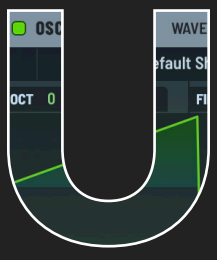
SERUM 2



User Guide

Serum 2

Serum 2 plug-in for Microsoft Windows and Apple macOS
A synthesizer instrument plug-in for VST3, Audio Unit, and AAX hosts

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W W W . X F E R R E C O R D S . C O M

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Welcome

Welcome to the *Serum 2 User Guide*. Serum 2 is an advanced virtual synthesizer that offers a unique blend of sound generation methods together with a powerful modulation system.

Note: Throughout this manual, we'll refer to Serum 2 (the latest version of Serum) simply as Serum.

Serum combines wavetable, subtractive, multi-sampled, sampled, granular, and spectral synthesis into an intuitive and cohesive workflow. This manual helps you get the most out of your instrument.

Serum Main Screen

Registering Serum

If you purchased Serum through Xfer Records, you are automatically registered for free updates.

If you purchased Serum through Splice.com, registration is automatic with the email you used at the time. If you are having trouble locating your registration on **`xferrecords.com`**, use the contact form and mention your Splice username.

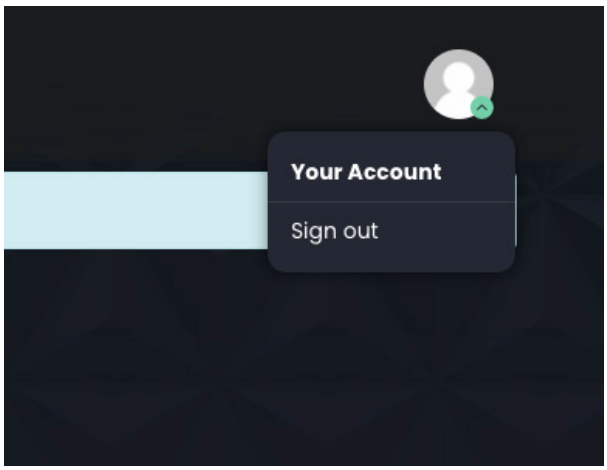
Getting in Touch

If you have any feedback about Serum, wish to pass on your comments regarding the software or this manual, or would like to send links to cool tunes made with Serum, please reach out to:

```
https://support.xferrecords.com
```

Downloading Serum

To download Serum, navigate to **`www.xferrecords.com`**, sign in to your account, and choose **Your Account** in the user menu.



Xfer Records User Menu

The account page appears.

All installers for your registered Xfer Records products are available on this page.

Download the installer for your operating system.

Installing Serum

Serum is available for both Apple macOS and Microsoft Windows. Serum is designed to be used within a host audio application that supports the VST3, AU, or AAX plug-in formats.

Important: You *must* select the VST3 version when installing the AU or AAX version of the plugin.

Installing on macOS and Windows-based PCs

1. Navigate to the folder to which you downloaded the software and double-click the file to launch the installer.

   A wizard guides you through the installation.

2. Follow the steps and complete the installation.

3. Open your DAW and rescan/refresh the plug-ins.

   Refer to the documentation that came with your DAW for details.

Uninstalling Serum on Windows-based PCs

1. Delete the Serum 2 folder from the following folder:

   `C:\Users\(username)\AppData\Roaming\Xfer\`

2. Delete the **Serum 2 Presets** folder.

   If you have Microsoft OneDrive enabled, delete the **Serum 2 Presets** folder in the following location:

   `C:\Users\(username)\OneDrive\Documents\Xfer\`

   Otherwise, delete the **Serum 2 Presets** folder in the following location:

   `C:\Users\(username)\Documents\Xfer\`

Note: Serum does not modify the Windows registry, therefore, you do not need to run an uninstaller application to completely remove Serum from your system.

Uninstalling Serum on macOS Computers

To uninstall Serum on macOS, remove the Serum 2 files and folders from the following locations:

- **`/Library/Audio/Plug-Ins/VST3/`**
- **`/Library/Audio/Plug-Ins/Components/`**
- **`/Library/Audio/Presets/Xfer Records/`**
- **`/Library/Application Support/Avid/Audio/Plug-Ins/`**
- **`~/Library/Preferences/Serum2Prefs.json`**
- **`~/Library/Application Support/com.xfer.serum2/Serum2.lic`**

Press **Cmd-Shift-G** to navigate to each folder quickly.

Exploring Serum

Serum is an incredibly versatile instrument designed for sound designers and musicians who seek creative control over every detail of their sound. Equipped with multiple oscillators, including dedicated noise and sub oscillators, Serum allows you to blend and shape a broad range of waveforms, from complex wavetables to granular, spectral, and sampled sounds.

Whether you're designing cutting-edge textures or classic synth tones, Serum offers the flexibility to explore both familiar and experimental sonic territories with ease.

In addition, Serum features dual filters, each offering a wide selection of filter types for precise tonal shaping. You can route oscillators through one or both filters, giving you detailed control over the harmonic structure and timbre of your sound.

Serum Interface

Complementing this, Serum includes an array of fully-configurable envelopes and LFOs (Low Frequency Oscillators) that you can assign to almost any control. This allows for intricate, evolving soundscapes with dynamic changes and rhythmic modulations.

To further expand the sonic possibilities, Serum integrates a sophisticated arpeggiator and MIDI clip system, offering advanced sequencing and performance options for live and studio use.

Taken together, the Serum interface is conveniently structured to help make sound design both easy and intuitive.

Exploring Sound Design in Serum

Serum provides you with multiple ways to create and manipulate sound, enabling you to design virtually any kind of sound you can imagine (and some you may not have thought possible).

Specifically, Serum offers you the following sound generation options, accessible through oscillators A to C (**OSC A**, **OSC B**, and **OSC C**), with each offering a distinct approach to sound sculpting:

- **Wavetable** — Produces an evolving soundscape by cycling through or morphing between multiple waveforms stored in a wavetable
- **Multisample** — Uses multiple recordings (samples) of instruments (or other sound generators) across various notes and dynamic ranges
- **Sample** — Uses single samples that can be played back at different pitches, layered, looped, or transformed with various effects
- **Granular** — Layers, stretches, rearranges, or modulates tiny sound fragments called grains to create new sound textures
- **Spectral** — Manipulates sound by analyzing and resynthesizing the frequency spectrum, creating new sounds with independent control over time and pitch

You can access the sound generators using a drop-down menu.

As you read through this manual, you'll learn how to use each option in detail to help craft the sounds you want.

Serum Oscillators

Getting Started

Getting started with Serum is quick and easy. This section provides a short overview to help you understand some common basic operations.

Adding Serum to a Track

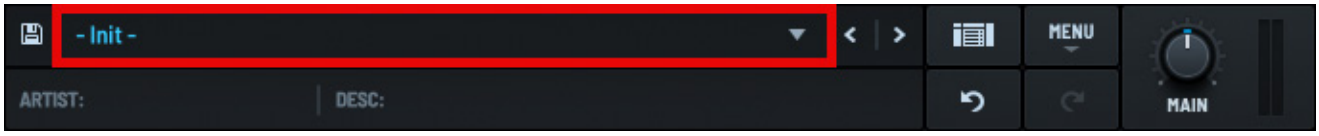
You can add Serum to an instrument (MIDI) track in your DAW using the same procedure that you use with any software instrument. At this point, sending MIDI notes to Serum should trigger the default sawtooth sound.

Similarly, you should hear the sawtooth sound if you click any of the piano keys at the bottom of the Serum user interface (UI).

Note: If you are not sure how to add an instrument to a track, refer to the documentation that came with your DAW.

Loading a Serum Preset

Serum lets you create amazing sounds from scratch. However, a good place to start is browsing and loading presets. To select a preset, click **- Init -** to display the Serum presets menu.



Serum Presets Menu

Navigate the menu to choose a preset.

After choosing a preset, you can click the **< >** arrows (to the right of the preset name) to navigate through a particular preset subfolder without having to repeatedly display the presets menu.

You can also choose a preset by clicking the  button (next to the main menu near the top right) to access the presets browser.

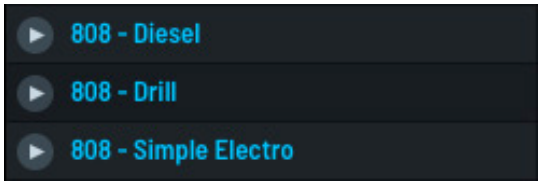
Click an entry in the list to load the corresponding preset. Most presets load immediately; presets with larger embedded samples, such as multisampled instruments, display a small green progress bar (directly beneath the preset name) when loading.

Serum Presets Browser

You can preview presets to help you quickly find the right sound.

Click the (play) button to preview the corresponding preset.

Serum plays an embedded preview clip (MIDI sequence) if the preset designer specified one. Otherwise, Serum plays a "fallback" clip to give you a sense of the preset.

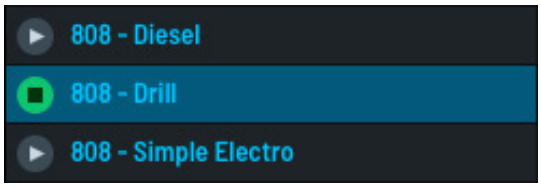


Presets Showing Play Buttons

You can choose the fallback clip from among three standard options. Serum also allows you to auto-preview clips. See "Performing Standard Preset Operations" on page 334 for more information.

Click another play button to preview the corresponding preset.

Click the button to stop the preview.



Preset Playing

Creating a New Sound

In addition to exploring Serum's rich set of factory presets, you can use Serum to create your own sounds from scratch.

If you just added Serum to a track, you will see the **- Init -** preset, which has a single wavetable oscillator enabled (**OSC A**) together with basic envelope and LFO modulators defined. This is a perfect blank slate from which to craft and evolve your sound.

Note: If you've already loaded a preset or made other changes and would like to start over, click the main menu and choose **Init Preset** in the context menu.

Choose Init Preset from the Main Menu

This initializes Serum to the default settings and sets the stage for you to work on your new sound.

For more information about all the options available in the main menu, see "Appendix A: Using the Main Menu" on page 323.

Saving Changes

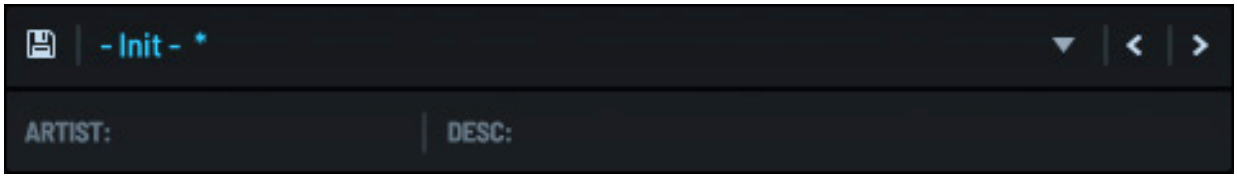
After crafting a new sound or modifying an existing preset, you can save the configuration as a new preset (that you can load later).

1. (Optional) Before saving a preset, add metadata to the preset.

   a. Double-click the **ARTIST** field directly below the preset name and type the artist name.

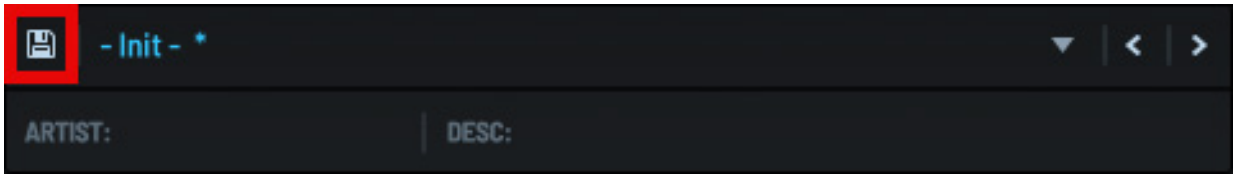
   b. Double-click the **DESC** field and add any relevant information related to the preset.

   You can add any type of information that you consider helpful or informative.



Preset Metadata (Artist and Description)

2. Click the (disk) button to the left of the preset name.



Save (Disk) Button

   A dialog appears allowing you to specify the file name and choose the location. Note that this file name becomes the preset name in Serum.

3. Click the **Save** button.

After you've already saved a preset and made modifications, click the (disk) button to save your changes. A dialog appears allowing you to type a new file name or overwrite your existing preset file.

If you already know that you want to use the same file name (and overwrite the existing preset file), Option-click (macOS)/Alt-click (Windows) the disk button. The preset saves using the same file name without displaying the dialog.

Embedding Content When Saving a Preset

You can embed a wavetable or sample into your preset when saving by clicking the  button.

After clicking, the button changes  to show that the feature is enabled.

Note: Alternatively, you can click the name of the wavetable or sample and choose **Embed in Preset** in the context menu.

In general, you should only consider doing this if you intend to share your preset with someone else.

Note that the option to embed a wavetable or sample into a preset is not available for factory-supplied content.

Embed Indicator

Dragging Audio to Your DAW

You can quickly export the last played note or chord (in WAV format) to your DAW.

Hover over the left of the Serum logo, and drag the wave icon that appears to an audio track in your DAW.

Exporting Audio by Dragging

Serum silently plays the note or chord using the corresponding pitch, velocity, and duration.

In addition to inserting the audio on to the track, Serum also captures the exported audio as a WAV-format file (stored in the **`Serum Presets > Renders`** folder).

Dragging Audio to Your DAW

Option-click (macOS)/Alt-click (Windows) the wave icon to open the **`Renders`** folder. Shift-drag the wave icon to copy the current (saved) preset to the macOS Finder/Windows Explorer.

Exploring Basic Operations

Serum includes a set of elements and on-screen controls designed to closely replicate the experience of using a hardware synthesizer, while also providing all the benefits of a digital computer.

This section describes common operations when using the on-screen controls, including options that you can use with most knobs and sliders. Subsequent chapters will describe specific options for individual controls and other elements of the interface.

Displaying Help (Tooltips)

You can display help for any control (such as a knob) by hovering the mouse pointer over the control and pausing for a moment.

For example, hovering over a regular control displays a tooltip similar to the one shown here.

This allows you to see information about a particular control or feature without having to open this manual.

Important: The **Help tooltips** global preference needs to be set to **SHOW** for tooltips to appear. Tooltips are enabled by default when you first install Serum.

Refer to “Exploring Global Settings” on page 312 for more information about modifying global preferences.

Tooltip Help

Tooltip Showing Modulation Sources

Hovering the mouse pointer over a control that has modulation assigned displays the modulation sources.

For example, hovering the mouse over the **CUTOFF** knob shows that **LFO 3** and **ENV 2** are assigned to modulate the filter cutoff.

Using the Serum Keyboard

Serum features an on-screen keyboard that you can use to directly play notes using your mouse, or monitor the notes being played on an external MIDI or computer keyboard. The on-screen keyboard also indicates which notes are being played through the Serum clip player and arpeggiator.

Serum Keyboard

You can also use the keyboard to specify basic settings including the following:

- TRANSPOSE — The number of semitones (positive or negative) to transpose all incoming and generated MIDI notes.
- KEY — The key to use throughout Serum, most notably in the CLIP and ARP modules.
- SCALE — The scale to use. Notes outside the scale automatically conform to the selected scale.
- SWING — Use to swing or shuffle the timing of the grid in the CLIP and ARP editors.
- OSC MAPPING — Edit the note and velocity ranges of the oscillators and arpeggiator.

Using Knobs and Sliders

To adjust a knob or slider, click and drag either up and down or left and right. A pop-up displays the current value allowing you to dial in a specific setting. Hold the **Shift** key to fine tune the adjustment.

If your mouse has a scroll wheel, you can also use it to adjust values up and down (without displaying the current value in a pop-up).

Double-click a knob or slider to display a text box showing the current value. Enter a new value for precise adjustments.

You can change the behavior of double-clicking a knob or slider to have Serum reset the control to its default setting, if you prefer.

See “Exploring Global Settings” on page 312 for more information.

Click-Dragging a Knob

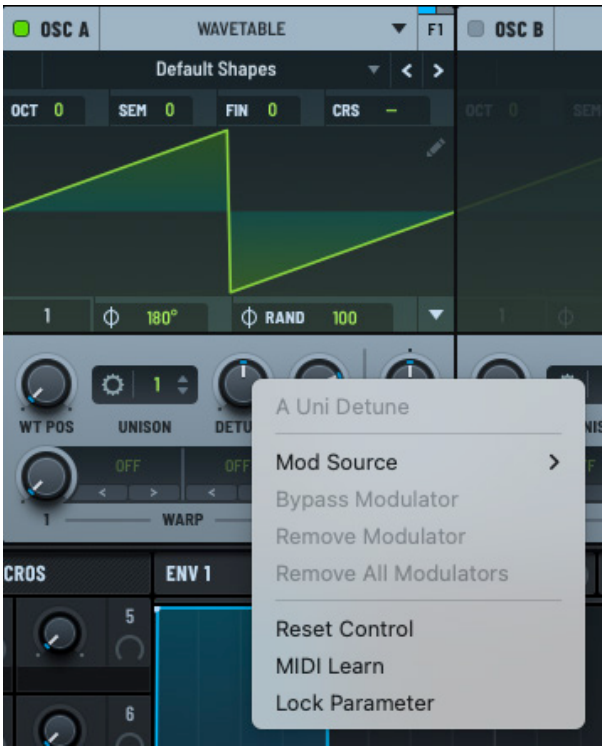
Right-click a knob or slider to open a context menu that displays the settings and operations available for that control.

You can use the menu to choose one or more modulation sources (such as **ENV 1** or **LFO 2**) for the control.

You'll read about modulating parameters later in this guide.

You can also choose to bypass or remove a modulator, or remove all modulators assigned to the control.

Note: Some controls offer additional menu options, depending on the context. You'll learn about these options in relevant sections of this guide.



Right-Click Context Menu

You'll also see the following options available on nearly all knob and slider menus:

- Reset Control — Resets the control to the default value.

  This is the same as Cmd-clicking (macOS) or Ctrl-clicking (Windows) the control.

- MIDI Learn — Activates MIDI learn mode. When enabled, Serum waits for an incoming MIDI CC value.

  After Serum receives a MIDI CC value, MIDI learn mode is deactivated and the CC# is assigned to the knob or slider. Note that the assignment is saved with the preset (patch).

- Lock Parameter — When enabled, locks the control setting (preventing a value change) when loading presets. You can, however, continue to adjust the control manually.

The **Reset Control** and **Lock Parameter** options appear in the context menu of almost every control in Serum.

In all cases, you can use these options to reset the control to the default value and lock a control parameter to prevent it from changing when loading presets, respectively.

When saving a DAW session, MIDI CC assignments are saved and recalled with the session.

When saving a preset, MIDI CC assignments are saved with the preset, but are only loaded if the **Load MIDI Map from Preset** preference is enabled on the **Global** page (this setting is disabled by default).

See “Preferences” on page 313 for more information about setting global preferences.

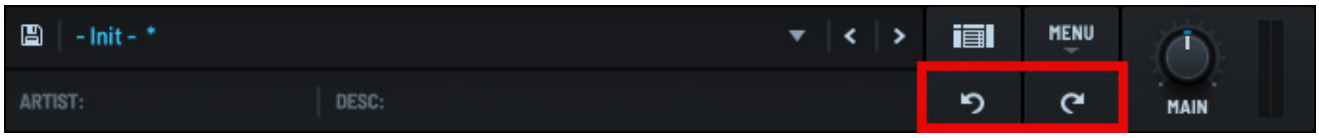
You can set the current MIDI CC assignments to load by default by choosing **Save MIDI Map** in the main menu and saving the MIDI map as **`default.SerumMIDIMap`** in the **`Serum 2 Presets/System/MIDI CC Maps`** folder.

This map then loads automatically when creating a new instance of Serum or choosing **Init Preset** in the main menu. The map also automatically loads when selecting a preset if the **Load MIDI Map from Preset** preference is disabled.

Undo and Redo

You can undo and redo just about any operation in Serum, encouraging you to effortlessly experiment in your sound design.

You can access the undo and redo buttons in the Serum header, near the top right.



Undo and Redo Buttons

Click the  button to undo your last operation (such as a change to a knob or an LFO shape, among others).

Click the  button to redo the last undo operation. This allows you to quickly compare and evaluate changes to your sounds side by side.

Controlling the Main Output Volume

You can control the main output volume of Serum using the **MAIN** knob (near the top right).

The stereo volume appears as a meter next to the knob.

Main Volume

Using Oscillators and Filters

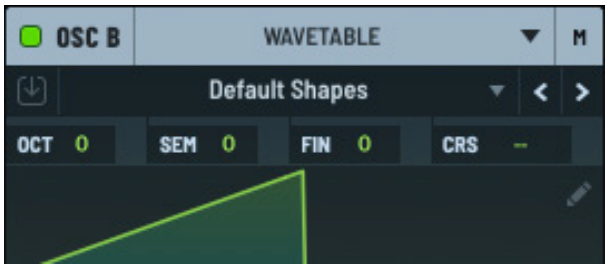
Serum generates sound using a set of oscillators, powered by a range of techniques that use both wavetables and samples. Serum then uses filters to sculpt the sound generated by the oscillators.

Each oscillator type features specific settings and parameters that you'll read about later in this guide. This section describes a series of operations that are common across most oscillator and filter types.

Enabling an Oscillator or Filter

You can enable an oscillator by clicking the label (containing the oscillator name and enable button). When enabled, the button turns green.

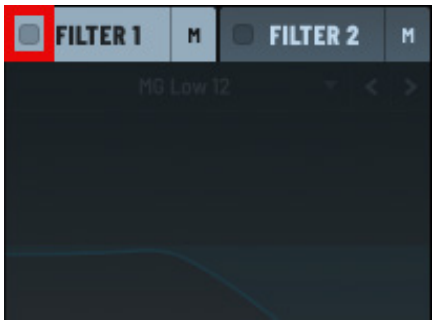
Enabling OSC B



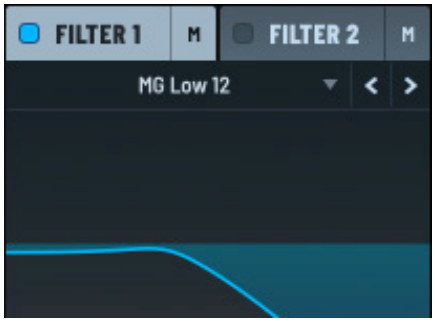
OSC B Enabled

You can also use the power button to mute (disable) an oscillator to either solo the other oscillators or free up CPU, as needed. Notice that when an oscillator is off, the entire panel is dimmed.

Enabling a filter is very similar, except that you need to click the power button directly.



Enabling Filter 1



Filter 1 Enabled

Choosing Oscillator or Filter Options

You can choose an oscillator or filter option using the associated drop-down menu.

Oscillator options accessible using the drop-down menu include wavetables, samples, multisamples, and more.

In the case of filters, you can choose the type of filter using the drop-down menu.

Click the **< >** arrows to quickly navigate to the previous and next menu option.

Alternatively, hover over the menu and use the mouse wheel to quickly rotate through menu options.

Choosing a Wavetable

Using Pitch Controls

Oscillator Pitch Controls

You can alter the pitch of the waveform using the **OCT** (octave), **SEM** (semitone), **FIN** (fine tuning in cents), and **CRS** (coarse) controls.

The **CRS** setting controls the pitch transpose that tunes or detunes continuous (no snap) semitones. **CRS** is most useful as a modulation destination or automation parameter for wide sweeps.

Serum uses separate controls for the four settings to facilitate automation and modulation (rather than a combined value such as 36.04).

It's often handy to assign an LFO to the octave setting, for instance, without having to count in 12 semitones.

There are times, however, when you might want to have an LFO control an oscillator pitch in a more coarse manner (a siren-type sound might require an LFO to modulate an oscillator pitch smoothly across an octave or more).

You can do this by modulating the CRS setting. (You'll read about modulating controls later in this guide.)

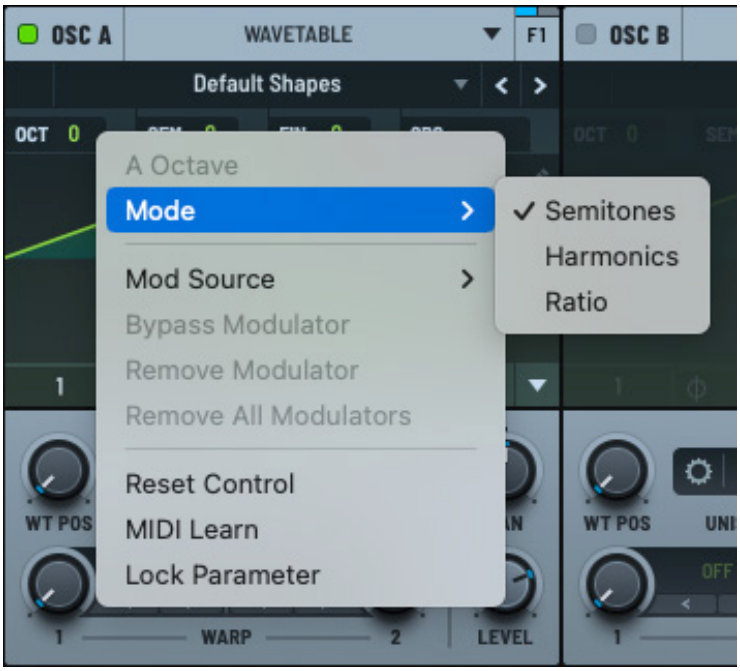
Use the **Global > Main Tuning** modulation destination to have all oscillators follow a coarse pitch change.

Setting the Octave or Semitone Mode

Serum further allows you to specify the mode for the **OCT** and **SEM** controls.

Right-click either control and choose the mode in the context menu.

You can select from the following options:



Selecting the Mode

- Semitones — Adjust the pitch in semitones, which are the intervals between two adjacent keys on a piano tuned to 12-tone equal temperament used in American/European musical tradition.

  This allows for fine-tuning, transposing, or creating intervals such as minor/major thirds, fifths, and more.

- Harmonics — Change the pitch by multiplying the base frequency using whole number harmonics.

  This generates pitches based on the harmonic series, which is useful for creating overtone-rich sounds, like organ tones or harmonic layers.

- Ratio — Set the pitch of the oscillator in relation to the base frequency using ratios, which is common in FM synthesis.

  In this case, the oscillator is tuned to a specific ratio relative to another oscillator to create complex timbres. When you select this mode, you can then set the specific source (SRC) and ratio.

- Step — Adjust the pitch up or down in periods and steps, as defined by the active MTS-ESP tuning.

  This option only appears when you are using microtuning and have an MTS-ESP tuning source available.

Routing an Oscillator or Filter

You can route the signal from any oscillator (**OSC A**, **OSC B**, **OSC C**, **SUB**, and **NOISE**) or any filters (**FILTER 1** and **FILTER 2**) to a range of targets.

The following table describes the routing targets:

Target	Description
Filter	Route the signal to either **FILTER 1**, **FILTER 2**, or by varying degrees to both.
Main	Route the signal to the main output, passing through the effects section.
Direct	Route the signal to bypass the filter and effects section and play "clean" along with the main output.
None	Route the signal to no output path. You can use this setting when you want to use an oscillator as a modulation source without having the sound included as part of the output signal.

You can access the routing options by clicking the button near the top right of an oscillator or filter.

Accessing the Signal Routing Settings

When you click, a set of controls appear that are relevant to the routing path.

For example, in the case of routing to the Serum filters, you have the option of routing the signal completely to **FILTER 1** (knob turned far left), completely to **FILTER 2** (knob turned far right), or to some combination of both (knob set somewhere in between).

By default, **OSC A** is automatically routed to **FILTER 1**, though **FILTER 1** is not enabled in new presets.

The other oscillators and filters are automatically routed to the **Main** output. You can change the routing at any time.

You can also send the signal to the Serum busses (**BUS 1** and **BUS 2**) by adjusting the corresponding knobs.

Filter Routing and Busses

To change the signal routing, click the ▼ button and choose another option, as appropriate.

Routing Menu

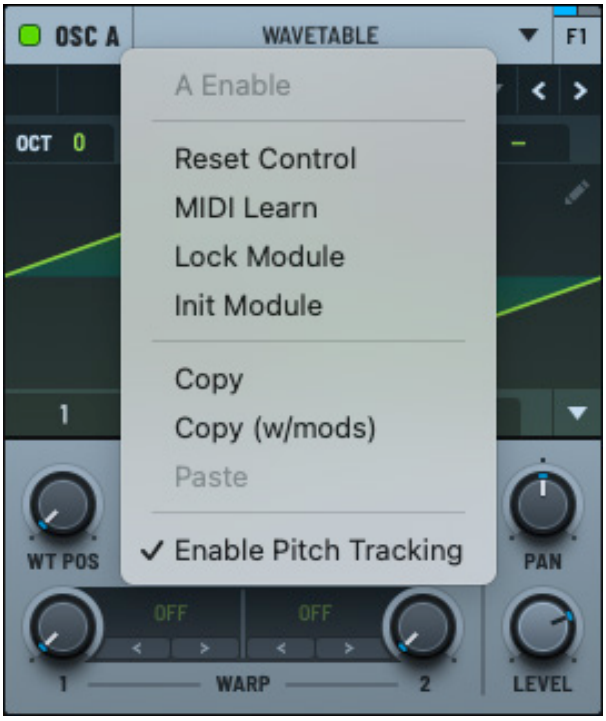
Accessing the Oscillator or Filter Menu

You can use the oscillator or filter menu to perform the most common operations related to the module.

These operations include locking the module, initializing the module, copying and pasting a module (with or without modulations), and enabling pitch tracking (in the case of oscillators).

Right-click the module label to access the context menu.

The next sections describe the specific operations you can perform.



Oscillator Menu

Locking a Module

You can lock a module by right-clicking the module label and choosing **Lock Module** in the context menu (see the **Oscillator Menu** above). This causes the parameters (settings) of the oscillator module to remain unchanged (locked) as you change presets.

For example, if you lock **OSC A**, the module remains the same even when you load a preset (that would normally change the oscillator module).

After locking a module, you can unlock individual controls (such as the **WT POS** knob, for example), as needed.

Note: This does not apply when you initialize the entire preset (by choosing **Init Preset** using the main menu). This initializes all settings and removes any module locks that you might have set.

Initializing a Module

After making changes to an oscillator or filter, you can return the module to an initialized state without affecting any other Serum setting.

Right-click the module label and choose **Init Module** in the context menu. The oscillator returns to initial settings (including removing all modulators assigned to the module).

Copying a Module

You can copy an oscillator or filter (with or without modulations) and paste it to another (similar) module. This works when copying and pasting modules in the same preset, as well as when you want to copy and paste between presets.

For example, consider the case where you configure an oscillator module with specific wavetable, unison, detune, and warp settings. In addition, you have **LFO 1** modulating the wavetable position (**WT POS** knob).

Right-clicking the oscillator label and choosing **Copy** in the menu copies the module to Serum's internal clipboard. Right-clicking another oscillator module and choosing **Paste** in the menu pastes the configuration (all settings except the **WT POS** modulation) to the corresponding module.

To copy and paste a module configuration including modulations, right-click the oscillator label and choose **Copy (w/mods)** in the menu. Pasting to a new module now includes the **WT POS** modulation from the earlier example.

As mentioned, this copy and paste operation works even after changing or initializing a preset. In addition, the module configuration stays on Serum's internal clipboard after pasting, allowing you to quickly paste the same configuration to multiple modules as needed. It is not possible, however, to copy a module from one instance of Serum to another.

Hold the Option/Alt key and drag the module label (next to the power button) to another (similar) module to copy the module without modulations.

Hold the Shift-Option or Shift-Alt keys and drag the module label to another module to copy with modulations.

Dragging from one module label to another without any keyboard modifiers swaps the two modules (including modulation assignments).

Enabling Pitch Tracking

Pitch tracking instructs the oscillator to adjust its pitch in response to the MIDI note or key being played. This ensures that the oscillator's frequency corresponds to the desired musical pitch. Pitch tracking is typically used with melodic and harmonic sounds, where the pitch needs to follow the keyboard.

Pitch tracking is enabled by default.

You can choose to disable pitch tracking by right-clicking the oscillator label and toggling **Enable Pitch Tracking** off in the context menu.

You might choose to disable pitch tracking with the following types of sounds:

- Drones or static sounds

  This produces a constant, unchanging pitch regardless of the notes played. This is common with ambient soundscapes that don't need pitch variation. This is also useful for layering a static tonal element beneath a dynamic lead or pad.

- Percussive sounds

  These sounds often don't rely on pitch tracking since their character is defined more by their transient and timbral qualities than specific pitch. You can use this with a variety of drum sounds, such as kick drums, snares, or hi-hats, as well as metallic or inharmonic percussive textures.

- Noise-based effects

  Noise signals (white, pink, or custom noise) aren't inherently pitched, so pitch tracking is irrelevant. You can also use this with wind, rain, or static effects, in addition to risers, sweeps, and impacts.

- Experimental sounds

  Disabling pitch tracking can lead to unexpected and unique sonic results. This can be helpful for creating unconventional or dissonant sounds. This can more easily allow you to explore textures where the focus is on timbre and modulation rather than pitch accuracy.

You can also disable pitch tracking to add layers without harmonic conflicts. A non-pitch-tracked oscillator can add texture or depth without interfering with the harmonic structure (such as a static sub-bass tone underneath a harmonic element).

When pitch tracking is disabled, the **Multisample**, **Sample**, **Granular**, and **Spectral** oscillators play C3 (MIDI note 60), whereas the **Wavetable** oscillator plays C-2 (MIDI note 0), allowing it to be used as an LFO.

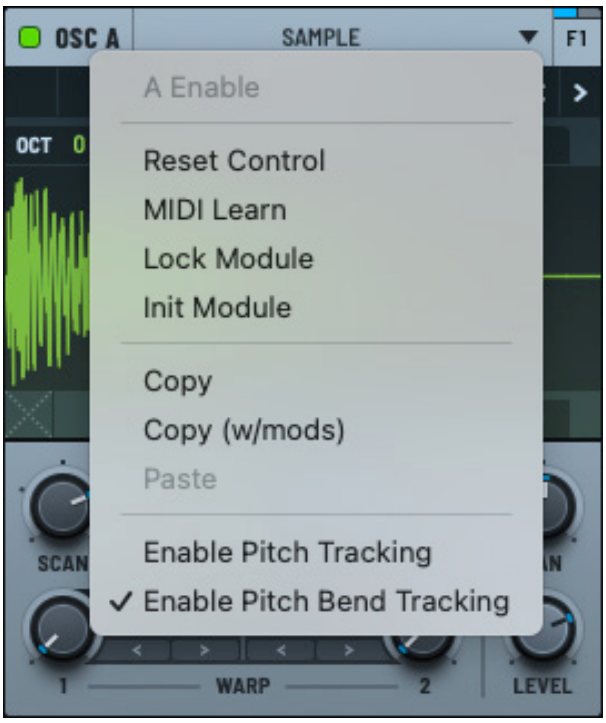
Pitch Bend Tracking

Pitch bend tracking determines whether MIDI pitch bend affects the frequency of an oscillator when pitch tracking is disabled.

The option appears on the Oscillator context menu only when **Enable Pitch Tracking** is disabled, and is only available for **Sample**, **Granular**, and **Spectral** oscillator types. Pitch bend tracking is enabled by default.

You might choose to disable pitch bend tracking for static, drone, or noise-based sounds so that they remain at a constant pitch, even when moving the pitch bend wheel.

Note that **Wavetable** and **Multisample** oscillator types always track pitch bend when pitch tracking is disabled.



Pitch Bend Tracking Enabled

Resizing the UI

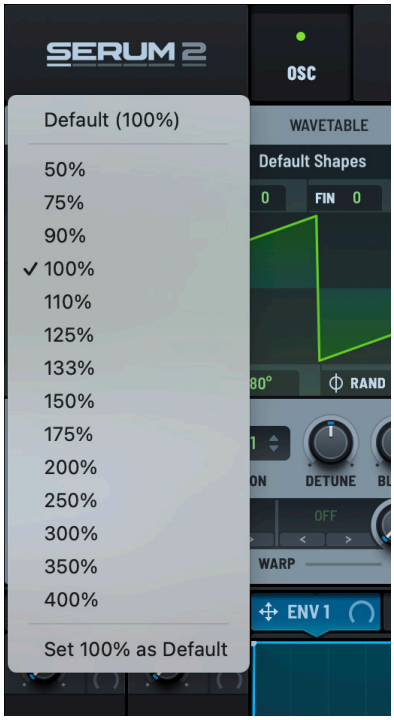
You can resize the Serum user interface to make it fit appropriately within your work environment.

