
ABSYNTH 6

Table of Contents

1. Disclaimer

Document authored by Native Instruments GmbH, Nicolas Sidi, Jan Ola Korte, Hannah Lockwood

Software version: 6.1 (04/2026)

Special thank you to all beta testers who contributed to making Absynth a better product!

2. Welcome to Absynth 6

Absynth 6 marks the return of a legendary semi-modular synthesizer. Built for textures that transform and breathe, it invites you to shape sounds that are entirely your own. Deep enough for complex sound design, and intuitive enough for newcomers, it opens up distinctive, living sounds for cinematic cues, ambient soundscapes, and electronic productions.

At the core of Absynth 6 is a hybrid engine that combines granular, FM, wavetable, and subtractive synthesis with deep modulation and a suite of creative effects. Its three oscillator channels give you structured unpredictability and endless room for exploration – perfect for sounds that evolve, shift, and surprise.

The Preset Explorer helps you find sounds by feel instead of scrolling through names. Navigate a sonic landscape where relationships between tones appear visually, leading you to unexpected discoveries and faster results.

This document shows you how to install and setup Absynth 6 and describes all features in detail, starting with the overview.

Document conventions

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

Italics	Indicates paths to locations on your hard disk or other storage devices.
Bold	Highlights important names, concepts, and software interface elements.
[Brackets]	References keys on the computer keyboard.
►	Denotes a single step instruction.
→	Denotes the expected result when following instructions.

The following three icons denote special types of information:

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

The **information** icon highlights essential information in a given context.

The **warning** icon alerts you of potential risks and serious issues.

What's new in Absynth 6

New in Absynth 6.1

The following features have been added or improved in Absynth 6.1:

- 32 new instrument presets and 100 new samples
- Undo / redo
- Audio modulator
- LFO retrigger
- Auto trigger
- Metadata editing for properties in the List browser
- Save / Save as functionality, including visual indication of preset modification

Presets saved in Absynth 6.1 will not open in Absynth 6.0.

3. Installation and setup

This chapter describes how to download and install Absynth 6 using Native Access and get started making music.

Installation using Native Access

Native Access is your go-to app for downloading, activating, and updating all your NI music creation tools including Absynth 6. If you are new to Native Instruments, you will first have to create your Native ID user account. To learn more about Native Access, visit our support page.

1. Download and install Native Access here.
2. Open the Native Access application.
3. Create a Native ID, if you do not have one already.
4. Login to Native Access using your Native ID.
5. Click **Library** on the left side of Native Access.
6. Click **Available** at the top of Native Access.
7. Click the **Applications** category to only show standalone products.
8. Click the **Install** button for Absynth 6.

→ The software is installed automatically.

Using Absynth 6 as a plugin in your DAW

The plugin version of Absynth 6 lets you use it as an instrument inside your DAW. You can run multiple instances of Absynth 6 alongside other instrument and effect plugins. When used as a plugin, Absynth 6 relies on your DAW to receive audio and MIDI, and its audio outputs are directly sent to your DAW's mixer.

Absynth 6 is available in VST3, Audio Units (AU), and AAX plug-in formats. Once installed, Absynth 6 will appear in the plugin lists of any compatible DAW installed on your system.

ⓘ If Absynth 6 does not show up in the plugin list of your DAW, you can read this article in our Knowledge Base for help.

To learn more about how to load and play instrument plugins like Absynth 6 in your DAW, refer to the documentation of your DAW.

Using Absynth 6 as a standalone application

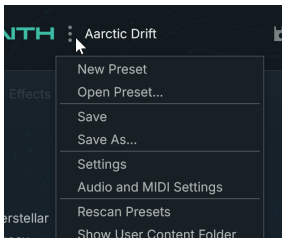
When you open Absynth 6 from your system's application folder, it launches as a standalone application independently of a DAW or another plugin host. The Absynth 6 standalone application receives MIDI from your MIDI keyboard or controller, and sends audio signals to your audio interface. For example, this allows you to use Absynth 6 as a live performance instrument on a dedicated computer.

When you open the Absynth 6 standalone application for the first time, you need to configure its audio and MIDI settings to connect Absynth 6 to your audio interface and to your MIDI keyboard or controller.

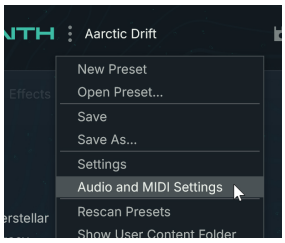
Configuring the audio and MIDI settings

When Absynth 6 is running as a standalone application, do as follows to configure its audio and MIDI settings:

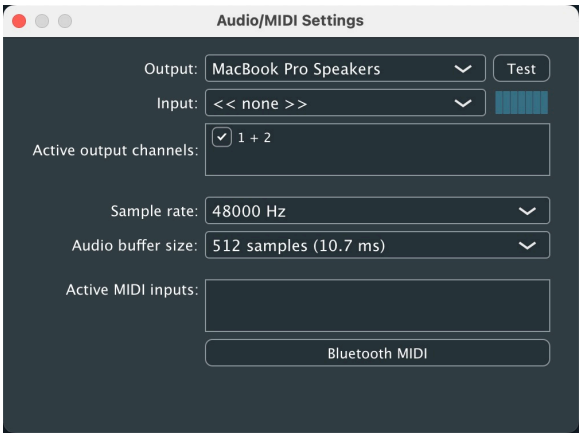
1. Click the three little dots in the Header of Absynth to open the Absynth menu:



2. Select **Audio/MIDI Settings** from the menu:



→ The **Audio/MIDI Settings** dialog opens.



In the **Audio/MIDI Settings** dialog, the **Active MIDI Inputs** setting at the very bottom is the only necessary setting for configuring MIDI. All the settings above relate to the audio configuration. The dialog includes the following settings, from top to bottom:

- **Audio device type**: Selects which of your operating system's device driver architectures Absynth 6 should use. Most professional audio devices provide ASIO, Core Audio (Mac) or WASAPI (Windows) drivers.
- **Device**: Selects the audio interface that Absynth 6 should use from all the connected audio interfaces that match the driver architecture chosen in the **Audio device type** selector above.
- **Active output channels**: Selects the pairs of outputs that will be available in Absynth 6.
- **Sample rate**: Selects the sample rate at which the selected audio interface Absynth 6 will operate.

- **Audio buffer size**: Selects the size of the audio buffer in samples. Small values will shorten the delay between pressing a key and hearing the resulting sound (this is called “latency”), but may cause drop-outs and stuttering when playing a lot of voices at the same time. Conversely, setting this to a higher value will make playback more reliable at the cost of more latency. Note that the **Audio buffer size** selector is not always available, as sometimes the latency is handled by your hardware drivers.

  ⓘ For some systems the **Audio buffer size** selector might only include a single entry corresponding to the buffer size currently selected. In that case, you can always click the **Control Panel** button below to open the control panel of your audio interface and select another buffer size from there.

- **Control Panel button**: Opens the control panel of the audio interface chosen in the **Device** selector.
- **Reset Device button**: Resets the selected audio interface. This suspends any existing audio stream.
- **Active MIDI inputs**: Selects the MIDI device(s) that you want to use for playing and controlling Absynth 6. The list shows all the MIDI devices currently connected to your computer. You can activate multiple MIDI devices and use them simultaneously.

ⓘ When Absynth 6 is used as a plugin in your DAW, the DAW handles the audio and MIDI communication with your external devices, so that you don’t need to configure the settings as described here. Read your DAW’s documentation for more information about configuring its audio and MIDI settings.

ⓘ In addition to the audio and MIDI settings, you might also want to adjust the surround configuration of Absynth among other instrument settings.

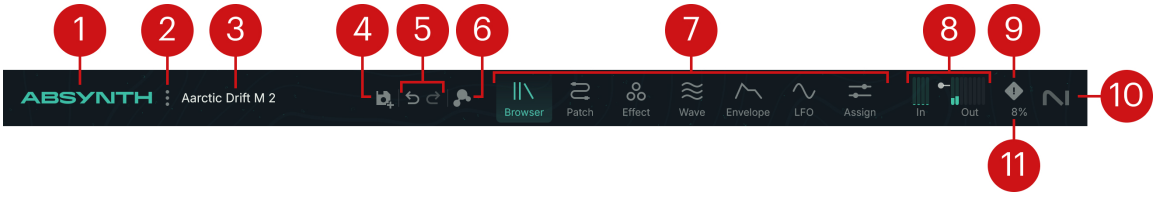
4. Absynth 6 overview

Absynth 6 consists of the following main areas:

1. **Header**: Contains global commands and indicators that let you switch between Absynth 6 presets, show the desired page of the instrument, check and adjust the input/output levels, and display the About screen. You can find more information in Absynth Header.
2. **Pages**: The biggest part of the Absynth 6 interface can show various pages, each page focusing onto a specific aspect of the instrument. You can select pages for display using the Page buttons in the Header.
3. **Macro controls**: Always visible at the bottom of the window, the Macro controls let you quickly adjust the Macros defined in the current preset. You can find more information in Macro controls.

Absynth Header

At the top of the instrument, the Header is always visible and provides a set of global commands, quick shortcuts, and indicators:



1. **Absynth logo**: Opens the About screen, which shows various information about the software like the version number, the serial number, and the credits.
2. **Absynth menu (three vertical dots)**: Provides a set of generic options and commands for creating, loading, and saving presets, adjusting various settings, changing the size of the user interface, and rescanning your preset library. Refer to Absynth menu.
3. **Preset selector**: Shows the name of the loaded preset. When you hover over the preset name with your mouse, three little arrows appear on the right that let you quickly load other presets from the current search results in the Browser: Clicking the left or right arrow loads the previous or next preset from the results, and clicking the downwards arrow opens the search results as a drop-down menu from which you can quickly select the desired preset for loading.
4. **Save / Save As… button**: Saves the current state of the instrument. The button operates based on whether a factory or user preset is loaded:
   - When a **factory preset** is loaded, the button shows a floppy disk with an additional "**+**" icon and acts as a Save As button: Clicking the button opens the **Save Preset As** dialog and lets you save a new user preset.

   - When a **user preset** is loaded, the button only shows a floppy disk and acts as a Save button: Clicking the button saves your changes to the current user preset, overwriting its contents.

     If the user preset has not been edited, the button is grayed out and inactive. When hovering over the button with your mouse, a down-pointing arrow appears below the button and lets you access the **Save As** command.

   > The **Save** and **Save As…** commands are also available in the Absynth menu.

5. **Undo and Redo buttons**: Clicking Undo (left arrow) reverts your last parameter change. Clicking Redo (right arrow) restores the parameter change that you have just undone. The Undo history (the list of your previous changes) is cleared when you load a new preset.
6. **Mutate button**: Triggers a new mutation of the current preset. This button is a shortcut for the **Mutate** button in the Mutator, at the bottom of the Browser page.

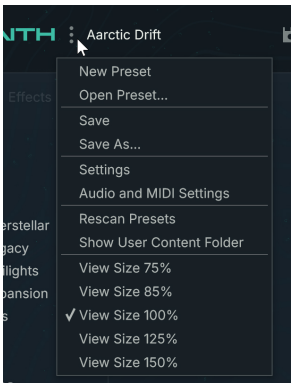
7. **Page buttons**: These buttons let you switch between the corresponding pages of Absynth 6. The highlighted button indicates the page currently displayed. The following buttons are available:
   - **Browser**: Opens the Browser page, which lets you explore and load your factory and user presets, and configure new mutations of the loaded preset. Refer to Browser page.
   - **Patch**: Opens the Patch page, which lets you combine and adjust the modules that make up the preset's sound. Refer to Patch page.
   - **Effect**: Opens the Effect page, where you can choose and configure the effect module used in the main channel of your patch. Refer to Effect page.
   - **Wave**: Opens the Wave page, which provides intuitive tools to adjust the existing waveforms or shape new ones from scratch. You can then use these waveforms in various modules of the instrument. Refer to Wave page.
   - **Envelope**: Opens the Envelope page, where you can design advanced envelopes to modulate the parameters of the instrument. Refer to Envelope page.
   - **LFO**: Opens the LFO page, which lets you configure up to three LFOs as additional modulation sources. Refer to LFO page.
   - **Assign**: Opens the Assign page, where you can configure the control, modulation, and automation of the Absynth 6 parameters in multiple ways. Refer to Assign page.
8. **Input/Output Level meters (In/Out)**: The **In** and **Out** LED bars indicate the level of the audio signals on the activated input and output channels, respectively. The LED bars for deactivated channels are grayed out and inactive. On the **Out** meters, a slider lets you adjust the overall output level of the instrument: You can drag the slider vertically with your mouse or double-click the slider and enter the desired value on your computer keyboard.

   ⓘ If Absynth 6 is running as a plugin, you can activate or deactivate the plugin's input and output channels in your DAW (refer to your DAW documentation for more details). If Absynth 6 is running as a standalone application, you can set up its output channels in the Audio/MIDI Settings dialog.

9. **Engine Restart button (! icon)**: Stops any audio stream and resets the audio engine. This can be useful when the instrument get stuck on notes or overload the CPU persistently. Any settings that you made in the instrument are preserved.
10. **NI logo**: Like the Absynth logo, clicking the NI logo opens the About screen, which shows various information about the software like the version number, the serial number, and the credits.
11. **CPU Load indicator**: Shows the CPU load in percent, giving you visual feedback about the available processing resources.

Absynth menu

You can open the Absynth menu by clicking the three dots right of the Absynth logo:



The Absynth menu contains the following entries:

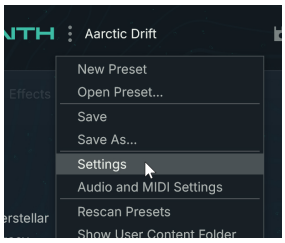
- **New Preset**: Creates a new empty patch containing only one basic Oscillator on Channel A. You can find more information on the patch structure in Patch page.
- **Open Preset...**: Opens a dialog that lets select and load another Absynth preset file on your hard drive. In the dialog, you can navigate to the containing folder, select the desired preset file, and click **Open** or double-click the file to load it. The selected preset will replace the current preset.
- **Save**: Saves the changes made to the loaded preset. If a factory preset is loaded or if you started from a new empty patch that has not been saved yet, the **Save** entry acts as the **Save As...** entry.
- **Save As...**: Saves the current state of the instrument as a new user preset. Clicking this entry opens a **Save Preset As** dialog in which you can navigate to the folder where you want to save your preset, type a new name for the preset, and click **Save** to save it. Instrument presets are saved in the NKSF file format, while effect presets are saved in the NKSFX file format. The saved preset will then appear in the instrument or effect category in the user content of the Browser.
- **Settings**: Opens a dialog providing global settings for Absynth. This dialog is described in Settings dialog.
- **Audio and MIDI Settings** (only visible in the standalone application): Opens the **Audio/MIDI Settings** dialog. This dialog is described in Configuring the audio and MIDI settings.
- **Rescan Presets**: Triggers a new scan of your Absynth 6 factory and user libraries. This is useful if you have manually moved some preset files or folders using the Finder (macOS) or the Explorer (Windows).
- **Show User Content Folder**: Opens the user content folder in the Finder (macOS) or Explorer (Windows).
- **View Size 75% – 150%**: Adjusts the size of the Absynth window.

> ⓘ You can define a default size in the General tab of the Settings dialog.

Settings dialog

The Settings dialog lets you adjust global settings for the instrument.

▶ To open the Settings dialog, click the three little dots in the Absynth Header and select **Settings** from the menu:

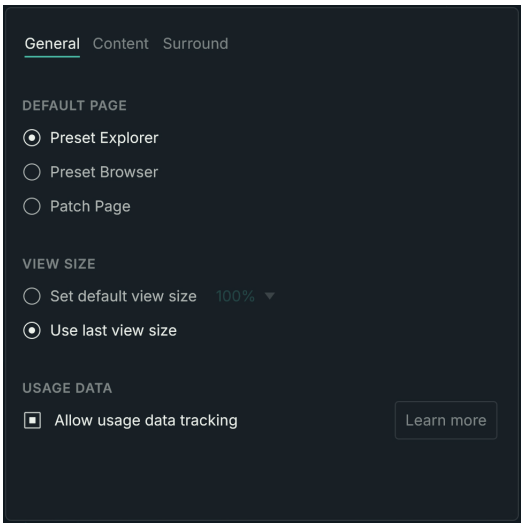


In the Settings dialog, the settings are organized into three tabs. You can show either tab by clicking its name at the top of the dialog:

- **General**: Lets you choose the default page, adjust the zoom setting, and activate the usage data tracking. Refer to General settings.
- **Content**: Lets you specify options affecting user presets. Refer to Content settings.
- **Surround**: Lets you adjust the surround configuration of Absynth. Refer to Surround settings.

General settings

The **General** tab of the Settings dialog contains the following settings:



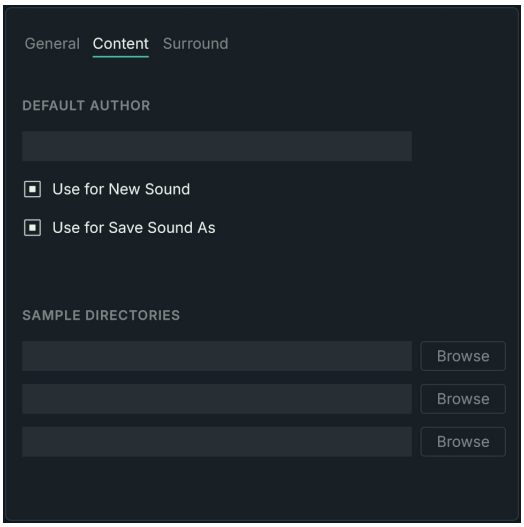
- **Default Page**: Chooses the page or view displayed first when you open Absynth. The following options are available: **Preset Explorer** selects the Preset Explorer view of the Browser, **Preset Browser** selects the the Preset List view of the Browser, and **Patch Page** selects the Patch page.
- **View Size**: Selects the default window size when you open Absynth. You can select **Set default view size** and choose a fixed size from the menu on the right, or select **Use last view size** to let Absynth reuse the same size as when you last closed it.

> ⓘ You can change the window size at any time using the dedicated entries in the Absynth menu.

- **Usage Data**: You can activate or deactivate the **Allow usage data tracking** check box. We encourage all users to keep Usage Data Tracking activated as it provides a valuable tool for understanding the performance and usage of Native Instruments applications, which in turn helps us improve our products. Clicking the **Learn more** button opens a link to a page with more information about usage data tracking on the Native Instruments website.

Content settings

The **Content** tab of the Settings dialog contains the following settings:



- **Default Author**: Specifies the default text that will be saved to the **Author** field of the Properties panel for new user presets, depending on how they were created:
  - **Use for New Sound**: If active, the default author will be applied when you select the **New Preset** entry from the Absynth menu.
  - **Use for Save Sound As**: If active, the default author will be applied when you select the **Save As…** entry from the Absynth menu or the Save As button from the Absynth Header.

  > You can manually edit the **Author** field in the Properties panel of the Preset List view in the Browser.

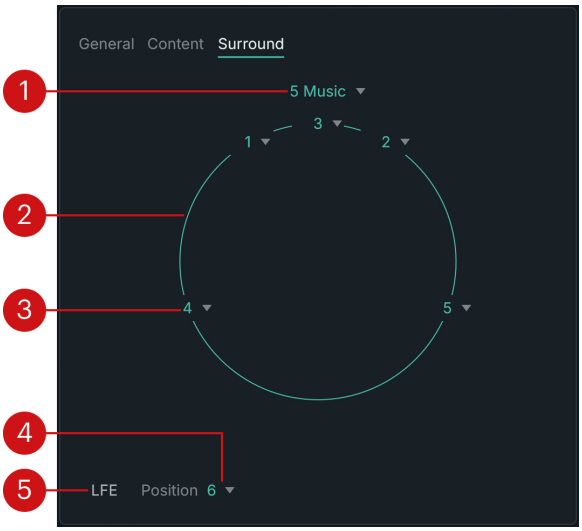
- **Sample Directories**: The three fields let you define the paths to the folders containing the samples used in your presets. To define or modify a path, click the **Browse** button on its right, navigate to the desired folder on your hard drive, and click **OK** to confirm.

Surround settings

Absynth 6 can have up to eight separate audio outputs. The **Surround** tab of the Settings dialog lets you assign the available output channels to specific positions in the panoramic field.

> If Absynth 6 is running as a plugin, the actual number of audio outputs depends on the track's settings in your DAW. If Absynth 6 is running as a standalone application, the number of audio outputs depends on how many output channels are activated in the Audio/MIDI Settings dialog.

The **Surround** tab contains the following elements:

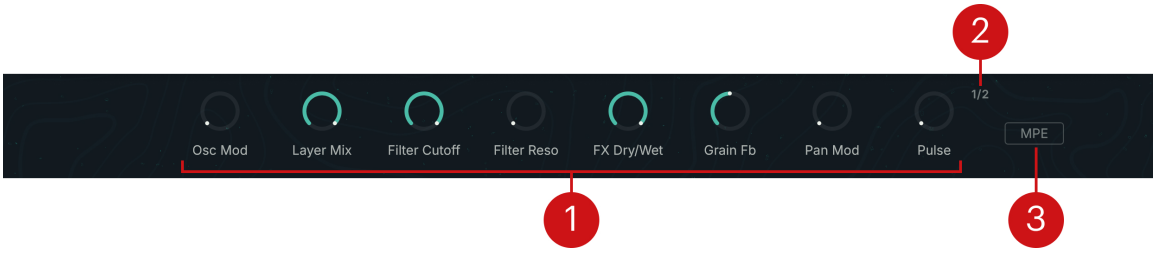


1. **Surround menu**: Shows the selected surround configuration. You can click the name and select another configuration from the menu. The name of each configuration indicates the number of included channels and a quick description of the channels' position.
2. **Field display**: Illustrates the spatial position of each channel on a circle centered on the listener.
3. **Channel menus**: The menus on the circle let you assign one output channel to each position in the current surround configuration.
4. **Position menu**: This menu is active only when **LFE** is on. It specifies the output channel of Absynth that should be used for the LFE signal (Low-Frequency Effects). When you select another configuration from the Surround menu, the next available channel is automatically selected in the **Position** menu.
5. **LFE button**: Activates or deactivates the additional LFE channel (Low-Frequency Effects). When on, you can assign an output channel to the LFE signal using the Position menu.

Macro controls

At the bottom of the Absynth window, the Macro controls are always visible and let you quickly adjust the values of the Macros defined in the current preset. Macros are customizable shortcuts to one or more parameters of the instrument. They are also available for host and MIDI automation. Each preset can contain up to 16 Macros.

The Macro controls contain the following elements:



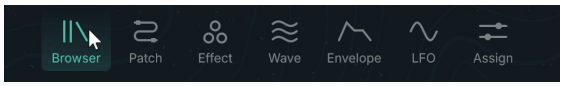
1. **Macros knobs**: The eight knobs adjust the Macros 1–8 or 9–16. The Macro names, behaviors, and target parameters vary with each preset. You can configure them in detail in the Assign page.

2. **Macro Page button**: Switches between the two pages of eight Macro knobs: When the button reads **1/2** the eight knobs adjust the Macros 1–8, when the button reads **2/2** the eight knobs adjust the Macros 9–16.
3. **MPE button**: Activates or deactivates MPE control globally for all Macros. If you want to use MPE (MIDI Polyphonic Expression) for a particular Macro, you need to activate this **MPE** button and to configure MPE support in the Macro assignment details of the Assign page.

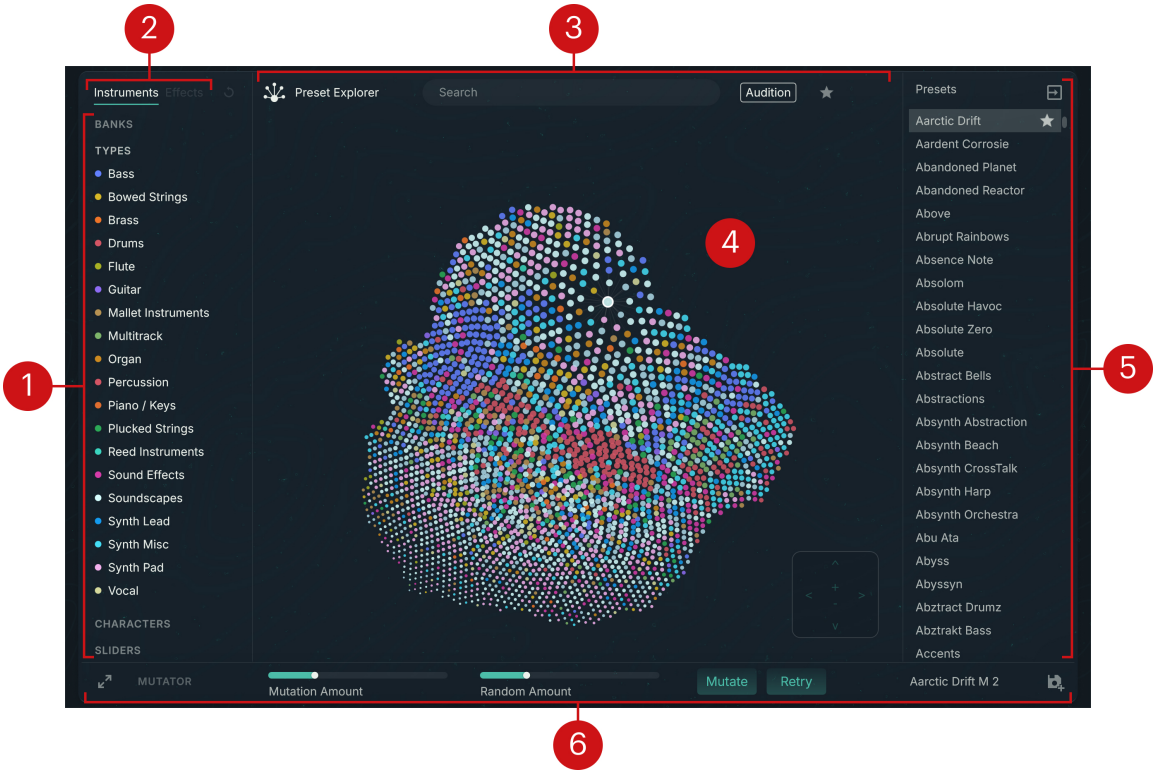
5. Browser page

The Browser page of Absynth 6 provides intuitive tools to quickly find presets according to their sonic characteristics. It also includes the Mutator, which proposes a totally different approach for generating new sounds from existing presets.

