameters to a varying degree:

- For information about single parameter assignment, see Assigning Macros to Single Parameters.
- For information about multiple parameter assignment, see Assigning Macros to Multiple Parameters.

You can conveniently manage existing assignments by removing, muting, and replacing them:

- For information about removing and muting Macros, see Removing and Muting Macros.
- For information about replacing Macros, see Replacing Macros.

Macros can also be renamed, which is especially useful when assigning them to multiple parameters. For more information, see Renaming Macros.

8.1. Assigning Macros to Single Parameters

The controls in the Header can take over the full range of a single parameter, which is called a 1:1 mapping.

For example, you can use the modulation wheel to control the Wavetable Position.

1. Drag and drop the modulation wheel icon (**M**) from the Header onto the **Wavetable Position**.

By moving the modulation wheel on your keyboard controller you can now control the **Wavetable Position**.

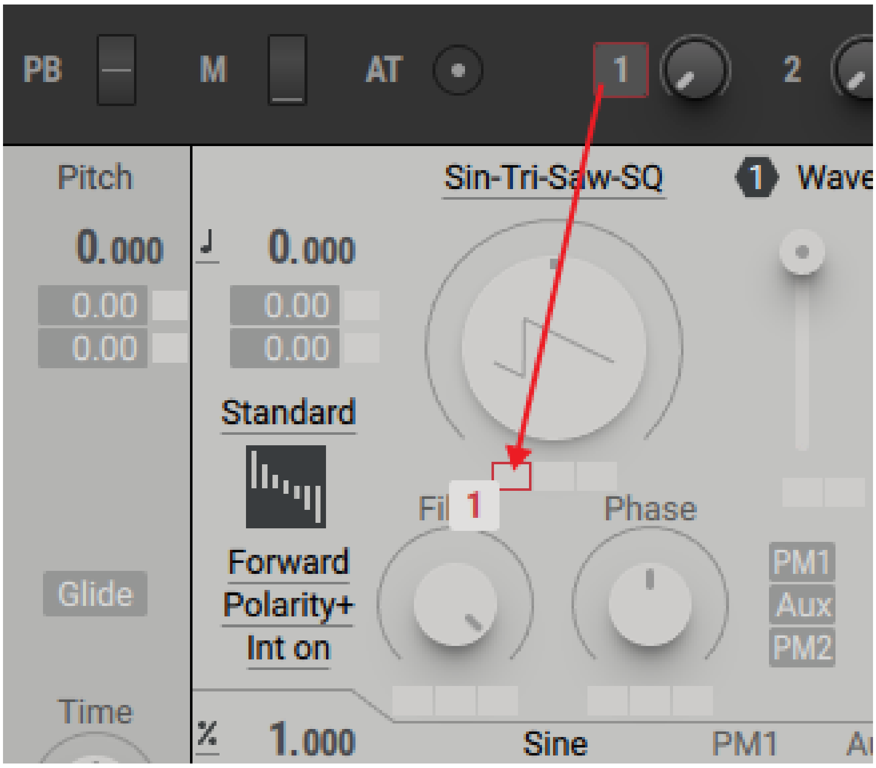
For information about removing or muting Macro assignments, see Removing and Muting Macros.

8.2. Assigning Macros to Multiple Parameters

The controls in the Header can adjust multiple parameters at the same time, and to a varying degree.

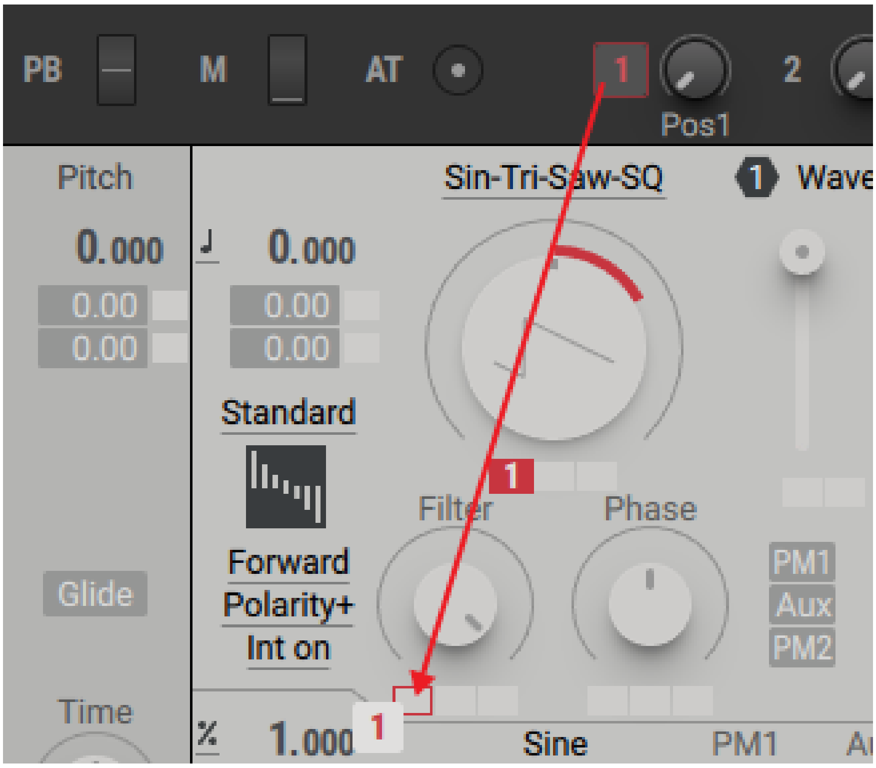
For example, you can use Macro **1** to control both the **Wavetable Position** and the Wavetable's **Filter** control:

1. Drag and drop the Macro icon (**1**) from the Header onto the first modulation slot of the **Wavetable Position**.



2. Click and drag the first modulation slot of the **Wavetable Position** to adjust the modulation amount.

3. Drag and drop the Macro icon (**1**) from the Header onto the first modulation slot of the Wavetable's **Filter** control.



4. Click and drag the first modulation slot of the Wavetable's **Filter** control to adjust the modulation amount.

By playing a note and turning Macro **1** you can now control the two parameters on the Wavetable oscillator at the same time.

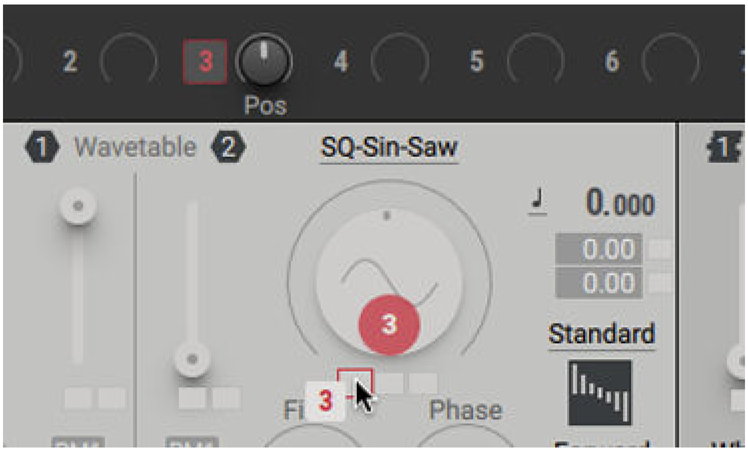
For information about removing or muting Macro assignments, see Removing and Muting Macros.

8.3. Replacing Macros

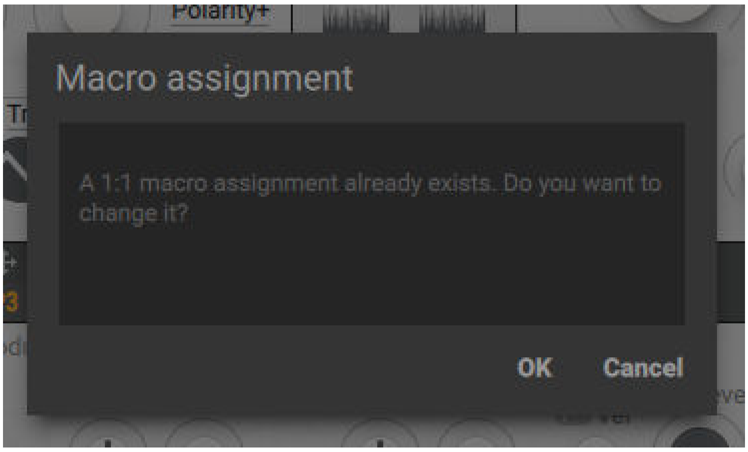
In some scenarios, you might want to remove a 1:1 Macro assignment and instead assign the Macro to a modulation slot of the same parameter, or vice versa.

To replace a 1:1 Macro assignment for a parameter with a modulation slot assignment:

1. Drag and drop the Macro icon from the Header onto the modulation slot.

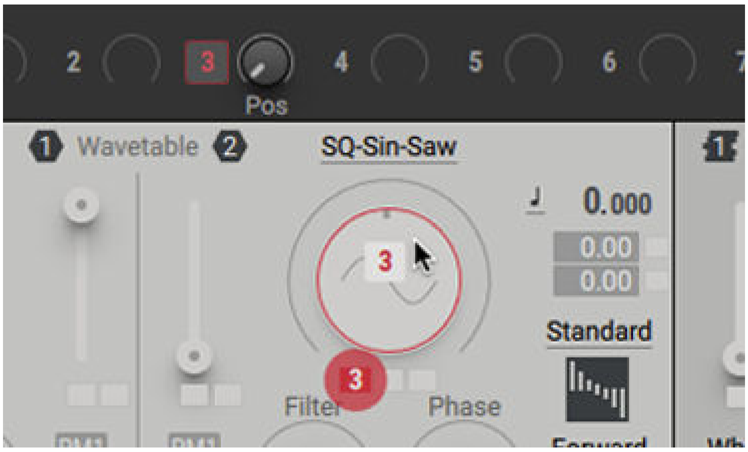


2. A dialog box appears, asking you to confirm the change. Click **OK** to remove the 1:1 Macro assignment, and replace it with a modulation slot assignment.

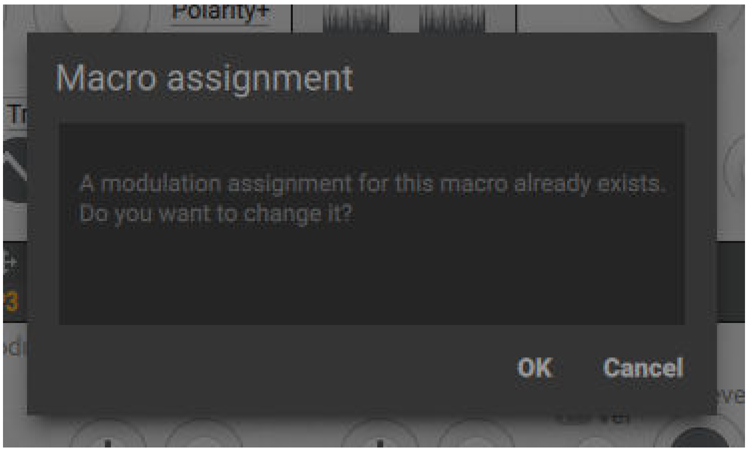


To replace a modulation slot assignment for a parameter with with a 1:1 Macro assignment:

1. Drag and drop the Macro icon from the Header onto the parameter's control.



2. A dialog box appears, asking you to confirm the change. Click **OK** to remove the Macro's modulation slot assignment, and replace it with a 1:1 Macro assignment.



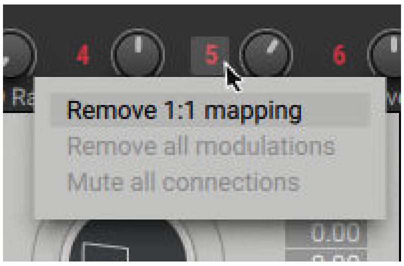
8.4. Removing and Muting Macros

You can remove or mute Macro assignments using the context menu for each Macro by right-clicking on the Macro number.

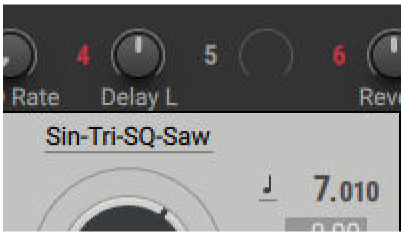
When a Macros is assigned to one control, it is referred to as 1:1 mapping.

To remove a Macro assignment from a single control:

1. Right-click the Macro number, located to the left of the corresponding control knob.



2. Select **Remove 1:1 mapping**.

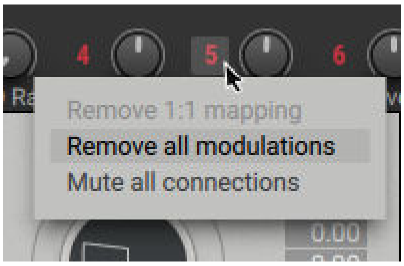


The assignment is now removed and the Macro control is blank.

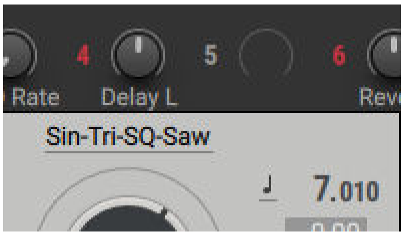
When a Macro is assigned to multiple parameters, you can remove all assignments at once.

To remove all modulations from a Macro control:

1. Right-click the Macro number, located to the left of the corresponding control knob.



2. Select **Remove all modulations**.



The assignments are now removed and the Macro control is blank.

You can mute Macro controls without removing the assignments. This is particularly useful if you want to bypass the modulation assignments for a particular Macro. This provides a quick AB reference for your modulation assignments.

1. To mute a Macro, follow the steps above, but select **Mute all connections**.

8.5. Renaming Macros

The Macro controls are automatically named when they are assigned to a parameter. In some instances, for example, when multiple parameters are assigned to the same Macro, you may want to rename the Macro to represent the full function of the control.

To rename a Macro control:

1. Double-click the Macro name, located under the corresponding control knob, to highlight it.

2. Type the new Macro name in the text field.

9. VOICE PAGE

The Voice page offers control over a range of parameters relating to voice setup and polyphony within MASSIVE X. Unison enables you to play several internal voices on top of one incoming note, and further controls allow you to modify these additional voices, producing powerful musical results. Voices can be detuned, stacked and spread to create a thicker, fuller, choir-like effect. Micro detuning can be used to recreate the often desired drifting and detuning qualities unique to analog synthesizers.

The Harmonizer section uses the additional voices to create harmonies, as determined by the selected Harmonization/Chord set. Modulating these parameters offers great creative potential, enabling you to explore different harmonies and create bold transitions.

Whether you are creating thicker, stacked sounds or chord-structures, the Voice page is a powerful tool to explore the full potential of the MASSIVE X synth engine.

9.1. Global Tune

In this section you can determine the global tuning and glide settings. The Tune parameter adjusts MASSIVE X's tuning in semitones and cents. The **Glide** button enables the glide effect (portamento), which has further options for customization in the Polyphony section of the Voice page. For more information on glide, see Glide.

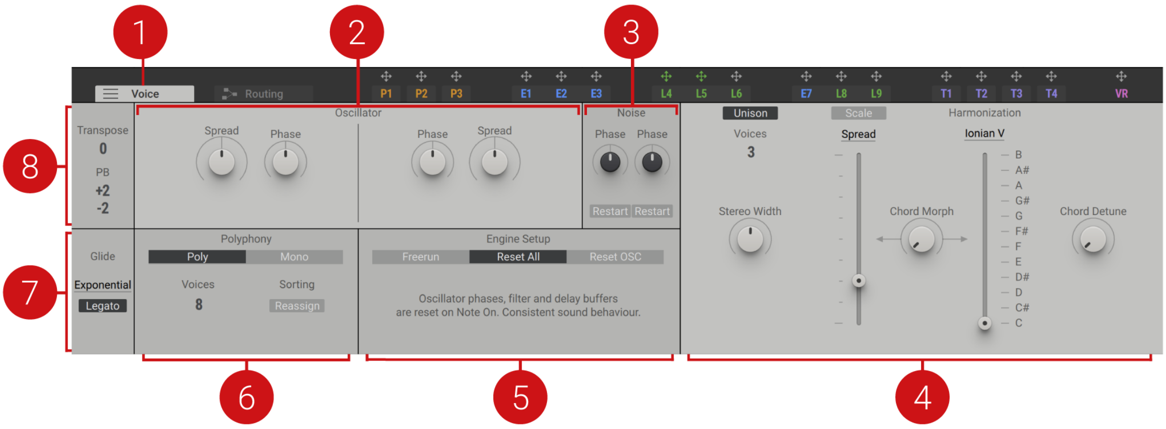
(**1**) **Tune**: Determines the global tuning of the oscillators in semitones and cents, from -64.000 to +64.000. The pitch can be modulated by routing a modulation source to either of the two modulation slots below.

(**2**) **Glide On/Off**: Activates or deactivates the glide effect.

(**3**) **Glide Time**: Adjusts the time it takes to glide from the first note pitch to the following note pitch. When the knob is turned fully left, there is no glide and the pitch will jump suddenly to the next. As the knob is turned right, the glide time increases, making the transitions slower between notes.

9.2. Overview of the Voice Page

This section provides an overview of the Voice page.



(**1**) **Voicing Tab**: Opens the Voice page in the editor.

(**2**) **Oscillator**: Controls the relative phase of the two oscillators, and the phase spreading of the added unison voices for both oscillators. For more information, see Oscillator.

(**3**) **Noise**: Determines the start position and restart behaviour of the noise samples. For more information, see Noise.

(**4**) **Unison**: Determines the unison play behaviour of MASSIVE X. For more information, see Unison.

(**5**) **Engine Setup**: The Engine Setup defines the Oscillator/Noise restart behavior and voice stealing approach. For more information, see Engine Setup.

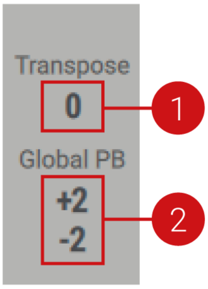
(**6**) **Polyphony**: Defines the polyphony and playback behavior. For more information, see Polyphony.

(**7**) **Glide**: Features parameters related to the glide effect (portamento). For more information, see Glide.

(**8**) **Global Voice**: Contains parameters that determine the global transposition and pitch bend settings. For more information, see Global Voicing.

9.3. Global Voicing

This section contains parameters that are related to the transposition, global pitch bend settings, and glide behavior. **Global PB** defines the way MASSIVE X responds to your master keyboard's pitch bend wheel, or corresponding MIDI controller. It provides controls for the upper and lower values of the pitch bend range when the pitch bend wheel of the master keyboard is at full up or full down position. Use this feature to set the maximum amount of pitch bend that can be applied to your sound. Setting a down value higher than the up value will invert the action on your pitch bend wheel.



(**1**) **Transpose**: Transposes the global pitch of the synthesizer in a range of -24 to +24 semitones (2 octaves).

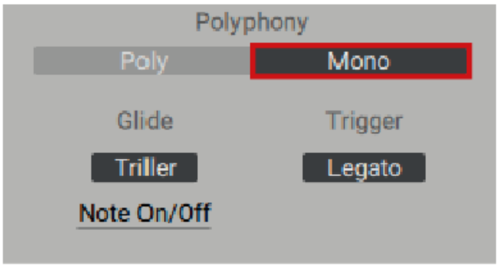
(**2**) **Global PB**: Determines the upper and lower values of the pitch bend control, in a range from -96 to +96 semitones.

9.4. Polyphony

This section allows you to define the polyphonic scheme of the MASSIVE X synth engine, which operates in either **Mono** (monophonic) or **Poly** (polyphonic) mode.

Mono Mode

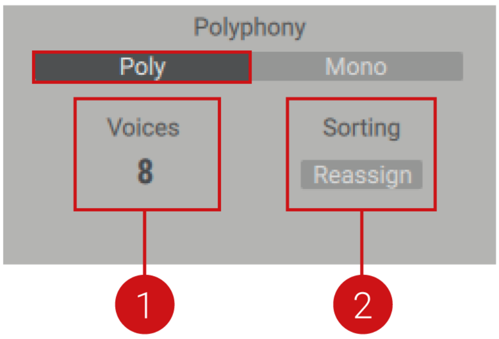
In **Mono** mode, only one note can be played at a time as with classic vintage monophonic synthesizers. However unison voices can still be stacked on top of this note creating huge sounds or even harmonies.



To learn more about the controls available in **Mono** mode, see Glide.

Poly Mode

Poly mode allows to play up to 64 notes at a time. The maximum number of **Voices** is set in the **Poly** panel. This is not including the unison voices, which can be stacked on top of this. Keep in mind that a higher number of voices also leads to a higher CPU load.

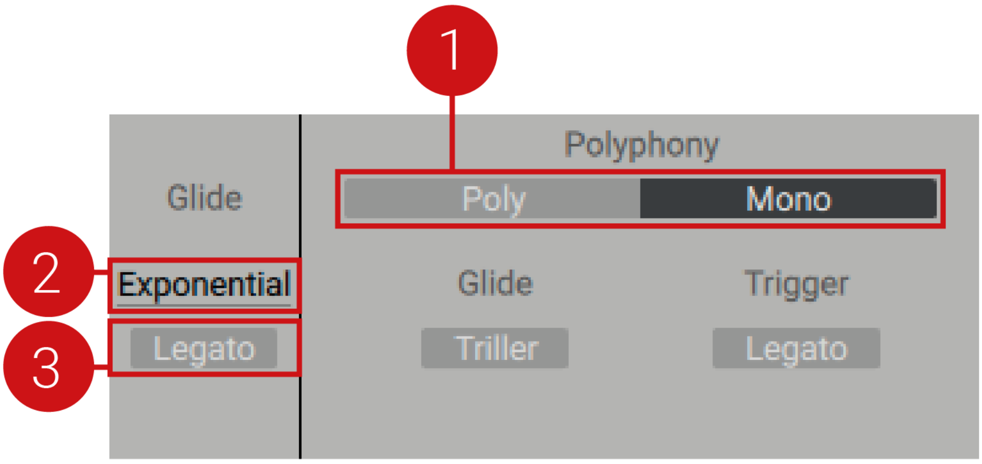


(**1**) **Voices**: Determines the maximum number of voices that can be played at the same time before voice stealing kicks in (1-64).

(**2**) **Voice Sorting**: Selects between **Rotate** and **Reassign** modes. **Rotate** is the most common mode in polyphonic synthesizers. Each new note is distributed to a new voice ID. In this case, its possible for two notes with the same pitch to be playing, a feature that is not possible with an acoustic instrument like piano or vibraphone. **Reassign** mode detects when you play the same note and allocates the same voice to the same note. This is good for piano and is particularly useful when you are trying to mimic acoustic behavior, as it won't cut off the pitch.

9.5. Glide

The Glide parameters determine the pitch transition between sequentially played notes, often called portamento. The **Glide** button and the **Glide Time** control are located on the upper left panel.

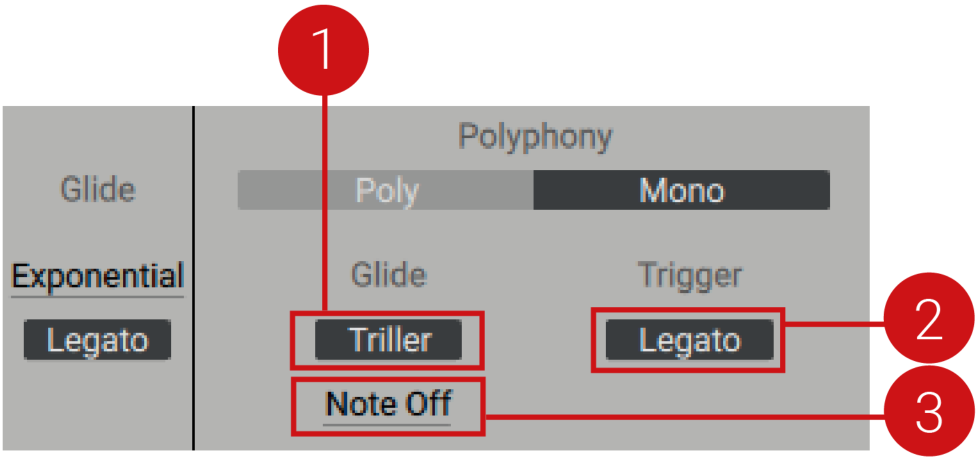


(**1**) **Polyphony**: Selects the operating mode (**Mono**, **Poly**). In **Mono** mode, the glide time is constant rate, and in **Poly** mode, it is constant time. The glide behaviour and available controls differ depending on the mode selected. Change the mode by clicking on the tab.

(**2**) **Glide Shape**: Selects the style of glide transition (**Linear**, **Exponential**, **Inverse Exp**). A **Linear** setting creates a straight, even transition. **Exponential** produces the equivalent of an analog 1-pole low-pass filter curve. Typically, this is the best choice for classic glide. **Inverse Exp** flips the characteristics of the LP glide, a feature not possible using analog filters.

(**3**) **Legato**: When active, the pitch glide only occurs when more than one note on the keyboard is pressed. When **Glide** is on and **Legato** is deactivated, a glide will always play on every note.

The following Glide parameters are only available if MASSIVE X is operating in **Mono** mode:



(**1**) **Triller**: Triller takes effect whenever a legato note is played, independent from the general **Legato** option. When active, Triller mimics the behavior you would typically find in a classic analog synth. If a legato note is released, the pitch glides back to the held note. When deactivated, all notes are killed when the legato note is released.

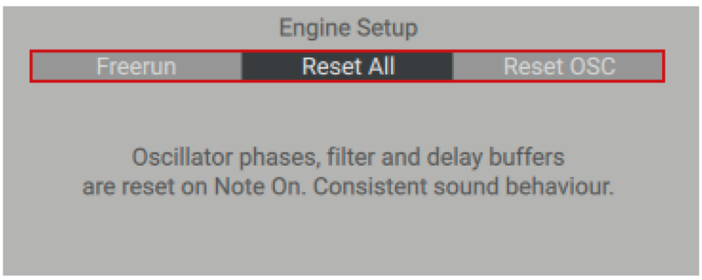
(**2**) **Trigger Legato**: Determines the envelope restart behavior during legato play. When deactivated, the envelopes and LFOs restart every time a note is played. When active, the envelopes are not re-triggered during legato playing.

If **Legato** and **Triller** are active in **Mono** mode, the Glide options menu is shown, with more detailed options for the pitch glide.

(**3**) **Glide options**: Selects one of three options (**Note On**, **Note On/Off**, **Note Off**) that determines the glide behavior in response to MIDI. When **Note On** is selected, the glide only occurs when notes are pressed. When the key is released, the note pitch immediately falls back without gliding. When **Note Off** is selected, the glide only occurs when notes are released. **Note On/Off** triggers the glide when notes are pressed and released.

9.6. Engine Setup

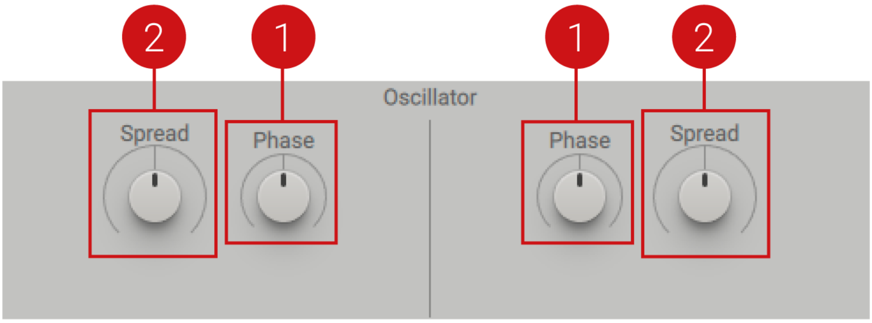
The Engine Setup section provides a selection of three modes (**Freerun**, **Reset All** and **Reset OSC**) that determines the reset behavior of the oscillator phases, filter, and delay buffers.



When **Reset OSC** is selected, the oscillator phases are restarted with each new incoming note. **Reset All** adds resetting of the filter and delay buffers, for example, in the Comb filter and Track Delay. The benefit is consistent sound behavior, where the same note will always sound the same, for example, if a filter is self-oscillating with a maximum resonance setting, **Reset All** resets the filter and when a new note is played it will require time for the filter to build up self-oscillation again. In **Freerun**, the self-oscillation would prevail, as the oscillator and filter resetting is deactivated. **Freerun** is the traditional mode that aims to recreate analog oscillator and filter behavior.

9.7. Oscillator

This section becomes available if the Engine Setup is operating in **Reset All** or **Reset OSC** mode. It allows you to set the start phase of the two oscillators and the phase spread of the additional unison voices. For information about the Oscillators, see Wavetable Oscillators.

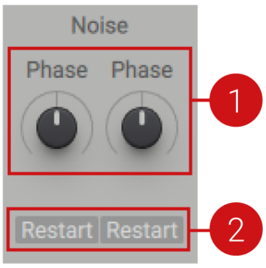


(**1**) **Phase**: Adjusts the restart phase of the oscillator. In middle position (0 degrees), the oscillator starts from the original phase. Turned fully left, the phase starts at -180 degrees and turned fully right, the phase starts at +180 degrees. Usually center position is used.

(**2**) **Spread**: Spreads out the starting phases for the added unison voices. At minimum position all voices start at the same phase position.