The easiest way to resize the interface is to click and drag the lower right corner of the UI (similar to how you would resize most other windows using your computer).

For more precise control, click the Serum 2 logo (near the top left) and choose a resize option in the menu that appears. The window resizes to match your selection.

After resizing, you can make this size the default setting by clicking the Serum 2 logo again and choosing **Set x% as Default,** where *x* is the current setting.

To return to the default setting of 100%, click the Serum 2 logo and choose **Default (100%)** using the menu.



Resize Menu

Using Wavetable Oscillators

As with the original Serum, wavetable oscillators remain at the heart of sound generation in Serum 2. Unlike many wavetable synthesizers, however, Serum tables are multi-cycle, offering a greater variety of sounds.

Serum oscillator playback has been carefully constructed to give you a high-frequency representation to the limits of the human ear, without the audible aliasing artifacts (Nyquist reflection) commonly found on most wavetable synthesizers.

While this requires more CPU during both load and run-time, Serum's advanced SSE (Streaming SIMD Extensions) optimizations effectively help to minimize the CPU expense. We think you'll agree that the benefits in sound purity make it worthwhile.

Oscillator Panel (with 2D Waveform)

What is a Wavetable?

A *wavetable* is a small amount of digital audio (sample data or waveform) that is played back in a looping fashion.

The frequency (pitch) of the resulting note is determined by the rate (the speed between the repeats) at which the waveform is played. The tone (timbre and harmonics) of the sound is determined by the content of the waveform.

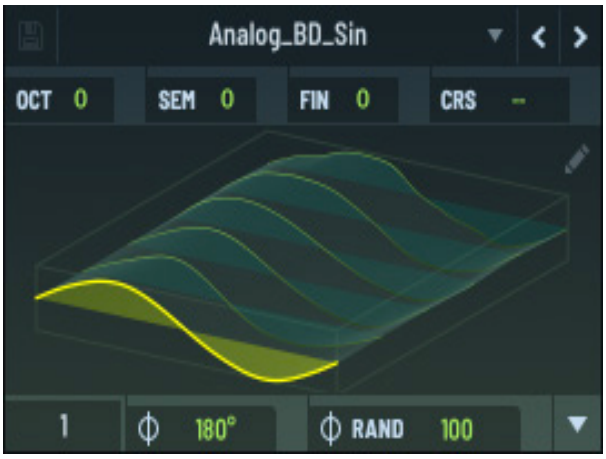
Anatomy of a Serum Wavetable

Wavetables in Serum consist of up to 256 subtables, or single-cycle waves (referred to hereafter as frames). You can think of this as (up to) 256 discrete waveforms joined together end-to-end in the parent file on disk.

In normal circumstances, you can hear one of these 256 frames at a time. However, if you assign automation to the **WT POS** knob, the sound starts to glide through the various frames, thereby becoming animated.

The wavetable in the illustration contains six frames, with yellow highlighting the current frame and green showing the five other frames.

If you increase the number of unison voices and turn up the unison **WT POS** control (in the unison settings), you'll hear multiple frames (subtables) playing back simultaneously.



Wavetable Frames

Strictly speaking, when Serum loads a wavetable, it uses 2048 samples for a frame (subtable) of the wavetable set. This means that the maximum file size is 2048 (samples) x 256 (frames) x 32 (bits), which is exactly 2 megabytes.

However, most wavetable files will not be this large. A good sounding wavetable can consist of just a few frames. The remaining frames can be interpolated (in the Wavetable Editor) to allow for smooth-sounding transitions.

These interpolated frames are generated through crossfading (mix blend) or spectral morphing (frequency + phase blend). These frames are computed at load time; Serum embeds the interpolation type rather than the interpolated waveforms (reducing disk space).

Using Wavetables

Serum features three multi-purpose oscillator modules, as well as dedicated **NOISE** and **SUB** oscillators (discussed separately later).

You can use any or all of the three oscillators in wavetable mode, taking advantage of advanced display and editing capabilities to help you get the greatest sonic possibilities.

This section describes how to use wavetables in Serum.

OSC A in 2D View

Exploring the Waveform Display

OSC A, **OSC B**, and **OSC C** all display a green waveform area with two viewing options: 2D and 3D.

Clicking on the waveform toggles between the two views.

In 2D mode, you see a single-cycle (frame) of the selected wavetable. You can use this view to see the warp feature in real-time (described later in this chapter).

The 3D view, in contrast, displays all frames (subtables) at-a-glance, with each frame represented by a green horizontal waveform, interpolated frames in gray, and the currently-selected frame in yellow.

Notice that the current frame number (wavetable position) appears in the lower left of the waveform area.

OSC A in 3D View (with wavetable position number)

Choosing a New Wavetable

You can load a new wavetable by clicking the drop-down menu and choosing an option.

Click the **< >** arrows to quickly navigate to the previous and next wavetable respectively.

Alternatively, hover over the menu and use the mouse wheel to quickly rotate through wavetable options.

Choosing a Wavetable

Editing a Waveform

You can take a closer look at a wavetable by hovering over the top-right corner of the waveform display and clicking the button.

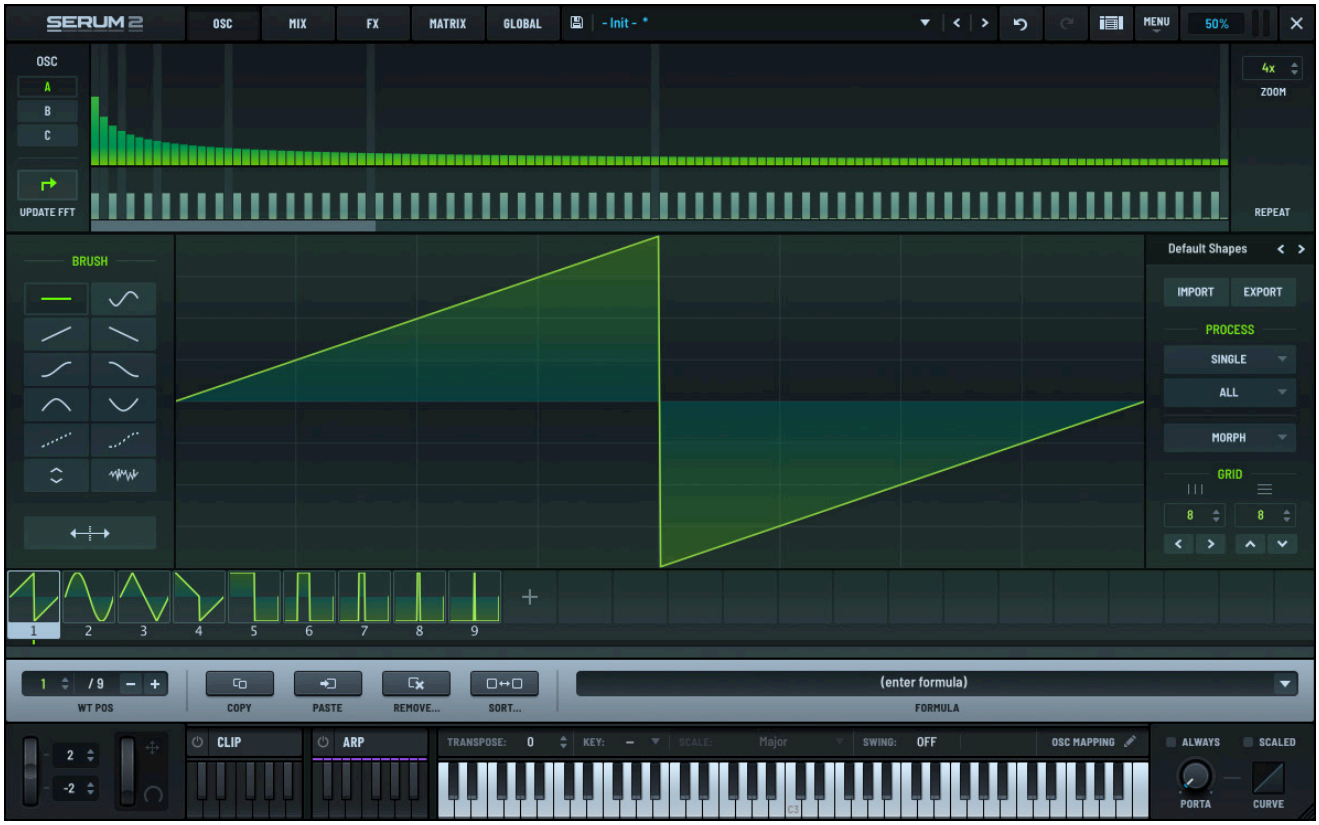
Wavetable Editor Button

Note that the button is initially dimmed until you hover directly over it.

Doing so opens the Wavetable Editor, presenting you with a rich set of tools to modify and manipulate the wavetable in multiple ways.

Click the button to close the Wavetable Editor.

You'll learn all about the Wavetable Editor and its many features in "Using the Wavetable Editor" on page 274.



Wavetable Editor

Phase

Use the **PHASE** control to specify where the oscillator begins playing back when a note is triggered.

It is the same concept as sample start on a sampler (except the “sample” is a very small waveform).

Since oscillators tend to be reasonably high frequency, you may not notice a difference when changing this knob.

In particular, if the adjacent **RAND** control is up considerably, you probably won’t hear a difference when adjusting the phase (because **RAND** alters the phase on each new note).

Oscillator Phase Control

Having said that, the effect can be very pronounced in the following scenarios:

- With a smooth waveform and fast-attack envelope

  If your wavetable is a sine wave, for instance, and the **ENV 1** attack is very fast (set to a small value), you will hear a click at the beginning of the sound if the **PHASE** control is set to a non-zero crossing (0% or 50% represent the zero crossings in a default sine wave).

- Using two or more oscillators with the **RAND** setting off

  The interaction between two or more oscillators can be very noticeable due to phase cancellation. Adjusting the start time of one of the oscillators results in a different tone.

  If the oscillator waveform display is in 2D (single table view), moving the **PHASE** control causes a yellow line to appear indicating where the note onset will occur.

Randomizing the Phase

Use the **RAND** (random) control to alter the **PHASE** knob value by a random amount for each new voice.

Oscillator Random Control

You might want to randomize the start phase of an oscillator for the following reasons:

- To provide a different start or click/thump to each note
- To provide a random tone to each note (when layering multiple oscillators), as the phase cancellation between the oscillators varies with each new note
- To reduce or remove the “laser zap” effect when unison (slightly detuned) notes are triggered

For example, do the following to hear the effects of the RAND control:

1. Set **PHASE** to 0 degrees.
2. Set **RAND** to 0 percent.
3. Raise the **UNISON** number to a high value, such as 8.
4. Lower the **DETUNE** slightly, for instance, to 10 o'clock (a value of 0.10).

At this point, you should hear a "laser zap" phasing sound resulting from the unison oscillators all starting together in phase, and slowly drifting apart from their detune.

Every time you trigger the note, you'll hear the same laser zap sound since the voice phases are restarting. While this can sometimes be cool, generally this is undesirable as the sweeping sound can be distracting.

As you raise the **RAND** control, notice how the effect becomes less pronounced as a random phase offset is introduced into each voice separately. By the time you reach 100% using the **RAND** control, you will no longer hear the "zap" sound.

Phase Legato

In the case of the "laser zap" phasing sound produced in the previous section, the default behavior is that when **Legato** is disabled (in the **VOICING** section) and a voice is stolen, the wavetable oscillator phase does not reset at note on. Therefore, you won't hear that "laser zap".

Also, in cases when **Legato** is enabled and Serum is playing polyphonically (including when not stealing a voice), the wavetable oscillator phase does reset at note on. This results in you hearing the zap even though legato is enabled and you want a smooth transition between notes.

The default behavior is how Serum 1 works, and is intended to avoid clicks at note on, which would be particularly evident with a simple sine wave.

However, with the above settings, we want the opposite behavior. You can toggle this by right-clicking the **PHASE** control and choose **Phase Legato** in the context menu.

Setting Phase Memory

You can set the phase memory for the wavetable using the phase memory drop-down menu. The phase memory specifies how phase and phase randomization should be determined for new notes.

Phase Memory Menu

The following table describes the options:

Option	Description
All Voices	New notes use the **PHASE** and **RAND** control settings for all voices. This is the default setting.
Contiguous	New notes continue with the phase of the previous note.
Per Voice	New notes start with the same editable phase each time.

If you set phase memory to **Per Voice** while one or more notes are sounding, Serum captures the current phase from the most-recently sounded note.

Wavetable Position

Use the **WT POS** knob to set the position within the wavetable. In other words, the knob selects the frame (subtable) that is currently audible.

Setting the **WT POS** knob to the minimum setting of 1 selects the first frame (highlighted in yellow).

Remember, yellow always indicates the currently-selected (audible) frame.

Right-click the **WT POS** knob and choose **Smooth Interpretation** in the menu to allow smoother waveform transitions without being destructive.

This helps keep the wavetable intact while morphing smoothly between positions.

Oscillator WT POS Control

Unison

Use the **UNISON** control to set the number of unison voices, effectively "stacking" oscillators in a way that is similar to playing multiple notes of the same pitch, but slightly detuned.

Click the field and drag to set the appropriate value. You can also double-click the field and type a value.

Serum allows you to stack up to 16 voices in unison, but doing so can result in a more "cloudy" sound and less like distinct voices. The classic magic number for unison is 7.

Serum has a special capability that keeps the level increase in check as you increase the number of unison voices, effectively maintaining the volume at the same level.

Note: Unison causes Serum to generate multiple voices, raising CPU usage. The color of the **UNISON** field changes as you increase the number of unison voices as a reminder of the CPU consumption.

Wavetable Unison Control

Wavetable Unison Settings

Click the button to display the unison settings.

The wavetable unison settings appear.

You can specify the following settings:

Setting	Description
MODE	The detune mode, from among the following: • Linear — The detuning between each additional voice increases in a consistent, linear fashion. This means that the pitch of each voice is spaced evenly in terms of frequency, creating a smooth spread that retains an even distribution. This mode can sound very controlled and smooth, giving a thick and coherent texture. • Super — Multiple voices are slightly detuned from each other but with a special emphasis on creating a dense and powerful sound, often with a slight stereo spread. Use this mode to create a lush, wide sound, especially with supersaw sounds, where the detuned voices give a rich, full sonic character. • Exp — The detuning between voices increases exponentially as you move away from the central pitch. This means that the spacing between voices gets wider more quickly as you move outward, creating a more dramatic spread compared to linear. This mode can create unique, rich textures, especially for sounds that need to be more aggressive or have a strong presence in a mix.

Setting	Description
MODE (cont.)	• Inv — The detuning behavior is inverted, with lower voices detuned more sharply compared to higher ones. Use this mode to create interesting phasing effects or to produce sounds with a less conventional detuning profile. • Random — Introduces a random element to the detuning of each voice. Instead of being evenly spaced, the voices are detuned unpredictably. This can create a more organic or chaotic sound, ideal for achieving textures that are less polished and more natural or experimental.
STACK	The unison stacking, which instructs Serum to not just duplicate the unison voices at the same pitch but instead transpose or harmonically adjust the voices. You can choose from among the following options: • Off — Do not stack the unison voices • 12 (1-3x) — Distribute voices between the original pitch and octave transpositions, with a range 1-3 octaves higher • 12+7 (1-3x) — Distribute voices between the original pitch, fifth, and octave transpositions, with a range 1-3 octaves higher • Center-12 — Transpose the center voices down one octave • Center-24 — Transpose the center voices down two octaves
WIDTH	The extent to which the unison voices are spread out across the stereo field, determining how wide or narrow the resulting sound feels in a stereo mix. When you increase the width parameter, the unison voices are spread farther apart in the stereo spectrum, placing some voices more to the left and others more to the right. This creates a sense of space and a wider, more immersive sound. Conversely, decreasing the width narrows the stereo spread, making the sound more centered and focused. This can be useful for sounds that need to be more direct or fit better in a dense mix without dominating the stereo field.

Setting	Description
RANGE	The extent or range of detuning applied to the unison voices, determining how far apart in pitch the individual voices are spread around the central frequency of the original sound. When the range is set to a low value, the detuning between the unison voices is minimal. The voices remain relatively close in pitch to the original frequency, creating a subtle and smooth chorusing effect. This can add warmth and a slight thickness to the sound without drastically changing its character. When the range is increased, the detuning becomes more pronounced, and the unison voices are spread further apart in pitch. This creates a more dramatic and sometimes more chaotic or dissonant effect, which can be useful for creating lush, wide textures or aggressive, detuned leads.
WT POS	The wavetable frame that the unison voice plays at a particular moment. By modulating this position, you can morph through different waveforms, creating dynamic, evolving sounds.
WARP 1	Spread out the warp amount applied to each voice around the current **WARP 1** knob position.
WARP 2	Spread out the warp amount applied to each voice around the current **WARP 2** knob position.

You can further adjust the unison setting using the **DETUNE** and **BLEND** knobs.

Unison Detune

Use the **DETUNE** knob to specify the tuning offset +/- for the additional voices. This is only applicable when unison is enabled (set to a value above 1).

Unison Blend

Use the **BLEND** knob to specify the level offset of the unison voices versus the “central” unison voice or voices (1 if unison is set to an odd number of voices and 2 if set to an even number).

You can think of **BLEND** as a wet/dry mix between a unison (wet) and non-unison (dry) sound. The default value of 75% is an even blend between all the voices. Note that this is only applicable when the number of unison voices is greater than two.

Warp

Setting the warp allows you to manipulate the playback (and sound) of the wavetable oscillator.

By default, warp is set to **OFF** (as displayed next to the corresponding knob). Clicking the current setting displays a menu from which you can choose from among the available warp modes.

You can also use the **< >** arrows to conveniently switch between different warp modes without having to open the menu. After selecting a mode, you can use the corresponding knob to set the depth.

Note that if the waveform is in 2D view (with a single frame visible), you can see how the warp mode affects the waveform for **Sync**, **Alt Warp**, and **Distortion** modes.

Wavetable Warp Control

Exploring the Warp Modes

Serum offers an extensive set of warp modes.

As with many parts of Serum, the best way to learn about them is through experimentation.

Wavetable Warp Menu

The following table describes the available warp modes:

Category	Warp Mode	Description
Off		The warp function is turned off.
Sync		Synchronize wavetable playback to an internal oscillator that restarts in sync with the original oscillator phase. The warp control sets the pitch of the internal oscillator; when increased, the harmonic content shifts upwards while retaining the pitch of the original oscillator. This can create a harmonious effect when the original and internal oscillator relate by a whole-number ratio (1:2, 1:5, and so on). Pitches outside of these values can cause a saw-wave to form at the end of each cycle. When you select **Sync** mode, a **WARP Var** fader control appears directly below the menu. Use this to adjust the smoothness of the sync from traditional “hard sync” to a very soft “soft sync.”
Alt Warp	Bend +	Pinch (bend) the waveform inwards (towards the middle of the wave cycle).
	Bend -	Pull (bend) the waveform outward (towards the edges of the wave cycle).
	Bend +/-	Allow for both of the above, depending on **WARP** knob value. A setting of 12 o’clock on the **WARP** knob (50%) represents no change to the sound.
	PWM	Push the entire waveform to the left. This is useful with square wave type sounds, especially for the classic PWM sound (but useful with other waveforms as well).
	Asym +	Similar to Bend, but in this case bend the entire waveform to the right instead of both halves of the duty cycle separately.
	Asym -	Bend the entire waveform to the left.
	Asym +/-	Bend the entire waveform to the left or right.
	Flip	Create an instantaneous polarity flip (often called phase inversion) on the waveform. The **WARP** knob determines where in the duty cycle the flip occurs.

Category	Warp Mode	Description
Alt Warp (cont.)	Mirror	Create a mirror-image of the waveform for the second half of the duty cycle. This has an “octaved” type of quality to the sound due to the doubling of the waveform into both halves of the wave cycle. For this setting, the **WARP** knob behaves similarly to the Asym +/- mode, except on both halves independently. Due to the mirroring of the waveform, this mode always has an audible effect.
	Remap 1	Custom remapping of the wave cycle. When you select this option, a pencil button appears that you can use to open a graph showing the way your waveform will remap. A diagonal line from bottom-left to top-right indicates no change to the waveform (y=x). The **WARP** knob determines the strength of the remap, from 0 (y=x) to 100% (what you see on the graph).
	Remap 2	Represents mirrored remapping. This is the same as Remap 1 but applies the graph to each half the waveform independently. This allows for symmetric remapping without the need to draw symmetric shapes on the graph.
	Remap 3	Represents sinusoidal remapping. This is another remapping option that saves you from having to draw fancy curves.
	Remap 4	Represents a 4x remapping. This is similar to Remap 2 (mirrored) but in this case the graph applies four times. This creates a more busy sound, and can be helpful when you want something nasty.
	Quantize	Similar to sample-and-hold, that is, a sample rate reduction. Compared to an SR Redux effect, this applies to the waveform itself. This causes the aliasing sound to follow the pitch perfectly (instead of having that “same ringing pitch on all notes” quality that a redux effect creates).
	Odd/Even	Proportionally vertically scale the waveform. Use this to emit only the odd or even harmonics from the signal or a mixture of the two. At 50%, you have the original signal. At 0%, you hear only the odd harmonics. At 100%, you hear only the even harmonics (which creates an octaving effect since the first harmonic is missing).

Category	Warp Mode	Description
Filter	LPF	Apply a low pass filter.
	HPF	Apply a high pass filter.
Distortion	Tube	Emulate the characteristics of analog tube amplification. The signal is subjected to nonlinearities that mimic the behavior of vacuum tubes, producing a warm, harmonically rich, and often smooth-sounding distortion.
	Soft Clip	Apply gentle, nonlinear compression to the signal, creating a smoother, less aggressive distortion compared to hard clipping. Use this to add warmth and character to a sound without introducing harsh or unpleasant artifacts.
	Hard Clip	Create a type of distortion that aggressively limits the signal by abruptly cutting off its peaks after it exceeds a certain threshold. This results in a sharp, harsh form of distortion that creates a more aggressive and intense sound compared to softer forms of clipping.
	Diode 1	Create a type of distortion that emulates the sound characteristics of analog diode clipping circuits, often found in classic guitar pedals and analog equipment. This mode adds a unique type of distortion that is both warm and aggressive, with specific tonal characteristics derived from the behavior of diodes in the signal path.
	Diode 2	Apply a sinusoidal transfer curve with increased hard clipping as drive is increased.
	Linear Fold	Create a type of wavefolding distortion that produces a distinctive and often aggressive sound by folding the waveform back on itself whenever it exceeds a certain amplitude threshold. This folding effect adds rich harmonic content and introduces a complex, metallic, or harsh character to the sound.
	Sine Fold	Create a type of wavefolding distortion effect that shapes the input waveform using a sine-based folding process. This effect introduces complex harmonic content and adds a smooth but dynamic character to the sound, often resulting in rich, evolving timbres that can range from warm and musical to intense and aggressive.

Category	Warp Mode	Description
Distortion (cont.)	Zero-Square	Modify the waveform in such a way that any part of the signal below a certain amplitude threshold is forced to zero, while parts of the waveform above the threshold are squared or otherwise drastically altered. This creates a sharp, abrupt change in the shape of the waveform, leading to a sound that is both harsh and harmonically rich.
	Asym	Create an asymmetric effect that applies different distortion characteristics to the positive and negative halves of an audio waveform. This type of waveshaping introduces a unique harmonic profile by treating one side of the waveform differently from the other, resulting in a sound that can range from subtle warmth to highly complex and rich overtones.
	Rectify	Create a type of waveshaping effect that modifies the signal by altering or “rectifying” the waveform, typically by flipping or removing one half of the waveform. This creates a distinct, harmonically rich, and sometimes harsh or metallic sound, often associated with aggressive or synthetic timbres.
	Sine Shaper	Create a type of waveshaping distortion that shapes the audio signal using a sine function. This type of distortion applies a nonlinear transformation to the input signal, resulting in a smoother, more rounded distortion that introduces harmonics in a musical and often warm manner.
	Stomp Box	Create a distortion effect that emulates the sound of classic distortion or overdrive pedals used by guitarists (stomp boxes). This effect reproduces the gritty, crunchy, and saturated tones characteristic of analog guitar pedals, bringing warmth, edge, and intensity to a wide range of sounds.
	Tape Sat.	Create a distortion effect that emulates the warm, rich sound characteristics of analog tape recording. Tape saturation is a form of soft-clipping distortion that occurs naturally when audio signals are recorded to magnetic tape, especially at higher levels. This effect is highly valued for its ability to add warmth, harmonic richness, and a sense of vintage character to an audio signal.

Category	Warp Mode	Description
Distortion (cont.)	Soft Sat.	Create a distortion effect that introduces subtle, smooth, and musical saturation to the signal. It is designed to gently enhance the harmonic content of the sound without introducing harsh or aggressive distortion. Soft saturation is often used to add warmth, fullness, and a natural, analog-like character to audio, making it a popular choice for enhancing digital recordings.
FM	FM (from other oscillator) For example, FM (B)	Perform frequency modulation using the other oscillator or filter. The other oscillator or filter must be enabled for this to work, however, you can turn down the volume of the other oscillator if you want to use the other oscillator simply as a modulation source.
	FM (from other oscillator) For example, FM (C)	
	FM (Noise)	
	FM (Sub)	
	FM (Filter 1)	
	FM (Filter 2)	
	Thru-Zero	Cause the carrier oscillator (the one producing the sound) to continue to oscillate correctly even when its frequency is modulated into negative values by the modulator oscillator. Normally, if the modulation drives the carrier oscillator's frequency below zero, it either clamps at zero (stops oscillating) or reflects back to a positive value (causing a discontinuity). With **Thru-Zero**, when the frequency modulation drives the carrier frequency below zero, the carrier oscillator doesn't stop or reflect; instead, it inverts its phase and continues oscillating. This allows for smooth and continuous modulation, resulting in a more natural and harmonically rich sound. The inversion of phase (caused by negative frequencies) adds new harmonic characteristics, making **Thru-Zero** especially useful for lush, metallic, or bell-like tones.

Category	Warp Mode	Description
FM (cont.)	Exp	Use an exponential scaling curve. This means small changes in the modulator's amplitude can cause dramatic changes in the carrier frequency, especially as the modulation depth increases. Compared to linear FM, exponential FM produces a broader and more pronounced harmonic spectrum. It is often described as brighter or harsher due to the rapid frequency sweeps caused by the exponential relationship.
	Linear	Use a linear scaling curve. This means that the modulation source (modulator oscillator) affects the frequency of the carrier oscillator in a direct, proportional manner. This ensures that the carrier oscillator maintains its overall pitch, even when modulated heavily. This makes it particularly useful in musical contexts, as it allows for predictable harmonic and inharmonic spectra without drastically detuning the base pitch. Linear is often described as smooth, clean, and more "musical" compared to other FM types such as exponential FM. This makes it great for bell-like tones, pads, and other complex but stable timbres. Note that linear FM is *thru-zero* but with a clamp at zero. This allows you to do the traditional "can't do thru-zero" FM.
PD	PD (from other oscillator) For example, PD (B)	Perform phase distortion using the other oscillator or filter. The other oscillator or filter must be enabled for this to work, however, you can turn down the volume of the other oscillator if you want to use the other oscillator simply as a modulation source. This is similar to FM except that the phase is modulated instead of the frequency.
	PD (from other oscillator) For example, PD (C)	
	PD (Noise)	
	PD (Sub)	
	PD (Filter 1)	
	PD (Filter 2)	
	PD (Self)	

Category	Warp Mode	Description
AM	AM (from other oscillator) For example, AM (B)	Perform amplitude modulation using the other oscillator or filter. The other oscillator or filter must be enabled for this to work, however, you can turn down the volume of the other oscillator if you want to use the other oscillator simply as a modulation source. This is similar to FM except that the amplitude is modulated instead of the frequency.
	AM (from other oscillator) For example, AM (C)	
	AM (Noise)	
	AM (Sub)	
	AM (Filter 1)	
	AM (Filter 2)	
RM	RM (from other oscillator) For example, RM (B)	Perform ring modulation using the other oscillator or filter. The other oscillator or filter must be enabled for this to work, however, you can turn down the volume of the other oscillator if you want to use the other oscillator simply as a modulation source.
	RM (from other oscillator) For example, RM (C)	
	RM (Noise)	
	RM (Sub)	
	RM (Filter 1)	
	RM (Filter 2)	
Swap Warps		Swap the **WARP 1** mode (on the left) with **WARP 2** mode (on the right).

Pan

Use the **PAN** knob to control the placement of the waveform in the stereo field (left to right).

Wavetable Pan Control

Level

Use the **LEVEL** knob to control the output volume of the oscillator.

Wavetable Level Control

Using Multisample Instruments

Serum features the ability to set any of the oscillators to function as a multisample instrument that utilizes an array of samples—recordings of an actual instrument—played across a range of pitches, intensities, and articulations.

With a multisample instrument, individual notes are recorded at different velocities (soft to loud) and sometimes with various playing techniques, such as plucking, bowing, or using different mallets.

When using a multisample instrument, Serum accurately selects the appropriate sample based on the MIDI input—evaluating both the pitch and velocity of the note—ensuring that the output sound mimics the real instrument's response as closely as possible. For example, playing softly might trigger a sample of a gently played note, while pressing hard would trigger a sample of the same note played with force.

Layering multiple distinct samples achieves the effect of rich, lifelike audio textures that respond accurately to different playing styles. This can capture the nuanced tonal variations and expressive characteristics of the original instrument, allowing for a highly realistic and responsive musical experience.

Selecting a Multisample Instrument

Using **OSC A**, **OSC B**, or **OSC C**, click the header and choose **Multisample** in the menu that appears.

Multisample in the Oscillator Menu

The oscillator switches to multisample mode.

Initially, the waveform display is empty since no instrument has been loaded.

Multisample Instrument

Click the drop-down menu and choose a multisample instrument.

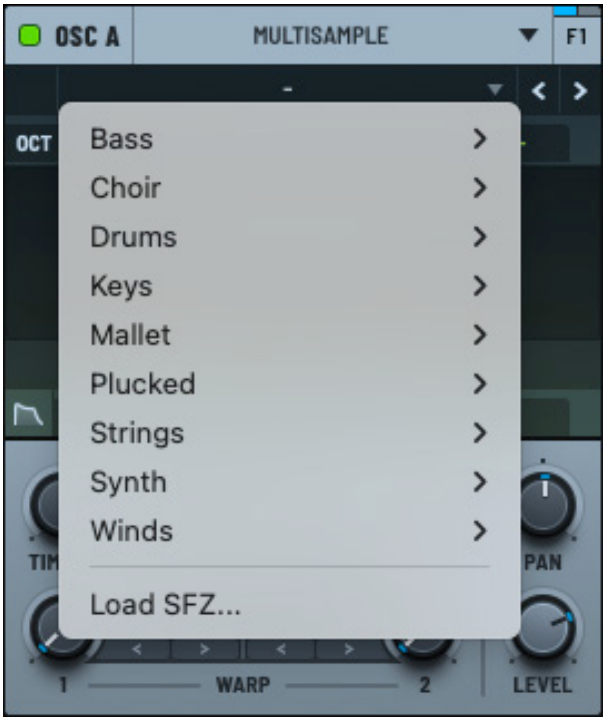
You can also load a multisample instrument using a **`.sfz`** file. Using the menu, choose **Load SFZ** and select the file using the dialog that appears.

SFZ files are text-based files that describe how audio samples (such as WAV files) should be played by a sampler.

The files map audio samples to specific notes, velocities, or other MIDI parameters. SFZ files can also include instructions for dynamic layers, key switches, round robins, and other articulations.

Furthermore, Serum supports SFZ envelope parameters to help shape the sound.

Serum does not natively support SoundFont 2 (SF2) files. There are, however, multiple open source applications that can convert files from SF2 to SFZ file formats.



Multisample Instrument Menu

After loading an instrument, the display shows the multisamples that comprise the instrument.

Playing a note highlights the specific sample, representing the corresponding pitch and velocity.

Multisample Instrument Loaded

Clicking in the display displays the waveform of the last note played.

Multisample Instrument (Note Played)

Setting the Multisample Envelope

You can modify the envelope by clicking the  button. The envelope pane appears, initially disabled.

SFZ files contain envelope information for playback. This can be per note, per group, or global.

Click the **OVERRIDE** button to activate the envelope display and override the envelope configuration from the SFZ file.

Use the knobs to adjust the envelope shape.

Multisample Envelope Pane

The following table describes the available controls:

Knob	Description
DELAY	The time (in milliseconds) before the envelope begins after a note is triggered, allowing you to introduce a pause before the attack phase starts. This is useful for creating rhythmic effects or gradual sound layering, giving more control over how and when the envelope affects the sound. Right-click the **DELAY** knob and choose **BPM Sync** in the context menu to synchronize the delay with the musical tempo. When selected, you can set the delay in beats and bars.
A (Attack)	The time it takes for the sound to reach its maximum level after a note is triggered. This shapes the initial onset of the sound, allowing for smooth fades or sharp, immediate beginnings depending on the setting.
H (Hold)	The amount of time the sound remains at its peak level after the attack phase is completed. This allows the sound to sustain momentarily before moving into the decay phase, adding emphasis and length to the peak of the sound.
D (Decay)	The time it takes for the sound to transition from its maximum level after the attack phase to the sustain level. This shapes the gradual reduction in volume, allowing you to create more natural fades or sharp drops in sound intensity depending on the setting.

Knob	Description
S (Sustain)	The steady level the sound maintains after the decay phase, as long as the note is held. Unlike other stages, the sustain level lasts indefinitely until the note is released, allowing for either continuous sound or a softer presence depending on the setting.
R (Release)	The time it takes for the sound to fade out after the note is released. This shapes the tail of the sound, allowing for smooth, gradual decays or quick cutoffs depending on the chosen release time.

You can also adjust the envelope by clicking and dragging each point directly on the graph. The cursor changes to indicate whether the point can be moved horizontally or freely in all directions, providing visual feedback for precise control.

Velocity Track

Use the **VEL TRACK** control to adjust the sensitivity of the note velocity. After enabling velocity tracking, click and drag in the field to modify the setting.

Random

Use the **RAND** control to randomize the initial phase of the sample. This is similar to the random setting for other oscillator modes. Click and drag in the field to modify the value.

Setting Multisample Parameters

You can set the timbre, unison (including detune and blend), and adjust the waveform warp for the multisample instrument.

You can also adjust the pan and level of the signal.

Multisample Parameters

Timbre

Use the **TIMBRE** knob to adjust the multisample timbre.

For multisamples with zone mapping, this control inversely adjusts mapped samples to pitch. This alters how the samples respond to changes in pitch across different key zones.

Normally, as you play higher notes, the pitch of the sample increases, and as you play lower notes, it decreases. When you inversely adjust the mapped samples to pitch, the opposite happens: higher notes trigger samples that are mapped to lower pitches, and lower notes trigger samples mapped to higher pitches.

This creates an unusual or experimental timbral effect, as the relationship between pitch and sample playback is reversed, potentially adding unique tonal qualities to the sound.

Unison

Use the **UNISON** control to set the number of unison voices, effectively "stacking" oscillators in a way that is similar to playing multiple notes of the same pitch, but slightly detuned.

Click the field and drag to set the appropriate value. You can also double-click the field and type a value.

This is similar to the unison setting for other oscillator modes.

Note: Unison causes Serum to generate multiple voices, raising CPU usage. The color of the **UNISON** field changes as you increase the number of unison voices as a reminder of the CPU consumption.

Click the  button to display the unison settings.

Multisample Unison Settings

You can specify the following settings:

Setting	Description
MODE	Set the detune mode, from among the following: • Linear — The detuning between each additional voice increases in a consistent, linear fashion. This means that the pitch of each voice is spaced evenly in terms of frequency, creating a smooth spread that retains an even distribution. This mode can sound very controlled and smooth, giving a thick and coherent texture. • Super — Multiple voices are slightly detuned from each other but with a special emphasis on creating a dense and powerful sound, often with a slight stereo spread. Use this mode to create a lush, wide sound, especially with supersaw sounds, where the detuned voices give a rich, full sonic character. • Exp — The detuning between voices increases exponentially as you move away from the central pitch. This means that the spacing between voices gets wider more quickly as you move outward, creating a more dramatic spread compared to linear. This mode can create unique, rich textures, especially for sounds that need to be more aggressive or have a strong presence in a mix. • Inv — The detuning behavior is inverted, with lower voices detuned more sharply compared to higher ones. Use this mode to create interesting phasing effects or to produce sounds with a less conventional detuning profile. • Random — Introduces a random element to the detuning of each voice. Instead of being evenly spaced, the voices are detuned unpredictably. This can create a more organic or chaotic sound, ideal for achieving textures that are less polished and more natural or experimental.
STACK	Set the unison stacking.