▶ To open the Browser page, click the **Browser** button at the top of the Absynth 6 window:



The Browser page contains the following areas and elements:



1. **Tag filter**: Lets you narrow down the search results according to descriptive attributes. Refer to Using the Tag filter.
2. **Instruments/Effects selector**: Switches between browsing the instrument presets and browsing the effect presets. You can browse the effect presets only from the Preset List view. In the Preset Explorer view the **Effects** label is grayed out and inactive.
3. **Browser global controls**: The elements at the top of the Browser page let you switch between the Preset List and Preset Explorer views of the Browser, search for a text string, activate audition, display only the Favorites, and include user presets in your search. Refer to Browser global controls.
4. **Search results**: The middle and biggest part of the Browser page shows all the presets matching your search criteria, as specified in the Tag filter on the left and in the global controls at the top. The Browser can display the search results in two different views:
   - The Preset Explorer view (depicted above) shows the presets as dots on a map. In this map the distance between dots indicates the sonic similarity between the presets.
   - The Preset List view shows the presets in a classic table layout with detailed information and sorting facilities.

5. **Right panel**: The content of the right panel depends on the Browser view: In Preset Explorer view the right panel shows the search results as a compact preset list, in Preset List view it shows the properties of the loaded preset.
6. **Mutator**: The Mutator lets you quickly generate variations of the current preset that go in a particular direction. Refer to Mutating your sound.

Browser global controls

The row at the top of the Browser page contains a set of global browsing controls:

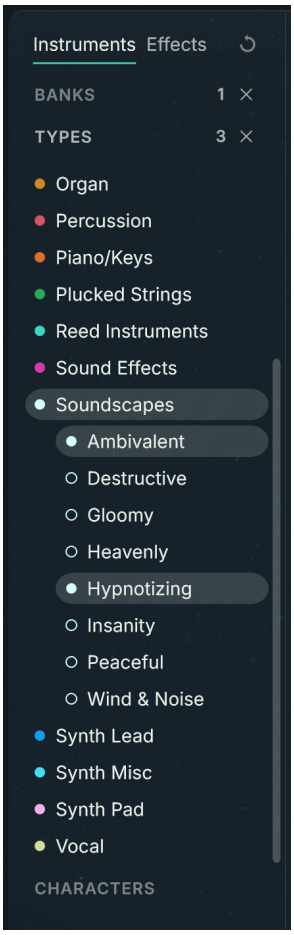
1. **Preset Explorer**: Switches the Browser between the Preset Explorer view (**Preset Explorer** button on) and the Preset List view (**Preset Explorer** button off).
2. **Search field**: Lets you type a text string to show only the presets containing that string in their name or author field.
3. **Audition**: When on, the Browser plays a glimpse of the presets' sound as you select them in the search results. This can help you choose between presets without having to load each of them. When **Audition** is on, you can click a preset to audition it, and double-click it to load it. When **Audition** is off, clicking the preset will load it directly.

   ⓘ Audition is available only for factory instrument presets.

4. **Favorites (star icon)**: Shows only your Favorites in the search results below. You can define any presets as Favorites by activating their star icon in the Preset List view or in the **Presets** pane of the Preset Explorer view. Favorites let you quickly create and recall a custom collection of your most beloved presets.
5. **User Content (user icon)**: Available only in the Preset List view. When on, the Browser shows only user presets. When off, the Browser shows both factory and user presets.

Using the Tag filter

On the left of the Browser page, the Tag filter lets you search for presets using descriptive attributes like their bank or NKS tags. NKS tags are used to characterize and classify presets based on various musical characteristics. Having the presets tagged in a meaningful way can encourage your sonic exploration and help you find the right presets quickly.



You can use the Tag filter as follows:

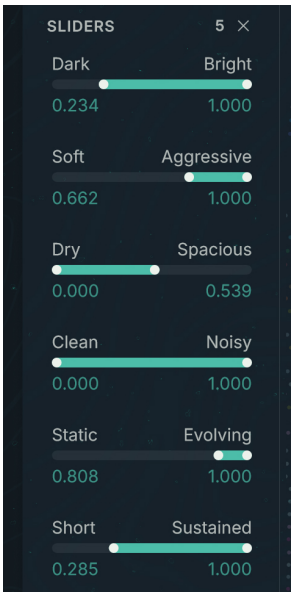
- The attributes are organized into the following categories: **Banks**, **Types**, and **Characters**. An additional **Sliders** category is available specifically for the Preset Explorer view of the Browser: The filters in this category work differently and are described in Using the Sliders category.
- Clicking a category header unfolds the category and shows its attributes. Only one category can be unfolded at a time.
- Each Type has a dedicated color. These colors appear on the dots in the Preset Explorer view to quickly recognize the preset types.
- Selecting a Bank, a Type, or a Character limits the search results to the presets with this attribute. You can select only one Bank and one Type at a time. If you select multiple Characters, the search results include all the presets having at least one of them.
- Selecting a Type unfolds its Subtypes. You can select one or more Subtypes to further refine your search. If you select multiple Subtypes, the search results include all the presets having at least one of them.
- The number of selected attributes (if any) appears next to each category header, indicating an active filtering in that category. You can click the little cross on the right to quickly deselect these attributes and reset the filtering for that category.

- In the top right corner, the **Reset button** (little cycling arrow) lets you globally reset the Tag filter by deselecting all attributes from all categories.

> Resetting the Tag filter does not affect the Search field. To display your entire collection of presets, make sure that you also delete any text string from the Search field.

Using the Sliders category

The Sliders category in the Tag filter is only available when the Browser is in Preset Explorer view. Instead of showing a list of fixed attributes, the Sliders category contains a set of sliders describing various sonic dimensions. Each dimension provides a continuous range from one sonic aspect to its opposite.

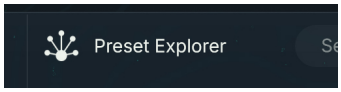


Each sliders provides two handles that define the minimum and maximum values for that particular sonic dimension. The colored bar between the handles corresponds to the range of allowed values: Only the presets within this range will appear in the search results. By default, the colored bars take up the full ranges and no presets are filtered. As you drag some of the handles with your mouse, you can progressively narrow down the search results regarding specific aspects of the sound, and get a unique list of presets corresponding to your very own collection of sonic criteria.

Preset Explorer view

The Preset Explorer view is the default view of the Browser. It provides a visual and intuitive way to explore your factory instrument presets by focusing onto their sonic aspects and similarities.

▶ To switch the Browser to Preset Explorer view, activate the Preset Explorer button at the top of the Browser page.



In the Preset Explorer view, your search results are represented as colored dots on a two-dimensional map:

- Each dot on the map represents a preset. The dot color indicates the preset Type, according to the color code of the Tag filter on the left. This lets you visualize the type of sound that you can expect from each preset.

  > The dots of the presets excluded from the search results are still visible but they are smaller, grayed out, and inactive.

- The distance between the dots illustrates the sonic similarity between the presets: The closer the dots, the closer their sounds.
- As you hover the mouse over a dot, the preset's name and Type/Character attributes appear in a tooltip box.
- You can move across the map by clicking anywhere in the map and dragging your mouse, and you can zoom in/out using the mouse wheel. You can also use the **Move controls** at the bottom right of the map: You can click the arrow buttons to move in the corresponding direction, and click the **+**/**-** buttons to zoom in or out.

- You can select another preset by clicking its dot. The selected dot is highlighted and the map is automatically centered on it. You can also select neighboring dots by pressing the left/right/up/down arrows on your computer keyboard: Each keystroke will select the next available dot in the direction pressed. You can use this to wander your preset libraries following your ears. On the way, you can quickly mark your discoveries as Favorites in the Presets panel on the right for later use.
- If the **Audition** button is on in the Browser global controls, selecting a dot auditions the preset, and double-clicking the selected dot (or pressing [Enter] on your computer keyboard) loads it.
  If **Audition** is off, selecting a dot directly loads the preset.

Presets panel

When the Browser is in Preset Explorer view, its right part contains the **Presets** panel. The **Presets** panel shows the same search results as the two-dimensional map, but arranged as a compact list:

You can hide or show the panel by clicking the arrow button in the top right corner.

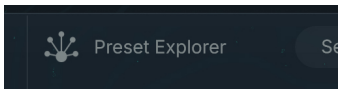
In the list, the presets are sorted alphabetically. The loaded preset is highlighted. If the **Audition** button is on in the Browser global controls above, you can select a preset to audition it or double-click the preset to load it. The last auditioned preset is slightly highlighted and shows a small speaker icon. If **Audition** is off, clicking a preset directly loads it.

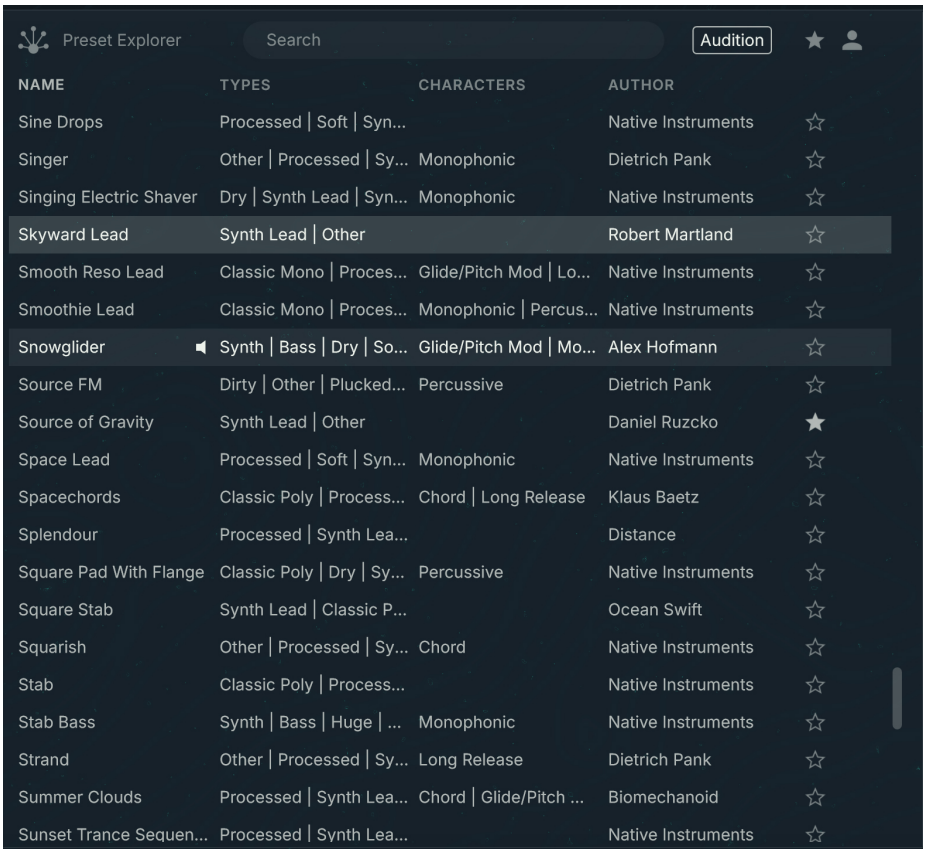
As you hover over presets from the list with your mouse, star icons appear next to each preset and let you manage your Favorites. You can click the star icons to add or remove their presets to/from your Favorites. The star icons are lit for the presets set as Favorites. You can then display only your Favorites by activating the star icon at the top in the Browser global controls.

Preset List view

The Preset List view of the Browser shows your search results in a table layout.

▶ To switch the Browser to Preset List view, deactivate the **Preset Explorer** button at the top of the Browser page.





In the Preset List view, each preset matching your search criteria appears as a row in the table. The table provides the following columns, from left to right:

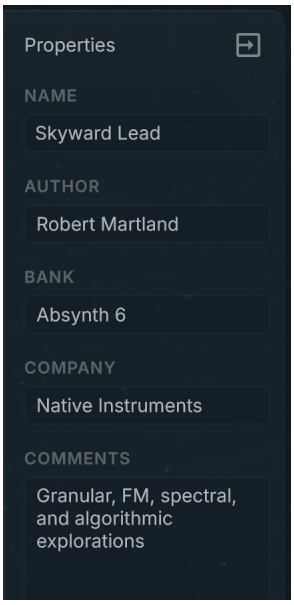
- **Name**: Shows the preset name. You can click the **NAME** column header to sort the presets by name.
- **Types**: Shows the Type and Subtype tags of the preset.
- **Characters**: Shows the Character tags of the preset.
- **Author**: Shows the preset author, as defined in the preset properties. You can click the **AUTHOR** column header to sort the presets by author.
- **Favorites**: The star icons in the rightmost column let you manage your Favorites. You can click the star icons to add or remove their presets to/from your Favorites. The star icons are lit for the presets set as Favorites. You can then display only your Favorites by activating the star icon at the top in the Browser global controls.

In the list, the loaded preset is highlighted. You can select another preset by clicking its row. You can also select the previous or next preset by pressing the up/down arrows on your computer keyboard.

If you are browsing factory instrument presets, and if the **Audition** button is on in the Browser global controls above, you can select a preset to audition it, and double-click a preset to load it. You can also load the selected preset by pressing [Enter] on your computer keyboard. The last auditioned preset is slightly highlighted and shows a small speaker icon. If **Audition** is off, clicking a preset directly loads it.

Properties panel

When the Browser is in Preset List view, its right part contains the **Properties** panel, which shows the properties (name, author, bank, company, and comments) of the loaded preset:



You can hide or show the panel by clicking the arrow button in the top right corner.

In the **Properties** panel, you can edit the **AUTHOR**, **BANK**, **COMPANY**, and **COMMENTS** properties by clicking inside either field (the active field is outlined) and entering the new text from your computer keyboard.

As with any parameter adjustment, your edits in the **Properties** panel are not stored automatically. To keep your changes, you need to save the preset, for example using the Save / Save As button in the Header.

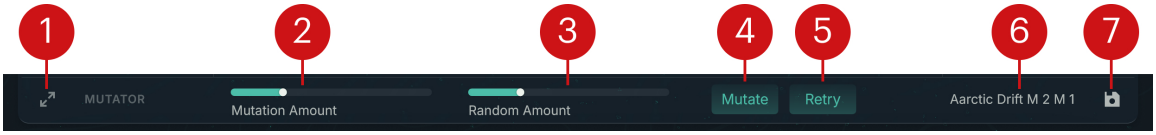
> Your edits in the **Properties** panel cannot be reverted using the Undo/Redo commands.

Mutating your sound

At the bottom of the Browser page, the Mutator lets you "mutate" the current preset to color its sound with other sonic characteristics. You simply need to describe these sonic characteristics in the Browser using the Tag filter, and the Mutator will take the search results as "direction" for the mutations. You can set the extent of the mutation (how far it is from the original preset) and add randomization to it. The result is a completely new sound based on the original preset, enhanced with your musical descriptions, and created with a single click. You can retrigger mutations as many times as you want, recursively mutate your preset, go back to a previous mutation, modify your search in the Browser to try another direction, and save any of the results as new user presets.

Mutator in compact view

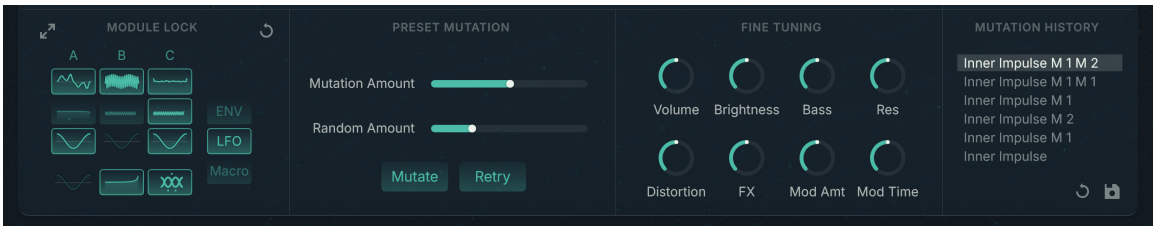
By default the Mutator is displayed in compact view. The compact view makes more room for the Browser's search results above, while providing the main mutation controls:



1. **View button (double arrow icon)**: Switches the Mutator to the extended view, which lets you configure the mutation process more in detail.
2. **Mutation Amount**: Adjusts the extent of the mutation. In other words it is the distance between the current sound and the next mutation. The more you drag the slider to the right, the farther the mutated preset will be from the original and the closer it will be to the presets in the search results.
3. **Random Amount**: Adjusts the amount of randomization applied to the mutated parameters. The more you drag the slider to the right, the more the parameters of the mutated preset will deviate from the road between the original preset and the presets in the search results.
4. **Mutate**: Triggers the mutation. Once the mutation is computed it is automatically loaded and it replaces the original preset (which can be itself a previous mutation). The mutation takes the name of the original preset and appends a "M" (for "mutation") and the rank number of the mutation: **M1** for the first mutation, **M2** for the second one, and so on.
5. **Retry**: Cancels the last mutation and triggers a new mutation starting from the previous one. The new mutation takes the name of the canceled one. If you have not triggered any mutation yet, the **Retry** button is grayed out and inactive.
6. **Mutation History menu**: Shows the name of the selected mutation. Clicking the mutation name opens a menu with all the mutations from the Mutation History. You can select any mutation from the menu to load it in the instrument.
7. **Save Mutation (floppy disk icon)**: Saves the selected mutation as a user preset. The saving procedure is the same as with the **Save As** command from the Absynth menu.

Mutator in extended view

In extended view the Mutator provides the set of controls of the compact view, along with additional controls for selecting the parts of the instrument that should be mutated and for adjusting further properties of the mutated sound.

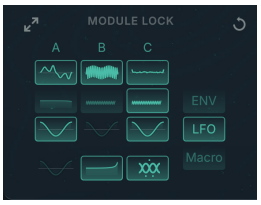


In the top left corner of the Mutator, clicking the View button (double arrow icon) switches back to compact view.

The extended view contains four sections, from left to right: Module Lock, Preset Mutation, Fine Tuning, and Mutation History.

Module Lock

The Module Lock section contains a small representation of your patch and its modules as configured in the Patch page. In this mini patch view you can select which components should be mutated and which should be locked. By locking individual components (or entire channels), you make sure that the mutation will retain their distinctive sonic feature.



The icons on the module buttons illustrate the module contents (for example, a particular waveform or effect) to easily distinguish them.

The module buttons can have three states:

- Fully lit: The module is active in the Patch page and included in the mutation. You can click the button to exclude the module from the mutations (the button turns dimmed).
- Dim lit: The module is active in the Patch page but excluded from the mutation. You can click the button to include the module in the mutations (the button turns fully lit).
- Grayed out: The module is deactivated in the Patch page and cannot be included in any mutation.

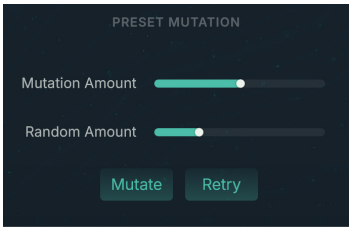
On the right, you can click the **ENV** (Envelope), **LFO**, and **Macro** buttons to include or exclude the envelopes, the LFOs, and the Macro assignments in/from the mutations, respectively. Included components are fully lit, excluded components are dim lit.

The lock states of the individual modules and components are saved with each preset. In the top right corner of the **Module Lock** section, the **Reset button** (little cycling arrow) lets you globally reset the lock states to their original values.

> Selectively including or excluding components in the **Module Lock** section can be a great help when mutating towards particular attributes. For example, if you have a bass sound and want it to be more percussive, make sure to activate the Envelope button, and maybe the Effect module. Or if you want an FM-like sound, activate the Oscillator modules and nothing else.

Preset Mutation

The Preset Mutation section provides the basic mutation controls:



- **Mutation Amount**: Adjusts the extent of the mutation. In other words it is the distance between the current sound and the next mutation. The more you drag the slider to the right, the farther the mutated preset will be from the original and the closer it will be to the presets in the search results.
- **Random Amount**: Adjusts the amount of randomization applied to the mutated parameters. The more you drag the slider to the right, the more the parameters of the mutated preset will deviate from the road between the original preset and the presets in the search results.

- **Mutate**: Triggers the mutation. Once the mutation is computed it is automatically loaded and it replaces the original preset (which can be itself a previous mutation). The mutation takes the name of the original preset and appends a "M" (for "mutation") and the rank number of the mutation: **M1** for the first mutation, **M2** for the second one, and so on.
- **Retry**: Cancels the last mutation and triggers a new mutation starting from the previous one. The new mutation takes the name of the canceled one. If you have not triggered any mutation yet, the **Retry** button is grayed out and inactive.

Fine Tuning

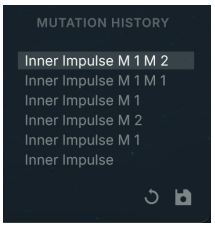
The Fine Tuning section provides quick access to eight characteristics of the sound. The eight controls let you quickly tweak the sound after a mutation, or simply after loading a preset.

Only the parameters that affect the current patch are active. The others are dimmed.

> The controls in the Fine Tuning section are intended to make on-the-fly adjustments to the sound when browsing or mutating presets. They are not meant for live performance and they can be neither automated nor modulated nor controlled via MIDI.

Mutation History

The **Mutation History** section shows the list of all the mutations created since you started Absynth, in chronological order (the oldest at the bottom).



You can click any entry from the list to load that mutation, replacing the current preset or mutation. This lets you quickly compare the sound of different mutations. If an older mutation is selected, clicking **Mutate** starts a new branch of mutations, which will be added as well to the Mutation History.

In the bottom right corner, two buttons are available:

- **Reset (little cycling arrow)**: Clears the Mutation History, removing all the mutations from the list.

> Resetting the Mutation History cannot be undone!

- **Save Mutation (floppy disk icon)**: Saves the selected mutation as a user preset. The saving procedure is the same as with the **Save As** command from the Absynth menu.

Mutation tips

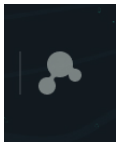
Here are a few more tips when using the Mutator:

- Setting the **Mutation Amount** slider at a higher value will actually generate more predictable results. This is because as the mutation amount increases, the Mutator algorithm is more likely to load a whole channel from a patch, which means that the envelope, filter, and oscillator will have been crafted for each other.
- The **Retry** button and the **Fine Tuning** controls are especially useful when you want to stick with results only in the category that you have selected in the Browser.
- If you deactivate the **Macro** button in the **Module Lock** section, your Macro assignments will likely stay mapped during mutations.

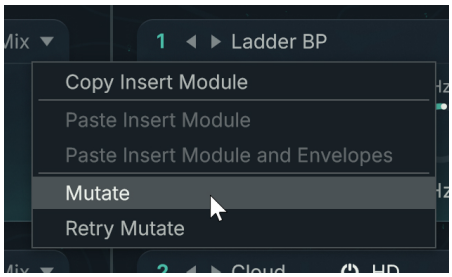
Mutation controls in other parts of the instrument

You can find mutation controls in other parts of Absynth 6, making mutations available even if another page is open:

- A Mutate button is available in the Absynth Header:



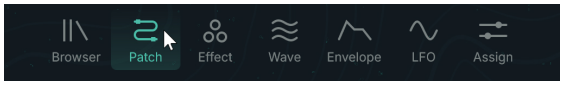
- A **Mutate** and a **Retry Mutate** commands are available in the Edit menu of each module in the Patch page so that you can selectively mutate single modules while working on your patch:



6. Patch page

The Patch page of Absynth 6 lets you assemble the components that will generate the preset's sound.

▶ To open the Patch page, click the **Patch** button at the top of the Absynth 6 window:



The **modules** represent the individual components in the Patch page. There are different types of modules: For example, the Oscillator modules are sound sources, while the Modulator, Filter, and Waveshaper modules process existing sounds. The semi-modular design of Absynth 6 allows you to choose the arrangement of some of the modules, while other modules have a fixed place in the signal flow.

Patch basics

The patch is structured into four channels: The signals start on three distinct Oscillator channels, then they are combined on the Main channel.

Oscillator channels

Absynth 6 provides three so-called Oscillator channels labeled **Channel A**, **Channel B**, and **Channel C**, and laid out vertically in the top part of the Patch page:

Each channel can contain up to three modules. The signal flows from top to bottom:

- The first module at the top is always a **source module** (sometimes globally referred to as **Oscillator**). Labeled **A**, **B**, and **C**, the source modules are the only sound sources in Absynth 6 and provide the sonic foundation for every preset. If you deactivate a source module, its entire channel turns off. Three categories of source modules are available: Oscillators, Samplers, and the Audio Input.
- The two modules below are **insert modules**. Labeled **1** and **2**, they process the sound coming from the previous module. Three categories of insert modules are available: Filters, Modulators, and the Waveshaper.

Elsewhere in the Absynth interface, the modules of the Oscillator channels are referred to using their **channel letter** and their **position** within the channel. For example:

- **Oscil A** would refer to the source module on channel A,
- **Filter B1** would refer to the Filter used as first insert module on channel B,
- **WS C2** would refer to the Waveshaper used as second insert module on channel C.

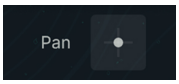
Channel levels

At the bottom of the Oscillator channels, the three **Volume** sliders let you adjust the level of the respective channels:

Surround panning the channels

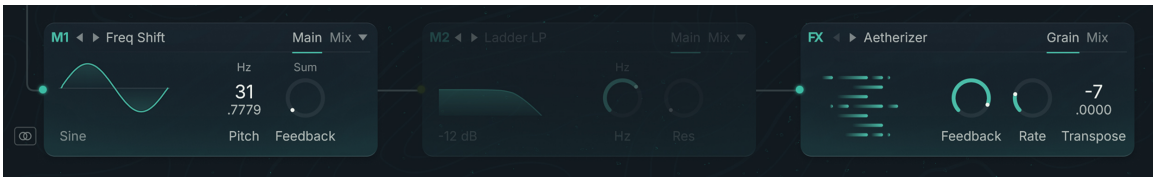
Located in the lower left corner of the Patch page, the Surround Pan switch lets you to activate or deactivate the Surround Pan mode.

In Surround Pan mode you can freely position the sound of each Oscillator channel in the surround field using the **Pan** X-Y control right of the **Volume** slider, at the bottom of the channel:



You can use surround panning to create impressive spatial effects, for example, by modulating the surround position of the channels independently of each other with an LFO or an envelope.

Main channel



The signals from the three Oscillator channels are mixed into the Main channel, which is arranged horizontally at the bottom of the Patch page. In the Main channel, the signal flows from left to right through three modules:

- The first two modules on the left are **insert modules**. Labeled **M1** and **M2** (for "Main 1" and "Main 2"), they process the sound coming from the three Oscillator channels above. You can use the same insert modules as in the Oscillator channels: Filters, Modulators, and the Waveshaper.
- The last module on the right is an **effect module**. Labeled **FX**, it is at the end of the module chain and its output signal leaves Absynth 6. You can choose from a set of advanced effects and adjust their most important parameters directly from the module. The detailed configuration of the effects takes place in the dedicated Effect page.

Monophonic and Polyphonic modes

The insert modules of the Main channel have two operating modes:

- In Monophonic mode, the signals from the three Oscillator channels are mixed beforehand and the module processes the mix.
- In Polyphonic mode, the module processes the signal from each Oscillator channel separately.

You can switch an insert module between both modes from the module's Edit menu.