9.8. Noise

This section is where you determine the phase restart position of the Noise sources. For more information of the Noise sources, see Noise.

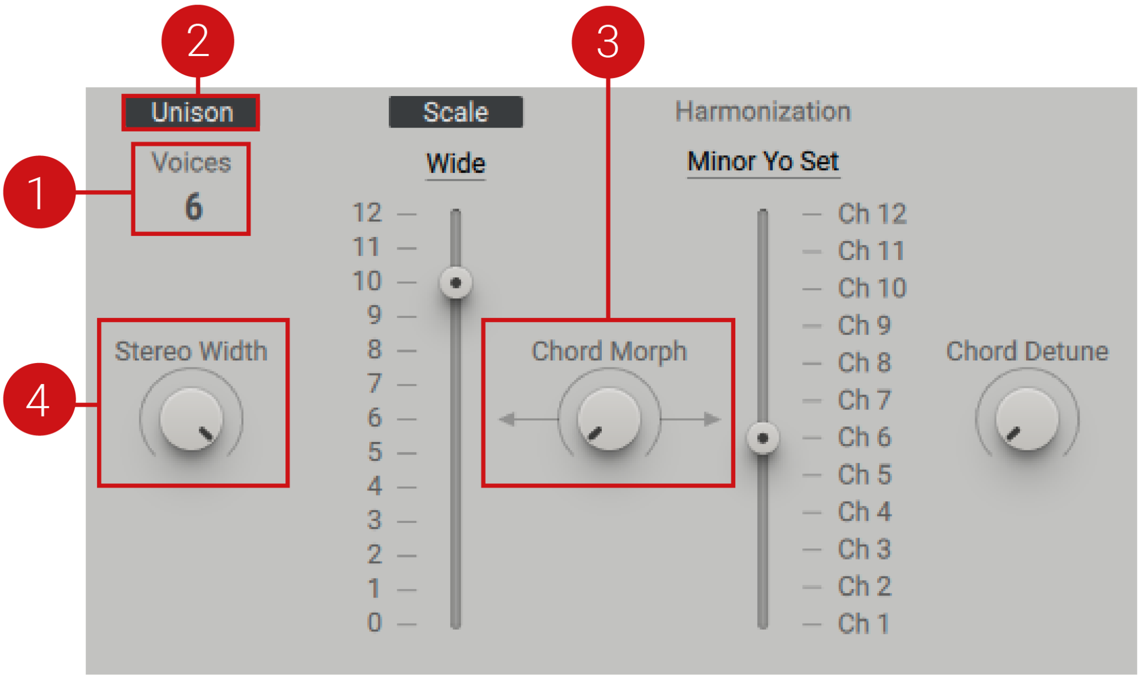


(**1**) **Phase**: Adjusts the start position for each noise source.

(**2**) **Restart**: When active, the noise source restarts at the set phase position, each time a key is pressed.

9.9. Unison

The Unison section can be used to fatten up and add life to your sounds. Additional voices with slightly different tunings can be used to recreate analog style "beating", the sound of multiple oscillators drifting in and out of phase. On the other side of the spectrum, the Unison section can be used to create chords and harmonies on top of single notes played. The additional unison voices comprise the full voice architecture as set up in the Routing section and are not limited to just the oscillators.



The left fader is used for detuning, to create the traditional unison effect that makes your sound thicker. On the right is the Harmonization fader, which allows you to choose the key when in Harmonize mode or a chord set when in Chord mode.

The Unison section is where you can determine the internal structure of voices, when triggered by one note from the MIDI keyboard. Grouping voices in different voice containers allows you to save notes, ensuring they can't be stolen. The idea of voices with voices inside comes from the inherent voicing limitations of analog and digital synthesizers, and the need to make use of all available voices.

The **Voices** parameter enables you to adjust the number of voices that will be played for each key that is pressed on the keyboard. When set to 1, only one voice will be played when a key is pressed and no unison effect will be heard. If the value is set higher, the corresponding number of voices will be triggered when a key is pressed. This stacking effect creates a thicker sound. Modifying the additional voices by spreading them across the stereo field creates a wider, fuller sound.

(**1**) **Voices**: Sets the number of voices (1-6) for Unison mode. When **Voices** is set to 1, there is no unison effect.

(**2**) **Unison**: Activates or deactivates Unison mode. When on, the Unison section and all its related controls are available.

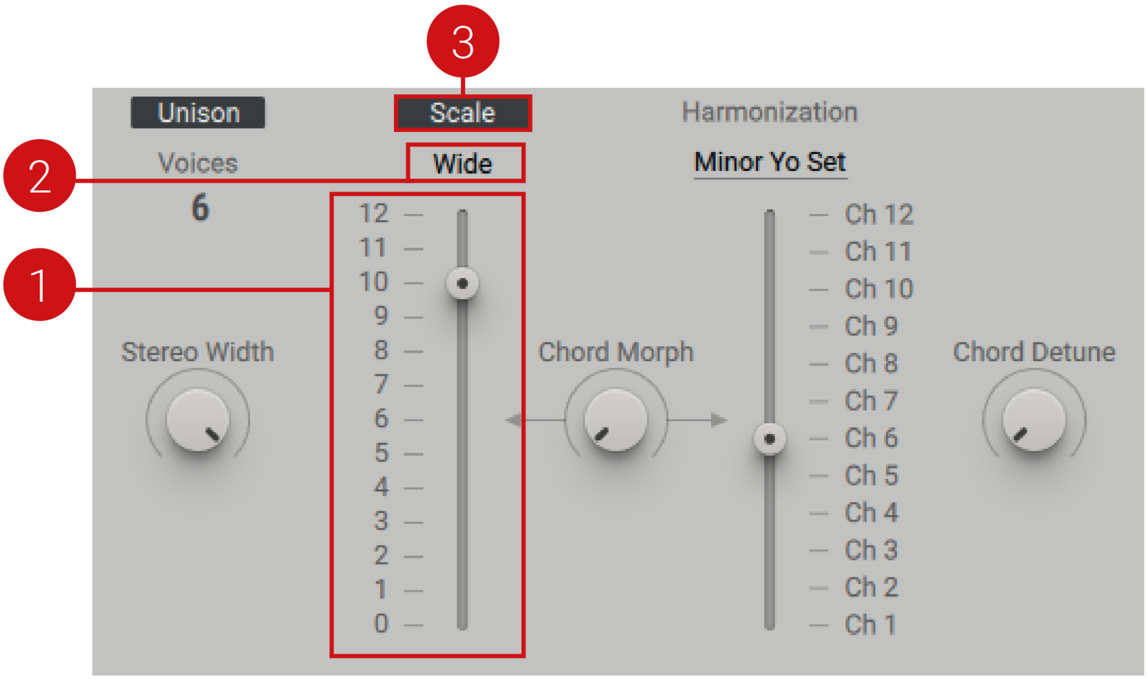
(**3**) **Chord Morph**: Morphs the tunings of the unison voices. This can create dramatic effects like the THX sound effect.

(**4**) **Stereo Width**: Adjusts the stereo spread of the voices. Turned fully left, all voices are centered mono. As the knob is turned right, the voices are spread in the stereo field.

9.9.1. Detune

The left side of the Unison section is where you can set detuning, and control the stereo spreading of the added unison voices. The result is more typical of the classic unison sound.

Spread and **Wide** tune modes can produce anything from slightly detuned moving timbres to musical intervals, creating a bigger, thicker sound. Through the inherent phase shifting of detuned oscillators sounding together, new timbres are produced. This is recreated in MASSIVE X by detuning unison voices slightly to create a choir-like effect. The strength of the effect is determined by the amount of spread and detuning applied, as well as the number of voices available.



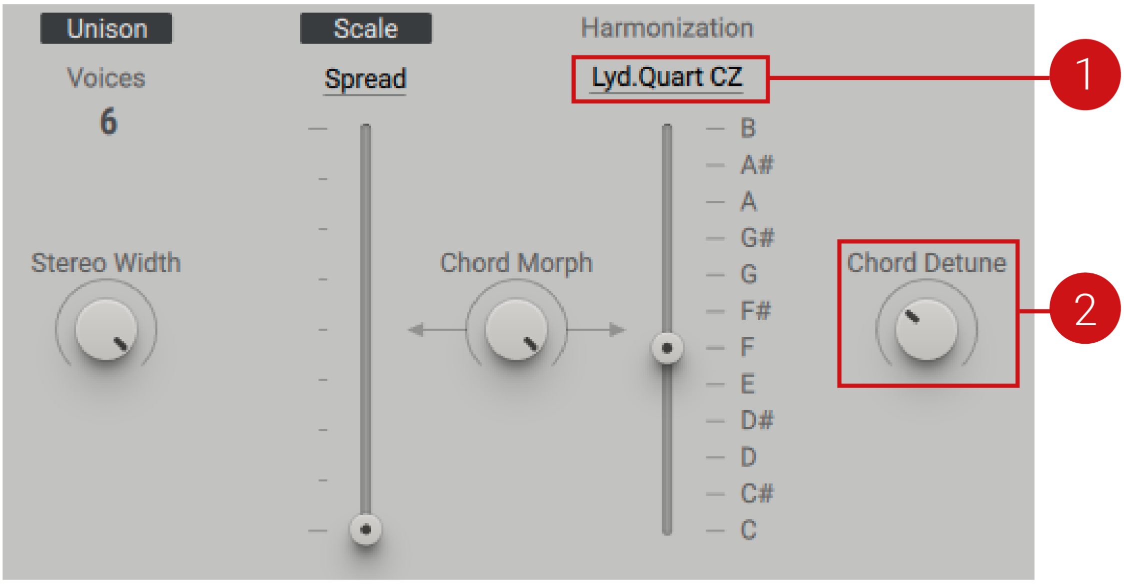
(**1**) **Spread Fader**: Adjusts the general amount of unison detuning. In **Wide** mode the fader has a semitone scale. Click on a number within the scale to choose perfect semitone offsets.

(**2**) **Spread Mode**: Selects from one of two modes (**Spread**, **Wide**) that determines the range of detuning. **Spread** mode is for micro detuning, and is fundamentally designed to make the signal sound more alive. Unequal spreading produces a less predictable sound. This effect can be properly heard with more than 3 voices and produces the best results when 4, 5 or 6 voices are selected. **Wide** mode offsets the tuning of unison voices by a maximum range of one octave. The equal distribution of the spreading produces a pattern in the beating that is more distinguishable. When **Wide** mode is selected the pitch can be shifted to a maximum of 1 octave.

(**3**) **Scale**: When enabled, the played voices are repitched according to the selected Harmonization set.

9.9.2. Harmonization

The Harmonization section allows you to build harmonies and perform within specific scales. Selecting between harmonize and chord modes will dictate the function of the right fader. In harmonize modes, the fader sets the key of the musical scale. In chord modes, different chord variations are stored across the fader.

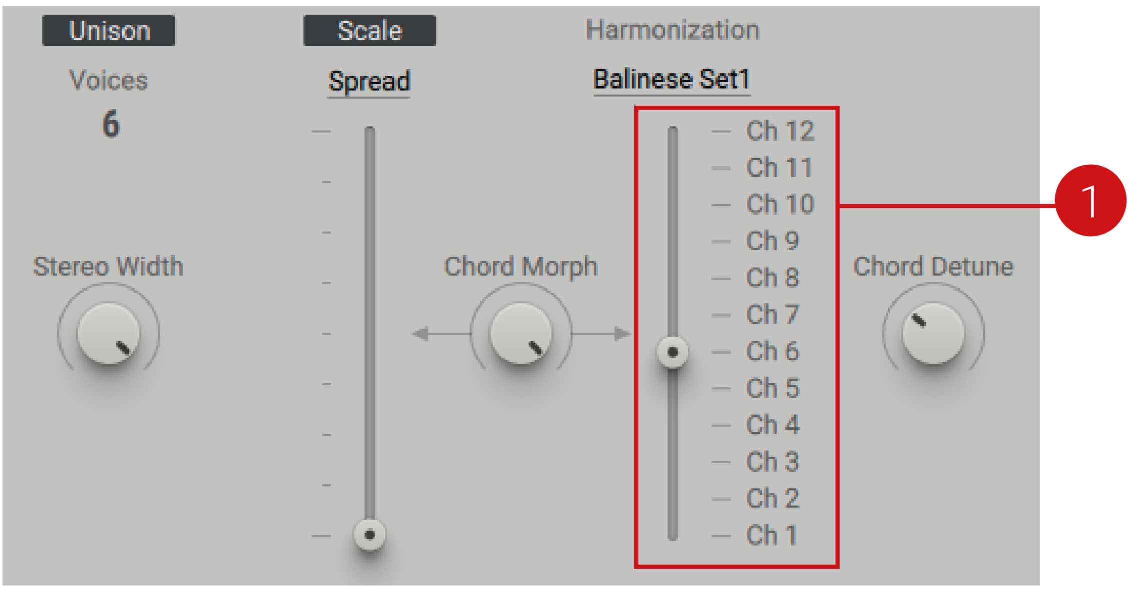


(**1**) **Harmonization menu**: Selects between harmonize and chord modes. In harmonize modes the fader determines the root note. In chord modes, the fader is comprised of set intervals to create chords based on the number of selected voices.

(**2**) **Chord Detune**: Slightly detunes the chord for a more pleasant or lively sound. Use the Tracker to control the chord and spreading.

Chord Modes

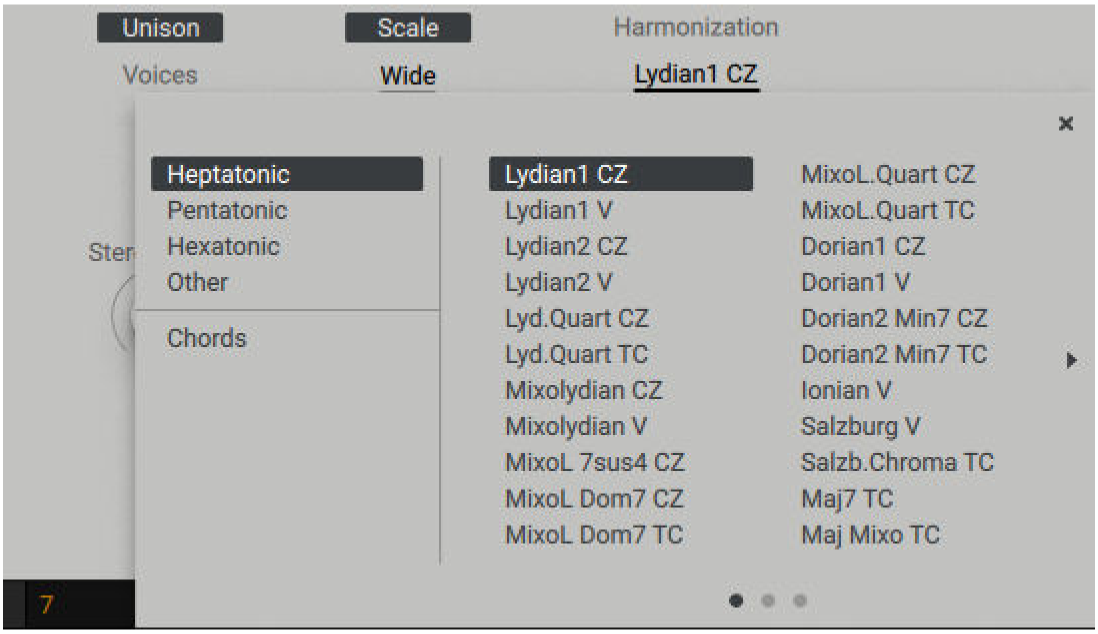
Chord modes enable you to play a variety of chords from a single MIDI note, providing parallel chords or transposition settings on top of the main pitch. When a chord mode is selected, the fader stores 12 variations of different chord settings. Adding the Performer to the chord fader allows you to tune through the different buffers, effectively jumping through your stored chords. The Tracker is another powerful option for modulation, allowing you to set the slider to an individual setting per note.



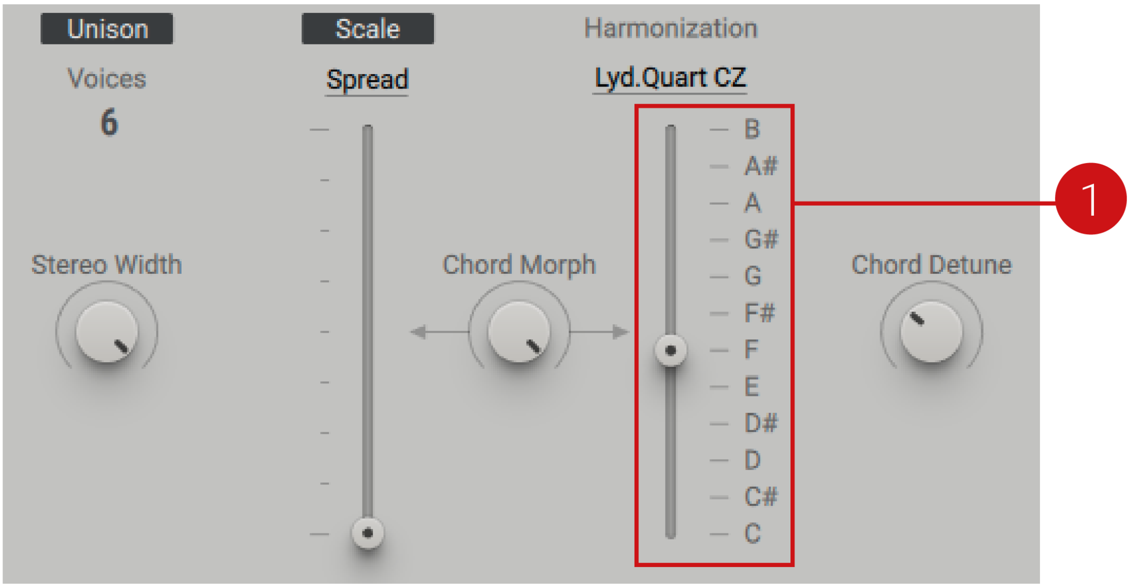
(**1**) **Chord fader**: Stores 12 variations of different chord settings.

Harmonize Modes

Harmonize modes allow you to select from a range of scales that determine the available set of notes from which you can build melodies and harmonies. Pitches from your MIDI device that are not within the set scale will be moved up or down, depending on the type of scale correction. Three settings determine the way scale correction is implemented; Catch Zone **(CZ)**, Variation **(V)** and Transition **(TC).**



- **Catch Zone** corrects notes when a note is not within the specified scale. In sharp scales, the higher note will be corrected and moved down, for example, in the Lydian scale, C# becomes C. In flat scales, the correction is reversed and notes are moved up.
- **Variation** rearranges the chord structure instead of correcting the chord, using the free buffers for a new sound.
- **Transition** mode is to be used for transitional chords, notable in classical music. Playing the transitional chords alone can produce undesired results as they are intended to be used to fill within the scale. The introduction of 4th chords leads to a more open sound.



(**1**) **Harmonize fader**: Determines the fundamental of the scale.

10. ROUTING

MASSIVE X is a semi-modular synthesizer with an open architecture. This means you can freely arrange and connect its modules to facilitate a wide range of different synthesis techniques, giving you the freedom to design and explore sounds without constrains.

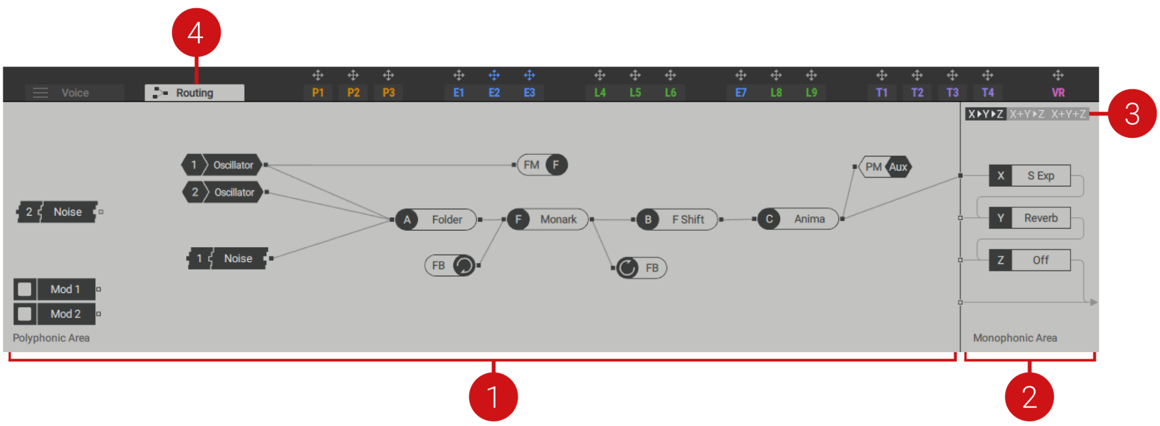
You can use and combine techniques like wavetable synthesis, phase modulation (PM, also called FM or frequency modulation), subtractive synthesis, physical modelling and various types of wave shaping, or distortion.

Modules are arranged and connected on the Routing page. Here you can combine all of MASSIVE X's sound generators and processors to create your sound.

10.1. Overview of the Routing Page

The following overview shows the Routing page's two main areas, the **Polyphonic Area** and the **Monophonic Area**. The Polyphonic Area is used to define the signal path that is processed independently for every single voice you play. The Monophonic Area contains three Stereo Effects that are globally applied to the sum of all polyphonic voices.

MASSIVE X opens with the preset **Init - Massive X**, which offers a pre-configured routing as shown in the screenshot below (based on subtractive synthesis with Wavetable oscillators). This way you can immediately start creating sounds without having to make connections. Alternatively, you can load the preset **Init - Massive X Blank** to start with a blank routing and make your own connections from scratch.



(**1**) **Polyphonic Area**: Contains modules that you can use to define the signal path of the polyphonic voices. In order to complete the signal path, one or more module outputs need to be connected to the inputs of the Monophonic Area (**2**). The following modules can be found in the Polyphonic Area:

- **Generators and processors**: The Oscillators, Noise sources, the Filter, and the Insert Effects are available in the Polyphonic Area. Black modules represent generators, gray modules represent processors. The icons shown on the modules can also be found on the corresponding module panel. For more information, see Generators and Processors.

- **PM Aux bus**: Makes the phase modulation inputs of the Oscillators accessible in the routing. This way you can use any source in the signal path to apply phase modulation to the Wavetable oscillators and experiment with noise, feedback, and effects in this application. For more information, see PM Aux Bus.
- **FB (Feedback) loop**: Makes the global feedback loop accessible in the routing. This way you can create a polyphonic feedback loop around modules to add chaotic and non-linear behavior. This feature is also useful for physical modeling sounds, especially when combined with the Comb filter. For more information, see Feedback Loop.
- **Mod (Modulation) modules**: Any modulation source can be assigned to the Modulation modules by dragging and dropping. This way you can use the modulation sources as generators in the signal path. For example, you can use the Switcher LFO in OSC mode as an additional oscillator, or the Exciter Envelope as an exciter for the Comb filter. For more information, see Modulation Modules.

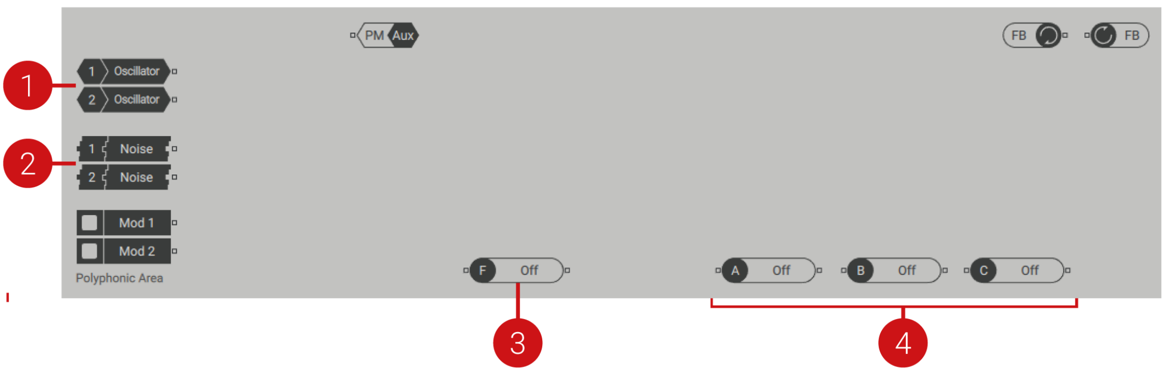
(**2**) **Monophonic Area**: Sums the polyphonic voices and applies the three Stereo Effects X, Y, and Z before sending the output signal to the host. The Monophonic Area provides four inputs, one for each Stereo Effect as well as a single input that is directly sent to the host.

(**3**) **Routing Options**: Three different routing options define the order of the effects in the signal path: **X > Y > Z** chains the three effects, **X + Y > Z** sends the sum of the X and Y effects to the Z effect, and **X + Y + Z** sums all three effects.

(**4**) **Routing Tab**: Opens the Routing page in the editor.

10.2. Generators and Processors

The generators and processors available in the Polyphonic Area of the Routing page are the basic components you can use to build your sound. They consist of the Oscillators, Noise sources, the Filter, and the Insert Effects.



(**1**) **Oscillators**: The Wavetable oscillators 1 and 2 are generators that each feature a single output. They can be connected to the processors and buses in the Polyphonic Area, or directly to an input of the Monophonic Area. For more information about the Oscillators, see Wavetable Oscillators.

(**2**) **Noise sources**: The Noise sources 1 and 2 are generators that each feature a single output. They can be connected to the processors and buses in the Polyphonic Area, or directly to an input of the Monophonic Area. For more information about the Noise sources, see Noise.

(**3**) **Filter**: The Filter is a processor that features a single input and a single output. It can be connected to the generators, processors, and buses in the Polyphonic Area, or directly to an input of the Monophonic Area. The Filter types Asimov, Blue Monark, Groian, Scanner feature a special FM bus in the routing:

You can connect any source in the signal path here to apply audio rate modulation to the filter frequency, also called filter FM (frequency modulation). Filter FM produces rich harmonics and distortion effects. For more information about the Filter, see Filter.

(**4**): The three Insert Effects A, B, and C are processors that each feature a single input and an single output. They can be connected to the generators, processors, and buses in the Polyphonic Area, or directly to an input of the Monophonic Area. By selecting OSC or PM OSC for the Insert Effects, you can also use them as generators. For more information about the Insert Effects, see Insert Effects.

10.3. PM Aux Bus

The PM Aux bus makes it possible to use any source in the signal path to apply phase modulation to the Wavetable oscillators. For example, you can experiment with noise, feedback, and effects in this context. For more information about phase modulation, see Phase Modulation.

The PM Aux bus has a dedicated module in the Polyphonic Area of the Routing page and additional controls on the panel of the Wavetable oscillators:

(**1**) **Aux assignment**: Assigns the signals received at the input of the PM Aux bus in the Polyphonic Area of the Routing page to the corresponding Wavetable oscillator.

(**2**) **Aux modulation amount**: Adjusts the amount of phase modulation applied from the PM Aux bus to the Wavetable oscillators.

(**3**) **PM Aux bus**: This bus features a single input that sends signals to the phase modulation function of the Wavetable oscillators. It can be connected to the generators and processors in the Polyphonic Area.

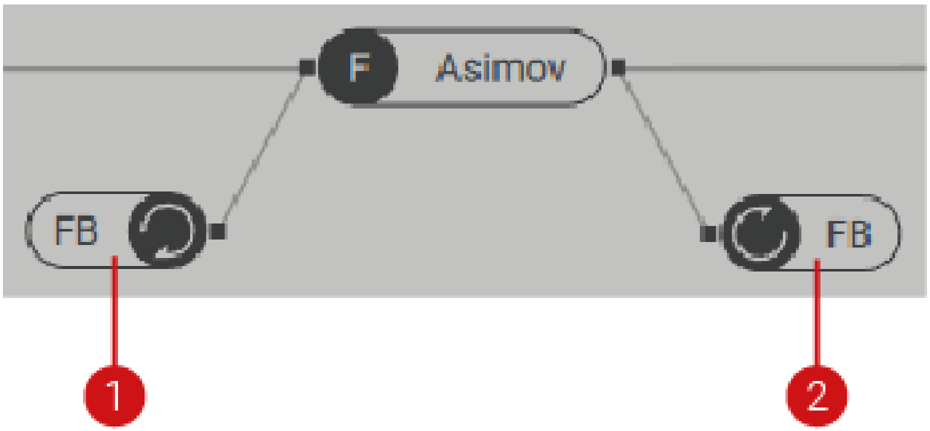
10.4. Feedback Loop

The feedback loop in the Polyphonic Area of the Routing page facilitates a connection from an output of a module to its input. Any number of modules can be chained in the feedback loop.

This way you can add chaotic and non-linear behavior to the voice, which creates organic and distorted sounds. You can use feedback to enhance the sound or as a dramatic effect, with high feedback levels causing sonic mayhem. The feedback loop is also useful for physical modeling sounds, especially when combined with the Comb filter.

Note that the feedback loop is polyphonic, meaning it is processed independently for every single voice you play. This way you can play chords and overlapping notes with your feedback sounds.