Setting	Description
WIDTH	The extent to which the unison voices are spread out across the stereo field, determining how wide or narrow the resulting sound feels in a stereo mix. When you increase the width parameter, the unison voices are spread farther apart in the stereo spectrum, placing some voices more to the left and others more to the right. This creates a sense of space and a wider, more immersive sound. Conversely, decreasing the width narrows the stereo spread, making the sound more centered and focused. This can be useful for sounds that need to be more direct or fit better in a dense mix without dominating the stereo field.
RANGE	The extent or range of detuning applied to the unison voices, determining how far apart in pitch the individual voices are spread around the central frequency of the original sound. When the range is set to a low value, the detuning between the unison voices is minimal. The voices remain relatively close in pitch to the original frequency, creating a subtle and smooth chorusing effect. This can add warmth and a slight thickness to the sound without drastically changing its character. When the range is increased, the detuning becomes more pronounced, and the unison voices are spread further apart in pitch. This creates a more dramatic and sometimes more chaotic or dissonant effect, which can be useful for creating lush, wide textures or aggressive, detuned leads.
WARP 1	Spread out the warp amount applied to each voice around the current **WARP 1** knob position.
WARP 2	Spread out the warp amount applied to each voice around the current **WARP 2** knob position.

You can further adjust the unison setting using the **DETUNE** and **BLEND** knobs.

Unison Detune

Use the **DETUNE** knob to specify the tuning offset +/- for the additional voices. This is only applicable when unison is enabled (set to a value above 1).

Unison Blend

Use the **BLEND** knob to specify the level offset of the unison voices versus the "central" unison voice or voices (1 if an odd number, 2 if an even number of unison).

You can think of **BLEND** as a wet/dry mix between a unison (wet) and non-unison (dry) sound. The default value of 75% is an even blend between all the voices. Note that this is only applicable when the number of unison voices is greater than two.

Warp

Setting the warp allows you to manipulate the playback/sound of the wavetable oscillator.

By default, warp is set to **OFF** (as displayed next to the corresponding knob). Clicking the current setting displays a menu from which you can choose from among the available warp modes.

You can also use the **< >** arrows to conveniently switch between different warp modes without having to open the menu. After selecting a mode, you can use the knob to set the depth.

See "Exploring the Warp Modes" on page 49 for detailed information about the available warp modes.

Pan

Use the **PAN** knob to control the placement of the waveform in the stereo field (left to right).

Level

Use the **LEVEL** knob to control the output volume of the oscillator.

Switching the Last Note Played to a Sample

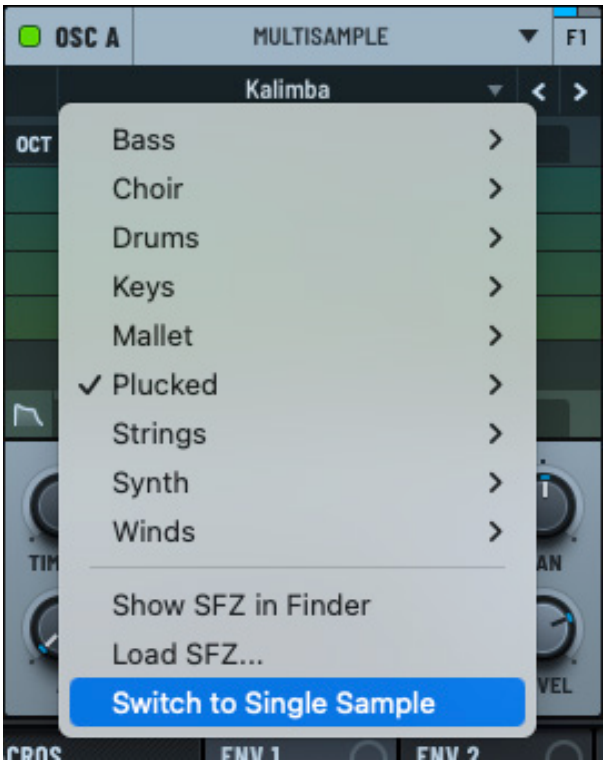
You can quickly switch the last multisample note that you played to a single sample. You might do this if you like a single sound but don't need the entire multisample.

Then later, if needed, you can also just as easily create a wavetable from that sample.

Begin by loading a multisample into an oscillator. After playing a note, click the **Multisample** menu and choose **Switch to Single Sample** in the menu.

The oscillator switches to **Sample** mode with the last-played note loaded as the sample.

To convert the sample to a wavetable, click the **Sample** menu and hover over **Switch to Wavetable** in the menu.



Multisample Instrument (Note Played)

The menu of import options appear. These are the same options that appear when you import audio as a wavetable in other areas of Serum.

Choose one of the menu options.

The oscillator switches to **Wavetable** mode with the converted wavetable loaded.

See "Importing Multi-Cycle Waveforms" on page 291 for a detailed description of each option.

Switch to Wavetable Menu

Using Sample Instruments

Serum features a versatile and intuitive sampler, designed for creatively manipulating and playing back audio samples. You can load a wide range of audio, from instrument sounds and vocals to field recordings, offering endless possibilities for sound design and musical experimentation.

In addition to the standard features you would expect in a sampler, Serum includes powerful slicing tools that are integrated with the **CLIP** mode within Serum, giving you even greater flexibility to rearrange and reimagine almost any type of audio source.

You can also quickly and easily convert samples to wavetables, offering even more creative possibilities.

When loading a sample, Serum assumes all samples to be tuned to C3 by default (following the standard that MIDI Note 69 is A3 at 440 Hz).

You can instruct Serum to assume a different tuning by adding the note name at the end of the file name. For example, naming a sample file **`Morning Bass F2.flac`** tells Serum to set F2 as the root of your sample, removing the need for you to manually adjust the pitch.

By informing Serum about the root note, Serum can now correctly map the sample. When you trigger F2, the sample will now play at its original pitch, while playing other notes (such as G2 or E2) will result in Serum pitch-shifting the sample accordingly.

Similarly, you can include flats and sharps in the file name. This means that naming a sample file **`Ashes Piano Eb3.flac`** or **`Ashes Piano D#3.flac`** produces the same result; the choice is yours whether to use flats or sharps.

Important: The note name (appearing at the end of the file name) must be preceded by a space or underscore character. No other separators are supported.

Serum additionally refers to pitch data embedded in the instrument chunk of a WAV file.

Selecting the Sampler

Using **OSC A**, **OSC B**, or **OSC C**, click the header and choose **Sample** in the context menu.

The oscillator switches to sample mode.

Sample in the Oscillator Menu

If you already chose a sample in the **Granular** or **Spectral** modes, that sample appears in the waveform display.

Otherwise, the display is empty.

Sample Instrument

Click the drop-down menu and choose a sample from the list of presets.

In addition to tonal and non-tonal factory presets, you can load Serum wavetables as samples.

You can also load a sample by choosing **Load Sample** in the menu and selecting the file using the dialog that appears.

Note: After loading a sample, you can use this menu to show the same location on your computer or reload the sample.

Sample Menu

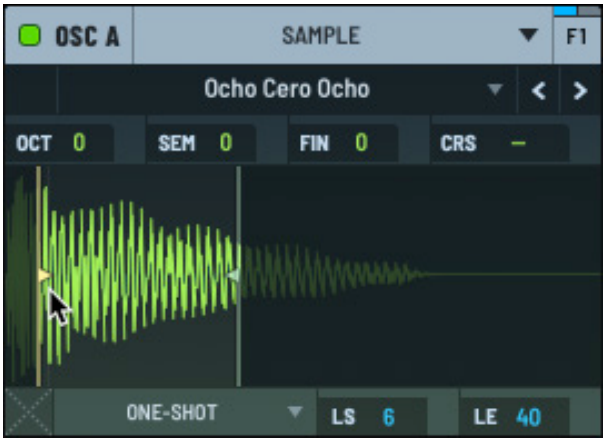
When a sample loads, the waveform appears in the display.

Sample Instrument Loaded

Setting the Sample Start and End

You can easily set the sample start and end points directly.

Hover over the sample and drag either (or both) of the markers that appear (on the left or right).



Setting Sample Start and End

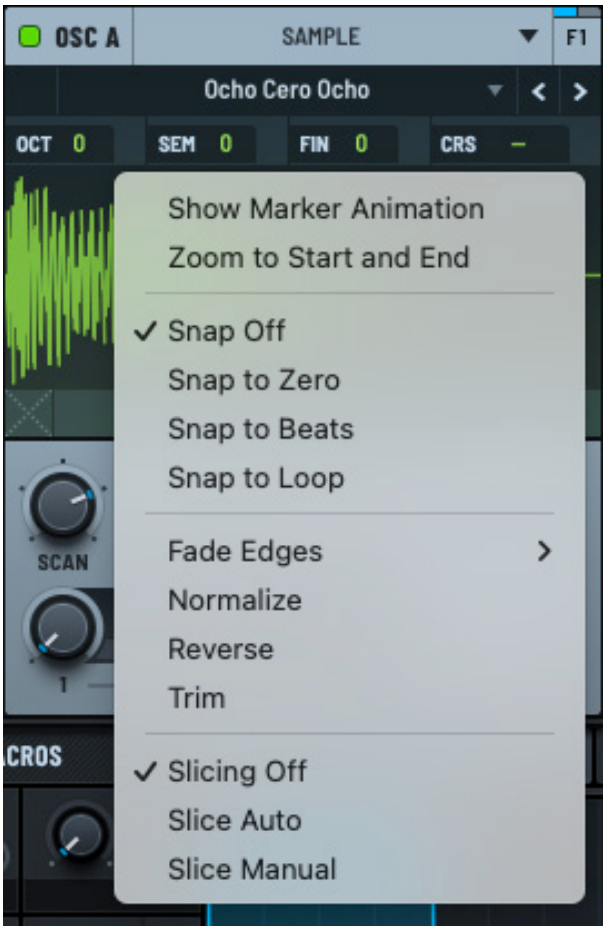
Performing Sample Operations

You can perform a range of operations on the loaded sample.

Right-click the sample and choose an operation using the context menu.

All operations available in **Sample** mode are also available when working in **Granular** and **Spectral** modes.

It's helpful to become familiar with these operations in **Sample** mode before switching to the other modes.



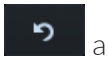
Sample Operations Menu

The following describes the operations you can perform:

Operation	Description
Show Marker Animation	Enable to have the start, end, loop start and loop end markers animate to show the effect of any assigned modulation. Note that dragging the markers is not allowed when this option is enabled.
Zoom to Start and End	Reset the display zoom setting to the default showing both the start and end markers. By default, when this setting is disabled (no check mark present), you can click-drag up and down in the waveform to zoom in and out, and drag left and right to pan the waveform left and right. When enabled (showing a check mark), the waveform maintains a consistent display showing the entire waveform from start to end, without the ability to zoom or pan using your mouse. **Important:** Selecting this option toggles the setting; you need to deselect this option to allow you to zoom the display again.
Snap Off	The next four menu items relate to how the sample playback start and end points or loop points are adjusted. With **Snap Off**, no snapping is applied. You can place the start and end, or loop points freely along the waveform, without any restrictions or alignment to specific reference points. This provides full flexibility when working with waveforms but requires careful manual placement to avoid clicks, pops, or timing issues.
Snap to Zero	Start, end, or loop points snap to the nearest zero-crossing in the waveform. A zero-crossing is a point where the waveform amplitude is zero. This prevents audio clicks or pops when the playback starts, ends, or loops, as abrupt transitions between non-zero amplitudes can create artifacts. This is ideal for ensuring smooth playback and transitions in the waveform.
Snap to Beats	Start, end, or loop points align to the nearest beat grid, based on the tempo of your track. This ensures that the waveform points or loops are musically synchronized with the track tempo. This is useful for rhythmic or tempo-synced loops, where precise timing is essential.

Operation	Description
Snap to Loop	The start or end points snap to the nearest pre-defined loop points in the waveform. This keeps the points aligned with the loop structure, ensuring seamless looping without unintended offsets. This is helpful when working on a pre-looped sample or creating a loop that must align perfectly.
Fade Edges	Apply a fade-in and fade-out at the start and end of the sample. You can choose a setting from 1ms to 128ms as well as **None**. This helps smooth out any abrupt changes in amplitude that could cause unwanted artifacts such as clicks or pops when the sample is triggered or looped.
Normalize	Normalize the sample by adjusting its overall volume to maximize the peak loudness without introducing distortion. This ensures consistent volume levels while preserving the original dynamics of the sample.
Reverse	Reverse the audio sample.
Trim	Trim the sample to the current start and end markers.
Slicing Off	See “Slicing Samples” below.
Slice Auto	
Slice Manual	

The **Fade Edges**, **Normalize**, **Reverse**, and **Trim** operations are non-destructive to the original sample file.

You can therefore easily undo and redo the operations using the  and  button buttons respectively.

In addition, you can choose **Reload Sample** in the sample menu if you need to clear all operations and return to the original sample.

Slicing Samples

Serum can help you slice audio samples into smaller segments, making it easier to trigger specific parts individually. You can use this technique to create new rhythmic patterns, isolate key elements, or remix a sample to fit a new sonic context.

Serum offers three slicing options:

- Slicing Off — Turn slicing off
- Slice Auto — Automatically slice the sample using a user-configurable threshold (sensitivity)
- Slice Manual — Automatically slice the sample, and then allow you to manually adjust the slices

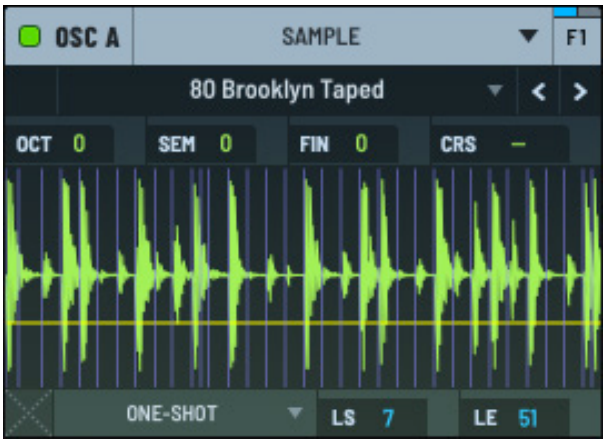
The process for slicing samples is identical in **Sample**, **Granular**, and **Spectral** modes.

Auto Slicing

Right-click the sample and choose **Slice Auto** in the context menu. Serum automatically slices the sample.

Notice the yellow horizontal line. This indicates the slicing threshold (sensitivity). You can move this line to adjust the slicing threshold.

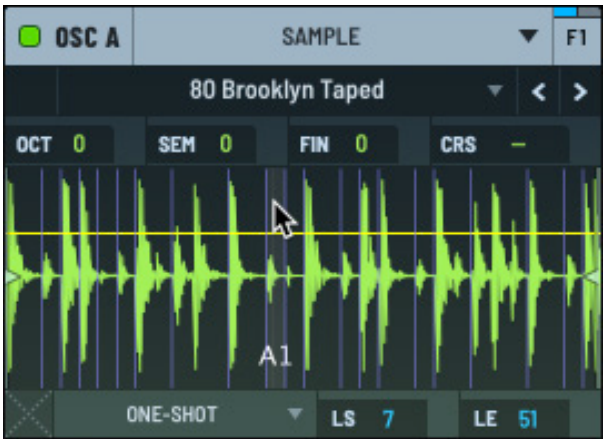
Dragging the line down decreases the slicing threshold, producing more slices. Dragging the line up increases the threshold, resulting in less slices.



Auto Slicing

After setting a threshold, hover the mouse over the slices. Serum displays the note assigned to the particular slice.

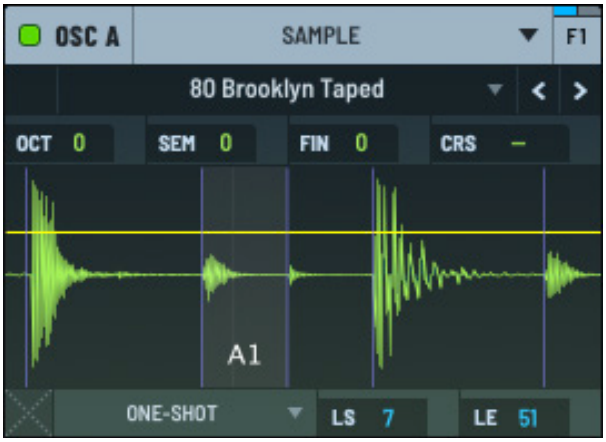
In this example, the highlighted slice is playable using A1.



Adjusting the threshold

Click and drag down and up to zoom in and out of the slices. You can also zoom in and out using the mouse wheel.

When you are zoomed in, you can drag left and right to pan the display.



Zooming In and Out

Manual Slicing

Right-click the sample and choose **Slice Manual** in the menu. Serum automatically slices the sample and then gives you the option to manually adjust the slices.

Zoom into the slices using your mouse wheel. Note that zooming using click and drag is disabled when manual slicing enabled.

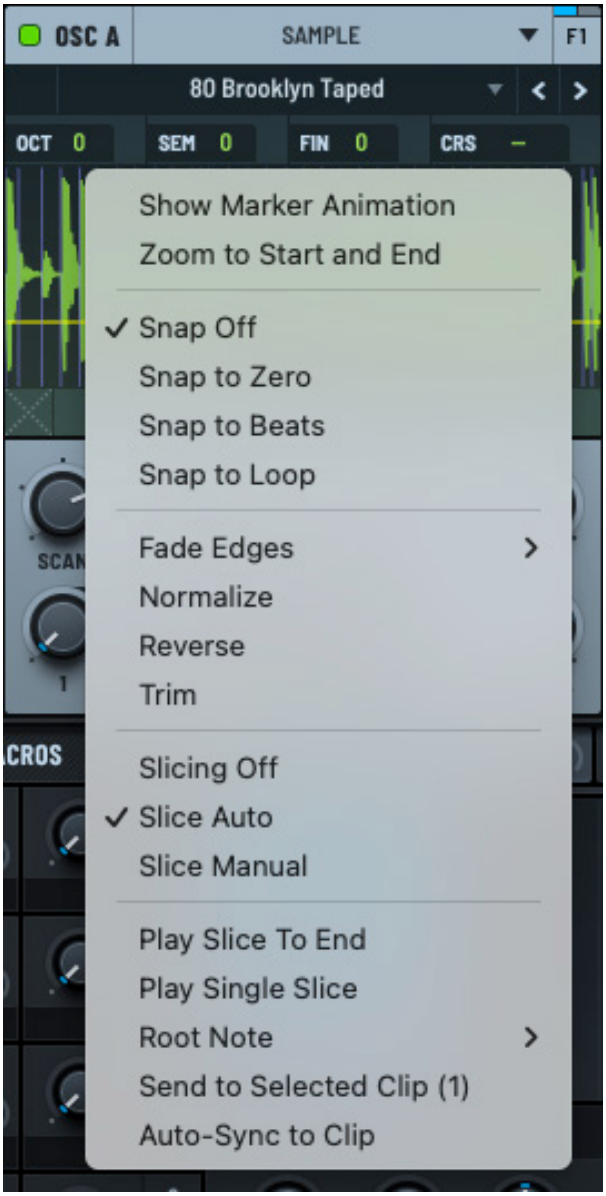
Grab a slice handle and move it to the new location. Continue adjusting other slices, as needed.

Option-click (macOS) or Alt-click (Windows) to add a slice. When the cursor is positioned over an existing slice, Option- or Alt-clicking removes the slice.

Manually Adjusting Slices

Slicing Options

If you select one of the slicing options (auto or manual), the following additional options appear in the context menu.



Extra Slicing Options

The following describes the options:

- Play Slice to End — Toggle this option to play from the triggered slice to the end of the sample.

  Otherwise playback stops at the next slice marker.

- Play Single Slice — Toggle this option to have Serum play just one of the slices whenever any note is triggered.

  The playback rate of the slice changes based on the trigger note (faster rate for notes higher in the register). You can assign a mod source to the "single slice" destination to control which slice is played.

  You can use this option in combination with **Play Slice to End** to play the entire sample using any note, with the playback rate increasing as you move up the register.

- Root Note — Set the note designated to play the first slice. You can choose any C note from C-1 to C8.

- Send to Selected Clip (x) — Send the slices to the currently-selected clip (by default, **Clip 1**) in the **CLIP** module, with each slice assigned to the corresponding trigger note.

  For example, if you select C3 as the root note, choosing this option assigns the slices to successive notes in the currently-selected clip, starting at C3.

- Auto-Sync to Clip — Similar to the previous option in that the slices are sent to the currently-selected clip, with each slice assigned to the corresponding trigger note. However, when you modify the slices (by changing the slice threshold), the clip is automatically modified to reflect the new slices.

Loop Menu

Use the loop menu to specify the loop mode for the sample.

Similar to the sample operations (described in the previous section), the loop menu options available in **Sample** mode are also available when working in **Granular** and **Spectral** modes.

Sample Loop Menu

You can choose from among the following options:

Option	Description
One-shot	The sample plays forward for the duration of the note. This technique is commonly used for sounds like drums, percussion hits, or sound effects, where the full duration of the sample is essential.
Fwd Loop	The sample plays from the start marker to the loop end marker, and then loops back to the loop start marker. This allows you to play the onset of the audio and then stay sustained in the loop. This method is ideal for sustaining sounds, such as a held violin note or a drone, where you want the sample to maintain a consistent, ongoing tone. Hover over the sample display and set the loop start and end by dragging the corresponding blue markers. Click and drag to move the loop to a new location.

Option	Description
Rev Loop	The sample plays from the start marker to the loop end marker and then reverses playback direction to loop back to loop start, which then loops backwards to the loop end. This creates a unique effect where the sound appears to play in reverse repeatedly, which can add an interesting, unconventional texture to the music. Hover over the sample display and set the loop start and end by dragging the corresponding blue markers. Click and drag to move the loop to a new location.
Fwd/Rwd Loop	The sample plays in a forward/reverse loop for the duration of the note. Note that the sample plays from the start marker uninterrupted until you reach the loop. This creates a seamless, ping-pong-like effect where the sound alternates between playing forward and in reverse, providing a smooth and uninterrupted looping experience. This type of playback is especially useful for creating evolving and dynamic textures, as it helps avoid the abrupt transitions or potential clicks that can occur with traditional forward-only looping. Hover over the sample display and set the loop start and end by dragging the corresponding blue markers. Click and drag to move the loop to a new location.
Tailed	The sample plays forward from halfway through the sample to the end (the tail) and then loops the tail of the sample as the amplitude decays. This allows the sound to fade out naturally rather than cutting off abruptly.
Relative Loop	The looped section of the sample changes dynamically based on the playback start position. Rather than always looping between fixed start and end points, the loop moves relative to the playback start marker. This can be useful when automating or modulating the start position.
Link Loop Length	The loop end marker moves relative to the loop start marker, keeping the loop length consistent. This can be useful when automating or modulating the loop start position.
Exit Loop on Release	When a key is released and the amplitude envelope is in the release phase, playback exits the loop and plays to the end of the sample.

Setting the Loop Start and End

You can set the loop start and end points either by dragging markers or by setting values in the **LS** (loop start) and **LE** (loop end) fields.

To drag markers, hover over the top of the sample display and drag either of the blue markers that appear, or between them to drag both together.

Alternatively, click and drag in the **LS** and **LE** fields to move the respective markers. To set a specific value, double-click the field and type the appropriate value.

You can move the loop start and end markers while the note is looping to help you find the best setting.

Setting Loop Start and End

Use the default modifier (Cmd/Ctrl-click or double-click, depending on **GLOBAL** setting) between the markers to reset the loop points to the playback start/end marker positions.

It is not possible to drag the loop markers outside the playback start/end markers.

You can, however, drag, automate, or modulate the markers so that the loop end marker is before the loop start marker. In this case, the loop direction is reversed.

Setting the Crossfade

You can set a crossfade on a sample loop to create smoother transitions at the loop points of the audio sample.

Without a crossfade, loops can sometimes result in abrupt or noticeable clicks, pops, or tonal inconsistencies, especially when the end and start of the loop have mismatched waveforms.

Crossfading helps address this by blending the overlapping regions at the loop boundaries.

Click the  button and drag up to set the crossfade amount. A light blue curve shows the crossfade graphically.

Setting a Loop Cross Fade

Setting Sample Parameters

You can set the crossfade, unison (including detune and blend), and adjust the waveform warp mode.

You can also adjust the pan and level of the signal.

Sample Parameters

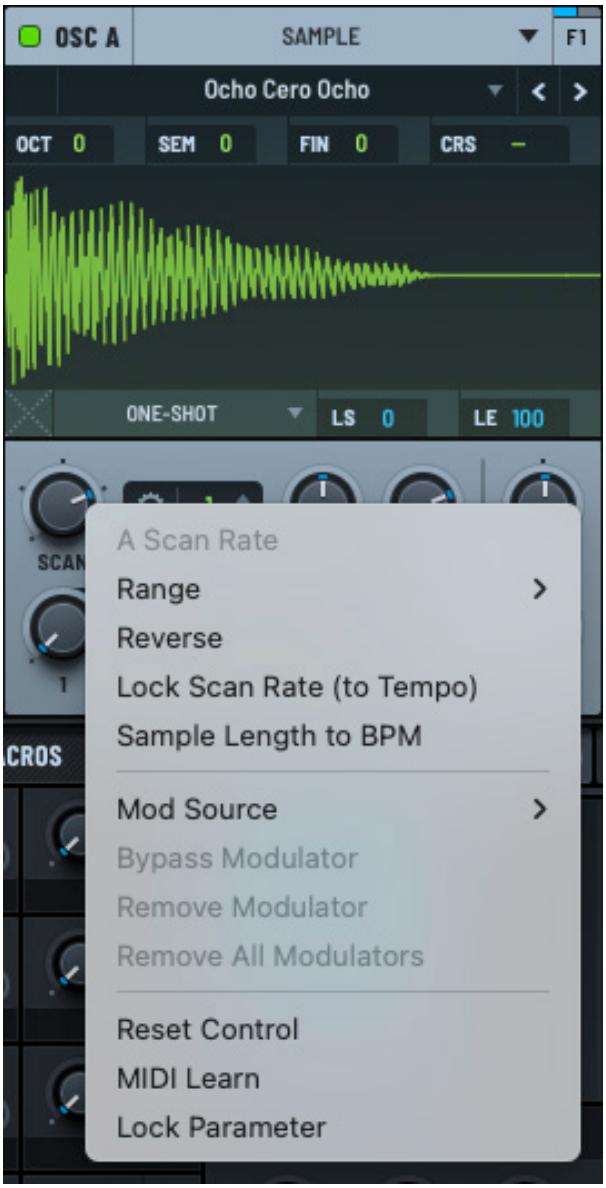
Scan

Use the **SCAN** knob to set the speed and direction of the sample playback.

You can adjust the following parameters related to the scan setting:

- Range — Set the range of the scan knob. The choices are: +/- 200% (default), +/- 400%, and +/- 800%.
- Reverse — Reverse the scan direction. You can automate and modulate this control to switch playback direction.
- Lock Scan Rate (to Tempo) — Change the scan rate when the tempo changes.
- Sample Length to BPM — Set the sample length based on the BPM set in your host DAW.

  The **SCAN** knob changes to **RATE**, allowing you to set the scan rate using beats and bars.



Sample Scan Menu

Unison

Use the **UNISON** control to set the number of unison voices, effectively "stacking" oscillators in a way that is similar to playing multiple notes of the same pitch, but slightly detuned.

Click the field and drag to set the appropriate value. You can also double-click the field and type a value.

This is similar to the unison setting for other oscillator modes.

Note: Unison causes Serum to generate multiple voices, raising CPU usage. The color of the **UNISON** field changes as you increase the number of unison voices as a reminder of the CPU consumption.

Click the  button to display the unison settings.

Sample Unison Settings

You can specify the following settings:

Setting	Description
MODE	Set the detune mode, from among the following: • Linear — The detuning between each additional voice increases in a consistent, linear fashion. This means that the pitch of each voice is spaced evenly in terms of frequency, creating a smooth spread that retains an even distribution. This mode can sound very controlled and smooth, giving a thick and coherent texture. • Super — Multiple voices are slightly detuned from each other but with a special emphasis on creating a dense and powerful sound, often with a slight stereo spread. Use this mode to create a lush, wide sound, especially with supersaw sounds, where the detuned voices give a rich, full sonic character.

Setting	Description
MODE (cont.)	• Exp — The detuning between voices increases exponentially as you move away from the central pitch. This means that the spacing between voices gets wider more quickly as you move outward, creating a more dramatic spread compared to linear. This mode can create unique, rich textures, especially for sounds that need to be more aggressive or have a strong presence in a mix. • Inv — The detuning behavior is inverted, with lower voices detuned more sharply compared to higher ones. Use this mode to create interesting phasing effects or to produce sounds with a less conventional detuning profile. • Random — Introduces a random element to the detuning of each voice. Instead of being evenly spaced, the voices are detuned unpredictably. This can create a more organic or chaotic sound, ideal for achieving textures that are less polished and more natural or experimental.
STACK	Set the unison stacking.
WIDTH	The extent to which the unison voices are spread out across the stereo field, determining how wide or narrow the resulting sound feels in a stereo mix. When you increase the width parameter, the unison voices are spread farther apart in the stereo spectrum, placing some voices more to the left and others more to the right. This creates a sense of space and a wider, more immersive sound. Conversely, decreasing the width narrows the stereo spread, making the sound more centered and focused. This can be useful for sounds that need to be more direct or fit better in a dense mix without dominating the stereo field.
RANGE	The extent or range of detuning applied to the unison voices, determining how far apart in pitch the individual voices are spread around the central frequency of the original sound. When the range is set to a low value, the detuning between the unison voices is minimal. The voices remain relatively close in pitch to the original frequency, creating a subtle and smooth chorusing effect. This can add warmth and a slight thickness to the sound without drastically changing its character. When the range is increased, the detuning becomes more pronounced, and the unison voices are spread further apart in pitch. This creates a more dramatic and sometimes more chaotic or dissonant effect, which can be useful for creating lush, wide textures or aggressive, detuned leads.

Setting	Description
START	Apply a random offset to the starting position for each unison voice. This creates subtle timing differences, which can make the sound more lively, complex, and textured.
SPAN	Apply a fixed offset to the starting position for each unison voice.
WARP 1	Spread out the warp amount applied to each voice around the current **WARP 1** knob position.
WARP 2	Spread out the warp amount applied to each voice around the current **WARP 2** knob position.

You can further adjust the unison setting using the **DETUNE** and **BLEND** knobs.

Unison Detune

Use the **DETUNE** knob to specify the tuning offset +/- for the additional voices. This is only applicable when unison is enabled (set to a value above 1).

Unison Blend

Use the **BLEND** knob to specify the level offset of the unison voices versus the "central" unison voice or voices (1 if an odd number, 2 if an even number of unison).

You can think of **BLEND** as a wet/dry mix between a unison (wet) and non-unison (dry) sound. The default value of 75% is an even blend between all the voices. Note that this is only applicable when the number of unison voices is greater than two.

Warp

Setting the warp allows you to manipulate the playback/sound of the wavetable oscillator.

By default, warp is set to **OFF** (as displayed next to the corresponding knob). Clicking the current setting displays a menu from which you can choose from among the available warp modes.

You can also use the **< >** arrows to conveniently switch between different warp modes without having to open the menu. After selecting a mode, you can use the knob to set the depth.

See "Exploring the Warp Modes" on page 49 for detailed information about the available warp modes.

Pan

Use the **PAN** knob to control the placement of the waveform in the stereo field (left to right).

Level

Use the **LEVEL** knob to control the output volume of the oscillator.

Switching a Sample to a Wavetable

You can quickly and easily create a wavetable from a sample in Serum.

With a sample already loaded, click the **Sample** menu and hover over **Switch to Wavetable** in the menu.

The menu of import options appears. These are the same options that appear when you import audio as a wavetable in other areas of Serum.

Choose one of the menu options.

The oscillator switches to **Wavetable** mode with the converted wavetable loaded.

Switch to Wavetable Menu

See “Importing Multi-Cycle Waveforms” on page 291 for a detailed description of each option.

Using Granular Synthesis

Serum features an easy-to-use granular synthesis mode that manipulates audio samples by breaking them down into tiny segments called grains, and then recombining them in various ways to create new textures and sounds. Each grain typically lasts only a few milliseconds and can be individually controlled in terms of pitch, shape, duration, and playback speed.

By layering, overlapping, and modifying these grains, you can create complex and evolving soundscapes, offering a high degree of flexibility and experimentation beyond traditional synthesis methods.

One of the key strengths of granular synthesis is its ability to transform audio in real-time, whether stretching sounds, altering pitch without affecting duration, or creating rich, atmospheric textures from even the simplest of recordings. Using the granular mode, you can create everything from shimmering, ethereal pads to glitchy, fragmented effects, offering limitless possibilities for sonic exploration.

Note: Granular synthesis can be CPU intensive, especially when compared to Wavetable, Sample, or Multisample modes.

Selecting Granular Synthesis

Using **OSC A**, **OSC B**, or **OSC C**, click the header and choose **Granular** in the context menu.

The oscillator switches to granular mode.

Granular in the Oscillator Menu

If you already chose a sample in the **Sample** or **Spectral** modes, that sample appears in the waveform display.

Otherwise, the display is empty.

Granular Synthesis

Click the drop-down menu and choose a sample from the list of presets.

In addition to tonal and non-tonal factory presets, you can load Serum wavetables as samples.

You can also load a sample by choosing **Load Sample** in the menu and selecting the file using the dialog that appears.

Note: After loading a sample, you can use this menu to show the same location on your computer or reload the sample.

Granular Menu

When a sample loads, the waveform appears in the display.

Granular Sample Loaded

Setting the Sample Start and End

You can easily set the sample start and end points directly.

Hover over the sample and drag either (or both) of the markers that appear (on the left or right).

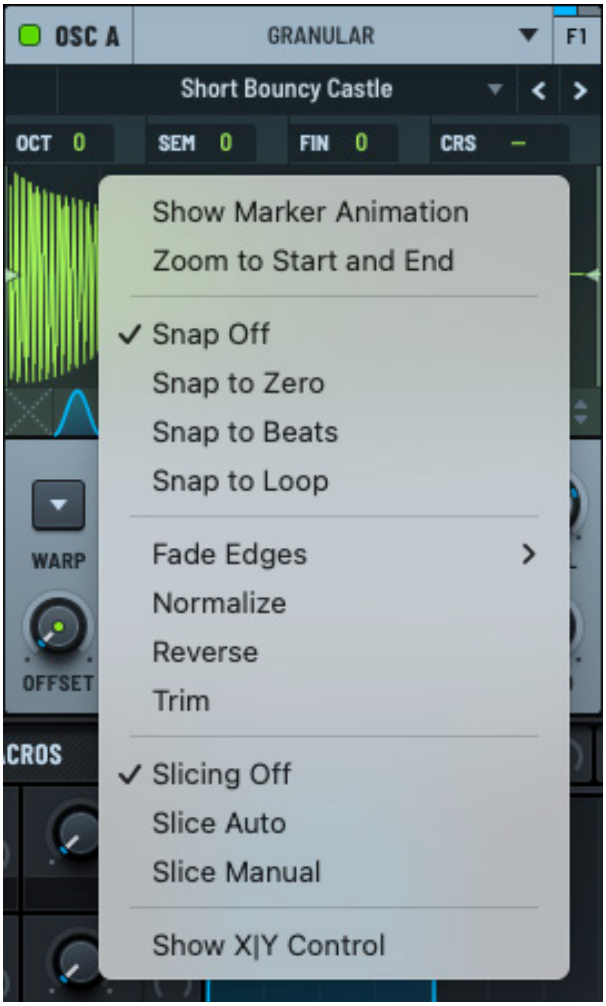
Setting Sample Start and End

Performing Granular Operations

You can perform a range of operations on the loaded sample.

Right-click the sample and choose an operation using the context menu.

Most of the operations available in **Granular** mode are also available in **Sample** mode. See “Performing Sample Operations” on page 71 for complete details about these operations.



Granular Operations Menu

Loop Menu

Use the loop menu to specify the loop mode for the sample.

Most of the loop options available in **Granular** mode are also available in **Sample** mode. See “Loop Menu” on page 77 for more information about these options.

Setting Loop Grains

You can select **Loop Grains** using the Loop menu. This sets the grain playback to respect loop markers.