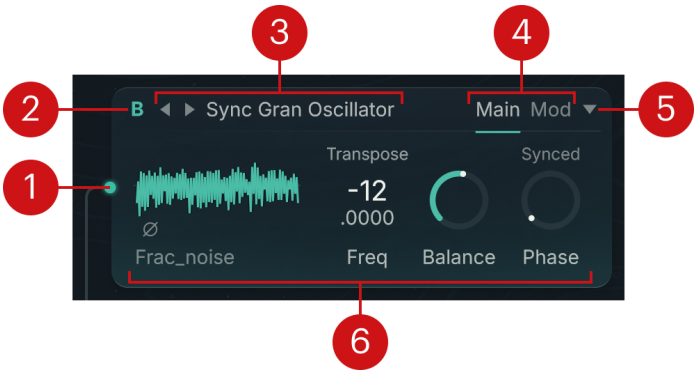
If the first insert module is set to Monophonic mode, the second insert module cannot operate in Polyphonic mode, since the first module has already summed the three Oscillator signals beforehand.

The difference between the two modes can be heard particularly clearly in the Waveshaper module: In Polyphonic mode every voice has its own, independent Waveshaper. The distortion affects every voice separately. In Monophonic mode, a single Waveshaper processes the different voices, which means that many of the played notes will interact. You can try out the effect by inserting a Waveshaper module into the Main channel. Play a couple of sounds while switching between both modes: You will notice that the Monophonic mode reacts with a significantly stronger distortion as you begin to play multiple notes. This is because the signals of the different voices are assembled before the Waveshaper input, which results in a higher input gauge. In Polyphonic mode, by contrast, the voices are distributed among multiple Waveshapers and produce lower signal levels.

For low level input signals, the Waveshaper in Monophonic mode works like a compressor and lends itself well to compressing and warming up the input signal.

Working with modules

The various modules available in the Patch page share a common layout and a set of generic features and workflows:



1. **Connector dot**: Clicking the dot activates or deactivates the module. The audio signal path is depicted by wires connecting the dots of the active modules. Inactive modules are grayed out and their dots are bypassed. On channel A–C, deactivating the source module deactivates the whole channel, and on an inactive channel, activating any of its module reactivates the entire channel.

   When a module is active, you can click anywhere on its left border to deactivate it. When a module is inactive, you can click anywhere in the module to activate it.

2. **Module label**: Identifies the module. The letter **A**, **B**, or **C** indicates the source module of the corresponding Oscillator channel, **1** and **2** indicate the first and second insert modules of an Oscillator channel, **M1** and **M2** indicate the first and second insert modules of the Main channel, and **FX** indicates the effect module at the end of the module chain.

3. **Type selector**: Shows the module type. Clicking the type name opens a menu from which you can switch the module to another type. You can also click the left/right arrows to switch to the previous or next type. The entries available in the Type menu vary with the role of the module (source, insert, or effect module).

4. **Panel tabs**: Clicking either tab shows the corresponding panel of parameters below.
   The available panels depend on the type of module (Oscillator, Filter, Modulator or Waveshaper) as well as on the specific mode for each module type. You will find a complete list of available panels in the module-specific sections.

5. **Edit menu (downward arrow)**: Provides editing functions like copy/paste commands and Mutation shortcuts. The exact menu entries vary with the module's role (source or insert) and channel (Oscillator channel or Main channel).

   The effect module has no Edit menu.

6. **Parameter panel**: A module can have up to three panels organizing the module parameters. Each panel can be called by clicking the corresponding tab at the top right. In most modules the left part of the **Main** panel shows a graphic display that visually illustrates the type of module, its function, or the selected waveform.

Waveform selector

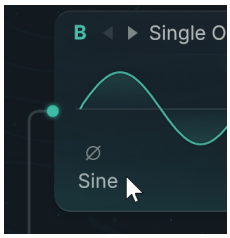
Waveforms are used in many parts of Absynth. Each time, the Waveform selector lets you choose the desired waveform.

In the Patch page, the Waveform selector is available in the following modules:

- Source modules: all the Oscillators.
- Insert modules: the Modulators (Freq Shift and Ringmod), the Waveshaper, and the Filters providing a feedback loop (LPF Feedback, Allpass Feedback, and Supercomb).

> The Waveform selector is also used in the LFO page, in the LFO controls of the Envelope page, as well as in the Transform panel for the **FM** and **Mix** transformations available in the Waveform view of the Wave page.

▶ To open the Waveform selector in a module, click the waveform name under the mini waveform display in the module parameters:



The Waveform selector contains a list with all the available waveforms:

In the list, the names between brackets indicate user-created waveforms.

To load a waveform:

1. To load a waveform, click its entry in the list.

→ The new waveform becomes immediately active and you can hear its effect on the sound when you press a key on your MIDI keyboard. The Waveform selector stays open so that you can test different waveforms.

2. When you are done, click anywhere outside the Waveform selector to close it.

At the top of the Waveform selector, you can also use the following buttons:

- **Simple**: Shows the list of simple waveforms. In this category, next to the standard forms like **Sine**, **Triangle**, **Saw**, or**Square**, you also find instrumental and atonal waveforms.
- **Morph**: Shows the list of morph waveforms. A morph waveform contains in fact two waveforms, which you can seamlessly blend ("morph") with one another.
- **New**: Creates a new user waveform based on the simple or morph waveform currently selected. Absynth automatically switches to the Wave page and selects this new waveform so that you can further edit it.
- **Edit**: This button is available only if a user waveform is selected in the list, which is indicated by the brackets around the name. Clicking **Edit** switches to the Wave page and selects this user waveform so that you can modify it.

Source modules

The source modules can be loaded in the first slot at the top of the channels **A**, **B**, and **C**. The source modules are the only sound sources in Absynth. The other modules will modify the sounds produced by the source modules.

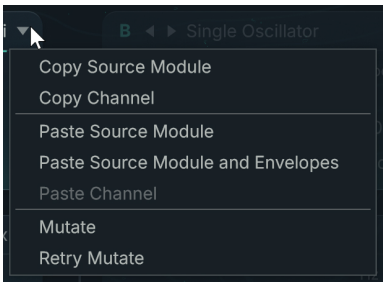
The source modules can use different types of synthesis, which are grouped into three categories:

- **Oscillators** are based on monocyclic waveforms played repeatedly to create a musical tone. The following types of Oscillators are available: **Single Oscillator**, **Double Oscillator**, **FM Oscillator**, **Ringmod Oscillator**, **Fractalize Oscillator**, and **Sync Granular Oscillator**.
- **Samplers** can play an audio recording in various ways. The following types of Samplers are available: **Sample Engine** and **Granular Engine**.
- The **Audio Input** module does not generate any audio by itself, instead it receives external audio and pass it on to the next modules.

You can choose the type of source module from the Type selector at the top of the first slots:

Edit menu for source modules

The Edit menu of a source module contains the following entries:

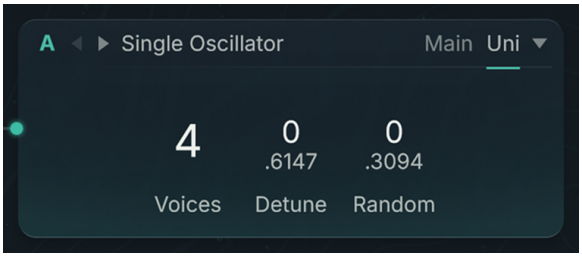


- **Copy Source Module**: Copies the settings of the module to the clipboard.
- **Copy Channel**: Copies the settings of all the active modules in that channel to the clipboard.
- **Paste Source Module**: Pastes into the source module the settings stored in the clipboard (using the command **Copy Source Module**).
- **Paste Source Module and Envelopes**: Pastes into the source module the settings stored in the clipboard (using the command **Copy Source Module**). Any envelopes linked to the copied source module are also inserted.
- **Paste Channel**: Pastes into the current channel the settings stored in the clipboard.
- **Mutate**: Mutates this particular module.
- **Retry Mutate**: Retries a mutation of the module. This entry is available only if you have already triggered a mutation in this session.

The **Mutate** and **Retry Mutate** commands use the mutation settings defined in the Mutator section of the Browser page.

Uni panel (Unisono)

The **Uni** panel (Unisono) is available for all the Oscillators except the Sync Granular Oscillator. The panel lets you quickly "stack" voices to produce fuller, stronger sounds.



The **Uni** panel contains the following parameters:

- **Voices**: Sets the number of voices produced by each note played. The highest value is 8.
- **Detune**: Adjusts the amount of detuning (in semitones) between the original voice and the additional voices. This parameter is available only if the **Voices** value is larger than 1, in other words, if there are some additional voices to detune. Even-numbered additional voices are transposed downwards, odd-numbered voices upwards.
- **Random**: Adjusts the amount of additional, random detuning (upwards and downwards) in semitones. The random detuning of the voices occurs with every note in the sequence. Subtly put to work, with a low **Voices** value, **Random** can produce the effect of a sloppy or imprecise intonation (for example, in order to imitate a string instrument without tension or to give atonal percussion sounds some natural variation). You can get some unpredictable and exciting results by playing around with different selections of notes.

Frequency menu and control

All the source modules except the Audio Input can be set to react to incoming notes in various ways. These ways are known as **Frequency modes**. The Frequency menu and the Frequency control (**Freq**) of the **Main** panel let you choose from the following Frequency modes:

- **Transpose**: The source module follows the pitch of the played note. The Frequency control (**Freq**) determines the transposition in semitones. For example, setting the Frequency control to 1.5 will transpose the played notes by one and a half tone. The Frequency control has a resolution of 1/1000 of a semitone.
- **Ratio**: Transposes the source module according to a frequency ratio (for example along the harmonic sequence: **2** corresponds with a transposition of 12 semitones, **3** with a transposition of 19 semitones, etc.). This mode is especially useful when using frequency modulation and ring modulation.
- **Hz**: Fixed pitch in Hertz. In this mode, also useful for frequency and ring modulation, the source module ignores the pitch of the played notes. The pitch of the source module does not react to information like pitch bend nor to signals coming from an LFO, but it can be controlled via an envelope.
- **Note**: Fixed pitch expressed as a MIDI note number. For example: 60.5 = C3 (middle C) plus a quarter-tone. This mode is similar to the Hz mode, but is more appropriate when you want to use the source module at a note-bound frequency.

> ⓘ You can switch between the **Hz** and **Note** modes to display the exact frequency of particular notes.

Single Oscillator

The Single Oscillator is the simplest source module. It includes only one oscillator.

The **Main** panel contains the parameters for the main oscillator:

- **Mini Waveform**: Shows the shape and the name of the loaded waveform. If a Morph waveform is loaded, you can click the shape and drag your mouse vertically to morph the waveform. Clicking the waveform name opens the Waveform selector and lets you choose another waveform.
- **Phase Invert button (phase symbol)**: Above the waveform name, the Phase Invert button lets you reverse the phase of the signal generated by the oscillator.
- **Frequency menu and control (Freq)**: These controls let you to define the oscillator's frequency as a transposition of the played note (**Transpose**), as a relationship with the played note (**Ratio**), as a fixed frequency (**Hz**), or as a fixed MIDI note number (**Note**). You can find more information in Frequency menu and control.
- **Phase Sync switch**: Setting the switch to **Synced** will reset the phase of the oscillator with every incoming MIDI note. When the Phase Sync switch is set to Fr **Free**, the oscillator will not be reset. When only one oscillator is active and you have set a single voice in the **Uni** panel, the mode **Free** has almost no effect. When you have a higher Unisono number of voices or multiple active oscillators, every note leads to an audible change in the sound.
- **Phase**: Sets the phase of the oscillator. This control is available only when the Phase Sync switch is set to **Synced**. The effect of the **Phase** control is only audible if a second oscillator is active. This parameter is useful when using integer values for frequency ratios between carrier and modulator, as it is the case in FM.

The **Uni** panel is common to most Oscillators and is described in Uni panel (Unisono).

Double Oscillator

The Double Oscillator contains two oscillators: the Main oscillator and the Mod oscillator (for "modulation"). The signals of the two oscillators are mixed.

The **Main** panel contains the same parameters as the Single Oscillator with an additional **Balance** control:

- **Mini Waveform**: Shows the shape and the name of the loaded waveform. If a Morph waveform is loaded, you can click the shape and drag your mouse vertically to morph the waveform. Clicking the waveform name opens the Waveform selector and lets you choose another waveform.
- **Phase Invert button (phase symbol)**: Above the waveform name, the Phase Invert button lets you reverse the phase of the signal generated by the oscillator.
- **Frequency menu and control (Freq)**: These controls let you to define the oscillator's frequency as a transposition of the played note (**Transpose**), as a relationship with the played note (**Ratio**), as a fixed frequency (**Hz**), or as a fixed MIDI note number (**Note**). You can find more information in Frequency menu and control.
- **Balance control**: Adjusts the balance between the Main and Mod oscillators in the output signal.
- **Phase Sync switch**: Setting the switch to **Synced** will reset the phase of the oscillator with every incoming MIDI note. When the Phase Sync switch is set to Fr **Free**, the oscillator will not be reset. When only one oscillator is active and you have set a single voice in the **Uni** panel, the mode **Free** has almost no effect. When you have a higher Unisono number of voices or multiple active oscillators, every note leads to an audible change in the sound.
- **Phase**: Sets the phase of the oscillator. This control is available only when the Phase Sync switch is set to **Synced**. The effect of the **Phase** control is only audible if a second oscillator is active. This parameter is useful when using integer values for frequency ratios between carrier and modulator, as it is the case in FM.

The **Mod** panel contains the parameters of the modulating oscillator:

- **Mini Waveform**: Shows the shape and the name of the loaded waveform. If a Morph waveform is loaded, you can click the shape and drag your mouse vertically to morph the waveform. Clicking the waveform name opens the Waveform selector and lets you choose another waveform.
- **Frequency menu and control (Freq)**: These controls let you to define the oscillator's frequency as a transposition of the played note (**Transpose**), as a relationship with the played note (**Ratio**), as a fixed frequency (**Hz**), or as a fixed MIDI note number (**Note**). You can find more information in Frequency menu and control.
- **Phase**: Sets the phase of the oscillator. This control is available only when the Phase Sync switch is set to **Synced**. The effect of the **Phase** control is only audible if a second oscillator is active. This parameter is useful when using integer values for frequency ratios between carrier and modulator, as it is the case in FM.

The **Uni** panel is common to most Oscillators and is described in Uni panel (Unisono).

A Double Oscillator uses less CPU resources than two Single Oscillators.

FM Oscillator

Frequency modulation was discovered in the late 1960's by John Chowning and achieved great popularity in the 1980's through Yamaha's DX7 synthesizer. Much has already been written about FM synthesis so we will not go into detail here.

In the FM Oscillator, the Mod oscillator modulates the frequency of the Main oscillator.

The **Main** panel contains the same parameters as the Single Oscillator with an additional **FM Index** control:

- **Mini Waveform**: Shows the shape and the name of the loaded waveform. If a Morph waveform is loaded, you can click the shape and drag your mouse vertically to morph the waveform. Clicking the waveform name opens the Waveform selector and lets you choose another waveform.
- **Phase Invert button (phase symbol)**: Above the waveform name, the Phase Invert button lets you reverse the phase of the signal generated by the oscillator.
- **Frequency menu and control (Freq)**: These controls let you to define the oscillator's frequency as a transposition of the played note (**Transpose**), as a relationship with the played note (**Ratio**), as a fixed frequency (**Hz**), or as a fixed MIDI note number (**Note**). You can find more information in Frequency menu and control.
- **FM Index**: Determines the depth of the frequency modulation.
- **Phase Sync switch**: Setting the switch to **Synced** will reset the phase of the oscillator with every incoming MIDI note. When the Phase Sync switch is set to Fr **Free**, the oscillator will not be reset. When only one oscillator is active and you have set a single voice in the **Uni** panel, the mode **Free** has almost no effect. When you have a higher Unisono number of voices or multiple active oscillators, every note leads to an audible change in the sound.
- **Phase**: Sets the phase of the oscillator. This control is available only when the Phase Sync switch is set to **Synced**. The effect of the **Phase** control is only audible if a second oscillator is active. This parameter is useful when using integer values for frequency ratios between carrier and modulator, as it is the case in FM.

The **Mod** panel contains the parameters of the modulating oscillator:

- **Mini Waveform**: Shows the shape and the name of the loaded waveform. If a Morph waveform is loaded, you can click the shape and drag your mouse vertically to morph the waveform. Clicking the waveform name opens the Waveform selector and lets you choose another waveform.
- **Frequency menu and control (Freq)**: These controls let you to define the oscillator's frequency as a transposition of the played note (**Transpose**), as a relationship with the played note (**Ratio**), as a fixed frequency (**Hz**), or as a fixed MIDI note number (**Note**). You can find more information in Frequency menu and control.
- **Phase**: Sets the phase of the oscillator. This control is available only when the Phase Sync switch is set to **Synced**. The effect of the **Phase** control is only audible if a second oscillator is active. This parameter is useful when using integer values for frequency ratios between carrier and modulator, as it is the case in FM.

The **Uni** panel is common to most Oscillators and is described in Uni panel (Unisono).

Ringmod Oscillator

In the Ringmod Oscillator, the signals from the Main and Mod oscillators are multiplied with one another.

The **Main** panel contains the same parameters as the Single Oscillator with an additional **Balance** control:

- **Mini Waveform**: Shows the shape and the name of the loaded waveform. If a Morph waveform is loaded, you can click the shape and drag your mouse vertically to morph the waveform. Clicking the waveform name opens the Waveform selector and lets you choose another waveform.
- **Phase Invert button (phase symbol)**: Above the waveform name, the Phase Invert button lets you reverse the phase of the signal generated by the oscillator.
- **Frequency menu and control (Freq)**: These controls let you to define the oscillator's frequency as a transposition of the played note (**Transpose**), as a relationship with the played note (**Ratio**), as a fixed frequency (**Hz**), or as a fixed MIDI note number (**Note**). You can find more information in Frequency menu and control.
- **Balance control**: Adjusts the balance between the Main and Mod oscillators in the output signal.
- **Phase Sync switch**: Setting the switch to **Synced** will reset the phase of the oscillator with every incoming MIDI note. When the Phase Sync switch is set to Fr **Free**, the oscillator will not be reset. When only one oscillator is active and you have set a single voice in the **Uni** panel, the mode **Free** has almost no effect. When you have a higher Unisono number of voices or multiple active oscillators, every note leads to an audible change in the sound.
- **Phase**: Sets the phase of the oscillator. This control is available only when the Phase Sync switch is set to **Synced**. The effect of the **Phase** control is only audible if a second oscillator is active. This parameter is useful when using integer values for frequency ratios between carrier and modulator, as it is the case in FM.

The **Mod** panel contains the parameters of the modulating oscillator:

- **Mini Waveform**: Shows the shape and the name of the loaded waveform. If a Morph waveform is loaded, you can click the shape and drag your mouse vertically to morph the waveform. Clicking the waveform name opens the Waveform selector and lets you choose another waveform.
- **Frequency menu and control (Freq)**: These controls let you to define the oscillator's frequency as a transposition of the played note (**Transpose**), as a relationship with the played note (**Ratio**), as a fixed frequency (**Hz**), or as a fixed MIDI note number (**Note**). You can find more information in Frequency menu and control.
- **Phase**: Sets the phase of the oscillator. This control is available only when the Phase Sync switch is set to **Synced**. The effect of the **Phase** control is only audible if a second oscillator is active. This parameter is useful when using integer values for frequency ratios between carrier and modulator, as it is the case in FM.

The **Uni** panel is common to most Oscillators and is described in Uni panel (Unisono).

Fractalize Oscillator

Experienced Absynth users will recognize the Fractalize Oscillator as a real time version of the **Fractalize** function available in the **Transform** menu of the Wave page. In the Fractalize Oscillator, the selected waveform is copied to itself so that smaller elements of the waveform become similar to the image of the whole. You can use the Fractalize Oscillator to add overtones to a plain waveform.

The **Main** panel contains the same parameters as the Single Oscillator:

- **Mini Waveform**: Shows the shape and the name of the loaded waveform. If a Morph waveform is loaded, you can click the shape and drag your mouse vertically to morph the waveform. Clicking the waveform name opens the Waveform selector and lets you choose another waveform.

- **Phase Invert button (phase symbol)**: Above the waveform name, the Phase Invert button lets you reverse the phase of the signal generated by the oscillator.
- **Frequency menu and control (Freq)**: These controls let you to define the oscillator's frequency as a transposition of the played note (**Transpose**), as a relationship with the played note (**Ratio**), as a fixed frequency (**Hz**), or as a fixed MIDI note number (**Note**). You can find more information in Frequency menu and control.
- **Phase Sync switch**: Setting the switch to **Synced** will reset the phase of the oscillator with every incoming MIDI note. When the Phase Sync switch is set to Fr **Free**, the oscillator will not be reset. When only one oscillator is active and you have set a single voice in the **Uni** panel, the mode **Free** has almost no effect. When you have a higher Unisono number of voices or multiple active oscillators, every note leads to an audible change in the sound.
- **Phase**: Sets the phase of the oscillator. This control is available only when the Phase Sync switch is set to **Synced**. The effect of the **Phase** control is only audible if a second oscillator is active. This parameter is useful when using integer values for frequency ratios between carrier and modulator, as it is the case in FM.

The **Mod** panel contains the following parameters:

- **Iterations**: Selects the number of similar repetitions that you want of the same sound, and how much you want to deviate from the original waveform. The value range is between 2 and 7. Higher values lead to more complex, lighter-sounding sounds (and heavier CPU loads).
- **Warp**: Defines the shift of the copied area relative to the original waveform: 0 corresponds to a position before the waveform, 1 to a position after the waveform. With a value of 0.5, the area is in the center of the waveform. Modulating this parameter with an LFO or an envelope can produce interesting movements inside the sound.
- **Amount**: Adjusts the mixing relationship between the original waveform and its copies.

The function **Fractalize** of the Wave page gives you an idea of what happens. Load a simple waveform (such as a sine) into the Wave page. Check under the Wave view: so far nothing has happened. Choose the entry **Fractalize** from the **Transform** menu. Set **Iterations** to **2** and **Displacement** to **9**. Slowly increase the **Displacement** value up to **25**. You see how the waveform distends. Change the **Iterations** value: The higher the value, the more dislocated the waveform becomes.

In the Fractalize Oscillator the **Uni** panel lacks a **Voices** parameter. Instead, the number of voices is set by the **Iterations** parameter on the **Mod** panel.

The Fractalize Oscillator works particularly well with waveforms containing few, yet strong harmonics: You will see new overtones emerge around the basic harmonics. By choosing the right waveform you can achieve interesting, formant-style effects. With harmonically dense waveforms (for example, the Saw from the factory waveforms) the effect is not as striking.

Sync Granular Oscillator

The Sync Granular Oscillator is similar to the Granular Engine module: It divides the waveform into very small parts, the so-called "grains," and then brings these grains back together. The difference between both modules is the input material used: In the Granular Engine the grains are taken from a sample, which is a much longer piece of audio, whereas the Sync Granular Oscillator uses one cycle of the desired waveform.

Before putting the grains back together (this is known as resynthesis), you can influence the grain "cloud": You can change the frequency of the grains, change the density of the cloud by adjusting how individual grains overlap, and modify the diffusion in the grain cloud.

These options allow you to create very effective sounds and, for example, convincingly simulate the blowing sound of a wind instrument such as a pipe or flute.

The **Main** panel contains the same parameters as the Single Oscillator with an additional **Balance** control:

- **Mini Waveform**: Shows the shape and the name of the loaded waveform. If a Morph waveform is loaded, you can click the shape and drag your mouse vertically to morph the waveform. Clicking the waveform name opens the Waveform selector and lets you choose another waveform.
- **Phase Invert button (phase symbol)**: Above the waveform name, the Phase Invert button lets you reverse the phase of the signal generated by the oscillator.
- **Frequency menu and control (Freq)**: These controls let you to define the oscillator's frequency as a transposition of the played note (**Transpose**), as a relationship with the played note (**Ratio**), as a fixed frequency (**Hz**), or as a fixed MIDI note number (**Note**). You can find more information in Frequency menu and control.
- **Balance control**: Adjusts the mix between the original waveform and the waveform produced by resynthesis. At 0 (knob at full left) only the original waveform can be heard, at 1 (knob at full right) only the resynthesized waveform.
- **Phase Sync switch**: Setting the switch to **Synced** will reset the phase of the oscillator with every incoming MIDI note. When the Phase Sync switch is set to Fr **Free**, the oscillator will not be reset. When only one oscillator is active and you have set a single voice in the **Uni** panel, the mode **Free** has almost no effect. When you have a higher Unisono number of voices or multiple active oscillators, every note leads to an audible change in the sound.
- **Phase**: Sets the phase of the oscillator. This control is available only when the Phase Sync switch is set to **Synced**. The effect of the **Phase** control is only audible if a second oscillator is active. This parameter is useful when using integer values for frequency ratios between carrier and modulator, as it is the case in FM.

The **Mod** panel contains the following parameters:

- **Frequency menu and control (Freq)**: These controls let you to define the oscillator's frequency as a transposition of the played note (**Transpose**), as a relationship with the played note (**Ratio**), as a fixed frequency (**Hz**), or as a fixed MIDI note number (**Note**). You can find more information in Frequency menu and control.
- **Density**: Adjusts the density of the grain cloud by setting how individual grains overlap. The values range from 3 to 8. Small values give a raw sound, higher value a more polished sound.
- **Diffusion**: Adjusts the diffusion, that is, the accidental scattering of the grain cloud.

Sample Engine

Two source modules are sample-based: Sample Engine and Granular Engine. These modules allow you to use Absynth as a sampler and play back audio data loaded in WAV, AIFF, MP3, OGG, or FLAC format.

The basic control is the same in both modules, so the following instructions for the Sample Engine also apply to the Granular Engine.

ⓘ In contrast to conventional sampling instruments, Absynth lacks the usual functions such as key mapping, velocity layering or AKAI import. Unlike the conventional samplers, Absynth's emphasis is not on the realistic reproduction of sampled instruments, but rather on the creative possibilities that sample-based synthesis provides.

You can load a sample in all three source modules of the patch; in a single preset you can use up to three different samples.

To load a sample into the Sample Engine, do the following:

1. Select **Sample Engine** from the Type menu of the source module.

→ The Sample selector on the **Main** panel reads **(No Sample)**.

2. Click this **(No Sample)** label.

→ A dialog appears that lets you select a sample file on your hard drive.

3. Choose a sample file and click **Open**.

→ The sample is now loaded and the Sample selector shows its name and a visual illustration of the sample.

Absynth can load audio data in AIFF, WAV, or FLAC format (stereo or mono) ranging from 16 to 32 bit and any sampling rate, and in MP3 or OGG format (stereo or mono) at any sample rate.