The feedback loop can be freely connected in the Polyphonic Area of the Routing page by using the two **FB** modules:



(**1**) **FB (feedback) loop output**: This bus features a single output that receives signals from the feedback loop input. It can be connected to the generators, processors, and buses in the Polyphonic Area.

(**2**) **FB (feedback) loop input**: This bus features a single input that sends signals to the feedback loop output. It can be connected to the generators and processors in the Polyphonic Area.

In the example above, the output of the Asimov filter is connected to the **FB** input (**2**), and the **FB** output (**1**) is connected to the input of the Asimov filter. This creates a feedback loop around the filter, making it sound distorted and behave in unexpected yet interesting ways.

10.4.1. Feedback Level Control

To control the amount of feedback, you can use the **FB** level control in the Amplifier module panel:

(**1**) **FB (feedback) level**: Adjusts the level, or volume of the feedback loop. This way you can control the chaotic behavior and distortion produced by the feedback.

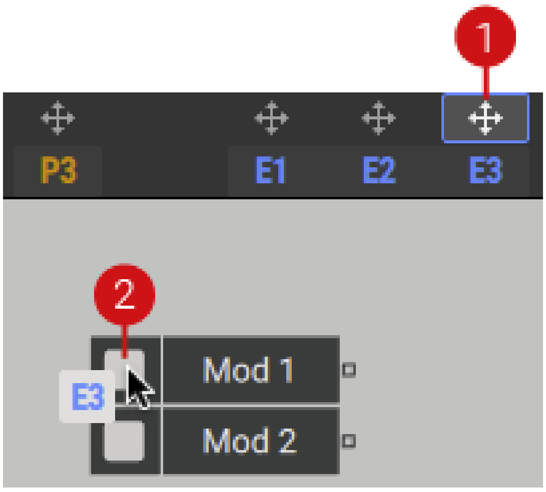
(**2**) **High-pass filter**: Enables a high-pass filter that cuts low frequency content in the feedback loop. When activated, you can avoid overloading the feedback loop with excessive bass.

10.5. Modulation Modules

The Modulation modules in the Polyphonic Area of the Routing page make it possible to use any of MASSIVE X's modulation sources as generators in the signal path. For example, you can use the Switcher LFO in OSC mode as an additional oscillator, or the Exciter Envelope as an exciter for the Comb filter.

To assign a modulation source to a Modulation module in the Polyphonic Area:

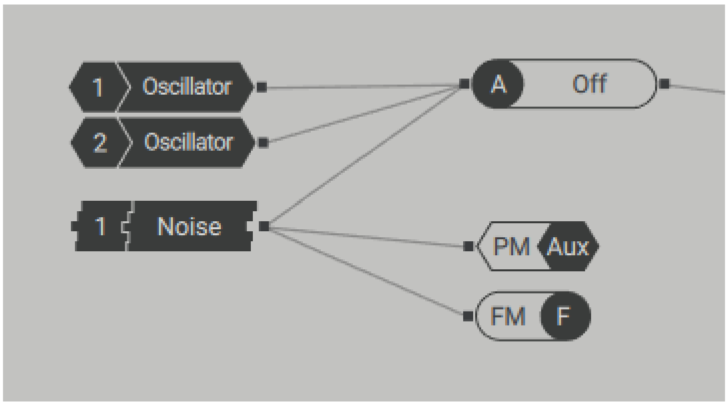
1. Click the modulation source's arrow icon (**1**) to select it, then click on the Modulation module slot (**2**) in the Polyphonic Area to assign it.



The signal produced by the modulation slot is now sent from the Modulation module's output and can be used anywhere in the signal path.

10.6. Using the Routing Page

On the Routing page, you arrange and connect the individual building blocks, or modules, that make up the synthesizer. Connections between the modules' inputs and outputs are established using wires. Outputs can be connected to any number of inputs and vice versa. This way you can distribute signals to multiple destinations or mix multiple outputs into the same input:



10.6.1. Routing Workflows

To make a connection between modules:

1. Click on an output to show all available inputs it can be connected to.
2. Click on the input you want to connect the output to.

Alternatively, you can click and drag from an output to an input to connect them.

To make an exclusive connection to an input, meaning that all existing connections to the input will be removed:

1. Click on an output to show all available inputs it can be connected to.
2. Right-click on the input you want to connect the output to exclusively.

To delete a wire:

- Double-click the wire you want to remove.

To delete all connections from a module:

- Double-click the module that you want to remove all connections from.

You can also disable, or bypass any number of modules directly on the Routing page. This provides a quick way of listening to the effect a generator or processor is having on the sound.

To bypass a module while keeping its connections intact:

- Right-click the module you want to bypass.

11. WAVETABLE OSCILLATORS

Two Wavetable oscillators form the basis for sound generation in MASSIVE X. These oscillators produce sound from digitally sampled, single-cycle waveforms that are arranged in a table, called wavetable. Scanning through the different waveforms in the table gives you access to a wide range of tones and sound colors that can be used in various musical forms.

Wavetable synthesis operates on a two-dimensional axis. On the x-axis, playback of the individual waveforms occurs. This playback, or readout, is controlled by the internal phase of the oscillator, which follows the pitch of incoming MIDI notes. The y-axis represents the wavetable itself, and the different waveforms that are stacked one above the next. Scanning the different waveforms along the y-axis produces variations in harmonic content. By modifying the readout of the waveform, you can create intricate timbral variations in our sound.

The beauty and power of wavetable synthesis becomes most evident when modulation is introduced to scan the table of waveforms. This process can be triggered by an envelope or modulated by an LFO, as well as manually altered with a controller. The dynamic results of this morphing is a defining sonic character in this style of synthesis. Wavetable synthesis inherently facilitates complex combinations of different waveforms, and this sonic potential is further enhanced with the **Wavetable Modes**, which dramatically impact that readout of the wavetable. The modes also determine the available controls and menus, significantly altering the behavior of the oscillator.

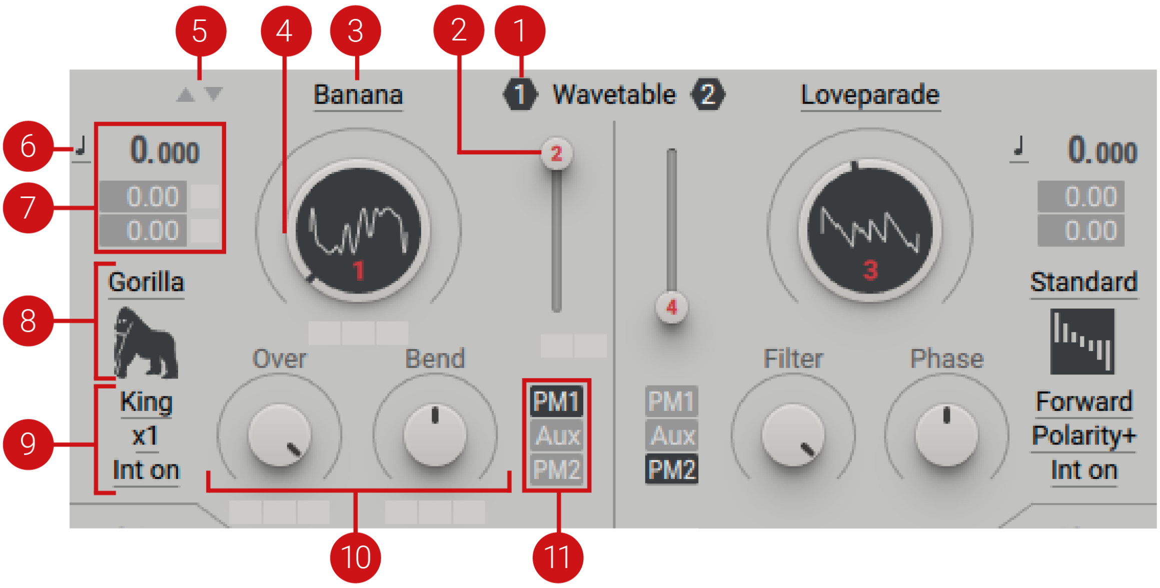
The Wavetable oscillators are defined by the selected wavetable and the wavetable mode. A diverse collection of wavetables has been crafted to cover a broad range of end uses, from more typical waveforms and PWM to FX and Harmonics. To start exploring the Wavetable oscillators, browse through the wavetables in the **Wavetable Menu** and experiment using these with the different **Wavetable Modes**.

Scanning through the waveforms included in the wavetable is done using the **Wavetable Position** control. Turning this knob scans through the waveforms included in the selected wavetable, the result of which is graphically displayed on the control. Intricate sweeping effects can be achieved by routing a modulation source to the **Wavetable Position knob**, producing a waveform that is constantly morphing and evolving.

The Wavetable oscillators can be used to replicate the timbre and articulation of acoustic instruments, or to shape new and abstract sounds. Complex drones, shimmery pads, abrasive leads and percussion can all be crafted using the oscillators as the core building block. When combined with the extensive modulation and routing options found in MASSIVE X, alongside the PM/Aux bus, noise sources, effects, and Modulators, the potential is unbounded.

11.1. Overview of the Wavetable Oscillators

This section provides an overview of the Wavetable oscillators. Controls for Wavetable oscillator 1 are numbered, and these controls are mirrored for Wavetable oscillator 2.



(**1**) **Bypass**: Bypasses the Wavetable oscillator. This will in effect turn the oscillator off.

(**2**) **Level fader**: Adjusts the output volume of the Wavetable oscillator.

(**3**) **Wavetable menu**: Opens a menu with available wavetables. The categories are presented on the left and the individual wavetables on the right. For more information on the wavetable categories, see Wavetables Menu.

(**4**) **Wavetable Position**: Scans through the waveforms included in the selected wavetable. The individual waveforms represented in each table range from 2-128. Changes in the waveform are visually represented.

(**5**) **Browse Arrows**: Hover over the Wavetable menu (**3**) to expose arrows that allow you to browse up and down through the wavetables, without opening the menu.

(6) Pitch Mode: Selects one of three modes (Keytrack, Fix, Ratio) that determines the response of the oscillator in relation to incoming MIDI pitch.

- In **Keytrack** mode, the pitch is locked to the main MIDI pitch.
- **Ratio** mode multiplies or divides the MIDI pitch in relation to the ratio setting. For example, a ratio of 3 adjusts the pitch to 3 times the frequency, producing the 3rd harmonic.
- **Fix** mode sets the fixed tuning of the oscillator, disregarding the incoming MIDI pitch. This mode displays MIDI note numbers, with the default set to 60 (middle C).

(7) Pitch: Adjusts the tuning of the oscillator in semitones and cents. The **Pitch** can be modulated to produce vibrato and arpeggio effects by routing a modulation source to either of the two modulation slots below.

(**8**) **Wavetable Modes**: Selects one of ten operating modes (Standard, Bend, Mirror, Hardsync, Wrap, Formant, ART, Gorilla, Random, Jitter), which determine the readout of the wavetables. For more information, see Wavetable Modes.

(**9**) **Wavetable Mode settings**: Menus and switches that vary, depending on the mode selected. Each mode has a specific set of menus and switches that directly relate to its operation. For more information on the settings relating to each specific Mode, see Wavetable Modes.

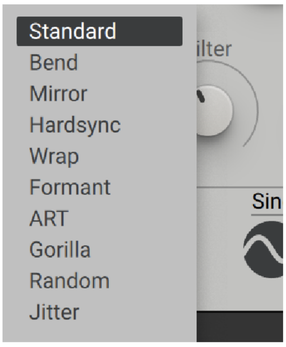
(**10**) **Wavetable Mode controls**: Two control knobs that vary, depending on the selected mode. Each mode has two knobs controls that directly relate to its operation. For more information on the controls relating to each Mode, see Wavetable Modes.

(11) PM/Aux assignment: When enabled, the Phase Modulation oscillators and/or Aux input are assigned to the corresponding Wavetable oscillator. For more information on the PM/Aux bus, see Phase Modulation.

11.2. Wavetable Modes

The Wavetable oscillator operates in one of ten different modes (Standard, Bend, Mirror, Hard-sync, Wrap, Formant, ART, Gorilla, Random, Jitter), which determine the readout of the waveta-bles and hence the behaviour of the Wavetable oscillator. The modes have a significant impact on sound and are essential to unlocking the full potential for sound design using MASSIVE X. Every mode features two dedicated parameters for real time manipulation and settings to define the be-haviour.

The following section provides an overview of each available mode and its corresponding controls, as well as suggestions for suitable end uses.



11.2.1. Standard Mode

The default Wavetable mode is Standard. Similar to Spectrum mode in original MASSIVE, the Fil-ter control is used to reduce the higher frequency harmonics heard in the selected waveform. By scanning through the set of band-limited waveforms that are usually assigned to specific pitches, a lowpass filtering effect is achieved.The function is similar to a low-pass filter cutoff, although the algorithm behind it is different from a standard filter design. When this filter is applied to a square wave it gradually becomes a sine wave, i.e. only the fundamental of the sine wave remains.

Standard mode contains the following parameters and controls:

(**1**) **Wavetable Modes**: Selects from a menu of ten operating modes that determine the readout of the wavetables.

(**2**) **Phase Direction**: Switches between two settings (Forward, Backward) which determine how the waveform is read out.

(**3**) **Polarity +/-**: Switches from positive to negative polarity, meaning the waveform is flipped.

(**4**) **Internal Phase On/Off**: When **Int on** is selected, the engine uses the main phase. Selecting **Int off** fixes the oscillator frequency at 0 Hz, bypassing the main phase and turning the oscillator into a waveshaper. The shaper must be used in conjunction with the PM oscillators or the PM Aux bus. The level of the PM oscillator and/or Aux input determines the amount of waveshaping, and the wavetable position and filter parameters control the shape function.

(**5**) **Filter**: Reduces the high frequency harmonics of the selected waveform. The effect is similar to adjusting the cutoff frequency of a low-pass filter.

(**6**) **Phase**: Adjusts the oscillator phase.

11.2.2. Bend Mode

In Bend mode, the readout curve of the wavetables can be shaped. You can raise and lower the readout speed depending on the position within the wavetable. Some parts of the waveform are compressed and other parts are expanded, as determined by the Bend, Strength and Direction parameter settings. Up-down mode can be used to create a hollow, square-like sound.

Bend mode contains the following parameters and controls:

(**1**) **Wavetable Modes**: Selects from a menu of ten operating modes that determines the readout of the wavetables.

(**2**) **Bend Curve**: Selects one of three settings (Strong, Medium, Gentle) that change the bend curve. With a **Strong** setting, the bending curve is significantly altered. **Medium** produces a less strong curve. With a **Gentle** setting the bend curve is more subtle, operating like the +/- Bend mode in MASSIVE.

(**3**) **Direction**: Selects one of three settings (Neutral, Up-Down, For-Back) that determine how the waveform is read out. With a **Neutral** setting, no directional change is applied and the waveforms are unaltered when played. **Up-Down** inverts every second cycle of the waveform. Flipping every second cycle cuts out all even harmonics (2, 4, 6, 8 etc.), meaning you cannot produce these harmonics with this setting. **For-Back** (Forward-Backwards) reads out every second cycle of the waveform backward. If the waveform is perfectly symmetrical, this has the same affect as **Up-Down** mode.

(**4**) **Filter**: Reduces the high frequency harmonics of the selected waveform. The effect is similar to adjusting the cutoff frequency of a low-pass filter.

(**5**) **Bend**: Bends the phase by accelerating and decelerating the readout of the waveform.

11.2.3. Mirror Mode

Mirror mode reads the wavetable back and forth. It has no sub menus. Exceeding a certain Ratio will force the waveform to be folded, producing a hard sync-style sound.

Mirror mode contains the following parameters and controls:

(**1**) **Wavetable Mode**: Selects from a menu of ten operating modes that determines the readout of the wavetables.

(**2**) **Bend**: Bends the phase by accelerating and decelerating the readout of the waveform.

(**3**) **Ratio**: Controls the depth ratio of the mirrored phase. You can successively change the range of mirroring, and even force it to folding, where the sonic results will enter hard-sync territory.

11.2.4. Hardsync Mode

Hardsync mode offers the classic hard sync sound, as well as other variations of this effect. Unlike with traditional analog synthesizers, no second oscillator is required to reset and achieve the effect. This mode produces a completely different sound to a standard wavetable readout. You can use the other Wavetable oscillator to add body to the sound. Alternatively, you can use the Insert oscillator with a sawtooth or square waveform in Lock mode. With a high amount setting, the sound is likely to be sharp. Soft and Grain settings can be used to smooth the reset; options that are not possible with classic analog hard sync.

Hardsync mode contains the following parameters and controls:

(**1**) **Wavetable Modes menu**: Selects from a menu of ten operating modes that determines the readout of the wavetables.

(**2**) **Window**: Selects one of three settings (Hard, Soft, Grain). **Hard** applies no smoothing window to the resetted waveform, resulting in the typical rolling sound of classic hard sync. **Soft** fades out the waveform before the reset occurs, producing a rounder and more even sound. This setting creates a smoother and softer sound sound, with a less exaggerated quality. **Grain** has an even softer effect, producing a very round waveform that fades in and out at the start of each sync. The resetting is completely smoothed out, resulting in a sound with less bite. With a round waveform, **Grain** is the same as the Formant mode found in MASSIVE.

(**3**) **Direction**: Selects one of three settings (Neutral, Up-Down, For-Back) that determines how the waveform is read out. With a **Neutral** setting, no directional change is applied and the waveforms are unaltered when played. **Up-Down** inverts every second cycle of the waveform. Flipping every second cycle cuts out all even harmonics (2, 4, 6, 8 etc.), meaning you cannot produce these harmonics with this setting. **For-Back** (Forward-Backwards) reads out every second cycle of the waveform backward. If the waveform is perfectly symmetrical, this has the same affect as **Up-Down** mode.

(**4**) **2nd level knob**: Controls the amplitude of every second resetted repetition of the cycle. Turning the knob fully left will produce the sound at an octave lower.

(**5**) **Ratio knob**: Sets the frequency ratio of the inaudible sync oscillator, which is used to reset the main oscillator.

11.2.5. Wrap Mode

Wrap mode is close to Hardsync mode, the difference between the two becomes evident when modulation is applied. When modulated, Hardsync mode will create more pitch artefacts. Starting in the centre it can create a ramp, cutting when it reaches the boundaries of the waveform.

Wrap mode contains the following parameters and controls:

(**1**) **Wavetable Modes menu**: Selects from a menu of ten operating modes that determines the readout of the wavetables.

(**2**) **Window**: Selects one of three sub-settings (Hard, Soft, Grain). The **Hard** setting applies a very subtle smoothing window to the end and start point of the fundamental cycle. Depending on the waveform, this results in the typical sound of classic hard sync. **Soft** applies a smoother window and **Grain** applies the smoothest window.

(**3**) **Direction**: Selects one of three settings (Neutral, Up-Down, For-Back) that determines how the waveform is read out. With a **Neutral** setting, no directional change is applied and the waveforms are unaltered when played. **Up-Down** inverts every second cycle of the waveform. Flipping every second cycle cuts out all even harmonics (2, 4, 6, 8 etc.), meaning you cannot produce these harmonics with this setting. **For-Back** (Forward-Backwards) reads out every second cycle of the waveform backward. If the waveform is perfectly symmetrical, this has the same affect as **Up-Down** mode.

(**4**) **Filter**: Reduces the high frequency harmonics of the selected waveform. The effect is similar to adjusting the cutoff frequency of a low-pass filter.

(**5**) **Ratio**: Sets the frequency ratio of the inaudible sync oscillator, which is used to reset the main oscillator.It operates in a centered way, so that the cycle-multiplications take place from the center of the cycle, equally to the left and right.

11.2.6. Formant Capture Mode

Formant mode manipulates the waveform so that it does not change the amplitude of the voice. This creates the impression that the waveform remains static over different pitches. Essentially, the sound is initially corrected so that it works as a static formant, erasing the original ‘Mickey Mouse’ effect. A Formant control then offers the ability to reintroduce the Mickey Mouse effect, independent from pitch. Each wavetable is imprinted with an extra metadata called Formant Center, which informs the engine where the original table was made. Formant mode uses only one specialised windowing algorithm, rather than two overlapping grains (as used in PSOLA Synthesis), for a clearer sound and lower CPU load.

This mode is most effectively used when there is a strong formant in the wavetable itself, hence why it does not work well with sine, sawtooth or square waveforms.

Formant Capture mode contains the following parameters and controls:

(**1**) **Wavetable Modes**: Selects from a menu of ten operating modes that determines the readout of the wavetables.

(**2**) **Direction**: Selects one of three settings (Neutral, Up-Down, For-Back) that determine how the waveform is read out. With a **Neutral** setting, no directional change is applied and the waveforms are always played one after the other. **Up-Down** inverts every second fundamental cycle of the waveform. **For-Back** (Forward-Backwards) reads out every second fundamental cycle of the waveform backwards.

(**3**) **2nd Level**: Controls the amplitude of the second repeat of the waveform. Turning the knob fully left will produce the sound an octave lower.

(**4**) **Formant**: This knob reintroduces the formant, or the ‘Mickey Mouse’ effect.

11.2.7. ART Mode

ART (Artificial Resonance Technology) mode utilises hard sync techniques and windowing to mimic a resonant filter. The basic idea for this mode was to create filters without filters. In contrast to standard analog modelling, the numerical approach to this mode means that more unusual filtering options can be explored. ART mode provides you with some of the fundamental sound principles of a filter, closely replicating a high resonating bandpass filter sweep. It combines this with the possibilities of a traditional wavetable scanning oscillator to create new variations of filtering sound, also expanding into the artificial filter territories that a real filter cannot achieve. This mode relies on modulation for the most effective and interesting results. It is the only mode where the FU-DB sub-mode can be found.

ART mode contains the following parameters and controls:

(**1**) **Wavetable Modes menu**: Selects from a menu of ten operating modes that determine the readout of the wavetables.

(**2**) **Window** Selects from three strength settings (Hard, Bity, Soft) that influence the shape of the impulse envelope. **Hard** is the most aggressive setting, creating a sharp sound. **Soft** is smooth sounding and the least aggressive setting. **Bity** has a balanced character and is a good starting point. The differences in the three modes can best be heard with low **Pitch** settings.

(**3**) **Direction**: Selects one of three settings (Neutral, Up-Down, For-back) that determine how the waveform is read out. With a **Neutral** setting, no directional change is applied and the waveforms are always played one after the other. **Up-Down** inverts every second cycle of the waveform. By flipping every second cycle, all even harmonics (2, 4, 6, 8 etc.) are cut out, meaning you cannot produce harmonics with this setting. **For-Back** (Forward-Backwards) reads out every second cycle of the waveform backwards. This setting will sound the same if the waveform is perfectly symmetrical. **FU-DB** (Forwards Upwards - Downwards Backwards) combines Up-down and For-back.

(**4**) **Body**: Selects one of two settings (Body, Nobody), which determine if the response of an artificial body is applied to the sound. When **Body** is active, this response adds bass to the sound. When **Nobody** is selected, you have the excitation response of an artificial filter.

(**5**) **Width knob**: Narrows or widens the envelope, mimicking the resonance impulse. The effect is similar to adjusting the resonance amount of a normal filter.

(**6**) **Pitch knob**: Adjusts the frequency of the artificial resonance, similar to the cutoff frequency on an analog filter.

To get familiar with ART mode, it is highly recommended to use a sine waveform in the Wavetable oscillator and experiment with all settings, aiming to produce some realistic filter sweeps. Then switch to a square waveform and note how much more aggressive the sound instantly becomes.

11.2.8. Gorilla Mode

Gorilla mode is for the vulgar and obscene! It is an aggressive sounding mode that produces uniquely exaggerated results. Input waveforms with minimal spectral complexity are most effective, as this mode already creates a high number of harmonics in the frequency spectrum.

Gorilla mode contains the following parameters and controls:

(**1**) **Wavetable Modes menu**: Selects from a menu of ten operating modes that determines the readout of the wavetables.

(**2**) **K!ngs**: Selects one of three settings (King, Kong, Kang), which determines the type of bend applied to the phase to achieve sonic variations. **King** is the cleanest sounding option, providing the classic screaming sound. **Kang** provides medium bending for a dirtier sound. **Kong** has maximal bending for the dirtiest sound.

(**3**) **Ratio**: Selects one of six ratios (x1, x2, x3, x4, x5, x6) that changes the frequency ratio of a second internal oscillator phase. This control has a strong impact on the overall 'hybrid color' of the sound. A ratio of x2 is recommended to achieve the prime Gorilla sound.

(**4**) **Internal Phase On/Off**: When **Int on** is selected, the engine uses the main phase. Selecting **Int off** fixes the oscillator frequency at 0 Hz, bypassing the main phase and turning the oscillator into a waveshaper. In this instance, the main phase is replaced by the phases provided by the modulation oscillators, which can be used to make the sound inharmonic. The shaper must be used in conjunction with the PM oscillators or the PM Aux bus. The level of the PM oscillator and/or Aux input determines the amount of waveshaping, and the wavetable position and filter parameters control the shape function. Using the main phase should be the standard option for this mode.

(**5**) **Over knob**: Controls the strength of the mode. Turn the knob right to introduce more of the 'Gorilla' effect.

(**6**) **Bend knob**: Creates the formants together with the **Over** control. For the best effect, the **Over** and **Bend** controls should be modulated together.

To get started, try the 'Banana' waveform with a **x2** ratio setting, and a modulation source applied to both the **Over** and **Bend** controls.

11.2.9. Random Mode

A special mode that is dedicated to creating unusual noise sounds. This mode uses two internal randomizers which are applied to the signal in several ways. Random mode is even more extreme than the effect created in Jitter mode.

Random mode contains the following parameters and controls:

(**1**) **Wavetable Modes menu**: Selects from a menu of ten operating modes that determines the readout of the wavetables.

(**2**) **Mode**: Selects one of three modes (Fluid, Thunder, Divide) that determine the behaviour of the **Pos J/Clk Div** and **Jitter** controls. The sonic outcome is highly dependant on the selected mode. In **Fluid** mode, the **PosJ/Clk Div** randomizes the wavetable position reader and the **Jitter** knob changes the amount of randomization applied to the oscillator’s fundamental frequency. In **Thunder** mode, the **PosJ/Clk Div** randomizes the position and also downclocks the position of the randomizer. The **Jitter** knob randomizes frequency independently to the Position Jitter control. In **Divide** mode, the **PosJ/Clk Div** downclocks the Jitter´s frequency randomizer. It does not add or subtract a random deviation on each cycle, but for example, only every 10th cycle. The **Jitter** knob changes the amount of randomization applied to the oscillator’s fundamental frequency.

(**3**) **P.Rnd (Pitch Random)**: Selects between two modes (Pitch Random, Pitch Switch). When **P.Rnd** is selected, randomization is applied every cycle, depending on the Jitter rate. The pitch setting determines if it will play faster or slower. **P. Switch** makes static, quantized deviations, rather than random speed deviations.

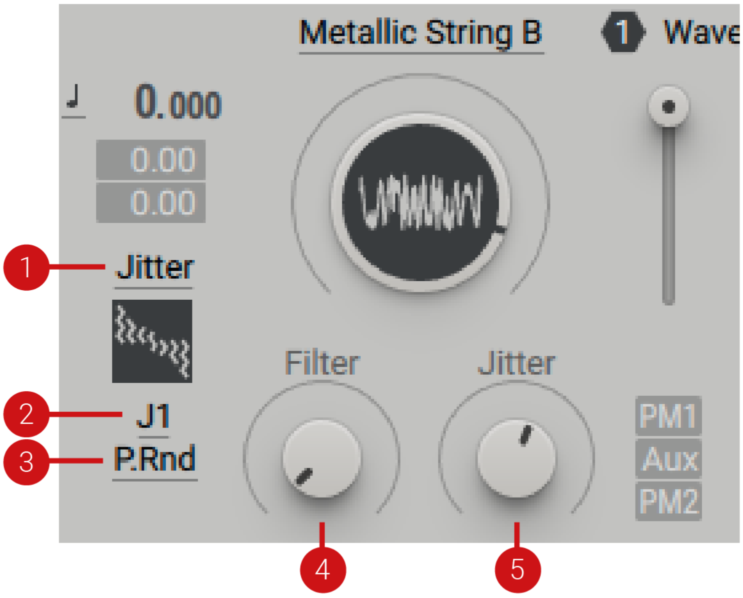
(**4**) **Position Jitter/Clock Divide knob**: Depending on the selected Mode, this knob operates in different ways, relating to the wavetable position readout and the down clocking of the internal randomizers.

(**5**) **Jitter knob**: Changes the amount of randomization applied to the oscillator´s fundamental frequency. Frequency randomization is only happening synchronous to the start of a wave cycle. The effect ranges from subtle jitter to total sound destruction.

11.2.10. Jitter Mode

Jitter mode introduces random deviations at the end of each cycle, making the signal read out faster or softer. This mode adds a glittery quality to the sound.

Jitter mode contains the following parameters and controls:



(**1**) **Wavetable Modes menu**: Selects from a menu of ten operating modes that determines the readout of the wavetables.

(**2**) **Jitter Rate**: How often the signal is flipped, every time a cycle is done. This allows you to spread out the randomization so that it does not sound too busy. **J1**: Every cycle, randomization is applied **J2**: Every 32 cycle **J3**: Every 128 cycle

(**3**) **P.Rnd (Pitch Random)**: Selects between two modes (Pitch Random, Pitch Switch). When **P.Rnd** is selected, randomization is applied every cycle, depending on the Jitter rate. The pitch setting determines if it will play faster or slower. **P. Switch** makes static, quantized deviations, rather than random speed deviations.