Granular Loop Menu

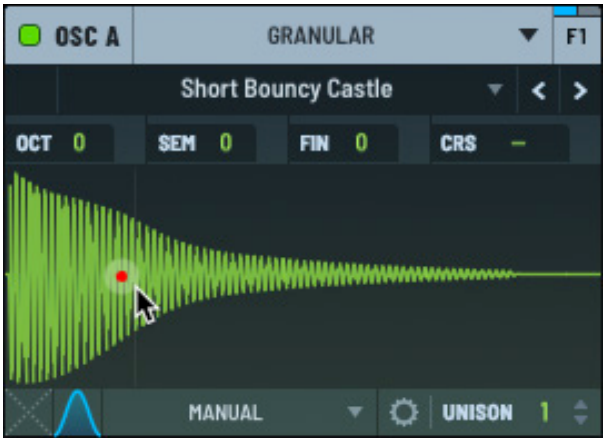
Setting Manual Mode

You can select **Manual** mode using the loop menu. When enabled, the playhead is replaced by a red X|Y dot and the **SCAN** knob changes to control the horizontal position of the dot.

When a note is played, the sample does not scan. Instead, you can freely automate or modulate the playback position.

You can do this by dragging any mod source to the X|Y dot to modulate either the position or the parameter that you assigned to the Y axis (if any).

Note that slicing is not available when **Manual** is selected.



Manual Mode

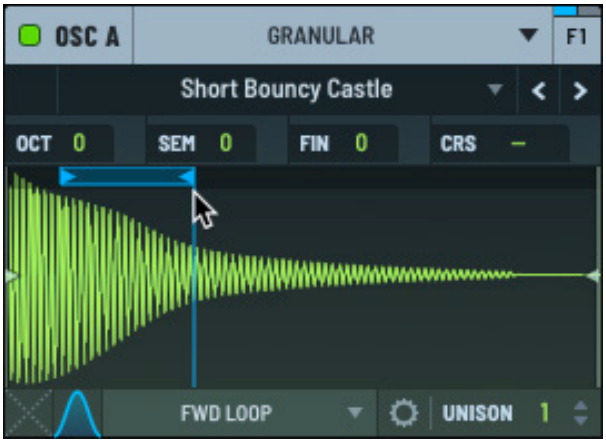
Setting the Loop Start and End

You can set the loop start and end points either by dragging markers or by setting values in the **LS** (loop start) and **LE** (loop end) fields.

To drag markers, hover over the top of the sample display and drag either of the blue markers that appear, or between them to drag both together.

Alternatively, click and drag in the **LS** and **LE** fields to move the respective markers. To set a specific value, double-click the field and type the appropriate value.

You can move the loop start and end markers while the note is looping to help you find the best setting.



Setting Loop Start and End

Use the default modifier (Cmd/Ctrl-click or double-click, depending on **GLOBAL** setting) between the markers to reset the loop points to the playback start/end marker positions.

It is not possible to drag the loop markers outside the playback start/end markers.

You can, however, drag, automate, or modulate the markers so that the loop end marker is before the loop start marker. In this case, the loop direction is reversed.

Setting the Crossfade

You can set a crossfade to individual grain playback.

Click the  button and drag up to set the crossfade amount. A light blue curve shows the crossfade graphically.

Note that since crossfade is applied to grain playback, you need to enable **Loop Grains** in the loop menu to have this take effect.

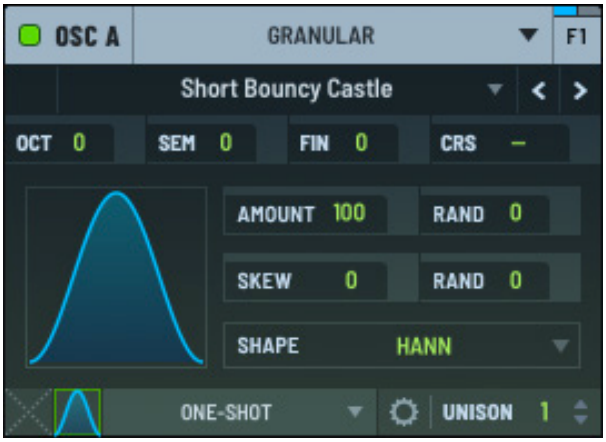
Setting a Loop Cross Fade

Window Amount

Click the  button to access the grain amplitude window settings.

The display changes to show the information you can set.

Option-click (macOS) or Alt-click (Windows) and drag the button to quickly change the window amount and skew without having to access the window settings.



Grain Amplitude Window Settings

The following table describes the settings you can specify:

Setting	Description
AMOUNT	Set the influence of the window curve. You can also set the per-grain randomization for the amount. Click the corresponding field and drag to set the appropriate value. You can also double-click the field and type a value.
SKEW	Set the window skew. You can also set the per-grain randomization for the skew. Click the corresponding field and drag to set the appropriate value. You can also double-click the field and type a value.
SHAPE	Set the grain window shape. A representation of the corresponding shape appears on the left.

The shape of each grain determines how its volume evolves over time, affecting the attack and decay characteristics of the sound. By selecting different grain shapes, such as smooth fades or abrupt cuts, you can dramatically alter the articulation and texture of the sound.

The following table describes the available grain window shapes:

Shape	Description
Hann	Apply a smooth, symmetrical fade-in and fade-out to each grain, using a cosine-shaped envelope to taper the volume. This creates a soft, natural sound with gradual attacks and decays, reducing any harsh transitions between grains for a more cohesive texture.

Shape	Description
Welch	Shape each grain using a smooth parabolic curve. This creates a rounded, more focused sound with a strong central emphasis, producing a natural yet distinct grain structure.
Gaussian	Shape each grain using a bell curve, where the volume increases to a peak in the center and then symmetrically decreases. This results in a smooth, gentle grain with soft transitions, producing a natural, rounded sound ideal for creating fluid textures.
Blackman-Harris	Apply a windowing function with steep slopes and a smooth central peak, offering a strong attenuation of the grain's edges. This results in grains with a well-defined central focus and minimized spectral leakage, producing a cleaner, more controlled sound with less interference between overlapping grains.
Sinc	Shape each grain with a distinctive oscillating pattern that gradually fades out, resembling the sinc function used in signal processing. This creates a grain with a sharp, precise center and oscillating tails, producing a unique texture that can add complexity and harmonic richness to the sound.
Tukey	Apply a shape that combines characteristics of both a rectangular and a tapered window, with a flat center portion and smoothly tapered edges. This allows for flexible control over the grain's attack and decay, offering a balance between sharp transitions and gradual fades, making it useful for adjusting the grain's prominence and blending.
Triangle	Shape each grain with a linear rise to a central peak followed by a symmetric linear decay, creating a simple, pointed envelope. This results in a grain with a clear, sharp attack and a smooth, evenly tapered release, providing a clean and minimal texture with straightforward transitions.
Trapezoid	Shape each grain with a gradual linear rise, followed by a flat, sustained middle section, and then a gradual linear decay. This creates a grain with a more extended, even body, allowing for a balanced sound that can blend sharp attacks with sustained tones for smoother transitions and consistent energy.
ExpDec	Shape each grain with an exponential decay, where the sound starts at full volume and quickly fades out in a curved, nonlinear fashion. This creates a grain with a sharp, pronounced attack followed by a rapid, smooth decay, useful for producing sharp, percussive textures or gradually fading sound effects.

Shape	Description
Exp Dec Rev	Shape each grain with a reversed exponential decay, where the grain starts at a low volume and rapidly rises to full volume in a curved, nonlinear fashion. This creates a grain with a smooth, swelling attack followed by a sharp peak, ideal for building tension or creating atmospheric effects with a gradual onset.

Unison

Use the **UNISON** control to set the number of unison voices, effectively "stacking" granular oscillators in a way that is similar to playing multiple notes of the same pitch, but slightly detuned.

Click the field and drag to set the appropriate value. You can also double-click the field and type a value.

This is similar to the unison setting for other oscillator modes.

Note: Unison causes Serum to generate multiple voices, raising CPU usage. The color of the **UNISON** field changes as you increase the number of unison voices as a reminder of the CPU consumption.

Click the  button to display the unison settings.

Granular Unison Settings

You can specify the following settings:

Setting	Description
MODE	Set the detune mode, from among the following: • Linear — The detuning between each additional voice increases in a consistent, linear fashion. This means that the pitch of each voice is spaced evenly in terms of frequency, creating a smooth spread that retains an even distribution. This mode can sound very controlled and smooth, giving a thick and coherent texture. • Super — Multiple voices are slightly detuned from each other but with a special emphasis on creating a dense and powerful sound, often with a slight stereo spread. Use this mode to create a lush, wide sound, especially with supersaw sounds, where the detuned voices give a rich, full sonic character. • Exp — The detuning between voices increases exponentially as you move away from the central pitch. This means that the spacing between voices gets wider more quickly as you move outward, creating a more dramatic spread compared to linear. This mode can create unique, rich textures, especially for sounds that need to be more aggressive or have a strong presence in a mix. • Inv — The detuning behavior is inverted, with lower voices detuned more sharply compared to higher ones. Use this mode to create interesting phasing effects or to produce sounds with a less conventional detuning profile. • Random — Introduces a random element to the detuning of each voice. Instead of being evenly spaced, the voices are detuned unpredictably. This can create a more organic or chaotic sound, ideal for achieving textures that are less polished and more natural or experimental.
STACK	Set the unison stacking.
DETUNE	Specify the tuning offset for the additional voices.
BLEND	Specify the level offset of the unison voices versus the “central” unison voice or voices (1 if an odd number, 2 if an even number of unison). You can think of **BLEND** as a wet/dry mix between a unison (wet) and non-unison (dry) sound. The default value of 75% is an even blend between all the voices. Note that this is only applicable when the number of unison voices is greater than two.

Setting	Description
WIDTH	The extent to which the unison voices are spread out across the stereo field, determining how wide or narrow the resulting sound feels in a stereo mix. When you increase the width parameter, the unison voices are spread farther apart in the stereo spectrum, placing some voices more to the left and others more to the right. This creates a sense of space and a wider, more immersive sound. Conversely, decreasing the width narrows the stereo spread, making the sound more centered and focused. This can be useful for sounds that need to be more direct or fit better in a dense mix without dominating the stereo field.
RANGE	The extent or range of detuning applied to the unison voices, determining how far apart in pitch the individual voices are spread around the central frequency of the original sound. When the range is set to a low value, the detuning between the unison voices is minimal. The voices remain relatively close in pitch to the original frequency, creating a subtle and smooth chorusing effect. This can add warmth and a slight thickness to the sound without drastically changing its character. When the range is increased, the detuning becomes more pronounced, and the unison voices are spread further apart in pitch. This creates a more dramatic and sometimes more chaotic or dissonant effect, which can be useful for creating lush, wide textures or aggressive, detuned leads.
START	Apply a random offset to the starting position for each unison voice. This creates subtle timing differences, which can make the sound more lively, complex, and textured.
SPAN	Apply a fixed offset to the starting position for each unison voice.
SPAWN PATTERN	The timing offset at which unison grain voices are spawned. With **Together** (default), all unison grain voices spawn at the same time. With the other options, the spawning of unison grain voices are offset into the period before the next spawn, as follows: • Even — The timing of unison grain voices is spread out evenly • Exp — The timing between unison grain voices increases over the period before the next spawn • Random — The timing between unison grain voices is randomly distributed over the period before the next spawn

Setting the X|Y Control

If you choose to show the **X|Y Control** using the context menu, a red dot appears in the display.

In addition, a new menu option appears in the context menu allowing you to select the **Y axis** parameter.

X|Y Control

Use the context menu to choose the **Y axis** parameter.

Y Axis Menu

The following table describes the available **Y axis** options:

Option	Description
None	No Y axis target.
Level	The oscillator level.
Warp	The WARP 1 setting.
Warp 2	The WARP B setting.
Density	The density of the grain cloud.
Grain Length	The duration of each grain.
Window Amt	The window amount for the sample. Sets the influence of the window curve.
Window Skew	The window skew.
Rand Offset	The per-grain randomization of the offset.
Rand Dir	The per-grain randomization of the direction.
Rand Pitch	The per-grain pitch randomization.
Rand Length	The per-grain length randomization.
Rand Pan	The per-grain randomization of the pan setting.
Rand Gain	The per-grain level randomization.
Rand Window	The window amount randomization.
Rand Skew	The window skew randomization.
Rand Warp	The WARP A randomization.
Rand Warp 2	The WARP B randomization.

After selecting a Y axis parameter (other than **None**), an additional menu option appears.

This allows you to choose how to modulate the **Y axis** parameter.

Select a modulation option, as appropriate.

Y Axis Modulation Menu

Setting Granular Parameters

You can set the scan rate, density, and length, as well as adjust the warp mode, among other settings.

You can also adjust the pan and level of the signal.

Granular Parameters

Warp Mode

Setting the warp allows you to manipulate the playback/sound of the sample.

Click the ▼ button to display the warp settings.

Click the ▲ button to hide the settings.

By default, warp is set to **OFF** (as displayed next to the corresponding knob). Clicking the current setting displays a menu from which you can choose from among the available warp modes.

Granular Warp Settings

You can also use the **< >** arrows to conveniently switch between different warp modes without having to open the menu. After selecting a mode, you can use the knob to set the depth.

See “Exploring the Warp Modes” on page 49 for detailed information about the available warp modes.

Scan

Use the **SCAN** knob to set the scan rate. The scan rate controls how quickly the Serum moves through the audio sample to generate grains.

Granular synthesis involves breaking an audio sample into tiny pieces called grains (usually lasting a few milliseconds) and then recombining them to create new sounds. Serum determines the playback start position of a grain by the current playhead position.

The scan rate specifies how quickly the playhead moves through the sample. Setting the scan rate higher spreads the start positions of grains out along the sample more, resulting in less overlap between grains.

Setting the scan rate lower causes movement through the sample to slow down. This can produce a more stretched, evolving, or drone-like sound.

In summary, a higher scan rate can be used to maintain rhythmic accuracy or to create fast-paced, glitch-like effects. A slower scan rate allows for dramatic time-stretching effects, creating elongated and ambient textures.

Setting the scan rate to a negative value reverses the direction of the playhead. Alternatively, setting the scan rate to 0 stops the playhead from moving.

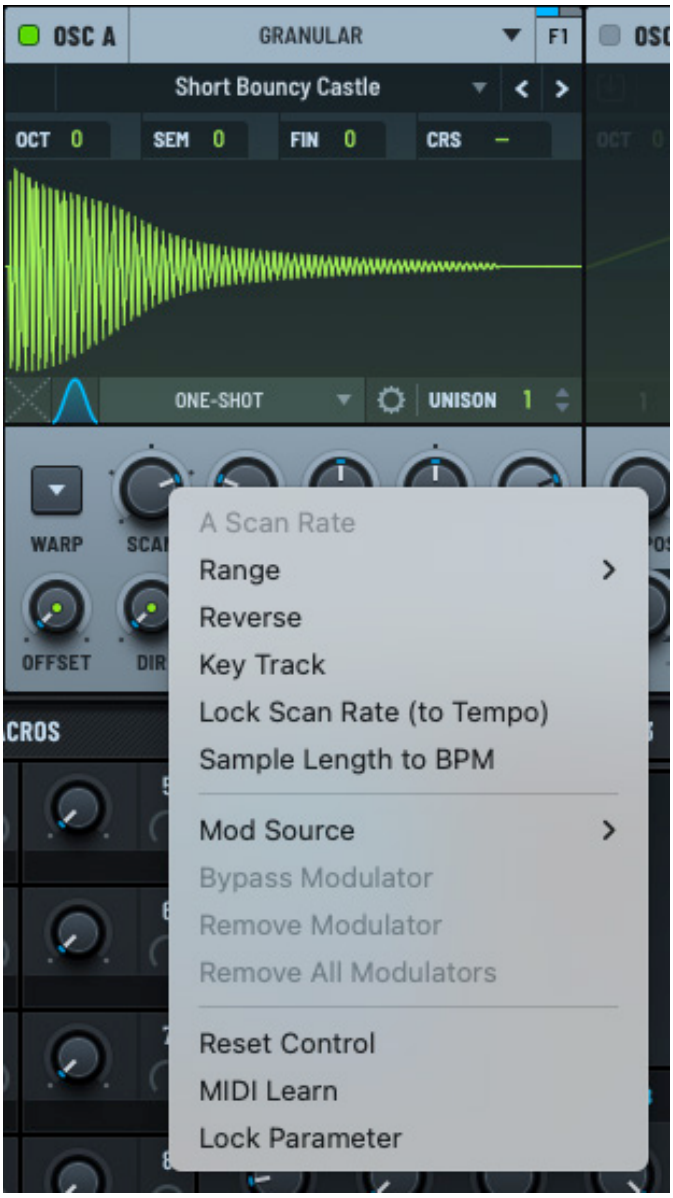
You can adjust the following parameters related to the scan setting:

- Range — Set the range of the scan knob. The choices are: +/- 200% (default), +/- 400%, and +/- 800%.
- Reverse — Reverse the scan direction. You can automate and modulate this control to switch playback direction.
- Key Track — Specify how the scan rate responds to the pitch of the note played.

  With Key Track disabled, the scan rate is fixed regardless of the key played.

  With Key Track enabled, the scan rate changes in proportion to the pitch of the note played (higher-pitched notes increase the scan rate).
- Lock Scan Rate (to Tempo) — Change the scan rate when the tempo changes.
- Sample Length to BPM — Set the sample length based on the BPM set in your host DAW.

  The **SCAN** knob changes to **RATE**, allowing you to set the scan rate using beats and bars.



Granular Scan Menu

Density

Use the **DENS** knob to set the density of the grain cloud.

Density defines the rate at which grains are spawned, according to one of three options that you can specify by right-clicking the knob and choosing from the context menu. The options are:

- Free — Grains are spawned at a rate defined in Hz
- BPM Sync — Grains are spawned at a bar/beat division of host tempo

  When selected, the context menu offers **Triplet** and **Dotted** as additional options, allowing you to specify whether triplet or dotted divisions can be selected with the knob.
- Grains — The spawn rate is calculated as a function of grain length such that a consistent number of grains, as set by the control, is playing at any given time

You can additionally specify two more options using the **DENSITY** knob context menu.

- Jump Start — Enable this option to have multiple grains spawn at note start so that the full density is heard immediately.

  When disabled, there is only a single spawning at note start and the sound builds to full density over subsequent spawnings, giving the note a softer start.
- Max Grains — Set to place a limit on the maximum number of grains that can play at any one time, including unison grain voices.

  Many grains playing simultaneously can consume a lot of CPU. Use this option to help reduce this consumption. If the oscillator tries to spawn a grain when the maximum number is playing, it will skip and wait until one has stopped playing before it spawns again.

Length

Use the **LENGTH** knob to set the duration of each grain.

Setting the duration of each grain allows you to control how long each sound fragment lasts, influencing the overall character of the resulting texture. Shorter grains can produce sharper, more rhythmic sounds, while longer grains create smoother, more sustained tones.

You can specify how the length is determined by right-clicking the **LENGTH** knob and choosing one of three options from the context menu:

- Free — Gain length can be set in seconds or milliseconds
- BPM Sync — Grain length can be set to a bar/beat division of host tempo

  When selected, the context menu offers **Triplet** and **Dotted** as additional options, allowing you to specify whether triplet or dotted divisions can be selected with the knob.

- Percent — Grain length is set to a percent of the density period. Note that this option is not available if you select the **Grains** option with the **DENSITY** knob.

Pan

Use the **PAN** knob to control the placement of the sample in the stereo field (left to right).

Level

Use the **LEVEL** knob to control the output volume of the oscillator.

Setting the Grain Randomization

You can set the grain randomization parameters by adjusting the lower row of knobs.

Grain Randomization

The following table describes the available knobs:

Knob	Description
OFFSET	Set the per-grain randomization of the offset.
DIR	Set the per-grain randomization of the direction.
PITCH	Set the per-grain pitch randomization.
RAND (LENGTH)	Set the per-grain length randomization.
RAND (PAN)	Set the per-grain randomization of the pan setting.
RAND (LEVEL)	Set the per-grain level randomization.

Reversing the Grain Playback Direction Randomization

You can reverse grains as part of the per-grain direction randomization.

Right-click the **DIR** knob and choose **Reverse Grains** in the context menu to toggle the reverse grains feature on or off.

You can automate and modulate this setting to switch playback direction of all grains at once.

Grain Randomization

Switching a Sample to a Wavetable

You can quickly and easily create a wavetable from a sample in Serum.

With a sample already loaded, click the **Granular** menu and hover over **Switch to Wavetable** in the menu.

The menu of import options appears. These are the same options that appear when you import audio as a wavetable in other areas of Serum.

Choose one of the menu options.

The oscillator switches to **Wavetable** mode with the converted wavetable loaded.

Switch to Wavetable Menu

See “Importing Multi-Cycle Waveforms” on page 291 for a detailed description of each option.

Using Spectral Synthesis

Serum features a spectral synthesis mode that generates sound by analyzing and manipulating the frequency spectrum of a sound, breaking it down into its individual frequency components or partials. Unlike other synthesis methods that operate directly on the waveform, spectral synthesis focuses on the harmonic and inharmonic content, allowing for precise control over the timbre and evolution of sound.

By altering specific frequency bands, adding or removing harmonics, or even shifting spectral content over time, this approach can create complex, evolving textures that can range from natural acoustic-like tones to entirely synthetic soundscapes. In practical use, spectral synthesis opens up unique possibilities for morphing and transforming sound in ways that are not achievable with other synthesis techniques.

For instance, by isolating and processing specific frequencies, it is possible to create sounds that gradually shift from one texture to another, or blend multiple sources into a single, coherent output. Additionally, spectral synthesis allows for dynamic filtering and precise spectral editing.

Note: This method can be CPU intensive but provides unparalleled flexibility in shaping sound at a fundamental, spectral level.

Selecting Spectral Synthesis

Using **OSC A**, **OSC B**, or **OSC C**, click the header and choose **Spectral** in the context menu.

The oscillator switches to spectral mode.

Spectral in the Oscillator Menu

If you already chose a sample in the **Sample** or **Granular** modes, that sample appears in the spectral display.

Otherwise, the display is empty.

Spectral Synthesis

Click the drop-down menu and choose a sample, as needed.

In addition to tonal and non-tonal factory presets, you can load Serum wavetables as samples.

You can also load a sample by choosing **Load Sample** in the menu and selecting the file using the dialog that appears.

Note: After loading a sample, you can use this menu to show the same location on your computer or reload the sample.

Spectral Menu

When a sample loads, the spectral representation appears in the display.

Click and drag up and down in the frequency spectrum to zoom the display.

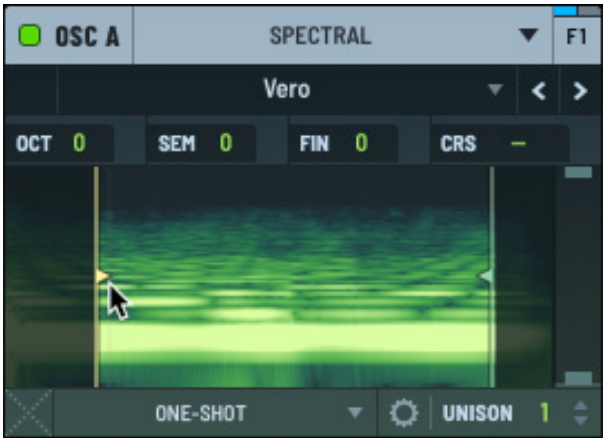
You can tell if the display is fully zoomed out by hovering over the display and checking whether you can see both the start and end markers.

Spectral Waveform

Setting the Sample Start and End

You can easily set the sample start and end points directly.

Hover over the sample and drag either (or both) of the markers that appear (on the left or right).

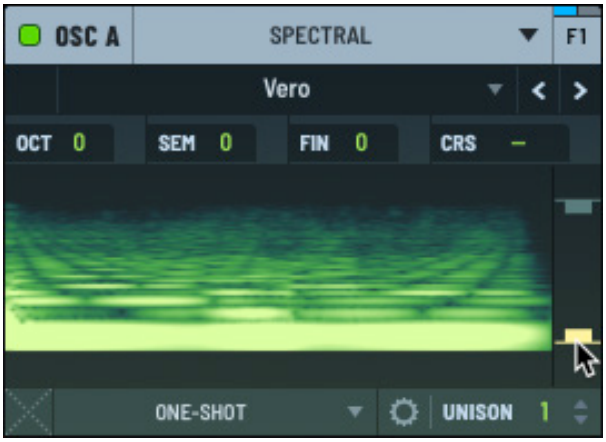


Setting Sample Start and End

Setting the Sample High and Low Frequencies

You can set the sample high and low frequency points directly by dragging the markers to the right of the spectrogram, as needed.

Right-click in the high-low pane to display the context menu. You can toggle options on and off using this menu.



Setting Sample Hi and Low

The following table describes the options you can choose:

Option	Description
Smooth	Apply a fourth-order Butterworth filter at the low and high frequency boundaries, for smoother edges.
Post Warp	Apply the low/high filtering after processing spectral warps.

You can drag and drop any modulation source (envelopes and LFOs) to the high and low frequency markers to modulate the respective control.

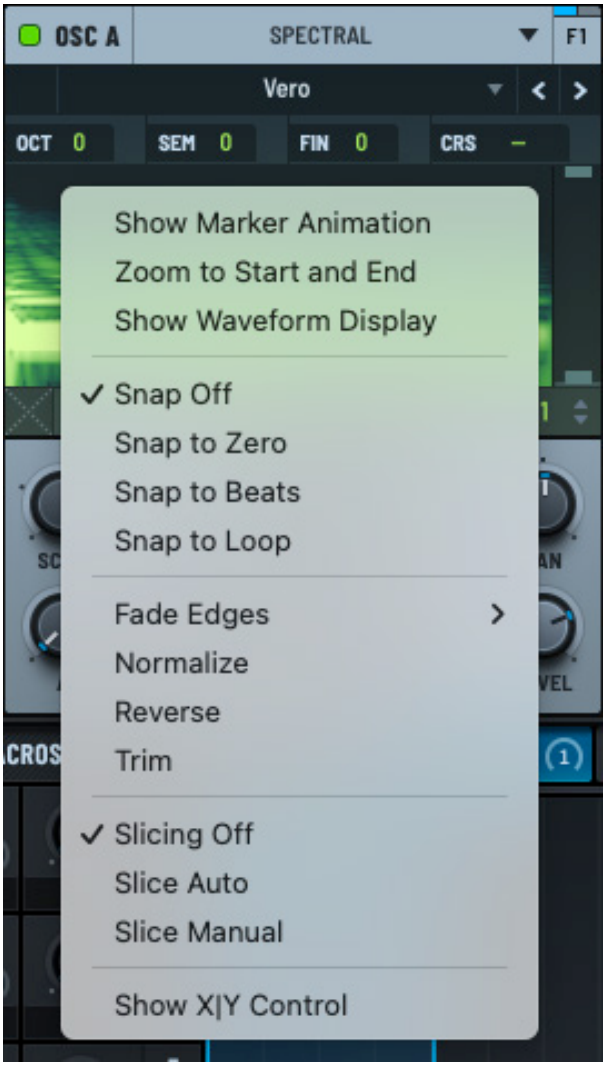
Modulating the Low Frequency

Performing Sample Operations

You can perform a range of operations on the loaded sample.

Right-click the sample and choose an operation using the context menu.

Most of the operations available in **Spectral** mode are also available in **Sample** mode. See “Performing Sample Operations” on page 71 for complete details about these operations.

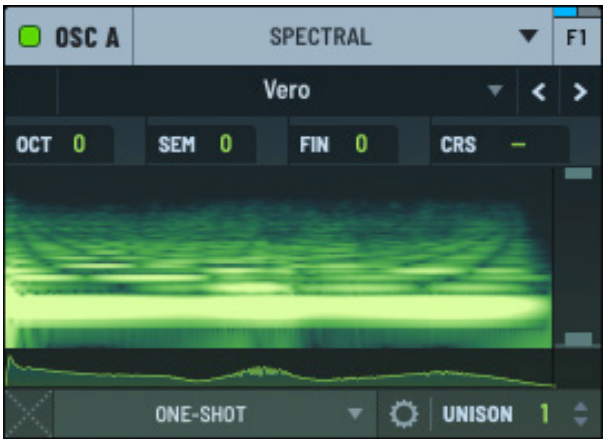


Spectral Context Menu

Showing the Waveform Display

To toggle the display of the waveform directly below the spectral display, right-click the sample and choose **Show Waveform Display** in the context menu.

Deselect the option to hide the waveform display.

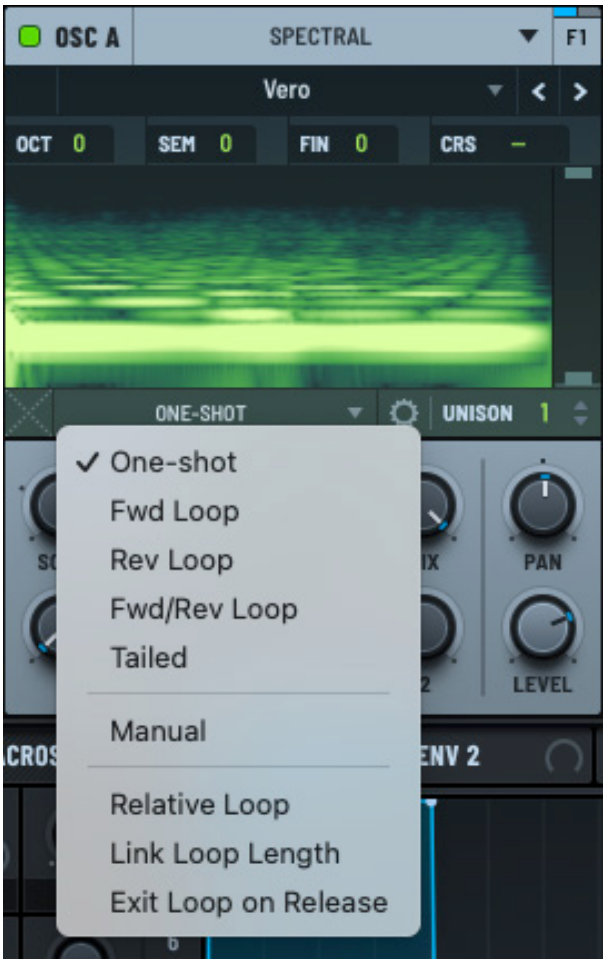


Waveform Display

Loop Menu

Use the loop menu to specify the loop mode for the sample.

Most of the loop options available in **Spectral** mode are also available in **Sample** mode. See “Loop Menu” on page 77 for more information about these options.



Spectral Loop Menu

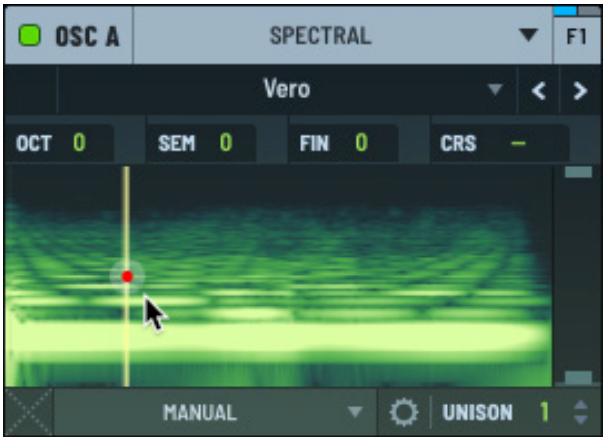
Setting Manual Mode

You can select **Manual** mode using the loop menu. When enabled, the playhead is replaced by a red X|Y dot and the **SCAN** knob changes to control the horizontal position of the dot.

When a note is played, the sample does not scan. Instead, you can freely automate or modulate the playback position.

You can do this by dragging any mod source to the X|Y dot to modulate either the position or the parameter that you assigned to the Y axis (if any).

Note that slicing is not available when **Manual** is selected.



Manual Mode

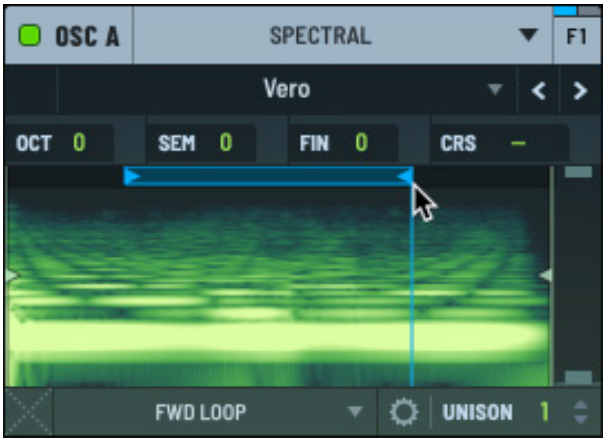
Setting the Loop Start and End

You can set the loop start and end points either by dragging markers or by setting values in the **LS** (loop start) and **LE** (loop end) fields.

To drag markers, hover over the top of the sample display and drag either of the blue markers that appear, or between them to drag both together.

Alternatively, click and drag in the **LS** and **LE** fields to move the respective markers. To set a specific value, double-click the field and type the appropriate value.

You can move the loop start and end markers while the note is looping to help you find the best setting.



Setting Loop Start and End

Use the default modifier (Cmd/Ctrl-click or double-click, depending on **GLOBAL** setting) between the markers to reset the loop points to the playback start/end marker positions.

It is not possible to drag the loop markers outside the playback start/end markers.

You can, however, drag, automate, or modulate the markers so that the loop end marker is before the loop start marker. In this case, the loop direction is reversed.

Setting the Crossfade

You can set a crossfade on a sample loop to create smoother transitions at the loop points of the audio sample.

Click the  button and drag up to set the crossfade amount. A light blue curve shows the crossfade graphically.

Setting a Loop Cross Fade

Unison

Use the **UNISON** control to set the number of unison voices, effectively "stacking" oscillators in a way that is similar to playing multiple notes of the same pitch, but slightly detuned.

Click the field and drag to set the appropriate value. You can also double-click the field and type a value.

This is similar to the unison setting for other oscillator modes.

Note: Unison causes Serum to generate multiple voices, raising CPU usage. The color of the UNISON field changes as you increase the number of unison voices as a reminder of the CPU consumption.

Click the  button to display the unison settings.

Spectral Unison Settings

You can specify the following settings:

Setting	Description
MODE	The detune mode, from among the following: • Linear — The detuning between each additional voice increases in a consistent, linear fashion. This means that the pitch of each voice is spaced evenly in terms of frequency, creating a smooth spread that retains an even distribution. This mode can sound very controlled and smooth, giving a thick and coherent texture. • Super — Multiple voices are slightly detuned from each other but with a special emphasis on creating a dense and powerful sound, often with a slight stereo spread. Use this mode to create a lush, wide sound, especially with supersaw sounds, where the detuned voices give a rich, full sonic character.

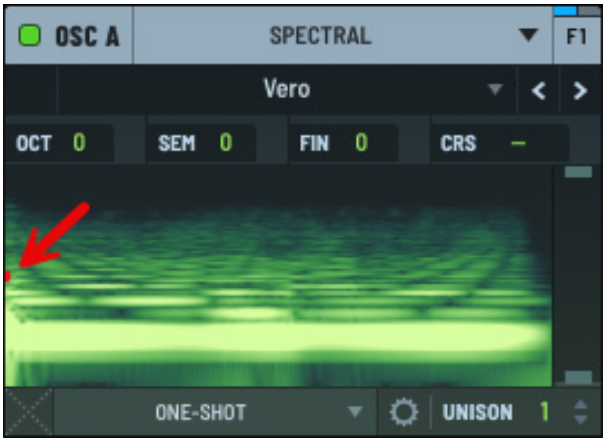
Setting	Description
MODE (cont.)	• Exp — The detuning between voices increases exponentially as you move away from the central pitch. This means that the spacing between voices gets wider more quickly as you move outward, creating a more dramatic spread compared to linear. This mode can create unique, rich textures, especially for sounds that need to be more aggressive or have a strong presence in a mix. • Inv — The detuning behavior is inverted, with lower voices detuned more sharply compared to higher ones. Use this mode to create interesting phasing effects or to produce sounds with a less conventional detuning profile. • Random — Introduces a random element to the detuning of each voice. Instead of being evenly spaced, the voices are detuned unpredictably. This can create a more organic or chaotic sound, ideal for achieving textures that are less polished and more natural or experimental.
STACK	The unison stacking.
DETUNE	The tuning offset for the additional voices.
BLEND	The level offset of the unison voices versus the “central” unison voice or voices (1 if an odd number, 2 if an even number of unison). You can think of **BLEND** as a wet/dry mix between a unison (wet) and non-unison (dry) sound. The default value of 75% is an even blend between all the voices. Note that this is only applicable when the number of unison voices is greater than two.
WIDTH	The extent to which the unison voices are spread out across the stereo field, determining how wide or narrow the resulting sound feels in a stereo mix. When you increase the width parameter, the unison voices are spread farther apart in the stereo spectrum, placing some voices more to the left and others more to the right. This creates a sense of space and a wider, more immersive sound. Conversely, decreasing the width narrows the stereo spread, making the sound more centered and focused. This can be useful for sounds that need to be more direct or fit better in a dense mix without dominating the stereo field.