The **Main** panel contains the following parameters:

- **Mini Sample**: Shows the shape and the name of the loaded sample. Clicking the sample name opens a dialog that lets you select and load another sample from your hard drive.
- **Mono/Stereo switch**: Stereo samples can be played in stereo or mono. If you have loaded a stereo sample, you can use the Mono/Stereo switch to switch between mono and stereo playback. If your sample cannot be switched to stereo, then it is a mono sample.
- **Frequency menu and Frequency control**: These controls let you to define the oscillator's frequency as a transposition of the played note (**Trans**), as a relationship with the played note (**Ratio**), as a fixed frequency (**Hz**), or as a fixed MIDI note number (**Note**). The sample will play at its original speed when the MIDI Note C3 is played. Lower notes produce a slower, lower sound, while higher notes sound higher and faster. You will find more information in Frequency menu and control.
- **Start**: Defines the starting point of the playback in the sample relative to the overall sample length.

The **Loop** panel contains the following control:

- **Loop Mode selector**: Selects from three loop modes. If **Off** is selected, the sample is played once (good for percussion sounds). If **Loop All** is selected, the entire sample is played in loop (this works best if you edit the loop beforehand to create a seamless loop). If **Loop Edit** is selected, you can adjust the start point and end point of the loop by dragging the left or right border of the highlighted region in the loop display, or you can move the entire loop on the timeline by clicking anywhere in the highlighted region and dragging your mouse horizontally.

Granular Engine

Like the Sample Engine, the Granular Engine uses a sample as basis for generating sound. However, this module divides the sample into many small **grains**, each containing a tiny fragment of the sound. In this way you can control the sample's pitch and duration independently of each other. In the Granular Engine module, the duration of the sample remains the same over the entire keyboard, while the notes determine the pitch.

The **Main** panel contains the same parameters as in the Sample Engine with an additional **Time %** control:

- **Mini Sample**: Shows the shape and the name of the loaded sample. Clicking the sample name opens a dialog that lets you select and load another sample from your hard drive.
- **Mono/Stereo switch**: Stereo samples can be played in stereo or mono. If you have loaded a stereo sample, you can use the Mono/Stereo switch to switch between mono and stereo playback. If your sample cannot be switched to stereo, then it is a mono sample.
- **Frequency menu and Frequency control**: These controls let you to define the oscillator's frequency as a transposition of the played note (**Trans**), as a relationship with the played note (**Ratio**), as a fixed frequency (**Hz**), or as a fixed MIDI note number (**Note**). The sample will play at its original speed when the MIDI Note C3 is played. Lower notes produce a slower, lower sound, while higher notes sound higher and faster. You will find more information in Frequency menu and control.
- **Time %**: Determines the playback speed of the sample. 50 % corresponds to half of the original speed, while 200 % doubles the speed of your sound. If you set **Time %** to zero, the sample for that length of time will "freeze." When you have frozen a sample, you can use the **Start** control to specify when exactly it should freeze.
- **Start**: Defines the starting point of the playback in the sample relative to the overall sample length.

The **Grain** panel contains the following parameters:

- **Density**: Determines the number of grains played simultaneously. You can set a value between 1 and 32. Lower values produce a thinner sound, while higher values produce a dense granular cloud. High **Density** values can also generate high CPU loads.
- **Grain Size ms**: Determines the length of the grains measured in milliseconds. Smaller values are good for percussive sounds, whereas higher values work well for pad-like sounds or string instruments. Small values tend to conceal the character of the sample and produce typical granular artifacts in the form of noise. The maximum grain size is 500 milliseconds.
- **Time Jitter**: Defines the randomness of the grains' playback speed. Zero means no randomness during grain playback, whereas 100 means that the grains are played completely at random, within the time window defined by the **Grain Size ms** parameter.
- **Freq Jitter**: Changes the pitch of individual grains randomly. Zero means no randomness, whereas 100 results in completely random pitches.
- **Amp Jitter**: Changes the amplitude ("loudness") of the individual grains randomly. A value of zero means no randomness, whereas 100 results in completely random amplitude levels.

Audio In

The Audio In source module does not produce signal itself, but rather transmits incoming audio signals from your DAW to its output. That makes it possible for any audio signal to work with the other modules in real time. You can also load Absynth 6 as an effect plug-in and let it process any audio track(s).

The **Main** panel is the only panel in the Audio In module. It contains the following parameters:

- **Mono/Stereo switch**: Selects between mono and stereo signals. If you set it to mono, a single Input menu is available. If you set it to stereo, a second Input menu appears.
- **Level control**: Adjusts the level of the input signal. At the default value of 0 dB, the signal passes through the input stage unchanged.
- **Input menu**: Chooses any of the six available audio inputs.

External audio signals run through Absynth 6's signal path like internally created signals do, thus they are also influenced by Absynth 6"s envelopes. This means that you do not hear audio signals from external sources unless Absynth 6 has received MIDI notes and used them to trigger the envelopes.

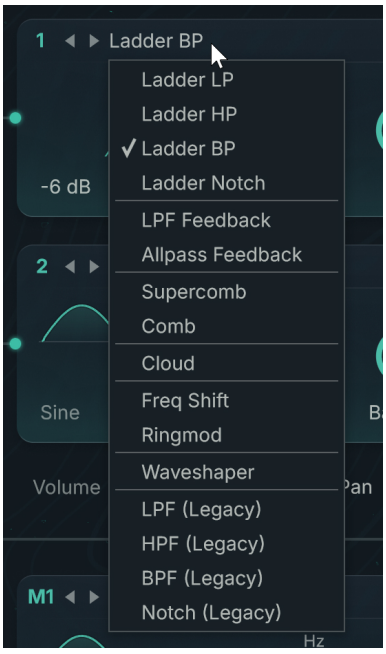
Insert modules

The insert modules can be inserted into the slots labeled **1** and **2** on every Oscillator channel **A**–**C** and into the slots labeled **M1** and **M2** on the Main channel.

The insert modules are grouped into three categories:

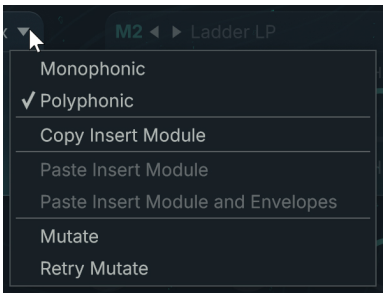
- **Filters** amplify or attenuate specific frequency contents in the input signal. The available Filter modules are described in Filters.
- **Modulators** use their own built-in oscillator to modulate the incoming signal. The available Modulator modules are described in Modulators.
- The **Waveshaper** uses a waveform to shape the input signal. You can find more information in Waveshaper.

You can choose the type of insert module from the Type selector at the top of each insert slot:



Edit menu for insert modules

The Edit menu of an insert module contains the following entries:

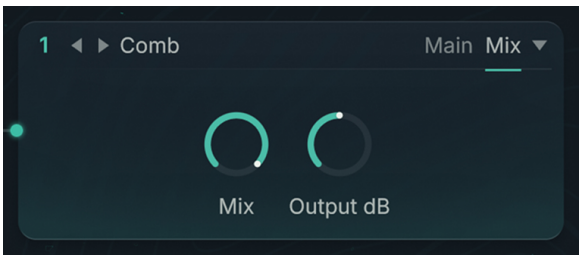


- **Monophonic/Polyphonic (modules on Main channel only)**: Switches the insert module between Monophonic and Polyphonic modes.
- **Copy Insert Module**: Copies the settings of the module to the clipboard.
- **Paste Insert Module**: Pastes into the module the settings stored in the clipboard (using the command **Copy Insert Module**).
- **Paste Insert Module and Envelopes**: Pastes into the module the settings stored in the clipboard (using the command **Copy Insert Module**), including any envelopes linked to the copied module.
- **Mutate**: Mutates this particular module.
- **Retry Mutate**: Retries a mutation of the module. This entry is available only if you have already triggered a mutation in this session.

The **Mutate** and **Retry Mutate** commands use the mutation settings defined in the Mutator section of the Browser page.

Mix panel

Every insert module provides a **Mix** panel with the following controls:



- **Mix**: Adjusts the balance between the original signal and the signal processed by the module.
- **Output dB**: Adjusts the output level of the module. This can help compensate for any level changes induced by the module's processing.

> The only exception is the **Mix** panel of the Ladder Notch module, which contains an additional **LP/HP** parameter.

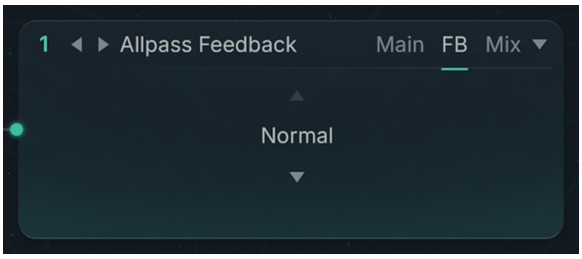
Filters

The Filter modules provide a number of filters to work with.

Feedback loop and Feedback panel

The **LPF Feedback**, **Allpass Feedback**, and **Supercomb** Filter modules provide a **feedback loop**. This feedback loop allows to send a portion of the filter's output back to its own input while adding an extra processing on the way.

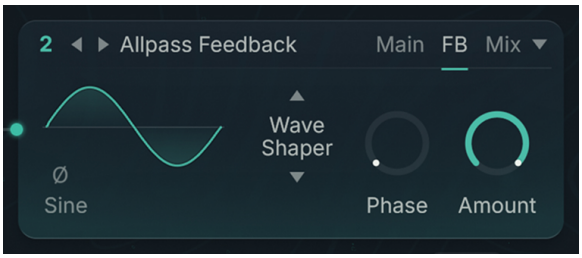
The feedback loop settings are located on the **FB** panel (Feedback) panel:



> The feedback loop is built inside the resonance loop of the Filters. Therefore, the amount of signal sent in the feedback loop is also affected by the **Feedback** parameter of the **Main** panel for these modules, which controls the filter resonance.

The Feedback panel contains a **Feedback Mode menu** selecting the feedback mode. If **Normal** is selected, no additional processing takes place within the feedback loop. The three other modes available in the Feedback Mode menu are described below.

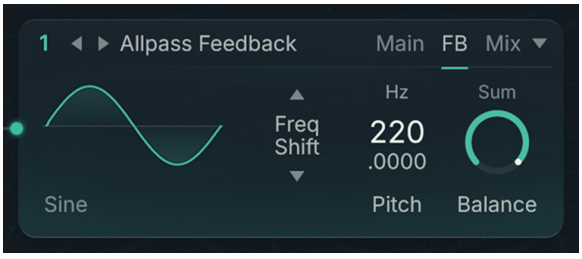
Wave Shaper feedback mode



When this mode is selected, the signal in the feedback loop is processed by a mini Waveshaper module. The parameters in the **FB** panel are the same as in the Waveshaper module except for the following differences:

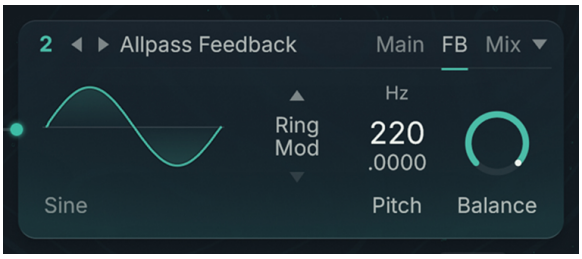
- A Phase Inverter button is available below the Mini Waveform.
- The **Input** control from the Waveshaper module is replaced with an **Amount** control adjusting the proportion of signal processed by the Waveshaper in the feedback loop.

Freq Shift feedback mode



When this mode is selected, the signal in the feedback loop is processed by a mini Freq Shift module. The parameters in the **FB** panel are the same as in the Freq Shift module except that the **Feedback** control from the Freq Shift module is replaced with a **Balance** control adjusting the proportion of signal processed by the Freq Shift.

Ring Mod feedback mode



When this mode is selected, the signal in the feedback loop is processed by a mini Ringmod module. The parameters in the **FB** panel are the same as in the Ringmod module.

Ladder LP

This lowpass filter is based on the classic ladder circuit used in early synthesizers.

The **Main** panel contains the following parameters:

- **Slope menu**: Selects the slope gradient of the signal attenuation in dB per octave. A value of -12 dB/octave means that the filter dampens a signal one octave higher by 12 dB more. The available values are -6 dB, -12 dB, -18 dB, and -24 dB.
- **Frequency control and Frequency menu**: The Frequency control adjusts the cutoff frequency in semitones (**Transpose**) or in Hertz (**Hz**) depending on the unit selected in the Frequency menu above.
- **Resonance control (Res)**: Adjusts the resonance at the cutoff frequency.

The **Mix** panel contains the following parameters:

- **Mix**: Adjusts the balance between the original signal and the signal processed by the module.
- **Output dB**: Adjusts the output level of the module. This can help compensate for any level changes induced by the module's processing.

Ladder HP

This highpass filter is based on the classic ladder circuit used in early synthesizers.

The **Main** panel contains the following parameters:

- **Slope menu**: Selects the slope gradient of the signal attenuation in dB per octave. A value of -12 dB/octave means that the filter dampens a signal one octave lower by 12 dB more. The available values are -6 dB, -12 dB, -18 dB, and -24 dB.
- **Frequency control and Frequency menu**: The Frequency control adjusts the cutoff frequency in semitones (**Transpose**) or in Hertz (**Hz**) depending on the unit selected in the Frequency menu above.
- **Resonance control (Res)**: Adjusts the resonance at the cutoff frequency.

The **Mix** panel contains the following parameters:

- **Mix**: Adjusts the balance between the original signal and the signal processed by the module.
- **Output dB**: Adjusts the output level of the module. This can help compensate for any level changes induced by the module's processing.

Ladder BP

This bandpass filter is based on the classic ladder circuit used in early synthesizers.

The **Main** panel contains the following parameters:

- **Slope menu**: Selects the slope gradient of the signal attenuation around the cutoff frequency, in dB per octave. A value of -12 dB/octave means that a signal one octave further away from the cutoff frequency would be dampened by 12 dB more. The available values are -6 dB and -12 dB.
- **Frequency control and Frequency menu**: The Frequency control adjusts the cutoff frequency in semitones (**Transpose**) or in Hertz (**Hz**) depending on the unit selected in the Frequency menu above.
- **Resonance control (Res)**: Adjusts the resonance at the cutoff frequency.

The **Mix** panel contains the following parameters:

- **Mix**: Adjusts the balance between the original signal and the signal processed by the module.
- **Output dB**: Adjusts the output level of the module. This can help compensate for any level changes induced by the module's processing.

Ladder Notch

A notch filter is the opposite of a bandpass filter: It rejects a particular frequency band. This notch filter is based on the classic ladder circuit used in early synthesizers. It combines a lowpass filter and a highpass filter.

The **Main** panel contains the following parameters:

- **Slope menu**: Selects the slope gradient of the signal attenuation around the cutoff frequency, in dB per octave. A value of -12 dB/octave means that a signal one octave closer to the cutoff frequency would be dampened by 12 dB more. The available values are -6 dB and -12 dB.
- **Frequency control and Frequency menu**: The Frequency control adjusts the cutoff frequency in semitones (**Transpose**) or in Hertz (**Hz**) depending on the unit selected in the Frequency menu above.
- **Resonance control (Res)**: Adjusts the resonance at the cutoff frequency.
- **Bandwidth**: Adjusts the width of the frequency band that is rejected.

The **Mix** panel contains the following parameters:

- **LP/HP**: Adjusts the balance between the lowpass and highpass filter outputs.
- **Mix**: Adjusts the balance between the original signal and the signal processed by the module.
- **Output dB**: Adjusts the output level of the module. This can help compensate for any level changes induced by the module's processing.

LPF Feedback

This lowpass filter is equipped with a feedback loop.

The **Main** panel contains the following parameters:

- **Slope menu**: Selects the slope gradient of the signal attenuation, expressed as a number of poles. The available values are **2-Pole**, **4-Pole**, and **8-Pole**, which correspond to -12 dB/octave, -24 dB/octave, and -48 dB/octave, respectively. A value of -12 dB/octave means that the filter dampens a signal one octave higher by 12 dB more.
- **Frequency control and Frequency menu**: The Frequency control adjusts the cutoff frequency in semitones (**Transpose**) or in Hertz (**Hz**) depending on the unit selected in the Frequency menu above.

- **Feedback**: Adjusts the level of the signal sent into the feedback loop, which affects the resonance at the cutoff frequency. This control is bipolar: At its default middle position (zero), the resonance is at minimum. From this middle position, turning the **Feedback** knob in either direction will increase the resonance of the filter.

The parameters of the **FB** panel (Feedback) let you configure the additional processing in the feedback loop. They are described in Feedback loop and Feedback panel.

The **Mix** panel contains the following parameters:

- **Mix**: Adjusts the balance between the original signal and the signal processed by the module.
- **Output dB**: Adjusts the output level of the module. This can help compensate for any level changes induced by the module's processing.

Allpass Feedback

Allpass filters allow all frequencies to pass through the output with similar strength, but modify the phase of the signal. Allpass filters are useful for creative filtering as they can easily produce phasing or resonating effects. With allpass filters, the number of poles define the number of peaks in the frequency spectrum of the filtered signal. With high resonance values, an 8-pole allpass filter can sound similar to a bell.

This allpass filter is equipped with a feedback loop.

The **Main** panel contains the following parameters:

- **Slope menu**: Selects the slope gradient of the signal attenuation, expressed as a number of poles. The available values are **2-Pole**, **4-Pole**, and **8-Pole**.
- **Frequency control and Frequency menu**: The Frequency control adjusts the cutoff frequency in semitones (**Transpose**) or in Hertz (**Hz**) depending on the unit selected in the Frequency menu above.
- **Feedback**: Adjusts the level of the signal sent into the feedback loop, which affects the resonance at the cutoff frequency. This control is bipolar: At its default middle position (zero), the resonance is at minimum. From this middle position, turning the **Feedback** knob in either direction will increase the resonance of the filter.

The parameters of the **FB** panel (Feedback) let you configure the additional processing in the feedback loop. They are described in Feedback loop and Feedback panel.

The **Mix** panel contains the following parameters:

- **Mix**: Adjusts the balance between the original signal and the signal processed by the module.
- **Output dB**: Adjusts the output level of the module. This can help compensate for any level changes induced by the module's processing.

Supercomb

The Supercomb filter is a Comb Filter equipped with a feedback loop and a tonal control borrowing characteristics from the Resonators and Pipe effects.

The Supercomb provides the following panels:

- The **Main** panel is the same as in the Comb Filter.
- The **FB** panel (Feedback) specifies the processing taking place in the feedback loop.
- The **Tone** panel lets you adjust the tone of the filter.

Main panel

The **Main** panel contains the following parameters:

- **Frequency control and Frequency menu**: The Frequency control adjusts the cutoff frequency in semitones (**Transpose**) or in Hertz (**Hz**) depending on the unit selected in the Frequency menu above.
- **Feedback**: Adjusts the amplification factor of the delayed signal. Higher values induce steeper frequency peaks and cancellations.

Feedback panel

The parameters of the **FB** panel (Feedback) let you configure the additional processing in the feedback loop. They are described in Feedback loop and Feedback panel.

Tone panel

The **Tone** panel contains the following parameters:

- **Tone Mode selector**: Selects from different resonance modes. The higher the feedback is turned up, the more pronounced the effect is. The Tone mode changes the color of the decay.
- **Tone**: This controls affects the sound differently according to the mode selected in the Tone Mode menu above. It is not available in **Clean** mode.
- **HP**: Low frequency damping. Low frequencies decay faster as this control is turned up.
- **LP**: High frequency damping. High frequencies decay faster as this is turned down. This control is useful for a more natural resonance.
- **Position**: Controls the ratio of the delay taps, as described for the Comb Filter. This control changes the color of the filter.

Don't hesitate to play with these controls to understand how they affect the sound. In particular, you can modulate the **Position** parameter to get nice effects!

Mix panel

The **Mix** panel contains the following parameters:

- **Mix**: Adjusts the balance between the original signal and the signal processed by the module.
- **Output dB**: Adjusts the output level of the module. This can help compensate for any level changes induced by the module's processing.

Comb

Comb filters change the sound by delaying the signal by a few milliseconds and mixing the delayed signal with the original. As a result, the level of certain frequencies may be raised or attenuated in the filtered signal.

Effects such as the phaser and the flanger use this phenomenon. By modulating the Comb's parameters, you can quickly produce a nice flanging effect.

The **Main** panel contains the following parameters:

- **Frequency control and Frequency menu**: The Frequency control adjusts the cutoff frequency in semitones (**Transpose**) or in Hertz (**Hz**) depending on the unit selected in the Frequency menu above.
- **Feedback**: Adjusts the amplification factor of the delayed signal. Higher values induce steeper frequency peaks and cancellations.

The **Mix** panel contains the following parameters:

- **Mix**: Adjusts the balance between the original signal and the signal processed by the module.
- **Output dB**: Adjusts the output level of the module. This can help compensate for any level changes induced by the module's processing.

Cloud

The Cloud filter is a reduced version of the Aetherizer effect, a granular delay with multiple feedback and tone controls. It inherits the most important parameters of the Aetherizer.

The Cloud filter does not have the panels usually found in other filter types. Instead, it provides the three following panels:

- The **Grain** panel controls the grain cloud.
- The **Tone** panel configure a filter.
- The **Mix** panel adjusts the mix between the dry and processed signals and the level of the output signal.

Grain panel

The **Grain** panel contains the following parameters:

- **HD switch**: Activates or deactivates the High Density mode. In High Density mode the grain cloud can reach higher densities, which results in a both fuller and more polished sound.
- **Rate Hz**: Defines the number of grains created during one second.
- **Transpose**: Sets the global transposition of the grains.
- **Delay ms**: Adjusts the pre-delay time involved in the grain creation,, in milliseconds.

Each parameter also has its own dedicated Jitter slider on the right that lets the value randomly deviate from the defined setting.

Tone panel

The **Tone** panel contains the following parameters:

- **Filter on/off switch**: Activates/deactivates the Cloud's internal filter. The remaining parameters on the panel are visible only if this **Filter** switch is turned on.
- **Frequency**: Defines the cutoff frequency of the internal filter.
- **Resonance**: Adjusts the resonance of the internal filter.
- **Quantize menu**: Activates/deactivates the filter quantization, and selects a quantization mode. The filter quantization distributes the possible cutoff frequencies over one of the scales available in the menu.
- **Transpose (all modes except Vowel and Quantize Off)**: Adjusts the base pitch of the scale on which the cutoff frequencies are quantized.
- **Vowel Mix (Vowel mode only)**: Lets you morph between different vowels.

The Frequency, **Resonance**, and Transpose / Vowel Mix parameters also have their own dedicated Jitter sliders on the right that let the values randomly deviate from the defined settings.

Mix panel

The **Mix** panel contains the following parameters:

- **Mix**: Adjusts the balance between the original signal and the signal processed by the module.
- **Output dB**: Adjusts the output level of the module. This can help compensate for any level changes induced by the module's processing.

You can find more information on all these parameters in Aetherizer.

LPF (Legacy)

This is the lowpass filter available in previous versions of Absynth.

The **Main** panel contains the following parameters:

- **Slope menu**: Selects the slope gradient of the signal attenuation in dB per octave. A value of -12 dB/octave means that the filter dampens a signal one octave higher by 12 dB more. The available values are -6 dB, -12 dB, and -24 dB.
- **Frequency control and Frequency menu**: The Frequency control adjusts the cutoff frequency in semitones (**Transpose**) or in Hertz (**Hz**) depending on the unit selected in the Frequency menu above.
- **Resonance control (Res)**: Adjusts the resonance at the cutoff frequency.

The **Mix** panel contains the following parameters:

- **Mix**: Adjusts the balance between the original signal and the signal processed by the module.
- **Output dB**: Adjusts the output level of the module. This can help compensate for any level changes induced by the module's processing.

HPF (Legacy)

This is the highpass filter available in previous versions of Absynth.

The **Main** panel contains the following parameters:

- **Slope menu**: Selects the slope gradient of the signal attenuation in dB per octave. A value of -12 dB/octave means that the filter dampens a signal one octave lower by 12 dB more. The available values are -6 dB and -12 dB.
- **Frequency control and Frequency menu**: The Frequency control adjusts the cutoff frequency in semitones (**Transpose**) or in Hertz (**Hz**) depending on the unit selected in the Frequency menu above.
- **Resonance control (Res)**: Adjusts the resonance at the cutoff frequency.

The **Mix** panel contains the following parameters:

- **Mix**: Adjusts the balance between the original signal and the signal processed by the module.
- **Output dB**: Adjusts the output level of the module. This can help compensate for any level changes induced by the module's processing.

BPF (Legacy)

This is the bandpass filter available in previous versions of Absynth.

The **Main** panel contains the following parameters:

- **Frequency control and Frequency menu**: The Frequency control adjusts the cutoff frequency in semitones (**Transpose**) or in Hertz (**Hz**) depending on the unit selected in the Frequency menu above.
- **Resonance control (Res)**: Adjusts the resonance at the cutoff frequency.

The **Mix** panel contains the following parameters:

- **Mix**: Adjusts the balance between the original signal and the signal processed by the module.
- **Output dB**: Adjusts the output level of the module. This can help compensate for any level changes induced by the module's processing.

Notch (Legacy)

This is the notch filter available in previous versions of Absynth.

The **Main** panel contains the following parameters:

- **Frequency control and Frequency menu**: The Frequency control adjusts the cutoff frequency in semitones (**Transpose**) or in Hertz (**Hz**) depending on the unit selected in the Frequency menu above.
- **Resonance control (Res)**: Adjusts the resonance at the cutoff frequency.
- **Bandwidth**: Adjusts the width of the frequency band that is rejected.

The **Mix** panel contains the following parameters:

- **Mix**: Adjusts the balance between the original signal and the signal processed by the module.
- **Output dB**: Adjusts the output level of the module. This can help compensate for any level changes induced by the module's processing.

Modulators

The Modulator modules use a built-in oscillator to modulate the incoming signal. Two Modulators are available: Freq Shift and Ringmod.

Freq Shift

The Freq Shift module produces a frequency shift by using a feedback loop. From a functional point of view, the frequency shift resembles the ring modulation, and it can also bring similar sonic results. From a technical point of view, though, there is the following difference: While the Ringmod module produces frequency sums as well as frequency differences, the Freq Shift module limits itself to either frequency sums or differences. Practically, this means that the Freq Shift sounds are subtler and easier to control than the Ringmod sounds. Because there is no interference between the sum and the difference frequencies, the Freq Shift frequently sounds cleaner than the Ringmod in situations involving complex input signals (for example, samples or saw tooth waveforms).

> ⓘ Frequency shifting is not equivalent to pitch shifting. A pitch shifter multiplies the frequencies of the input signal by a certain factor: As a result, the harmonic relations between the frequencies are maintained. By contrast, a frequency shifter adds a certain value to the frequencies of the input signal, which alters the harmonic relations the input signal may contain.

