EROSIA

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1. Disclaimer

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Software version: 1.0 (08/2025)

2. Welcome to Erosia

Erosia is a cinematic sound design engine that listens closer than the ear can. Each texture begins with the smallest detail: a bow drawn across glass, a hinge under strain, ice cracking across metal. Captured up close, these fragile moments expand into vast cinematic landscapes. Friction becomes motion, decay becomes beauty, and the overlooked becomes a powerful storytelling tool for composition and sound design.

Four layers form Erosia's core: a Grain layer that fractures and animates textures, plus three interchangeable Sampler or Wavetable layers. Together, they create instruments that feel alive, organic, and unpredictable. Streamlined Play and FX pages let you shape sound quickly and intuitively, while the Transform control evolves timbres, moving seamlessly between tension, release, and decay.

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

We hope you enjoy Erosia!

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.

3. Installation and setup

Before making music with Erosia, you must install and set up the necessary software. Follow these instructions to get started.

Installing Erosia using Native Access

Native Access is your go-to app for downloading, activating, and updating all your NI music creation tools including Erosia. 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 **Kontakt** category to only show products related to Kontakt.
8. Click **Install** for the following products:
   - Erosia
   - Kontakt or Kontakt Player

→ The software is installed automatically.

> If the software is already installed, click the **Updates** tab and install the available updates before proceeding.

Loading Erosia in Kontakt

Once installed, you can start using Erosia in Kontakt. Erosia is not an independent plug-in, so you first need to open an instance of Kontakt or Kontakt Player.

Kontakt offers two ways to load an instrument, the Library browser and the side pane browser.

To load an instrument using the Library browser:

1. Open Kontakt as a plug-in in your host software (DAW) or as a stand-alone application.
2. By default, Kontakt opens the Library browser on first launch. If you have turned this off, click **Library** in the Kontakt header to open the Library browser.
3. In the Library browser, make sure that the **Instruments** category is selected at the top (this should be the case by default), otherwise click **Instruments** to select that category.
4. Locate Erosia in the Library browser. You can use the search bar at the top to quickly find it.
5. Click on the arrow icon (**>**) in the top right corner of the instrument's artwork to load the instrument and its first preset.
6. Alternatively, you can click the instrument's artwork to display its presets in the list on the right of the browser window.

7. Double click any preset to load it. The first entry, identified by a keyboard icon, loads the instrument with its default preset.

To load an instrument using the side pane browser:

1. Open Kontakt as a plug-in in your host software (DAW) or as a stand-alone application.
2. In the side pane on the left, make sure that the **Instruments** category is selected (this should be the case by default), otherwise click **Instruments** to select that category.
3. Locate Erosia's artwork tile below.
4. Click on the arrow icon (**>**) in the top right corner of the instrument's artwork to load the instrument and its first preset.
5. Alternatively, you can click the instrument's artwork to display the list of its presets.
6. Double click any preset to load it. The first entry, identified by a keyboard icon, loads the instrument with its default preset.

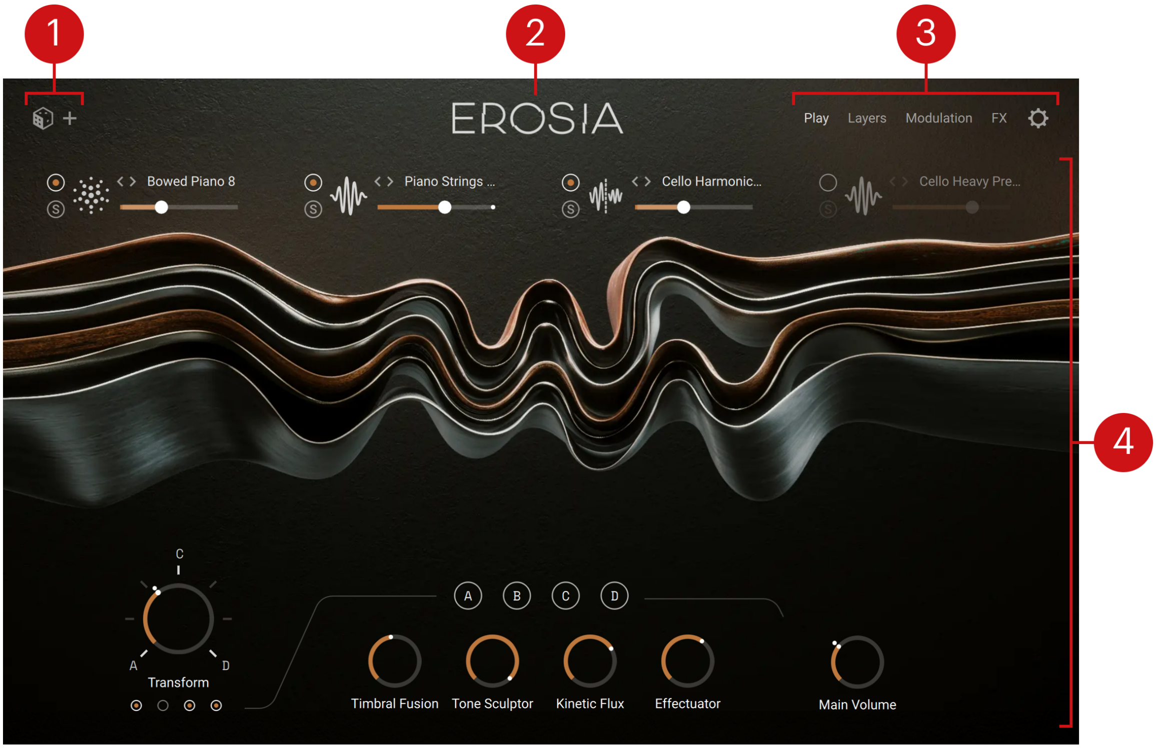
If you are new to Kontakt and want more information, visit Kontakt Player and Kontakt.

4. Erosia overview

This chapter introduces the main concepts and areas of Erosia.

To generate sound, Erosia uses four layers playing simultaneously but using up to three different sound engines: Layer 1 always uses a **grain engine**, while layers 2–4 can be individually set to use a **sample engine** or a **wavetable engine**. All three engines use source samples to produce their sound, but in different ways. Each engine has its own synthesis technique, engine parameters, and dedicated source samples.

The Erosia window can display various pages along with a set of global controls, which are always visible at the top of the instrument:



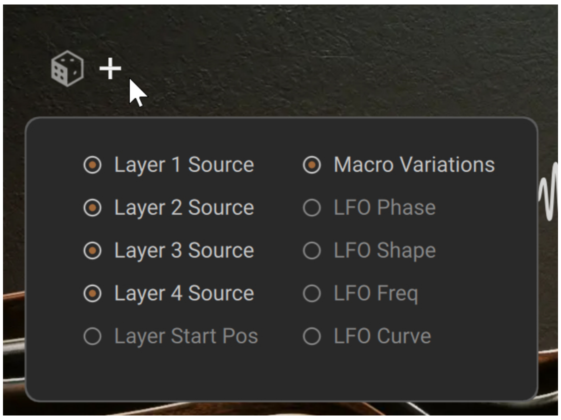
1. **Randomize controls**: These controls let you quickly set new random values for specific parameters in the instrument. Refer to Randomize controls.
2. **About**: Clicking the instrument name opens the About screen, which displays the credits for this instrument. You can click the About screen to close it again.
3. **Page buttons**: Clicking either button opens the corresponding page below.
   - **Play**: Opens the Play page, which is the instrument's default page and is depicted above. The Play page contains a set of basic controls for each of the four layers, an animated artwork, and the Transform and Macro controls. Refer to Play page.
   - **Layers**: Opens the Layers page, which lets you adjust the sound of the four layers in detail. Refer to Layers page.
   - **Modulation**: Opens the Modulation page, which lets you configure the internal modulation sources of the instrument. Refer to Modulation page.
   - **FX**: Opens the FX page, where you can set up two insert effect chains and a send effect to process the sound of the four layers. Refer to FX page.
   - **Options**: Opens the Options page, which contains pitch, voicing, engine, and response settings for the instrument. Refer to Options page.

4. **Page area**: This is the main part of the window and can display the various pages of the instrument.

Randomize controls

In the top left corner of the instrument, the Randomize controls let you quickly set new random values for specific parameters.

- **Randomize button (dice)**: Sets new random values for the selected target parameters.
- **Targets button ("+")**: Opens a panel where you can choose which parameters will be randomized when clicking the dice. The selected parameters are highlighted in the panel. You can activate or deactivate parameters by clicking their labels. By default, the **Layer 1–4 Source** and **Macro Variations** entries are selected.



The following target parameters are available:

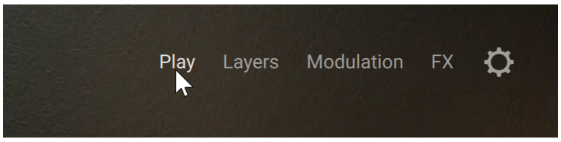
- **Layer 1–4 Source**: Random source sample in the layer 1–4.
- **Layer Start Pos**: Random start position in all four layers. This affects the **Scan** control in the grain layer, the **Start** control in sample layers, and the **Position** control in wavetable layers.
- **Macro Variations**: Random sets of values for the four Macro Variations, each variation including random values for the four Macros.
- **LFO Phase**, **Shape**, **Freq**, and **Curve**: Random values for the corresponding parameters in both LFOs.

ⓘ The targets selected in the panel are not saved with the instrument presets, so that the same targets will stay selected when you switch between instrument presets.