Setting	Description
RANGE	The extent or range of detuning applied to the unison voices, determining how far apart in pitch the individual voices are spread around the central frequency of the original sound. When the range is set to a low value, the detuning between the unison voices is minimal. The voices remain relatively close in pitch to the original frequency, creating a subtle and smooth chorusing effect. This can add warmth and a slight thickness to the sound without drastically changing its character. When the range is increased, the detuning becomes more pronounced, and the unison voices are spread further apart in pitch. This creates a more dramatic and sometimes more chaotic or dissonant effect, which can be useful for creating lush, wide textures or aggressive, detuned leads.
SPAN	Apply a fixed offset to the starting position for each unison voice.
START	Apply a random offset to the starting position for each unison voice. This creates subtle timing differences, which can make the sound more lively, complex, and textured.
WARP 1	Spread out the warp amount applied to each voice around the current **WARP 1** knob position.
WARP 2	Spread out the warp amount applied to each voice around the current **WARP 2** knob position.

Setting the X|Y Control

If you choose to show the **X|Y Control** using the context menu, a red dot appears in the spectral display.

In addition, a new menu option appears in the context menu allowing you to select the **Y axis** parameter.



X|Y Control

Use the context menu to choose the **Y axis** parameter.

Y Axis Menu

The following table describes the available **Y axis** options:

Option	Description
None	No Y axis target.
Level	The oscillator level.
Warp	The **WARP 1** setting.
Warp 2	The **WARP 2** setting.
Spec Flt Cutoff	The spectral filter cutoff.
Spec Flt Wet/Dry	The spectral filter mix (wet/dry).
Freq Lo	The low frequency (near the bottom right of the spectral display).
Freq Hi	The high frequency (near the top right of the spectral display).

After selecting a Y axis parameter (other than **None**), an additional menu option appears.

This allows you to choose how to modulate the **Y axis** parameter.

Select a modulation option, as appropriate.

Y Axis Modulation Menu

Setting Spectral Parameters

You can set the sample playback, spectral filter cutoff, and adjust the waveform warp mode.

You can also adjust the pan and level of the signal.

Spectral Parameters

Scan

Use the **SCAN** knob to set the speed and direction of the sample playback.

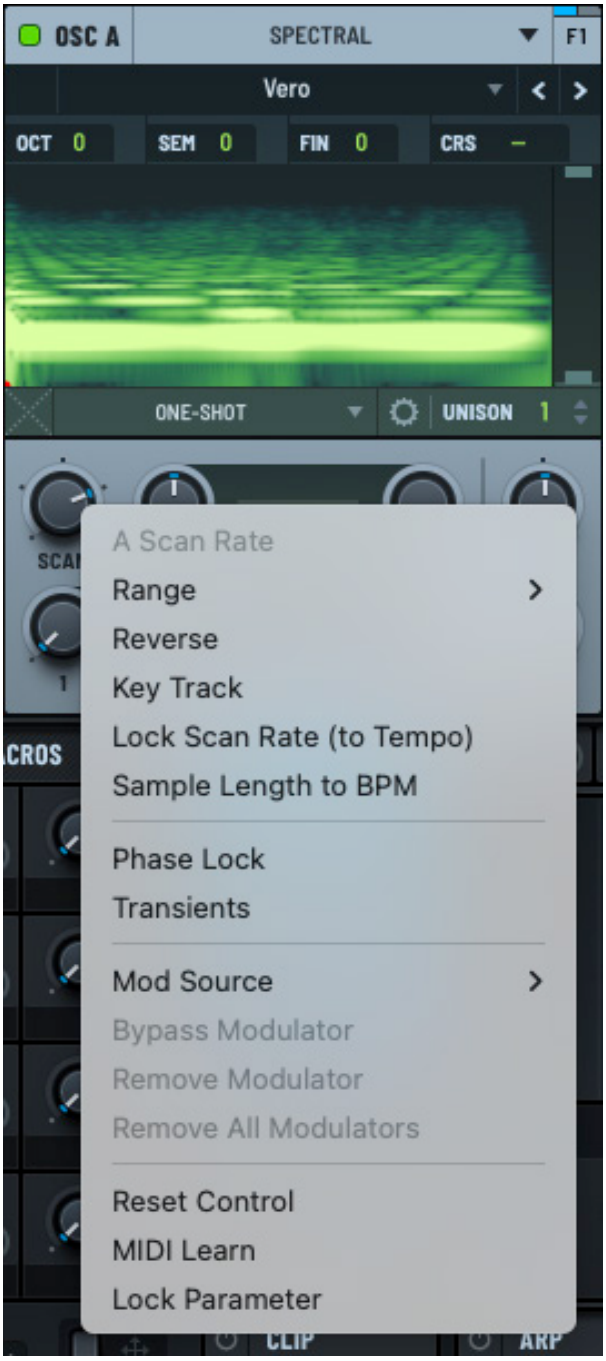
You can adjust the following parameters related to the scan setting:

- Range — Set the range of the scan knob. The choices are: +/- 200% (default), +/- 400%, and +/- 800%.
- Reverse — Reverse the scan direction. You can automate and modulate this control to switch playback direction.
- Key Track — Specify how the scan rate responds to the pitch of the note played.

  With Key Track disabled, the scan rate is fixed regardless of the key played.

  With Key Track enabled, the scan rate changes in proportion to the pitch of the note played (higher-pitched notes increase the scan rate).
- Lock Scan Rate (to Tempo) — Change the scan rate when the tempo changes.
- Sample Length to BPM — Set the sample length based on the BPM set in your host DAW.

  The **SCAN** knob changes to **RATE**, allowing you to set the scan rate using beats and bars.



Spectral Scan Menu

Setting the sample length to BPM allows drum loops or samples to sync with the DAW tempo while still letting you pitch the sample up and down using the keyboard while keeping the tempo consistent.

You can also control the phase lock and transients.

- Phase Lock — Adjust the FFT phases to minimize the audible phase change between FFT blocks

  This can result in a less “smeared” sound, more faithful to the original sample. Consider using this with tonal samples.

- Transients — Preserves transients that would otherwise be smeared by FFT processing

  Consider using this with percussive sounds or drum loops.

Cut

Use the **CUT** knob to set the cutoff of the spectral filter.

This sets the cutoff point of the spectral filter, determining which frequencies are allowed to pass through or are filtered out.

By adjusting the **CUT** knob, you can control the range of frequencies that shape the sound, effectively removing unwanted high or low spectral content to refine the tone and texture of the output.

Filter

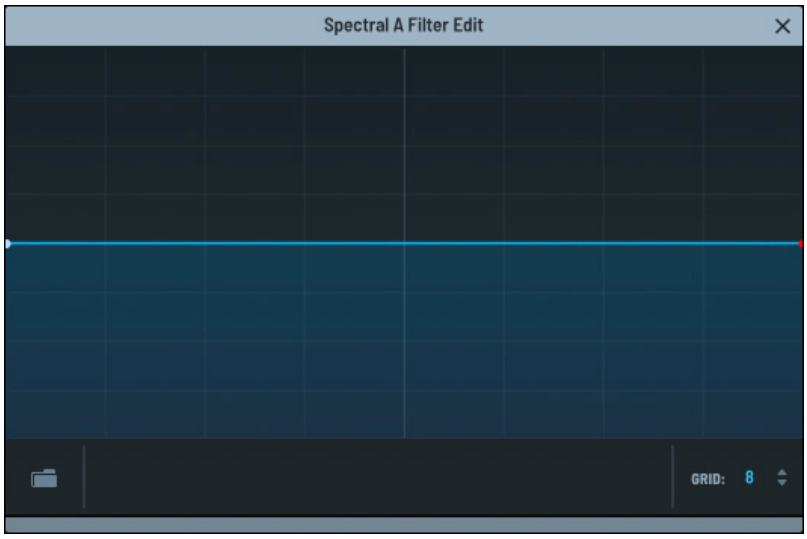
You can create a custom filter curve, choose a filter preset, or choose a wavetable to act as the filter.

Creating a Custom Curve

Click the **FILTER** display to show the spectral filter editor.

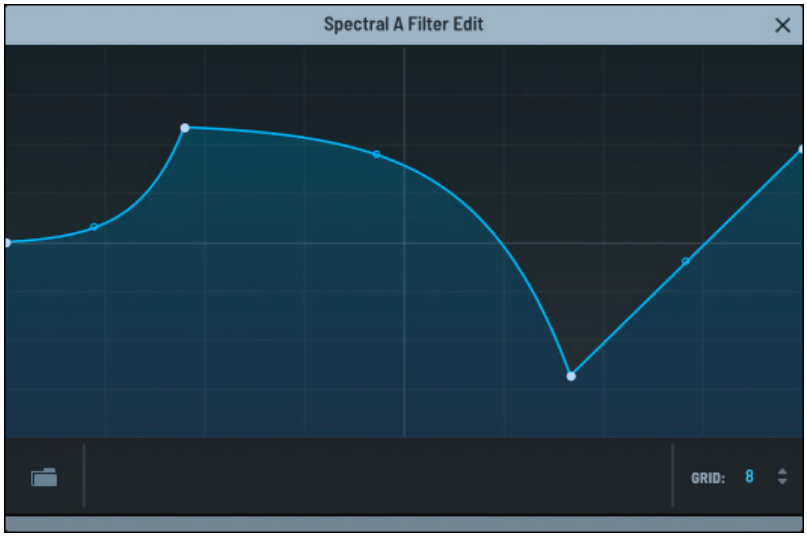
Filter Display

The spectral filter editor appears in a dialog.



Spectral Filter Mask Editor

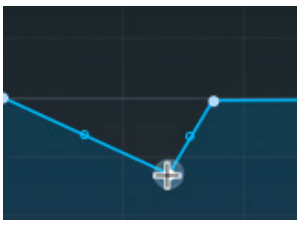
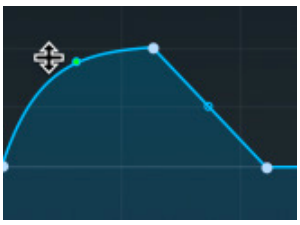
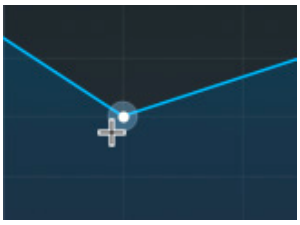
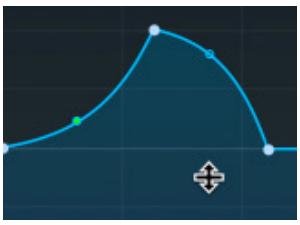
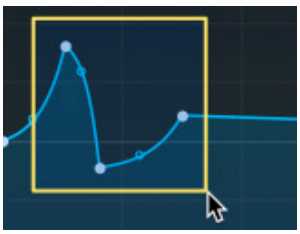
Modify the filter by adding new points and dragging curves.



Spectral Filter Mask Editor (Modified)

The following table describes operations you can perform when editing the filter:

Operation	Graph	Description
Double-click		Add a new point to the mask or remove an existing point.

Operation	Graph	Description
Drag a point		Move a point to a new location.
Drag a curve point		Create or modify a curve between points.
Option/Alt drag a point		Move a point to a new location, constrained to the current grid.
Option/Alt drag a curve point		Create or modify curves between all points simultaneously.
Click and drag to select		Select multiple points. This allows you to drag multiple points simultaneously.

Click and drag in the **GRID** field to set the number of grid divisions. To set a specific value, double-click the field and type the appropriate value.

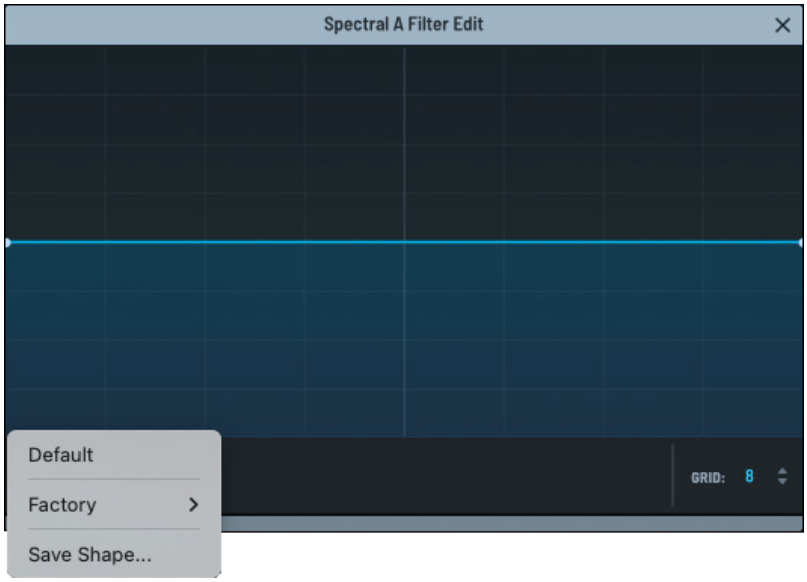
Choosing a Filter Preset

You can choose a factory filter preset, click the **FILTER** display to show the spectral filter editor.

In the filter editor, click the  button and choose a preset in the context menu.

After the preset loads, you can modify the filter curve as needed. Use the menu to save the preset for future use.

Alternatively, choose **Default** in the menu to return to the draft filter curve.



Filter Presets Menu

Choosing a Wavetable

To use a wavetable as a filter, right-click the **FILTER** display and choose a curve in the context menu.

After choosing a preset, a thumbnail of the filter shows to indicate your selection.

Important: It's not possible to edit the filter if you selected a wavetable.

Filter Menu

To return to creating a custom filter, right-click the **FILTER** display and choose **Curve Filter** in the menu.

You can then click the **FILTER** display, as before, to display the spectral filter editor.

Filter Selected

Mix

Use the **MIX** knob to control the balance between the wet (processed) and dry (unprocessed) signal.

Setting the Warp Mode

Setting the warp allows you to manipulate the playback/sound of the sample.

By default, warp is set to **OFF** (as displayed next to the corresponding knob). Clicking the current setting displays a menu from which you can choose from among the available warp modes.

You can also use the **< >** arrows to conveniently switch between different warp modes without having to open the menu. After selecting a mode, you can use the knob to set the depth.

See “Exploring the Warp Modes” on page 49 for detailed information about the available warp modes.

Pan

Use the **PAN** knob to control the placement of the sample in the stereo field (left to right).

Level

Use the **LEVEL** knob to control the output volume of the oscillator.

Switching a Sample to a Wavetable

You can quickly and easily create a wavetable from a sample in Serum.

With a sample already loaded, click the **Spectral** menu and hover over **Switch to Wavetable** in the menu.

The menu of import options appears. These are the same options that appear when you import audio as a wavetable in other areas of Serum.

Choose one of the menu options.

The oscillator switches to **Wavetable** mode with the converted wavetable loaded.

See “Importing Multi-Cycle Waveforms” on page 291 for a detailed description of each option.

Switch to Wavetable Menu

Using the Sub Oscillator

Serum features a sub oscillator that generates a waveform pitched below the primary oscillators, adding depth and weight to the low end of a sound. You can select from a variety of simple waveforms, including sine, square, triangle, and more, providing a clean, stable foundation without introducing excessive harmonic complexity.

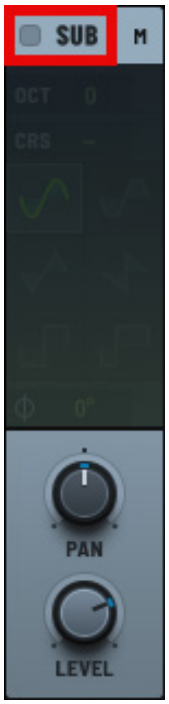
Using the sub oscillator enhances bass presence by reinforcing low frequencies, giving your sound more power and fullness. This is especially useful for creating bass-heavy sounds like sub-bass, deep bass for electronic music, or warm, rich pads.

The sub oscillator also helps create a thicker, more robust tone, making it ideal for leads, plucks, and pads, as it adds body without overwhelming the higher frequencies.

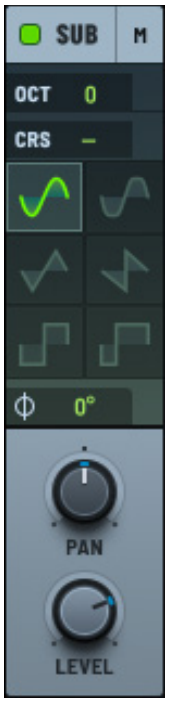
Sub Oscillator

Exploring the Sub Oscillator

To enable the sub oscillator, click the header (with the oscillator name and power button). When enabled, the button turns green.



Enabling the SUB Osc



SUB Osc Enabled

You can set several parameters related to the sub oscillator.

Pitch

Use the **OCT** (octave) and **CRS** (coarse) controls to alter the pitch of the waveform.

The **CRS** setting controls the pitch transpose that tunes or detunes continuous (no snap) semitones. **CRS** is most useful as a modulation destination or automation parameter.

Waveform

You can select the waveform to use as the basis of the sub oscillator. The following table describes the waveform options:

Icon	Waveform	Description
	Sine	A sine wave is a smooth, pure waveform that contains only the fundamental frequency, without any harmonics or overtones. It produces the cleanest and most basic form of sound, characterized by its rounded, steady oscillation. A sine wave is particularly effective at reinforcing the lowest frequencies without adding any additional complexity to the sound. This makes it ideal for creating deep, focused sub-bass, as it delivers a pure and powerful low end that sits well in a mix without clashing with other frequencies.
	Rounded Rect	A rounded rectangle wave is a variation of the square wave that has smoother transitions between its high and low points, as opposed to the sharp, immediate jumps of a traditional square wave. This results in a waveform that still has a pronounced pulse-like quality but with a less harsh, more rounded sound due to the gentler edges. A rounded rect wave produces a sound with some harmonic content but softer than that of a square wave, making it useful for adding more depth and texture to the low end without being too aggressive. It strikes a balance between the pure, smooth tone of a sine wave and the rich harmonic content of a square wave.
	Triangle	A triangle wave is a simple waveform that resembles a triangle in shape, with a linear rise and fall. It is similar to a sine wave in that it emphasizes the fundamental frequency, but it has a small amount of harmonic content, primarily odd harmonics, giving it a slightly richer sound than a sine wave, though still softer and smoother than a square wave. A triangle wave adds depth to the low end with a bit more texture and brightness than a sine wave, while still maintaining a relatively clean sound. It's a great choice for adding subtle harmonic detail to a bassline without overpowering other elements in the mix, making it effective in genres that require both warmth and clarity in the lower frequencies.

Icon	Waveform	Description
	Saw	A sawtooth wave is a waveform characterized by a sharp rise followed by an abrupt drop, containing both odd and even harmonics. This rich harmonic content makes it one of the most complex and versatile waveforms in synthesis. When used in a sub oscillator, a saw wave can add a much more aggressive and gritty texture to the low end. It introduces not only the fundamental bass frequency but also a wide range of harmonics that give the sound a bright and energetic character. This makes the saw wave useful for creating powerful, bold basslines that cut through a mix, especially in genres like dubstep, techno, and electro, where a more pronounced and dynamic low end is desired. However, because the saw wave is so harmonically rich, it can sometimes overpower the other elements in the mix if not carefully managed. In the sub oscillator role, it can be used to fatten up a sound, adding both depth and brightness, but it should be applied when you want your low-end frequencies to have a strong presence and a more complex tonal quality compared to simpler waveforms like sine or triangle waves.
	Square	A square wave is a waveform that alternates sharply between its high and low states, creating a distinct, pulse-like sound. It is rich in odd harmonics, which gives it a fuller, more aggressive tone compared to smoother waveforms like sine or triangle waves. A square wave can add a bold, punchy quality to the low end. Its strong harmonic presence makes it ideal for basslines that need to stand out or cut through a dense mix, especially in genres like electronic, rock, and synthwave. The square wave's harmonics provide additional texture and brightness, making the bass sound more present and dynamic compared to simpler waveforms. However, the sharp transitions of a square wave can sometimes produce a harsher sound in the low frequencies. This makes it a good choice when you want your bass to have an assertive, more aggressive edge, adding character and energy to the foundational elements of a track.
	Pulse	A pulse wave is a variation of the square wave where the high and low states are not equal, resulting in an asymmetrical waveform.

Phase

Use the **PHASE** control to specify where the oscillator should begin playing back when a note is triggered. This is similar to the phase control in other parts of Serum. It is also the same concept as sample start on a sampler (except the sub oscillator “sample” is a very small waveform).

Right-click the **PHASE** control and choose Contiguous in the menu that appears to have new notes continue with the phase of the previous note.

Pan

Use the **PAN** knob to control the placement of the waveform in the stereo field (left to right).

Level

Use the **LEVEL** knob to control the output volume of the sub oscillator.

Using the Noise Oscillator

Serum features a dedicated noise oscillator that offers a wide range of factory-supplied noise samples that you can use to add depth, texture, realism, and expressiveness to your sound design.

Since the noise oscillator is actually a stereo sample player featuring high-quality playback, you can also use it to load your own samples for even greater versatility.

For example, you could use the noise oscillator as a modulation source for many parameters (though the results can, admittedly, be chaotic). Samples like drum loops, for example, can lead to very interesting results.

The noise oscillator also appears in the **WARP** section of all three wavetable oscillators, allowing you to apply frequency, phase, amplitude, and ring modulation (using the sampler as the modulator).

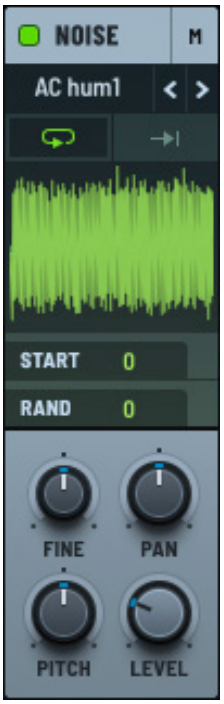
Noise Oscillator

When using the noise oscillator as a modulator, you should turn down the volume of the noise oscillator (to just take advantage of the modulation effect).

Exploring the Noise Oscillator

To enable the noise oscillator, click the header (with the oscillator name and power button). When enabled, the button turns green.

Enabling the Noise Osc



Noise Osc Enabled

Loading a Preset

To load a noise preset, click the current sample name and choose an option in the menu that appears.

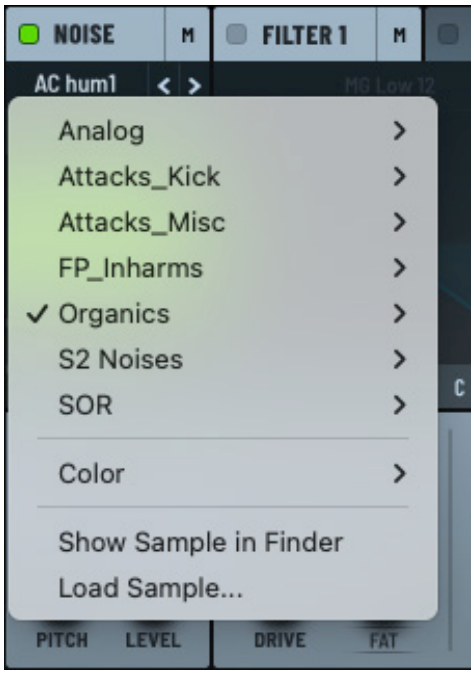
After choosing a preset, you can use the **< >** arrows to conveniently switch between noise samples without having to open the menu.

Note that there is a separate category for noises named after colors.

If you choose one of the noise color modes, a **STEREO** control becomes available.

At a setting of 0, the noise is mono; at 100, there is no correlation between left and right signals.

In addition, a **FILTER** knob is available to high or low pass the signal offering you further control.



Noise Menu

The following table offers a brief description of the difference between these different noises:

Noise Color	Description
White	White noise has equal energy at all frequencies, meaning it covers the entire audible spectrum (20 Hz to 20 kHz) with equal intensity. It sounds like a constant hiss or the static between radio stations. Because of its equal distribution across frequencies, white noise has a bright, harsh sound.
Pink	Pink noise has equal energy per octave, which means the energy decreases as the frequency increases. Specifically, it has 3 dB less energy per octave as the frequency doubles. It has a warmer, less harsh sound compared to white noise, with more emphasis on lower frequencies.
Brown	Brown noise, also called Brownian noise or red noise, has energy that decreases even more rapidly than pink noise, at 6 dB per octave. It has a deep, rumbling sound, with much more emphasis on lower frequencies and very little high-frequency content.
Geiger	Geiger noise doesn't have a uniform frequency spectrum like the others. Instead, it is chaotic, consisting of random clicks or bursts that are not evenly distributed across time or frequency. It sounds like the clicks of a Geiger counter, hence the name. The noise is made up of sporadic, random bursts with unpredictable intervals between them.

Loading a Sample

Noise oscillator sounds are simply mono or stereo WAV files. This means that you can use your own samples as noise sounds.

To load your own sample in the noise oscillator, click the current sample name to display the **NOISE** menu and choose **Load Sample**. A dialog appears allowing you to locate the sample on your computer.

Alternatively, drag a sample from the Finder (macOS) or Explorer (Windows) to the oscillator waveform pane.

Embedding the Sample in the Preset

You can embed a noise sample into your preset (when saving the preset) by displaying the **NOISE** menu and choosing **Embed in Preset**.

This allows you to streamline your presets by packaging your noise samples together with the Serum settings.

Noise oscillator playback uses high-quality real time interpolation. This is because with noise sounds, the high-frequencies are important. Serum strives to offer the best quality whenever possible, even at the expense of additional CPU usage (although this feature is heavily optimized with both SSE2 and pre-calculations when a file loads).

Since high-quality playback does require more CPU processing, it is worth noting that mono sounds do use slightly less CPU resources (one channel instead of two can add up with chords). The main consideration here is that if you have a mono noise source, don't export it as a stereo file for Serum since you will be wasting both disk space and CPU resources (on playback).

One Shot/Looping

By default, the noise oscillator is set to loop samples.

This is indicated by the highlighted looping button. 

Select the  (one shot) switch to have the sample (noise) stop when playback reaches the end of the sound file.

The button highlights to show that it's enabled. 

Looping is typically good with noise sounds, but one-shot mode is useful for attack sounds (to add a percussive punch/attack transient to a sound, for instance).

Start

Use the **START** control to set the phase start (as a percentage). Click and drag in the field to set the value. Alternatively, double-click the field and type the appropriate value.

This setting for the noise oscillator is similar to the phase controls in other parts of Serum. However, since a noise sound is a lot longer than an oscillator waveform, it might be easier to think of this as "sample start."

 You can also automate the control, which results in a sort of lo-fi "scratching" effect.

Random

Use the **RAND** control to randomize the start phase. This prevents the noise from being identical each time you press a note.

Holding a chord is a good example of when you would likely want to add randomness, otherwise the same noise sounds for all the notes.

Pitch

Use the **PITCH** knob to specify the base pitch/frequency for the noise oscillator. The default (50%) is nominal pitch, that is, the original pitch of the sound file (assuming that the keytrack switch is off).

Fine

Use the **FINE** knob to fine tune the pitch setting.

Pan

Use the **PAN** knob to control the placement of the noise in the stereo field (left to right).

Level

Use the **LEVEL** knob to control the output volume of the noise oscillator.

Using the Filter Modules

Serum features an advanced filter module that offers per-voice filtering of one or more oscillators. Filters are one of the most powerful tools for shaping and sculpting sound, allowing you to modify the frequency content of a sound, selectively emphasizing or reducing certain frequencies.

By controlling which parts of the sound's spectrum are allowed to pass through and which are cut off, you can use Serum's filters to drastically alter a sound's character, making it warmer, brighter, darker, or even more aggressive.

Serum includes an extensive collection of filters, featuring numerous variations on low-pass, high-pass, band-pass, notch, comb, and more. Combined with Serum's extensive modulation capabilities, you can use the filters to create a range of sounds including rhythmic sweeps, evolving pads, and even vocal-like formant sounds.

With playful tweaking and clever modulation, you can use the filter module as a creative playground, exploring endless possibilities for sound manipulation and transformation.

Serum Filter Modules

Exploring the Filter Modules

To enable or disable a filter module, click the corresponding power button. This offers you an easy way to disable a filter entirely, if needed.

Enabling Filter 1 and 2

Filter 1 and 2 Enabled

Routing an Oscillator

To route an oscillator to a specific filter module, select the corresponding switch:

- S - SUB oscillator
- A - OSC A
- B - OSC B
- C - OSC C
- N - NOISE oscillator

Filter Routing

Filter Type

To choose a filter type, click the current filter setting. A pop-up menu appears showing a hierarchical collection of available filter types.

Use the **< >** arrows to conveniently switch between filter types without having to open the menu.

Alternatively, hover over the menu and use the mouse wheel to quickly rotate through menu options.

Filter Menu

Filter Types and Var Parameter Functions

The following table describes the available filter types:

Category	Filter Type	Description	Var Function
Normal	MG Low 6/12/18/24	Ladder Low-Pass Filter. The number represents the db per octave slope of the filter. The topology is based on filters made famous in classic Moog synthesizer designs.	**FAT:** Add saturation to the filter resonance signal path. This has a taming effect on filter resonance and enriches the resulting harmonic content.
	Low 6/12/18/24	State-Variable Low-Pass Filter. The number represents the db per octave slope of the filter.	**FAT:** Add saturation to the filter resonance signal path. This has a taming effect on filter resonance and enriches the resulting harmonic content.

Category	Filter Type	Description	Var Function
Normal (cont.)	High 6/12/18/24	State-Variable High-Pass Filter. The number represents the db per octave slope of the filter.	**FAT:** Add saturation to the filter resonance signal path. This has a taming effect on filter resonance and enriches the resulting harmonic content.
	Band/Peak/Notch 12/24	State-Variable Band/Peak/Notch Filter.	**FAT:** Add saturation to the filter resonance signal path. This has a taming effect on filter resonance and enriches the resulting harmonic content.
Multi	LH/LB/LP/LN/ HB/HP/HN/BP/ BN/PP/PN/NN	Dual SVF Filters. The first letter is primary, the second letter is secondary (for example, BP is Band+Peak). The resonance control applies equally to both filters.	**FREQ:** Set the cutoff frequency of the second SVF filter.
	LBH/LPH/LNH/ BPN	Morphing SVF filters (for example, Lowpass <-> Bandpass <-> Highpass)	**MORPH:** Smoothly transition between the filter states.
Flanges	Cmb L/Flg L/Phs L	Comb/Flanger/Phaser with a low-pass filter in the internal feedback circuit. Set the **MIX** knob to 50% for best results.	**LP FREQ:** Set the cutoff frequency of the low pass filter affecting the internal feedback circuit.
	Cmb H/Flg H/Phs H	Comb/Flanger/Phaser with a high-pass filter in the internal feedback circuit. Set the **MIX** knob to 50% for best results.	**HP FREQ:** Set the cutoff frequency of the high-pass filter affecting the internal feedback circuit.
	Cmb HL/Flg HL/ Phs HL	Comb/Flanger/Phaser with a high pass + low pass filter in the internal feedback circuit. Set the **MIX** knob to 50% for best results.	**HL WID:** Expand the bandwidth allowed through the internal feedback circuit around the filter cutoff frequency.

Category	Filter Type	Description	Var Function
Misc	Low/Band/High EQ 6/12	Filters with morphable frequency responses. The number represents the db per octave slope of the filter. Resonance has no effect on 6 db per octave variations.	**DB +/-:** Increase or decrease gain of the pass band. Extreme settings morph the frequency response of the filter, for example, the Low EQ 12 filter blends from a high pass (when db gain is set to 0) to a low shelf.
	Ring Mod/Ring Modx2	Apply ring modulation to the input signal at a frequency set by the cutoff control. The x2 variant features a second ring modulation.	**SPREAD** (only available in the x2 filter variant): Control the distance between the first and second ring modulator frequencies.
	SampHold/ SampHold-	Apply a sample-and-hold distortion. The minus variant outputs the difference of the sample-and-hold distortion and the input signal.	**N/A**
	Combs/Allpasses/ Reverb	Generate phase smearing through combinations of delays and all-pass filters.	**DAMP:** Soften the feedback path of the filter.
	French LP	Unique distorting low-pass filter. The filter responds non-linearly to input signals.	**BOEUF:** A secondary resonance control. Combinations of the primary and Boeuf resonance values produce unique results.
	German LP	“Zero-Delay Feedback” low-pass filter.	**N/A**
	Add Bass	Phase-rotated low pass filter with a touch of drive. Not a typical synth filter, but maybe you’ll find a use!	**THRU:** Add a phase-rotated dry signal.

Category	Filter Type	Description	Var Function
Misc (cont.)	Formant-I/II/III	Formant 'vowel' filters. The **CUTOFF** knob morphs between various formants. This is great for adding vocal-like characteristics to patches.	**FORMNT:** Shift the formants to generate a broader range of possible filter permutations.
	Bandreject	Attenuate a specific range to very low levels. It has the opposite effect of a band-pass filter.	**WIDTH:** Adjust the overall width of the filter's attenuated "notch".
	Dist.Comb 1/2 LP/BP	Combination comb filter and pass filter. The comb filter (version 1 is a positive feedback comb filter; version 2 is a negative feedback comb filter) is applied to the feedback path of the pass filter.	**COMBFRQ:** Set the frequency of the comb filter.
	Scream LP/BP	A high-feedback filter effect that can have a scream-like quality.	**SCREAM:** Set the cutoff frequency for the feedback circuit. The **DRIVE** knob affects the amount of scream (raise **DRIVE** above 50% to hear the scream).
New	Wsp	Circuit model of a classic synth filter that buzzes (and burbles!).	**MORPH:** Blend between LPF/Notch/HPF.
	DJ Mixer	Xfer freeware DJM filter plugin.	**N/A**

Category	Filter Type	Description	Var Function
New (cont.)	Diffusor	All-pass diffusor stage for making things sound phasey and blurry in a cool way.	**STAGES:** Specify how many APF stages are used.
	MG Ladder	Clean circuit model of classic transistor ladder VCF.	**SMOOTH:** Slow the rate at which the cutoff is modulated.
	Acid Ladder	Circuit model of a diode ladder VCF ubiquitous in acid music.	**SMOOTH:** Slow the rate at which the cutoff is modulated.
	EMS Ladder	Clean circuit model of the VCF that makes Dr. Who sounds.	**SMOOTH:** Slow the rate at which the cutoff is modulated.
	MG Dirty	MG Ladder, but all the distortion is there and you are overdriving the circuit.	**PAIN:** How far you're holding a lighter from the circuit board (this is physically correct; not a joke).
	PZ SVF	Drawable filters.	**SMOOTH:** Slow the rate at which the cutoff is modulated.
	Comb 2	Comb filter with crazy resonance and stuff.	**FRQ2**
	Exp MM	Multimode (LPF/Notch/HPF) output from the classic synthesizer expander module.	**MIX:** blends between LPF/Notch/HPF.
	Exp BPF	BPF output from the classic synthesizer expander module.	**N/A**

Filter Display Options

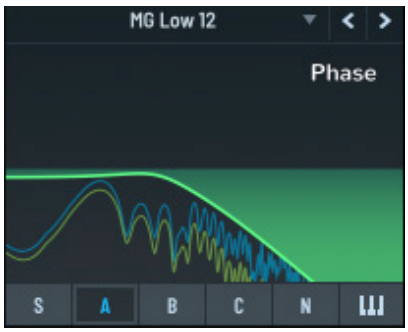
You can set display options for the filter to show frequency, FFT, and phase information, as needed.

Right-click the filter display and choose an option using the context menu.

Filter Display Menu

The following table describes the available display options:

Option	Display	Description
Frequency Response		Shows how the filter affects the amplitude of different frequencies in the audio spectrum, visualized as a graph with frequency on the horizontal axis and amplitude (gain or attenuation) on the vertical axis. Use this view to understand which frequencies are being boosted, attenuated, or left unaffected by the filter.
Frequency Response & FFT		Combines the filter frequency response curve with a real-time FFT (Fast Fourier Transform) analysis of the audio input. The FFT displays the actual frequency content of the signal, overlaid with the filter's effect. Use this view to show a dynamic visualization of both the audio signal spectrum and how the filter is shaping it in real time, allowing for more precise adjustments.