The **Main** panel contains the following parameters:

- **Mini Waveform**: Shows the shape and the name of the loaded waveform. If a Morph waveform is loaded, you can click the shape and drag your mouse vertically to morph the waveform. Clicking the waveform name opens the Waveform selector and lets you choose another waveform.
- **Pitch menu and control**: These controls let you to define the oscillator's frequency as a transposition of the played note (**Transpose**), as a relationship with the played note (**Ratio**), as a fixed frequency (**Hz**), or as a fixed MIDI note number (**Note**).

  > ⓘ The Pitch menu and control are similar to the Frequency menu and control available in the Oscillator modules. You can find more information in Frequency menu and control.

- **Sum/Difference switch**: With the switch set to **Sum**, the Freq Shift module produces frequency sums, otherwise it produces frequency differences.
- **Feedback**: Adjusts the amount of feedback. A higher feedback generates a richer sound.

A light detuning between the input signal and the Modulator signal, combined with a moderate **Feedback** value, can induce a nice phasing effect. Very low Frequency values (e.g., 1 Hz) can lead to similar effects.

The **Mix** panel contains the following parameters:

- **Mix**: Adjusts the balance between the original signal and the signal processed by the module.
- **Output dB**: Adjusts the output level of the module. This can help compensate for any level changes induced by the module's processing.

Ringmod

The Ringmod module produces ring modulation and is similar to the Ringmod Oscillator from the source modules. The amplitudes of the input signal and Modulator signal are multiplied with each other.

The **Main** panel of the Ringmod module contains the following parameters:

- **Mini Waveform**: Shows the shape and the name of the loaded waveform. If a Morph waveform is loaded, you can click the shape and drag your mouse vertically to morph the waveform. Clicking the waveform name opens the Waveform selector and lets you choose another waveform.
- **Pitch menu and control**: These controls let you to define the oscillator's frequency as a transposition of the played note (**Transpose**), as a relationship with the played note (**Ratio**), as a fixed frequency (**Hz**), or as a fixed MIDI note number (**Note**).

  The Pitch menu and control are similar to the Frequency menu and control available in the Oscillator modules. You can find more information in Frequency menu and control.

- **Balance**: Adjusts the balance between the input signal and the modulated signal.

The **Mix** panel contains the following parameters:

- **Mix**: Adjusts the balance between the original signal and the signal processed by the module.
- **Output dB**: Adjusts the output level of the module. This can help compensate for any level changes induced by the module's processing.

Waveshaper

The Waveshaper module uses a waveform to shape an input signal. This idea is used in guitar amplifiers and distortion effects. Waveshaping reacts to the amplitude of the input signal: The sound changes depending on the volume envelope of the oscillator or any other form of oscillator-volume control. Additionally, waveshaping emphasizes phasing and detuning effects in the signal.

You can always use the Waveshaper module when you want to enrich a signal with harmonics. Its action ranges from subtle density to cutting distortion. The Waveshaper reacts to changes in the amplitude. The level of distortion changes with every fluctuation of the input signal, which generates a very dynamic sound.

The **Main** panel contains the following parameters:

- **Mini Waveform**: Shows the shape and the name of the loaded waveform. If a Morph waveform is loaded, you can click the shape and drag your mouse vertically to morph the waveform. Clicking the waveform name opens the Waveform selector and lets you choose another waveform.
- **Input**: Adjusts the input level in dB. Increasing this value usually strengthens the distortion. This strengthening is not linear, however, and depends on the selected waveform and the **Phase** value. You can experiment with different settings to get the sound that you want.
- **Phase**: Adjusts the phase of the waveform. This parameter has an extreme effect on the sound, especially when working with complex waveforms, since these will be very sensitive to phase changes.

If you want to make yourself more comfortable with the effect of the Waveshaper, you could create a new waveform for the Waveshaper and work on it in the Spectrum view of the Wave page while listening to the results.

The **Mix** panel contains the following parameters:

- **Mix**: Adjusts the balance between the original signal and the signal processed by the module.
- **Output dB**: Adjusts the output level of the module. This can help compensate for any level changes induced by the module's processing.

Effect modules

The effect modules can be inserted in the third and rightmost position in the Main channel at the bottom of the Patch page. This makes them the last processing in the signal path of Absynth. The effect modules host the effects configured in the Effect page.

You can switch between the available effect modules by using the Type selector at the top of the modules:

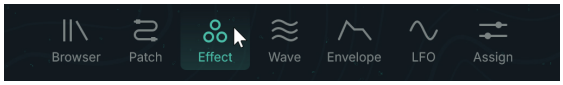
The **Main** panel of the effect modules shows the most important parameters of the respective effects. These parameters vary with each effect, you can find their description in Effect page. You can click the effect thumbnail in the left part of the **Main** panel to jump directly to that effect in the Effect page.

The **Mix** panel of the effect modules contains two knobs: **Dry** and **Wet**. These knobs let you adjust the balance between the unprocessed signal (**Dry**) and the processed signal (**Wet**). They correspond to the **Dry** and **Wet** knobs in the **Input Mix** section of the Effect page.

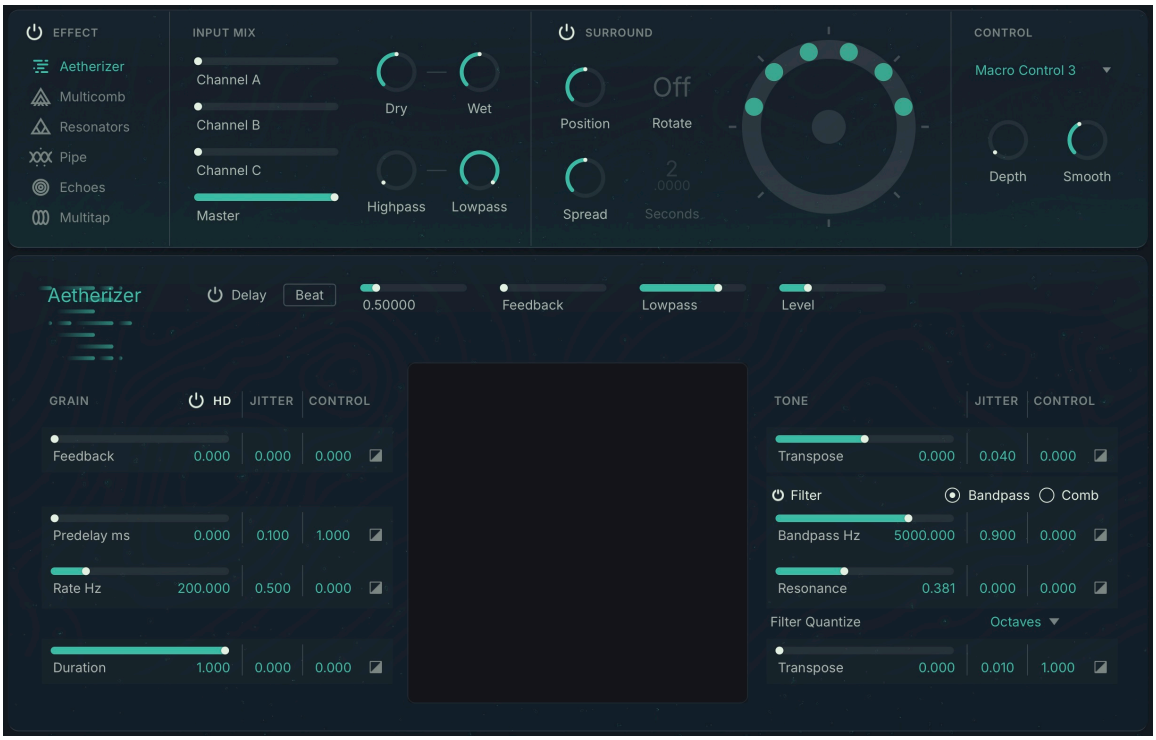
7. Effect page

The Effect page of Absynth 6 lets you choose and configure an effect that is applied to the sound coming from the three channels on the Patch page. Additionally, the Surround panner enables you to place sounds in the surround field when using a multichannel output configuration.

▶ To open the Effect page, click the **Effect** button at the top of the Absynth 6 window:



The Effect page consists of the following elements and sections:



- **Effect On/Off**: Bypasses the Effect page, routing the channels on the Patch page directly to the output. Bypassing the Effect page reduces the CPU load.
- **Effect**: Selects one of the following effects that will be be applied to your sound. You can edit the selected effect in the lower area of the Effect page.
  - Aetherizer
  - Multicomb
  - Resonators
  - Pipe
  - Echoes
  - Multitap
- **Input Mix**: Lets you adjust the signal levels coming from the Patch page, global dry/wet controls, and apply input filtering. You can find more information in Input Mix section.
- **Surround**: Lets you adjust the controls of the Surround Panner. You can find more information in Surround section.

- **Control**: Lets you select Macro controls, and adjust their modulation depth and smoothing. You can find more information in Control section.

Input Mix section

The Input Mix section lets you adjust the signal levels coming from the Patch page before they are processed by the effect selected on the Effect page. Additionally, you can set the dry and wet effect levels, and apply input filtering to the signal.

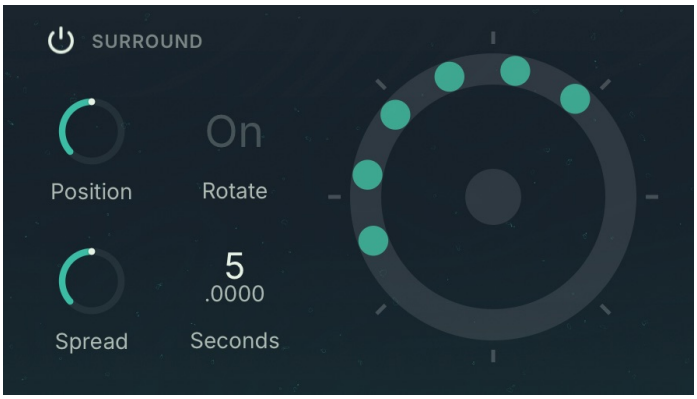
- **Channel A/B/C**: Adjusts the levels of the individual channel signals coming from the Patch page.
- **Master**: Adjusts the level of the main signal coming from the Patch page, which is composed according to each channel's **Volume** and **Pan** controls on the Patch page.
- **Dry**: Adjusts the level of the dry signal coming from the Patch page at the output of the instrument.
- **Wet**: Adjusts the level of the effect signal at the output of the instrument.
- **Highpass**: Applies a high-pass filter to the input signal of the selected effect, removing low frequencies as you turn up the control.
- **Lowpass**: Applies a low-pass filter to the input signal of the selected effect, removing high frequencies as you turn down the control.

Surround section

The Effect page integrates Absynth 6's Surround Panner. All of the available effects support multichannel applications, from multiple delay taps to independent resonators, which can be placed separately in the surround field.

Using the Surround Panner greatly expands the possibilities for spatial sound design. For example, you can make echoes come from behind your listening position, or you can make the Pipe effect circle around you.

The Surround Panner can also be used if a stereo configuration is selected in the Surround tab of the Settings dialog. In this case the surround signal will be mixed down to stereo.



- **Surround On/Off**: Switches the Surround Panner on or off.
- **Position**: Moves the position of all effect signals around a circle in the surround field. The relative positions of the signals remain constant. At center position, the effect signals are oriented towards the front of the surround field. Turning the control to the left moves the signals in counter-clockwise direction, turning it to the right moves them in clockwise direction.
- **Spread**: Distributes the effect signals across a circle in the surround field. When turned fully to the left, all signals are played back in the same position. Turning the control to the right increases the distances between them, relative to the individual **Pan** settings of the effect (if applicable).
- **Rotate**: Routes an internal LFO to **Position** and creates a circular movement of all effect signals in the surround field.
- **Seconds**: Adjusts the duration of the circular movement of the effect signals created by the **Rotate** function. Increasing the value slows the movement down.
- **Display**: Shows the position of the individual effect signals in the surround field, each represented by a dot. Each effect is represented according to the number of individual effect signal it produces. The front of the surround field is at the top.

Control section

The Control section lets you define how the effect responds globally to modulation coming from the selected Macro control. Each effect has a different set of controls that can be modulated, with dedicated **Control** parameters to adjust the individual modulation amount. Adjusting **Depth** globally increases or decreases the relative modulation amount across all **Control** parameters.

The **Dry** and **Wet** controls in the **Input Mix** section can be independently assigned to Macro controls by right-cicking them and using the context menu.

- **Control selector**: Selects the Macro control used for modulating the effect.
- **Depth**: Adjusts the global modulation amount applied from the selected Macro control. You can fine-tune the modulation amounts for individual effect controls using the respective **Control** parameters.
- **Smooth**: Applies smoothing to the modulation, creating soft transitions between different values, even if the modulation contains sharp edges and fast transitions.

Aetherizer

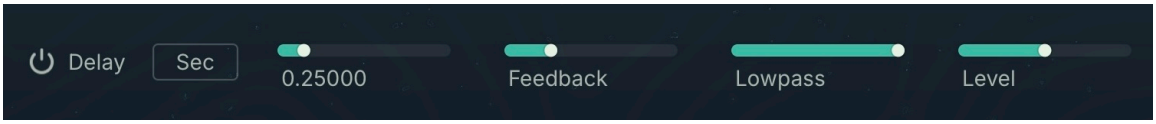
Aetherizer is a granular processor that allows you to process the signal on the level of small fragments of sound, called grains. You can use transposition, a feedback loop, and a bandpass or comb filter, before sending the signal to a delay including its own feedback loop.

Each grain is played back at a random position in the stereo field. When using the Surround Panner, each single grain is reproduced at its own surround position. Similarly, all processing on the grain level can be randomized to produce evolving and unpredictable results. You can also control all parameters (including the randomization parameters) using envelope, LFOs, or Macro controls.

Delay section

The Delay section adds another dimension to the grain clouds by applying echo effects to the output of the granular processing.

The Delay section contains the following controls:



- **Beat/Sec**: Switches the Delay Time control between time intervals (**Sec**) and musical beats in quarter notes (**Beat**).
- **Time**: Defines the post-delay duration.
- **Feedback**: Defines the proportion of the incoming signal that is sent to the post-delay's feedback loop (range: 0.00 to 99.99 %).
- **Lowpass**: Adjusts the cutoff frequency (in Hertz) of a lowpass filter within the feedback loop.
- **Level**: Adjusts the gain (in dB) of the delayed signal.

Grain section

The Grain section lets you adjust the Aetherizer's granular processing, allowing you to control the shape of the grain cloud.

> The **Jitter** and **Control** parameters allow for randomization and modulation of the individual controls. You can find more information in Jitter and Control.

The Grain section contains the following controls:

- **HD switch**: Activates or deactivates the High Density mode. In High Density mode the grain cloud can reach higher densities, which results in a both fuller and more polished sound. The High Density mode can also generate higher CPU loads.
- **Feedback**: Amount of grain-level feedback.
- **Predelay ms**: Defines the position in the incoming signal at which the grains are taken. Increasing the Predelay value makes the grain start later in the input signal.
- **Rate Hz**: Rate at which the input signal is chopped (in other words, it is the number of grains taken from the input signal during one second).
- **Duration**: Adjusts the length of each grain. This parameter influences the density of the sound: less duration means less overlapping grains, thus making the sound thinner.

ⓘ The actual grain duration also depends on **Rate Hz**, the faster the rate, the shorter the grains.

Tone section

The Tone section lets you adjust the tonal quality of the grains, allowing your to control the pitch and filtering of each grain.

ⓘ The **Jitter** and **Control** parameters allow for randomization and modulation of the individual controls. You can find more information in Jitter and Control.

The Tone section contains the following controls:

- **Transpose**: Adjusts the transposition of the grains (in semitones).
- **Filter On/Off**: Enables/disables the filter at the grain level.
- **Filter Mode**: Selects between two filter modes, Bandpass and and Comb.
- **Bandpass/Comb Hz**: Defines the cutoff frequency of the grain-level filter.
- **Resonance**l: Defines the resonance of the grain-level filter.

When using **Comb** mode, turn the down **Rate Hz** in the Grain section to make low pitches more audible. For example, if **Rate Hz** is set to 200 Hz and **Comb Hz** to 50 Hz, the grain is too short for the pitch to build up in the comb filter.

Filter Quantize quantizes the filter frequencies of the grains according to musical scales or formants. The following controls are available:

- **Filter Quantize Mode**: Selects the musical scale or **Formant** mode that will determine the filter frequencies of the individual grains.
- **Transpose**: Transposes the selected scale in semitones. This control is not available when **Vowel** mode is selected.

The quantization of the cutoff frequency can only happen if you allowed the cutoff frequency to “move” around the basic Frequency control value. You can achieve this by setting the Random Frequency control to its right at a non-zero value (see below for more details about the Random parameters).

Jitter and Control

In both the Grain and Tone sections each control can be randomized and modulated using the Macro control. By applying randomization and modulation, the grain cloud comes to live, creating interesting textures, layers, and movements.

The following parameters are available for each control:

- **Jitter**: Adjusts the amount of randomization applied to the respective control. At a value of 0, the control is not randomized and all grains share this same value. By turning up **Jitter**, the control is increasingly randomized and each grain is played back according to different parameter values.

- **Control**: Adjusts the amount of modulation applied from the Macro control selected in the Control section. You can find more information in Control section.

Display

The Aetherizer display shows you a visual representation of your grain cloud. The vertical axis is the pitch, the horizontal axis is the time. Each grain is represented by a small triangle followed by an horizontal trail (which represents the grain's length).

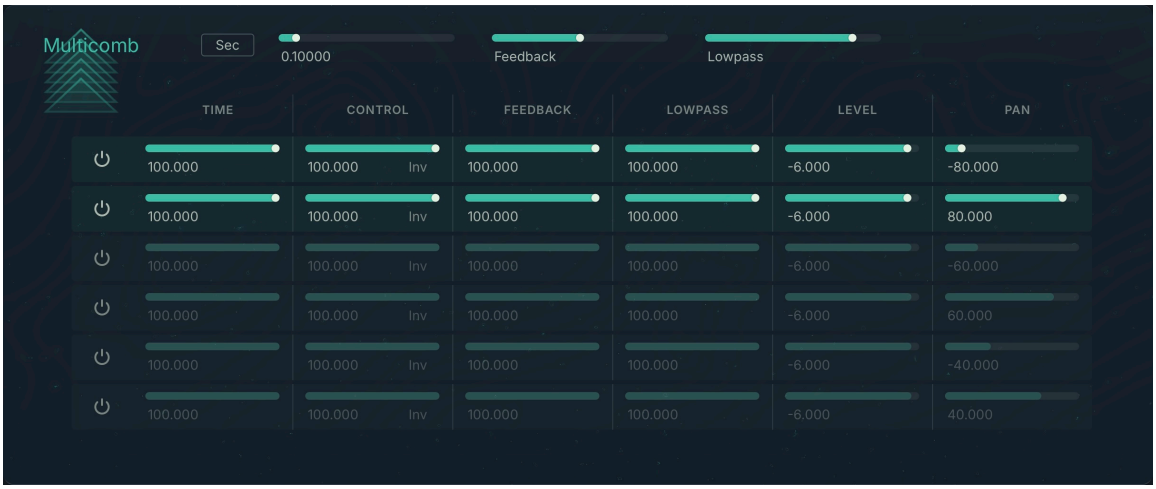
The visuals change according to the adjustments you make to the controls:

- **Duration** changes the trails' lengths.
- **Rate** changes the distance of the grains (and their length).
- **Transpose** moves grains on the y-axis.
- **Transpose Jitter** spreads grains across the y-axis.
- **Bandpass/Comb Hz** changes the color of grains according to the filter frequency.
- **Resonance** changes the intensity of the grains' color.

Multicomb

Multicomb specializes in modulation effects, offering up to six independent comb filters that use low-pass filters in the feedback loop. By applying modulation, you can easily achieve flanging and other effects based on phase shifts. For subtle to moderate phase modulations, choose medium values. High feedback rates produce tune-able resonance effects that sound like chords.

Multicomb contains the following controls:



- **Beat/Sec**: Switches the Time control between time intervals (**Sec**) and musical beats in quarter notes (**Beat**).
- **Time**: Determines the maximum delay of the feedback signal, i.e. the combfilter's frequency. A high value results in a long delay and thus a long vibration, i.e. a deep frequency.
- **Feedback**: Defines the maximum level of the feedback signal, as a percentage value related to the original signal's level. At high levels, the combfilter has a high resonance, and the sound's overtones produce a saw-tooth curve.
- **Lowpass**: Determines the cut-off frequency of the Lowpass filter, which is run through via the feedback signal (in Hz).

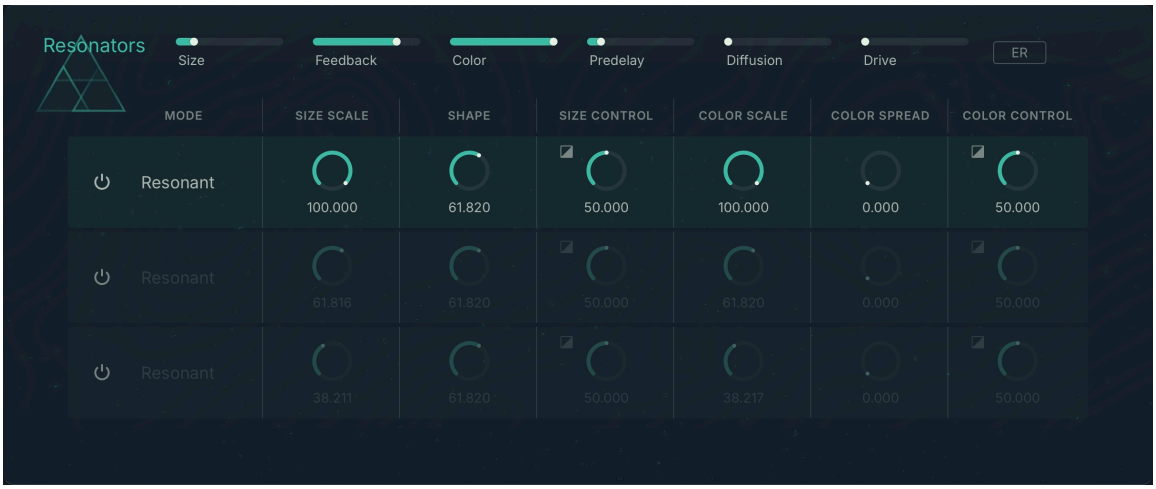
Each of the individual comb filters contains the following controls:

- **On/Off**: Allows each of the six comb filters to be turned on or off.
- **Time**: Defines the frequency (and hence the delay time) of the respective comb filter. This is measured in percentage, and correlates to the value you have selected in the Master Time control.
- **Control**: Adjusts the amount of modulation applied from the Macro control selected in the Control section to the **Time** control (in percent).
- **Inv**: If you turn on the Inversion switch, you invert the effect of the Macro Control's control signal (see the section on Modulation und Sensitivity control).
- **Feedback**: Establishes the feedback signals' level of every individual filter as a percentage, which correlates to the value you choose in the Master Feedback control.
- **Lowpass**: Controls the cut-off frequency of each Lowpass filter. This is measured in percent, which again correlates to the value set forth in the Master Lowpass control.
- **Level**: Determines the level of each respective comb filter signal at the output signal in dB.
- **Pan**: Positions the signal of the respective comb filter within the panorama of the output signal. The value 0 corresponds to the position on the extreme left and 1 corresponds to the position on the extreme right. 0.5 corresponds to the middle position.

Resonators

Resonators consists of three virtual resonant bodies. These resonators can produce echo and reverb effects from infinite spaces to shorter, brighter sounding rooms, or simulate the sound of specific objects such as metal bars, brake drums and bottles.

Resonators contains the following controls:



- **Size**: Sets the maximum delay-length between 0% and 100%.
- **Feedback**: Controls the feedback amount (from 0 to 100).
- **Color**: Controls the filters inside the resonators. The effect depends on which feature is selected in the Mode menu (see below).
- **Predelay**: Sets the delay in milliseconds, before the onset of the resonation effect.
- **Diffusion**: Creates random delay variations. A good adjustment will reduce how much the resonator "rings" less, and higher values make it sound grainier. Range: 0 to 100.
- **Drive**: Regulates the input level. This parameter determines the input signal's degree of saturation. If a high input level distorts the signal, the resonators react differently to when stimulated by a clean signal. This is a very powerful setting, since it also affects all other parameters.
- **ER**: Turns the "early reflections" on or off.

Each of the individual echo paths contains the following controls:

- **Mode**: Selects one of the following resonator modes.
  - **Clean**: Dampened delay. The sound flutters and is less diffuse
  - **Natural**: More diffuse variation. Sounds rather dark.
  - **Resonant**: More diffuse, brighter sound.
  - **Synthetic**: Like **Resonant**, just more extreme. The parameter Tone control (see above) produces very different effects depending on which operating mode you are using. Have an experiment with this parameter's extreme settings.
- **Size Scale**: The resonator size is the determined by the value set by the Master section's Size control multiplied by this setting.
- **Shape**: Allows you to adjust the various delay ratios. For a hall effect, changing this value would be like changing the space's volume or form.
- **Size Control**: Adjusts the amount of modulation applied from the Macro control selected in the Control section to the **Size** control (in percent). The Inversion switch inverts the sensitivity.
- **Color Scale**: The sounds created here are the result of a fixed Master Tone value multiplied by the setting selected in Tone Scale control.
- **Color Spread**: Increasing the value of this produces a more diffuse sound and also reduces feedback.
- **Color Control**: Adjusts the amount of modulation applied from the Macro control selected in the Control section to the **Color** control (in percent). The Inversion switch inverts the sensitivity.

Pipe

Pipe replicates the physical qualities of resonating bodies and resembles a so-called waveguide. Unlike waveguides based on physical modelling, Pipe's algorithm does not attempt to realistically simulate existing instruments or other natural resonating bodies. It is helpful to imagine Pipe as a kind of string or pipe. The string model using an exciter at a variable position of the string (**Input Position**) and two pickups, which can also be placed freely along the string (**Output Left Position**, **Output Right Position**).

You can modulate the string's length and the pickups' position using LFOs or a MIDI Controller. This way, various flanging, pitch-shifting and rotary speaker effects can be achieved. These effects are particularly apparent when the modulation of the pickups are modulated in opposite directions. LFO modulation applied to the output positions in opposite directions creates interesting panning effects. If the two points intersect, the output becomes mono.

When **Output left Position** or **Output right Position** cross **Input Position**, a muffled side tone can be heard. When modulating **Length**, any values crossing the output position can produce a muffled click. However, **Output left Position** and **Output right Position** can cross freely without artifacts.