(**4**) **Filter knob**: Reduces the high frequency harmonics of the selected waveform. The effect is similar to adjusting the cutoff frequency of a low-pass filter.

(**5**) **Jitter knob**: Adjusts the strength of the effect from soft to strong. You can create noise by combining high pitch with a high Jitter setting.

11.3. Wavetables Menu

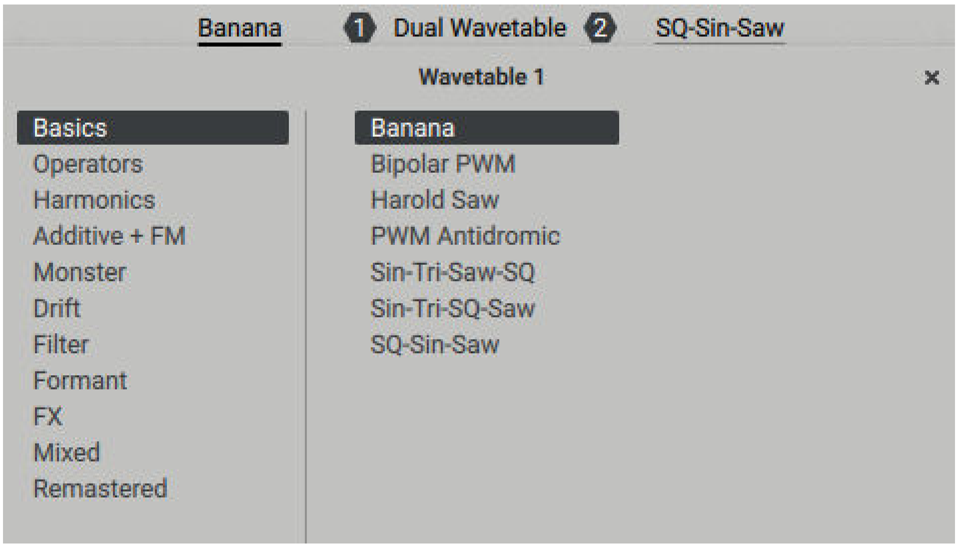
The wavetables in MASSIVE X have been categorized to help streamline your search, and provide you with an idea of the proposed use case for the wavetable.

The wavetables are not strictly separated, and they can serve as the starting point for various kinds of sound design. However, the sonic content of some wavetables makes them more suitable in certain use cases than in others. For example, some wavetables are best suited for sounds based on subtractive synthesis, other are specifically made to be used for phase modulation.

In some cases, these wavetables are not particularly exciting on their own, but work well in combination with other wavetables or specific wavetable modes. Experimenting with all controls is key to understanding how the wavetable relates to the respective wavetable mode.

For more information about wavetable modes, see Wavetable Modes.

The following section provides an overview of the wavetable categories:



- **Basics**: These wavetables do not have a distinct character, and serve as the basis for subtractive synthesis. The most basic table, **SQ-SIN-SAW**, is a good starting point for any type of synthesis. Basics also consists of some classic analog waveforms, including PWM (pulse width modulation). The **Banana** wavetable is especially designed for **Gorilla** mode, and produces its best results with modulation applied to the mode's **Over** and **Bend** controls.
- **Operators**: These wavetables are specifically made to be used for phase modulation, but they can also add body to a layered sound. **WM** (width modulation) is provided in wavetables that contain only few harmonics, allowing for a more dynamic and animated sound.
- **Harmonics**: These wavetables feature a specific effect that emphasizes the harmonic series when scanning through the wavetable. They combine well with subtle filtering or EQ, and produce interesting results when applying modulation to the wavetable position.
- **Additive + FM**: These wavetables feature sounds typically associated with additive or FM synthesis. They are ideal for producing cold, shimmery, metallic, or glassy sounds.
- **Monster**: These wavetables have a particularly brutal tone, and a strong character. They combine well with EQ, unisono and distortion. With additional phase modulation, or when used with modes like **Bend** or **Mirror**, even more aggressive timbres can be achieved. Applying modulation to the wavetable position with a sawtooth or triangle LFO creates the wobble sounds syn-

onymous with EDM or dubstep. To tame these sounds, you can use clean filters like **SV Parallel** or **SV Serial**. The included wavetables named Gorilla can produce a similar effect as the Gorilla mode when scanning through the wavetable. All content in this category is usually played in the low to very low register.

- **Drift**: These wavetables provide hard transients that work well with subtractive filtering, from standard and more complex **PWM** wavetables, to newly invented tables with the **Drift** feature. The tables create the illusion of multiple, slightly detuned oscillators beating against each other, allowing you to achieve a drifting effect with only one oscillator. To use this feature, apply a slowly moving triangle LFO to the wavetable position and alter the **Rate** of the LFO to produce the illusion of oscillator drift.
- **Filter**: These wavetables are mimicking filter sounds, including BP (band-pass) and LP (low-pass) options. They combine well with EQ, unisono and distortion. Scanning through the wavetable emulates the effect of adjusting a filter's cutoff frequency at high resonance. The results can sound similar to the wavetables in the **Harmonics** category. Due to the similarity of the effect, these wavetables are best not combined with resonant filters. Rather, they can effectively replace a resonant filter in the sound, while adding capabilities beyond normal filters.
- **Formant**: These wavetables mimic characteristics of the human voice and can be used for creating vowel sounds. They work particularly well in **Formant Capture** mode. The vowel sounds become especially distinguished when modulation is applied to the wavetable position. Some of these wavetables are based on voice recordings, which were then transformed into wavetables.
- **FX**: These wavetables produce complex sound colors. They are suitable for sound effects rather than standard oscillator sounds, or to add shimmer and texture to layered sounds. When transposed down drastically and used without pitch tracking, they can serve as noise generators without distinct pitch for sound design. You can use them to emulate environmental sounds, or to create inharmonic bell sounds. These wavetables are most effective with strong modulation applied to the wavetable position.
- **Mixed**: These wavetables are made up of a variety of highly unrelated waveforms with different sound colors. As such, they do not morph smoothly like most typical wavetables, but can be used to create interesting stepping effects.
- **Remastered**: These wavetables are legacy content from the original MASSIVE, which as been remastered for MASSIVE X.

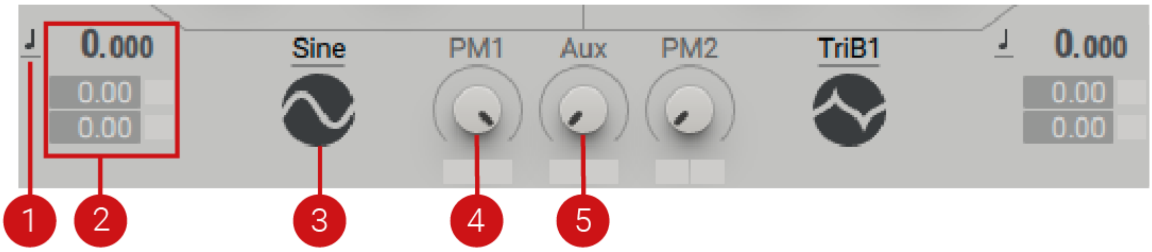
11.4. Phase Modulation

Phase modulation (PM) is a synthesis technique used in classic FM (frequency modulation) synthesizers. It creates rich harmonic spectra that can be used for a wide range of sounds including plucked basses, digital pianos, bells, and abstract sound scapes. In MASSIVE X, phase modulation is combined with the wavetable oscillators, which further increases the sonic possibilities.

The PM section offers two PM oscillators and an additional Aux input. Each of these sources can be assigned to one or both oscillators. The PM oscillators provide a variety of waveforms that cannot be heard by themselves, but instead are used to modulate the phase of the wavetable oscillators.

The PM Aux bus makes the phase modulation inputs of the oscillators accessible on the Routing page. This allows for phase modulation feedback or cross-modulation from other sound sources in MASSIVE X. Via the PM Aux bus, any source in the signal path can be used to apply phase modulation to the Wavetable oscillators.

The two PM oscillators have the same set of controls, which are mirrored on either side of the PM section:



(**1**) **Pitch Mode**: Selects one of three modes (Keytrack, Fix, Ratio) that determines the response of the PM1 oscillator in relation to incoming MIDI pitch.

- In **Keytrack** mode, the pitch is locked to the main MIDI pitch.
- **Ratio** mode multiplies or divides the MIDI pitch in relation to the ratio setting. For example, a ratio of 3 adjusts the pitch to 3 times the frequency, producing the 3rd harmonic.
- **Fix** mode sets the fixed tuning of the oscillator, disregarding the incoming MIDI pitch. This mode displays MIDI note numbers, with the default set to 60 (middle C).

(**2**) **Pitch**: Adjusts the tuning of the PM1 oscillator in semitones and cents. The **Pitch** can be modulated by assigning a modulation source to either of the two modulation slots below.

(**3**) **PM1 menu**: Selects from six waveforms (Sine, Tri, TriB1, TriB2, TriB3, SinN) that sets the waveform type for the Phase Modulation oscillator. The selected wave cycle is visually illustrated below the menu.

(**4**) **PM1 amount**: Adjusts the amount of phase modulation applied from the PM oscillator to the Wavetable oscillators. This control can be modulated by assigning a modulation source to either of the two modulation slots below the knob.

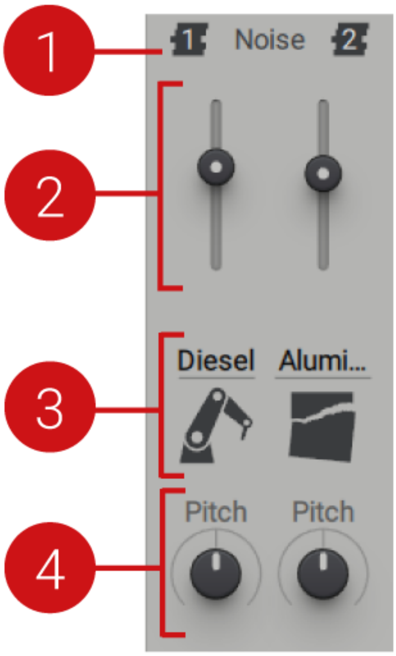
(**5**) **Aux**: Adjusts the amount of phase modulation applied from the PM Aux bus to the Wavetable oscillators. The PM Aux bus makes the phase modulation inputs of the oscillators accessible on the Routing page. Via the PM Aux bus, any source in the signal path can be used to apply phase modulation to the Wavetable oscillators. For more information about the PM Aux bus, see PM Aux Bus.

12. NOISE

The Noise section provides two noise sources each offering the ability to add textures and atmospheric content to your sound beyond the typical white and pink noise offered on most synthesizers. A wide range of categories is provided, each containing a large number of individual sounds. Besides the standard noise sounds, there are plenty of different recordings from machines, animals, and environments.

Each Noise source can be routed separately within the routing section providing a great deal of flexibility when it comes to placement of the noise sources. Further sound design opportunities become available when the Pitch and Level parameters of each noise source are modulated.

The Noise section contains the following parameters and controls:



(**1**) **Bypass switch**: Bypasses the noise source.

(**2**) **Level slider**: Adjusts the output volume of the noise source.

(**3**) **Noise menu**: Opens a menu with available noises. The categories are presented on the left and the individual noises on the right.

(**4**) **Pitch knob**: Determines the base pitch/frequency for the noise source. At center position, the Noise samples playback at their original pitch. Use the Pitch control to speed up or slow down the playback.

13. FILTER

The Filter is a key element in MASSIVE X's voice architecture. It offers nine different types of filters that cover a wide range of different applications. It can be used to balance and correct the tone of a sound, to sculpt complex timbres from raw waveforms, and to transform sounds beyond recognition.

Additionally, a number of filter types can be set into oscillation and become sound generators on their own. In some instances, this can be achieved by increasing the resonance for self-oscillation, in other instances the filter can be excited using the special Exciter envelope or noise signals for physical modelling sounds.

This chapter covers the Filter section, including all available filter types, their respective controls, and suggestions for most effective use.

Selecting where to place the Filter in the signal chain is done via the Routing page. The Filter is represented by a circular icon (**F**).

The following section provides an overview of the Filter section:

(**1**) **Filter Menu**: Selects the filter type.

(**4**) **Parameter Controls**: This area hosts a range of buttons and knobs that control different parameters of the selected filter type. Each filter type has a different selection of controls that relate to its behavior and operation.

13.1. Asimov

This low-pass filter is based on the paradigm of the filter found in a classic bass synthesizer from the 80s, despite not being a direct clone. Its defining characteristic is the lack of self-oscillation, providing a huge sweet-spot for resonant filter sounds. Additionally, it adds high-pass filtering in the feedback path for a controlled bass response. The filter has a squelchy sound that makes it suitable for acid bass lines and works well with distortion.

Asimov contains the following parameters and controls:

- **Filter mode**: Selects one of three modes (LP1, LP2, LP4) that determine the steepness of the low-pass filter. **LP1** (low-pass 1-pole) has a slope of 6 dB/Oct, **LP2** (low-pass 2-pole) has a slope of 12 dB/Oct, and **LP4** (low-pass 4-pole) has a slope of 24 dB/Oct.
- **Freq**: Adjusts the cutoff frequency of the low-pass filter. Frequency content above the cutoff frequency is attenuated, creating a darker sound.
- **Res**: Adjusts the resonance amount of the low-pass filter. Turning **Res** to the right increases the resonance, causing the frequency content at the cutoff frequency to become more pronounced.
- **KTR**: Adjusts the amount of key tracking, which is the degree to which the filter's cutoff frequency follows the MIDI pitch.
- **Gain**: Adjusts the input level and increases the amount of saturation applied to the signal.
- **FM**: Adjusts the amount of audio rate modulation applied to the filter frequency, also called filter FM (frequency modulation). The FM source needs to be connected to the FM bus on the Routing page, otherwise the **FM** control will not have an effect.

13.2. Blue Monark

This multi-mode filter is a polyphonic adaptation of the filter found in NI's Monark, with additional modes and slightly different non-linear behavior. Its defining characteristic is the ability to drive the filter into saturation using the **Gain** control. Additionally, it offers audio-rate modulation of the filter frequency and is capable of self-oscillation. Feedback can be patched in a flexible manner on the Routing page. The filter has a warm and fat character that makes it suitable for overdriven bass sounds as well as classic leads.

Blue Monark contains the following parameters and controls:

- **Filter mode**: Selects one of seven modes (LP1, LP2, LP4, BP, Peak, HP, Dual Notch) that determine the response and steepness of the filter. Three low-pass modes are available, each attenuating frequency content above the cutoff frequency with the given slope. **LP1** (low-pass 1-pole) has a slope of 6 dB/Oct, **LP2** (low-pass 2-pole) has a slope of 12 dB/Oct, and **LP4** (low-pass 4-pole) has a slope of 24 dB/Oct. **BP** (band-pass) mode attenuates frequency content above and below the cutoff frequency. **Peak** mode adds a resonant filter peak at the cutoff frequency. **HP** (high-pass) mode attenuates frequency content below the cutoff frequency. **Dual Notch** mode attenuates frequencies in two narrow frequency bands (or notches) around the cutoff frequency.
- **Freq**: Adjusts the cutoff frequency of the filter. The effect of this control on the sound depends on the selected filter mode. For more information, see **Filter mode** above.
- **Res**: Adjusts the resonance amount of the low-pass filter. Turning **Res** to the right increases the resonance, causing the frequency content at the cutoff frequency to become more pronounced.
- **KTR**: Adjusts the amount of key tracking, which is the degree to which the filter's cutoff frequency follows the MIDI pitch.
- **Gain**: Adjusts the input level and increases the amount of saturation applied to the signal.
- **FM**: Adjusts the amount of audio rate modulation applied to the filter frequency, also called filter FM (frequency modulation). The FM source needs to be connected to the FM bus on the Routing page, otherwise the **FM** control will not have an effect.

13.3. Comb

The Comb filter works differently from an analog filter: it delays the input signal and feeds it back onto itself, similar to an echo effect. However, the Comb is optimized for very short delay times, causing interferences in the audible range. This creates regularly spaced peaks and troughs in the frequency response, resembling the appearance of a comb. The effect gets more pronounced as feedback is increased. The Comb can be used as a tuned resonator in physical modeling, a complex filter for oscillator signals, or to create flanging effects. Instead of a typical filter frequency control with adjustable key tracking, it features the same Pitch control as found on the oscillators. This way, the incoming MIDI pitch is perfectly tracked, which facilitates its applications as a tuned resonator and complex harmonic filter.

Comb contains the following parameters and controls:

- **Filter mode**: Selects one of three modes (Exciter, OSC, Flanger) that cater to different applications. **Exciter** is suitable for using Comb as a resonator in physical modeling (for example Karplus-Strong synthesis). This is done by setting the Comb filter into oscillation with a suitable signal, for example from the Exciter envelope via the Routing page's Modulation Sources, or the Noise source controlled by an envelope. **OSC** is suitable for using the Comb filter with periodic signals produced by an oscillator to create complex harmonic filtering effects. **Flanger** is suitable when using Comb as a flanging effect with a wide range of different signals.
- **Pitch Mode**: Selects one of three modes (Keytrack, Fix, Ratio) that determine the response of the Comb in relation to incoming MIDI pitch. In **Keytrack** mode, the pitch is locked to the main MIDI pitch. **Ratio** mode multiplies or divides the MIDI pitch in relation to the ratio setting. For example, a ratio of 3 adjusts the pitch to 3 times the frequency, producing the 3rd harmonic. **Fix** mode sets the fixed tuning of the Comb, disregarding the incoming MIDI pitch. This mode displays MIDI note numbers, with the default set to 60 (middle C).
- **Pitch**: Adjusts the tuning of the Comb in semitones and cents. The **Pitch** can be modulated to produce vibrato and arpeggio effects by routing a modulation source to either of the two modulation slots below. In technical terms, the tuning adjusts the size of the delay buffer inside of the Comb. A larger delay buffer causes a longer delay time, which results in a lower filter frequency. A smaller delay buffer causes a shorter delay time, which results in a higher filter frequency.

- **FBW**: Switches between two different pickup points for the output in the signal flow. When disabled, the output is picked up after the delay. When enabled, the output is picked up before the delay. This setting only takes effect in combination with specific configurations on the Routing page, for example when using the **FB** modules for external feedback around the Comb filter or when mixing the Comb filter's output signal with the input signal as part of a parallel routing.
- **Feedback Polarity**: Switches between positive and negative polarity for the feedback signal. With positive feedback polarity the Comb filter produces all harmonics, while with negative feedback polarity it produces only odd harmonics.
- **FB**: Sets the level of the internal feedback loop of the Comb. Higher settings increase the feedback level, resulting in a stronger resonance of the filter.
- **AP Freq**: Adjusts the frequency of an all-pass filter in the Comb's feedback path. By changing the all-pass filter frequency, you can use the Comb to produce inharmonic spectra.
- **LP Freq**: Controls the cutoff frequency of a low-pass filter in the feedback path. Decreasing the low-pass filter frequency attenuates the high-frequency content of the feedback signal, resulting in a dampened sound.

13.4. Creak

Creak is an experimental filter that stems from research into flangers. It produces strong resonances in the frequency spectrum and is characterized by its distinct non-linear behavior. Combined with the feedback loop on the Routing page, you can apply heavy distortion and spectral transformation to a sound. The filter has an aggressive and wild character that makes it suitable for radical sound design.

Creak contains the following parameters and controls:

- **Filter mode**: Selects one of four unique modes (Driven, Gnarl, Nosy, Euer). **Driven** and **Gnarl** produce different sets of harmonically spaced resonances similar to a flanger, **Nosy** produces formants with a nasal quality, and **Euer** produces formants with a vocal quality.
- **Freq**: Adjusts the frequency of the filter. Unlike a typical cutoff control, it shifts the resonances across the frequency spectrum without attenuating broad frequency bands.
- **Res**: Adjusts the intensity of the resonances in the frequency spectrum produced by the filter.

- **KTR**: Adjusts the amount of key tracking, which is the degree to which the filter's frequency follows the MIDI pitch.
- **Mix**: Blends between the input signal and the filtered signal.

13.5. Groian

Groian is a hybrid between a filter and a flanger. It features a delay with feedback in the filter's resonance path. This structure produces strong resonances in the frequency spectrum that are superimposed with the basic response of the filter. Combined with the feedback loop on the Routing page you can apply heavy distortion and spectral transformation to a sound. Self-oscillation is possible, however it becomes unstable towards lower frequencies. The filter has a highly resonant character that makes it suitable for creating vocal or even metallic sounds.

The behavior of the filter is sensitive to the level of the input signal with stronger self-oscillation at lower input levels.

Groian contains the following parameters and controls:

- **Filter mode**: Selects one of four modes (LP4, BP, Peak, HP) that determine the response of the filter. **LP4** (low-pass 4-pole) mode attenuates frequency content above the cutoff frequency with a slope of 24 dB/Oct. **BP** (band-pass) mode attenuates frequency content above and below the cutoff frequency. **Peak** mode adds a resonant filter peak at the cutoff frequency. **HP** (high-pass) mode attenuates frequency content below the cutoff frequency.
- **Freq**: Adjusts the cutoff frequency of the filter. The effect of this control on the sound depends on the selected filter mode. For more information, see **Filter mode** above.
- **Res**: Adjusts the resonance amount of the low-pass filter. Turning **Res** to the right increases the resonance, causing the frequency content at the cutoff frequency to become more pronounced.
- **KTR**: Adjusts the amount of key tracking, which is the degree to which the filter's frequency follows the MIDI pitch.

- **Character**: Adjusts the intensity of the additional resonances in the frequency spectrum produced by the delay in the resonance path.
- **FM**: Adjusts the amount of audio rate modulation applied to the filter frequency, also called filter FM (frequency modulation). The FM source needs to be connected to the FM bus on the Routing page, otherwise the **FM** control will not have an effect.

13.6. Scanner

This multi-mode filter is inspired by the raw sound of a number of analog monophonic synthesizers from the 80s. Its defining characteristic is the pronounced resonance behavior, which you can use to carve out the harmonics of a signal when doing filter sweeps. Strong non-linear properties add harmonic distortion to the resonance. The filter has a dirty character that makes it suitable for adding texture and character to stacked pads and leads.

The behavior of the filter is sensitive to the level of the input signal with stronger self-oscillation at lower input levels.

Scanner contains the following parameters and controls:

- **Filter mode**: Selects one of four modes (LP1, LP2, BP2, Peak) that determine the response of the filter. Two low-pass modes are available, each attenuating frequency content above the cutoff frequency with the given slope: **LP1** (low-pass 1-pole) with a slope of 6 db/Oct and **LP2** (low-pass 2-pole) with a slope of 12 db/Oct. **BP2** (band-pass 2-pole) attenuates frequency content above and below the cutoff frequency with a slope of 12 db/Oct. **Peak** mode adds a resonant filter peak at the cutoff frequency.
- **Freq**: Adjusts the cutoff frequency of the filter. The effect of this control on the sound depends on the selected filter mode. For more information, see **Filter mode** above.
- **Res**: Adjusts the resonance amount of the low-pass filter. Turning **Res** to the right increases the resonance, causing the frequency content at the cutoff frequency to become more pronounced.

- **KTR**: Adjusts the amount of key tracking, which is the degree to which the filter's frequency follows the MIDI pitch.
- **Gain**: Adjusts the input level and increases the amount of saturation applied to the signal.
- **FM**: Adjusts the amount of audio rate modulation applied to the filter frequency, also called filter FM (frequency modulation). The FM source needs to be connected to the FM bus on the Routing page, otherwise the **FM** control will not have an effect.

13.7. SVF

This multi-mode filter is based on the popular state-variable filter topology and serves as a tool for a wide range of filtering tasks. Its defining characteristic is the controlled behavior and universal applicability. The filter has a clean character that makes it suitable for any sound that requires tonal shaping without adding color or distortion.

> You can use the Exciter envelope via the Modulation sources on the Routing page to briefly trigger self-oscillation of the SVF at high **Res** settings. This so called filter pinging produces a damped sine wave that can be played via MIDI by using key tracking (**KTR**).

SVF contains the following parameters and controls:

- **Filter mode**: Selects one of four modes (LP, BP, Peak, HP) that determine the response of the filter. **LP** (low-pass) mode attenuates frequency content above the cutoff frequency with a slope of 12 dB/Oct. **BP** (band-pass) mode attenuates frequency content above and below the cutoff frequency with a slope of 6 db/Oct. **Peak** mode (a band-pass filter mixed with the input signal) adds a resonant filter peak at the cutoff frequency. **HP** (high-pass) mode attenuates frequency content below the cutoff frequency with a slope of 12 dB/Oct.
- **Freq**: Adjusts the cutoff frequency of the filter. The effect of this control on the sound depends on the selected filter mode. For more information, see **Filter mode** above.

- **Res**: Adjusts the resonance amount of the low-pass filter. Turning **Res** to the right increases the resonance, causing the frequency content at the cutoff frequency to become more pronounced.
- **KTR**: Adjusts the amount of key tracking, which is the degree to which the filter's cutoff frequency follows the MIDI pitch.
- **Res Boost**: Increases the range of the **Res** control to allow for a stronger resonance.

13.8. SVF Parallel

This dual multi-mode filter is based on the popular state-variable filter topology and serves as a tool for a wide range of filtering tasks. The two filters are arranged in a parallel configuration, meaning the input signal is sent to both of them separately, while the output signal is the sum of their individual outputs. Its defining characteristic is the controlled behavior and universal applicability. The filter has a clean character that makes it suitable for any sound that requires tonal shaping without adding color or distortion. Due to the parallel configuration of two filters, it is capable of producing vocal formants as well as complex wobble sounds.

You can use the Exciter envelope via the Modulation sources on the Routing page to briefly trigger self-oscillation of the SVF Parallel at high **Res** settings. This so called filter pinging produces damped sine waves that can be played via MIDI by using key tracking (**KTR**).

SVF Parallel contains the following parameters and controls:

- **Filter mode**: Selects one of seven modes that combine different responses of each of the two parallel filters. Six combinations of a 12 dB/Oct **LP** (low-pass) filter, a 12 dB/Oct **HP** (high-pass) filter, and a 6 dB/Oct (**BP**) (band-pass) filter are available, as well as the special Plateau mode.
- **Freq**: Adjusts the cutoff frequencies of the two parallel filters. Both cutoff frequencies are offset by the same amount. Their relative position in the frequency position is set with the **Bandwidth** control.

- **Res**: Adjusts the resonance amount of the low-pass filter. Turning **Res** to the right increases the resonance, causing the frequency content at the cutoff frequencies to become more pronounced.
- **Bandwidth**: Spreads the cutoff frequencies of the two parallel filters in the frequency spectrum. At minimum setting, both filters share the same cutoff frequency, which makes the resonance much more pronounced. As **Bandwidth** is increased, the cutoff frequency of one filters moves down in frequency, while the other moves up. This way you can distribute the filter resonances to create formant frequencies.
- **KTR**: Adjusts the amount of key tracking, which is the degree to which the filter's cutoff frequency follows the MIDI pitch.
- **2nd Filter**: Blends between single and dual configurations of the filter by mixing in the second filter and adjusting the amount of frequency spread introduced by the **Bandwidth** control.

13.9. SVF Serial

This dual multi-mode filter is based on the popular state-variable filter topology and serves as a tool for a wide range of filtering tasks. The two filters, a high-pass and a low-pass, are arranged in a serial configuration, meaning the input signal is sent to the high-pass filter, the output of which is going into the low-pass filter. Its defining characteristic is the controlled behavior and universal applicability. The filter has a clean character that makes it suitable for any sound that requires tonal shaping without adding color or distortion. Due to the serial configuration of two filters, it is capable of producing vocal formants as well as balancing the overall tonal quality of a sound.

You can use the Exciter envelope via the Modulation sources on the Routing page to briefly trigger self-oscillation of the SVF Serial at high **Res** settings. This so called filter pinging produces damped sine waves that can be played via MIDI by using key tracking (**KTR**).

SVF Serial contains the following parameters and controls:

- **Freq**: Adjusts the cutoff frequencies of the two serial filters. Both cutoff frequencies are offset by the same amount. Their relative position in the frequency position is set with the **Bandwidth** control.
- **Res**: Adjusts the resonance amount of the low-pass filter. Turning **Res** to the right increases the resonance, causing the frequency content at the cutoff frequencies to become more pronounced.
- **Bandwidth**: Spreads the cutoff frequencies of the two serial filters in the frequency spectrum. At minimum setting, both filters share the same cutoff frequency. As **Bandwidth** is increased, the high-pass filter's cutoff moves down in frequency, while the low-pass filter's cutoff moves up. This way you can distribute the filter resonances to create formant frequencies.
- **KTR**: Adjusts the amount of key tracking, which is the degree to which the filter's cutoff frequency follows the MIDI pitch.
- **HP Peak**: Gradually turns the high-pass filter into a peak filter (a band-pass filter mixed with the input signal) with only a resonant peak but no filter slope. This way you can add formant frequencies without filtering out the rest of a signal.
- **LP Peak**: Gradually turns the low-pass filter into a peak filter (a band-pass filter mixed with the input signal) with only a resonant peak but no filter slope. This way you can add formant frequencies without filtering out the rest of a signal.