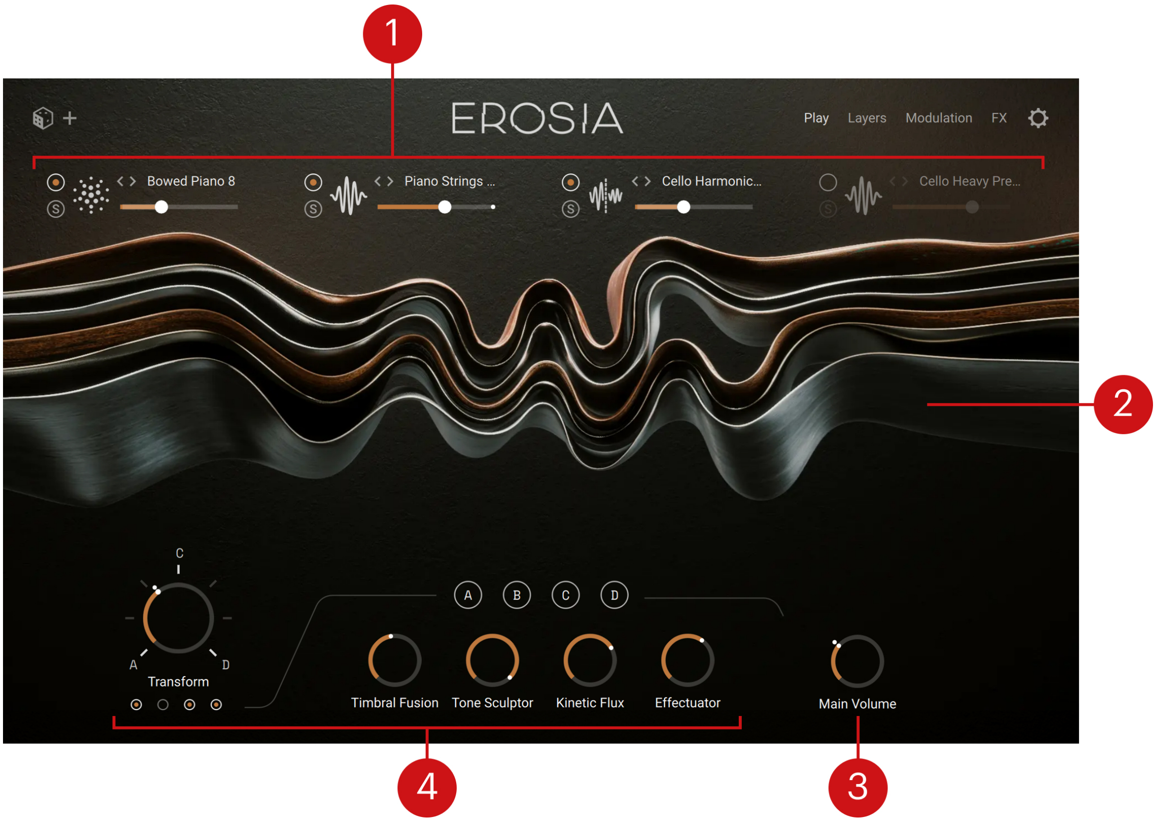
5. Play page

When you first open Erosia, the Play page shows up by default. You can also display the Play page at any time:

▶ To open the Play page, click the **Play** button in the top right corner of the instrument.



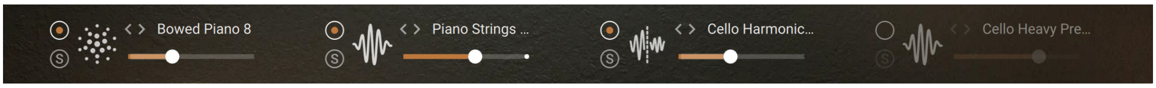
The Play page contains the following elements:



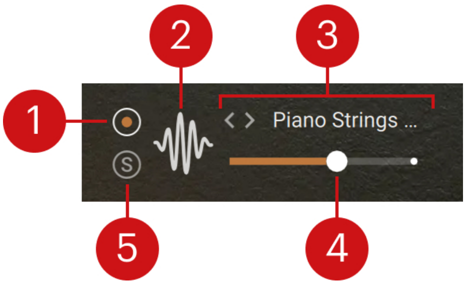
1. **Layer Inspector**: Provides basic controls for the four layers of Erosia. For each layer, you can switch it on and off, see the source sample currently in use, switch to another source by clicking the next and previous buttons, browse all available sources by clicking the source name, adjust the layer volume, and solo this layer to hear it without the other layers. Refer to Layer Inspector.
2. **Erosia artwork**: The artwork varies with the position of the Transform knob at the bottom left.
3. **Main Volume knob**: Adjusts the overall volume level of the instrument.
4. **Transform and Macro controls**: The four Macros are remote controls that you can assign to the desired parameters in the instrument. The Macros belong to the various modulation sources available in Erosia. In addition you can define different sets of values for the Macros, called **Macro variations**, and morph between the variations using the **Transform** knob. Refer to Transform and Macro controls.

Layer Inspector

The Layer Inspector contains a basic set of controls for each of the four layers 1–4.



For each layer, the Layer Inspector contains the following elements:



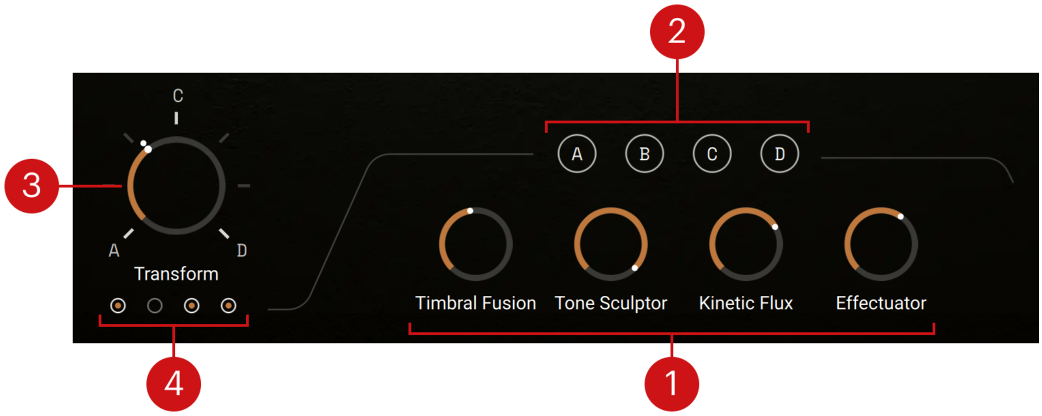
1. **Layer on/off switch**: Activates or deactivates the layer. Inactive layers don't generate any sound.
2. **Layer icon**: Indicates the type of layer: grain, sample, or wavetable. You can click the icon to quickly open the layer details in the Layers page.
3. **Source selector**: Displays the name of the sound used as source in the layer. You can click the sound name to open the Source browser and select another source for that layer. Alternatively, you can click the left and right arrows to quickly load the previous or next sound from the browser's result list without opening the browser.
4. **Layer Level slider**: Adjusts the volume level of the layer.
5. **Solo** (**S**): Mutes all the other layers. Soloing a layer can be useful to focus on that layer when designing your sound.

> The four Layer Level sliders can be modulated. For more information, refer to Modulating your sound.

Transform and Macro controls

At the bottom of the Play page, the Transform and Macro controls let you use a custom set of controls for the instrument, define variations of their values, and morph the sound between the different variations.

The Transform and Macro controls contain the following elements:



1. **Macros**: The four Macro knobs (**Timbral Fusion**, **Tone Sculptor**, **Kinetic Flux**, and **Effectuator**) can control one or more parameters in Erosia. They are already assigned to parameters in the factory presets, but you can freely modify their assignments, these will be saved with your user presets. The Macros belong to the various modulation sources available in the instrument. They follow the common workflow for assigning modulation. Refer to Modulating your sound for more information.
2. **Variation Edit buttons (A–D)**: Each button lets you define values for the four Macros and store them into one Macro Variation. When you activate a Variation Edit button, the button lights up and you can adjust the four Macro knobs below to create a sound to your liking. When you are done, you can deactivate the lit button to quit edit mode and store your new Macro values into that Macro Variation. Your new Macro Variation will be used at the chosen position(s) of the **Transform** knob.
3. **Transform knob**: Morphs the sound between the two Macro Variations on either side of the knob's current position. For example, if the **Transform** knob is on a **B** graduation, when you turn it towards a **C** graduation the sound will progressively morph from the Macro Variation B into the Macro Variation C. You can click any letter on the knob's scale to enter edit mode for that Variation. As you hover over the **Transform** knob with the mouse, little up/down arrows appear next to each letter:

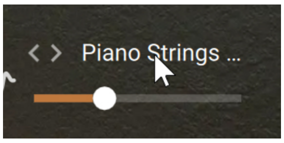
   

   Clicking the arrows lets you assign another Macro Variation to that position on the scale.
4. **Position on/off switches**: These switches are active by default. Deactivating a switch will remove the Macro Variation from the corresponding position on the **Transform** knob's scale: The leftmost switch removes the Variation located at full left on the knob scale, the second switch removes the second Variation from the left on the scale, and so on. Each time, the remaining Macro Variations are moved on the scale to spread equally over the full range of the knob. Activating a switch will re-insert the Variation at the corresponding position in the sequence.