Option	Display	Description
Phase Response & FFT		Shows how the filter affects the phase of different frequencies, indicating the degree of phase shift applied to each frequency in the signal, alongside the real-time FFT of the audio. Use this view to understand phase-altering filters and how they might impact phase-sensitive tasks like stereo imaging or when combining multiple signals.

You can quickly cycle through the different display modes by Option-clicking (macOS) or Alt-clicking (Windows) in the filter display.

Setting Filter Parameters

You can set a range of filter parameters, including the cutoff, resonance, drive, pan, mix, and level. In addition, different filter types offer a variable parameter depending on context.

Cutoff

Use the **CUTOFF** knob to set the primary cutoff frequency for the filter (with just a couple exceptions, such as vowels for formant filters).

Use the  (keytrack) switch to offset the cutoff using MIDI notes. With most filter types, one octave of MIDI corresponds to precisely one octave of filter frequency control.

If the keytrack switch is enabled, this tracks the pitch of the first oscillator (**OSC A**, **OSC B**, or **OSC C**) that has pitch tracking enabled (including portamento).

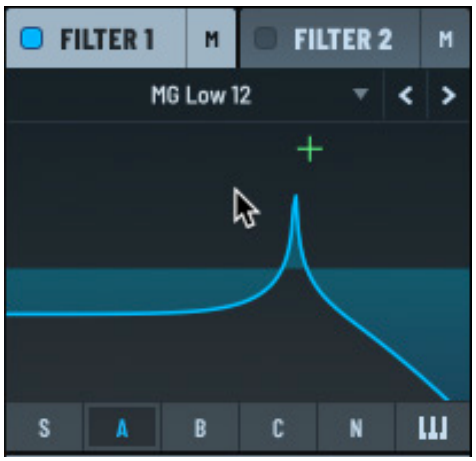
If none of the oscillators have pitch tracking enabled, the filter tracks the input MIDI note number.

Resonance

Use the **RES** knob to set the resonance (feedback) of the filter circuit.

You can graphically adjust the filter cutoff and resonance (in combination) by clicking and dragging in the filter display.

This is a quick way to experiment with filter settings in your sound design.



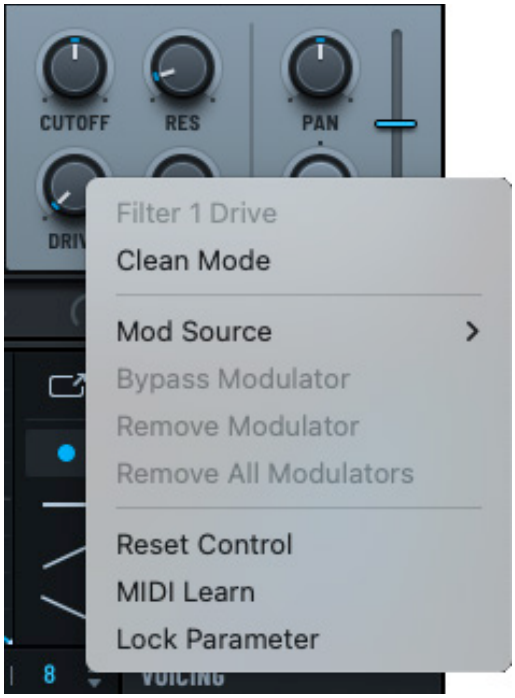
Graphical Adjustments

Drive

Use the **DRIVE** knob to increase the gain into the filter circuit and can impart some coloration (mild distortion) to the sound.

Right-click the knob and choose **Clean Mode** in the context menu to have the filter pre-gain stage the filter input -24 dB (with a +24 dB boost post-filter).

This reduces saturation and input drive in the filter models.



Drive Clean Mode

Fat (and Others)

By default, the **FAT** knob appears when the filter type is **MG Low 6** (as part of the **Init** preset, for instance). However, this knob changes based on the filter type you select using the menu.

Refer to the table above for details about the knob that appears here.

Pan

Use the **PAN** knob to create a cutoff offset for the left and right signals. At the default setting of 50% (12 o'clock) this knob has no effect. When turned to the left (counter-clockwise) the left channel cutoff increases, and the right channel cutoff decreases.

When turned to the right (past 12 o'clock clockwise), the opposite happens; the left cutoff decreases and the right channel cutoff increases.

Mix

Use the **MIX** knob to control the wet/dry amount for the filter. The default (100%) means 100% wet.

Level

Use the **LEVEL** slider to adjust the filter output level (in decibels).

Using the Mixer

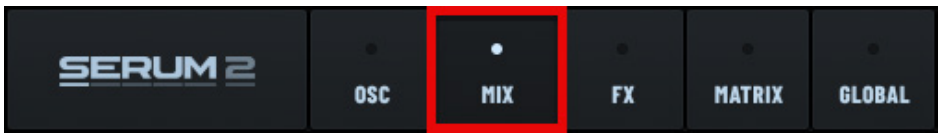
Serum features an advanced mixer that you can use to blend and balance the principal sound sources in your patch, including the sub oscillator, the three main oscillators (**OSC A**, **B**, and **C**), and the noise oscillator. In addition, you can include the output from **FILTER 1** and **2** in the mix, together with the output from two internal Serum busses.

In short, the mixer provides you with precise control over the individual elements of your sound. Using the Serum mixer, you can create complex, layered sounds with ease, adjusting the volume, pan, and routing of each component to achieve the perfect sound.

Serum Mixer (MIX)

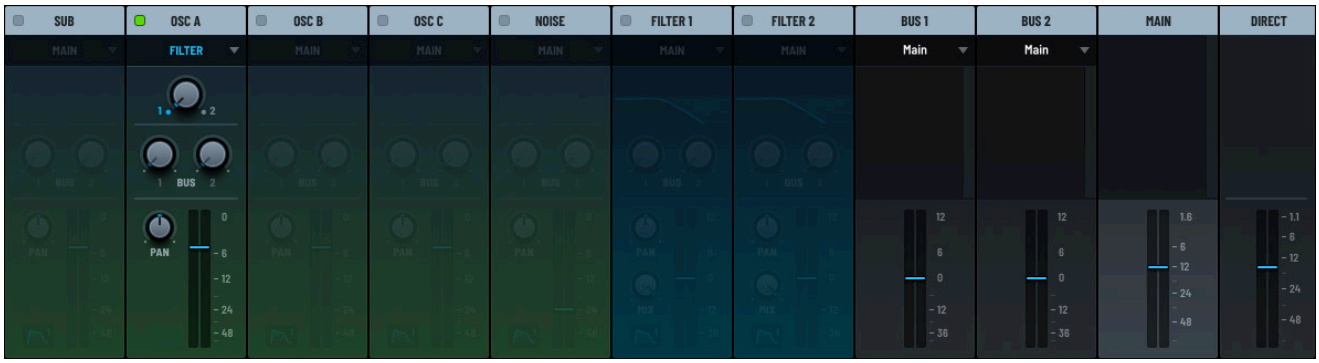
Exploring the Mixer

Click the **MIX** tab to access the Serum mixer.



Accessing the Serum Mixer (MIX)

The mixer shows the elements of Serum that you have enabled.



Mixer for the Init Patch

For the **Init** patch, for instance, only **OSC A** is enabled and, therefore, this is the only channel enabled in the mixer. Click the header to enable a channel.

Note that this enables the oscillator or filter throughout Serum.

Oscillator Header

Each oscillator channel offers the same controls.

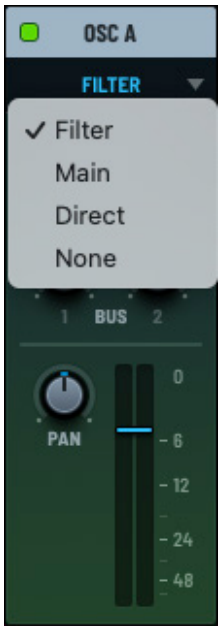
Use the channel menu to specify the signal routing.

For example, in the illustration below, the output of **OSC A** is currently being routed to the filter module.

However, you could choose to route the signal to an alternative destination.

Note that, by default, **OSC A** routes through **FILTER 1**, while the **OSC B**, **OSC C**, **SUB**, and **NOISE** oscillators route through the **MAIN** output by default.

Oscillator Channel



Oscillator Channel Menu

Hold the Option/Alt key and drag the oscillator channel label to another oscillator label to copy the oscillator without modulations.

Hold the Shift-Option or Shift-Alt keys and drag the oscillator channel label to another oscillator label to copy with modulations.

Dragging from one oscillator channel label to another without any keyboard modifiers swaps the two oscillators (including modulation assignments).

The following table describes the signal routing options:

Option	Description
Filter	Route the signal to the filter module. Use the top knob to set how much of the signal is sent to **FILTER 1** and **FILTER 2**. 
Main	Route the signal to the main output. When you choose this option, the envelope button  appears near the bottom left of the channel. When enabled, the oscillator output level is affected by **ENV 1**. When disabled, **ENV 1** has no effect on the oscillator output level and it terminates after the longest release of any of the envelopes. You can use this to make oscillators appear closer to being free running.
Direct	Route the signal to direct output, bypassing the filter and effects sections, outputting "clean" along with the main output. Similar to **MAIN**, when you choose this option, the envelope button  appears near the bottom left of the channel.
None	Disable the output of the source (the oscillator, for example).

Sending to the Busses

You can send oscillator signals to **BUS 1**, **BUS 2**, or both.

To send a signal to either bus, click and drag the corresponding **BUS** knob. To set a specific value, double-click the knob and type the appropriate value in the field that appears.

Sending to the Busses

Sending a signal through auxiliary effects busses offers multiple advantages, including the following:

- Consistent sound and cohesion

  Using a shared bus for effects such as reverb or delay can create a more cohesive sound across the mix. For example, sending multiple tracks to a single reverb bus can make them sound as if they're in the same space, creating a sense of depth and unity.

- More precise control

  Auxiliary busses allow you to control the wet/dry balance more precisely, as you can adjust the send levels from each track to the effect bus. This flexibility makes it easier to blend the effect subtly or aggressively, depending on the needs of the mix.

- Parallel processing

  Aux busses enable parallel processing, where the dry signal and the processed signal run simultaneously. This is useful for effects like parallel compression, where you maintain the original dynamics while adding enhanced processing to the signal.

- CPU efficiency

  Instead of applying the same effect to multiple oscillators individually, you can send them to a single auxiliary bus. This reduces the CPU load since only one instance of the effect is running.

Setting Pan and Levels

You can set the pan and levels for each oscillator as part of creating a balanced and clear audio experience.

Click and drag the **PAN** knob left or right, as appropriate. To set a specific value, double-click the knob and type the appropriate value in the field that appears.

Use negative values for left and positive values for right.

Similarly, click and drag the level up or down, as needed. To set a specific value, double-click the control and type the appropriate value in the field that appears.

Setting Pan and Levels

You can use panning to effect the following:

- Create a stereo image — Panning places sound elements across the stereo field (left to right), giving the mix width and spatial dimension.

  By positioning elements in different parts of the stereo field, you create a more realistic and immersive sound, emulating how we naturally hear sounds around us.

- Prevent clashing — Panning helps separate elements that occupy similar frequency ranges.

  For example, if you have two midrange-heavy sound elements, panning them apart reduces the likelihood of them competing for space and allows each to be heard more clearly.

- Add depth and realism — Proper use of panning mimics the natural placement of sound elements in a room or on stage. It helps give the listener a sense of depth and positioning, making the mix feel more dynamic and engaging.

You can set the levels to control the following:

- Ensure balance — Adjusting levels is essential for achieving a clear and balanced mix.

  Setting the volume of each element ensures that the most important parts are prominent while supporting elements are appropriately audible without overpowering the mix.

- Maintain dynamics — Setting levels effectively preserves the dynamic relationships between sound elements.

  This allows you to better manage contrast between elements.

- Controls focus — Levels can guide listener focuses.

  By raising or lowering the volume of certain elements, you can guide the listener's attention and shape the overall feel of the sound, making certain parts stand out more prominently while others remain in the background.

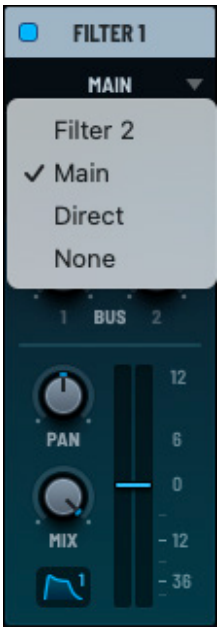
Mixing Filters

You can control how the filter output integrates into the overall mix.

Important: Even after enabling a filter (by clicking in the corresponding header), the filter appears dimmed until you send an oscillator signal to the filter.

By default, **FILTER 1** and **FILTER 2** send through the **MAIN** output.

Use the channel menu to specify the signal routing.

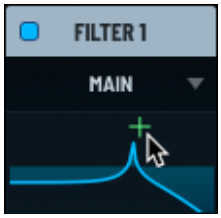


Filter Channel Menu

Filter Channel Menu

Similar to the filter module, you can graphically adjust the filter cutoff and resonance (in combination) by clicking and dragging in the filter display.

This is a quick way to experiment with filter settings while mixing!



Filter Adjustments

The following table describes the signal routing options from the filters:

Option	Description
Filter 1 or Filter 2	Route the signal to the other filter, either **FILTER 1** or **FILTER 2**.
Main	Route the signal to the main output. When you choose this option, the envelope button  appears near the bottom left of the channel. When enabled, the oscillator output level is affected by **ENV 1**. When disabled, **ENV 1** has no effect on the oscillator output level and it terminates after the longest release of any of the envelopes. You can use this to make filters appear closer to being free running.
Direct	Route the signal to direct output. Similar to **MAIN**, when you choose this option, the envelope button  appears near the bottom left of the channel.
None	Disable the output of the source (the filter, for example).

Routing to the Busses

You can route filter signals to **BUS 1**, **BUS 2**, or both.

To route a signal to either bus, click and drag the corresponding **BUS** knob. To set a specific value, double-click the knob and type the appropriate value in the field that appears.

Routing to the Busses

Setting Pan, Mix and Levels

You can set the pan, mix, and levels for each filter. Click and drag the **PAN** knob left or right, as appropriate.

To set a specific value, double-click the knob and type the appropriate value in the field that appears. Use negative values for left and positive values for right.

Similarly, click and drag the **MIX** knob to set the dry/wet blend of the filter mix.

For most filters, the recommended mix setting is 100% (all the way to the right).

Note: The **MIX** knob has no effect for Combs-type filters.

Setting Pan, MIX and Levels

Finally, click and drag the level up or down, as needed. To set a specific value, double-click the control and type the appropriate value in the field that appears.

Mixing the Busses

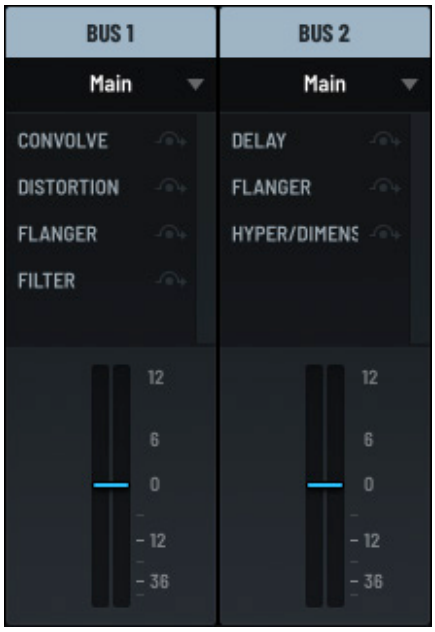
You can set the routing and overall levels for the FX busses.

By default, **BUS 1** and **BUS 2** route through the MAIN output.

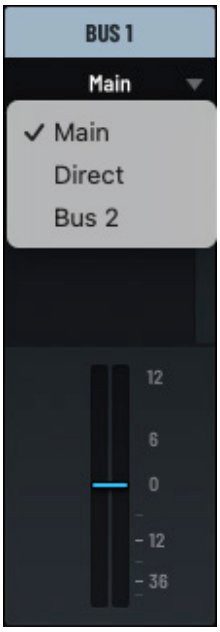
Use the channel menu to specify the signal routing.

Click the  button to bypass the corresponding FX module on the bus.

When enabled, the button highlights in red to show that the FX module is being bypassed. 



Mixing Busses



Filter Channel Menu

The following table describes the signal routing options from the busses:

Option	Description
Main	Route the signal to the main output.
Direct	Route the signal to direct output.
Bus 1 or Bus 2	Route the signal to the other bus, either **BUS 1** or **BUS 2**.

Setting Levels

You can set the level for each bus.

Click and drag the level up or down, as appropriate.

To set a specific value, double-click the control and type the appropriate value in the field that appears.

Setting Levels

Setting the Main and Direct Levels

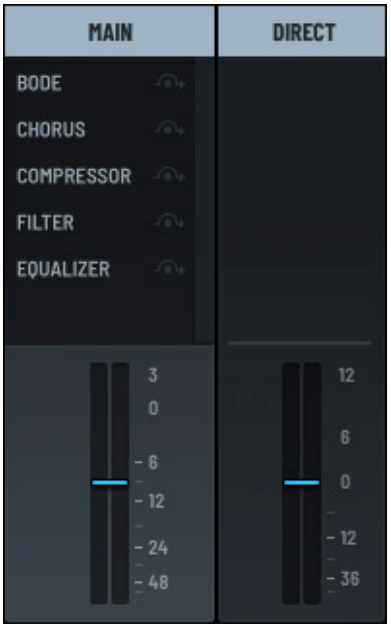
You can set the **MAIN** and **DIRECT** levels for the mix.

Click and drag the levels up or down, as appropriate.

To set a specific value, double-click the control and type the appropriate value in the field that appears.

Click the  button to bypass the corresponding FX module on the main channel.

When enabled, the button highlights in red to show that the FX module is being bypassed. 



Setting Main and Direct Levels

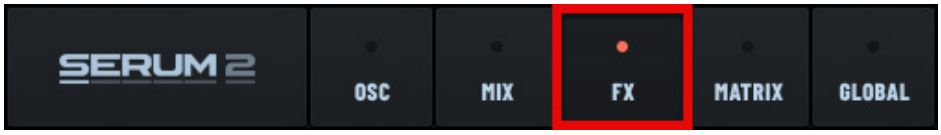
Using Serum FX

Serum features an effects section with 13 different FX processors that you can use in any order or combination, including multiple instances of the same processor. There are also three types of splitter modules that allow FX processing to be applied to a particular part of the signal.

Serum Effects (FX)

Using the FX Module

Click the **FX** tab to access the effects module.



Accessing Serum FX (Effects)

An empty rack appears that you can populate with any of the 13 effects modules, in any order.

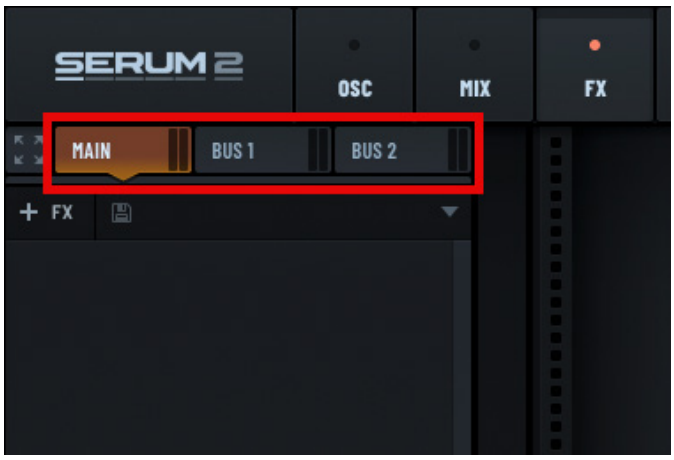
Click the  button (near the top left) to expand the FX rack and list view. Alternatively, press Option-F (macOS)/Alt-F (Windows) to expand the view.

This provides more rack space to display modules without scrolling.

FX Rack and List Views (Expanded)

Click the  button to revert the FX rack (and list view) back to its original size. Similarly, press Option-F (macOS)/Alt-F (Windows) to revert the view to the original size.

Selecting a Rack



Serum FX Racks

Serum offers three FX racks: **MAIN**, **BUS 1**, and **BUS 2**. Each FX rack processes the audio signal on the corresponding channel.

Click one of the tabs to select the corresponding FX rack.

Loading Rack Presets

A quick way to get started is to load rack presets, available using the presets drop-down menu.

The preset populates the rack (including the list view on the left).

At any time, you can initialize an FX rack by choosing **Init** as the factory preset.

FX Presets Menu

The following shows an example FX preset:

Acid Dist Delay Rack Preset

Adding Modules

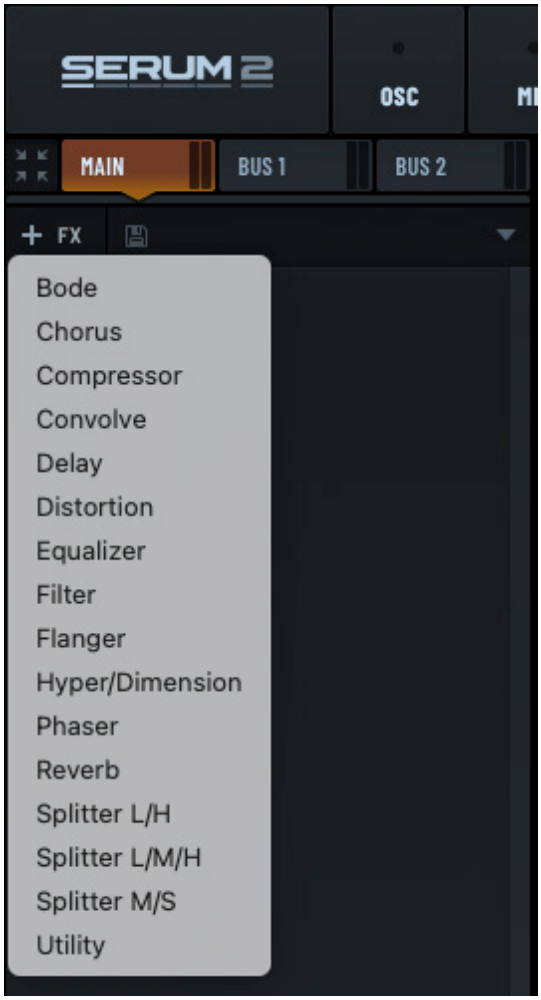
Click the + FX button and choose a module in the list that appears.

Alternatively, you can add a module to the rack by right-clicking in the rack, choosing **Add FX Module**, and then choosing the FX module in the menu that appears.

The module appears in the rack.

After you place an FX module in the rack, all audio routed to **MAIN** passes through the module (and then to the master volume and output).

Note that the signal flow is top to bottom through the rack.



Adding an FX Module

Module Added to the Rack

Reordering Modules

To reorder effects, click and drag an effect to the new location in the list view (on the left) or in the rack view (on the right). A yellow line indicates where the module will land.

Reordering Modules

Copying a Module

You can copy an FX module to create a duplicate of the module on a rack, with or without assigned modulations.

To copy a module without modulations, Option-drag (macOS) or Alt-drag (Windows) an existing module to the appropriate location on the rack. A yellow line indicates where the module will land, which is helpful when placing a module between two existing modules.

Copying a Module

To copy a module with assigned modulations, Shift-Option-drag (macOS) or Shift-Alt-drag (Windows) an existing module to an empty location on the rack.

Bypassing a Module

Every FX module features a bypass button  that allows you to easily bypass the module. This button appears to the right of the module both in the list and rack view.

Clicking the button bypasses the module in the signal routing. When enabled, the button highlights in red  to show that the module is being bypassed.

Bypass Effect Button

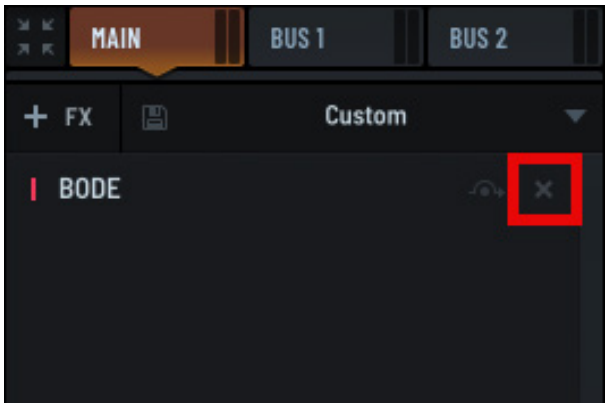
Bypassing a module is really intended for temporary use. For example, you can use bypass to hear your sound with and without a given FX (without having to set the **MIX** knob to 100% dry).

Option-clicking (macOS) or Alt-clicking (Windows) on a bypass button toggles bypass for all FX on the bus. This also works with the FX bypass buttons on the **MIXER** page.

Removing a Module

To deactivate an effect, you should generally remove the module from the rack (you can always add it back later).

Click the  button for the corresponding module in the list view (on the left).



Remove an FX Module

Saving a Rack as a Preset

After creating a new custom rack, or modifying an existing rack, you can save the rack as a new preset

Click the  button. A dialog appears allowing you to type the rack preset name. By default, the preset is saved in a standard user location so that Serum can easily find it later.

Exploring FX Rack Operations

Serum makes it easy to manage your FX racks by offering a series of operations that you can quickly access by right-clicking in the background of the rack.

The following table describes the operations you can perform:

Operation	Description
<Select a module>	Add a module to the rack.
Cut FX Bus	Cut (remove) all modules from the current rack. You can then paste the modules into another rack. Use this when you want to move all modules from one rack (such as **MAIN**) to another rack (such as **BUS 1**).
Copy FX Bus	Copy all modules from the current rack. You can then paste the modules into another rack. Use this when you want to duplicate all modules in one rack (such as **MAIN**) in another rack (such as **BUS 1**).
Paste FX Bus	Paste the contents of the rack clipboard (after a cut or copy operation) to the currently-selected rack.
Clear FX Bus	Clear the currently-selected rack. This removes all modules from the rack.
Lock FX Bus	Enable this option to have the modules in the selected rack remain in place when changing presets (no modules will be loaded to that rack). Note that modulation assignments to module parameters in the locked rack are cleared when changing presets.
Lock All FX Busses	As the name implies, locks all FX busses (in the manner described above).
Load FX Bus	Load a user preset rack. Serum displays a dialog allowing you to choose the appropriate rack.
Save FX Bus	Save the current rack as a user preset.

Modulating FX Parameters

You can modulate most FX parameters, similar to the way you can modulate standard Serum synth controls.

Click and drag an envelope (**ENV**) or **LFO** to the modulation destination.

Assigning to Modulation to an FX Module

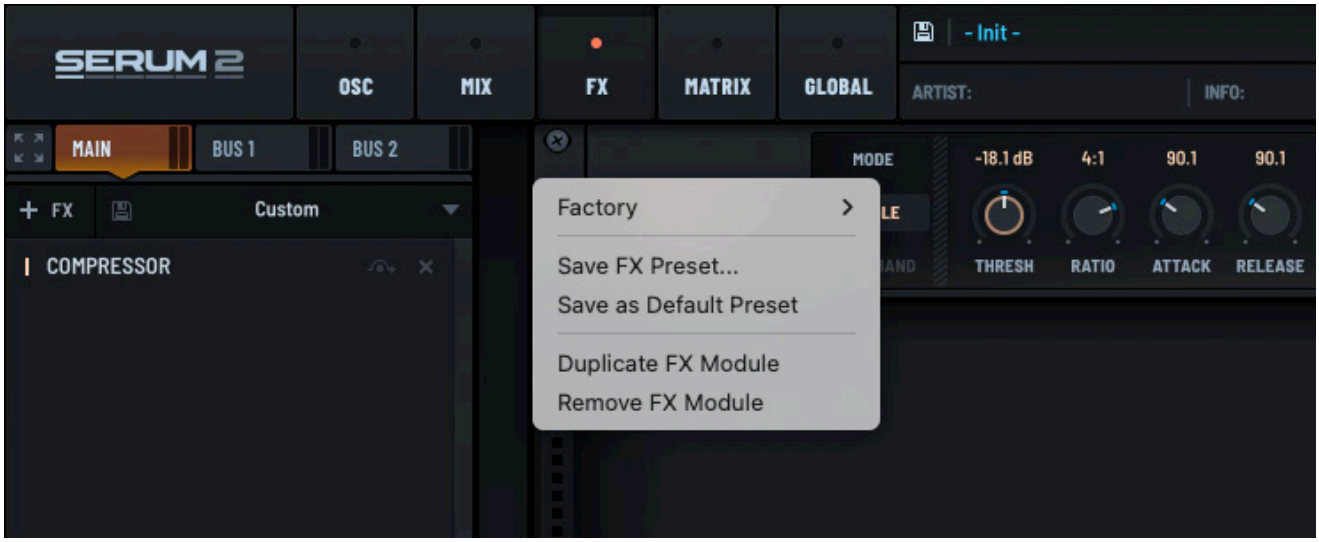
The FX rack is a DSP process that operates on the sum output of the synth engine (rather than per voice). You can think of this as “the effects are monophonic” or “the effects are like plug-in inserts after Serum.” This is sometimes referred to as paraphonic behavior, especially in conjunction with the filter effect.

Therefore when playing polyphonic synth parts (strummed chords, for example) keep in mind that automating FX controls with per-voice mod sources (such as an envelope) results in the effect parameter modulation being modulated/retriggered by each new note.

Exploring FX Module Operations

When working with individual modules in an FX rack, you can load presets, save your current module settings as a preset, as well as duplicate or remove a module.

Click the  button to display the module menu. Note that the color of the button varies depending on the color of the module. A drop-down menu appears.



Module Menu

The following table describes the operations you can perform:

Operation	Description
Factory	Load a factory preset for the FX module.
Save FX Preset	Save the module configuration as a user preset.
Save as Default Preset	Set the current module configuration as the default preset when adding the same type of module to an FX rack.
Duplicate FX Module	Duplicate the module on the rack (with the current module settings).
Remove FX Module	Remove the module from the rack.

Exploring Individual FX Modules

This section describes the controls available of each FX module.

Bode

The **BODE** module offers an implementation of the Bode frequency shifter, a device that shifts the frequency of an audio signal by a fixed amount, resulting in a unique sound effect.

Named after electronic music and audio signal processing pioneer Harald Bode, this frequency shifting can create dissonance, phasing effects, or a sense of movement in sound, which you can creatively apply to your sound design.

Bode Module

The **BODE** module offers the following controls:

Control	Description
MONO INPUT	Enable to route mono input to the module.
SHIFT	The percentage of the range to which to apply the pitch shift. Right-click the knob and choose **Retrig** in the context menu to have the module restart the effect for each new note.
RANGE	The range of the bode shift.
DIR	The direction of the bode shift. Setting to the center causes both channels to go in opposite directions.
WIDTH	Used in conjunction with the **DIR** knob, **WIDTH** specifies whether both Bode channels (up and down) are used, or if only a single channel is used.
DELAY	The delay time.
BPM	Specifies whether the delay time is synced to the BPM or measured in Hertz (Hz).
FEED	The amount of delay fed back into the Bode shifter, which can produce pitched delays.
BALANCE	The delay input mix between a down and up shifted signal.
BLUR	Use to create chorus and wow/flutter effects.
MIX	The wet/dry amount for the effect, from 0 (100% dry) to 100 (100% wet).
LEVEL	The output level of the module (in decibels).

Chorus

The **CHORUS** module offers a four-voice chorus effect, with two left and two right chorus taps.

Chorus Module

The **CHORUS** module offers the following controls:

Control	Description
RATE	The rate of the chorus. The units depend on the **BPM** setting. When **BPM** is on, the **RATE** knob snaps to musical time (from 8 bars to 1/32nd note). When **BPM** is off, the **RATE** knob is in Hertz (Hz), between 0 Hz and 20 Hz.
BPM	Specify whether the chorus is synced to the host BPM. See the **RATE** knob (above) for more information.
DELAY 1	The amount of delay (in milliseconds) between the dry signal and the first stereo pair of chorus voices.
DELAY 2	The amount of delay (in milliseconds) between the dry signal and the second pair of chorus voices.
DEPTH	Specifies how much the chorus LFO modulates the delay times described above (how much pitch warble occurs).
FEEDBACK	The feedback amount of the chorus voices (how much of the chorus voice output appears back at the input of the chorus module). This creates a more pronounced “ringing” to the chorus.
LPF/HPF	The cutoff frequency in Hertz (Hz) of the low pass/high pass filter after the chorus wet effect. This is useful for a more (or less) “warm doubling” of the signal. Click the label to toggle between LPF and HPF.
MIX	The wet/dry amount for the effect, from 0 (100% dry) to 100 (100% wet).
LEVEL	The output level of the module (in decibels).

Compressor

The **COMPRESSOR** module reduces the volume of loud sounds or amplifies quiet sounds, thereby reducing or compressing the dynamic range of an audio signal.

Compressor Module

The **COMPRESSOR** module offers the following controls:

Control	Description
MODE	The compressor mode, either **SINGLE** or **MULTIBAND**. The mode determines how the compressor processes audio signals. **SINGLE** — Select to configure the module as a single-band compressor. This causes the compressor to affect the entire frequency spectrum of the audio signal as one unified band. In other words, it applies compression uniformly across all frequencies. Single-band compressors are often used for overall dynamic control of a track or mix. They are simpler and more straightforward, making them ideal for general-purpose compression. However, when a single-band compressor is triggered by a loud frequency, it compresses the entire signal. This can cause other frequencies, such as mids and highs, to be compressed as well, even if they don't need it, leading to less precise dynamic control. **MULTIBAND** — A multiband compressor divides the audio signal into multiple frequency bands, allowing you to compress each band independently. This provides more precise control over the dynamics of different parts of the frequency spectrum. Select this option to make the compressor become a multiband upwards/downwards compressor. This is an extreme setting, but you may find a use for it. The individual bands are separately user-adjustable, and you can assign modulations using the modulation matrix. This is useful for side-chaining just the low end out of the way of a kick or bass.
THRESH	The threshold (in dB) for the compression to start engaging. A setting of 0 equals 0 dB (no compression, unless the input signal is overloaded); a setting of 100% equals -120 dB (almost always compressing). Typically, you would set this to around the middle of the range (for example, around -12 dB), but the setting is dependent on your input signal strength and the amount of compression you want.
RATIO	The strength of the gain reduction. Typical compression is between 2:1 and 4:1. If you set the compression knob to maximum, **Limit** appears. This offers a completely different DSP circuit (a true peak limiter and not a compressor); therefore, the other controls behave differently when the limiter is engaged. Specifically, the attack time range changes to 0-10ms and the makeup gain range changes to 0-36dB. **Note:** Setting the ratio to **Limit** can introduce latency. Right-click the knob and choose **Limiter Latency Comp** in the context menu to have the module report this latency to the host.

Control	Description
ATTACK	The amount of time in milliseconds for the gain reduction to engage. A longer (slow) attack is useful for letting some signal through before the gain reduction takes place, resulting in a “punch,” “snap,” or “bite” (sounds are hard to describe using language!). A shorter/faster attack tames peaks more completely.
RELEASE	The amount of time for the gain reduction to be removed.
GAIN	The amount of makeup gain. This is a good way to boost quiet signals. The control allows for approximately 30dB of boost for the compressor and 36dB for the limiter so be careful (that’s a large amount of gain). Basically, a little can go a long way.
X-LOW	(Enabled when **MULTIBAND** is selected) Sets the low crossover for the multiband split.
BELOW	(Enabled when **MULTIBAND** is selected) Sets the compression ratio below the threshold.
X-HIGH	(Enabled when **MULTIBAND** is selected) Sets the high crossover for the multiband split.
H	(Enabled when **MULTIBAND** is selected) Sets the high band gain.
M	(Enabled when **MULTIBAND** is selected) Sets the mid band gain.
L	(Enabled when **MULTIBAND** is selected) Sets the low band gain.
MIX	The wet/dry amount for the effect, from 0 (100% dry) to 100 (100% wet).
LEVEL	The output level of the module (in decibels).

Convolve

The **CONVOLVE** module allows you to apply the impulse response of a signal you select (representing the characteristics of a room’s reverb or a specific filter) to your sound design.

The most common use is to create a convolution reverb, where the impulse response of a real acoustic space (such as a concert hall) is convoluted with your audio signal to simulate how that signal would sound if played in that space. This allows for highly realistic and complex reverb effects.

You can also use the **CONVOLVE** module to blend sounds in unique ways, applying an effect that alters your sound in a manner different from simple filtering or other modulation techniques.