14. INSERT EFFECTS

There are two different types of effects available in MASSIVE X; Insert Effects and Stereo Effects. Insert Effects can be used as parts of the polyphonic voice structure, while Stereo Effects are used on the voice mix at the final stage of the signal chain. This chapter will cover the Insert Effects section, including all available effects, their respective controls and suggestions for most effective use.

Selecting where to place the Insert Effects in the signal chain is done via the Routing page. The three Insert Effects are represented by circular icons (**A**, **B**, **C**).

Individually, they can be placed anywhere in the voice structure. They can also be chained together, or inserted separately on any sound source. Where you choose to place the effect in the signal path will have significant impact on the final sound and the way the effect behaves. These crucial sonic differences will become obvious as you experiment with different routing options and effects combinations.

The following section provides an overview of the Insert Effects section:

(**1**) **Insert Effect A Menu**: Selects an Insert Effect for slot **A**. The active Effect panel on display is highlighted with an underline.

(**2**) **Insert Effect B Menu**: Selects an Insert Effect for slot **B**.

(**3**) **Insert Effect C Menu**: Selects an Insert Effect for slot **C**.

(**4**) **Parameter Controls**: This area hosts a range of buttons and knobs that control different parameters of the effect. Each Insert Effect has a different selection of controls that relate to its behavior and operation.

14.1. Anima

Anima can be used to enhance oscillator signals by altering their frequency content. The results range from subtle colorization to harmonic transformations that are achieved by adding new sidebands in the frequency spectrum.

As an experimental derivative of effects like comb filters and flangers, its internal structure involves a unique combination of delay lines, audio rate modulation, and feedback.

Similar to tuned comb filters, Anima tracks the MIDI pitch of the instrument. This way, it can be tuned relative to the pitch of the input signal, allowing you to explore a range of interesting sounds.

Anima contains the following parameters and controls:

- **Pitch**: Adjusts the center frequency of the tuned delay lines in Anima's internal structure.
- **Mix**: Blends between the input signal and the effect signal. Turn the control fully left to bypass the effect, or right to mix in the effect. Anima is commonly used with high to maximum **Mix** settings.
- **FB**: Adjusts the amount of feedback in Anima's internal structure. Anima is commonly used with high to maximum **FB** settings.
- **+/-**: Selects one of two routing modes of the effect and (**+**) and (**-**). A positive polarity setting is represented by the (**+**) icon and is the default setting. Press the icon to change to a negative polarity (**-**). Each of the two modes features a unique set of signal polarities within Anima's internal structure. This parameter has a strong influence on the overall sound character of the effect.
- **Smear**: Adjusts the distribution of tuned delay lines around the center frequency as set with **Pitch**.
- **Amount**: Adjusts the amount of internal modulation applied to the tuned delay lines.
- **Rate**: Sets the rate of the internal modulation oscillator.

- **Fast/Slow**: Selects one of two basic modes (Slow, Fast). When **Slow** is selected, the internal modulation oscillator runs at LFO rates. This produces sounds reminiscent of a comb filter combined with vibrato. When **Fast** is selected, the internal modulation oscillator runs at audio rates, relative to the instrument's MIDI pitch. This produces interesting harmonic transformations by adding new sidebands in the frequency spectrum, similar to frequency modulation or phase modulation.

14.2. Bit Crusher

The Bit Crusher degrades a signal by reducing the bit depth, which is the number of bits used to represent amplitude in digital audio. A high bit depth results in an accurate representation of a signal, while a low bit depth adds noise and distortion. This can be used for lo-fi effects ranging from subtle noise textures to extreme distortion that turns any signal into clicks and pulses.

The Bit Crusher contains the following parameters and controls:

- **Crush**: Adjusts the bit depth and therefore the strength of the bit reduction effect. By turning the control fully left, the lowest number of bits is used, resulting in the strongest effect. By turning the control fully right, the highest number of bits is used, closely resembling the input signal.
- **Mix**: Blends between the input signal and the effect signal. Turn the control fully left to bypass the Bit Crusher, or turn right to mix in the effect. Turning the control fully right results in a wet only signal.
- **Offset**: Adds an offset to the input signal, making it asymmetric. This can be used to create variations of the basic bit reduction effect.
- **Norm**: Subtracts the input signal from the effect signal, making the bit reduction effect more pronounced.
- **HQ**: Activates high quality anti-aliasing. By default, **HQ** is deactivated. Press the **HQ** button for less harsh, smooth lo-fi results.

14.3. Utility

This module helps you correct, level out, or finalize your sound. If a sound is too bright, too bass heavy, too loud or soft, this is the tool to use. It is especially effective when used in conjunction with the Tracker, as well as in Feedback scenarios, where it can be used together with the Comb filter to create an overblown flute and similar effects. This enables you to scale the amplitude in order to produce a balanced sound over the range of the keyboard. The filters have no nonlinearities or resonance, resulting in a clean and neutral sound.

The Utility contains the following parameters and controls:

- **Filter Type**: Two menus that offer four filter types (**HP1**, **HP2**, **LP1**, **LP2**) to select from. If you use two **LP2** filters with the same settings you will achieve a 4-pole filter characteristic.
- **Freq**: **Freq 1** and **Freq 2** determine the cutoff point for the corresponding filters.
- **Level**: The Level knob sets the volume of the VCA (voltage controlled amplifier). Two controls determine the behavior of the VCA .
  - **Range**: Defines the maximal amplification factor that can be achieved with the VCA. Turning the control from fully counter-clockwise to centre position provides a range from 1% to 100%. Turning **Range** from the center position to fully right provides a maximum of 500%.
  - **Curve**: Bends the response character of the VCA. At centre position, the VCA behaves linearly. This is the default setting for this module. Turned fully left, it produces a strong bend with the amplification factor rising exponentially towards the end. At full right position, it bends strongly in the opposite way, so the amplification factor rises logarithmically towards the end.

14.4. Folder

Sitting somewhere between saturation and hard sync effect, the Folder starts to fold, or copy, the waveform of the input signal when driven with high pre-amplification settings. The sonic results are varied and dependant on the input signal, the selected Mode and the Drive and Offset settings. Four modes (Sinus, Triangle, Wrap, Spiky) determine the character of the folding, from smooth to distinctly rough. The harmonic repetitions can to create sounds reminiscent of hard sync-style effects, which you can incorporate subtly with low Mix settings, or turn up for an intense, brutal sound. The Folder can also share sonic correlations to phase modulation sounds. This is achieved most effectively by combining a sinus input signal with Sinus mode .

The Folder contains the following parameters and controls:

- **Mode**: Select one of four modes (Sinus, Triangle, Wrap, Spiky). The wave of each mode is visually represented below the Mode selector.
- **Mix**: Blends between the input signal and the effect signal. Turn the control fully left to bypass the effect, or turn right to mix in the effect.
- **Drive**: Pre-amplifies the signal before it reaches the folding algorithm.
- **Offset**: Shifts the operating point of the algorithm.
- **HQ**: Activates high quality anti-aliasing. By default, HQ is deactivated. Press the HQ button to turn on anti-aliasing.

14.5. Frequency Shifter

The Frequency Shifter shifts the frequency of the incoming signal. Unlike a pitch shifter, the Frequency Shifter is able to create inharmonic partials from a sound comprised of harmonic partials, adding a metallic character to your sound. A Shepard phaser effect can be achieved when slow modulation and feedback is applied, with the Mix control set to 50%.

The Frequency Shifter contains the following parameters and controls:

- **Range**: Selects one of two frequency ranges (Wide, Narrow). **Wide** makes the Frequency Shifter operate over a wide frequency range, useful for achieving wild and animated sounds. The **Narrow** setting provides good control for relatively small shifting factors. This range is scaled with the **Freq** control.
- **Mix**: Blends between the input signal and the effect signal. Turn the control fully left to bypass the effect, or right to mix in the effect.
- **Freq**: Shifts the partials of a signal. In centre position, the effect is neutral. Turning the control right shifts the partials upwards, and turning left shifts the partials downwards. Due the nature of the algorithm, partials can be reflected upwards again, if the downshifted partials are exceeding zero hertz. The encoder itself behaves in a nonlinear manner, resulting in a finer resolution around the centre.
- **FB**: Adjusts the amount of feedback. When this control is turned up, the output of the Frequency Shifter is reflected back in to the input, creating a Shepard flanger effect.

14.6. Distortion

This module offers different nonlinear distortion functions that are especially useful for polyphonic sound design. Nonlinear distortion is produced when the output of a signal is not exactly proportional to the input signal, generating harmonics. The Distortion provides five modes for different nonlinear functions, Mix and Drive controls and a HQ button that activates high quality anti-aliasing. Unlike the Stereo Effects' Nonlinear Lab module, the Distortion does not offer internal memorizing, loading or cabinet simulation.

The Distortion contains the following parameters and controls:

- **Mode**: Selects one of five modes that offer different nonlinear functions (tanH, hypB, sin C, H.clip, Rectify). Each mode is visually represented below the Mode selector.
- **Mix**: Blends between the input signal and the effect signal. Turn the control fully left to bypass the effect, or right to mix in the distortion.
- **Drive**: Controls how hard the saturation stages are driven, from subtle to extreme.
- **HQ**: Enables antialiasing in order to improve the audio quality of the nonlinear functions.

> Enabling **HQ** introduces half a sample latency, which can cause phase artifacts when mixed with the original signal. This can also have consequences when used in feedback loops created with audio routing.

14.7. Insert Oscillator

The Insert Oscillator (OSC) is an independent source for three classic synth waveforms; Sine, Sawtooth, and Pulse, which can be used in addition to the main Wavetable Oscillators. When used simultaneously over the three Insert Effects slots and alongside the two Wavetable Oscillators, it expands to a five oscillator synthesizer. These Insert Oscillators can also be phase locked to either Wavetable Oscillator, creating a classic Sub Oscillator. The Insert Oscillator is one of the most powerful tools in the Insert Effects, greatly expanding the potential of the MASSIVE X synth engine.

The Insert Oscillator contains the following parameters and controls:

- **Mode**: Selects the waveform of the oscillator (Sine, Saw, Pulse). **Sine** and **Saw** modes offer a phase shift control while **Pulse** mode offers control over PW (pulse width).
- **Invert**: Inverts the polarity of the waveform. When the button is activated, the waveform is flipped.
- **Pitch Mode**: Selects one of three modes (Keytrack, Fix, Ratio) that determines the response of the oscillator in relation to incoming MIDI pitch. In **Keytrack** mode, the pitch is locked to the main MIDI pitch. **Ratio** mode multiplies or divides the MIDI pitch in relation to the ratio setting. For example, a ratio of 3 adjusts the pitch to 3 times the frequency, producing the 3rd harmonic. **Fix** mode sets the fixed tuning of the oscillator, disregarding the incoming MIDI pitch. This mode displays MIDI note numbers, with the default set to 60 (middle C).
- **Pitch**: Adjusts the tuning of the oscillator in semitones and cents. The **Pitch** can be modulated by routing a modulation source to either of the two modulation slots below, allowing you to produce vibrato and arpeggio effects.
- **Lock**: Enables you to lock the pitch to one of the main Wavetable oscillators. When inactive, the pitchbox is shown, providing MASSIVE X's standard oscillator tuning. When the **Lock** button is active, settings for **Ratio** and the Sync source are provided.
  - **Lock Ratio**: Selects one of four ratios (1:1, 1:2, 1:3, 1:4) that determines the Tune Ratio in relation to the chosen Source oscillator. At **1:1**, the Insert oscillator runs at exactly the same pitch as the Source oscillator. With a **1:2**, the Insert oscillator is 1 octave lower (-12 semitones). **1:3** runs the Insert oscillator at 1/3 of the speed, equating to approximately 1 octave and a fifth lower (-19 semitones). At **1:4**, the Insert oscillator runs 2 octaves lower (-24 semitones).
  - **Pitch Source**: Selects if the pitch should be locked to Wavetable oscillator 1 or 2. The **Ratio** cannot be modulated as it is hard-locked to the Source oscillator, but modulation can be applied to the Insert oscillator's **Phase**/**PW** parameter.
- **Phase/PW**: The label and functionality of this control changes according to the selected mode. Sine and Sawtooth modes offer a **Phase** offset control, allowing you to phase shift the oscillator. The Phase control is used to set the Phase relationship between oscillators when locked to the Wavetable oscillators or if the Engine is set to resetting behaviour on the Voice page. When Pulse is selected, the **Phase** control switches to **PW**, which adjusts the width of the Pulse wave. When set to the center value, a square wave is produced. Adjusting the **PW** control

makes the Pulse wave asymmetric, introducing even harmonics that create a thinner sound. Applying modulation to the **PW** parameter is a classic technique used to add dynamism to a waveform, known as PWM (pulse width modulation).

- **Mix**: Blends between the input signal and the independently delivered waveform. Turn the control fully left to bypass the effect, or turn right to mix in the waveform. The Mix control has the same functionality across all modes.

14.8. PM Oscillator

The Phase Modulation oscillator features a sine wave oscillator that can be Phase modulated by it's input signal and placed anywhere in the voice structure. The rich phase modulated audio signal can be mixed in with the original input. Combining the three Insert Effects with the two Wavetable oscillators, creates the potential for a seven operator FM synth.

The flexibility in routing reveals the true power of PM oscillator. Complex Wavetable oscillators can be used to modulate the operators, or the operators can be used to phase modulate the Wavetable oscillators via the Aux bus. You can even combine it with filters, ring modulation or any other Insert Effects. With flexible routing, you can achieve cross-feeding of operators or phase feedback chains within itself.

The PM Oscillator contains the following parameters and controls:

- **Invert**: Inverts the polarity of the waveform. When the button is activated, the waveform is flipped.
- **Pitch Mode**: Selects one of three modes (Keytrack, Fix, Ratio) that determines the response of the oscillator in relation to incoming MIDI pitch. In **Keytrack** mode, the pitch is locked to the main MIDI pitch. **Ratio** mode multiplies or divides the MIDI pitch in relation to the ratio setting. For example, a ratio of 3 adjusts the pitch to 3 times the frequency, producing the 3rd harmonic. **Fix** mode sets the fixed tuning of the oscillator, disregarding the incoming MIDI pitch. This mode displays MIDI note numbers, with the default set to 60 (middle C).
- **Pitch**: Adjusts the tuning of the oscillator in semitones and cents. The **Pitch** can be modulated by routing a modulation source to either of the two modulation slots below, allowing you to produce vibrato and arpeggio effects.

- **PM**: Adjusts the Phase Modulation amount. This works as an amplifier for the incoming signal added to the sine oscillator's phase.
- **Mix**: Blends between the input signal and the sine operator signal. Turn the control fully left to bypass the effect, or turn right to mix in the waveform.

14.9. Ring Modulator

Ring modulators mix the frequencies of two different waveforms, and output the sum and difference of the frequencies present in each. This process of ring modulation produces a signal rich in partials. Neither original signals are prominent in the final output, allowing you to transform your signal.

The Ring Modulator offers a Sine wave oscillator that can be used to ring modulate any input signal. This can be freely placed into the voice structure via the Routing page like any other Insert effect. Using ring modulation with one of the main oscillators, when the two frequencies are not harmonically related, can create metallic or bell-like sounds.

The Ring Modulator contains the following parameters and controls:

- **Invert**: Inverts the polarity of the waveform. When the button is activated, the waveform is flipped.
- **Pitch Mode**: Selects one of three modes (Keytrack, Fix, Ratio) that determines the response of the oscillator in relation to incoming MIDI pitch. In **Keytrack** mode, the pitch is locked to the main MIDI pitch. **Ratio** mode multiplies or divides the MIDI pitch in relation to the ratio setting. For example, a ratio of 3 adjusts the pitch to 3 times the frequency, producing the 3rd harmonic. **Fix** mode sets the fixed tuning of the oscillator, disregarding the incoming MIDI pitch. This mode displays MIDI note numbers, with the default set to 60 (middle C).
- **Pitch**: Adjusts the tuning of the oscillator in semitones and cents. The **Pitch** can be modulated by routing a modulation source to either of the two modulation slots below, allowing you to produce vibrato and arpeggio effects.
- **Mix**: Blends between the input signal and the sine waveform. Turn the control fully left to bypass the effect, or turn right to mix in the waveform.

14.10. Sample and Hold

The Sample and Hold module provides the opportunity for classic Sample and Hold effects. The internal oscillator samples the incoming signal, holding this value until the next sample is taken. This effect results in a 'stepped' quantized audio signal. The oscillator itself is not heard directly, but can be tuned in the same way as the other oscillators. The Sample and Hold effect can be used to create a range of different distortion effects. Try setting low Pitch values to recreate the sounds of old digital synthesizers. For classic sample rate reduction, Key Tracking must be deactivated.

The Sample and Hold module contains the following parameters and controls:

- **HQ**: When deactivated, the effect operates in a typical way with a clear digital character. When activated, a high quality algorithm treats the steps in the waveform caused by downsampling, producing an analog sound.
- **Pitch Mode**: Selects one of three modes (Keytrack, Fix, Ratio) that determines the response of the oscillator in relation to incoming MIDI pitch. In **Keytrack** mode, the pitch is locked to the main MIDI pitch. **Ratio** mode multiplies or divides the MIDI pitch in relation to the ratio setting. For example, a ratio of 3 adjusts the pitch to 3 times the frequency, producing the 3rd harmonic. **Fix** mode sets the fixed tuning of the oscillator, disregarding the incoming MIDI pitch. This mode displays MIDI note numbers, with the default set to 60 (middle C).
- **Pitch**: Adjusts the tuning of the oscillator in semitones and cents. The **Pitch** can be modulated by routing a modulation source to either of the two modulation slots below, allowing you to produce vibrato and arpeggio effects.
- **Mix**: Blends between the input signal and the delayed signal. Turn the control fully left to bypass the effect, or turn right to mix in the delayed signal.

14.11. Track Delay

Track Delay operates unlike a standard delay effect. The module provides two basic modes of operation. In Key Track mode the input signal can be shifted in relation to MIDI pitch, and mixed with the original signal, in an inverted or normal manner. For example, a saw input can be transformed into a square wave by phase shifting the signal by 180 degrees and subtracting the inverted mix from the input. Modulating the Phase results in true PWM (pulse width modulation), and this can be adapted to any input signal as WM. Fix mode can be used to mimic pickup effects, placing the pickup on a virtual string. Track Delay can also be used as a polyphonic chorus effect.

The Track Delay contains the following parameters and controls:

- **Modes**: Selects one of two modes (Key Track, Fix) that define the main operating mode. In **Key Track** mode, the delay time is locked to the main MIDI pitch. In **Fix**mode, the delay time is not locked to the main MIDI pitch and operates in the time domain. The range is determined by the **Range** menu.
- **Range**: Selects one of three options for each of the two operating modes (**180°/10msec**, **360°/50msec**, **720°/150msec**) phase degree/milliseconds.
- **Phase**: Adjusts the **Phase** when **Key Track** mode is selected. In **Fix** mode, it determines the delay time.
- **Mix**: Blends between the input signal and the delayed signal. Turn the control fully left to bypass the effect, or right to mix in the delay signal. Center position is recommended. At this position, the **Trim** encoder can be used to control the center level.
- **Invert**: Switches between Normal and Invert to determine how the delayed signal is mixed. An Inverted polarity is recommended.
- **Lag**: Adjusts the smoothing of delay time changes.
- **Trim**: When the **Mix** encoder is set to center position, the **Trim** knob controls the center level.

15. AMPLIFIER

The **Amp** (Amplifier) is the final module in the signal path of the voice. It controls the level and panning before the signal enters the Stereo Effects section.

The Amp Envelope (**E1**) is hard-wired to control the Amp level. The additional **Level** control can be used to adjust the signal level going into the Stereo Effects, while the **Pan** control enables the distribution of the sound to the left or right stereo channel and can also be modulated for creative panning effects.

The additional **FB** control adjusts the amount of feedback through the voice's feedback loop. The feedback loop can be freely connected in the Polyphonic Area of the Routing page. For more information about the feedback loop, see Feedback Loop.

The Amp section contains the following parameters and controls:

Level: Adjusts the output level of the amplifier. Double-click to reset to the default value.

Pan: Distributes the sound to the left or right stereo channel. Double-click to reset to the default value.

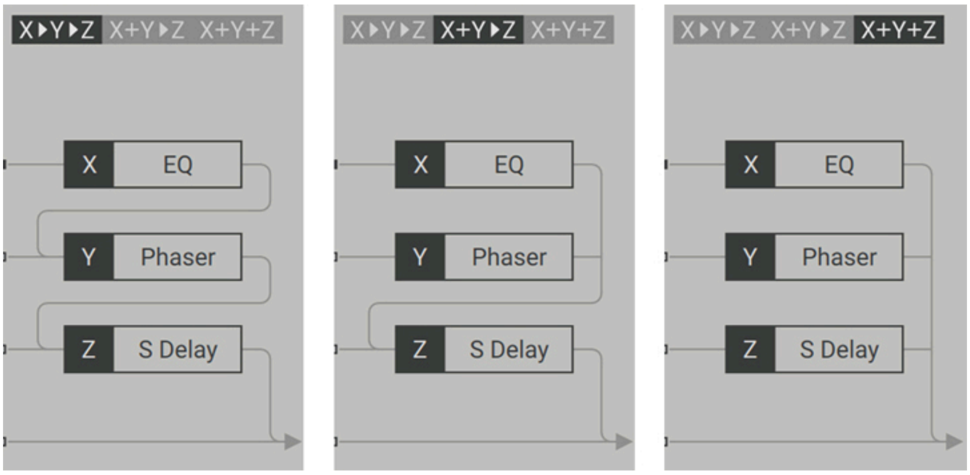
FB (Feedback) level: Adjusts the level or volume of the feedback loop. This way you can control the chaotic behavior and distortion produced by the feedback.

High-pass filter: Enables a high-pass filter that cuts low-frequency content in the feedback loop. When activated, you can avoid overloading the feedback loop with excessive bass.

16. STEREO EFFECTS

Stereo Effects are the final stage that the signal passes through before reaching the main output. A wide range of effects are offered, from time-based effects like flanging, phasing and chorus, to spatial effects that allow you to adjust and expand the stereo field. Versatile distortion, saturation and overdrive effects can be produced, and combined with a true stereo reverb or delay.

Effects can be loaded into three slots (**X**, **Y**, **Z**), and three routing options offer different ways for the effects to be mixed: **X > Y > Z** chains the three effects, **X + Y > Z** sends the sum of the X and Y effects to the Z effect, and **X + Y + Z** sums all three effects.



Stereo Effects Routing options

The following section provides an overview of the Stereo Effects section:

(**1**) **Stereo Effect X Menu**: Selects a Stereo Effect for slot **X**. The active Effect panel on display is highlighted with an underline.

(**2**) **Stereo Effect Y Menu**: Selects a Stereo Effect for slot **Y**.

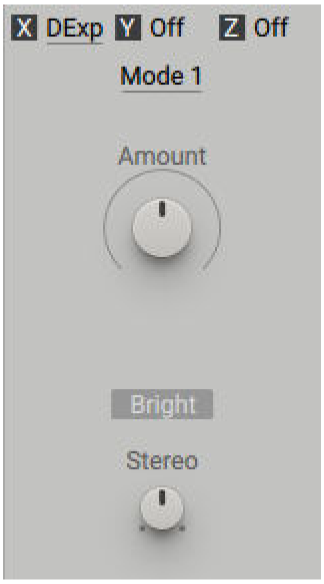
(**3**) **Stereo Effect Z Menu**: Selects a Stereo Effect for slot **Z**.

(**4**) **Parameter Controls**: This area hosts a range of buttons and knobs that control different parameters of the effect. Each Stereo Effect has a set of controls that relate to its behaviour and operation.

16.1. Dimension Expander

The Dimension Expander is a Chorus with stereo expansion capabilities, that can add an extra dimension to your sound. The results range from subtle enrichment to a shimmery, expanded stereo sound.

The Dimension Expander contains the following parameters and controls:



- **Modes**: Selects one of four modes of the effect (1 - 4), ranging from subtle to strong.
- **Amount**: Adjusts the strength of the effect.
- **Bright:** When on, the character is neutral. When off, the effect produces a slightly filtered sound.
- **Stereo**: Morphs the output of the device from mono to stereo. Turn the control fully right for a complete stereo effect.

16.2. Reverb

The Reverb offers a range of modes that mimic different room types and spaces, designed for various kinds of applications.

The Reverb contains the following parameters and controls:

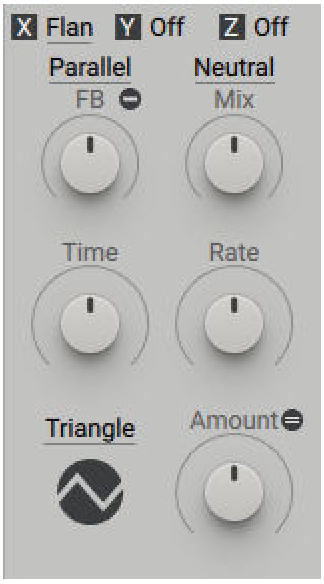
- **Modes**: Selects one from seventeen different Reverb modes (Late, Large Hall, Jazz Hall, Stage, Med Synth, Fat Synth, Reflective, Rave Cellar, Small Early, Small Dense, Micron, Tight, Metolla, Wave, Woosh, Non Linear, Wanderlust), that determines the sound character and behavior of the Reverb effect.

- **Send**: Determines the amount of input signal that will be routed into the reverb engine.
- **Mix**: Blends between the input signal and the effect signal, determining how much of the reverb effect is mixed with the dry signal. Turn the control fully left to bypass the effect, or turn right to mix in the reverb.
- **Size**: Adjusts the size of the simulated room, or reverb effect. This control affects the duration of the reverb tail.
- **Color**: Adjusts the coloration of the reverb using an internal filter to subtly change the sound from dull to bright.
- **Delay**: Determines the pre-delay of the reverb effect. No delay is added when the control is turned fully left.
- **Chorus**: Applies a chorusing effect to the reverb. This can add movement and depth to the sound, producing a lively reverb that evolves over time.

16.3. Flanger

The Flanger is a sonically diverse module, featuring seven different play modes, ranging from classic stomp box-style to wild and complex flange effects. The delay range of typical flanger is expanded, allowing you to also create unusual and unique effects. To achieve a classic flanger sound, the range must be chosen carefully.

The Flanger contains the following parameters and controls:



- **Modes**: Selects one of seven modes that determines the type of modulation and the internal routing of the flanger. The available parameters vary depending on the chosen mode.
  - **Parallel**: The LFO modulation is identical for the left and right channels while the audio inputs are still true stereo. This mode is best suited for the classic stomp box flanger sound.
  - **Wide**: The LFO modulation of the left and right channel is slightly offset, resulting in a wider, stereophonic sound.
  - **Inverse**: The LFO modulation is shifted by 180 degrees (inverse modulation) on one side, resulting in an even wider stereophonic sound.

  - **Difference**: The internal signals are mixed so that the output stages show the sum and the differences of this operation. This results in a sound with a metallic character, that is most clearly heard when the **Time** control is set to a short delay time. The Triangle **LFO Shape** is recommended for this mode.
  - **Cross**: The LFO uses inverse modulation. The internal signals are mixed and the output stages show the sum and the differences of this operation. As Cross mode relies on a specific, internal modulation to achieve its result, the **LFO Shape** is predefined and can not be changed. While it shares similarities to Difference mode, it creates stable frequency sidebands resulting in less movement and sonic wobble.
  - **Cross Astral**: A variation of Cross and Difference modes, using more complex internal modulation. The **LFO Shape** is fixed and the shape menu is removed. This mode can be particularly effective for raw pads or pure waveforms.
  - **Manual**: In Manual Mode, independent Time control for the left and right channels is available and the internal LFO modualtion is removed. Applying different modulations to the left and right sides can produce dramatic results.
- **Mix**: Blends between the input signal and the effect signal, determining how much phasing is applied. Turn the control fully left to bypass the effect. At center position, the largest frequency gaps are created. Turning the control fully right creates a pure flanger effect.
- **FB**: Sets the amount of signal that will feed back from the delays output into the delays input. The more feedback, the more the frequency peaks and gaps are modeled out of the sound spectrum.
- **FB +/-**: This switch heavily influences the overall characteristics of the flanger. It determines whether the feedback signal is mixed in with normal or inverse polarity. Negative feedback produces uneven harmonics from the comb filter circuit. Positive feedback creates even and odd harmonics. Cross and Cross Astral modes do not offer **Feedback** polarity.
- **Time**: Sets the delay time or fundamental frequency of the flanger. The LFO modulation is applied around this center frequency. In Manual mode, independent **Time** control of the left and right channel is available.
- **Rate**: Sets the Modulation Rate of the LFO.
- **LFO Shape**: Selects one of three modulation shapes (Triangle, Logarithmic, Sine). The LFO shapes are tweaked to be musical rather than mathematically perfect shapes. Besides the standard **Triangle** and **Sine** shapes, a **Logarithmic** shape is an option to recreate classic flanger sounds.
- **Amount**: Sets the amount of **Time** modulation by the LFO. Classic flange sounds can be achieved with lower **Amount** settings.
- **Constant Amount Button**: When deselected, the LFO amplification applied by the **Amount** control works in a typical manner. A faster LFO rate setting will apply stronger detuning to the comb filter. When **Constant Amount** is active, the amount of detuning applied is fixed, regardless of the LFO rate. This leads to very interesting pitch shifting effects when using internal Triangle modulation with high **Feedback** values.