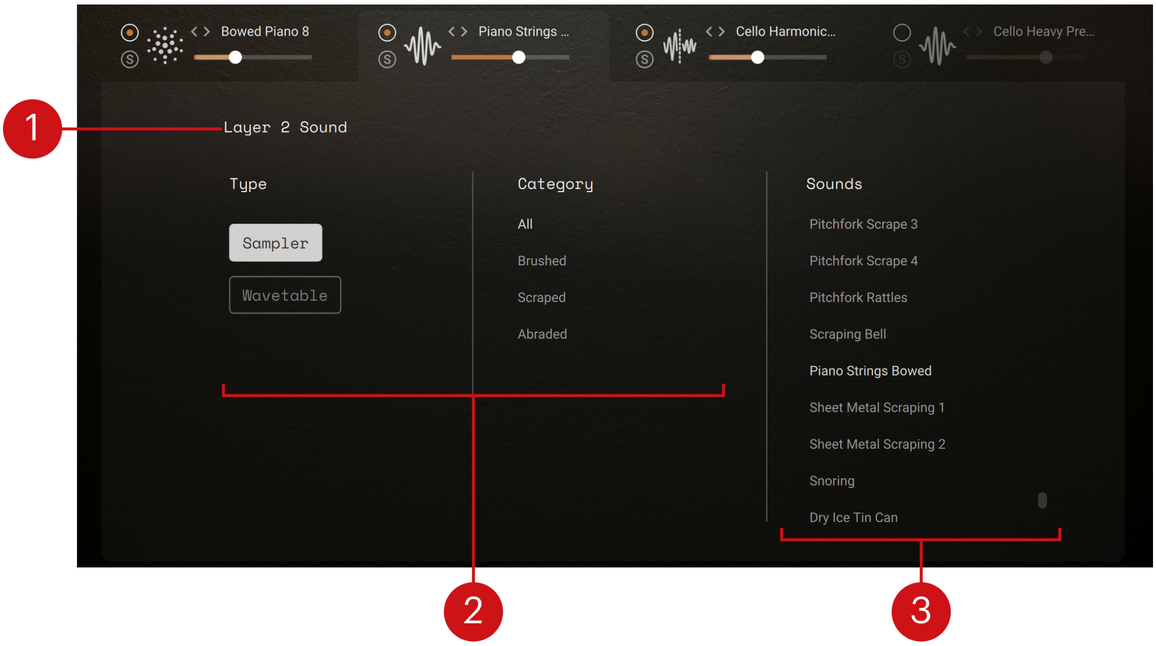
6. Source browsers

The Source browsers let you browse and load the source sounds used in the four layers.

▶ To open the Source browser for a particular layer, click the name of its current source sound in the Layer Inspector of the Play page or in the Layer tabs of the Layers page:



The Source browsers contain the following elements:

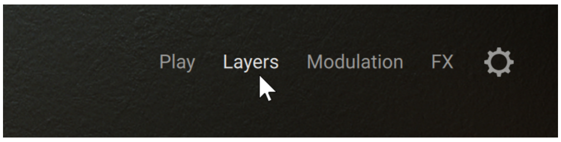


1. **Browser name**: Indicates the layer for which you are browsing source sounds.
2. **Tag filter**: You can click the tags describing the sound that you are looking for. The **Sounds** list on the right is updated accordingly. Only one tag can be selected in each column.
   - For all types of layers (grain, sample, and wavetable), the Tag filter provides a **Category** column. At the top of the column, the **All** tag shows all the available sources.
   - For the sample and wavetable layers (layers 2–4), the Tag Filter also provides a **Type** column with two entries: **Sampler** and **Wavetable**. Clicking either entry will determine the engine used in that layer. The tags in the **Category** column vary with the engine selected in the **Type** column.
3. **Sounds list**: Shows the source sounds available for that layer with the selected Type and Category tags. The loaded source is highlighted. You can click another source from the list to load it into the layer and directly hear its sound in context. Double-clicking a source loads it and closes the browser. The scroll bar on the right lets you display the remaining entries.

7. Layers page

The Layers page of Erosia lets you adjust the sound of its four layers in detail.

▶ To open the Layers page, click the **Layers** button in the top right corner of the instrument.

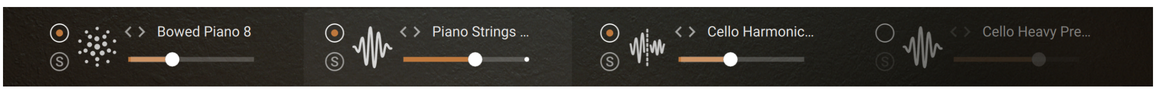


The Layers page contains the following elements:

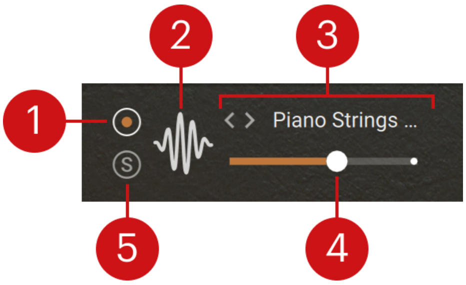
1. **Tab headers**: Each header provides a set of basic controls for one layer, and lets you display the layer's details below. All the controls in the areas below apply to the layer selected here. Refer to Tab headers.
2. **Layer parameters**: Each tab lets you adjust in detail the sound of a particular layer. Refer to Layer parameters.

Tab headers

At the top of the Layers page, the tab headers let you switch between the layer tabs, and provide for each layer the same set of basic controls as in the Layer Inspector of the Play page:



Each tab header contains the following controls:



1. **Layer on/off switch**: Activates or deactivates the layer. Inactive layers don't generate any sound.
2. **Layer icon**: Indicates the type of layer: grain, sample, or wavetable. Clicking the icon opens the corresponding layer tab below.
3. **Source selector**: Displays the name of the sound used as source in the layer. You can click the sound name to open the Source browser and select another source for that layer. Alternatively, you can click the left and right arrows to quickly load the previous or next sound from the browser's result list without opening the browser.
4. **Layer Level slider**: Adjusts the volume level of the layer.
5. **Solo** (**S**): Mutes all the other layers. Soloing a layer can be useful to focus on that layer when designing your sound.

The four Layer Level sliders can be modulated. For more information, refer to Modulating your sound.

Layer parameters

The layer parameters are organized into various sections, which are available for all three types of layers (grain, sample, and wavetable) unless otherwise indicated.

On the layer tabs, the signal flows from top to bottom and from left to right:

- In the top row, the layer engine generates the sound as configured in the first row in the Main and Grain, Sample or Wavetable sections. In grain and sample layers, the sound additionally passes through the Fragment section.
- The sound then goes through the bottom row and is successively processed by the layer's internal effects of the Lo-Fi, Harmonic Resonator, and Filter sections.
- Finally, the FX Bus section sends the sound to the desired effect bus A or B.

The layer sound will then pass through the effect chain on the selected bus for further processing. This is described in FX page.

Main section

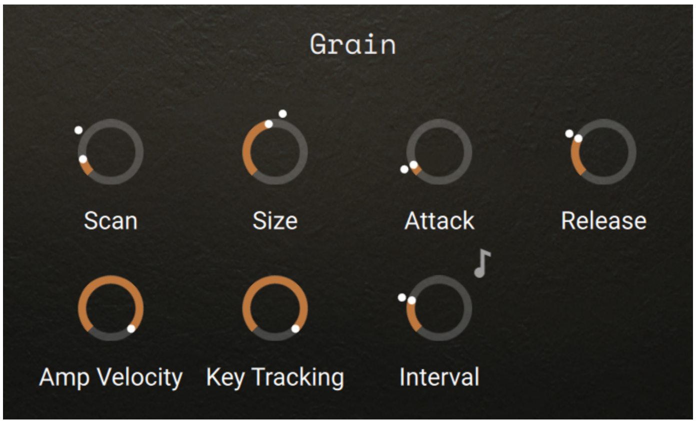
The Main section contains basic controls for the layer.

- **Pan**: Adjusts the panoramic position of the layer in the stereo field.
- **Pitch**: Adjusts the coarse pitch of the layer in semitones.
- **Detune**: Adjusts the fine pitch of the layer in cents.

Grain section (grain layer only)

The **Grain** section lets you configure the grain engine of the layer.

> The **Grain** section is only available in the grain layer, which corresponds to the leftmost tab header.

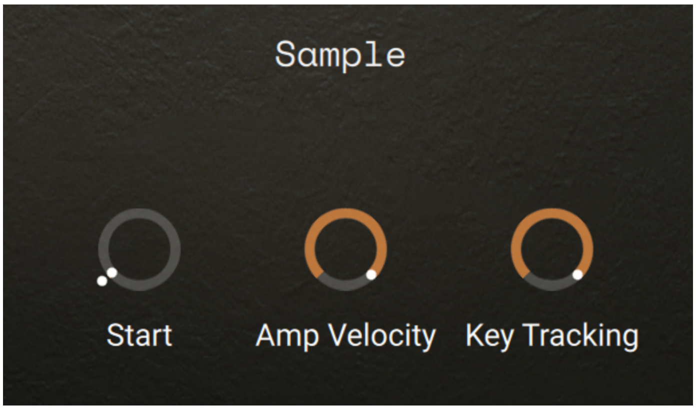


- **Scan**: Adjusts the position in the source sample from which the grains are created. The position is set as percentage of the range between the left and right boundaries of the source sample.
- **Size**: Adjusts the duration of each individual grain.
- **Attack**: Adjusts the fade applied at the beginning of each grain. You can use this to soften the edges of the grains.
- **Release**: Adjusts the fade applied at the end of each grain. You can use this to smoothen the release of the grains.
- **Amp Velocity**: Adjusts how much the note velocity will affect the amplification level. At full left, the sounds will always have the same level no matter how strong you hit the keys. The more you turn the knob to the right, the stronger the note velocity will affect the level of the resulting sounds.
- **Key Tracking**: Adjusts how much the note pitch will affect the pitch of the sounds. At full left, the sounds will always have the same pitch no matter which key you press on your keyboard. At full right, the sound pitches follow exactly the key pitches.

- **Interval**: Adjusts the time between the generation of individual grains from the source sample.
- **Interval Sync (note icon)**: Allows you to set the Interval value relative to Kontakt's main tempo, which is the tempo of your DAW if Kontakt or Kontakt Player is running as a plug-in.

Sample section (sample layers only)

The **Sample** section lets you configure the sampler engine of the layer.



- **Start**: Adjusts the start position of the playhead in the source sample.
- **Amp Velocity**: Adjusts how much the note velocity will affect the amplification level. At full left, the sounds will always have the same level no matter how strong you hit the keys. The more you turn the knob to the right, the stronger the note velocity will affect the level of the resulting sounds.
- **Key Tracking**: Adjusts how much the note pitch will affect the pitch of the sounds. At full left, the sounds will always have the same pitch no matter which key you press on your keyboard. At full right, the sound pitches follow exactly the key pitches.