Pipe contains the following controls:

- **Beat/Sec**: Switches the Master Length control between time intervals (**Sec**) and musical beats in quarter notes (**Beat**).
- **Length**: Determines the virtual string's maximum length. In terms of application this means: that's where the maximal delay can be set.
- **Feedback**: Determines the level of the feedback signal in percentage of the original level; at higher values, the virtual string resonates longer, at lower values it is silenced faster.
- **Lowpass:** Defines the cutoff-frequency of the Lowpass filter, through which the feedback signal passes (in Hz). It changes the string's sustain characteristic.
- **Control Delay**: Adjusts the amount of modulation applied from the Macro control selected in the Control section to the **Length** control (in percent).
- **Inversion**: If you turn on the Inversion switch, the effect of the control signal received via the Macro Control is inverted (see the section on Modulation und Sensitivity control).
- **Input Position**: Adjusts the position of the exciter on the string.
- **Output Left/Right Position**: Adjusts the position for the the pickup on the string.
- **Output Left/Right Control**: Adjusts the amount of macro control applied to **Position**.
- **Display**: Shows the string model along with the positions of the exciter (**In**) and the two pickups (**Out L**, **Out R**), including modulation.

Echoes

Echoes is tailored towards creating classic echo-effects. The input signal can be delayed individually in up to three delay paths, each with its own filter configuration including low-pass, high-pass and all-pass filters. The **Pan** controls enable the three delay paths' output signals to be separated in the stereo field. Each delay path can be directly mixed to the output using the **Gain** control. At the same time, it can be sent back to its own input using the **Feedback** control. An echo can lose high and low frequencies when sent through a filter and thus sound duller or more ethereal. The **Allpass** filter introduces phase shifts that produce interesting results, particularly using short delay times and when modulated by an envelope.

Echoes contains the following controls:

- **Beat/Sec**: Switches the Time control between time intervals (**Sec**) and musical beats in quarter notes (**Beat**).
- **Time**: Determines the maximum delay of the three taps.
- **Feedback**: Defines the feedback intensity.

Each of the individual echo paths contains the following controls:

- **Time**: Determines the delay of each echo (in percent) relative to the value set in Time control in the Master section
- **Feedback**: Sets the level of an echo's signal, which is routed back to the input, as a percentage of the original signal's level.
- **Control**: Adjusts the amount of modulation applied from the Macro control selected in the Control section to the **Time** control (in percent).
- **Lowpass**: Determines the cut-off frequency of the Lowpass filter that is applied to the signal in the feedback loop. The value in Lowpass control can range from 1.0 to 22050 Hz.
- **Highpass**: Determines the cut-off frequency of the Highpass filter that is applied to the signal in the feedback loop. The value in Highpass control can range from 1.0 to 22050 Hz.
- **Allpass**: Determines the cut-off frequency of an Allpass filter, which is similar to the Allpass 4 function in the Filter Module in the Patch Window. Allpass filters let all frequencies pass through, but change the signal's phase depending on the filter frequency. The Allpass filter produces its most impressive results when the parameters are modulated, thus creating phasing effects.
- **Level**: Defines the output signal's level of the respective echo (in dB). A value of -6dB reduces the signal level by half.

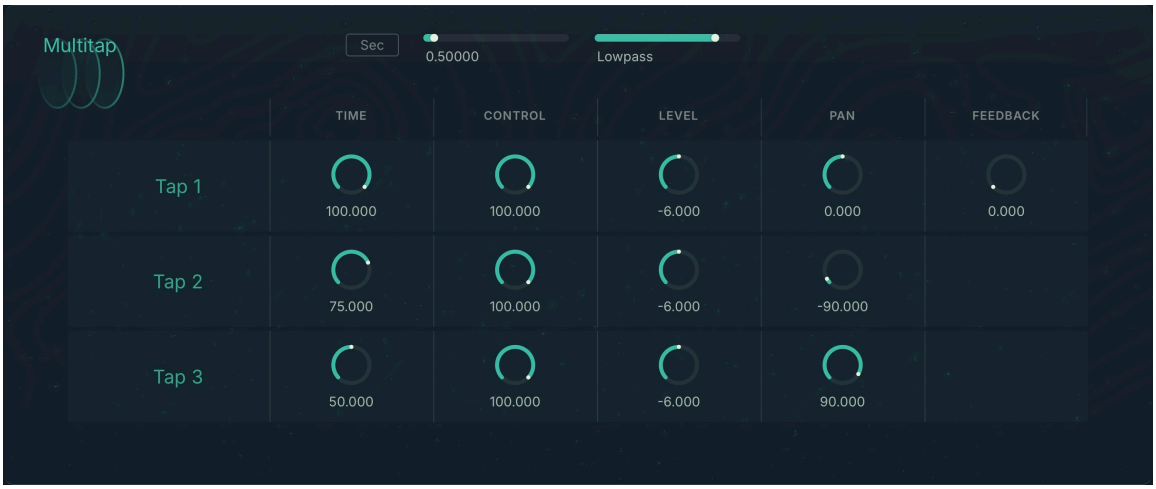
- **Pan**: Positions the respective echo signal in the panorama of the output signal. A 0 value corresponds to a position at the extreme left, 1 a position at the extreme right and 0.5 to the middle.

Multitap

Multitap is a simple delay with three taps. The input signal is delayed according to the time set for each tap, creating three copies of the input signal. Interesting effects can be achieved by routing the delayed signal back to the input after the first tap, producing dense delay clusters.

Try to shift the positions of the three taps during the tonal progression through an envelope or an LFO. By setting the modulation source on the three taps' Pan controls and feeding the Effect module with short, differentiated sounds, the signals will move around in a more powerful way.

Multitab contains the following controls:



- **Beat/Sec**: Switches the Time control between time intervals (**Sec**) and musical beats in quarter notes (**Beat**).
- **Time**: Determines the maximum delay of the three taps.
- **Lowpass**: Determines the cut-off frequency of the lowpass filter, which the feedback signal runs through (in Hz).

Each of the individual delay taps contains the following controls:

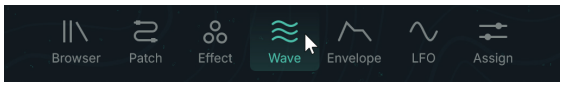
- **Time**: Determines the delay of each respective tap in percent of the value selected in the Time control in the Master section.
- **Control**: Adjusts the amount of modulation applied from the Macro control selected in the Control section to the **Time** control (in percent).
- **Level**: Defines the signal proportions for each tap at the output in dB. For example, a value of -6dB reduces the level by half.
- **Pan**: Positions the respective tap's signal in the panorama of the output signal. A 0 value corresponds to a position at the extreme left, 1 to a position at the extreme right and 0.5 to the middle.

- **Feedback**: This parameter exists only for the first delay tap. It regulates the level of the signal, which is re-routed to the input, as a percentage of the original signal's level.

8. Wave page

The Wave page of Absynth 6 provides intuitive tools to shape your user waveforms. User waveforms can be based on factory waveforms or created from scratch. You can use these waveforms in various places in the instrument, including many modules and the LFOs. User waveforms are saved with each preset and are available only in this preset. However you can reuse waveforms from other presets using copy/paste.

▶ To open the Wave page, click the **Wave** button at the top of the Absynth 6 window:



The Wave page opens automatically when you click **Edit** or **New** in a Waveform selector. Waveform selectors are available in various modules of the Patch page, as well as in the LFOs configured in the LFO page and Envelope page.

The Wave page contains the following areas:

1. **Preset Waveforms list**: Displays all the user waveforms in the preset. Each entry shows the waveform name and a thumbnail of its shape. Selecting an entry will open the waveform in the Waveform Editor on the right. User waveforms always have brackets around their name to easily distinguish them from factory waveforms in the Waveform selectors. By default, a user waveform takes the name of the waveform from which it is coming. You can rename it by double-clicking its entry and typing a new name from your computer keyboard.

   User waveforms are always created by starting from an existing waveform, as described in Waveform selector.

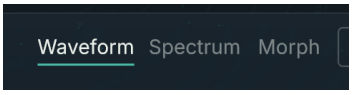
2. **Assigned To list**: Shows the component(s) using the selected waveform.

3. **Waveform Editor**: Displays and lets you modify the selected waveform. You can find more information in Waveform Editor.

Waveform Editor

The Waveform Editor is the main area of the Wave page. It lets you adjust the user waveform selected in the **PRESET WAVEFORMS** list on the left. The Waveform Editor provides different views that focus on different aspects of the waveform.

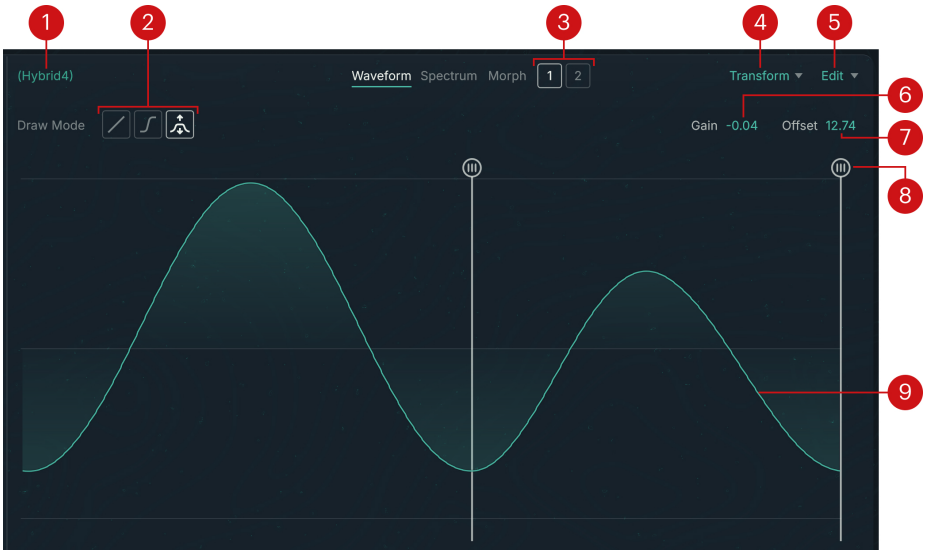
At the top of the Waveform Editor, the Waveform View selector lets you switch between the following views:



- **Waveform**: Shows the Waveform view, which lets you edit the waveform in the time domain.
- **Spectrum**: Shows the Spectrum view, which lets you edit the waveform in the frequency domain.
- **Morph (only available for Morph waveforms)**: Shows the Morph view, where you can configure the morphing between both waveforms.

Waveform view

The Waveform view lets you shape the selected waveform in the time domain. It contains the following controls:

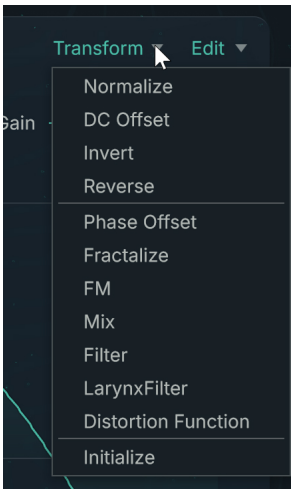


1. **Waveform name**

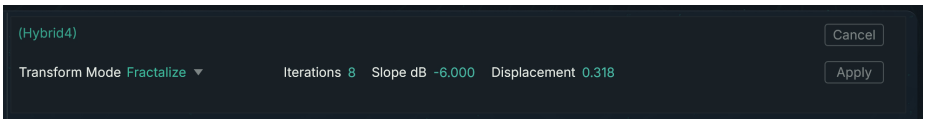
2. **Draw Mode selector**: Switches between three graphical modes for shaping the waveform with your mouse. The available modes are, from left to right:
   - **Line mode**: Builds the waveform with straight lines. The line starts from the Anchor point (marked by a vertical bar through the waveform display) and ends where you click. At every click, the Anchor point jumps to the clicked position and you can directly shape the next segment from there.
   - **Curve mode**: Works like the Line mode but inserts semi-cosine curves instead of straight lines. This results in smoother transitions between segments, which can help attenuate harsh high frequency contents.
   - **Stretch mode**: When you select this mode, a second Anchor point appears in the waveform display. Clicking and dragging your mouse vertically lets you expand or compress the part of the waveform between the Anchor points. You can drag the handles at the top of the bars to select another part of the waveform.
3. **1/2 switch (Morph waveforms only)**: Selects either of the base waveforms for editing.
4. **Transform menu**: Lets you apply various transformations to the waveform. You can find more information in Transform menu (Waveform view).
5. **Edit menu**: Provides copy/paste commands for waveforms. Selecting **Copy Waveform** stores the current waveform into the clipboard. You can then select **Paste Waveform** in another waveform to replace it with the copied waveform.
6. **Gain**: Adjusts the waveform's amplification level in dB.
7. **Offset**: Adjusts the amplitude offset of the waveform relative to the x-axis (0 dB). With **Offset** at 0 dB, the highest and lowest points of the waveform are at the same distance from the x-axis.
8. **Anchor point**: Defines a reference point on the waveform when using one of the drawing modes. Line mode and Curve mode use one Anchor point, while Stretch mode uses two. You can move the Anchor point(s) by dragging the top handle horizontally with your mouse.
9. **Waveform shape**: Represents the current waveform. Any edits that you make using the mouse, the functions from the Transform menu, or the copy/paste commands, will be directly mirrored in the display and in the resulting sound.

Transform menu (Waveform view)

The **Transform** menu lets you apply various transformations to the displayed waveform. In the menu, the transformations are split into two main groups:



The transformations from the upper group are directly applied to the waveform, whereas the transformations from the lower group open a dedicated **Transform panel** with additional settings at the top of the page:



In the Transform panel the following elements are common to all transformations:

- **Transform Mode menu**: Switches to another transformation.
- **Cancel**: Closes the panel and leaves the original waveform untouched.
- **Apply**: Transforms the original waveform according to the panel settings and closes the panel.

The other parameters in the Transform panel depend on the selected transformation. As you change the parameter values, the waveform display and the resulting sound are updated in real time to show you the effect of your actions. You can then choose to apply these changes or to cancel them.

The **Transform** menu of the Waveform view provides the following transformations:

Normalize: Normalizes the waveform amplitude. The waveform's minimum and maximum values are adjusted to the maximum value range.

DC Offset: Adjusts the amplitude offset of the waveform so that the waveform's energy will be evenly distributed between its positive and negative sections, which can help avoid unwanted audio artifacts. This is different from centering the waveform by setting the **Offset** control to zero.

Invert: Flips the waveform vertically (effectively inverting the waveform's phase).

Reverse: Flips the waveform horizontally.

Phase Offset: Lets you adjust the phase of the waveform. In the Transform panel that opens, the **Offset** parameter adjusts the phase offset.

Fractalize: Adds smaller copies of the waveform to the waveform itself, so that parts of the waveform become similar to the shape of the whole. You can use it to add overtones to the waveform. This allows for extraordinary waveform manipulations and is particularly good for creating powerful, organic waveforms. The Transform panel opens and provides the following parameters:

- **Iterations**: Selects how many times the waveform is copied to itself, which affects the quantity of details added to the original waveform. The value range is between 2 and 8. Higher values lead to more complex, lighter-sounding sounds.
- **Slope dB**: Adjusts the mixing relationship between the original waveform and its copies. This controls the fractalization's intensity: The higher the value, the richer the sound in overtones and noise.
- **Displacement**: Defines the shift of the copied area relative to the original waveform: 0 corresponds to a position before the waveform, 1 to a position after the waveform. With a value of 0.5, the area is in the center of the waveform. Modulating this parameter with an LFO or an envelope can produce interesting movements inside the sound.

FM: Applies frequency modulation. The current waveform acts as the carrier. The Transform panel opens and provides the following parameters:

- **Modulation waveform**: Shows the name of the waveform used as modulator. By default the **Sine** waveform is selected. Clicking the name opens a Waveform selector where you can choose another modulation waveform.
- **Amount**: Determines the intensity of the frequency modulation.

- **Main Freq**: Defines the denominator of the frequency ratio between modulator and carrier.
- **Mod Freq**: Defines the numerator of the frequency ratio between modulator and carrier.
- **Phase**: Adjusts the phase of the modulator relative to the carrier.

Mix: Mixes the waveform with another one. In the Transform panel that opens, your original waveform is labeled **A** and the additional waveform is labeled **B**. The panel provides the following parameters:

- **Mix waveform**: Shows the name of the waveform mixed with the original one. By default the **Sine** waveform is selected. Clicking the name opens a Waveform selector where you can choose another waveform.
- **Mix A**: Adjusts the level of the original waveform.
- **Mix B**: Adjusts the level of the additional waveform.
- **Freq Ratio B**: Adjusts the frequency ratio between the additional waveform and the original waveform.
- **Phase B**: Adjusts the phase of the additional waveform relative to the original waveform.
- **Phase Invert switch (phase symbol)**: Inverts the phase of the additional waveform.

Filter: Applies a filter to the waveform. The Transform panel opens and provides the following parameters:

- **LP/HP/BP**: Selects between low-pass (**LP**), high-pass (**HP**), and band-pass (**BP**) filtering.
- **Frequency**: Adjusts the cutoff frequency of the filter.
- **Resonance**: Adjusts the resonance at the cutoff frequency.

Larynx Filter: Simulates the resonant frequencies of the human vocal tract as it produces different vowels. Each vowel has its own set of characteristic frequencies, called formants. The Transform panel opens and provides the following parameters:

- **Vowel**: Selects from different vowel models, each having a distinct set of formants.
- **Resonance**: Adjusts the global resonance of the filter.
- **Shift**: Adjusts the frequencies of all the formants simultaneously while maintaining the frequency ratios between the formants. This modifies the perceived timbre of the voice without affecting the global pitch of the vowel.

Distortion Function: Applies an adjustable distortion function to the waveform. The Transform panel opens and provides the following parameters:

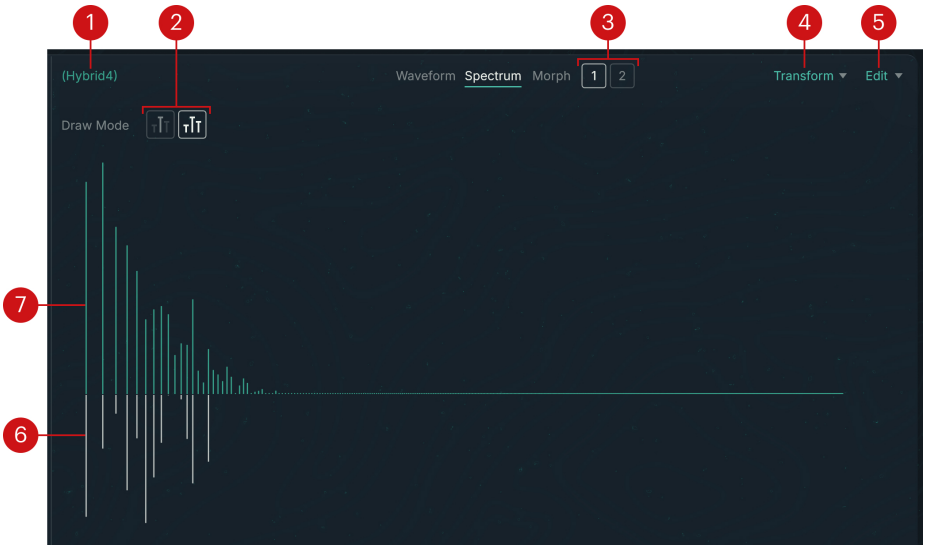
- **Amount**: Adjusts the distortion strength.
- **Asymmetry**: Adjusts a distinct distortion for the positive values and for the negative values. At the minimum value 0, the distortion is the same on both sides. At the maximum value 1, only one side is distorted.
- **Mix**: Adjusts the balance between the original signal and the distortion output.
- **Phase Invert**: Inverts the phase of the distortion function.

Initialize: Resets the waveform to zero.

Spectrum view

The Spectrum view of the Wave page lets you shape the selected waveform in the frequency domain. It shows the first 512 harmonics of the selected waveform as a series of bars: The colored bars in the top half show the harmonics' amplitudes, and the white bars in the lower half show their phases. The fundamental frequency, also called first harmonic, is represented by the first pair of colored and white bars at full left, followed on its right by the higher harmonics in ascending order.

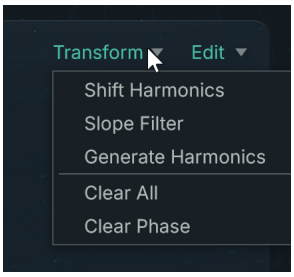
The Spectrum view contains the following controls:



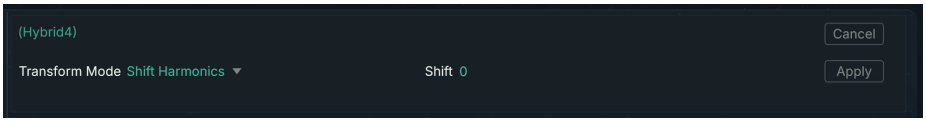
1. **Waveform name**
2. **Draw Mode selector**: Switches between two graphical modes for editing the harmonics with your mouse. The available modes are, from left to right:
   - **Single Harmonic mode**: In this mode you can click a bar and drag your mouse to adjust the height of this particular bar only. The horizontal movements of the mouse have no effect, and its vertical movements are applied relative to the bar's current height. This mode can be used for precise manipulations.
   - **Multi Harmonics mode**: In this mode you can click anywhere in the display and drag your mouse to draw the amplitudes or the phases of multiple harmonics in one go. As you drag the mouse, each bar under the mouse cursor will be set to the exact height of the cursor. Depending on whether you first clicked in the upper half or in the lower half of the display, your mouse movements will affect only the amplitudes or only the phases of the harmonics.
3. **1/2 switch (Morph waveforms only)**: Selects either of the base waveforms for editing.
4. **Transform menu**: Lets you apply various transformations to the spectrum. You can find more information in Transform menu (Spectrum view).
5. **Edit menu**: Provides copy/paste commands for waveforms. Selecting **Copy Waveform** stores the current waveform into the clipboard. You can then select **Paste Waveform** in another waveform to replace it with the copied waveform.
6. **Phase bars**: The white bars in the lower half of the display represent the phases of the harmonics. The zero is on the middle line and the phase values grow downwards.
7. **Amplitude bars**: The colored bars in the upper half of the display represent the amplitudes of the harmonics. The zero is on the middle line and the amplitude values grows upwards.

Transform menu (Spectrum view)

The **Transform** menu lets you apply various transformations to the displayed spectrum. In the menu, the transformations are split into two groups:



The transformations from the lower group are directly applied to the spectrum, whereas the transformations from the upper group open a dedicated **Transform panel** with additional settings at the top of the page:



In the Transform panel the following elements are common to all transformations:

- **Transform Mode menu**: Switches to another transformation.
- **Cancel**: Closes the panel and leaves the original spectrum untouched.
- **Apply**: Transforms the original spectrum according to the panel settings and closes the panel.

The other parameters in the Transform panel depend on the selected transformation. As you change the parameter values, the spectrum display and the resulting sound are updated in real time to show you the effect of your actions. You can then choose to apply these changes or to cancel them.

The **Transform** menu of the Spectrum view provides the following transformations:

- **Shift Harmonics**: Shifts all the harmonics to the left or to the right. The Transform panel opens with a single **Shift** parameter, which adjusts how far the harmonics will be shifted. Negative values of **Shift** will move the harmonics to the left (lower frequencies), positive values will move them to the right (higher frequencies). The available values range from -64 to 64.
- **Slope Filter**: Progressively attenuates or amplifies the harmonics on either side of a particular harmonic called **crossover harmonic**. The attenuation/amplification of the other harmonics grows with their distance to crossover harmonic. You can adjust each side independently. The Transform panel opens and provides the following parameters:
  - **Crossover**: Specifies the harmonic used as central point. The remaining harmonics will be amplified or attenuated on either side independently.
  - **Low dB per Octave**: Adjusts the slope of the attenuation or amplification for the harmonics lower than the crossover harmonic (on its left in the display).
  - **High dB per Octave**: Adjusts the slope of the attenuation or amplification for the harmonics higher than the crossover harmonic (on its rightin the display).

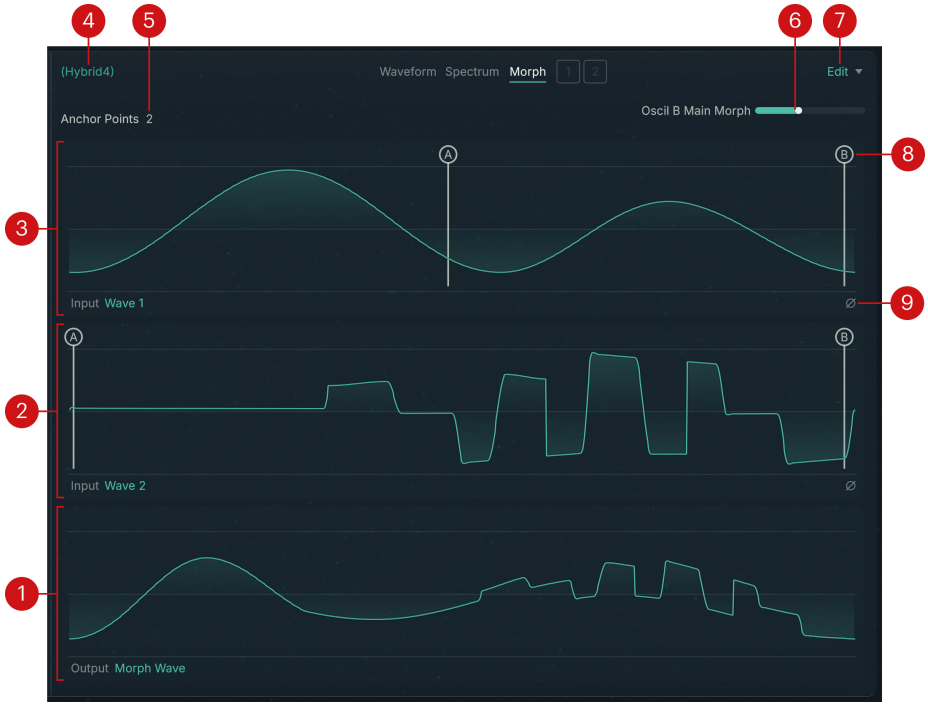
- **Generate Harmonics**: Creates a series of harmonics with a decreasing slope. The Transform panel opens and provides the following parameters:
  - **Slope dB**: Adjusts how fast the amplitudes decrease in the series.
  - **Balance**: Adjusts the mix between the original spectrum (**Balance** set to 0) and the generated series (**Balance** set to 1).
  - **Series Mult**: Includes only specific multiples in the generated series: For example, setting **Series Mult** to 2 will generate only the 2nd, 4th, 6th, (…) harmonics, while setting it to 3 will generate only the 3rd, 6th, 9th, (…) harmonics.
  - **Series +/-**: Adjusts the starting harmonic in the series. Changing this value results in shifting the entire series to the next or previous harmonics. For example, this can be useful to bring back the fundamental into a series including only specific multiples.
- **Clear All**: Removes all the harmonics from the spectrum except the first one (the fundamental frequency).
- **Clear Phase**: Clears all the phase bars from the spectrum while leaving the amplitude bars untouched. This aligns the phases of all harmonics.

Morph view

The Morph view of the Wave page is available only when a morph waveform is selected in the **PRESET WAVEFORMS** list on the left. The Morph view lets you adjust the waveform's morphing in detail.