16.4. Nonlinear Lab

The Nonlinear Lab offers a multitude of overdrive and distortion characteristics. Several optional speaker cabinet simulations further expand the sonic flexibility of the Nonlinear Lab. The available parameters and controls are dependant on the chosen Drive type and Cabinet.

The Nonlinear Lab contains the following parameters and controls:

- **Drive type**: Selects one of three different HQ saturator/distortion models (Hard Clip, Soft Clip, Overdrive).

  Hard Clip: A standard tanH (=hyperbolic tangens) saturator. This model offers great standard distortion that works particularly well for acidic sounds.

  - **HP Pre**: Adjusts the frequency of the high-pass filter, applied to the input signal before the saturator. Use this control to avoid overloading the saturator stage with excessive bass.
  - **HP Post**: Frequency of the high-pass post-stage (after saturator). Adjusts the bass output of the saturator.
  - **Comp**: Adjusts the output level compensation for the **Drive** amount. The output level of different **Drive** amounts is highly dependent on the input signal. The **Comp** control helps to keep the output level under guard.

  Soft Clip: The most gentle distortion mode of the Non Linear Lab. It works well for a wide number of signals like bass, guitar and drum sounds. This mode offers three different models of different sonic flavors.

  - **HP Pre**: Adjusts the frequency of the high-pass filter, applied to the input signal before the saturator. Use this control to avoid overloading the saturator stage with excessive bass.
  - **Models**: Selects one of three sub models for this mode (Glue, Warm, Hot). **Glue** offers compression with relatively subtle distortion. **Warm** offers compression with medium distortion while the **Hot** setting features compression with strong distortion.
  - **Comp**: Adjusts the output level compensation for the **Drive** amount. The output level of different **Drive** amounts is highly dependent on the input signal. The **Comp** control helps to keep the output level under guard.

  Overdrive: An extreme, intense distortion effect, that produces exaggerated sonic results.

  - **HP Pre**: Adjusts the frequency of the high-pass filter, applied to the input signal before the saturator. Use this control to avoid overloading the saturator stage with excessive bass.
  - **Inertia**: Determines how fast the loading of the stage occurs, depending on the frequency. Turn the control fully right for the fastest response.

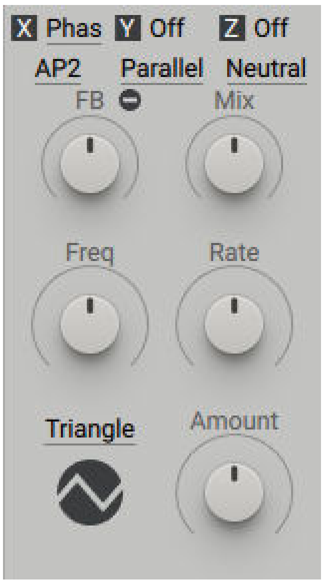
  - **Comp**: Adjusts the output level compensation for the **Drive** amount. The output level of different **Drive** amounts is highly dependent on the input signal. The **Comp** control helps to keep the output level under guard.
- **Stereo**: Selects between Mono and Stereo. In **Mono**, the left and right sides are mixed together and sent to the monophonic saturator and cabinet stages. The dry path remains stereo. In **Stereo** mode, the device operates in true stereo.
- **Drive**: Controls how hard the saturation stages are driven, from subtle to extreme.
- **Mix**: Blends between the unaltered input signal and the affected output of the Lab.
- **Cabinet Stage**: Selects one of six different guitar cabinet simulations (California, West Coast, Classy, Hi Gain, Crank, British). The cabinets drastically shape the overall colour of the output. Bypass the cabinet stage by selecting **Off**. Without a cabinet selected, the distortions may sound raw, and are particularly suitable for raw synth sounds.
- **Bass**: Finely adjusts the bass response and boominess of the cabinet model.
- **Vari**: Selects one of three sub-models of the cabinet (A, B, C).

16.5. Phaser

The Phaser can produce a wide array of sonic results exceeding the conventional limits of standard phasers. Phasing is a modulation effect that sends a signal through a series of allpass filters. Each filter alters the phase of a set frequency, and an LFO is used to modulate this phase shifting, producing a characteristic sweeping sound. When the effect signal is mixed with the original, the out of phase frequencies create notches and peaks in the frequency spectrum. The number of allpass filters (**Stages**) determines the amount of notches and peaks, and the **Feedback** control sends the effect signal back through the series of allpass filters, increasing the resonance for an even more intense sound.

For a simple and classic guitar stomp-box style phaser, a relatively low Feedback setting should be used, with a positive polarity setting. For a sonically complex sound, use a higher number of allpass stages. Fine-tuned coloration settings are predetermined, with each option producing varied sonic characteristics typically found in different phaser effects. The wide sonic range and potential combinations makes the Phaser effect highly powerful and versatile.

The Phaser contains the following parameters and controls:

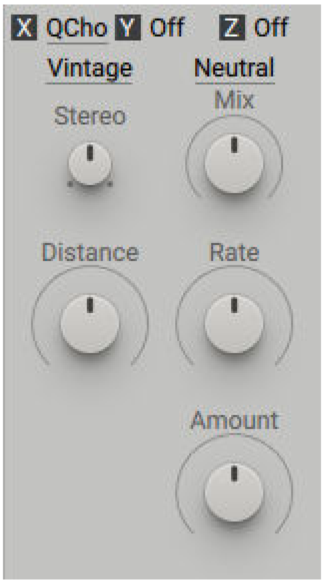


- **Stages**: Selects the number of allpass (AP) filter stages (2, 4, 5, 6, 8). Additionally there are two modes that offer Barber Pole phasing. In these modes the traditional LFO modulation controls are replaced by a **Freq** control that is used to create infinite upwards-downwards motion through the frequency spectrum. The two Barber modes differ in range. The AR mode features audio rate modulation.
- **Modes**: Selects one of five modes that determines the type of modulation and the internal routing of the phaser. The available parameters vary depending on the chosen mode.
  - **Parallel**: The LFO modulation is identical for the left and right channels while the audio inputs are still true stereo. This mode is best suited for the classic stomp box phaser sound.
  - **Wide**: The LFO modulating the flanger is identical for the left and right side, but one side is slightly offset, resulting in a wider, stereophonic sound. The modulation itself remains monophonic.
  - **90°**: The LFO phase is shifted by 90 degrees on one side, producing a bigger difference between the left and right channels. This results in a wider, more stereophonic sound.
  - **180°**: The LFO modulating the flanger is shifted by 180 degrees (inverse modulation) on one side, resulting in an even wider stereophonic sound. In Barber mode the movement is inversed between both sides.
  - **Manual**:In Manual mode, the internal LFOs are switched off and you have individual control over the frequency parameter of both channels. Applying different modulations to the left and right sides produces dramatic effects.
- **Color**: Selects one from three different settings (Neutral, Stomp, Narrow) that determine the coloring of the phaser. For classic stomp-box behavior, the **Stomp** setting is ideal. **Narrow** does not influence the spectrum around the allpass peaks as strong as the other settings, making it suitable for darker, bass-heavy sounds. Use the **Neutral** setting to produce a neutral tone.
- **Mix**: Adjusts the amount of phasing applied. Turn the control fully left, to bypass the effect. Center position typically produces the maximum notch effect. Turned fully right, the pure phaser is heard. The best position will be dependant on the number of stages, the polarity setting and the desired effect. The sweet spot is very often around the middle or close to fully wet.
- **FB**: Controls how much feedback is applied to the signal. The more feedback the stronger the allpass filters will resonate. This is similar to your typical lowpass resonance filter. The more resonance, the more the frequency peaks, and gaps are modeled out of the sound spectrum. For a classic phaser sound, set the range between 30%-70%.
- **FB +/-**: Switches between Positive and Negative polarity settings, which determine whether the internal feedback is mixed in with normal or inversed polarity. This greatly influences the overall characteristics of the phaser, particularly with stage settings **2** and **4**. Negative feedback produces funky, formant-style sounds. Positive feedback produces round, bass-heavy sounds.
- **Freq**: Determines the basic operating frequency of the phaser. The LFO modulation is applied around this center frequency. In **Barber** mode the **Freq** parameter controls the rate of the upwards/downwards movement through the frequency spectrum.
- **Rate**: Determines the Modulation Rate of the LFO. In **Barber** mode, the Rate of the modulation is controlled by the **Freq** parameter.
- **LFO Shape**: Select one of four modulation shapes (Triangle, Round, Ramp Up, Ramp Down).
- **Amount**: Sets the range of how much the LFO is animating the allpass filters. This control is only available when an LFO is active.

16.6. Quad Chorus

Choruses are used to enrich sounds by adding spatial movement and giving them an ensemble-like quality. The Quad Chorus comes with multiple characteristics and different play modes. The Chorus is true stereo.

The Quad Chorus contains the following parameters and controls:



- **Modes**: Selects one of five modes that determine how the four delay lines are activated and modulated in different ways. The modes are based on the most successful classic chorus effects.
  - **TriVintage**: Modeled after a famous device, this mode use only three of the available four delays. It can reproduce the classic sound of a triphase chorus, but also offers additional controls, expanded ranges and stereo possibilities.
  - **Quadron**: Follows the same principles as TriVintage mode, but is optimized for a stereo signal. It can be used for subtle to over the top chorusing.
  - **Random**: This mode uses random walk generators instead of the internal LFOs. This avoids audible modulation patterns, making it ideal for subtle chorusing.
  - **Even**: Offers a special combination of two synced LFO's, spread and applied to four delay lines.
  - **Manual**: Gives you direct access to the basic core of the algorithm, in this case the four delays. You can use this to create a subtle aural room effects, or modulate it with the LFO´s, envelopes or performers. You can access the delay lines by the four controls, labeled **Time Left1**, **Time Left2**, **Time Right1**, **Time Right2**.
- **Flavor**: Selects one from five different settings (Neutral, Light, Controlled, Warm, Dark) that determine the coloring of the chorus. Which flavor fits best is highly dependant on what you want to achieve, but their names suggest the general character of each setting.
- **Mix**: Blends between the input signal and the effect signal, determining how much chorus is applied. Turn the control fully left to bypass the effect. The sweet spot often lies around center position.
- **Stereo**: Adjusts the width of the stereo field. Turn left for a narrow stereo field, and turn right to widen the stereo field of the chorused signal.

- **Distance**: The time distance of the modulated delay lines to the original sound. Turn the control right for longer settings that will result in a recognizable delay effect. It can also produce interesting results for auxiliary modulation when the internal LFOs are not used. The Distance control is not available in **Manual** mode.
- **Rate**: Sets the Modulation Rate of the internal LFOs. The LFO shapes cannot be chosen, as they are an integral part of the models. The Rate control is not available in Manual mode.
- **Amount**: Sets the range of how much the LFO is animating the delay lines. The Amount control is not available in **Manual** mode.

16.7. Equalizer

The Equalizer is used to balance and shape the frequency content of a sound. Internal algorithms make this equalizer sound particularly musical.

The Equalizer contains the following parameters and controls:

- **Freq**: Sets the frequency of the high shelf filter (1.2kHz to 23.6kHz).
- **Hi Gain**: Adjusts the level control of the high shelf filter (-24dB to +24 dB). Center position is 0dB.
- **Freq**: Sets the frequency of the parametric mid band (90Hz to 14kHz). Center position is 2.14 kHz.
- **Mid Gain**: Controls the boost factor of the mid band (from -24dB to +24 dB). Center position is 0dB.
- **Q**: The Quality Factor (Q) gives you control over the sharpness or bandwidth of the filter. A wide and round setting is produced when the control is turned left. Turning the control right creates a narrow resonance.
- **Low Gain**: Adjusts the level of the low shelf filter (-36dB to +18 dB). Center position is 0dB. The frequency range is dependant on the level boost, sitting between 120Hz and 260Hz.

16.8. Stereo Expander

This module is the next iteration of the Dimension Expander found in the original MASSIVE. It is cleaner, more versatile and less CPU intensive than its predecessor. The Stereo Expander can be used to create room-style spatial effects for a wide range of mono and stereo sound sources. It has a clear, diffused character sound.

The Stereo Expander contains the following parameters and controls:

- **Amount**: Adjusts the strength of the effect.
- **Time**: Adjusts the delay time of the effect. Turning the control right increases the delay time, making the sound appear more distant.

16.9. Stereo Delay

The Stereo Delay offers individual time control over the left and right channels, making it an extremely powerful and versatile stereo delay effect.

The Stereo Delay contains the following parameters and controls:

- **Route**: Determines if the internal feedback is in Parallel or Cross mode. When **Parallel** is selected, the output from the left delay is fed into the left delay input and the output from the right delay is fed into the right delay input. They do not interact with each other. When **Cross** is selected, the output from the left delay is fed into the input of the right delay, and the output from the right delay is fed into the left delay input. This creates different reflection patterns that are typically longer and more complex. By default, Route is set to Parallel.
- **Color**: Selects between three different characteristics (Neutral, Warm, Hot). With a **Neutral** setting, the internal EQ filters have no resonance and there are almost no nonlinearities, making this the most clean sounding with a neutral character. With a **Warm** setting, the internal EQ filters have some resonance, creating a warm character sound. **Hot** has strong nonlinearities and the internal EQ filters have a strong resonance that is very audible in the final sound.
- **FB**: Determines the amount Feedback applied to the signal. Applying more Feedback increases the number of echoes. Each one of these repetitions will gradually fade out as new ones are produced, with shorter delay times typically causing reflections to disappear faster than longer delay times. The Stereo Delay rescales the reflection levels so that the decay is time independent.
- **Mix**: Blends between the input signal and the effect signal. Turn the control fully left to bypass the effect, and right to mix in the delay.
- **Sync**: Selects one of two basic time manipulation modes (Sync, Free). In **Sync** mode, the fader scans through five individually assignable, synchronized times in a quantized manner. Adjust the individual dominators and denominators by clicking a number and dragging the mouse up or down. This delay's sophisticated algorithms enables you to jump from one time division to another without audio artefacts. This also applies when the host temp is changing the master tempo of the effect. This makes it possible to modulate the faders. When **Free** is selected, the stepped fader(s) is replaced by a continuous fader, operating in a manner typical to vintage delay.
- **Mono**: When activated, **Mono** modulation is switched on. The effect remains in stereo, but the delay time is controlled by a single fader for both channels.
- **Latch**: This button enables you to recall the delay times with note-on messages received from the synth engine. This means you will only hear the change of delay times, when a new note is pressed. This enables you to synchronize the delay changes to actual notes playing and is available in both **Sync** and **Free** modes.
- **Send**: Determines the amount of input signal that will be routed into the delay engine.
- **Flutter**: Moves the delay time slightly, creating a more lively effect and avoiding fixed phase relationships to the input signal. This control can be likened to a subtle chorusing.
- **Color**: Adjusts the internal filters of the delay circuitry. Turn the control left for a strong lowpass effect that becomes weaker towards center position. At center position the color is neutral. Turn the control right to increase the highpass filtering.

17. PERFORMERS

The three Performers are specialized sequencers for modulation. They provide a flexible way to apply complex, rhythmical movements to parameters that usually are achieved by automation created in a host sequencer. Therefore they are key to make intricate modulations part of a MASSIVE X sound.

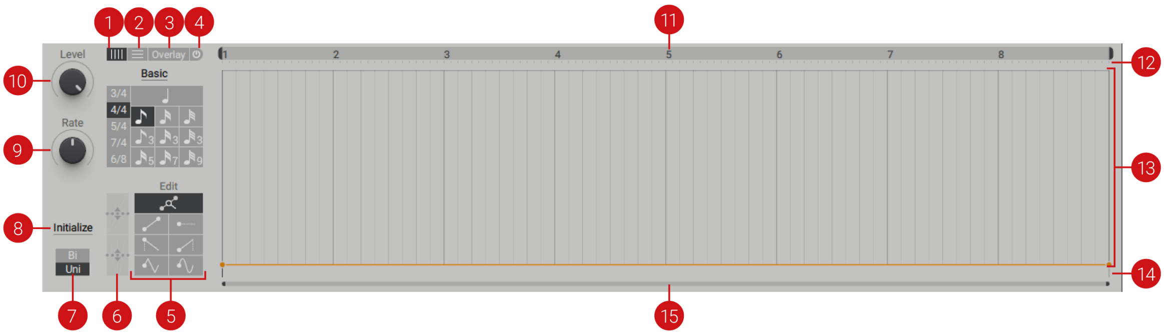
They are accessed and assigned via the Navigation Bar using the tabs labeled **P1**, **P2**, and **P3**. For information about assigning the Performers to parameters, see Assigning Modulation.

Basic and Custom editor options that enable you to quickly paint complex modulation shapes onto the Performer grid in order to form up to 12 patterns. The default Basic editor offers flexibility, while the Custom editor allows you to create a complex rhythmic grid for tailored and precise results.

The Performers can be controlled using the Remote Octave, which allows you to switch between patterns remotely using MIDI notes as key switches, or by selecting one of the 12 patterns using the mouse in the footer of MASSIVE X. For more information about the Remote Octave, see Remote Octave.

17.1. Overview of the Performers

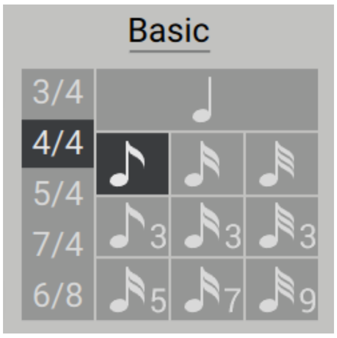
This section provides an overview of the Performers' features, including the Grid options, the Overlay, and the Paint tools.



(**1**) **Vertical grid tab** (X-Axis): Enables you to set the rhythmic timing for painting shapes on the grid, and contains two sets of tools: **Basic** and **Custom**. You can switch between these tools by clicking the underlined **Basic** / **Custom** label:

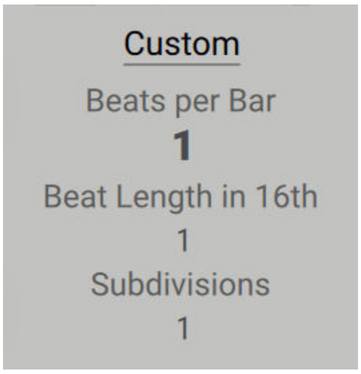
Basic Tools

The **Basic** tools are the default option to define the rhythmic grid for adding modulation shapes to the Performer grid using the painting tools (**5**). Use Basic mode to define the time signature and rhythmic resolution of the grid from quarter notes down to 32nd note divisions.



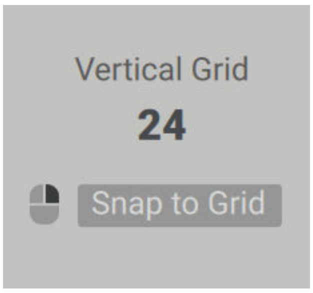
Custom Tools

The **Custom** tools are used to create custom time signatures with up to eight different sections. Use the Custom Grid to create your own complex time signatures. When using Custom mode, set the number of beats per bar, a beat length, and subdivisions.



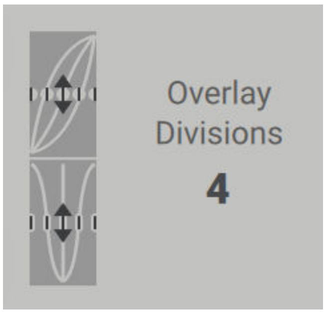
You can safely switch between the Basic and Custom grid without losing your modulation pattern. Switching between them only changes the underlying grid, not the pattern.

(**2**) **Horizontal grid tab** (Y-Axis): Provides options to set the number of steps for parameter values (quantization) in the Performers' Horizontal grid. By default, the number of steps is set to 24 (bipolar -24 to +24, unipolar 0 - +24). The **Snap to Grid** option ensures the grid value is adhered to even when making fine adjustments by holding the right mouse button and dragging.



An important thing to bear in mind when using the grids is that they are not separate, they simply work together to provide flexibility when adding modulation.

(**3**) **Overlay grid tab** : When the overlay is switched on (**4**), the overlay placement and width can be adjusted using the three handles that appear in the Select zone (**14**). The number of divisions is set by dragging up or down over the **Overlay Divisions** value. To the left, the overlay also features stretch controls that can be used like the ones found next to the painting tools. The difference is that the grid itself is stretched (not the position of the modulation points of a multi-selection). This makes it easy to use the overlay as a guide for creating modulation sections that speed up or slow down. The overlay can also be used to create wider grid values than quarters or odd divisions over any irregular timing.



(**4**) **Overlay On/Off switch**: Displays or hides the grid overlay.

(**5**) **Painting tools**: The selected brush defines the modulation shape that is painted on the grid (**13**). The length of a shape is determined by the length of the grid division. For more information on the painting tools, inserting and editing modulation, see Editing Modulation, and Painting Modulation Painting Tools.

(**6**) **Stretch tools**: Select multiple points in the Select zone and use these tools to stretch or compress the modulation curves in the pattern by dragging upwards or downwards. The upper tool stretches or compresses points to the left or right. The lower tool stretches or compresses points inwards or outwards.

(**7**) **Range switch**: Sets the range of the Performer to bipolar or unipolar.

(**8**) **Initialize**: Click to clear all modulation data and reset the grid to bipolar or unipolar.

(**9**) **Rate**: Set the speed of the Performer based on the tempo of your host; at the center position, the speed is the same as your host. At the far-left position, the rate is at an eighth of the host, and at the far-right, it is eight times that of the host.

(**10**) **Level**: Sets the output level of Performer modulation. When turned to the left position, the output is at zero, when turned to the right position, it's at 100%.

(**11**) **Start/End markers**: You can drag the marker handles to define the start and endpoints of the section of the modulation you want to play. The marked section will play according to the playback mode selected in the Performer Grid Overview. For more information on selecting playback modes in the Performer Grid Overview, see Overview of Remote Octave.

Double click the Start/End marker bar to adjust the playback area to the visible section in the editor, as set by the Zoom bar.

(**12**) **Select zone**: Click and drag in this area to select multiple modulation points. For more information on editing modulation, see Editing Modulation.

(**13**) **Grid**: The area where modulation is painted onto the flexible grid using the painting tools (**5**).

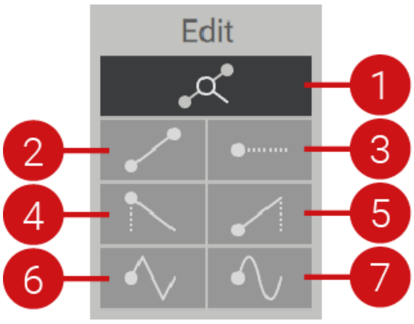
(**14**) **Segment Edit zone** : Use this area to edit single or multi-selected segments. Right-click and drag to fine-tune segments on the grid without them snapping to the quantize value. Double-click on a segment or multi-selection to delete it.

(**15**) **Zoom bar**: Click and drag the handles to zoom in and out of the Performer timeline. Double-click the Zoom bar to zoom into the first bar of the grid.

17.2. Painting Tools

The painting tools enable you to paint modulation shapes onto the rhythmic grid.

Here is an overview of the painting tools:



(**1**) **Edit**: Use to add modulation points to the Performer grid. The points are connected as straight lines to neighboring points.

(**2**) **Line**: Creates lines from the start to the endpoint of a grid segment.

(**3**) **Step**: Creates static values that extend to the end of a grid segment.

(**4**) **Ramp A**: This tool creates descending ramps.

(**5**) **Ramp B**: This tool creates ascending ramps.

(**6**) **Triangle**: Creates triangle shapes.

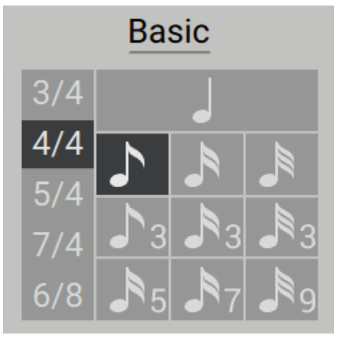
(**7**) **Sine**: Creates sine shapes.

17.3. Painting Modulation

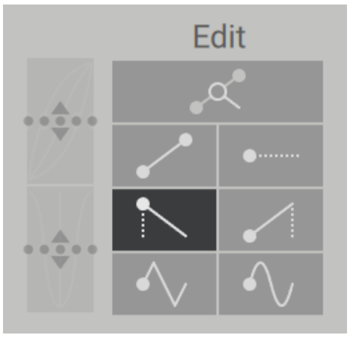
Select one of the shape brushes to start painting modulation on the grid. As you use the brushes to create shapes, they will snap to the selected time signature division and note value. For more information on selecting time signatures and note values, see Overview of the Performers.

To draw modulation onto the grid:

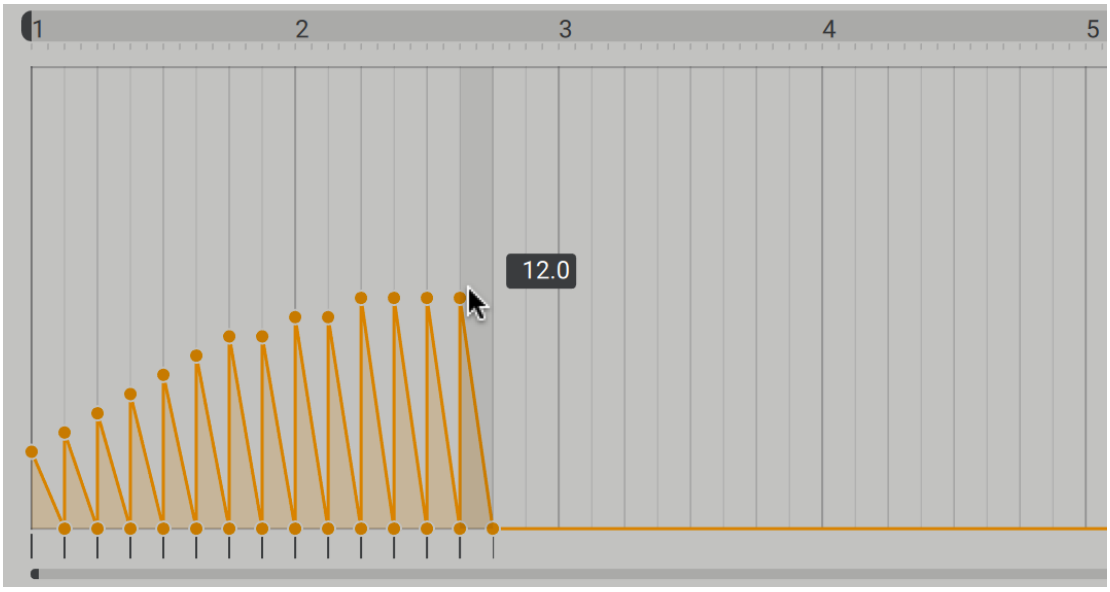
1. Select a time signature and grid note value.



2. Select a paint tool.



3. Click and drag the mouse across the grid to paint modulation.



4. Move the mouse up and down while painting to change the MIDI data amount.

The Performer only starts playing when it is assigned to a parameter of an active module. For more information on assigning Performers to parameters, see Assigning the Performers.

17.4. Editing Modulation

This section provides an overview of the features available for editing modulation in the Performer grid.

Selecting Modulation

To select modulation points:

- In the Selection Zone, drag across any selection of modulation points to highlight them.

For more information on the Selection Zone, see Overview of the Performers.

Deleting Modulation

You can delete individual or multiple modulation points.

To delete a single modulation point:

- Double-click the modulation point.

To delete multiple modulation points:

- In the Select zone, select the points that you want to delete, then double-click in the Segment Edit zone.

Alternatively, you can select a modulation point and drag left or right. Any existing points from the selected modulation point to the target location are deleted as the mouse is moved.

To delete all modulation within a Performer:

- Click the **Initialize** button and select **Bipolar** or **Unipolar** or double-click the pattern in the Remote Octave overview to reset the grid.

To delete all three modulation patterns for all three Performers:

- Double-click in the lower grid area of the Remote Octave overview.

For more information on the Remote Octave, see Overview of the Performers.

Moving Modulation

You can move modulation points in values quantized or unquantized to the grid value.

To move a selection of modulation points in steps quantized to the grid:

- In the Select zone, select the points that you want to move, then in the Segment Edit zone, drag them horizontally.

To fine-tune a selection of modulation points unquantized to the grid:

- In the Select zone, select the points that you want to move, then in the Edit zone, drag horizontally holding the right mouse button.

For more information on the Select zone and Segment Edit zone, see Overview of the Performers.

Bending a Modulation Curve

- Place the mouse over an existing modulation curve on the grid, then click and drag upwards or downwards.

18. REMOTE OCTAVE

Remote Octave provides a special key zone with access to 12 optional variations of the Performers **P1**, **P2**, **P3**. These alternative sets can be switched remotely using MIDI notes as key switches or by selecting one of the 12 patterns using the mouse in the footer of MASSIVE X. This way drastic changes can be performed or programmed by triggering remote key switches from your host.