Fragment section (grain and sample layers only)

The **Fragment** section lets you split the audio into short fragments and modify their playback speed before reassembling them together.

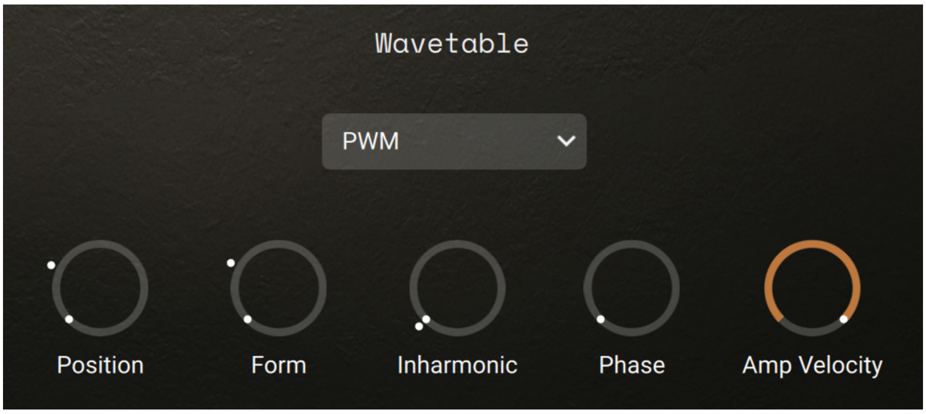
The **Fragment** section contains the following controls:

- **On/off switch**: Activates or deactivates the section. When deactivated, the section does not process the audio.
- **Speed**: Changes the playback speed of the fragments. This value is displayed as a percentage of the original speed, so 200 % doubles the playback speed, while 50 % halves it. The control goes all the way down to 0%, which freezes the sound at its current playback position.

- **Smooth**: In order to reduce artifacts during playback, the fragments are crossfaded into each other. This control adjusts the shape of these very short crossfades. Higher values will result in a smoother sound, while lower values will generate a buzzing sound.
- **Length**: Adjusts the size of the fragments in milliseconds.

Wavetable section (wavetable layers only)

The **Wavetable** section lets you configure the wavetable engine of the layer.



- **Type menu**: Selects the type of wave shaping applied to the oscillator's phase. This fundamentally changes the way the wavetable is read out, therefore bending and warping the resulting waveform.
- **Position**: Morphs between the waveforms included in the loaded wavetable.
- **Form**: Adjusts the amount of wave shaping applied to the oscillator's phase. The type of wave shaping can be selected using the Type menu.
- **Inharmonic**: Adjusts how far the partials are stretched away from the harmonic series.
- **Phase**: Adjusts the reset point of the oscillator's phase. This determines the sound's start position in the waveform when a new note is triggered.
- **Amp Velocity**: Adjusts how much the note velocity will affect the amplification level. At full left, the sounds will always have the same level no matter how strong you hit the keys. The more you turn the knob to the right, the stronger the note velocity will affect the level of the resulting sounds.

Lo-Fi section

The **Lo-Fi** section adds various digital artifacts like quantization noise or aliasing by reducing the audio quality.

- **On/off switch**: Activates or deactivates the section. When deactivated, the section does not process the audio.
- **Bit**: Re-quantizes the signal to an adjustable bit depth. Fractional bit levels (such as 12.4 bits) are possible and can add considerable "grit". Audio CDs have a quantization depth of 16 bits, old samplers frequently used 8 or 12 bits, and 4 bits evoke memories of countless irritating children's toys.
- **KHz**: Re-samples the signal to an adjustable sample rate. The re-sampling is done without any kind of (usually mandatory) low-pass filtering, which causes all kinds of wonderful aliasing artifacts. The sample rate goes all the way down to 50 Hz, which will not leave much of the original signal.
- **Noise**: Adds hiss to the sound.
- **Nz Color**: Adjusts the frequency characteristic of the noise and acts as a low-pass filter.

Harmonic Resonator section

The **Harmonic Resonator** section provides an advanced filtering with three separate bandpass filters (**H1**, **H2**, **H3**). Each filter band has a slope of 12 dB/octave. In addition, at high **Resonance** values the filter will begin to oscillate and produce sound, even if there is no signal present at the input. This effect is known as self-oscillation.

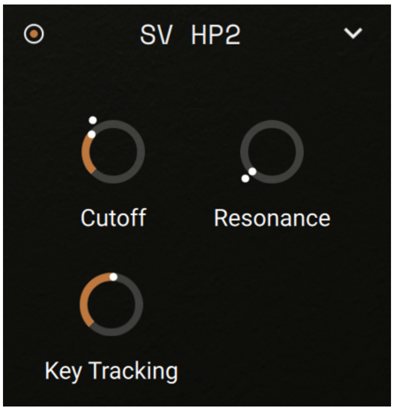
- **On/off switch**: Activates or deactivates the section. When deactivated, the section does not process the audio.
- **H1 Frequency**: Adjusts the cutoff frequency of the first filter band (**H1**), which directly affects the cutoff frequencies of the three filter bands in unison, since the other two bands have cutoff frequencies defined relative to this one. The cutoff frequency is measured in octaves relative to the pitch of the played key. The available values range from **-3** to **+3** octaves.

- **H2 Semitones**: Adjusts the cutoff frequency of the second filter band (**H2**) in semitones as an offset in relation to the first band (**H1**). With a value of **0**, both filters will have identical cutoff frequencies, while increasing the value will set the second cutoff frequency higher than the first. The available values range from **0** to **60** semitones.
- **H3 Semitones**: Adjusts the cutoff frequency of the third filter band (**H3**) in semitones as an offset in relation to the second band (**H2**). With a value of **0**, both filters will have identical cutoff frequencies, while increasing the value will set the third cutoff frequency higher than the second. The available values range from **0** to **60** semitones.
- **H1, H2, H3 Resonance**: These adjust the resonance (boost at the cutoff frequency) for each filter band. Values of **98 %** or higher will result in self-oscillation.
- **Gain**: As high resonance settings can significantly increase the signal level, the instrument will automatically reduce the output level in such cases. You can compensate this with the **Gain** control, but be careful: It is easy to get excessive volume levels from this filter.
- **Pitch Link**: When this is active, the cutoff frequencies of the three filter bands will follow the global pitch of the layer, as set by the **Pitch** and **Detune** controls in the Main section.

Filter section

At the bottom right of the Layers page, the Filter section lets you process the layer sound using one of many available filters.

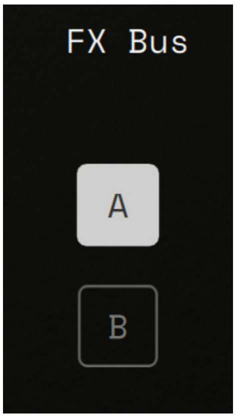
The Filter section contains the following elements:



- **On/off switch**: Activates or deactivates the section. When deactivated, the section does not process the audio.
- **Filter menu**: Shows the name of the current filter at the top of the slot. You can click that name and choose another filter from the menu, or select **None** to leave the slot empty.
- **Filter parameters**: The filter parameters vary with the selected filter. You can find a list of the available filters and a description of their parameters in Filter reference.

FX Bus section

The **FX Bus** section is the last step of the sound on the **Layers** page. Here you can choose which of the two available effect chains A or B should process the layer's sound.



The FX Bus section contains only one element:

- **A/B Bus switch**: Selects the effect bus **A** or **B** to which the layer’s signal will be routed.

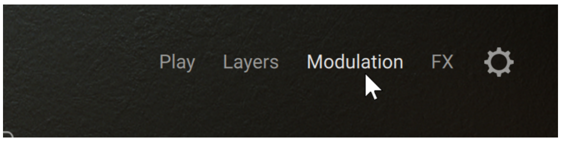
You can choose and configure the effects loaded into each effect chain using the FX page.

8. Modulation page

The Modulation page of Erosia lets you configure its internal modulation sources, or **modulators**.

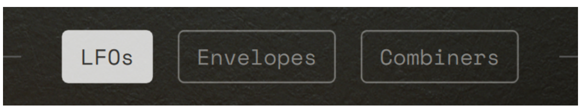
> These modulators are listed among other modulation sources in the Modulation menus appearing next to many parameters of Erosia. For a list of the available modulation sources, and to learn how to assign modulation, refer to Modulating your sound.

▶ To open the Modulation page, click the **Modulation** button in the top right corner of the instrument.



Erosia provides six modulators: two LFOs, two envelopes, and two combiners.

At the top of the Modulation page, the **Modulator Type selector** lets you display the parameters for the two modulators of either type:



- **LFOs**: Shows the parameters of the LFO 1 and LFO 2 modulators. Refer to LFOs.
- **Envelopes**: Shows the parameters of the Envelope 1 and Envelope 2 modulators. Refer to Envelopes.

- **Combiners**: Shows the parameters of the Combiner 1 and Combiner 2 modulators. Refer to Combiners.

LFOs

When the Modulator Type selector is set to **LFOs** at the top of the Modulation page, the page shows the parameters for the two available LFOs: **LFO 1** on the left, **LFO 2** on the right.

Each LFO provides the following elements:

- **Oscillator Display**: Shows the shape of one LFO cycle with the current settings. The horizontal axis represents the time, and the vertical axis represents the modulation value. The white vertical line indicates the current playback position of the LFO for the last played note. You can click or click-and-drag anywhere in the display to manually edit the shape. When you do this, the Waveform selector below automatically switches to **Custom**.
- **Steps**: Adjusts the number of steps in one cycle, from 1 to 128.
- **Waveform selector**: Selects the waveform shape from the following available waveforms: **Sine**, **Tri** (triangle), **Saw**, **Pulse** (or "square"), **Random**, and **Custom**. Each time you select **Random**, new random values are generated for every step. Selecting **Custom** recalls the last waveform that you have manually edited.
- **Playback Mode selector**: Selects how the shape should be played back. In **Linear >** mode, the shape is read from left to right at a steady rate. In **Linear ><** mode, the shape is read back and forth at a steady rate. In **Circular** mode, the shape is read back and forth and the playback slows down near the left and right boundaries.
- **Invert**: Reverses the waveform along the time axis, so that the end of the cycle becomes its beginning.
- **Freq**: Adjusts the oscillator frequency, which specifies how many LFO cycles will be read within one second.