Convolve Module

The **CONVOLVE** module offers the following controls:

Control	Description
IMPULSE	Click to choose from a menu of impulse responses. Use the < > arrows to advance through the modes without having to open the menu. You can load impulse responses from outside the **`Serum 2 Presets`** folder by dragging and dropping the files on the IR display or selecting **Load IR** from the context menu. If you do this, the option to **Embed in Preset** appears in the menu, and an embed icon appears at the top right of the IR display. This allows you to embed the impulse response into the preset, similar to what you can do with oscillators.
SIZE	The size with which to stretch or contract the impulse.
TONE	Use to filter the impulse.
ϕ MIN	The convolution minimum phase. This converts the IR to a minimum-phase representation, keeping the frequency response unchanged and eliminating echoing.
PRE-DLY	The convolution pre-delay. This offsets the impulse in time.
BPM	Specifies whether the pre-delay is synced (BPM or millisecond based.
ATTACK	The convolution attack. Use this to fade in the impulse.
DECAY	The convolution decay. Use to shorten the impulse.
DAMP	Use to shorten the high frequency of the impulse.
IR GAIN	The impulse volume.
MIX	The wet/dry amount for the effect, from 0 (100% dry) to 100 (100% wet).
LEVEL	The output level of the module (in decibels).

Delay

The **DELAY** module records an input signal and then plays it back after a period of time. The delayed signal may be played back multiple times, or fed back into the recording, to create the sound of a repeating, decaying echo.

Delay Module

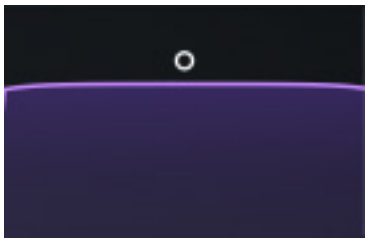
The **DELAY** module offers the following controls:

Control	Description
MODE	The type of delay, from among the following: **NORMAL** — Specifies a standard stereo delay with independent left and right channels/times. **PING-PONG** — Sets the outputs of the left and right delays to feed into one another. **TAP-> DELAY** — Sets both delays to mono and in series, causing the left signal to fire once with no feedback (tap) followed by the right signal operating as a typical delay (feedback is applied here). Select the **High Quality** option to render the output of the module in higher quality.
Delay Times	The delay times. There are two settings available for both the left and right channels. The upper input is the base delay time for the channel. The lower box is a (scalar) offset for the delay time you set above. For instance, a value of 1.1 means that the corresponding delay time sounds at 110% of its value. Dragging the lower inputs displays Trip or Dot when you reach 133% (1.333) and 150% (1.5) respectively. This allows you to quickly set triplet or dotted values for the delay times.
BPM/MS	Specifies whether the delay times are in tempo-based units (quarter note, for example) or in milliseconds.
LINK 	Enable to have the right-channel delay times link with (kept the same as) the left channel delay times.
FEEDBACK	The amount of the delayed signal appearing back at the input of the delay. This is useful for controlling how many delay repeats are audible.
FREQ (Frequency)	The filter cutoff frequency (in Hertz) for the delay filter.
Q (Resonance)	The bandwidth for the delay filter (how much low pass and high pass are applied). Technically, this is the opposite of a standard Q control, where a larger Q value typically signifies a narrow filter bandwidth. However, in this case, a maximum value means minimum filtering/maximum bandwidth.
MIX	The wet/dry amount for the effect, from 0 (100% dry) to 100 (100% wet).
LEVEL	The output level of the module (in decibels).

Manipulating the Delay Filter

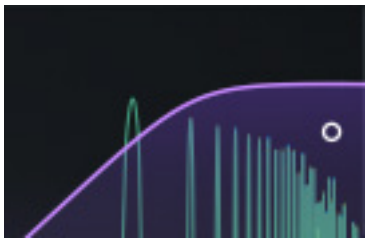
In addition to using the **FREQ** and **Q** knobs, you can manipulate the delay filter using your mouse.

By default, the delay filter appears like this.



Click the drag in the display to set the frequency and Q simultaneously.

Double-click the display to toggle a real-time frequency overlay.

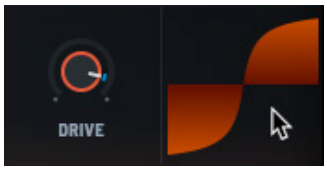


Distortion

The **DISTORTION** module offers 13 types of distortion, including two dual-waveshaper modes that allow you to create your own custom distortion.

Distortion Module

The **DISTORTION** module offers the following controls:

Control	Description
MODE	Click to choose from a menu of distortion types (**Tube**, by default). You can use the **< >** arrows to advance through the modes without having to open the menu.
OFF/PRE/POST	A switch to enable filtering, which you can set to either pre-distortion (**PRE**) or post-distortion (**POST**).
TYPE	The filter type for the distortion module. Drag the red control between left and right to morph from low pass to bandpass to high pass.
FREQ	The cutoff frequency for the filter (when the filter is enabled). Double-click the field to display a text box, allowing you to enter a frequency value. Right-click the knob and choose **Key Track** in the context menu to have frequency respond to the pitch of the note played. With **Key Track** disabled, the frequency is fixed regardless of the key played. With **Key Track** enabled, the frequency changes in proportion to the pitch of the note played (higher-pitched notes increase the frequency).
Q	The resonance for the filter (when the filter is enabled). Double-click the field to display a text box, allowing you to enter a resonance value. You can set a high value to create squelchy feedback. Typically, lower values are more common.
DRIVE	Generally, the gain boost for the distortion, with the following exceptions: • With Downsample filter type, the **DRIVE** knob controls the sample rate reduction amount. • With **X-Shaper** and **X-Shaper (Asym)** modes, the DRIVE knob affects a morph between the two waveshapes. You can also set the drive by dragging up or down in the graphic display. 
MIX	The wet/dry amount for the effect, from 0 (100% dry) to 100 (100% wet).
LEVEL	The output level of the module (in decibels).

Understanding X-Shaper (Dual Waveshaper) FX Modes

The X-Shaper is a dual crossfading waveshaper. Selecting **X-Shaper** in the distortion menu causes **Edit A** and **Edit B** buttons to appear directly below the menu.

Clicking either button displays a pop-up "X-Y" graph editor for the respective waveshaper. In both cases, the X (horizontal) axis represents the input level, and the Y (vertical) axis represents the corresponding remapped output level for an input level.

The **DRIVE** knob, described above, controls the blend between the two waveshaping graphs (the **DRIVE** knob at 0% presents waveshaper A while 100% presents waveshaper B).

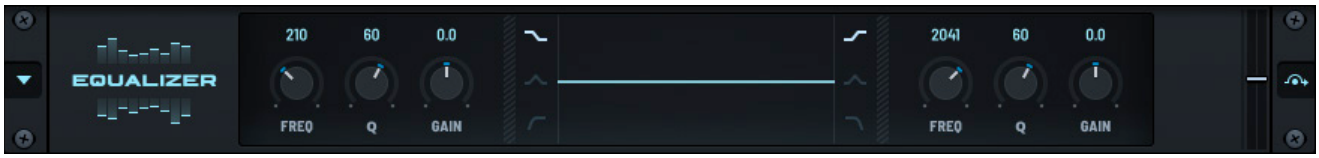
Note that **X-Shaper** is a symmetric waveshaper, with the lower-left point on the graph representing silence (-INF dB for input and output). Similarly, the top-right point represents the highest level (0 dB for input and output).

In contrast, **X-shaper (Asym)** is an asymmetric waveshaper. In this case, the middle of the graph represents silence (-INF dB input and output), the top-right represents the highest positive value to the signal, and the lower-left represents the highest possible negative value.

Asymmetric distortion allows you to bring out even-order harmonics that are not typically found in a standard symmetric distortion (such as clipping). This is often the case in guitar amps; one pole distorts (for example, fatline) while the other pole remains relatively undistorted.

Equalizer

The **EQUALIZER** module offers two-band parametric control.



Equalizer Module

You can set the type of each of the two bands using the corresponding three-state switches. The left band offers low-frequency (LF) adjustment and enables low shelf, peaking, or high pass filtering. The right band offers high-frequency (HF) adjustment and enables high shelf, peaking, or low pass filtering.

The **EQUALIZER** module offers the following controls:

Control	Description
FREQ (L)	The frequency (in Hz) for the low EQ band.
Q (L)	The Q (resonance) for the low EQ band.
GAIN (L)	The gain boost/cut (in dB) for the low EQ band. This knob has no effect if you selected **High Pass** as the low EQ band type.

Control	Description
FILTER TYPE	Click the icons to select a **Shelf**, **Peak**, or **High Pass** filter type (on the left) and **Shelf**, **Peak**, or **Low Pass** filter type (on the right).
FREQ (R)	The frequency (in Hz) for the high EQ band.
Q (R)	The Q (resonance) for the high EQ band.
GAIN (R)	The gain boost/cut (in dB) for the high EQ band. This knob has no effect if you selected **Lowpass** as the high EQ band type.
LEVEL	The output level of the module (in decibels).

Filter

The **FILTER** module operates identically to the per-voice synth filter found on the main **OSC** tab, except that in this case, it runs as a master effect.

Filter Module

The **FILTER** module offers the following controls:

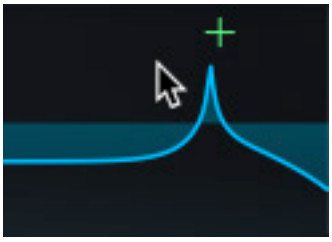
Control	Description
TYPE	Click to choose from a menu of filter types (**MG Low 6**, by default). You can use the **< >** arrows to advance through the filter types without having to open the menu. See "Filter Types and Var Parameter Functions" for more information about the available filter types.
CUTOFF	The primary cutoff frequency for the filter (with just a couple exceptions, such as vowels for formant filters). **TIP:** You can recreate certain paraphonic vintage synth behaviors by applying an envelope to this knob. Right-click the knob and choose **Key Track** in the context menu to have cutoff frequency respond to the pitch of the note played. With **Key Track** disabled, the cutoff frequency is fixed regardless of the key played. With **Key Track** enabled, the cutoff frequency changes in proportion to the pitch of the note played (higher-pitched notes increase the cutoff frequency.
RES	The resonance (feedback) of the filter circuit.

Control	Description
DRIVE	The gain into the filter circuit. The setting can impart some coloration (mild distortion) to the sound. Right-click the knob and choose **Clean Mode** in the context menu to have the filter pre-gain stage the filter input -24 dB (with a +24 dB boost post-filter).
FAT/FREQ/MORPH/ LP FRO/HP FRO/ HL WID/LP FRO/ DB +/-/SPREAD/ DAMP/BOEUF/ THRU/FORMNT/ WIDTH/COMBFRO/ SCREAM/STAGES/ SMOOTH/PAIN/ FRO2	This is a variable knob with different functions depending on the selected filter type. For instance, with "dual" filters, the **VAR** knob controls the second filter cutoff frequency.
PAN	A cutoff offset for the left and right signals. At the default setting of 50% (12 o'clock) this knob has no effect. When turned to the left (counter-clockwise) the left channel cutoff increases, and the right channel cutoff decreases. When turned to the right (past 12 o'clock clockwise), the opposite happens; the left cutoff decreases and the right channel cutoff increases.
MIX	The wet/dry amount for the effect, from 0 (100% dry) to 100 (100% wet).
LEVEL	The output level of the module (in decibels).

You can graphically adjust the filter cutoff and resonance (in combination) by clicking and dragging in the filter display.

This is a quick way to experiment with filter settings in your sound design.

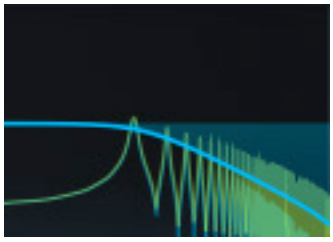
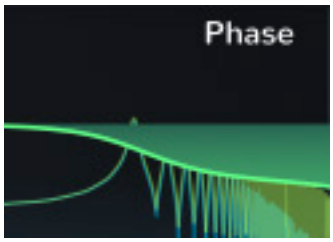
Right-click in the display to access a menu allowing you to choose display options for the filter.



Graphical Adjustments

The following table describes the available display options.

Option	Display	Description
Frequency Response		Shows how the filter affects the amplitude of different frequencies in the audio spectrum, visualized as a graph with frequency on the horizontal axis and amplitude (gain or attenuation) on the vertical axis.

Option	Display	Description
Frequency Response & FFT		Combines the filter frequency response curve with a real-time FFT (Fast Fourier Transform) analysis of the audio input. The FFT displays the actual frequency content of the signal, overlaid with the filter's effect.
Phase Response & FFT		Shows how the filter affects the phase of different frequencies, indicating the degree of phase shift applied to each frequency in the signal, alongside the real-time FFT of the audio.

You can quickly cycle through the different display modes by Option-clicking (macOS) or Alt-clicking (Windows) in the filter display.

Flanger

The **FLANGER** module works by cyclically varying phase shift into one of two identical copies of a signal and then recombining them.

Flanger Module

The **FLANGER** module offers the following controls:

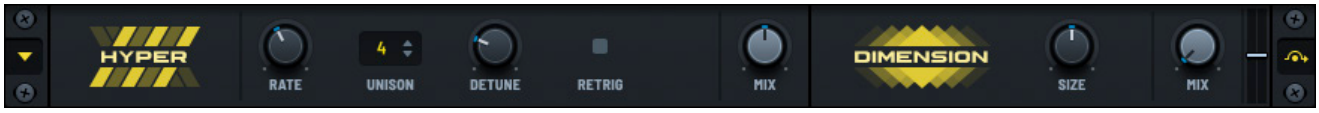
Control	Description
RATE	The rate of the flanger. The units depend on the **BPM** setting. When **BPM** is on, the **RATE** knob snaps to musical time (from 8 bars to 1/32nd note). When **BPM** is off, the **RATE** knob is in Hertz (Hz), between 0 Hz and 20 Hz.
BPM	Specifies whether the flanger sweep is synced to the host BPM. See the **RATE** knob (above) for more information.
DEPTH	Specifies how much the flanger LFO influences the sound, in other words, how much (or deep) the flanger operates.
FEEDBACK	The feedback amount of the flanger circuit, which makes the effect more pronounced ("ringing").

Control	Description
PHASE	The stereo phase offset for the LFO influence over the flanger (the left flange and right flange offset). A setting of 0% sets both left and right to the same frequency. A setting of 50% represents 180 degrees, meaning that the left and right have opposite frequencies. In this case, the flanger sweep "rises" on the left while "falling" on the right, or vice versa.
MIX	The wet/dry amount for the effect, from 0 (100% dry) to 100 (100% wet).
LEVEL	The output level of the module (in decibels).

Hyper/Dimension

The **HYPER/DIMENSION** module is a micro-delay chorus with a variable number of voices (1-7). In addition you can configure the Hyper/Dimension effect to retrigger on every MIDI note, which adds to the potential simulation of a unison.

To conserve CPU, consider using the **HYPER** effect as an alternative to high unison settings.



Hyper/Dimension Module

The **HYPER** module offers the following controls:

Control	Description
RATE	The speed at which the various hyper voices oscillate sharp/fat in pitch.
UNISON	The number of chorus voices. If you only want to use the **DIMENSION** effect and not the **HYPER** effect, set the **UNISON** to 0.
DETUNE	The amount/depth of the hyper voice oscillations (sharp/fat in pitch).
RETRIG	When turned on, resets all hyper voices to start over from a zeroed-pitch offset. This provides a laser-like zap effect for each note-on event. You might use this on certain monophonic patches, for example.
MIX	The wet/dry amount for the effect, from 0 (100% dry) to 100 (100% wet).
LEVEL	The output level of the module (in decibels).

The **DIMENSION** effect is a pseudo-stereo effect consisting of four delay lines summed out-of-phase and slowly amplitude-modulated to provide a subtle amount of motion to the effect. This is useful for adding a perceived width to an otherwise mono signal.

The **DIMENSION** module offers the following controls:

Control	Description
SIZE	Adds an extra layer of phased delays.
MIX	The wet/dry amount for the effect, from 0 (100% dry) to 100 (100% wet).
LEVEL	The output level of the module (in decibels).

Phaser

The **PHASER** module filters a signal by creating a series of peaks and troughs in the frequency spectrum.

Phaser Module

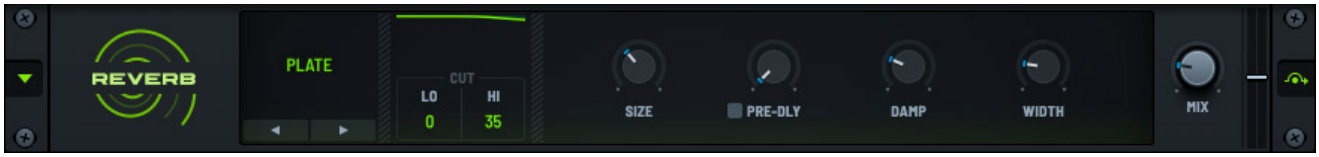
The **PHASER** module offers the following controls:

Control	Description
RATE	The rate of the phaser. The units depend on the **BPM** setting. When **BPM** is on, the **RATE** knob snaps to musical time (from 8 bars to 1/32nd note). When **BPM** is off, the **RATE** knob is in Hertz (Hz), between 0 Hz and 20 Hz.
BPM	Specifies whether the phaser sweep is synced to the host BPM. See the **RATE** knob (above) for more information.
POLES	The number of stacked phaser poles.
DEPTH	Specifies how much the phaser LFO influences the sound.
DEPTH 2	The offset between phaser stages.
FREQ	The base frequency for the phaser effect.
FEEDBACK	The feedback amount of the phaser circuit. A higher setting makes the effect more pronounced (“ringing”).
PHASE	The stereo phase offset for the LFO influence over the phaser (left flange and right flange offset). A setting of 0% sets both left and right to the same frequency. A setting of 50% represents 180 degrees, with left and right at opposite frequencies (in other words, the phaser sweep “rises” on the left while “falling” on the right, or vice versa.

Control	Description
MIX	The wet/dry amount for the effect, from 0 (100% dry) to 100 (100% wet).
LEVEL	The output level of the module (in decibels).

Reverb

The **REVERB** module offers a plate and hall reverb, using a modified version of the Tal Reverb algorithm (courtesy of Togu Audio Line).



Reverb Module

The **REVERB** module offers the following controls:

Style	Control	Description
TYPE		Click to choose from a menu of reverb types (**PLATE**, by default). You can use the < > arrows to advance through the reverb types without having to open the menu.
PLATE	LO CUT	Suppress low frequencies from the reverb. 0% means no effect on lows, 100% means no lows at all.
	HI CUT	Suppress high frequencies from the reverb. 0% means no effect on highs, 100% means no high frequencies at all.
	SIZE	The length of the reverb.
	PRE-DLY	The offset time for the reverb.
	DAMP	An additional high-frequency cut for the reverb. The knob controls how fast this high-frequency attenuation occurs. 0% mean no damping, 100% means maximum damping.
	WIDTH	Expand or collapse the stereo width of the reverb. 100% means maximum width.
HALL	LO CUT	Suppress low frequencies from the reverb. 0% means no effect on lows, 100% means no lows at all.
	HI CUT	Suppress high frequencies from the reverb. 0% means no effect on highs, 100% means no high frequencies at all.
	SIZE	The room size (reverb time + dimension).

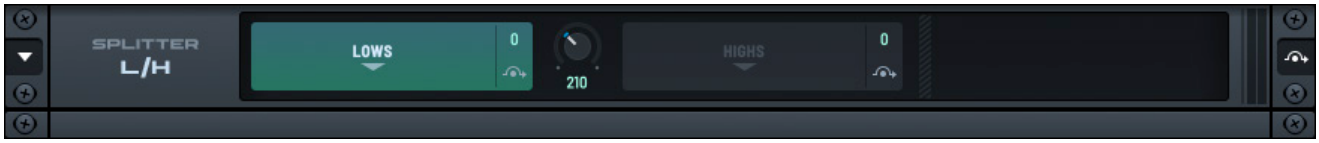
Style	Control	Description
HALL (cont.)	PRE-DLY	The amount of time (in milliseconds) before reverberation occurs. Using pre-delay allows you to give the impression that a sound is close to you but in a large room. You can also use it to separate your transient from the reverb or create a delay-like echo.
	DECAY	The amount of decay (in milliseconds).
	SPIN RATE	Set the speed of the LFO used to modulate time differences.
	SPIN DEPTH	Modulate time differences with an LFO to create a sense of movement within the reverb.
VINTAGE	LO CUT	Suppress low frequencies from the reverb. 0% means no effect on lows, 100% means no lows at all.
	HI CUT	Suppress high frequencies from the reverb. 0% means no effect on highs, 100% means no high frequencies at all.
	SIZE	The length of the reverb.
	PRE-DLY	The amount of time (in milliseconds) before reverberation occurs. Using pre-delay allows you to give the impression that a sound is close to you but in a large room. You can also use it to separate your transient from the reverb or create a delay-like echo.
	ER SIZE	The length of the early reflection part of the reverb.
	DECAY	The amount of decay (in milliseconds).
	DAMP	The speed at which high frequencies decay.
	DIFF A	The diffusion of the reverb.
	DIFF B	Dampen the diffusion stage of the reverb.
	CHORUS	The top value sets the speed that the reverb is modulated. The bottom value sets the pitch depth that the reverb is modulated.
NITROUS	LO CUT	Suppress low frequencies from the reverb. 0% means no effect on lows, 100% means no lows at all.
	HI CUT	Suppress high frequencies from the reverb. 0% means no effect on highs, 100% means no high frequencies at all.
	SIZE	The length of the reverb.
	PRE-DLY	The offset time for the reverb.
	FEEDBACK	The amount of reverb fed back into the input signal.
	DIFFUSION	The diffusion of the reverb.

Style	Control	Description
	MODE	The nitrous mode, from among the following: • Space • Marble • Rectangle • Hexagon • Box
NITROUS (cont.)	CHORUS	The top value sets the speed that the reverb is modulated. The bottom value sets the pitch depth that the reverb is modulated.
BASIN	LO CUT	Suppress low frequencies from the reverb. 0% means no effect on lows, 100% means no lows at all.
	HI CUT	Suppress high frequencies from the reverb. 0% means no effect on highs, 100% means no high frequencies at all.
	SIZE	The length of the reverb.
	PRE-DLY	The offset time for the reverb.
	FEEDBACK	The amount of reverb fed back into the input signal.
	CHORUS	The top value sets the speed that the reverb is modulated. The bottom value sets the pitch depth that the reverb is modulated.
MIX		The wet/dry amount for the effect, from 0 (100% dry) to 100 (100% wet).
LEVEL		The output level of the module (in decibels).

Splitter L/H

The **SPLITTER L/H** module divides the audio signal into distinct low and high-frequency bands, enabling you to design and apply dedicated FX racks for each band independently.

This setup provides precise control over your sound, allowing for tailored processing that enhances both the low-end punch and high-end clarity.



Splitter L/H Module

When using this module, you can build two separate racks, one to handle the **LOWS** and another to process the **HIGHS**. The list view (on the left) shows both racks. The rack view, however, only shows the currently-selected rack. Click the **LOWS** or **HIGHS** panel to display the corresponding rack.

Splitter L/H Module Example

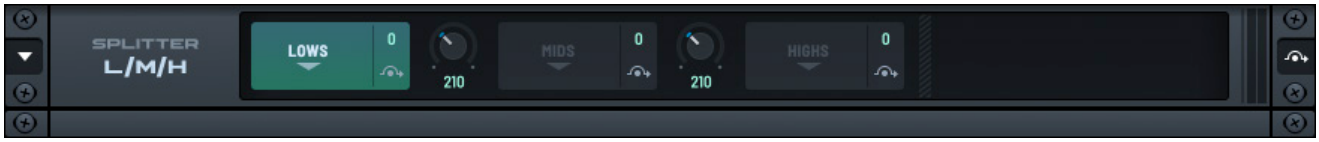
The **SPLITTER L/H** module offers the following controls:

Control	Description
LOWS	Right-click in the panel and choose an FX module to add using the menu that appears. Continue doing this until you finish building the low-end rack. Each time you add a module, it appears below the **SPLITTER L/H** main module. Note the bypass button in the **LOWS** panel. Enabling this bypasses the low-end rack.
SPLIT FREQ	The crossover frequency for the low and high bands.
HIGHS	Right-click in the panel and choose an FX module to add using the menu that appears. Continue doing this until you finish building the high-end rack. Each time you add a module, it appears below the **SPLITTER L/H** main module. Note the bypass button in the **HIGHS** panel. Enabling this bypasses the high-end rack.
LEVEL	The output level of the module (in decibels).

Splitter L/M/H

The **SPLITTER L/M/H** module divides the audio signal into low, mid, and high-frequency bands, offering even greater flexibility than the **SPLITTER L/H** module. This configuration enables you to design and apply dedicated FX racks for each band independently, allowing for fine-tuned control over your sound.

With the addition of the **MIDS** panel, you gain the ability to shape and process the critical midrange frequencies separately, ensuring that elements like vocals, guitars, and synths stand out or blend seamlessly. This setup provides precise control, enhancing the low-end punch, midrange presence, and high-end clarity, for a more refined and impactful mix.



Splitter L/M/H Module

When using this module, you can build three distinct racks, one to handle the **LOWS**, one for the **MIDS**, and another to process the **HIGHS**. The list view (on the left) shows all three racks. The rack view, however, only shows the currently-selected rack. Click the **LOWS**, **MIDS**, or **HIGHS** panel to display the corresponding rack.

Splitter L/M/H Module Example

The **SPLITTER L/M/H** module offers the following controls:

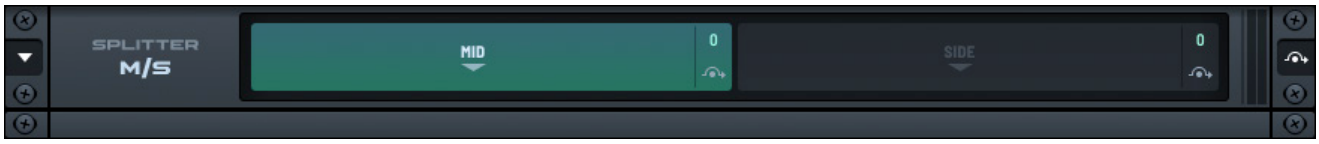
Control	Description
LOWS	Right-click in the panel and choose an FX module to add using the menu that appears. Continue doing this until you finish building the low-end rack. Each time you add a module, it appears below the **SPLITTER L/M/H** main module. Note the bypass button in the **LOWS** panel. Enabling this bypasses the low-end rack.

Control	Description
SPLIT FREQ	The crossover frequency for the low and mid bands.
MIDS	Right-click in the panel and choose an FX module to add using the menu that appears. Continue doing this until you finish building the low-end rack. Each time you add a module, it appears below the **SPLITTER L/M/H** main module. Note the bypass button in the **MIDS** panel. Enabling this bypasses the mid-end rack.
SPLIT FREQ	The crossover frequency for the mid and high bands.
HIGHS	Right-click in the panel and choose an FX module to add using the menu that appears. Continue doing this until you finish building the high-end rack. Each time you add a module, it appears below the **SPLITTER L/M/H** main module. Note the bypass button in the **HIGHS** panel. Enabling this bypasses the high-end rack.
LEVEL	The output level of the module (in decibels).

Splitter MS

The **SPLITTER M/S** module divides the audio signal into mid and side bands. This configuration enables you to design and apply dedicated FX racks for each band independently, allowing for precise control over the central and spatial elements of your sound.

Use it to enhance focus and presence by processing the mid channel, or to control depth and width by processing the side channel, giving you a powerful tool for shaping your mix.



Splitter M/S Module

When using this module, you can build two separate racks, one to handle the **MID** band and another to process the **SIDE** band. The list view (on the left) shows both racks. The rack view, however, only shows the currently-selected rack. Click the **MID** or **SIDE** panel to display the corresponding rack.

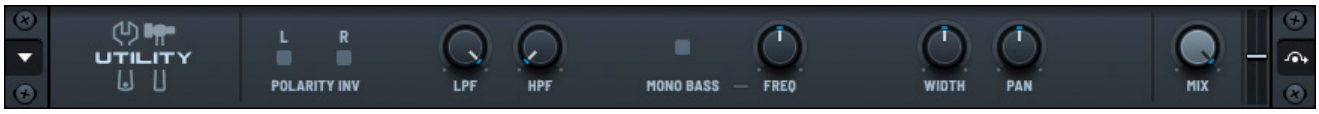
Splitter M/S Module Example

The **SPLITTER M/S** module offers the following controls:

Control	Description
MID	Right-click in the panel and choose an FX module to add using the menu that appears. Continue doing this until you finish building the mid-band rack. Each time you add a module, it appears below the **SPLITTER M/S** main module. Note the bypass button in the **MID** panel. Enabling this bypasses the mid-band rack.
SIDE	Right-click in the panel and choose an FX module to add using the menu that appears. Continue doing this until you finish building the side-band rack. Each time you add a module, it appears below the **SPLITTER M/S** main module. Note the bypass button in the **SIDE** panel. Enabling this bypasses the side-band rack.
LEVEL	The output level of the module (in decibels).

Utility

The **UTILITY** module offers a series of "utility" functions including polarity inversion, basic low and high pass filters, stereo width and balance, and more.



Utility Module

The **UTILITY** module offers the following controls:

Control	Description
POLARITY INV	Invert the polarity of the audio signal on the left and right channel of the signal respectively.
LPF	The low pass filter.
HPF	The high pass filter.
MONO BASS/FREQ	Enable for mono bass, which forces frequencies below the threshold (set using the **FREQ** control) to be monophonic.
WIDTH	The stereo width.
PAN	The stereo balance.
MIX	The wet/dry amount for the effect, from 0 (100% dry) to 100 (100% wet).
LEVEL	The output level of the module (in decibels).

Exploring Sound Modulation

Serum offers advanced options for sound modulation, allowing you to create dynamic sounds with motion. Modulating sound in Serum principally involves working in the following three areas:

1. Envelopes
2. LFOs
3. Modulation Matrix

Serum Sound Modulation

Using Envelopes

The **Envelopes** module offers four modulation sources, labeled **ENV 1**, **ENV 2**, **ENV 3**, and **ENV 4**. The following shows the **Envelopes** area of Serum.

Serum Envelopes

Configuring Envelopes

Select an envelope 1 by clicking the corresponding envelope tab.

Modify the envelope directly using your mouse. 2

Alternatively, adjust the **ATK** (attack), **HOLD**, **DEC** (decay), **SUS** (sustain), and **REL** (release) knobs to change the envelope parameters. 3

Click the (lock) button 4 to auto-normalize the zoom.

Alternatively, drag the mouse through the zoom slider to manually zoom the envelope display. 5

Envelope Operations

Lock Button	Description
Locked	The envelope always zooms to perfectly fill the display area. This means that adjusting envelope times simply changes how much time is represented on the ruler scaling below the envelope.
Unlocked	Allows you to zoom in and out of the envelope by dragging up and down in the zoom area directly below the Lock button.

See “Modifying Envelopes” later in this chapter for details about changing envelope shapes.

You can use all four envelopes, though **ENV 1** is considered special within Serum because it is used with the amplifier, controlling the output volume of each voice. You can still use **ENV 1**, however, assigning it to any parameter as you would the other envelopes.

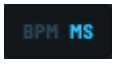
When you select **ENV 2**, **ENV 3**, or **ENV 4**, the envelope of **ENV 1** is faintly visible (in gray) in the background.

Inverting the Legato Setting

When the main **LEGATO** switch is enabled, envelopes do not retrigger if a second note is played while a first note is still held. You can invert this option to force an envelope to always trigger at note on, even when legato is enabled.

To do this, right-click the envelope graph and choose **Legato Inverted** in the context menu.

Setting Envelope Parameter Units

You can choose to alter values in milliseconds or in subdivisions of a note by clicking 

(**MS** or **BPM**).

When you switch from **MS** to **BPM**, Serum calculates the nearest subdivision to the millisecond value for each control based on host tempo.

Setting the Envelope Verticals

You can select whether the vertical lines that appear in the graph background are placed at time (ms) or beat intervals.

Right-click on the graph and select either **Time** or **Beats** in the **Grid** sub-menu.

Modifying Envelopes

You can modify envelope shapes either by setting values using the knobs in the envelope window, or by manipulating the envelope waveform directly.

To modify an envelope directly, click the corresponding envelope tab, such as **ENV 2**, **ENV 3**, or **ENV 4** and graphically adjust the envelope points using your mouse.

Alternatively, adjust the respective knob settings using your mouse.

Important: Remember that **ENV 1** is a special envelope in Serum (it controls the output volume) and is therefore always active.

Graphical Envelope Edits

You can modify the following envelope parameters using the knobs and directly on the graph:

Control	Description
ATK	Attack. The time for the initial run-up from start to peak, beginning with the note on event.
HOLD	Hold. The time that the envelope stays at full volume before entering the decay phase.
DEC	Decay. The time for the subsequent run-down from the attack level to the designated sustain level.
SUS	Sustain. The level during the main sequence of the sound's duration, until the note off event.
REL	Release. The time for the level to decay from the sustain level to zero after the note off event occurs.

You can modify the following parameters directly using your mouse (only).

- Attack curve
- Decay curve
- Release curve

Curve Adjustments

Assigning an Envelope to a Control

You can assign an envelope to a control (such as a knob), causing the envelope to modulate the control.

For example, you might choose to have an envelope modulate the wavetable position (**WT POS**) setting of **OSC A**.

To do this, click and drag the **ENV 1** tab to the **WT POS** knob in the **OSC A** panel.

As you are dragging, notice that an **ENV 1** label hovers next to the mouse pointer.

The pointer adds a **+** sign as you hover over an assignable knob (in this case, the **WT POS** knob).

The **+** sign indicates that you are over a valid mod destination.

Assigning ENV 1 to WT POS

When you release the mouse button, Serum automatically makes the connection causing **ENV 1** to now affect the **OSC A** wavetable position.

Right-click an envelope tab to bypass and remove destinations assigned to the envelope.

Bypass and Remove Assignments

Using LFOs

Serum offers ten low frequency oscillators (LFO) that each feature a set of independent controls. The following shows the LFO area of Serum.

Serum LFOs

Note: When you initialize a patch, only **LFO 1** to **LFO 6** are visible. **LFO 7** to **LFO 10** becomes visible after you use (assign) **LFO 6**.

Configuring LFOs

Select an LFO 1 by clicking the corresponding LFO tile.

Modify the LFO graph directly 2 using your mouse.

Change tools 3 to get different drawing results.

Fine-tune using the LFO Editor, if needed. 4

Adjust knobs and controls 5 to change various parameters.

LFO Operations

Note that the settings are independent for each LFO.

To copy all LFO settings from one LFO to another, Option-click-drag (macOS) or Alt-click-drag (Windows) an LFO title to another.

For example, to copy all the **LFO 1** settings to **LFO 2**, Option/Alt-click-drag the **LFO 1** tab to **LFO 2**.

Drawing an LFO Graph

Serum offers a set of tools to help you create LFO graphs using your mouse.

LFO Graph Tools

Note that each tool relies on the current grid setting. The following table describes the LFO drawing tools available:

Type	Tool	Use to ...
Point		Manipulate points in the graph, including adding new points, moving points, deleting points, and adjusting curves (between main points).
Flat		Add flat lines between points, based on the current grid size.
Ramp Up		Add ramp ups between points, based on the current grid size.
Ramp Down		Add ramp downs between points, based on the current grid size.

Modifying LFOs

When modifying points on an LFO graph, you can do the following:

Action	Task
Double-click	Add or remove points.
Shift-click	Draw steps at the grid size (step sequencer).
Option-click-drag a point (macOS) Alt-click-drag a point (Windows)	Snap the point to the grid size.
Option-click-drag any curve point (macOS) Alt-click-drag any curve point (Windows)	Move all curve points at once.
Click-drag on the background	Select multiple points.
Cmd-click-drag a point (macOS) Ctrl-click-drag a point (Windows)	Select multiple points for relative movement. Rainbow colors appear on the points. Dragging them makes closer points move more, and further points move less.
Ctrl-click (macOS) Right-click (Windows)	Display a context menu showing additional features, such as setting the segment shape for Shift-click, removing all selected points, or assigning the start or loopback points.
Shift-Cmd-click (macOS) Shift-Ctrl-click (Windows) on a point (in **Envelope** mode)	Set the point as the loopback position (or the very last point if you intend no loopback position). This is simply a shortcut to avoid the menu.