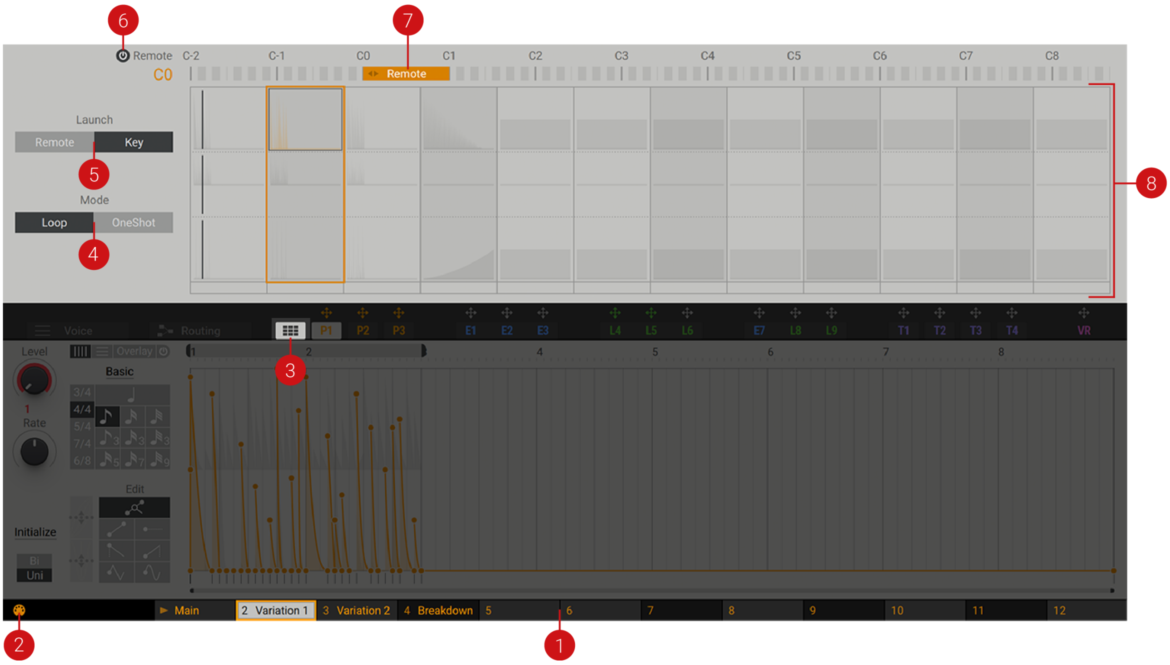
You can think of the Remote Octave as a remote control on your keyboard controller to select different Performer modulation patterns. This can be used in a performance to change modulations on the fly for variation, or for precise control when programming remote sequences in your favorite host sequencer.

In the Performer Grid view, a simultaneous overview of all three Performers is available, making it easy to copy, swap, and delete the 12 patterns from each Performer. You can also set up how patterns are triggered and how they can be changed with key switches using the Remote Octave.

For more information about the Performers, see Performers.

18.1. Overview of Remote Octave

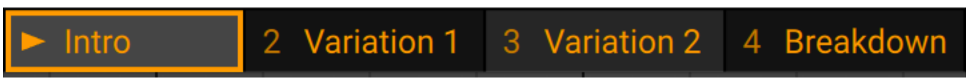
This section provides an overview of the Remote Octave features.



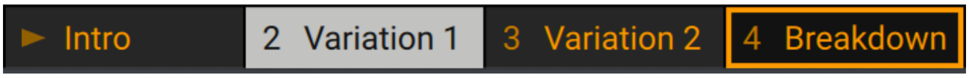
(**1**) **Remote Octave Keys**: The bottom area holds the 12 keys of the Remote Octave. This area displays information about the activity states and is used to manually change the active patterns of all Performers using the mouse. The patterns here directly correlate to each note of the chromatic scale within the Remote Key Zone (**7**).

Here's an overview of the Remote Octave key states:

- A Play icon next to the name of a pattern indicates the Performer pattern currently playing.



- An orange outline highlights which pattern will play when the next launch trigger is received. This signal can be triggered by a change of the Remote Octave (for example, switching via the mouse in the footer or MIDI note in Remote Octave zone) or by a MIDI note, see (**5**) for more information.



- An orange fill indicates a pattern that is currently open in the Editor and playing. Selecting a key with a right-click will open the associated pattern to be edited while another pattern is playing.

- A gray fill indicates the pattern currently being edited but not playing. It is possible to have one pattern playing while editing another, to do this, right-click a pattern other than the one playing.

(**2**) **Remote MIDI control**: This button mutes incoming MIDI notes to the Remote Octave key zone (defined on the Performer Grid view (**8**). This function can be used, for example, to disable remote switching from a host sequencer while editing and listening to the modulation effect of a Performer pattern.

(**3**) **Performer Grid view**: The Grid option button becomes visible when the Performer editor is selected and shown in the lower area of MASSIVE X. It opens the Grid view that holds playback options for the Performer and an overview of all Performer patterns.

(**4**) **Trigger modes**: This selector switches between **Loop** and **OneShot** modes.

- **Loop**: The section within the Start /End Markers will repeat.
- **OneShot**: The section within the Loop Markers will play once only.

The Performer options are globally applied to all three Performers.

(**5**) **Performer Launch options**: This selector switches between the Performer Launch options, as follows:

- **Remote**: Starts or retriggers a modulation pattern directly when a pattern is changed or reselected. Use the Remote option when you want to sequence modulation pattern changes with your host sequencer.
- **Key**: Starts or retriggers a pattern when a note is played. Use Key when you want to restart the modulation pattern every time a note is played.

(**6**) **Remote Power button**: This button enables/disables control of Remote Octave key switches via MIDI. This deactivates the key zone filtering so that the whole keyboard range can be used to play notes.

(**7**) **Remote Key Zone**: The key zone for MIDI note control of the Remote Octave can be shifted up and down in octaves by dragging the Remote handle left and right.

(**8**) **Performer Grid Overview** This is used to select a modulation pattern for editing. It features the **P1** at the top, **P2** in the center row and **P3** at the bottom row. The column holding the patterns currently playing is highlighted with an orange outline. The pattern visible in the editor has a black outline. There is also a playback cursor as seen in the Performer editor. The 12 pattern variations are placed next to each other on the same row. The Grid offers an overview of all modulation patterns and provides a convenient way to copy and swap pattern sets and individual patterns.

- To copy a pattern, drag and drop a pattern onto a different slot.
- To swap the patterns, right-click and drag and drop between two slots.

19. MODULATORS

Modulation is a key component of all forms of synthesis. It is the way in which you shape a signal and the tool with which to breath life and movement into a sound. MASSIVE X provides nine modulation sources in the form of Modulators, which can be used to control various parameters across the synth. Modulator 1 is a dedicated **Amp-Envelope**, hard-wired to control the Amplifier level. Each of the remaining eight Modulators can be assigned to any of the four modulation sources: **Modulation Envelope**, **Exciter Envelope**, **Switcher LFO,** and **Random LFO**. On a basic level, Modulators can be used to apply LFOs or envelopes to create simple contours or add motion to a sound. However, through an extensive network of modulation routing possibilities, as well as the specific controls of the Modulators themselves, more intricate arrangements can be constructed.

Envelopes are one of the most common and powerful tools for contouring your synth sound. In more typical use cases, envelopes control the loudness of an amplifier, shape the tone and color of a filter, or adjust the pitch of a signal. They can also be used for more creative objectives, like applying modulation to the speed of a sequence, the delay time of a stereo effect, or the rate of another Modulator.

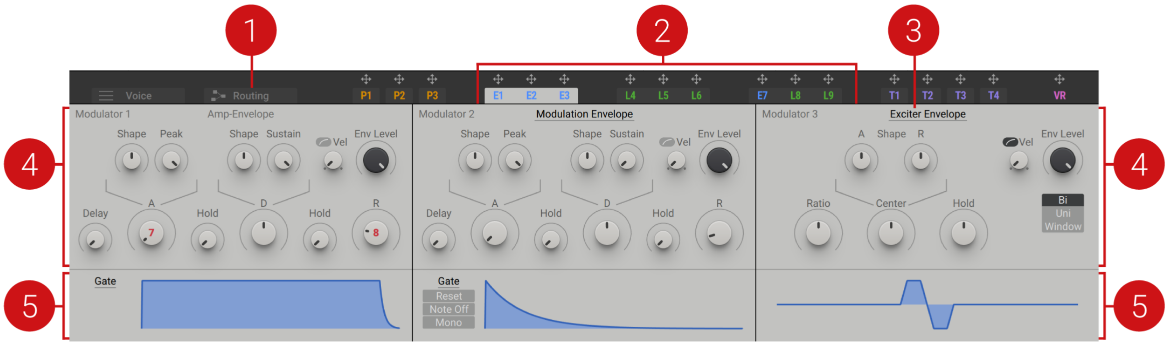
While envelopes create motion by defining the contour over a set of time and level based stages, LFOs typically have a fixed wave shape, useful for producing cyclical, predictable rhythmic modulation. LFOs (low frequency oscillators) produce a signal below the human range of hearing. As the signal cannot be heard, it is ideal for producing movement and animation, adding a sense of motion and depth to a sound. Modulation effects can become a defining character of a sound, for example, vibrato when an LFO is applied to the pitch of an oscillator, or tremolo when applied to an amplifier. Slow LFOs allow for subtle, elongate changes over time, while higher frequency rates can be used as a source for frequency modulation, amplitude modulation, and other types of audio-rate modulation.

The Modulators are accessed and assigned via the Navigation Bar using the tabs labeled e.g. **E1**, **E2**, **E3** for envelopes and **L1**, **L2**, **L3** for LFOs. For information on how to assign the Modulators to parameters, see Assigning Modulation.

The flexible and comprehensive range of modulation sources and routing possibilities provide considerable power for sound design. When properly exploited, the Modulators can go beyond your basic modulation needs, facilitating more advanced applications, like physical modeling, velocity-sensitive envelope shaping, multi-stage envelopes for complex shaping and so much more.

19.1. Overview of the Modulators

This section provides an overview of the Modulator section:



(**1**) **Routing Tab**: Two Modulation modules can be accessed via the Routing page, allowing you to use modulation sources as generators in the signal path. For more information, see Modulation Modules.

(**2**) **Modulators**: Displays the type and number of the Modulator assigned to each slot. Envelopes are represented with a blue **E** and LFOs are represented with a green **L**. Click on a Modulator to display the corresponding parameters. When a Modulator is selected, its tab is highlighted as in the image above. Click on the arrow icon above the Modulator to assign it to a modulation slot. For more information, see Assigning Modulation.

(**3**) **Modulator menu**: Selects one of four modulation sources (Modulation Envelope, Exciter Envelope, Switcher LFO, Random LFO).

(**4**) **Controls**: The knobs and menus in this section are used to edit the shape and behavior of the Modulators.

(**5**) **Display**: Provides a visual representation of the current shape of the Envelope, as determined by the trigger and knob settings. Moving the controls will show how each parameter influences the shape of the Envelope. Additional menus and settings relating to the behavior of the Modulator are also found here.

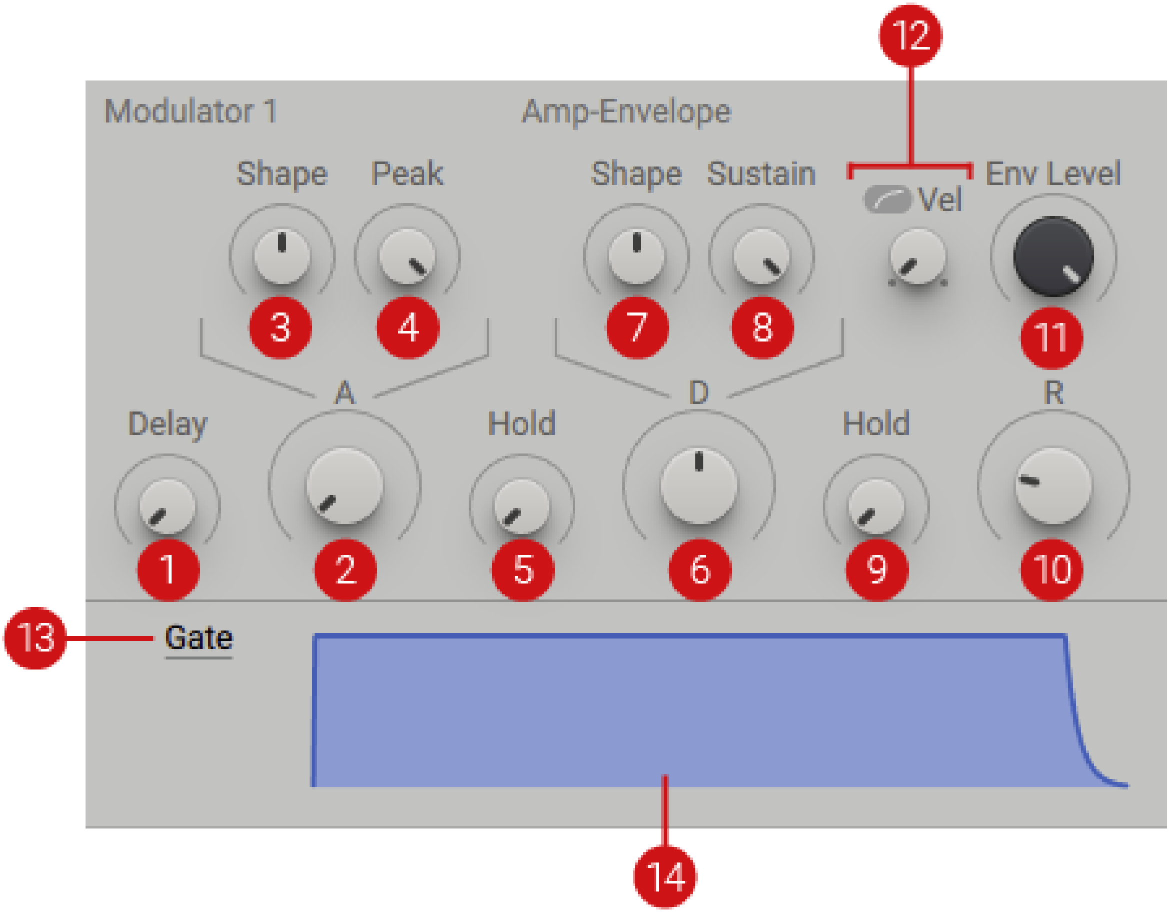
19.2. Amp Envelope

The Amp Envelope is used to control the response and shape of the Amplifier. Hardwired to the Amplifier, it offers individual scaleable curve controls over the **Attack** and **Release** stages, as well as **Delay** and **Hold** stages that provide additional control over the Amplifier's articulation.

In relation to human perception of amplification, small changes at low amplitude typically have a distinct and pronounced effect, while small changes made at higher volumes produce a less obvious result. Exponential shapes, as found on the **Shape** controls are especially effective for controlling the amplitude as they work to balance out the logarithmic shape of the amplifier, producing a smoother and more continuous change in volume.

The **Trigger** modes allow you to adapt to the way the Amplifier receives MIDI triggers. **Gate** mode is particularly useful when playing with a keyboard, as it provides a way to control the sound depending on the duration and intensity at which a key is pressed and held. For more information about the Amplifier, see Amplifier.

The Amp Envelope contains the following parameters and controls:



(**1**) **Delay**: Applies a delay to the onset of the envelope. When turned fully left, no delay time is applied and the envelope starts at the **Attack** stage. Turning the knob right increases the delay time.

(**2**) **Attack (A)**: Adjusts the time the envelope takes to reach the peak level. Turned fully left, the envelope will start immediately. As you turn the control right, the **Attack** becomes longer, and your sound will have a smoother start.

(**3**) **Attack Shape**: Changes the curve of the **Attack** stage of the envelope. Turning the knob left to right fades the curve from exponential to linear to logarithmic.

(**4**) **Attack Peak**: Defines the maximum level that can be reached. The scope of this control is dependant on the **Velocity** setting.

(**5**) **Hold**: Determines the fixed amount of time that the peak level of the envelope is held between the end of the **Attack** stage and the start of the **Decay** stage.

(6) Decay (D): Adjusts the amount of time it takes to fall from the attack's maximum **Peak** level to the level defined by the **Sustain** control. Turned fully left, the **Decay** stage will start immediately, and turning the knob right increases the Decay time.

(7) Decay Shape: Changes the curve of the **Decay** stage of the envelope. Turning the knob left to right fades the curve from exponential to linear, and then to logarithmic.

(**8**) **Decay Sustain**: Sets the amplitude of the **Sustain** stage.

(**9**) **Hold**: Determines the fixed amount of time that the peak level of the envelope is held between the end of the **Sustain** stage and the start of the **Release** stage.

(**10**) **Release (R)**: Defines the amount of time it will take for the envelope to fall from the set **Sustain** level and fade to zero.

(**11**) **Env Level**: Defines the overall level of the envelope.

(**12**) **Velocity**: Controls the influence of the incoming MIDI note’s velocity on the overall amplitude of the envelope. When turned fully left, the envelope amplitude is not influenced by the velocity of the incoming notes. When the fader is turned fully right, the overall envelope amplitude is directly proportional to the velocity of the incoming notes. The **Velocity button** changes the slope of the **Velocity** control from linear to logarithmic, when activated.

13) **Trigger**: Selects one of three settings (Gate, OneShot, LoopGate), that determine the envelope's response to incoming MIDI notes.

- With **Gate** is selected, the envelope is started and read out until its end when triggered. If the key is released before the **Sustain** stage of the envelope, it will immediately jump to the **Release** stage.
- With **OneShot** selected, the envelope is read to end, even if the key is released before the **Sustain** stage
- **LoopGate** creates a loop between the **Attack** and **Decay** stages.
- **Loop** creates a loop from the entirety of the envelope, including the **Release** stage.

(**14**) **Display**: Provides a visual representation of the envelope shape, based on how the knobs and menus are set. Move the controls described above to see how each parameter influences the shape of the envelope.

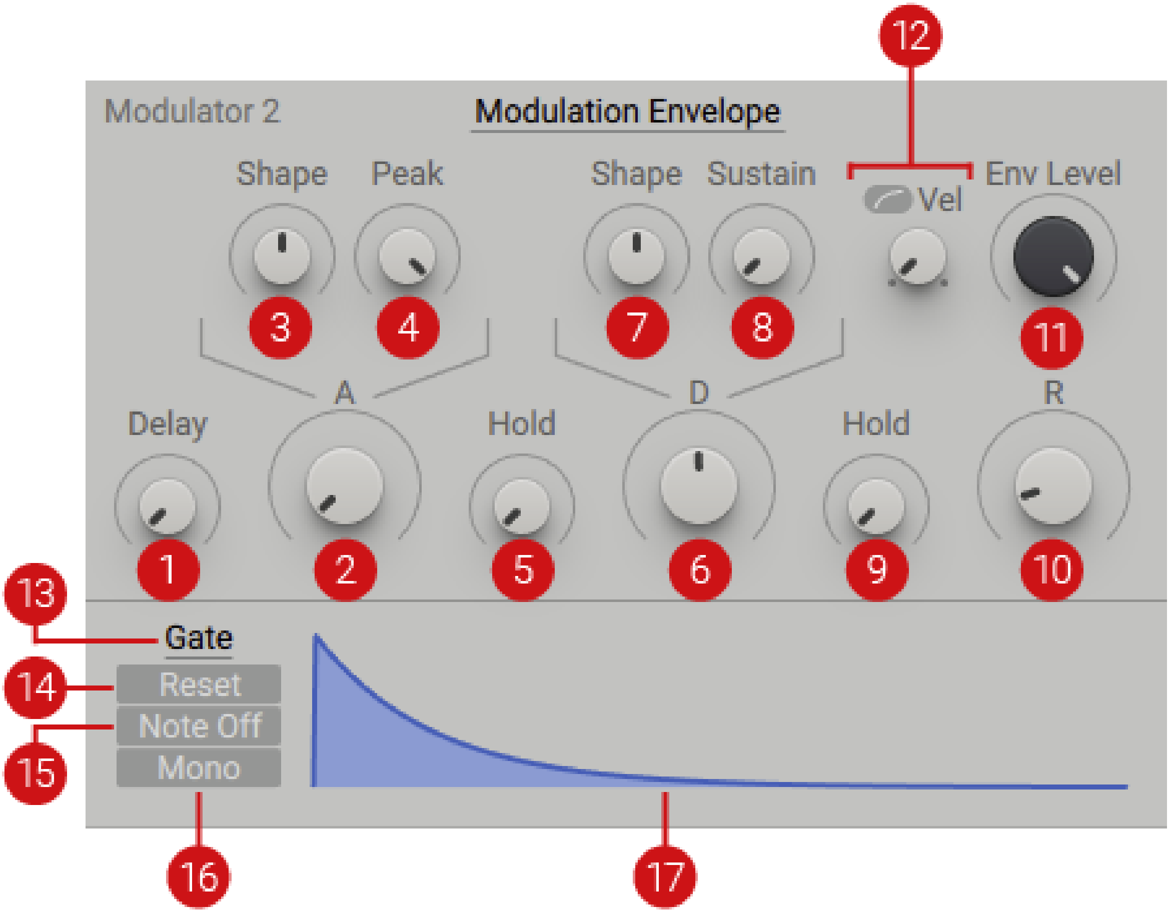
19.3. Modulation Envelope

The Modulation envelope in MASSIVE X offers a highly flexible and precise way of shaping your signal. Through comprehensive routing options, this envelope can be used to control virtually any aspect of the signal.

Alongside the standard **Attack, Decay, Sustain** and **Release** parameters, one **Delay** stage and two **Hold** stages provide additional control over the shape and contour of the envelope, enabling you to create sounds that seemingly have multiple sustain stages. The **Delay** control is typically used to delay the start of modulation, or to delay the impact of a modulation source. It occurs before the start of the envelope and can be particularly useful in sound design, for example, creating a measured time between the onset of the Amplifier envelope and a Filter envelope. By delaying the Filter envelope, a double attack-style effect is produced, opening a world of intricate sound design possibilities.

Attack Shape and **Decay Shape** allow you to determine the slope of the envelope when rising and falling. This allows for more precise sculpting and contouring, as different shapes are more suitable for different musical uses. Linear shapes set a direct path from one stage to the next, while exponential curves are commonly used to create pitch sweeps, as they create a more effective continuous sweep than a linear or logarithmic envelope.

The Modulation Envelope contains the following parameters and controls:



(**1**) **Delay**: Applies a delay to the onset of the envelope. When turned fully left, no delay time is applied and the envelope starts at the **Attack** stage. Turning the knob right increases the delay time.

(**2**) **Attack (A)**: Adjusts the time the envelope takes to reach the peak level. Turned fully left, the envelope will start immediately. As you turn the control right, the **Attack** becomes longer, and your sound will have a smoother start.

(**3**) **Attack Shape**: Changes the curve of the **Attack** stage of the envelope. Turning the knob left to right fades the curve from exponential to linear, and then to logarithmic.

(**4**) **Attack Peak**: Defines the maximum level that can be reached. The scope of this control is dependant on the **Velocity** setting.

(**5**) **Hold**: Determines the fixed amount of time that the peak level of the envelope is held between the end of the **Attack** stage and the start of the **Decay** stage.

(6)Decay (D): Adjusts the amount of time it takes to fall from the attack's maximum **Peak** level to the level defined by the **Sustain** control. Turned fully left, the Decay stage will start immediately, and turning the knob right increases the Decay time.

(7)Decay Shape: Changes the curve of the **Decay** stage of the envelope. Turning the knob left to right fades the curve from exponential to linear to logarithmic.

(**8**) **Decay Sustain**: Sets the amplitude of the **Sustain** stage.

(**9**) **Hold**: Determines the fixed amount of time that the peak level of the envelope is held between the end of the **Sustain** stage and the start of the **Release** stage.

(**10**) **Release (R)**: Defines the amount of time it will take for the envelope to fall from the set **Sustain** level and fade to zero.

(**11**) **Env Level**: Defines the overall level of the envelope.

(**12**) **Velocity**: Controls the influence of the incoming MIDI note's velocity on the overall amplitude of the envelope. When turned fully left, the envelope amplitude is not influenced by the incoming note's velocity. When the fader is turned fully right, the overall envelope amplitude is directly proportional to the incoming note's velocity. The **Velocity button** changes the slope of the **Velocity** control from linear to logarithmic, when activated.

(**13**) **Trigger**: Selects one of four settings (Gate, OneShot, LoopGate, Loop), that determine the envelope's response to incoming MIDI notes.

- When **Gate** is selected, the envelope starts and is read out until its end. If the key is released before the **Sustain** stage of the envelope, it will immediately jump to the **Release** stage.
- With **OneShot** selected, the envelope is read to end, even if the key is released before the **Sustain** stage.
- **LoopGate** creates a loop between the **Attack** and **Decay** stages.
- **Loop** creates a loop from the entirety of the envelope, including the **Release** stage.

(**14**) **Reset**: When active, the envelope will restart each time a note is triggered.

(**15**) **Note Off**: When active, the envelope is triggered with the note-off stage, for example when a key is released.

(**16**) **Mono**: When activated, all incoming notes receive the same envelope shape, regardless of pitch.

(**17**) **Display**: Provides a visual representation of the envelope shape, based on how the knobs and menus are set. Move the controls described above to see how each parameter influences the shape of the envelope.

19.4. Exciter Envelope

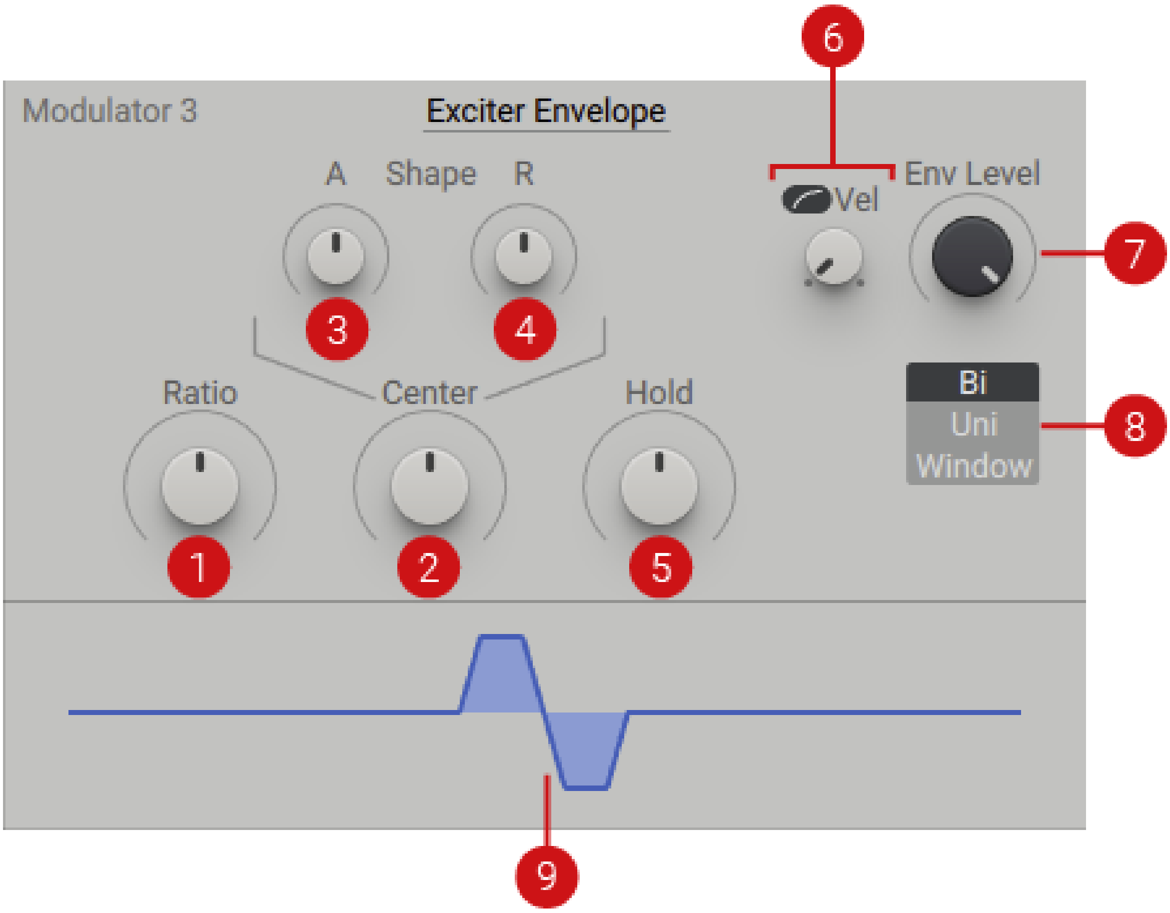
The Exciter envelope is particularly fast envelope that has been specifically designed to trigger the Comb filter, producing a resonator in physical modelling. It also effective for creating percussive sounds, with its quick attack and short release times.

The Exciter is unlike a standard ADSR based, multi-stage envelope generator. It is an Attack-Release envelope with short times, functioning in a single cycle, with a defined centre point and **Hold** stage that maintains the peak level from **Attack** to **Release** stages.

To use this envelope as an exciter for the Comb filter, it must be assigned to one of the Modulation Sources via the Routing page, routed to the Comb filter and then the filter routed to the output. This enables you to set the Comb filter into oscillation and create tones from the **Feedback**. A low **Ratio** setting produces the short envelope burst needed to create sound from the filter. For more information, see Comb.