- **Freq Sync (note icon)**: When Freq Sync is on, the **Freq** values are set in note values relative to the main tempo of Kontakt, which is the tempo of your DAW if Kontakt / Kontakt Player is used as a plug-in. When Freq Sync is off, the **Freq** values are set in Hertz.
- **Curve**: Adjusts the curvature of the shape. At the default center position, the original shape is left untouched. Negative values of **Curve** will pull the shape downwards, positive values will pull it upwards.
- **Phase**: Adjusts the oscillator phase. Changing the phase will shift the shape along the time axis.
- **Attack**: Adjusts how long it takes for the LFO to reach its specified shape from the moment it is triggered. With **Attack** at full left (minimum value), the LFO directly starts with the shape shown on the display. At higher values, the LFO will need longer before getting fully active.
- **Multiply**: Specifies how many times the shape should be read during one LFO cycle. The default value is **1**, and the available values range from **0.125** (1/8, that is, 8 LFO cycles are needed for the full shape to be read) to **8** (8 shapes are read within one LFO cycle).
- **Interpolate**: Smooths the LFO by blending between adjacent step values. At low settings the LFO is stair-stepped; as you increase it, the LFO becomes progressively smoother.
- **Retrigger Mode menu**: Specifies when the LFO should be retriggered from the beginning of the cycle. The following modes are available:
  - **Mono Retrig**: The LFO plays the shape in loop, and it is retriggered for a new note only if all previous notes have been released.
  - **Poly Retrig**: The LFO plays the shape in loop, and it is retriggered for every new note.
  - **Rnd Retrig**: The LFO plays the shape in loop, and it is randomly retriggered or not for every new note.
  - **One Shot**: The LFO plays the shape once, and it is retriggered for every new note.
  - **Mono One Shot**: The LFO plays the shape once, and it is retriggered for a new note only if no previous note is still being held.
  - **Env Mode** (Envelope mode): The LFO plays the shape in one direction as long as the note is held, and in the other direction when the note is released. The LFO is retriggered for every new note.
- **Sample Trigger menu** (LFO 1 only): The Sample Trigger menu lets you trigger the held notes at every step of the LFO. The following options are available:
  - **Smpl Trig Off** (Sample Trigger Off): This deactivates the feature. The notes are triggered only as you play them.
  - **As Played**: The notes are triggered on each step in the order that you have played them.
  - **Up**: The notes are triggered on each step from the lowest pitch to the highest.
  - **Down**: The notes are triggered on each step from the highest pitch to the lowest.
  - **Random**: The notes are triggered on each step in a random order.
  - **Chord**: All the held notes are played together on each step.

Envelopes

When the Modulator Type selector is set to **Envelopes** at the top of the Modulation page, the page shows the parameters for the two available envelopes: **Envelope 1** on the left, **Envelope 2** on the right.

Each envelope provides the following elements:

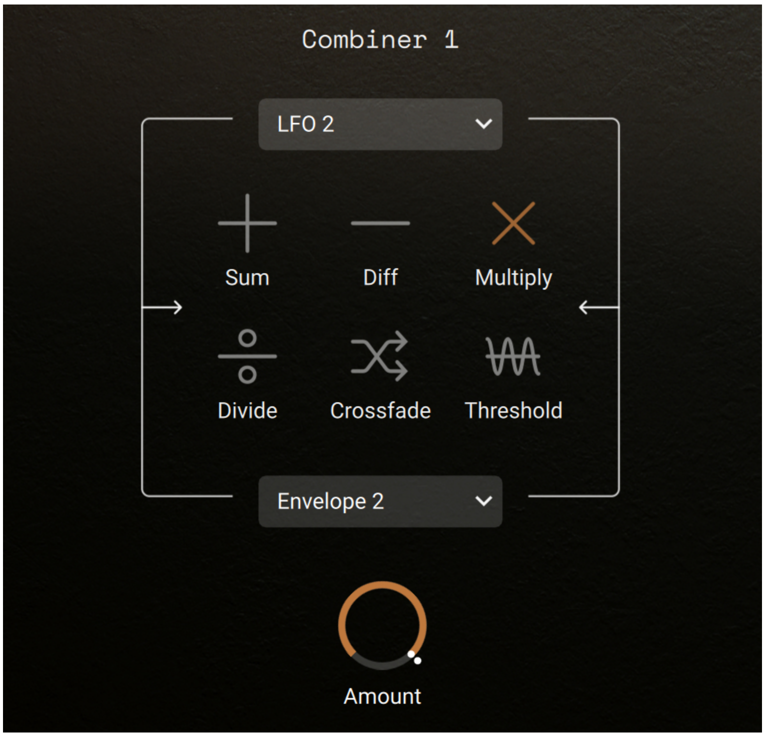
- **Envelope Display**: Shows the shape of the envelope with the current settings.
- **Attack**: Adjusts the time it takes for the envelope to reach its maximum once triggered.
- **Attack Curve**: Adjusts the curvature of the envelope shape during the attack phase. At the default center position, the envelope grows at a steady rate. At negative values the attack phase starts slowly and ends at a faster rate, and inversely at positive values.
- **Decay**: Adjusts the time it takes for the envelope to go from its maximum value to its sustain value.
- **Sustain**: Adjusts the value that the envelope keeps once it has decayed and as long as the note is held. This value is measured as a percentage of the envelope's maximum value: At **100 %** the sustain value is the maximum value, and at **0 %** the sustain value is zero.
- **Release**: Adjusts the time it takes for the envelope to return to zero once the note has been released.

Combiners

When the Modulator Type selector is set to **Combiners** at the top of the Modulation page, the page shows the parameters for the two available combiners: **Combiner 1** on the left, **Combiner 2** on the right.

A combiner lets you generate a new modulation source by merging two existing modulation sources through an operator.

Each combiner provides the following elements:



- **Source 1 menu (top menu)**: Selects the first modulation source to combine. The menu contains the usual modulation sources, except for the combiner itself, and an additional **0 value** entry, which lets you use the zero value with the selected operator.
- **Operator selector**: Selects the operation applied. The following operators are available:
  - **Sum**: Adds the source 2 to the source 1.
  - **Diff**: Subtracts the source 2 from the source 1.
  - **Multiply**: Multiplies the source 1 by the source 2.
  - **Divide**: Divides the source 1 by the source 2.
  - **Crossfade**: Linearly blends between the sources 1 and 2 using the **Amount** knob as mix control. With **Amount** at **0 %** only the source 1 passes through, at **100 %** only the source 2.
  - **Threshold**: Applies a minimum floor to the source 1 using the **Amount** value as the threshold. The values of source 1 below the threshold are raised up to it, the values above pass through unchanged.

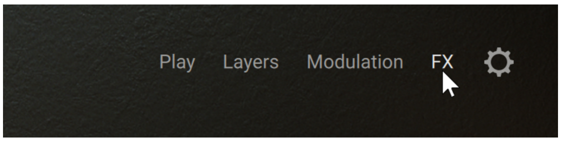
    The **Threshold** operator uses a single modulation source.
- **Source 2 menu (bottom menu)**: Selects the second modulation source to combine. The menu contains the usual modulation sources, except for the combiner itself, and an additional **0 value** entry, which lets you use the zero value with the selected operator. The Source 2 menu is not available with the **Threshold** operator.

- **Amount**: The action of the **Amount** control varies with the selected operator:
  - With the **Sum**, **Diff**, **Multiply**, and **Divide** operators, **Amount** adjusts the overall level of the resulting signal. For example, you can use it to compensate for the very high levels that might be generated by the operator.
  - With the **Crossfade** operator, **Amount** adjusts the blend between both source signals.
  - With the **Threshold** operator, **Amount** sets the threshold level under which the values of the first source are raised.

9. FX page

The FX page of Erosia lets you set up two insert effect chains, a send effect, and a limiter for your layers.

▶ To open the FX page, click the **FX** button in the top right corner of the instrument.



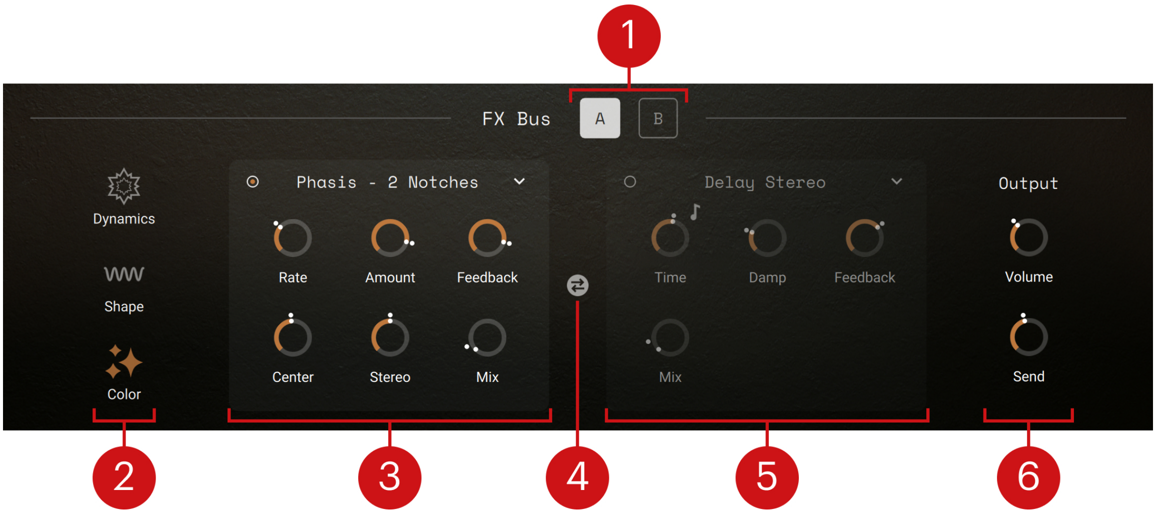
The FX page contains the following sections:

- In the upper part, the **FX Bus** section lets you select and configure the effects in the two available effect busses A and B. Each bus can contain a chain of up to six effects (two effects at three different stages), and you can route each individual layer to either effect bus. Refer to FX Bus section.
- In the lower left part, the **Ambience** section provides a delay/reverb unit available as send effect for the output of the two effect busses above. Refer to Ambience section.
- In the lower right part, the **Limiter** processes the overall sound of the instrument. Refer to Limiter.