A morph waveform allows you to dynamically blend two waveforms into one. You can edit each of the base waveforms using the tools in the Waveform view and Spectrum view, and configure how they are morphed using the Morph view.

The Morph view contains the following elements:



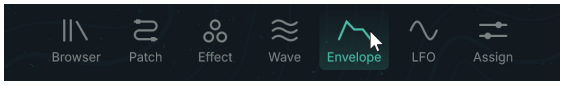
1. **Morph Wave display**: Shows the waveform shape resulting from the morphing. The shape changes as you move the **Morph** slider at the top right, and when you modify the position of the various Anchor points in the **Wave 1** and **Wave 2** waveforms above.

2. **Wave 2 display**: Shows the shape of the second base waveform.
3. **Wave 1 display**: Shows the shape of the first base waveform.
4. **Waveform name**
5. **Anchor Points menu**: Selects the number of Anchor points used by the morphing. You can choose between 2, 3, and 4 Anchor points. These Anchor points will appear as vertical bars labeled **A**, **B**, **C**, and **D** in the **Wave 1** and **Wave 2** displays.
6. **Morph slider**: Adjusts the morphing ratio between the two base waveforms. With the slider at full left, the morph wave mirrors Wave 1. With the slider at full right, the morph wave mirrors Wave 2. In other words, the **Morph** parameter describes the position of the morph waveform on the way from Wave 1 to Wave 2.
7. **Edit menu**: Provides copy/paste commands for waveforms. Selecting **Copy Waveform** stores the current waveform into the clipboard. You can then select **Paste Waveform** in another waveform to replace it with the copied waveform.
8. **Anchor points**: These points define references for the morphing. Each Anchor point in Wave 1 is linked to the point with the same letter in Wave 2, so that the segment between two Anchor points, for example A and B, in Wave 1 will be morphed into the segment between the same Anchor points A and B in Wave 2. You can move the Anchor points by dragging their letter horizontally with your mouse. Matching different segments from the two base waveforms can produce interesting effects in the morph waveform and makes morphing a subtle design tool.
9. **Phase Invert button (phase symbol)**: Inverts the phase of that waveform.

9. Envelope page

The Envelope page of Absynth 6 lets you design advanced envelopes for modulating the parameters of the instrument.

▶ To open the Envelope page, click the **Envelope** tab in the Header:



The Envelope page contains the following areas:

1. **Envelope list**: Contains all the envelopes used in the preset. You can select the desired envelopes from the list to open them in the Envelope Editor on the right. The Envelope list lets you also show only specific envelopes, and create new envelopes. You can find more information in Envelope list.
2. **Envelope and breakpoint settings**: Provides global settings and commands for the envelope under focus in the Envelope Editor. If a breakpoint is selected, the right part shows the settings for that particular breakpoint. You can find more information in Envelope and breakpoint settings.
3. **Envelope Editor**: Shows and lets you visually edit the selected envelopes and their breakpoints. You can find more information in Envelope Editor.
4. **Master Envelope Macros**: The four Envelope Macro knobs can be freely assigned to breakpoints, which makes these breakpoints available for MIDI control. You can find more information in Envelope Macros.

Envelope list

The Envelope list contains all the envelopes used in the preset.

In this list, the envelope names describe the modulation targets: category of module (Oscil, Filter, Mod, Chan, etc.), channel and position (A, A2, B1, C, M1, etc.), and target parameter.

Selecting an envelope from the list will show it in the Envelope Editor on the right. Multiple envelopes can be selected simultaneously: They will be stacked in the Envelope Editor. You can select multiple contiguous entries by holding down the [Shift] key and selecting the upper entry and the lower entry from the list. You can select multiple non-contiguous entries by holding down the [Command] key (macOS) or the [Ctrl] key (Windows) and selecting the desired entries from the list.

The Envelope list provides two additional controls at the top:

- **New Envelope button**: Creates a new envelope to modulate a new target parameter. You can find more information in Creating a new envelope.
- **View menu**: Offers a quick way to select specific groups of envelopes and show them in the Envelope Editor. The **View** menu includes the following display options:
  - **All**: Selects all the envelopes from the list.
  - **None**: Clears the current selection.
  - **Channel A–C**: Selects all the envelopes that belong to the modules of channel **A**–**C**, respectively.
  - **Master Channel**: Selects all the envelopes that belong to the modules of the Main channel.
  - **All Engines**: Selects all the envelopes that belong to the source modules.
  - **All Filters**: Selects all the envelopes that belong to the Filter modules.
  - **All Modulators**: Selects all the envelopes that belong to the Modulator modules.
  - **All Waveshapers**: Selects all the envelopes that belong to the Waveshaper modules.
  - **All Effects**: Selects all the envelopes that belong to the effect module.

Creating a new envelope

You can create a new envelope modulation by using either of the following two methods:

Method 1: Parameter's context menu

▶ In the appropriate page of Absynth, navigate to the target parameter that you want to modulate, right-click it (Windows) or [control]-click it (macOS), and select **Create a New Envelope…** from the context menu.

→ Absynth automatically switches to the Envelope page with the new envelope selected and ready for editing.

Method 2: New Envelope dialog

1. In the Envelope page, click the **New Envelope** button at the top of the Envelope list.

→ The New Envelope dialog opens.

2. In the mini patch view on the left, select the module containing the parameter that you want to modulate.

→ The list on the right shows all the module's parameters available for modulation.

3. Select the desired parameter from the list.
4. At the bottom of the dialog, click **Add** to create the new envelope.

→ The dialog closes and a new envelope automatically opens in the Envelope Editor, replacing any envelope(s) previously displayed.

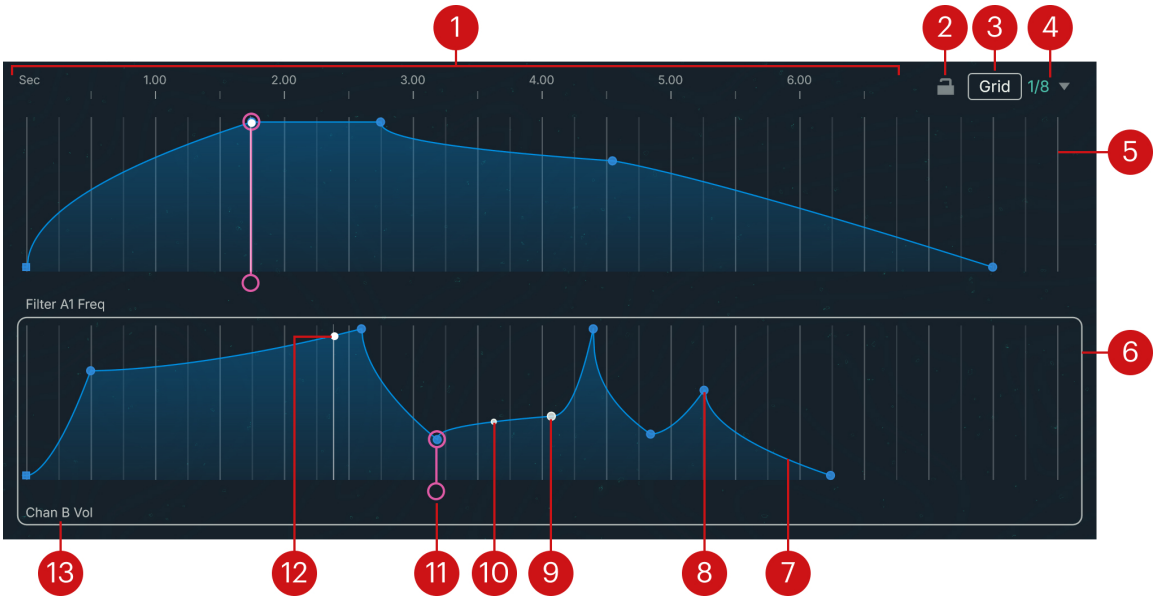
When the New Envelope dialog is open, you can also use the following buttons at the bottom of the dialog:

- **Pin**: When **Pin** is on, the dialog stays open when you click the **Add** button. This allows you to quickly create envelopes for multiple parameters in a row, and edit them at a later stage.
- **Cancel**: Closes the dialog without creating any envelope.

Envelope Editor

The Envelope Editor displays the envelopes selected in the Envelope list on the left, and lets you visually create, edit, and delete breakpoints with the mouse.

The Envelope Editor contains the following elements:



1. **Timeline**: Measures the time from the beginning of the envelope, in seconds. All the displayed envelopes use the same timeline, which helps shape the envelopes in relation to each other. You can click anywhere in the timeline and drag your mouse vertically to adjust the zoom factor and horizontally to scroll in the timeline.
2. **Lock button**: When the Lock button is off (default setting), as you drag a breakpoint all the following breakpoints move in the timeline accordingly to maintain the current time intervals between the breakpoints. If you activate Lock, the positions of the following breakpoints are locked, and moving a breakpoint in the timeline only changes the position of this breakpoint. Lock can be very useful for rhythmical envelopes where the breakpoints should stay synchronized with the beats.
3. **Grid button**: Activates or deactivates the grid. When the grid is active, gray vertical lines are displayed at regular intervals. As you drag a breakpoint closer to a grid line, the breakpoint will snap to the line. You can use it to quickly set perfect timings across different stages in the envelope or across multiple envelopes.

   ⓘ When the grid is active, you can temporarily deactivate snapping by [Shift] +dragging the breakpoint.

4. **Grid Size menu**: Adjusts the size of the grid intervals. The available values are 1/8th, 1/16th, or 1/32nd of a second.
5. **Grid line**: The grid lines are visible only when the **Grid** button is on.
6. **Focused envelope**: The white frame indicates the envelope under focus. At the top of the page, the envelope and breakpoint settings always relate to the focused envelope. You can click anywhere in another envelope image or its background to switch the focus onto it.
7. **Envelope shape**: Represents the envelope's amplitude values across the timeline. The shape is made of a series of breakpoints and transitions between them. When an envelope modulates a parameter, the value set in the Patch page represents the maximal value that the envelope can reach. The envelope will reduce this value according to its shape.

8. **Breakpoint**: Defines a target point in the envelope, for which you can precisely adjust the time and amplitude values. The transition between the previous breakpoint and this breakpoint will be interpolated according to the breakpoints' positions and the chosen **Slope** value. You can create a breakpoint by double-clicking anywhere in the envelope display, and delete a breakpoint by double-clicking it. You can find mode information on creating and deleting breakpoints in Creating and deleting breakpoints.
9. **Selected breakpoint**: The white dot indicates the selected breakpoint. You can drag the breakpoint with your mouse to change its position. The parameters of the selected breakpoint are displayed in the breakpoint settings above the Envelope Editor.
10. **Slope handle**: You can drag this handle with your mouse to adjust the curve of the transition before the selected breakpoint. The corresponding **Slope** parameter is available in the **BP Settings** panel of the breakpoint settings above the Envelope Editor.

    ⓘ The slope handle is not available if the transition is in step mode. You can find more information about the **Slope**/**Step** control in Breakpoint settings: BP Settings panel.

11. **Sustain/Release marker**: Sets the breakpoint as the sustain or release point of the envelope. The exact function of the marker depends on the envelope mode. You can move the marker onto the desired breakpoint by clicking. the round handle at the bottom and dragging your mouse horizontally. Any breakpoint can be set as sustain/release point except the first and the last breakpoints.
12. **Playback position**: The white vertical line with the white dot running on the envelope shape represents the current playback position.
13. **Envelope name**: Mirrors the name of the parameter modulated by the envelope.

Creating and deleting breakpoints

The same command lets you create or delete breakpoints depending on whether you use it on an existing breakpoint or not.

To create a breakpoint in an envelope:

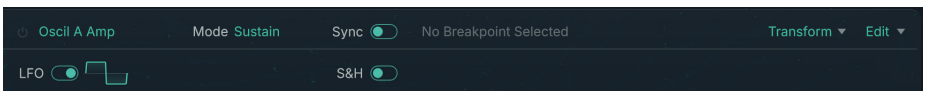
▶ Double-click anywhere in the envelope display to create a breakpoint at that position. You can also [Ctrl]-click (Windows) or [command]-click (macOS) anywhere in the envelope display.

To delete a breakpoint from an envelope:

▶ Double-click a breakpoint in the envelope display to delete it from the envelope. You can also [Ctrl]-click the breakpoint (Windows) or [command]-click the breakpoint (macOS).

Envelope and breakpoint settings

At the top of the Envelope page, two rows of **envelope settings** provide parameters and commands to adjust the envelope under focus in the Envelope Editor.



If a breakpoint is selected in the Envelope Editor, the right part of the envelope settings shows additional controls for the breakpoint. These controls are described in Breakpoint settings.

The **upper row** of settings contains the following elements, from left to right:

- **Envelope on/off switch**: Activates or deactivates the envelope. This control is not available for the amplitude envelopes of the source channels (labeled **Oscil A–C Amp**).
- **Envelope name**: Indicates the target parameter of the envelope modulation.
- **Mode menu**: Selects from different envelope modes that define the envelope behavior. You can find more information in Envelope modes.
- **Sync switch** (Release, Sustain, and Loop modes only): When **Sync** is on, the envelope is synchronized with the tempo of your DAW.
- **Beat control** (Retrigger mode only): Adjusts the delay (in beats) after which the envelope is retriggered.
- **Transform menu**: This menu is visible only when no breakpoint is selected in the Envelope Editor. The **Transform** menu lets you apply transformations to the envelope. You can find more information in Transform menu (Envelope settings).
- **Edit menu**: This menu is visible only when no breakpoint is selected in the Envelope Editor. The **Edit** menu provides the following editing commands for the envelope:
  - **Copy Envelope**: Stores the envelope into the clipboard.
  - **Paste Envelope**: Inserts the envelope from the clipboard, replacing the existing envelope.
  - **Initialize**: Resets the envelope to the default envelope, which contains four breakpoints all set to the maximum amplitude.
  - **Delete**: Deletes the envelope and its modulation.

For envelopes in Release, Sustain, Loop, or Retrigger mode, the **lower row** of envelope settings lets you configure an internal LFO to modulate the envelope. You can find more information in Envelope LFO.

Envelopes in Control Driven mode and in Link mode provide additional parameters described in Control Driven mode and Link mode, respectively.

Envelope modes

The **Mode** menu of the envelope settings lets you choose from the following envelope modes:

- The Release, Sustain, Loop, and Retrigger modes are based on the usual envelope concept in which the envelope modulates the parameter value over time.
- In Control Driven mode you can choose a Macro or MIDI control to set the position in the envelope instead of following the time.
- In Link mode the envelope can mirror the shape of another envelope.

Release mode

With an envelope in Release mode, when you play a note the envelope runs from the first to the last breakpoint in one go. If you release the note before the Sustain/Release marker has been reached, the envelope jumps to the marker and continues from there to the end. The Release mode is useful for percussive sounds, and for piano-like sounds with sustain.

If the Sustain/Release marker is located near the beginning, the Release envelope will always run in the same way, no matter how long the note is held.

Sustain mode

In Sustain mode the envelope works like a classic ADSR envelope. When you play a note, the envelope runs until the Sustain/Release marker and stays at the marker position as long as you keep the note held. When you release the note, the envelope runs on the remaining path to the end. If you release the note before the Sustain/Release marker has been reached, the envelope jumps to the marker and continues from there to the end.

Loop mode

With an envelope in Loop mode, two markers let you define the loop start and end points. When you play a note, the envelope runs until the Loop End marker, jumps back to the Loop Start marker and repeats the looped section as long as you keep the note held. When you release the note, the envelope jumps to the Loop End marker and runs and continues from there to the end.

Retrigger mode

In Retrigger mode, when you play a note the envelope restarts at regular intervals as long as you keep the note held. You can adjust the retrigger interval by using the **Beat** control in the envelope settings. If the envelope reaches the Sustain/Release marker before being retriggered, it stays at the marker position until it is retriggered. When you release the note, the envelope jumps to the Sustain/Release marker and continues from there to the end. The Retrigger mode is especially useful for creating rhythmic, looped paths.

Control Driven mode

While an envelope in any other mode is controlled by the time, in Control Driven mode the movement on the horizontal axis is controlled by a Macro. The MIDI value 0 corresponds to the beginning of the envelope, 64 to its midpoint, and 127 to its end.

For example, if you draw a Control Driven envelope containing only two breakpoints connected by a straight line, the Macro will control the envelope parameter in a proportional (linear) way in the range between the values of the two breakpoints. By drawing more complex shapes you can create advanced non-linear controls for your parameters. You will find many examples of Control Driven envelopes in the factory presets.

> ⓘ The Control Driven mode is not available for the amplitude envelopes of the source modules (labeled **Oscil A–C Amp** in the instrument).

In Control Driven mode the envelope provides one dedicated setting:

- **Control**: Selects the control assigned to the envelope. The available controls are the **Macros 1–16**, **Mod Wheel**, **Aftertouch**, **Sustain Pedal**, and **Velocity**.

Link mode

The Link mode lets you connect the edited envelope to another envelope (called the “source” envelope). In Link mode, all the changes made to the source envelope will also affect the edited envelope. Even though you cannot directly modify the breakpoints of the edited envelope, you can control how it will translate the source envelope by adjusting the available scaling parameters.

> ⓘ The Link mode is not available for the amplitude envelopes of the source modules (labeled **Oscil A–C Amp** in the instrument).

In Link mode the envelope provides the following settings:

- **Source menu**: Selects the source envelope.
- **Time %**: Scales the time intervals between breakpoints as a percentage of the intervals in the source envelope. This makes the envelope faster or slower. A value of 50 % will double the speed of the source envelope, a value of 200 % will halve it.
- **Amount %**: Scales the amplitudes of the breakpoints as a percentage of the amplitudes in the source envelope. This scaling adjusts the overall depth of the envelope modulation compared to the source envelope.
- **Offset**: Adjusts an offset globally added to the amplitudes of the breakpoints. Negative offset values will decrease the amplitudes.
- **Slope %**: Scales the slopes of the transitions between breakpoints as a percentage of the slopes in the source envelope. Scaling down the slopes makes the transitions start faster and end more softly, and vice versa.

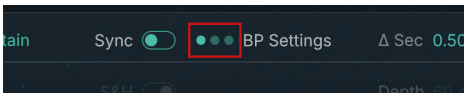
The Link mode not only saves time when editing similar envelopes, but can also serve for creative purposes. You could, for example, link the cutoff frequencies of the filters on multiple channels and choose slightly different time or slope scaling values to create a more organic sound.

Breakpoint settings

The breakpoint settings appear in the right half of the envelope settings as soon as a breakpoint is selected in the envelope display.

> The breakpoint settings are not available for envelopes in Link mode, these envelopes reuse instead the settings of another envelope.

The breakpoint settings are organized into three panels that you can successively show by clicking the **three little green dots** or the panel label on their right:

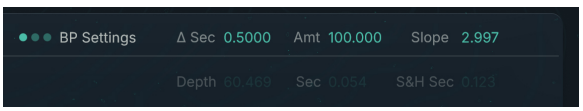


the following panels are available: BP Settings, BP Macros, and Assign BP To.

Breakpoint settings: BP Settings panel

The **BP Settings** panel lets you numerically adjust the position of the breakpoint and the slope of the previous transition. Your adjustments in this panel are equivalent to dragging the breakpoint or the slope handle with your mouse in the envelope display of the Envelope Editor.

The panel contains the following settings:

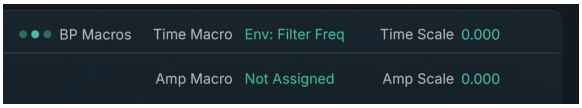


- **Δ Sec / Sec**: Adjusts the breakpoint's time value (horizontal axis). Clicking the label lets you switch between **Δ Sec** and **Sec**. **Δ Sec** adjusts the time of the breakpoint relative to the previous breakpoint, while **Sec** adjusts the time of the breakpoint relative to the beginning of the envelope.

- **Amt / [Unit]**: Adjusts the breakpoint's amplitude value (vertical axis). Clicking the label lets you switch between **Amt** and the particular unit (for example, **Hz** or **dB**) of the modulated parameter. **Amt** adjusts the amplitude as a percentage of the original parameter value. **[Unit]** adjusts the amplitude as an absolute value; in this case the value range might vary with the parameter.
- **Slope / Step**: Adjusts the shape of the transition before the breakpoint. Clicking the label lets you switch between **Slope** and **Step**. **Slope** adjusts the curve of the transition: A value of 1 corresponds to a straight line; for values between 0 and 1 the transition starts fast and ends slowly; for values above 1 the transition starts slowly and ends fast. If you switch to **Step**, the envelope will keep the value of the previous breakpoint until it reaches this breakpoint.

Breakpoint settings: BP Macros panel

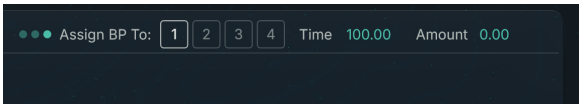
The **BP Macros** panel lets you use Macros to control the position of the breakpoint. The panel contains the following settings:



- **Time Macro**: Selects the control assigned to the breakpoint's time (horizontal position). The available controls are the **Macros 1–16**, **Mod Wheel**, **Aftertouch**, **Sustain Pedal**, and **Velocity**. Selecting **Not Assigned** deactivates the time control.
- **Time Scale**: Adjusts the range of the Macro control relative to the original time value. Negative **Time Scale** values invert the control.
- **Amp Macro**: Selects the control assigned to the breakpoint's amplitude (vertical position). The available controls are the **Macros 1–16**, **Mod Wheel**, **Aftertouch**, **Sustain Pedal**, and **Velocity**. Selecting **Not Assigned** deactivates the amplitude control.
- **Amp Scale**: Adjusts the range of the Macro control relative to the original amplitude value. Negative **Amp Scale** values invert the control.

Breakpoint settings: Assign BP To panel

The **Assign BP To:** panel lets you assign the breakpoint to one of the four Envelope Macros. The panel contains the following settings:



- **1–4 selector**: Assigns or unassigns the corresponding Envelope Macro to this breakpoint. Only one Envelope Macro can be assigned at a time. The cell of the assigned Envelope Macro is highlighted.
- **Time**: Adjusts the depth of the control over the breakpoint's time value as a percentage of the original time value. If no Envelope Macro is assigned to the breakpoint, the **Time** parameter is grayed out and inactive.
- **Amount**: Adjusts the depth of the control over the breakpoint's amplitude value as a percentage of the original amplitude value. If no Envelope Macro is assigned to the breakpoint, the **Amount** parameter is grayed out and inactive.

Envelope LFO

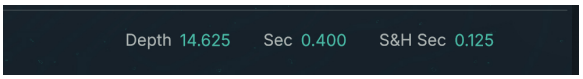
For all envelope modes except the Control Driven and Link modes, you can modulate the envelope amplitude using a dedicated low-frequency oscillator (LFO). The LFO wave will modulate the entire envelope, but every breakpoint can set different depth and rate values. The resulting LFO modulation is visually displayed in the Envelope Editor.

The LFO parameters are available in the **lower row** of the envelope settings, above the envelope displays.

The LFO parameters on the left affect the entire envelope:

- **LFO switch**: Activates or deactivates the LFO modulation of the envelope amplitude. The remaining LFO parameters will be visible only when the **LFO** switch is active.
- **Waveform thumbnail**: Depicts the shape of the waveform used by the LFO. Clicking the thumbnail opens a Waveform selector where you can choose another waveform or create your own waveform for the LFO.
- **S&H switch (Sample & Hold)**: Activates or deactivates the Sample & Hold mode. When **S&H** is off (default setting), the LFO steadily follows the selected waveform. When **S&H** is on, the LFO "samples" the waveform by extracting a value at regular intervals, and maintains this value until the next value is sampled. This process generates a staircase modulation signal based on the selected waveform.

If a breakpoint is selected in the Envelope Editor, additional LFO parameters for this breakpoint appear on the right in the **BP Settings** panel of the breakpoint settings:



- **Depth**: Adjusts the amount of modulation, from 0 to 100. At 0 the LFO has no effect on the envelope. At 100 the LFO oscillations go down to zero.

> By setting **Depth** to zero on chosen breakpoints, you can limit the LFO modulation to the remaining parts of the envelope.

- **Sec**: Adjusts the period of the oscillation in seconds, which defines the rate of the LFO (a short period making a high rate).
- **S&H Sec**: Available only when the Sample & Hold switch (**S&H**) is on, this adjusts the time interval between two sampled values, in seconds.

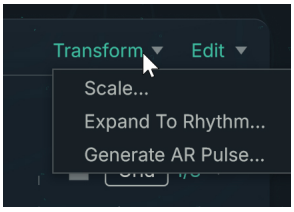
> If the breakpoint's LFO settings are not visible, click the three little dots in the row above until **BP Settings** is displayed.

Between breakpoints, the values of the LFO settings evolve according to the **Slope**/**Step** setting of the next breakpoint. You can find more information on this setting in Breakpoint settings: BP Settings panel.

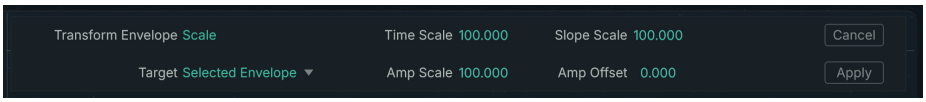
Transform menu (Envelope settings)

The **Transform** menu lets you apply various transformations to the envelope under focus or to all the envelopes displayed.

> The **Transform** menu is visible only if no breakpoint is selected in the Envelope Editor. If a breakpoint is selected, click anywhere in the background of the envelope display to deselect the breakpoint.



Selecting either transformation from the menu opens a dedicated **Transform panel** with additional settings at the top of the page:



In the Transform panel the following elements are common to all transformations:

- **Transform Envelope menu**: Switches to another transformation.
- **Target menu**: Specifies the envelope(s) affected by the transformation. If you select **Selected Envelope** (default setting) the transformation will affect only the envelope under focus (indicated by the white frame in the Envelope Editor). If you select **All Visible Envelopes** the transformation will affect all the envelopes shown in the Envelope Editor.
- **Cancel**: Closes the panel and leaves the envelope(s) untouched.
- **Apply**: Transforms the envelope(s) according to the panel settings and closes the panel.

The other parameters in the Transform panel depend on the selected transformation. As you change the parameter values, the shape of the envelope(s) and the resulting sound are updated in real time to show you the effect of your actions. You can then choose to apply these changes or to cancel them.