You can also use the Exciter envelope to briefly trigger self-oscillation of resonant filters at high **Res** settings. This so-called filter pinging produces a damped sine wave that can be played via MIDI by using key tracking (**KTR**). The **SVF** filter types are especially suitable for this purpose.

The Exciter Envelope contains the following parameters and controls:



(**1**) **Ratio**: Controls the length of the envelope. Turning the knob right, increases the value, resulting in a longer envelope.

(**2**) **Center**: Tilts the envelope towards the **Attack** stage when turned left, or towards the **Release** stage when turned right.

(**3**) **Attack Shape**: Changes the curve of the attack (**A**) stage of the envelope from parabolic to logarithmic.

(**4**) **Release Shape**: Changes the curve of the release (**R)** stage of the envelope from parabolic to logarithmic.

(**5**) **Hold**: Increases the width of the peak stage of the envelope, determining how long the level of this stage will be held.

(**6**) **Velocity**: Controls the influence of the incoming MIDI note's velocity on the overall amplitude of the envelope. When turned fully left, the envelope amplitude is not influenced by the velocity of the incoming notes. When the fader is turned fully right, the overall envelope amplitude is directly proportional to the velocity of the incoming notes. The **Velocity button** changes the slope of the **Velocity** control from linear to logarithmic, when activated.

(**7**) **Env Level**: Defines the overall level of the envelope.

(**8**) **Polarity**: Selects one of three settings (Bi, Uni. Window) that determine the direction of modulation.

- When **Bi** is selected, the modulation is bidirectional.
- With **Uni** selected, the modulation is unidirectional and moves only in one direction.
- When **Window** is selected, the envelope is unidirectional and mirrored at the center. This removes the independent controls for attack and release, and offers only one shape control as the stages are now identical.

(**9**) **Display**: Provides a visual representation of the envelope shape, based on how the knobs and menus are set. Move the controls described above to see how each parameter influences the shape of the envelope.

19.5. Switcher LFO

The Switcher LFO offers a suite of LFO types, optimized for different use cases. The LFO is made up of the Rate, Shape, and Amp sections, running from left to right. The Rate section determines the speed and the play mode of the LFO. Depending on the mode selected, the **Rate** knob will change its appearance and functionality. **Osc** mode, provides an extremely fast modulation option that can be used in sync with the pitch of the Wavetable oscillators. The Shape section offers 16 different waveform shapes, from the typical Sine, Triangle, Saw and Square, to Ramp, Random, Spike and more.

Assigning a modulation source to the **Shape** selector allows you to scan through the different LFO shapes, creating some dramatic sonic effects. The **Fall/Rise** control in the Amp section can be used to determine the fade in and fade out of the LFO. The envelope is not velocity sensitive, and always restarts each time a MIDI gate is received. This can be particularly useful in musical applications, for example, when using the LFO for pitch vibrato, it enables you to fade in the vibrato slowly. The flexible shapes allow you to customise your LFO to tailor very specific modulation tasks.

The Play modes determine whether the LFO loops continuously or plays through one cycle only, and how the LFO cycle resets in response to incoming MIDI triggers. In addition to standard Gate and Restart behavior, **Loop REL** mode can be used to create interesting release effects.

The Switcher LFO contains the following parameters and controls:

(**1**) **Rate mode**: Selects one of three modes (Sync, Free, OSC), which determine the rate in which the LFO repeats its cycles.

- In **Sync** mode, the LFO is synchronized to the host tempo. It provides a slider with five positions, each of which can be programmed to your desired time division. Editing the numerator and denominator separately allows you to produce specific time divisions, and also facilitates more esoteric ones. **Sync** mode organizes the denominator values in a non-linear manner, to provide quick access to the most common values. Standard values **/4**, **/8**, **/16**, **/32** are then followed by triplets **/12** (8-note triplets), **/24** (16-note triplets), **/48** (32-note triplets), and so on, until 99. The change of **Rate** happens as soon as the slider hits a new position, with no fading in between.
- In **Free** mode, the **Rate** is absolute and independent from the host tempo. The slider subdivision are replaced with a continuous knob to control the speed of the LFO. The range, in Hz, extends from very slow to very fast rates. The knob is scaled in this respect, giving finer control in the middle ranges. The overall rate range is 0.004 Hz to approximately 60 Hz. Centre position is around 5.3 Hz, which is ideal for pitch vibrato.
- In **Osc** mode, the LFO operates at audio rate, becoming an additional keytracking oscillator. The values generated by the **Rate** control are linked to incoming MIDI notes, turning **Rate** into a control for transposition, ranging from zero when turned fully right, down to -96 semitones lower than the note pitch.

(**2**) **Latch Rate**: When activated, changes to the **Rate** (via direct control or modulation) are latched until the next note is received.

(**3**) **Waveform Selector**: Selects one of sixteen waveforms for the LFO. The new shape is updated with each cycle, providing a synchronized switch when heavily modulated.

(**4**) **Latch Shape**: When activated, changes to the **Shape** (via direct control or modulation) are latched until the next note is received.

(**5**) **LFO Level**: Amplifies the output of the LFO. Turning the knob right increases the amplification level.

(**6**) **Polarity**: Selects from three Polarity settings (Bi, Uni, Uni Z) that determine how the envelope in the Amp section behaves.

- When **Bi** (Bipolar) is selected, the output range of the LFO is -100% to +100%. The waveform always starts at 0%, independent of which shape is selected.
- **Uni** (UniPolar) offers a range of 0 to 100%. The waveform will always start at 50%, independent of the selected shape.
- **Uni Z** (UniPolar Zero) also has a range of 0 to 100% . The differences between **Uni** and **Uni Z** can only be heard when the oscillator is restarted with a new note trigger. The **Polarity** setting is reflected in the modulation.

(**7**) **Delay**: Applies a delay to the onset of the LFO. When turned fully left, no delay time is applied and the LFO starts immediately. Turning the knob right increases the delay time.

(**8**) **Fall/Rise**: Adjusts the way the LFO fades in or out. At centre position the LFO has infinite falling time, working as though it is always on. Turning the knob right produces a short rising ramp that becomes longer as it is turned fully right. Turning the knob left creates a very small decay fall time, which extends to a very long falling time as the control is turned fully left. The display, located below the **Fall/Rise** knob, provides a visual representation of the shape of the fades.

(**9**) **Mono**: When deactivated, the LFO is polyphonic. Each incoming note receives its own LFO. When activated, the LFO is monophonic and all incoming notes receive the same LFO, regardless of pitch.

(**10**) **Midi**: Switches between Midi and Remote mode for resetting and latching the LFOs. When **Midi** is selected the LFO resets and latches based on incoming MIDI note events. If **Remote** is selected the LFO resets and latches when you change key switches in the Remote Octave.

(**11**) **Play mode**: Selects from six settings (Loop, Loop RST, Loop GTE, Loop REL, 1shot, 1shot REL), that determine the general behavior of the LFO.

- **Loop** is the most classic setting, with the LFO running in an infinite loop, regardless of whether notes are being played or not.
- **Loop RST** (Loop Restart) also runs in an infinite loop, but will immediately jump to a given phase if a MIDI trigger is received from Zone or Remote.
- **Loop GTE** (Loop Gate) follows the same behavior as Loop Restart, but cuts the LFO as soon as the note is released.
- **Loop REL** (Loop Release) deactivates the LFO as soon as note is received. When the note is released, the LFO resets and the modulation is in effect. In this mode, it is important to set a long release time on the Amp envelope, or the effect will not be heard.
- When **1shot** (One Shot) is selected, only one cycle of the LFO will be played. Note on will always restart the LFO and go to zero after the cycle, regardless of the LFO value or polarity setting. This ensures predictable results, regardless of the LFO waveform.
- **1shot REL** (One Shot Release) follows the same principles as **1shot** mode, but creates a cycle when the note is released. In this mode it is also important to set a long **Release** time on the Amp envelope, or the effect will not be heard.

When in **Sync** mode, try modulating the **Rate** with another Switcher LFO. While sweeping through the positions on the slide, experiment in programming subtle variations in time divisions.

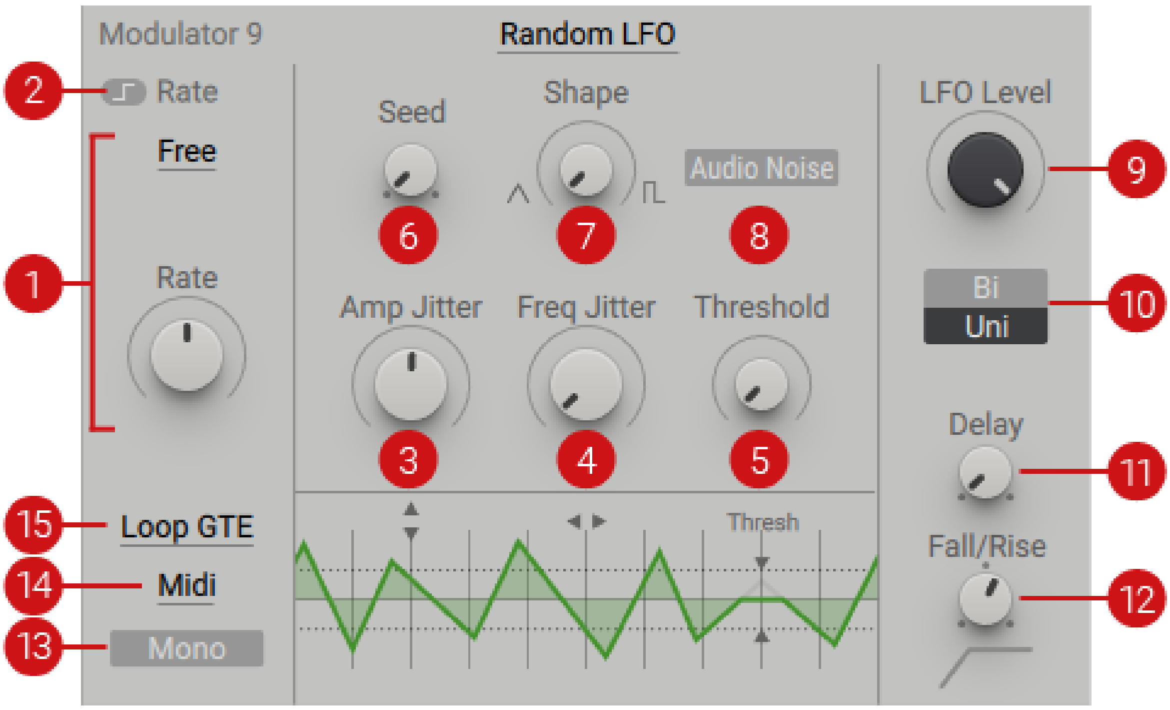
19.6. Random LFO

Adding randomness to your signal can inject human-like character to an otherwise highly controlled sound. This naturalistic quality is grounded in chance, probability and human error; the essence of what distinguishes a drummer from a drum machine. The Random LFO is a specialized LFO that generates different kinds of controllable variable random numbers, and provides controls to alter the range of randomness applied. It shares the same range of Rate and Amp controls as the Switcher LFO, excluding the **Uni Z** polarity setting. The **Shape** and **Jitter** controls offer further access to determine the type and amount of randomness produced.

The Random LFO can also be used as a generator by activating the **Audio Noise** button. Assigning the LFO to a Modulation source via the Routing page enables you to use the LFO as a noise source, in the audio signal path. For more information on using Modulators as generators, see Modulation Modules.

The Amp Jitter can be used to produce a precise amount of random values. Mixing between the two extremes, particularly with fast **Rates** settings, has a significant impact on the color of the noise. The independent controls can produce a wide range of noises and random modulations.

The Random LFO contains the following parameters and controls:



(**1**) **Rate mode**: Selects one of three modes (Sync, Free, OSC), which determine the rate in which the LFO repeats its cycles.

- In **Sync** mode, the LFO is synchronized to the host tempo. It provides a slider with five positions, each of which can be programmed to your desired time division. Editing the numerator and denominator separately allows you to produce specific time divisions, and also facilitates more esoteric ones. **Sync** mode organizes the denominator values in a non-linear manner, to provide quick access to the most common values. Standard values **/4**, **/8**, **/16**, **/32** are then followed by triplets **/12** (8-note triplets), **/24** (16-note triplets), **/48** (32-note triplets), and so on, until 99. The change of **Rate** happens as soon as the slider hits a new position, with no fading in between.

- In **Free** mode, the **Rate** is absolute and independent from the host tempo. The slider subdivision are replaced with a continuous knob to control the speed of the LFO. The range, in Hz, extends from very slow to very fast rates. The knob is scaled in this respect, giving finer control in the middle ranges. The overall rate range is 0.004 Hz to approximately 60 Hz. Centre position is around 5.3 Hz, which is ideal for pitch vibrato.
- In **Osc** mode, the LFO operates at audio rate, becoming an additional keytracking oscillator. The values generated by the **Rate** control are linked to incoming MIDI notes, turning **Rate** into a control for transposition, ranging from zero when turned fully right, down to -96 semitones lower than the note pitch.

(**2**) **Latch Rate**: When activated, changes to the **Rate** (via direct control or modulation) are latched until the next note is received.

(**3**) **Amp Jitter**: Adjusts the amount of random modulation applied to the amplitude of the signal. Turned fully left, the basic waveform of the LFO is unaltered. Turning the knob right increases the amount of unpredictable values. When turned fully right, a completely random value is produced with each cycle.

(**4**) **Freq Jitter**: Adjusts the amount of random modulation applied to the frequency of the signal. Turned fully left, the basic waveform of the LFO is unaltered. Turning the knob right increases the random frequency fluctuation. This can produce results similar to white noise, if **Amp Jitter** is set to a low value. **Freq Jitter** is only available in **Free** and **Osc** modes.

(**5**) **Threshold**: Provides further treatment for the amplitude, and only has an effect if the **Amp Jitter** is turned up. Random values below a set **Threshold** are forced to zero, which can be used to drastically thin out the noise.

(**6**) **Seed**: Adjusts the seed that feeds the random sequence introduced by **Amp Jitter** (**3**) and **Freq Jitter** (**4**). This control is only available in the play modes **Loop RST**, **Loop GTE**, and **Loop REL** (**15**). Adjusting **Seed** produces a new random sequence, starting with the next reset event. You can use this to explore different randomly acquired waveforms and repeat them with every new note.

(**7**) **Shape**: Adjusts the shape and smoothness of the ties between values. Technically speaking, it's applying a linear interpolation to the values. When turned fully left, a smooth interpolation is produced and when turned right, hard steps are created.

(**8**) **Audio Noises**: When activated, the LFO operates in audio rate. This turns the LFO into a Noise generator, which when assigned to a modulation source in the Routing page, can be used in the audio signal path.

(**9**) **LFO Level**: Amplifies the output of the LFO. Turning the knob right increases the amplification level.

(**10**) **Polarity**: Selects from two Polarity settings (Bi, Uni) that determine how the envelope in the Amp section behaves. When **Bi** (Bipolar) is selected, the output range of the LFO is -100% to +100%. The waveform always starts at 0%, independent of which shape is selected. **Uni** (UniPolar) offers a range of 0 to 100%. The waveform will always start at 50%, independent of the selected shape.

(**11**) **Delay**: Applies a delay to the onset of the LFO. When turned fully left, no delay time is applied and the LFO starts immediately. Turning the knob right increases the delay time.

(**12**) **Fall/Rise**: Adjusts the way the LFO fades in or out. At center position the LFO has infinite falling time, working as though it is always on. Turning the knob right produces a short rising ramp that becomes longer as it is turned fully right. Turning the knob left creates a very small decay fall time, which extends to a very long falling time as the control is turned fully left. The display, located below the **Fall/Rise** knob, provides a visual representation of the shape of the fades.

(**13**) **Mono**: When deactivated, the LFO is polyphonic. Each incoming note receives its own LFO. When activated, the LFO is monophonic and all incoming notes receive the same LFO, regardless of pitch.

(**14**) **Midi**: Switches between Midi and Remote mode for resetting and latching the LFOs. When **Midi** is selected the LFO resets and latches based on incoming MIDI note events. If **Remote** is selected the LFO resets and latches when you change key switches in the Remote Octave.

(**15**) **Play mode**: Selects from six settings (Loop, Loop RST, Loop GTE, Loop REL, 1shot, 1shot REL), that determine the general behavior of the LFO.

- **Loop** is the most classic setting, with the LFO running in an infinite loop, regardless of whether notes are being played or not. When turning **Amp Jitter** (**3**) and **Freq Jitter** (**4**) to the right, the waveform will randomly change over time.
- **Loop RST** (Loop Restart) also runs in an infinite loop, but will immediately jump to a given phase if a MIDI trigger is received from Zone or Remote. When turning **Amp Jitter** (**3**) and **Freq Jitter** (**4**) to the right, the waveform will change randomly over time, but start the same random sequence again upon receiving a MIDI trigger.
- **Loop GTE** (Loop Gate) follows the same behavior as Loop Restart, but cuts the LFO as soon as the note is released. When turning **Amp Jitter** (**3**) and **Freq Jitter** (**4**) to the right, the waveform will change randomly over time, but start the same random sequence again when the note is released.
- **Loop REL** (Loop Release) deactivates the LFO as soon as note is received. When the note is released, the LFO resets and the modulation is in effect. In this mode, it is important to set a long release time on the Amp envelope, or the effect will not be heard. When turning **Amp Jitter** (**3**) and **Freq Jitter** (**4**) to the right, the waveform will change randomly over time, but start the same random sequence again when the note is released.
- When **1shot** (One Shot) is selected, only one cycle of the LFO will be played. Note on will always restart the LFO and go to zero after the cycle, regardless of the LFO value or polarity setting. This ensures predictable results, regardless of the LFO waveform.
- **1shot REL** (One Shot Release) follows the same principles as **1shot** mode, but creates a cycle when the note is released. In this mode it is also important to set a long **Release** time on the Amp envelope, or the effect will not be heard.

To get started, try applying the Random LFO to the **Pitch** of an oscillator, or the **Wavetable Position**. Turn the **Shape** control toward the square shape to create rhythmic variations in pitch and timbre. Experiment with all the settings to hear the range of noises and random voltages you can create.

20. TRACKERS

The four Trackers are modulation sources that provide deep functionality for advanced keyboard tracking. They map incoming MIDI control data like pitch and velocity to modulation that you can apply to any parameter. This enables you to exactly define how your sound responds to the MIDI input.

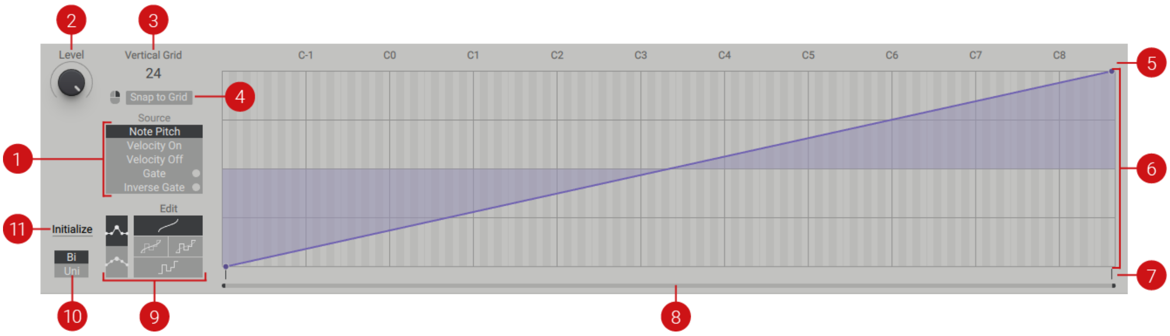
The Tracker's Tracking view allows you to view and create curves that represent the relationship between the MIDI input and the modulation produced by the Tracker. Each Tracker can be used with one of five available sources: **Note Pitch**, **Velocity On**, **Velocity Off**, **Gate**, and **Inverse Gate**.

As an example, you can use **Note Pitch** to define the response of your sound to the pitch values of incoming MIDI notes. When setting the Tracker's curve to a linear, rising ramp, it can be applied to parameters for standard keyboard tracking, meaning low notes produce low parameter values, and high notes produce high parameter values. In this case, the values produced by each MIDI note increase linearly across the keyboard, from left to right. By changing the Tracker's curve from a linear, rising ramp to other shapes or even adding irregular offsets to it, you can break up this relationship. For instance, inverting the curve to a falling ramp would result in high parameter values when playing low notes and vice versa, or adding extreme offsets to specific notes would produce high parameter values only when playing the corresponding keys.

You can access and assign the Trackers in the Editor via the Navigation Bar using the tabs labeled **T1**–**T4**. For information about assigning the Trackers to parameters, see Assigning Modulation.

20.1. Overview of the Trackers

This section provides an overview of the Tracker’s features, including the Tracking view, the Source options, and the painting tools.



(**1**) **Source**: Selects the type of MIDI control data processed with the Tracker. The following sources are available:

- **Note Pitch** is the MIDI note value. This option retains the pitch value until another note is played. Use this source if you want to track the pitch of the notes.
- **Velocity On** is standard, note-on velocity. This option retains the velocity value until another note is played. Use this source if you want to track the velocity of the notes.
- **Velocity Off** is note-off velocity. This option retains the velocity value until another note is released. Use this source if you want to track the note-off velocity of the notes.

- **Gate** is standard, note-on velocity. This option resets the velocity value to 0 when a note is released. Use this source if you want to track the velocity of the notes you are playing, but only for as long as a note is held.
- **Inverse Gate** is note-off velocity. This option resets the velocity value to 0 when another note is played. Use this source if you want to track the note-off velocity of the notes you are playing, but only for as long as no other note is played.

> This is useful for controlling envelope parameters when a note is released without affecting the notes that follow.

(**2**) **Level**: Sets the output level of modulation produced by the Tracker. At the left position, the output is at zero, at the right position, it's at 100%.

(**3**) **Vertical grid**: Sets the vertical resolution of values in the Tracking view (**9**). By default, the number of available values is set to 24 (bipolar -24 to +24, unipolar 0 - +24).

(**4**) **Snap to Grid**: Ensures the grid value is adhered to even when making fine adjustments by holding the right-mouse button and dragging.

(**5**) **Select zone**: Click and drag in this area to select multiple breakpoints. The segments between them can be edited using the Segment Edit zone (**7**).

(**6**) **Tracking view**: Here you can draw curves and offsets that define the Tracker's modulation output using the different edit modes (**6**). The content of the other Trackers is shown as greyed out curves or offsets in the background (depending on which type of content is active in the Tracker being edited.

(**7**) **Segment Edit zone** : Use this area to edit single or multiple segments between breakpoints. Clicking and dragging moves the segments to a new position in the Tracking view. Right-clicking and dragging allows you to make fine-adjustments without snapping to the grid. Double-clicking deletes segments.

(**8**) **Zoom bar**: Zooms and navigates in the Tracking view (**6**). Clicking and dragging the handles zooms in and out. Clicking and dragging the bar scrolls the contents. Double-clicking the bar zooms out completely.

(**9**) **Edit mode**: Determines whether the Tracker's modulation output is defined by a curve, offsets, or a combination of the two in the tracking view (**8**). For more information, see Tracking View and Edit Modes.

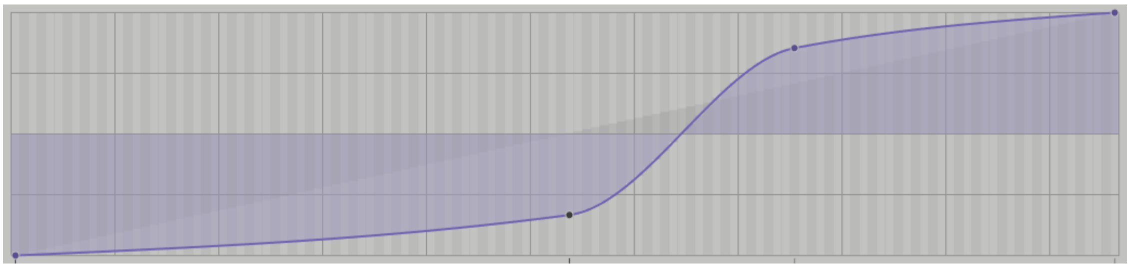
(**10**) **Range switch**: Sets the range of the Performer to bipolar or unipolar.

(**11**) **Initialize**: Clears all curves and offsets, and resets the grid to bipolar or unipolar.

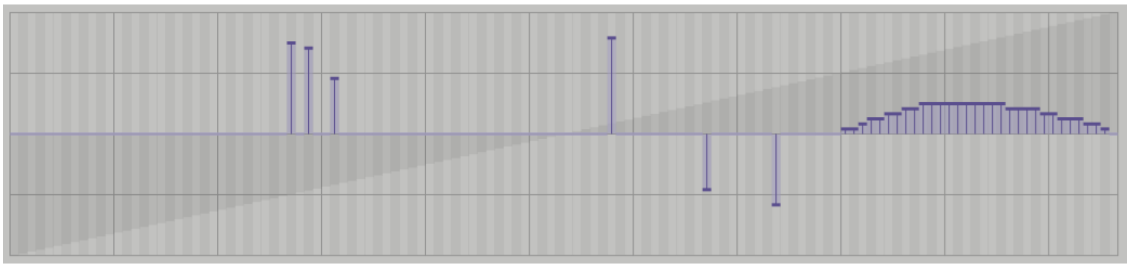
20.2. Tracking View and Edit Modes

The Tracking view is used to establish the relationship between the MIDI input on the horizontal axis and the Tracker's modulation output on the vertical axis. It can contain a curve, offsets, or a combination of the two. This determines not only how you interact with the Tracker, but also its modulation output.

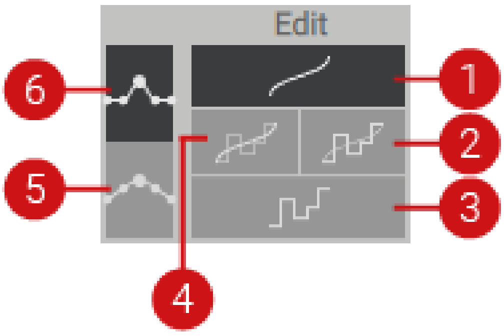
- The curve establishes the relationship between the MIDI input and the modulation output across the whole range. This is useful for generating modulation that responds to the MIDI input in a coherent way, adjacent MIDI input values will produce progressively changing modulation values. The shape of the curve can be set using any number of breakpoints, creating separate segments between them:



- The offsets establish the relationship between specific values of the MIDI input and the modulation output. This is useful for generating modulation that responds to the MIDI input in unexpected ways. Each MIDI input value can produce any modulation value, hence abrupt changes in modulation are possible from one MIDI input value to the next. The offsets can be painted in for each value individually:



You can switch between curve and offsets using the different Edit modes:



(**1**) **Curve mode**: Only the curve is active and can be edited. The following mouse interactions are available:

- Clicking on the Tracking view creates a new breakpoint.
- Clicking and dragging an existing breakpoint moves it to a new position.
- Right-clicking and dragging allow you to make fine-adjustments in one direction without snapping to the grid.

(**2**) **Combined Offset mode**: Both the curve and the offsets are active, however only the offsets can be edited. The following mouse interactions are available:

- Clicking and dragging adjusts the offset for multiple values.

- Right-clicking and dragging sets multiple values to 0.
- Clicking, holding, and then dragging up and down adjusts the offset for a single value.
- Right-clicking, holding, and then dragging up and down allows for fine-adjustments for a single value without snapping to the grid.

(**3**) **Offset mode**: Only the offsets are active and can be edited. The following mouse interactions are available:

- Clicking and dragging adjusts the offset for multiple values.
- Right-clicking and dragging sets multiple values to 0.
- Clicking, holding, and then dragging up and down adjusts the offset for a single value.
- Right-clicking, holding, and then dragging up and down allows for fine-adjustments for a single value without snapping to the grid.

(**4**) **Combined Curve mode**: Both the curve and the offsets are active, however only the curve can be edited. The following mouse interactions are available:

- Clicking on the Tracking view creates a new breakpoint.
- Clicking and dragging an existing breakpoint moves it to a new position.
- Right-clicking and dragging allow you to make fine-adjustments in one direction without snapping to the grid.

(**5**) **Absolute breakpoint editing**: Sets the behavior when editing breakpoints to absolute. Moving a breakpoint only affects the breakpoint and its corresponding segments. This option is only available in Curve mode (**1**) and in Combined Curve mode (**4**).

(**6**) **Relative breakpoint editing**: Sets the behavior when editing breakpoints to relative. Moving a breakpoint also affects the other breakpoints by shifting their position relative to a virtual rubber band. This option is only available in Curve mode (**1**) and in Combined Curve mode (**4**).

21. VOICE RANDOMIZATION

Voice Randomization is a modulation source that allows you to add pseudo-random variation to your sound. It generates a fixed modulation value per voice that can be applied to parameters, giving them a different value depending on which voice is played.

For example, this can be used to create subtle differences in tuning between voices like on an analog synth, or to create dramatic effects by completely changing the sound per voice.

You can assign Voice Randomization via the Navigation Bar using the arrow icon labeled **VR**:

For information about assigning Voice Randomization to parameters, see Assigning Modulation.