FX Bus section

At the top of the FX page, the **FX Bus** section lets you set up the effect chains on the two busses A and B. Each bus includes three stages that can host up to two effects each.

The **FX Bus** section contains the following elements:



1. **FX Bus selector**: Shows the contents of the effect bus **A** or **B**. All the remaining elements in the **FX Bus** section are specific to the bus selected here.
2. **Effect Stage selector**: Clicking **Dynamics**, **Shape**, or **Color** shows the two effect slots of this effect stage.
3. **Effect slot 1**: Hosts one effect and shows its parameters. At the top of the slot, the effect menu shows the name of the loaded effect. You can click that name and choose another effect from the menu, or select **None** to leave the slot empty. In the top left corner, the on/off switch lets you bypass the slot. Different effects are available in the different slots and stages. Each effect provides its own set of parameters.
4. **Swap button**: Swaps the two slots on that stage.
5. **Effect slot 2**: Hosts one effect and shows its parameters. At the top of the slot, the effect menu shows the name of the loaded effect. You can click that name and choose another effect from the menu, or select **None** to leave the slot empty. In the top left corner, the on/off switch lets you bypass the slot. Different effects are available in the different slots and stages. Each effect provides its own set of parameters.
6. **Output section**: Includes controls adjusting the following levels:
   - **Volume**: Adjusts the output level of the effect bus. After this control, the sound is routed both to the **Limiter** and to the **Send** control below.
   - **Send**: Adjusts the level of the signal sent to the delay/reverb unit in the **Ambience** section. As you raise the **Send** value, more of the bus sound will be processed by the delay/reverb unit.

   > The send path is an additional path: The output of each effect bus is routed anyway to the **Limiter** at the specified **Volume** level, no matter if some of it is additionally sent through the **Ambience** section.

Dynamics, Shape, and Color stages

In either effect bus, the sound passes through all three stages from top to bottom, and in each stage through both effect slots from left to right.

In the **Dynamics** stage, the available effects are based on the following Kontakt effects:

- Effect slot 1: The various **Compressor** effects are based on the Solid Bus Comp compressor of Kontakt with different attack settings. The **Transient Master** effects are based on the Transient Master of Kontakt with the Smooth switch on or off.
- Effect slot 2: The various **Saturate** effects are based on the corresponding modes of the Saturate effect in Kontakt.

In the **Shape** stage, the available effects are based on the following Kontakt effects:

- Effect slot 1: The **Tube** and **Transistor** effects are based on the corresponding modes of the Distort effect in Kontakt. The **Bit Reduction** effect is based on the Lo-Fi effect of Kontakt. The two **Skreamer** effects are based on the Skreamer effect of Kontakt, the **Skreamer 50/50** effect having the **Clean** knob turned at full level in Kontakt.
- Effect slot 2: This slots provides the same collection of filters as the **Filter** section of the Layers page. You can find a complete list of the available filters and a description of their parameters in Filter reference.

In the **Color** stage, the available effects are based on the following Kontakt effects:

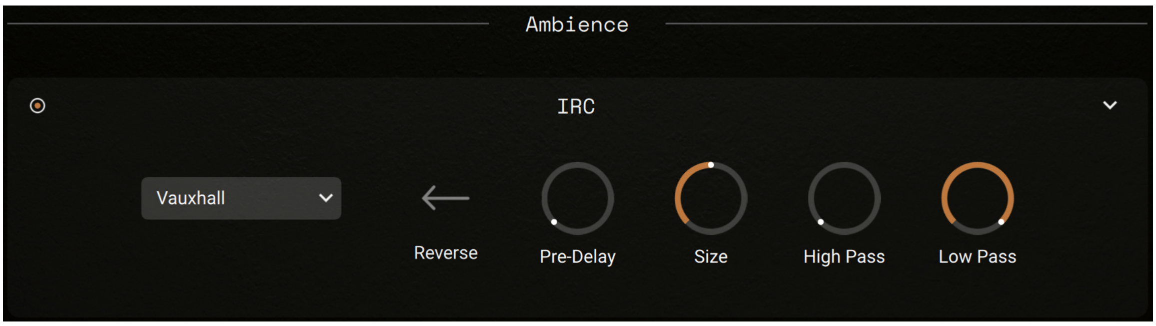
- Effect slot 1: The various effects are based on the corresponding Choral, Phases, and Flair effects of Konkakt.
- Effect slot 2: The various **Delay** effects are based on the Delay effect of Kontakt with various Pan settings.

You can find more information on these effects and a description of their parameters in the Effect reference chapter of the Kontakt user guide.

Ambience section

At the bottom left of the FX page, the **Ambience** section provides a delay/reverb unit available as send effect for the sound coming from the two effect busses A and B above.

The **Ambience** section contains the following elements:



- **On/off switch**: Turns the effect on or off.
- **Effect menu**: Shows the name of the current effect at the top of the slot. You can click that name and choose another effect from the menu, or select **None** to leave the slot empty.
- **Effect parameters**: Each effect provides its own set of parameters.

The effects available in the Effect menu are based on the following Kontakt effects:

- The various **Replika** effects (**Modern**, **Analog**, **Tape**, **Vintage**, **Diffusion**, and **Diffusion (Dense)**) are based on the corresponding modes of the Replika Delay in Kontakt. The **Diffusion (Dense)** effect uses the Diffusion mode with the **Dense** switch turned on in Kontakt.
- The **IRC** effect is based on the Convolution effect of Kontakt, with an additional menu providing a selection of impulse responses.

You can find more information on these effects and a description of their parameters in the Effect reference chapter of the Kontakt user guide.

Limiter

At the bottom right of the FX page, the **Limiter** works as a final processor on your signal chain. It has two key functions: to ensure that the signal level stays below 0 dB, thus avoiding digital clipping, and to increase the overall perceived volume of the signal. It is acting as a final means of protecting your signal from clipping without imposing too much coloration to your sound. Reducing the **Threshold** will reduce the dynamic range of the signal and boost the overall volume. However, setting the **Threshold** too low can result in a squashed and dull sound.

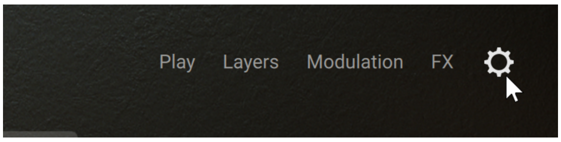
The **Limiter** contains the following controls:

- **On/off switch**: Turns the effect on or off.
- **Threshold**: Sets the threshold above which the limiter kicks in. Simultaneously, the signal will be automatically amplified to take advantage of the headroom made available by the limiting. To simply prevent the signal from clipping, leave the **Threshold** at 0 dB. To make the signal louder, reduce the **Threshold** value by turning the control to the left. The available values range from -40.0 dB to 0.0 dB.
- **Gain Reduction meter**: Displays the amount of gain reduction. Limiting works best if this meter responds only to occasional level peaks. Permanent gain reduction indicates that the **Threshold** is set too low.

10. Options page

The Options page of Erosia lets you adjust various settings affecting the pitch, voicing, engine, and response of the instrument.

▶ To open the Options page, click the cogwheel icon in the top right corner of the instrument.



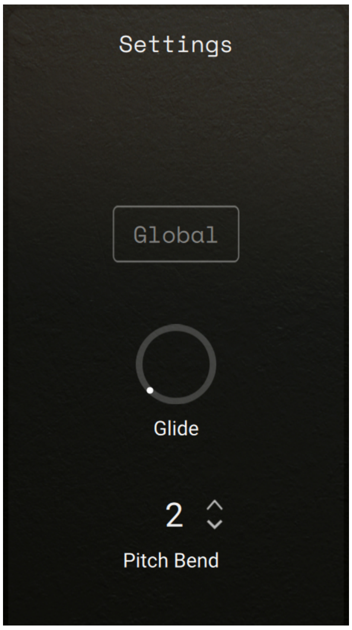
The Options page contains the following elements:

- **Voices**: Adjusts the number of voices that the instrument can play simultaneously. If you play more notes at once than the **Voices** value, the oldest notes are cut. The maximum **Voices** value is **64** and its minimum is **Mono** (one single voice). If you select **Legato**, the instrument uses a single voice as with **Mono**, but it will retrigger the previous note if it is still being held as the current note is released.
- **Engine Resolution**: Selects an engine resolution. **Max** provides the best audio results but uses more CPU resources, **Eco** uses the CPU more sparingly, and **Enhanced** provides a compromise between audio quality and CPU consumption.
- **Settings section**: Contains global and pitch-related options. Refer to Settings section.
- **Velocity**, **Key Tracking**, **Mod Wheel**, and **Aftertouch sections**: These sections let you adjust how the instrument responds to the velocity, key pitch, modulation wheel, and aftertouch data, respectively. Refer to Response curves.

Settings section

In the left part of the Options page, the **Settings** section contains global and pitch-related settings.