The **Transform** menu provides the following transformations:

- **Scale…**: Scales the various parameters of the envelope(s). You can choose which parameters should be scaled, and to which extent. The Transform panel contains the following parameters:
  - **Time Scale**: Scales the time intervals between breakpoints as a percentage of the original intervals. This scaling makes the envelope faster or slower.
  - **Amp Scale**: Scales the amplitudes of the breakpoints as a percentage of the original amplitudes. This scaling modifies the overall depth of the envelope modulation.
  - **Slope Scale**: Scales the slopes of the transitions between breakpoints as a percentage of the original slopes. Scaling down the slopes makes the transitions start faster and end more softly, and vice versa.
  - **Amp Offset**: Adjusts an offset globally added to the amplitudes of the breakpoints. Negative offset values will decrease the amplitudes. The available values range from -100 to 100: At -100 all the amplitudes are null. At 100 all the amplitudes are at the maximum.
- **Expand To Rhythm…**: Creates a rhythmic pattern by triggering the envelope(s) on selected steps. The Transform panel contains the following parameters:
  - **Num Beats**: Adjusts the number of steps in the pattern.
  - **BPM**: Adjusts the speed of the pattern in beats per minute.
  - **Duration**: Adjusts the length of the steps, measured in beats.
  - **Step Sequencer**: Shows the individual steps as numbered cells in a row. You can create a rhythmic pattern by activating or deactivating the desired cells. Only the active cells will trigger the envelope(s).
- **Generate AR Pulse…**: Creates a series of attack-release pulses based on the original envelope(s). The Transform panel contains the following parameters:
  - **Num Beats**: Adjusts the number of pulses.
  - **BPM**: Adjusts the speed of the pulse in beats per minute.
  - **Duration**: Adjusts the length of the pulses, measured in beats.
  - **Shape**: Adjusts the attack time of the pulses.
  - **Slope**: Adjusts the slope of the attack and release stages in the pulses. Smaller **Slope** values make the attack and release transitions start faster and end more softly, and vice versa.
  - **Min Level**: Sets the lowest amplitude value that the pulse can reach.

Envelope Macros

The four Envelope Macros (**1**–**4**) allow you to control envelope breakpoints from a common knob and using MIDI.

As with generic Macro controls, setting up the Envelope Macros includes two parts: assigning breakpoints to an Envelope Macro, and (if desired) assigning that Envelope Macro to a MIDI control.

To assign a breakpoint to an Envelope Macro:

1. Select the breakpoint in the envelope display.

→ The breakpoint settings appear in the top right of the page.

2. In the breakpoint settings, click the three little dots until the third dot lights up and the **Assign BP To** panel shows up.
3. Activate the switch **1**–**4** corresponding to the desired Envelope Macro.

→ The breakpoint is now controlled by the Envelope Macro.

Now, to assign the Envelope Macro to a MIDI control:

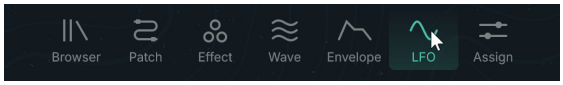
1. Right-click (Windows) or [control]-click (macOS) the Envelope Macro knob **1**–**4** in the **MASTER ENVELOPE MACROS** and select **Midi Learn** from the context menu.
2. Move the MIDI control on your MIDI controller or MIDI keyboard.

→ The Envelope Macro is now controlled by this MIDI control.

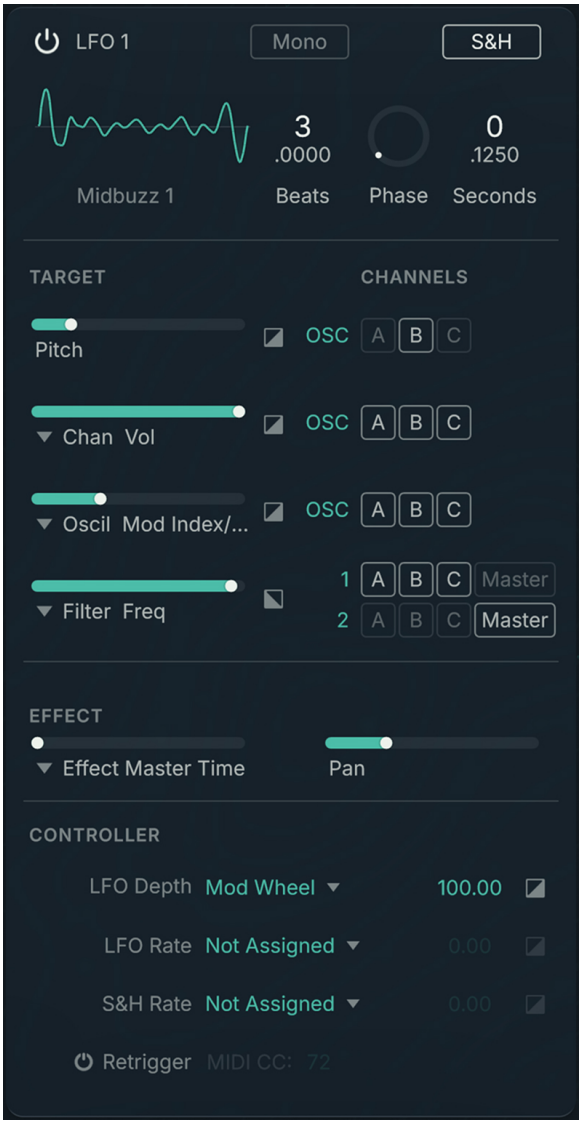
10. LFO page

The LFO page of Absynth 6 lets you configure up to three LFOs as modulation sources for a wide range of different controls across the instrument. LFOs, or low frequency oscillators, produce periodic modulation signals that can be used for slowly evolving or rapid movements of controls.

▶ To open the LFO page, click the **LFO** button at the top of the Absynth 6 window:



Each of the three LFOs on the LFO page consists of the following controls and sections:



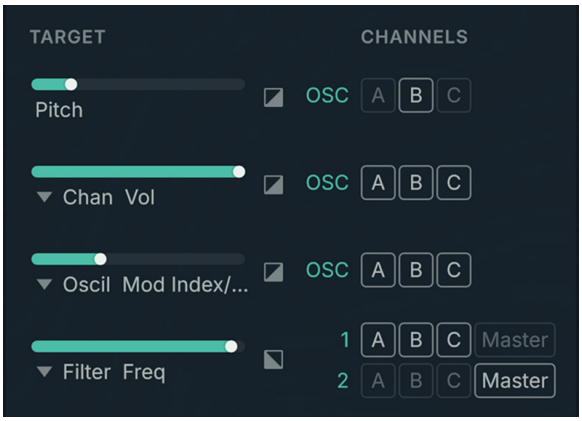
- **LFO On/Off**: Switches the LFO on or off. When switched off, the LFO does not produce a modulation signal.
- **Mini Waveform**: Shows the shape and the name of the loaded waveform. If a Morph waveform is loaded, you can click the shape and drag your mouse vertically to morph the waveform. Clicking the waveform name opens the Waveform selector and lets you choose another waveform.

- **Mono/Poly**: Switches the LFO between monophonic and polyphonic operation. When switched to Mono, the LFO modulates all the voices produced by the instrument. When switched to Poly, each voice is modulated by its own individual copy of the LFO.
- **LFO Rate**: Adjusts the speed of the LFO either in time intervals (**Seconds**) or quarter notes synced to the host tempo (**Beats**). You can switch between **Seconds** and **Beats** by clicking the label.
- **Phase**: Adjusts the initial phase position of the LFO.
- **S&H**: Switches Sample-and-Hold mode on or off. When **S&H** is active, the modulation produced by the LFO is turned into a staircase waveform by holding its momentary values at a frequency set using the S&H Rate control.
- **S&H Rate**: Adjusts the frequency of the Sample-and-Hold mode when **S&H** is active, either in time intervals (**Seconds**) or quarter notes synced to the host tempo (**Beats**). You can switch between **Seconds** and **Beats** by clicking on the label. The sample-and-hold frequency determines how often and for how long the values of the modulation are held.
- **Target section**: Lets you assign the modulation from LFO to controls on the Patch page You can find more information in Target section.
- **Effect section**: Lets you assign the modulation from the LFO to the Effect page and the panning. You can find more information in Effect section.
- **Controller section**: Lets you assign the modulation from the Macro controls the LFO. You can find more information in Controller section.

Target section

The Target section lets you assign the modulation from the LFO to the pitch of the instrument, along with three flexible assignments for controls on the Patch page.

The Target section consists of the following controls:

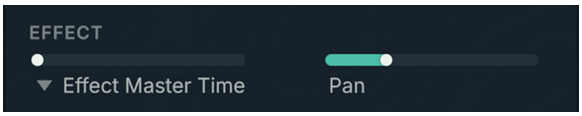


- **Pitch**: Adjusts the amount of modulation applied to the pitch of the instrument.
- **Modulation amount**: Adjusts the amount of modulation applied to the control assigned using the Modulation Target selector.
- **Modulation Target selector**: Assigns the modulation from the LFO to a control on the Patch page.
- **Invert**: Inverts the phase of the modulation for the respective assignment.
- **Channels**: Routes the modulation from the LFO to the channels on the Patch page. The modulation will be applied to the assigned control on all selected channels.

Effect section

The Effect section lets you assign the modulation from the LFO to the panning of the instrument, along with a flexible assignment for a control on the Effect page.

The Effect section consists of the following controls:

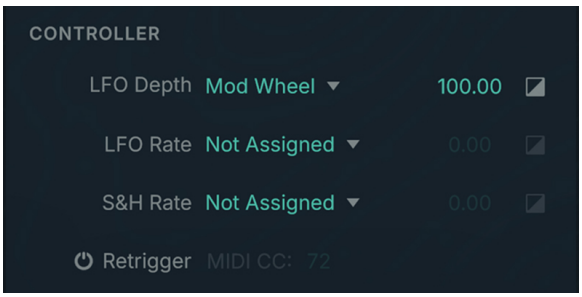


- **Modulation amount**: Adjusts the amount of modulation applied to the control assigned using the Modulation Target selector.
- **Modulation Target selector**: Assigns the modulation from the LFO to a control on the Effect page.
- **Pan**: Adjusts the amount of modulation applied to the panning of the instrument. Both left-right panning as well as front-back panning in surround configurations is supported.

Controller section

The Controller section lets you assign modulation from the Macro controls to the LFO itself. You can modulate the following parameters: **LFO Depth** (the amplitude of the LFO oscillations), **LFO Rate**, and **S&H Rate** (these two parameters are described in LFO page).

The Controller section consists of the following controls:



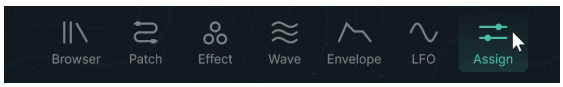
- Each LFO parameter that can be modulated provides a row with the following controls:
  - **Modulation Source selector**: Assigns the modulation from the selected Macro control to the respective LFO control.
  - **Modulation amount**: Adjusts the amount of modulation applied to the respective LFO control from the Macro control selected using the Modulation Source selector.
  - **Invert**: Inverts the phase of the modulation for the respective assignment.
- **Retrigger on/off**: Only applies when the LFO is in monophonic mode (**Mono** displayed at the top of the LFO). In monophonic mode the same LFO modulates all the voices of the instrument: The LFO starts with the first note and keeps running as long as notes are playing. If you activate **Retrigger**, you can reset the phase of the LFO to its start position (defined by the **Phase** parameter) by setting a MIDI continuous controller to any value higher than 0. You can specify the desired MIDI continuous controller using the **MIDI CC** parameter.

> When the LFO is in polyphonic mode, each voice of the instrument triggers its own copy of the LFO, no matter if **Retrigger** is active or not.

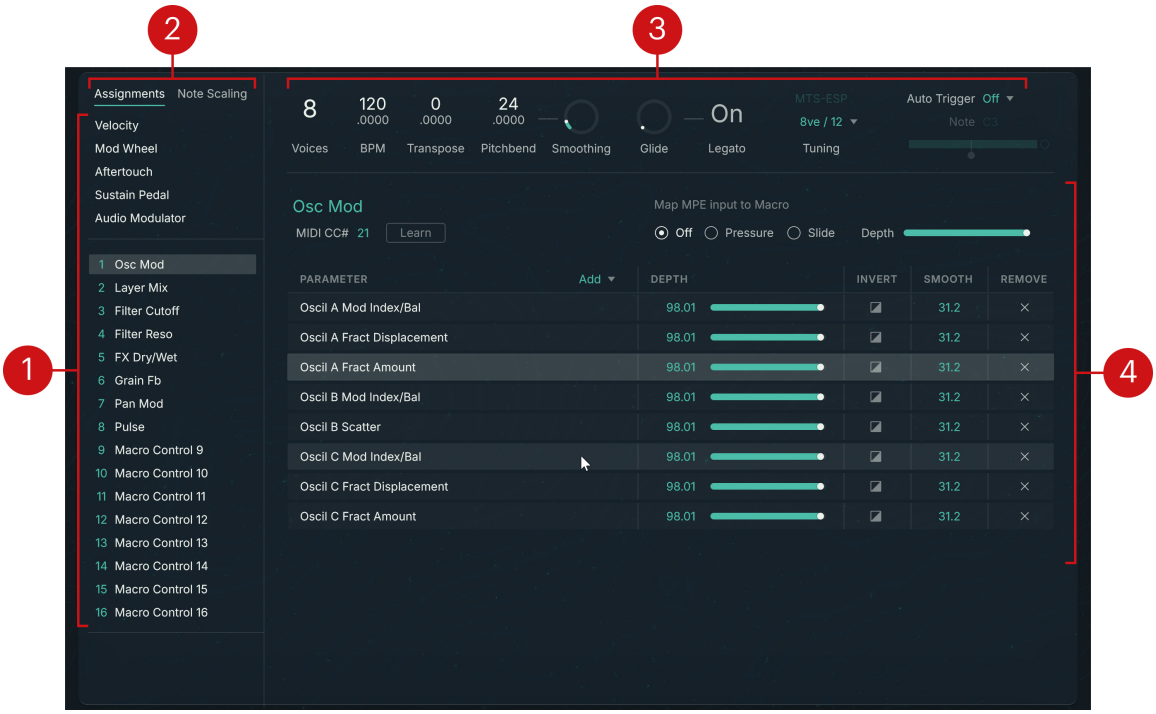
11. Assign page

The Assign page of Absynth 6 lets you configure the Macros, MIDI control, note scaling, and tuning of the instrument.

▶ To open the Assign page, click the **Assign** button at the top of the Absynth 6 window:



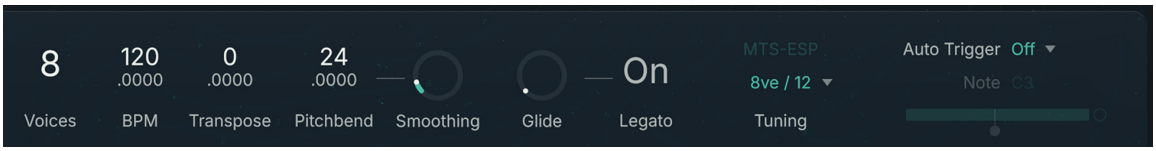
The Assign page contains the following areas and elements:



1. **Control List** / **Parameter List**: When **Assignments** is selected above, this shows a list of the MIDI and Macro controls. When **Note Scaling** is selected above, it shows a list of the parameters that can be modulated by the played pitch. In any case, clicking an entry from the list displays its details on the right.
2. **Assignments / Note Scaling switch**: Switches the Assign page between showing the MIDI and Macro assignments and the note scaling settings.
3. **Global Response area**: Shows global settings affecting how the instrument reacts to your input. You can find more information in Global Response.
4. **Assignment Details** / **Note Scaling Details**: Lets you configure the details for the control or parameter selected in the list on the left.

Global Response

At the top of the Assign page, the Global Response area lets you adjust how the instrument reacts to your input. It contains the following settings:

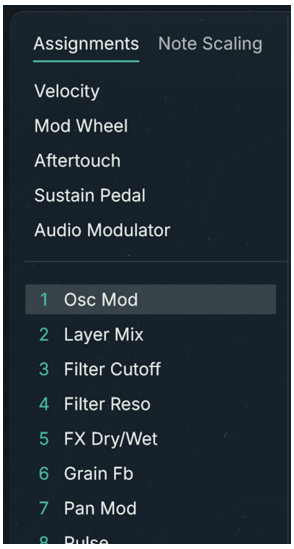


- **Voices**: Defines the number of voices produced by Absynth 6. This setting affects all the presets.
- **BPM**: Sets the global tempo, measured in beats per minute. This setting is available only if Absynth 6 is running as a standalone application. The **BPM** value affects, for example, the effects that depend on tempo.
- **Transpose**: Shifts the global pitch, by semitones. The resolution is by 1/10 cent, which is the minimal step width and can also be described as one thousandth of a semitone.
- **Pitchbend**: Adjusts the pitch range available for the MIDI Pitchbend input, in semitones.
- **Smoothing**: Adjusts how fast the pitch reacts to Pitchbend changes. For higher **Smoothing** value, the pitch will take longer to reach its new value.
- **Glide**: Adjusts the duration of a gradual pitch transition between played notes (glissando). The default value is 0 and the notes are generated directly at their pitches without any transition. If you raise the **Glide** value, the next notes will start from the pitch of the previous note and gradually move to their own pitch.
- **Legato**: When on, the **Glide** setting affects only the notes played legato, that is, with no silence in between. When off, the **Glide** setting affects all the notes.
- **Tuning menu**: Selects a global tuning. Clicking the displayed value opens a menu where you can select another tuning from the list of predefined tunings. You can also select **Import Tuning File** at the top of the menu: This opens a dialog in which you can select a custom tuning file. Beforehand, make sure to place your tuning files in the *Tuning* subfolder of your Absynth user folder. You can click the **Show Tuning Folder…** at the bottom of the dialog to quickly open this folder in the Finder (macOS) or Explorer (Windows). Absynth supports MTS-ESP microtunings.
- **Auto Trigger**: Lets Absynth trigger notes automatically. The **Auto Trigger** menu lets you select from the following modes:
  - **None**: Automatic note trigger is inactive.
  - **Always On**: Absynth holds a note whose pitch is specified by the **Note** control. To modify the pitch value, you can click it and drag your mouse vertically or horizontally, or double-click the value and enter a new pitch.
  - **Audio Input**: Notes are triggered depending on level of the incoming audio. The note pitch is specified by the **Note** control. To modify the pitch value, you can click it and drag your mouse vertically or horizontally, or double-click the value and enter a new pitch. Below, the **Threshold slider** lets you set the level above which the incoming audio will trigger a note.

MIDI and Macro assignments

When **Assignments** is selected in the top left corner of the Assign page, you can review and modify the MIDI and Macro assignments for your preset.

On the left, the **Control List** shows the **Velocity**, **Mod Wheel**, **Aftertouch**, **Sustain Pedal**, and **Audio Modulator** controls followed by the 16 Macro controls:



▶ Select a control from the Control List to show its assignment details on the right.

→ The selected control is highlighted.

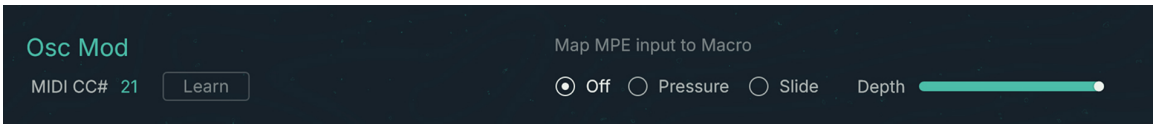
In the Control List, you can rename the Macros by double-clicking their entry, typing a new name, and pressing [Enter] on your computer keyboard. The Macro name will be updated accordingly everywhere in the instrument, as well as on your Kontrol S-Series keyboard.

> ⓘ Unassigned Macros are named **Macro Control X** where **X** is the Macro's number. When you assign a Macro for the first time, it automatically takes the name of its target parameter. You can modify the Macro name at any time to your liking.

On the right, the **Assignment Details** show the assignments for the selected control:

Control settings

The top part of the Assignment Details shows the name of the selected control (green) and its general settings. The control settings apply to all its targets.



The available settings depend on the selected control:

- The **Velocity**, **Mod Wheel**, and **Sustain Pedal** MIDI controls don't have any settings.
- The **Aftertouch** MIDI control provides a **Polyphonic Aftertouch** button: When active, you can control the listed targets for each played note independently.

  To use Polyphonic Aftertouch, your MIDI keyboard or controller must support it.

- The settings of the **Audio Modulator** control let you configure how the audio sources will be translated into modulation signals. These settings are described in Audio Modulators.
- The 16 Macros provide the following settings, from left to right:
  - **MIDI CC#**: Lets you specify the MIDI CC number from which you want to control the Macro. To change the value, you can click it and drag your mouse vertically, or double-click it and type the number followed by [Enter] on your computer keyboard. You can also click the **Learn** button on the right and move the desired control element on your MIDI controller.
  - **Map MPE input to Macro**: Lets you use MPE (MIDI Polyphonic Expression) signals to control the Macro. You can select the **Pressure** or the **Slide** MPE signal and adjust its modulation amount using the **Depth** slider. Selecting **Off** deactivates MPE for this Macro.

    To use MPE, your MIDI controller or keyboard must support it.

Audio Modulators

The Audio Modulators can convert Absynth's internal audio signals into modulation signals that you can assign to parameters. Absynth provides four Audio Modulators, labeled **A**, **B**, **C**, and **D**. Each modulator has its own audio source and settings, and it can be turned on or off independently,

An Audio Modulator acts as an **envelope follower**: It creates a modulation signal that reproduces the amplitude curve of the audio source, with some optional adjustments.

In addition, each Audio Modulator can act as an **envelope trigger**: When the processed source signal reaches a certain threshold level, the Audio Modulator can trigger the envelopes of the desired module in your patch.

When you select **Audio Modulator** in the Control List on the left, its control settings appear on the right:

- **A/B/C/D buttons**: Clicking either letter shows the settings of the corresponding modulator (whether it is active or not). Underneath the letters, the lit dots indicate the active modulators.

- **Modulator on/off switch**: Activates or deactivates the modulator. When the modulator is inactive, the remaining settings on the right are grayed out and the little dot under the modulator letter is unlit.
- **Source menu**: Selects the module whose audio output will be used to modulate the target parameters.
- **Post Envelope button**: Selects whether the source audio is tapped after the amplitude envelope has been applied (button on) or before (button off).
- **Gain control**: Adjusts the amplification or attenuation applied to the audio source before it is converted to a modulation signal. For example, this can be useful for the modulation signal to reach a specific range of values for the target parameter(s).
- **Attack control**: Adjusts how quickly the modulation signal will follow the upward movements of the source level. Higher Attack values lead to slower reactions.
- **Decay control**: Adjusts how quickly the modulation signal will follow the downward movements of the source level. Higher Decay values lead to slower reactions.
- **Trigger menu**: Selects a target module for the envelope trigger. Once you have selected a module, all the envelopes assigned to this module will be triggered every time that the modulation signal passes the threshold level. If **None** is selected (default setting), the envelope trigger is off.

  ⓘ If you select **Engine A**, **B**, or **C** from the menu, every time that the signal reaches the threshold, the corresponding source module A, B or C will play the note again.

- **Level meter and Threshold slider**: The horizontal bar shows the current level of the resulting modulation signal. When a module is selected in the **Trigger** menu above, an additional slider lets you adjust the threshold value that will trigger the envelopes of the module. On the right, the dot lights up when the envelopes are triggered.

Target list

The lower part of the Assignment Details shows the assignment's targets in a table layout, with one row for each target.

Each row contains the following elements, from left to right:

- **Parameter Descriptor**: Identifies the target parameter with an explicit label. For example, for a module the label could include the channel letter, the module position in the channel, the panel in the module, and the parameter name.

- **Depth**: Adjusts how strong the parameter value is affected by the assignment, as a percentage of the current parameter value.
- **Invert switch**: Reverses the modulation. The parameter value will decrease when the controller value increases.
- **Smooth**: Adjusts how long it takes for the parameter to reach its new values upon changes in the controller value.

  ⓘ The **Smooth** setting is not available when adjusting the **Velocity** assignments (**Velocity** selected on the left).

- **Remove**: Removes this target parameter from the list. The parameter will not be controlled by the MIDI or Macro control anymore.

You can add target parameters by clicking the **Add** button above the table, and selecting the desired parameter from the menu. Upon your selection a new row will appear in the table with the assignment settings for the new target.

You can also add or edit a parameter assignment from the page containing the parameter:

1. Switch to the page containing the parameter that you want to control, and locate that parameter in the page.
2. Right-click (Windows) or [control]-click (macOS) the parameter and select **Modulation Settings…** from the menu.

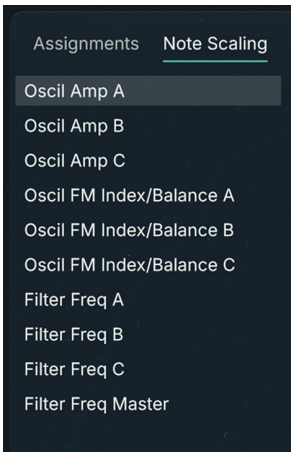
→ A Modulation Settings dialog appears with the parameter name in the header and its modulation settings below.

3. If you want to control the parameter using a MIDI or Macro control, click the menu in the **Macro** section and select the desired control from the list. If you want to control the parameter using key velocity, activate the **Velocity** switch. If you prefer to control the parameter using an Audio Modulator, click the menu in the **Audio Modulator** section and select either modulator **A**, **B**, **C**, or **D**.
4. In either section, you can adjust the **Depth** value and activate the Invert switch directly from the dialog. This is equivalent to using the **Depth** and Invert elements from the Assign page's Target list described above.

Note scaling

When **Note Scaling** is selected in the top left corner of the Assign page, you can configure the modulation of specific parameters according to the played key. For example, you could make the frequency of a bandpass filter follow the pitch of the key.

On the left, the Parameter List shows the available parameters:



The following parameters are available for note scaling:

- **Oscil Amp A/B/C**: Amplitude envelope of the Main oscillator in the source module on channel **A**/**B**/**C**.
- **Oscil FM Index/Balance A/B/C**: Ratio between the Main and Mod oscillators in the source module on channel **A**/**B**/**C**.
- **Filter Freq A/B/C/Master**: Cutoff frequency of the Filter modules on channel **A**/**B**/**C** or on the Main channel.

▶ Select a parameter from the list to show its scaling details on the right.

On the right, the Note Scaling Details show the note scaling for the selected parameter:

The top part of the Note Scaling Details lets you adjust the parameter modulation value for each individual keys. It contains the following controls:

- **Key**: Selects the desired pitch. To select another pitch you can drag the value vertically with your mouse, or double-click it and type the desired MIDI Note number on your computer keyboard.
- **MIDI Note switch**: When on, you can use your MIDI keyboard to quickly select the **Key** value. This can be useful when adjusting the modulation for several keys in a row.

- **Scale**: Adjusts the modulation value for the selected pitch. The value is expressed as a percentage of the original, non modulated parameter value.

The bottom part of the Note Scaling Details shows the modulation curve as a series of vertical bars representing the scale values over the full key range, from the lowest pitch on the left to the highest pitch on the right. You can quickly draw a modulation curve with your mouse as follows:

1. Click anywhere in the vertical bars and drag your mouse to draw the desired modulation curve.