The section contains the following elements:



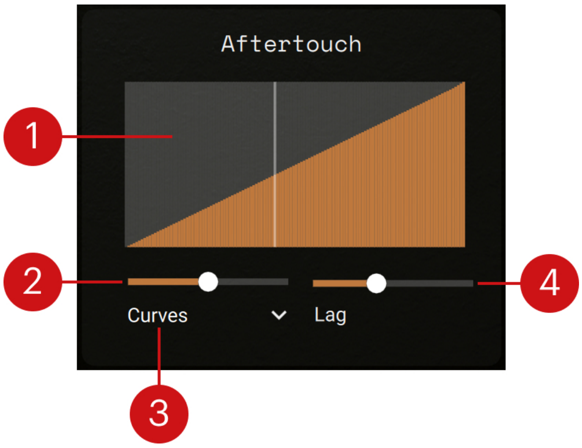
- **Global**: When **Global** is active, the **Glide** setting, the **Pitch Bend** settings, and the four response curves keep their current values when you load another factory or user preset. This allows the instrument to respond in the same way to your MIDI keyboard as you switch between instrument presets. When **Global** is inactive, all your setting adjustments apply to the current preset only, and their values will change when you will load another preset.
- **Glide**: Adjusts the duration of the pitch sweep between the current note and the next one. The available values range from 0 to 10 seconds.
- **Pitch Bend**: Adjusts the range of the Pitch Bend wheel above and below the note pitch, in semitones. The available values are **0** (no pitch bend), **1**–**12** (one octave), and **24** (two octaves).

Response curves

The **Velocity**, **Key Tracking**, **Mod Wheel**, and **Aftertouch** response curves let you adjust how the instrument responds to the note velocity, note pitch, modulation wheel, and aftertouch data, respectively, that are received from the outside world. For example, you could adapt some of these response curves to your own playing or to your particular MIDI keyboard.

> Each response curve affects a signal that is available as modulation source in Erosia, thus the response curve will also indirectly affect the behavior of any parameter modulated by this source.

The four response curves provide similar editing tools:



1. **Response Display**: Shows the current response curve. The horizontal axis represents the input values (for example, coming from your MIDI keyboard), and the vertical axis represents the resulting values in the instrument. The white vertical line indicates the current input. You can freely adjust the curve by clicking and dragging your mouse in the display. You can click or click-and-drag anywhere in the display to manually edit the curve. When you do this, the Mode selector below automatically switches to **Custom**.

2. **Curvature slider**: Adjusts the curvature of the shape. At the default center position, there is no curvature and the original shape is left untouched. Negative values (slider in the left half) will pull the shape downwards, positive values (slider in the right half) will pull it upwards.

3. **Shape selector**: Selects a predefined or a free shape. Selecting **Curves** resets the shape to a diagonal line, for which the input and the output are proportional. Selecting **Custom** recalls the last shape that you have manually edited. If you manually edit a predefined shape, the Shape selector automatically switches to **Custom**.
   Velocity section only: Selecting **Fixed** resets the shape to a horizontal line, in which the output velocity keeps the same value for all input velocities. In this particular case, the Curvature slider adjusts the value of the fixed velocity.

   > Whichever entry you select in the Shape selector, the resulting shape will be affected by the current Curvature value.

4. **Lag slider** (**Aftertouch** section only): Adjusts a lag before the incoming aftertouch data starts being read. For some presets this can be useful to let the attack of the sound develop before the aftertouch data kicks in and modifies the sound.

11. Modulating your sound

Erosia makes an extensive use of parameter modulation. The instrument lets you build advanced modulating schemes in an intuitive way and generate continuously evolving sounds that you can play dynamically.

Modulation is the modification of a parameter value relative to its original value. For example:

- An envelope modulates the amplitude of the sound when a note is played, with different stages (like attack, decay, sustain, and release) that modify the sound level relative to the original velocity of the played note.
- An LFO (low-frequency oscillator) modulates the pitch of a sound around the original pitch of the played note.

A modulation is specified by its **target** (which parameter is affected), its **source** (where does the modification come from), and its **amount** (how much the target value is affected by the source).

The following sections describe how to create and edit a modulation and provide a list of the available modulation sources in Erosia.

Creating and editing a modulation

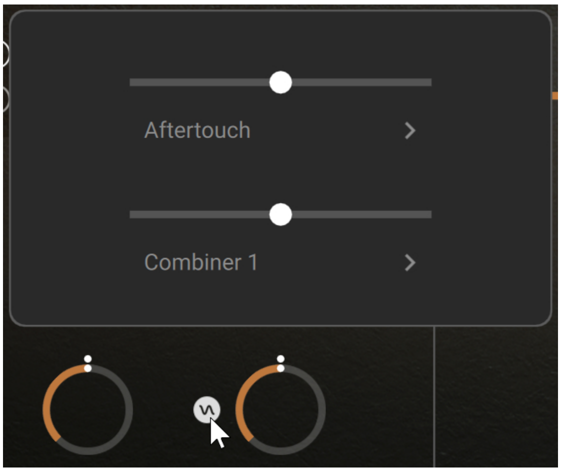
In Erosia you can use the same simple workflow to modulate most continuous parameters in the form of a rotary knob or a slider.

To create or edit a modulation:

1. Navigate to the page and tab containing the control that you want to modulate.
2. Hover with your mouse over the rotary knob or slider.

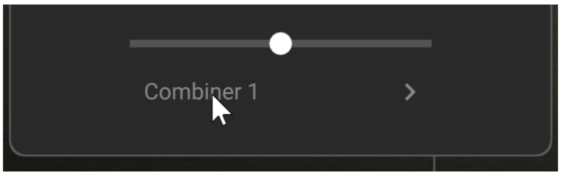
→ A Modulation icon appears next to the control:

3. Click that Modulation icon to open the Modulation panel:

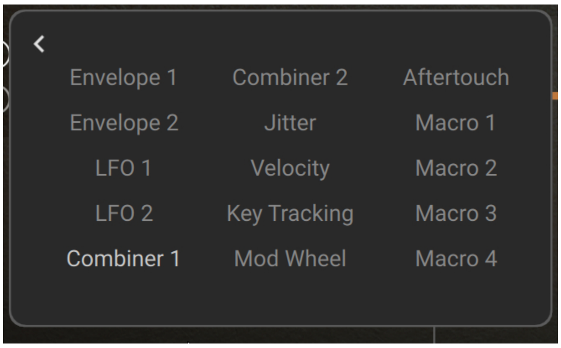


→ The Modulation panel shows the two modulations currently assigned to this control. Each modulation includes a modulation source and an Amount slider. When a slider is at the center, that modulation is inactive.

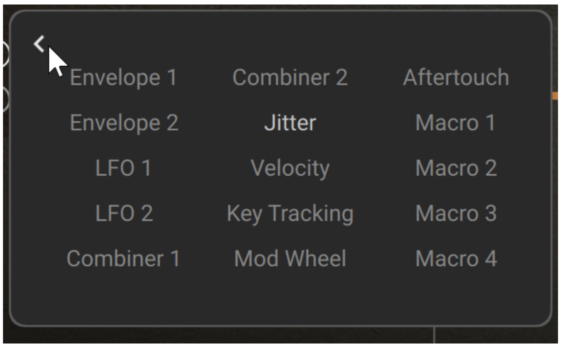
4. If you want to replace a modulation source, click its name.



→ The Modulation Source panel opens up and lists the available modulation sources, with the current source highlighted.



5. Select another source, and click the arrow in the top left corner to close the Modulation Source panel.



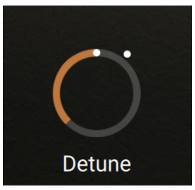
→ The Modulation panel now shows your new source.

6. Adjust the Amount slider of the modulation to fit your needs. Any value other than zero (center position) will activate the modulation.

→ On the rotary knob, a gray outer ring (or a segment for sliders) indicates the global amount of modulation for that control.

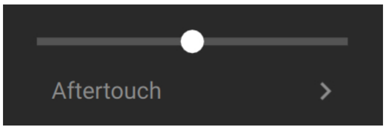
7. Click the Modulation icon again to close the Modulation panel.

→ You new modulation is now active. When you play a note, the second dot outside the knob shows the modulated value in real time:



> All the rotary knobs that are available for modulation will show the second dot outside the knob. With no active modulation, this dot simply follows the knob's main cursor. With an active modulation, the dot shows the modulated value, which is set relative to the value of the main cursor.

For each modulation source, the Modulation panel provides the following controls:



- **Modulation Amount slider**: Adjusts the intensity of the modulation, that is, how much this modulation source will affect the parameter. This defines the range of the modulation values. With the slider in the center, the modulation source has no influence over the parameter. The current value appears as you drag the slider. The further you drag the slider to the left or to the right, the stronger the modulation will be. With the slider on the right side, increasing the modulation value will increase the parameter value (direct ratio). With the slider on the left side, increasing the modulation value will decrease the parameter value (inverse ratio). You can reset the slider and remove the modulation by [command]+clicking (Mac) or [Ctrl]+clicking (Windows) the slider.
- **Modulation Source name**: Indicates the selected modulation source. Clicking the name or the right arrow opens the Modulation Source panel, where you can select another source.

Available modulation sources

The following modulation sources are available:

- **Envelope 1** and **Envelope 2**: Internal modulators whose configuration is described in Envelopes.
- **LFO 1** and **LFO 2**: Internal modulators whose configuration is described in LFOs.
- **Combiner 1** and **Combiner 2**: Internal modulators whose configuration is described in Combiners.
- **Jitter**: Random values.
- **Velocity**: How hard you hit the keys.
- **Key Tracking**: The pitch of the played keys.
- **Mod Wheel**: Modulation wheel.
- **Aftertouch**: The pressure that you apply on held keys. This can be polyphonic (a distinct value for each individual note) if it is available on your MIDI keyboard.
- **Macros** (**Timbral Fusion**, **Tone Sculptor**, **Kinetic Flux**, and **Effectuator** controls): Located at the bottom of the Play page, these four knobs respectively appear as **Macro 1**–**4** in the Modulation Source panels.

> Some of the modulation sources listed above might not be available for specific control elements for conflicting reasons: For example, the **Transform** control cannot be modulated by any of the Macros, since it does control these very Macros through the Macro Variations.

12. User presets

You can save and manage your own user presets for Erosia. When you save a user preset, all the parameter adjustments and instrument settings are stored within the preset. User presets let you save your own sounds, use them in other projects and across computers, share them with other users, or create backups.

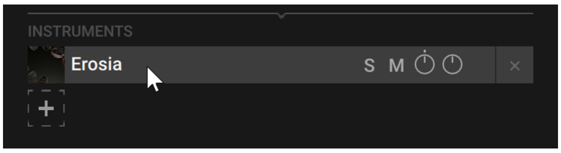
User presets are saved on your computer as Snapshots (.nksn file extension), which are Kontakt's underlying file format for instrument presets.

Saving a user preset

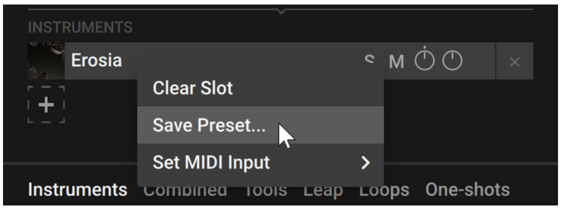
You can save user presets to recall your favorite sounds and settings at any time, share them with others, or create backups.

When using Kontakt's Default view, you can save user presets using the Navigator in the side pane:

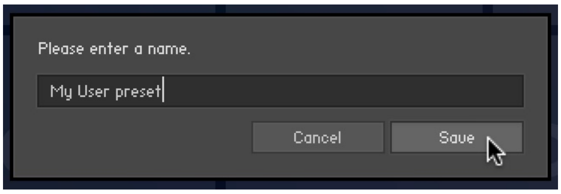
1. In the Navigator, right-click the Erosia slot.



2. Click **Save Preset…** in the context menu to open the Save dialog.



3. Enter a name for your new user preset and click **Save**. If you enter the same of an existing user preset, you will be given the option to replace it by clicking **Overwrite**.



→ The user preset is saved and added to your user content in the library.

> You can also save user presets using the Instrument Header. For more information, refer to the Kontakt user guide or the Kontakt Player user guide.

All the user presets are automatically stored in the default User Content folder. You can transfer any of your presets to another computer by copying the respective Snapshot files.

The default User Content folders are:

Mac OS X:	*Macintosh HD/Users/<User Name>/Documents/Native Instruments/User Content/*

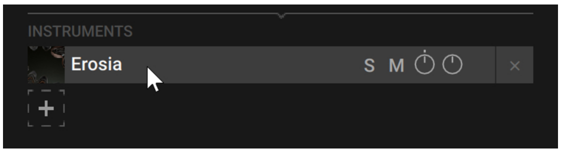
Windows: *C:\Users\<User Name>\My Documents\Native Instruments\User Content*

Please make sure that you include your *Documents* / *My Documents* folder in your regular data backups.

Loading a user preset

When using Kontakt's Default view, you can load user presets using the Navigator and the browser in the side pane:

1. In the Navigator, click the Erosia slot.



2. In the side pane browser below, activate the User Content button to show your user presets.

→ The Results list below displays the user presets available for Erosia.

3. Double-click the desired user preset from the list to load it.

You can also load user presets and factory presets using the Kontakt Browser or the Instrument Header. For more information, refer to the Kontakt user guide or the Kontakt Player user guide.

Deleting a user preset

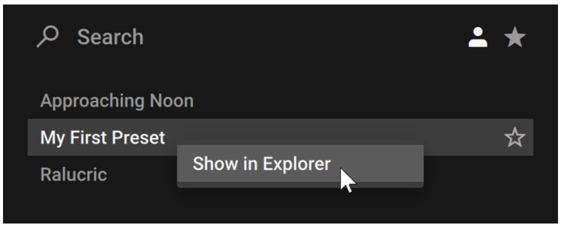
When using Kontakt's Default view, you can delete user presets using the Navigator and the browser in the side pane:

1. In the Navigator, click the Erosia slot.

2. In the side pane browser below, activate the User Content button to show your user presets.

→ The Results list below displays the user presets available for Erosia.

3. Right-click the desired user preset from the list and select **show in Finder/Explorer**.



→ The folder containing this user preset opens up on your desktop.

4. Delete the file from the disk.

→ The user preset is removed from your library on the next launch of Kontakt / Kontakt Player.

> You can also delete user presets using the Kontakt Browser or the Instrument Header. For more information, refer to the Kontakt user guide or the Kontakt Player user guide.

13. Filter reference

The Filter menus of Erosia offer a large selection of filters. These filters can be split into two groups:

- The upper part of the Filter menus contains **classic filters**: **Ladder LP1**, **Ladder LP2**, **Ladder LP4**, **SV LP2**, **SV LP4**, **Ladder HP1**, **Ladder HP2**, **Ladder HP4**, **SV HP2**, **SV HP4**, **Ladder BP2**, **Ladder BP4**, **SV BP2**, **SV BP4**, **Ladder Peak**, and **Par LP/HP** filters. Refer to Classic filters.
- The lower part of the Filter menus contains **creative filters**: **Phaser**, **Vowel 1**, **Vowel 2**, **Formant 1**, **Formant 2**, **Spectral 1** to **5**, and **Spectral Frq Bst** filters. Refer to Creative filters.

> The Filter menus are available in the Filter section of the Layers page and in the FX Bus section of the FX page.

Classic filters

The classic filters include the **Ladder LP1**, **Ladder LP2**, **Ladder LP4**, **SV LP2**, **SV LP4**, **Ladder HP1**, **Ladder HP2**, **Ladder HP4**, **SV HP2**, **SV HP4**, **Ladder BP2**, **Ladder BP4**, **SV BP2**, **SV BP4**, **Ladder Peak**, and **Par LP/HP** filters.

The name of each classic filter provides information about its characteristics:

- Algorithm:
  - **SV** (State Variable) filters have a clean character.
  - **Ladder** filters are based on the classic ladder circuit used in early synthesizers.
- Attenuated frequencies:
  - **LP** (lowpass) filters attenuate harmonics and signals above the cutoff frequency.
  - **HP** (highpass) filters attenuate signals below the cutoff frequency.
  - **BP** (bandpass) filters attenuate signals above and below the cutoff frequency.
  - **Peak** filters are bandpass filters with a very narrow band.
  - The **Par LP/HP** filter is the combination of a highpass filter and a lowpass filter routed in parallel (the signal is split and routed through each separately). As a result the filter removes a specific frequency band from the signal.
- Steepness of the attenuation curve:
  - **1-pole and 2-pole filters** (for example, **LP1** or **BP2**) have a rather gentle slope that preserves more of the original sound.
  - **4-pole filters** (for example, **LP4** or **BP4**) affect the sound more frankly on their dedicated frequency range.

All the classic filters provide the following controls:

- **Cutoff**: Adjusts the frequency below and/or above which signals will be attenuated.
- **Resonance**: With values greater than 0, this control will boost a small frequency range around the cutoff frequency.
- **Key Tracking**: Adjusts to which extent the **Cutoff** value follows the pitch of the played key.

Creative filters

The creative filters include the **Phaser**, **Vowel 1** and **Vowel 2**, **Formant 1** and **Formant 2**, **Spectral 1** to **5**, and **Spectral Frq Bst** filters.

Phaser

The **Phaser** filter creates a distinct comb filter effect by using an all-pass filter design that radically alters the phase relations in your signal. It provides the following controls:

- **Cutoff**: Adjusts the center working frequency of the phaser's comb filter effect. Changing this parameter will alter the tonality of your sound in a distinct and not always easily predictable way.
- **Resonance**: Adjusts the depth and narrowness of the notches that the phaser imposes on the frequency spectrum, and thereby the intensity of the effect.
- **Key Tracking**: Adjusts to which extent the **Cutoff** value is following the pitch of the played key.

Vowel 1 and Vowel 2

The **Vowel 1** and **Vowel 2** filters simulate the resonant frequencies of the human vocal tract in order to produce vowel-like sounds. They provide the following controls:

- **Cutoff**: Adjusts the center frequency of the filter. Various distinct frequencies across the spectrum will produce different vowels.
- **Resonance**: With values greater than 0, this control will emphasize the frequencies around the center frequency in order to create a sharper sound and enhance the effect.
- **Key Tracking**: Adjusts to which extent the **Cutoff** value is following the pitch of the played key.

Formant 1 and Formant 2

The **Formant 1** and **Formant 2** filters are also designed to mimic the frequency response of the human vocal tract. These filters can be used to emulate the “talk box” effect. They provide the following controls:

- **Talk**: Controls the shape of the frequency response. This control can be used to morph between vowel-sounds.
- **Size**: Controls the center of the frequency response, analogous to the **Cutoff** control of the other filters.
- **Key Tracking**: Adjusts to which extent the **Size** value is following the pitch of the played key.

Spectral 1–5 and Spectral Frq Bst

The **Spectral 1**–**5** and **Spectral Frq Bst** (Frequency Boost) filters are based on a 3-band equalizer (EQ). They use movement of the EQ frequency and gain parameters to sweep across the spectrum, creating dynamic tonal shifts. They have different movement shapes and ranges, so that each one creates its own distinct sweeping pattern. The **Spectral Frq Bst** filter type uses only one band to boost or cut the specified frequency range. These filter types provide the following controls:

- **Freq**: Sets the center frequency in the spectrum where the filter is focused.
- **Gain**: Adjusts the emphasis or attenuation at the **Freq** value.
- **Key Tracking**: Adjusts to which extent the **Freq** value is following the pitch of the played key.

14. Credits

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