Modifying LFO Controls

The LFO module features a complete set of controls that you can use to set the LFO type, LFO mode, BPM, grid size, and other options. The following table explains each of the LFO controls:

Control	Description
TYPE	The LFO type, from among the following: • Normal • Path • Chaos: Lorenz • Chaos: Rossler • S&H
MODE (Retrig)	Specifies how the LFO behaves when a new note is played. • **FREE** — The LFO follows the host clock and ignores note timing. • **RETRIG** — Retriggers the LFO, causing the LFO to start with a new note. Use this setting when you want the LFO to always have the same timing with new notes. • **ENVELOPE** — Similar to **RETRIG** however the LFO plays through a single cycle before stopping. It's possible to loop a segment of the LFO while in envelope mode using the loopback point, accessible by Ctrl-clicking (macOS) or right-clicking (Windows) a point and choosing **Set Loopback Point Here**. This causes the LFO to play through, then cycle back to the selected loopback point.
MONO	Select whether the LFO is monophonic or polyphonic. By default, LFOs are polyphonic allow independent modulation for each voice. With **MONO** enabled, the same modulation is applied to all voices.
SHAPE	Displays a pop-up menu allowing you to load an LFO preset. Note that this overwrites the current LFO graph. The menu also offers the ability to save the current LFO graph as a user-defined preset.
DIRECTION	The direction of the LFO, from among the following: • Forward • Reverse • Ping Pong

Control	Description
GRID	The grid size of the LFO graph. The visual grid background in the LFO graph changes as you adjust this setting. Double-click the horizontal or vertical grid box and specify a number to set a corresponding grid. The grid is helpful when Alt-clicking (to snap points) or Shift-clicking (to draw step segments) on the LFO graph. This allows you to align modulation to the rhythm of your production, or create arpeggiator-like pitch modulations.
HOST	Specifies whether the LFO is always synced to the global song position. When **BPM** is enabled, the **HOST** switch determines whether the LFO playback position "jumps" if you change the LFO rate. When **HOST** is enabled, the phase is "anchored" to the host transport position. For example, when changing the rate from ¼ note to 1 bar, the phase may jump to have the playback properly fixed to the bar cycle.
BPM/HZ	Set the time value to snap to song tempo-based units (1/4 note, 1/8 note, and so on) or Hertz (free time).
RATE	The playback speed of the LFO. This determines the amount of time represented on the LFO graph area. The LFO rate is in beat-synced units by default (when BPM is enabled) but can also be set to a frequency in Hertz (when HZ is enabled). When **BPM** is selected (see above), right-click the **RATE** knob and choose **Swing** in the context menu to add swing to the LFO (using the **SWING** setting above the keyboard). When **HZ** is selected, right-click the **RATE** knob and choose **10x** in the menu to have the range of the rate control (in Hz) multiplied by a factor of 10, allowing for faster LFO rates.
TRIP/DOT	Set triplet and dotted time on the rate control respectively. These are useful for avoiding triplet or dotted times when automating the LFO rate, that is, for avoiding dotted and triplet times when you know you want evenly beat-divisible time.
RISE	The amount of time for the LFO graph shape to have influence over the LFO output. The LFO begins with a fixed output (imagine the LFO graph as a fat horizontal line, with the value of the left-most point of the LFOTool graph) and slowly (based on the rise time) becoming the shape of the visible graph. This knob is useful for having the LFO slowly influence your sound.

Control	Description
DELAY	The amount of time before the rise begins. The LFO has a fixed output, as described above. After the delay time period, the rise begins.
SMOOTH	Smooth the LFO output. This is useful for avoiding abrupt jumps in the LFO output, without having to draw ramps on every segment of the LFO graph.
PHASE	Sets the start position of the LFO phase. Right-click the **PHASE** knob and choose **Snap to Grid** in the context menu to have the phase value snap to the vertical grid lines (as defined by the grid parameter at the bottom right of the LFO display).

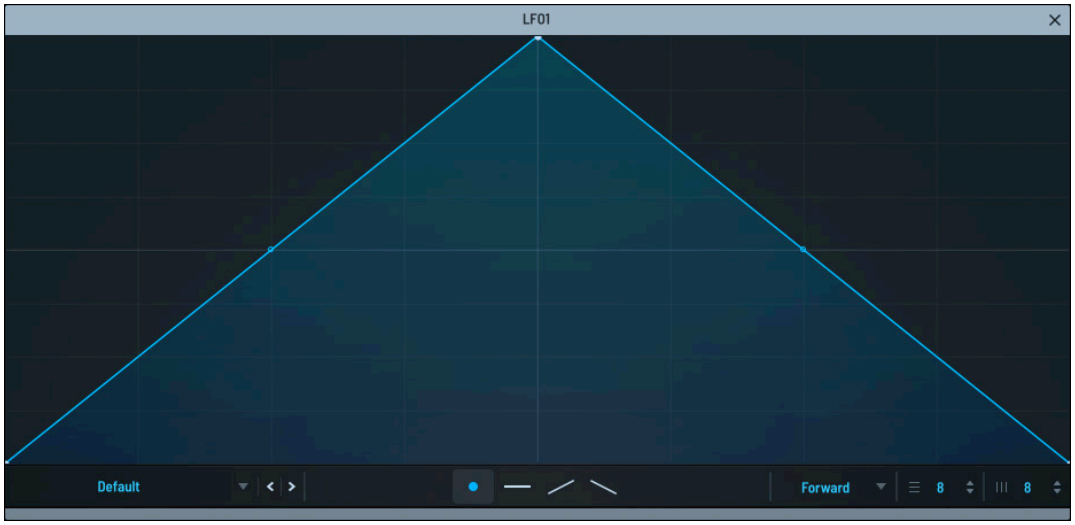
Unlike Serum 1, the **HOST** switch now has an effect when **BPM** is disabled.

With **BPM** disabled, the **HOST** switch determines whether phase is calculated from the host transport sample position on retriggering an LFO. One reason to enable the **HOST** switch is to ensure that, even though an LFO is set to **FREE**, it plays back exactly the same every time a song is played through.

Conversely, a reason to disable the **HOST** switch is to allow a truly free-running LFO. Each time playback of a song starts from the beginning, the LFO phase will continue as if it had been free-running since the last time playback started.

Using the LFO Editor

Click the  button to access a dedicated LFO Editor featuring a larger canvas.



LFO Editor

You can use the same range of tools in the LFO Editor, with the advantage of being able to make finer adjustments using the larger canvas.

Assigning an LFO to a Control

You can assign an LFO to a control (knob), causing the LFO to modulate the control. For instance, to assign **LFO 2** to the **DETUNE** knob in **OSC C**, click and drag from the **LFO 2** tab to the **DETUNE** knob in the **OSC C** panel.

As you are dragging, notice that an **LFO 2** label hovers next to the mouse pointer The pointer adds a **+** sign as you hover over an assignable knob (in this case, the **DETUNE** knob).

Assigning LFO 2 to DETUNE

The **+** sign indicates that you are over a valid mod destination.

When you release the mouse button, Serum automatically makes the connection causing **LFO 2** to now affect the **OSC C** detune.

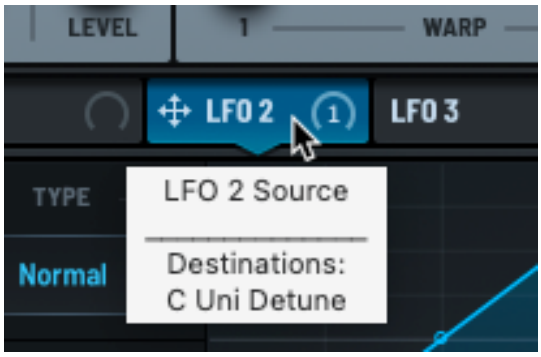
After setting this modulation, notice that a number 1 now appears next to **LFO 2**.

This indicates that **LFO 2** has one destination.

Mod Source with One Destination Assigned

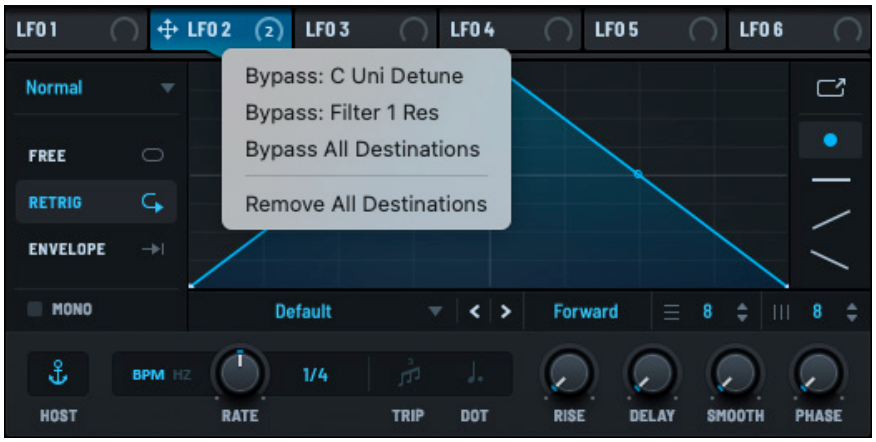
Hover the mouse pointer over the mod source to display a tooltip showing the destinations.

In this case, **LFO 2** displays **C Unison Detune** as the assigned destination.



Hovering to Show Destinations

Right-click an LFO tab to bypass and remove destinations assigned to the LFO.



Bypass and Remove Assignments

Setting the Modulator Depth

When you connect a modulation source with a destination, such as **LFO 2** with **OSC C DETUNE**, a blue halo appears around the knob (in this case, the **DETUNE** knob).

This indicates the **LFO 2** depth on the **DETUNE** knob, defining how much influence the LFO has over the control (knob) position.

DETUNE Modulator Depth at 100%

A smaller blue halo appears to the top left of the knob. Hovering over this small halo displays an Up/Down arrow control.

Click and drag the arrow control to change the modulation depth amount.

As you drag the arrow, notice how the halo shrinks or expands to show the range of modulation.

Setting Modulator Depth

Alternatively, as a shortcut, you can also Option/Alt-click and drag on the main knob (the **DETUNE** knob itself, for example) to change the modulation depth.

Similar to the smaller blue halo, a gray halo appears to the top left of controls that also have a modulator assigned, but when the modulator source is not currently selected.

To adjust the modulator depth in this case, simply click the corresponding mod source tab and adjust the depth using the halo.

Alternatively, you can also adjust the depth using the Modulation Matrix. See “Using the Modulation Matrix” on page 209 for complete details.

Modulator Deselected

When creating modulations by dragging-and-dropping, the default type assigned depends on whether the control is centered. For instance, if you drag to a **DETUNE** knob, which is centered, Serum assumes you want the mod source to pan both left and right so bidirectional is chosen.

If you drag to a control that is not centered, such as the filter resonance (the **RES** knob), Serum assumes you want the modulation to add value only, and unidirectional is used.

You can change the type setting without visiting the modulation matrix window by Shift-Option-clicking (macOS) or Shift-Alt-clicking (Windows) on the knob with a visible (blue/yellow) modulation assignment.

Setting Negative Modulation Depths

As described in the previous section, a blue halo around a control indicates the modulation depth of the corresponding mod source. The blue color indicates a positive value. You can also set the depth to a negative value using the same Click-drag operation.

When the value becomes negative, the color of the halo changes to a lighter blue. This indicates that the depth amount is inverted (as the LFO output goes up, the influence on the control goes down).

Copying a Wavetable Shape to an LFO

You can copy the current wavetable shape from any of the main oscillators (**OSC A**, **OSC B**, **OSC C**) to an LFO, making it quick and easy to define complex LFO shapes.

To do so, load the appropriate wavetable in the oscillator and use the corresponding **WT POS** knob to select the appropriate frame.

Copying a Wavetable to an LFO

Next, select the LFO tab to which you want to copy the wavetable shape, and click the **Default** menu.

Choose **Wavetable A to LFO** to copy the currently-selected frame in **OSC A** to the LFO. The same applies for oscillators B and C.

Copying an LFO Shape to a Wavetable

You can similarly copy the current LFO shape to an oscillator.

Choose or draw the appropriate shape in an LFO and then Option-drag (macOS)/Alt-drag (Windows) the corresponding LFO tab to the wavetable display for either **OSC A**, **OSC B**, or **OSC C**.

Copying an LFO Shape to a Wavetable

Modulating LFO Points

You can modulate one or more LFO points (or curves) on an LFO graph, allowing you to create dynamic or evolving modulations.

LFO point modulation works using LFO busses. Busing allows multiple point modulation by a single source without requiring you to define duplicate assignments.

You can assign modulation to an LFO point using:

- Bus menus
- Drag and drop

Modulating LFO Points Using Menus

You can quickly define LFO point modulation using LFO bus menus.

In the LFO display, right-click (Ctrl-click on macOS) a point and choose **Modulate X** or **Modulate Y** in the menu that appears.

LFO Point Bus Menu

Select an LFO bus (initially only **LFO Bus 1** appears) and choose a modulation source. The modulation source you choose will control the LFO point movement. The LFO graph updates to show a shaded bar representing the modulation range (from minimum to maximum).

You can optionally drag to select multiple points in the LFO graph and configure modulation for these points in a single assignment.

Modulating LFO Points Using Drag and Drop

You can also specify a modulation source by dragging the source, such as **ENV 2**, over an LFO point and dropping on the **X** or **Y** button that appears.

Modulating by Dragging a Source

This allows you to modulate the point either horizontally (X) or vertically (Y), adding the source using the next available LFO bus.

Using Context Menus with Controls

Each control has a context menu that gives you quick access to useful functions. You can access the context menu by Ctrl-clicking (macOS) or right-clicking (Windows) the control.

The context menu that appears offers the following options:

Menu Option	Description
Mod Source (submenu)	Display all possible modulation sources for the control/knob. Use this menu to quickly configure a connection without having to drag from the modulation source or visiting the Modulation Matrix window.
Aux Source (submenu)	Display all possible auxiliary sources for the control/knob.
Edit Custom Curve	Open a curve editor to define a custom mapping applied to the mod source.
Bypass Modulator	Bypass the current modulation connection (between the currently-selected modulation source, such as **LFO 1**, and the control/knob). After selecting the option, a check mark appears next to the menu item and the halo around the control/knob turns gray indicating that the modulation connection is bypassed. You can reverse this by selecting the menu item again (uncheck the option). You can also bypass a modulator by right-clicking the source tile (such as **LFO 1**) and choosing the corresponding option.

Menu Option	Description
Remove Modulator	Remove the connection between the modulation source and the control/knob.
Remove All Modulators	Remove all connections to the control/knob from all modulation sources.
Reset Control	Reset the control/knob to the default value. This is the same as Cmd-clicking (macOS) or Ctrl-clicking (Windows) the control.
MIDI Learn	Activate MIDI learn mode. When enabled, Serum waits for an incoming MIDI CC value. After Serum receives a MIDI CC value, MIDI learn mode is deactivated and the CC# is assigned to the control/knob. The assignment is saved with the preset (patch).
Remove MIDI cc	Remove the MIDI CC# assignment, if any.
Lock Parameter	When enabled, lock the parameter (or module) setting (preventing a value change) when loading presets.

Setting Velocity and Notes

You can use the MIDI velocity and note values, customized through a user-defined graph (curve), to modulate the full range of parameters available in Serum. By mapping these MIDI inputs to specific modulation targets using a graph, you gain precise control over how velocity and note data affect various aspects of the sound.

For example, velocity can dynamically influence parameters like volume, filter cutoff, or even the brightness of the timbre, enabling nuanced expression that responds directly to your playing intensity.

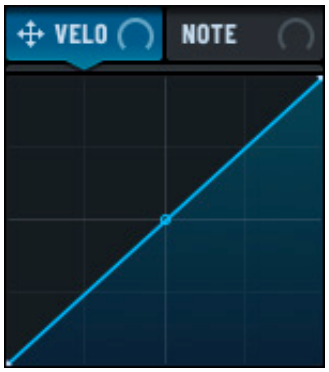
Similarly, you can map note values to parameters such as oscillator pitch, filter resonance, or effects settings, allowing the sound to evolve based on the pitch being played. With the flexibility to design and shape the response curve, you can fine-tune how each parameter behaves across the velocity range or note spectrum.

This approach not only enhances the expressiveness of your sounds but also opens up creative possibilities for crafting unique, dynamic, and musically responsive presets.

Serum Velocity and Notes

Velocity Settings

You can use the velocity tab to define the MIDI velocity graph that you can later use to modulate the range of Serum parameters.



Velocity Graph

Click to select the **VELO** tab, if necessary.

Draw the graph using the tools and operations described in the next section.

Then modulate one or more Serum controls (such as the filter cutoff, for instance) using the procedure described later in this section.

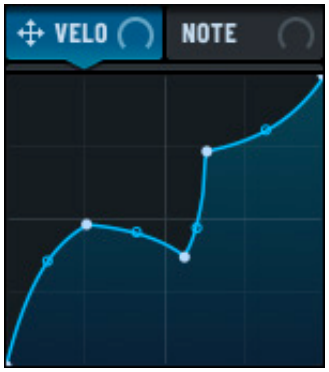
Right-click the **VELO** tab and enable **Legato (Portamento Time)** in the context menu to have portamento applied to the velocity curve when a note is triggered and another is already held.

Using the same context menu, choose **Init Graph** to remove all added points and reset the graph to a straight diagonal line.

Drawing the Graph

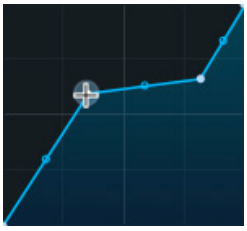
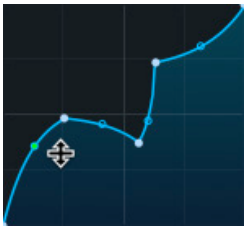
You can modify the velocity graph by manipulating the graph directly, adding new points and dragging curves as needed using your mouse.

Begin by ensuring that the **VELO** tab is selected.



Modified Velocity Graph

The following table describes operations you can perform when editing the graph:

Operation	Display	Description
Double-click		Add a new point to the mask or remove an existing point.
Drag a point		Move a point to a new location.
Drag a curve point		Create or modify a curve between points.
Option/Alt drag a curve point		Create or modify curves between all points simultaneously.

Modulating a Control

You can assign the velocity graph to a control (such as a knob), causing the graph to modulate the control.

For example, you might choose to have the velocity graph modulate the filter cutoff (**CUTOFF**) setting of **FILTER 1**.

To do this, click and drag the **VELO** tab to the **CUTOFF** knob in the **FILTER 1** panel.

As you are dragging, notice that an **VELO** label hovers next to the mouse pointer. The pointer adds a **+** sign as you hover over an assignable knob (in this case, the **CUTOFF** knob).

The **+** sign indicates that you are over a valid mod destination. When you release the mouse button, Serum automatically makes the connection causing **VELO** to now affect the **FILTER 1** cutoff setting.

Velocity Modulating Cutoff

Note Settings

You can use the note tab to define the MIDI note graph that you can later use to modulate Serum parameters.

Click to select the **NOTE** tab, if necessary.

Draw the graph using the tools and operations described in the next section.

Then modulate one or more Serum controls (such as the filter resonance, for instance) using the procedure described later in this section.

Note Graph

Right-click the **NOTE** tab and enable **Legato (Portamento Time)** in the context menu to have portamento applied to the note curve when a note is triggered and another is already held.

Using the same context menu, choose **Init Graph** to remove all added points and reset the graph to a straight diagonal line.

Drawing the Graph

You can modify the note graph by manipulating the graph directly, adding new points and dragging curves as needed using your mouse.

Begin by ensuring that the **NOTE** tab is selected. Then use your mouse to adjust the graph.

Refer to the table in the "Velocity Settings" on page 202 section for details about the various drawing operations available.

Modified Note Graph

Modulating a Control

Similar to the **VELO** graph, you can assign the **NOTE** graph to a control (such as a knob), causing the graph to modulate the control. For example, you might choose to have the note graph modulate the filter resonance (**RES**) setting of **FILTER 1**.

To do this, click and drag the **NOTE** tab to the **RES** knob in the **FILTER 1** panel.

As you are dragging, notice that an **NOTE** label hovers next to the mouse pointer. The pointer adds a **+** sign as you hover over an assignable knob (in this case, the **RES** knob).

The **+** sign indicates that you are over a valid mod destination. When you release the mouse button, Serum automatically makes the connection causing **NOTE** to now affect the **FILTER 1** resonance value.

Note Modulating Resonance

Using Macros

Serum features eight macros that you can use to simplify the control of multiple parameters simultaneously. Macros provide an efficient way to design, perform, and tweak sounds without having to manually adjust numerous parameters one by one.

Macros Pane

For example, instead of adjusting filter cutoff, resonance, and pan individually, you can assign these parameters to a single macro.

Macros are great for experimenting with sound because small changes to a macro setting can result in complex shifts across multiple sound elements. Macros are also powerful in live performance setups, where fast and intuitive control is essential.

In addition to being a modulation source, macros can also serve as a destination. This offers incredible flexibility when setting up modulation.

For example, consider the case where you want a second aux source to modulate a destination. You could set a macro as an aux source and then modulate it with another entry in the modulation matrix, using both a main and aux source.

Assigning a Macro

You can assign a macro to a control by dragging and dropping the macro selector to the appropriate control. As you are dragging, notice that a label hovers next to the mouse pointer The pointer adds a **+** sign as you hover over an assignable knob (in this case, the **DETUNE** knob).

The **+** sign indicates that you are over a valid mod destination.

When you release the mouse button, Serum automatically makes the connection causing the macro to now affect the **OSC A** detune.

Assigning a Macro to a Control

After setting the macro, notice that a number 1 now appears next to **MACRO 1**.

This indicates that **MACRO 1** has one destination.

Macro with One Destination Assigned

Hover the mouse pointer over the macro to display a small tooltip showing the destinations.

In this case, **MACRO 1** displays **A Unison Detune** as the assigned destination.

You can repeat this process and assign the same macro to multiple controls.

You can then manipulate the macro (perhaps assigned to a knob or slider on a physical controller) as you would a mod wheel.

Note: As with other modulators, you can set the modulator depth to offer even finer control over the destination. See "Setting the Modulator Depth" for more information.

Hovering to Show Destinations

To swap macros, drag and drop a macro over another macro.

For example, if **MACRO 1** is assigned to the **WT POS** knob and **MACRO 2** is assigned to the **PAN** knob, dragging and dropping either macro to the other macro swaps the assignments.

Using Oscillators and Filters as Modulation Sources

You can use the output of any oscillator or filter as a modulation source.

Drag the module label to a control to create the modulation assignment.

At this point, the control is modulated by the output of the corresponding oscillator or filter.

Using an Oscillator as a Modulator

Using the Modulation Matrix

Serum features an easy-to-use modulation matrix that shows all configured modulations as a list. This at-a-glance view allows you to quickly select the routing and amounts for the various modulation connections.

Note: Serum offers 64 modulation matrix slots in a patch. You can use these slots to modify or scale up to 64 destination parameters, one per slot, with 49 different modulation sources.

Serum enhances the typical modulation matrix found on many synthesizers by integrating this matrix with the drag-and-drop style of routing described in “Exploring Sound Modulation”.

Serum Modulation Matrix

Dragging a modulation source to a control/knob causes the routing to automatically appear in the matrix and vice-versa. This gives you extra flexibility when viewing, configuring, and modifying your modulation assignments.

Exploring the Modulation Matrix

Click the **MATRIX** tab to access the modulation matrix.

Accessing the Modulation Matrix

You can use the modulation matrix to configure the following options:

Option	Description
SOURCE	The modulation source, such as LFO 1 for example. Note that certain modulation sources are only available to set using the modulation matrix (it's not possible to drag the source to the control). These mod sources include: • Active Voices, derived by dividing the number of active voices by the maximum number of voices allowed by the **POLY** count setting. • Note-On Alt. 1/2, which switches between 0 and 1 at each note-on. The state of **Note-On Alt 2** is the inverse of **Note-On Alt 1**. • Note-On Rand (Discrete). The **Note-On Rand 1/2** sources send the same value to each assigned destination at note-on. Destinations assigned to **Note-On Rand (Discrete)** each get a different value at note-on. • Note-on Rand 1 and 2, which are two separate random numbers generated on a note on event, in case you need two different random values for each note on event. • Oscillators, allowing you to use the output of any oscillator (including **NOISE** and **SUB**) as a mod source. • Release Velocity • Voice Index, updated at note-on with a value derived from the current index of the Voice panel divided by the number of active steps. • Voice Mod 1/2, updated at note-on with the value of the current step in the Voice panel for Mod 1/2. • Expression/MPE X/Y/Z, which are Note Expression or MIDI Polyphonic Expression (MPE) axes. • Filters, allowing you to use the output of either filter as a mod source.

Option	Description
SOURCE (cont.)	• Mod Wheel • Aftertouch (channel pressure) • Poly Aftertouch • Pitch Bend • Fixed (not really a modulation source, but you can use this to allow a modulation assignment to get a fixed value with slider depth control, should you want that for some reason).
CRV (Curve)	Scale the mod source to respond in a non-linear fashion. A 50% value indicates a linear setting. When the curve is gray, it is bypassed. Double-click the curve (or right-click and choose **Editable Curve** in the menu) to display a curve editor that you can use to define the remapping curve. Draw the appropriate curve, and use the **RISE** and **FALL** smoothing controls to act as a slew limiter on the source.
AMOUNT	The modulation depth, set using a bi-directional slider. Moving the slider to the left sets a negative value, causing the mod source output to be inverted before heading to its destination. Cmd-clicking (macOS) or Ctrl-clicking (Windows) resets the value to the (zero) default.
POL (Polarity)	Specify whether the modulation is unidirectional or bi-directional. You can achieve similar sonic results with either, it depends on whether you prefer the destination control (the knob position) to be at the beginning (unidirectional) or the middle (bi-directional).
DESTINATION	The modulation destination. Use the pop-up menu to select the destination (the parameter modified by the mod source).
OUT	A graph that shows the shape of the modulation output.
AUX SOURCE	A secondary source to determine the amount of modulation. Use the pop-up menu to select the aux source.
INV	Invert the auxiliary source signal. Normally, the two modulation sources are “multiplied” together so that one is scaling the other. For instance, if **LFO 1** is the **SOURCE**, and **ModWheel** is the auxiliary source (**AUX SOURCE**), the **LFO 1** influence is inaudible unless the **ModWheel** is raised above zero. This is the default setting. When set to inverse, the setting is the same as above except that the secondary source (**AUX SOURCE**) is value-inverted. For instance, in the above example, there would be no modulation if the **ModWheel** is at maximum, and there would be full modulation if the **ModWheel** is at minimum.

Option	Description
CRV (Curve)	Scale the auxiliary source to behave in a non-linear fashion. A setting of 50% designated linear. The curve is bypassed when it is gray.
OUTPUT	Scale the final modulation output, allowing for fine tuning.
	Select to bypass this row of the modulation matrix (causing it now to have no effect). The button changes  to show that it is enabled.
	Click to remove this row of the modulation matrix. This removes the modulation assignment from the patch.

Click the  button (near the top left) to expand the matrix to show more rows. Alternatively, press Option-F (macOS)/Alt-F (Windows) to expand the view.

Modulation Matrix (Expanded)

Similarly, click the  button to revert the modulation matrix down to the default size. Similarly, press Option-F (macOS)/Alt-F (Windows) to revert the view to the original size.

Moving Modulations in the Matrix

To move a modulation row to another location, click and drag the modulation handle  to another location.

Moving a Modulation

Bypassing a Modulation

Every row features a bypass button  that allows you to easily bypass the modulation.

Clicking the button bypasses the modulation in the signal routing, indicated by the button updating  to show that the row is being bypassed.

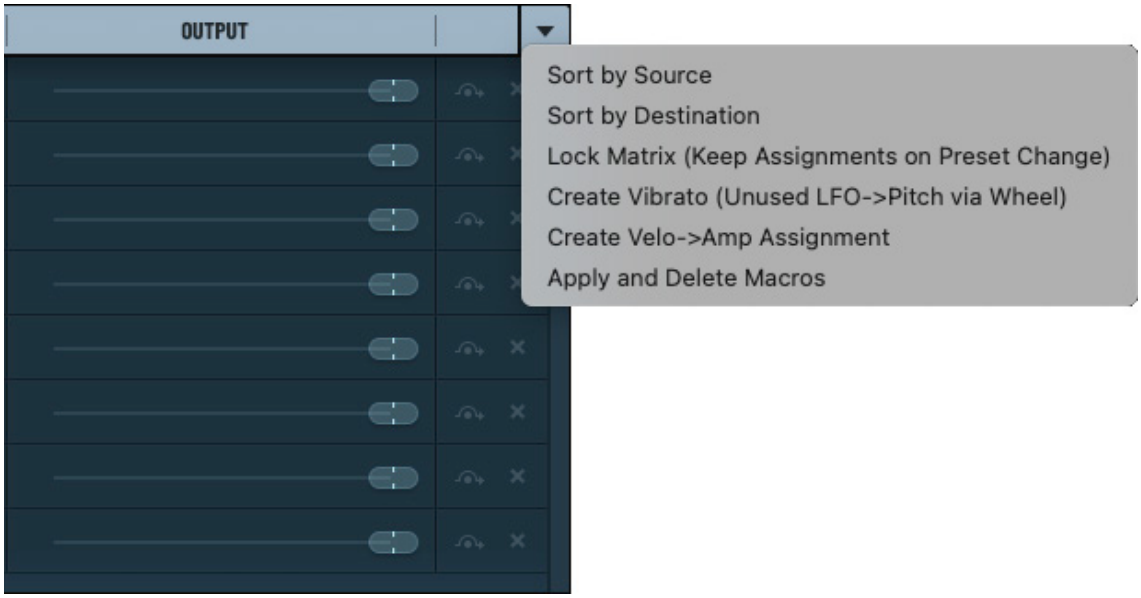
Removing a Modulation

You can remove a modulation assignment directly from the modulation matrix. To do so, click the  button for the corresponding modulation.

Performing Matrix Operations

You can perform a range of additional operations on the modulation matrix including sorting the matrix, locking the modulations (even when you change presets), and creating specialized new modulation assignments.

Click the ▼ button (near the top right) and choose an option in the menu that appears.



Modulation Matrix Menu

The following table describes the operations you can perform:

Operation	Description
Sort by Source	Sort the modulation matrix (ascending) by the **SOURCE** column.
Sort by Destination	Sort the modulation matrix (ascending) by the **DESTINATION** column.
Lock Matrix (Keep Assignments on Preset Change)	Lock the modulation matrix, which keeps the modulation assignments when you change presets or initialize a new preset.
Create Vibrato (Unused LFO->Pitch via Wheel)	Create a new modulation that maps the next available **LFO** to "Main Tuning" using the **Mod Wheel**.
Create Velo->Amp Assignment	Create a new modulation that maps **VELO** to the Amp.
Apply and Delete Macros	"Bake" the macro adjustments into the current preset. Specifically, for any parameter assigned to a macro, update the current value of the parameter to include any offset from the macro. Then remove all modulation assignments for all macros from the modulation matrix.

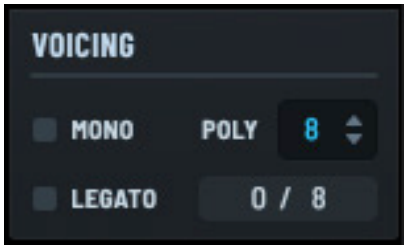
Setting Voicing and Portamento

Voicing is the simultaneous vertical placement of notes in relation to each other. Portamento is a pitch sliding from one note to another.

Serum Voicing and Portamento

Voicing Settings

The **VOICING** section contains controls that allow you to change how Serum behaves when multiple notes play at once.



Serum Voicing Controls

Mono

Use the **MONO** switch to enable monophonic mode, causing Serum to only allow one active note at a time. If a new note is pressed while a note is already playing, the earlier note is interrupted (technically the note is re-pitched to the new pitch).

Legato

LEGATO is only audible when **MONO** is enabled. When a monophonic voice is interrupted, the state of the LEGATO switch determines whether the envelopes/LFOs retrigger.

When **LEGATO** is enabled with **MONO** off, Serum behaves paraphonically (especially noticeable when **ENV 2** is modulating a filter). This means you can control envelope retriggering modes for the effects assignments here also.

When **LEGATO** is enabled, the envelopes do not retrigger, which results in a smooth change to the new note. However, sometimes you want the envelopes to retrigger so each note has the same definition; in these cases, set **LEGATO** to off.

Note that you can set individual envelopes to behave opposite to the **LEGATO** setting. See “Inverting the Legato Setting” on page 185 for more information.

Poly

Use the **POLY** (polyphony) setting to specify the number of simultaneous notes that can be played. Sometimes, often for CPU reasons, you might want to place a limit on the simultaneous notes.

For instance if you are sending a flurry of notes (arpeggio) to a patch that has a long release (say 10 seconds), this could end up producing a large number of voices, potentially overloading your CPU.

Typically, eight voices is enough and 16 is generally considered a lot. Note that this control is disabled when **MONO** is enabled.

Polyphony Count

This shows the number of voices playing against the total number of voices allowed. In the example, “0 / 1” indicates that 0 out of a total possible number of 1 voices is playing.

If a note is played, the display would change to “1 / 1”. Similarly, if you enable a second oscillator, the display would show “0 / 2” when no notes are playing. In other words, Serum totals the number of voices you have active in a patch.

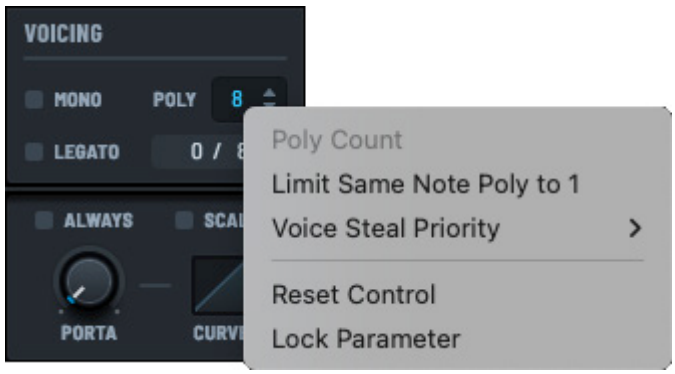
Single note voice count in Serum includes unison voices and, if one or more oscillators are set to **GRANULAR** mode, grain count as well.

Limiting Polyphony

You can choose to limit same note polyphony. This means that if a note (for example, note number 60) is already sounding, pressing the same note again won't trigger another layer of the same note. This prevents "stacking" multiple instances of the same note, ensuring clarity in the sound and avoiding unintended overlaps or muddiness.

Right-click the **POLY** field and choose **Limit Same Note Poly to 1** in the context menu.

This type of polyphony control is useful for instruments or patches that need to remain clean, such as bass lines, monophonic synths, or drum kits where a single sound per note is preferred.



Serum Voicing Controls

Voice Steal Priority

In cases when the number of voices exceeds the polyphony setting, you can specify which active note should be terminated when a new note is triggered.

Right-click the **POLY** field and use the the **Voice Steal Priority** submenu to choose which voices get priority (are not stolen) when Serum performs voice stealing, from among the following:

Option	Description
Newest	Terminate the earliest played note.
Oldest	Terminate the most recently played note.
Highest	Terminate the lowest-pitched note.
Lowest	Terminate the highest-pitched note.
Velocity	Note with the lowest input velocity.

Portamento Settings

Portamento creates a slow glide/pitch bend from one note to another. It is most commonly used (and is most useful) when **MONO** is also enabled.

In this case, when one note plays and then another, the pitch slowly changes from the first note to the second.

Serum Portamento Controls

Porta

Use the **PORTA** knob to control the rate of glide from one note to another.

Curve

Use the portamento **CURVE** to adjust the contour of glide from one note to another. If set convex (typical use), the note pitch departs the beginning pitch quickly and slows down as it nears the destination note frequency.

If set concave (dragged down below half), the opposite is true. In this case, the pitch slowly departs the source pitch and later rapidly arrives at the destination pitch.

Always

When **ALWAYS** is activated, the portamento occurs on a new note even if no note is currently playing. When disabled, a note must be held for portamento to occur on the (second) note.

Scaled

The **SCALED** switch is potentially useful for melodic leads when you want a less noticeable portamento on short intervals.

When activated, the portamento rate is adjusted based on the distance between the source and destination pitches. For example, if the portamento is a glide between two notes one octave apart, the **PORTA** knob time value is used.

If the portamento is a glide between notes less than one octave apart, the time is faster and vice-versa (notes larger than an octave are progressively slower).

Using Clips

Serum features a versatile and fun-to-use **CLIP** module that you can use to create, fine tune, and play a series of MIDI clips, directly within Serum.

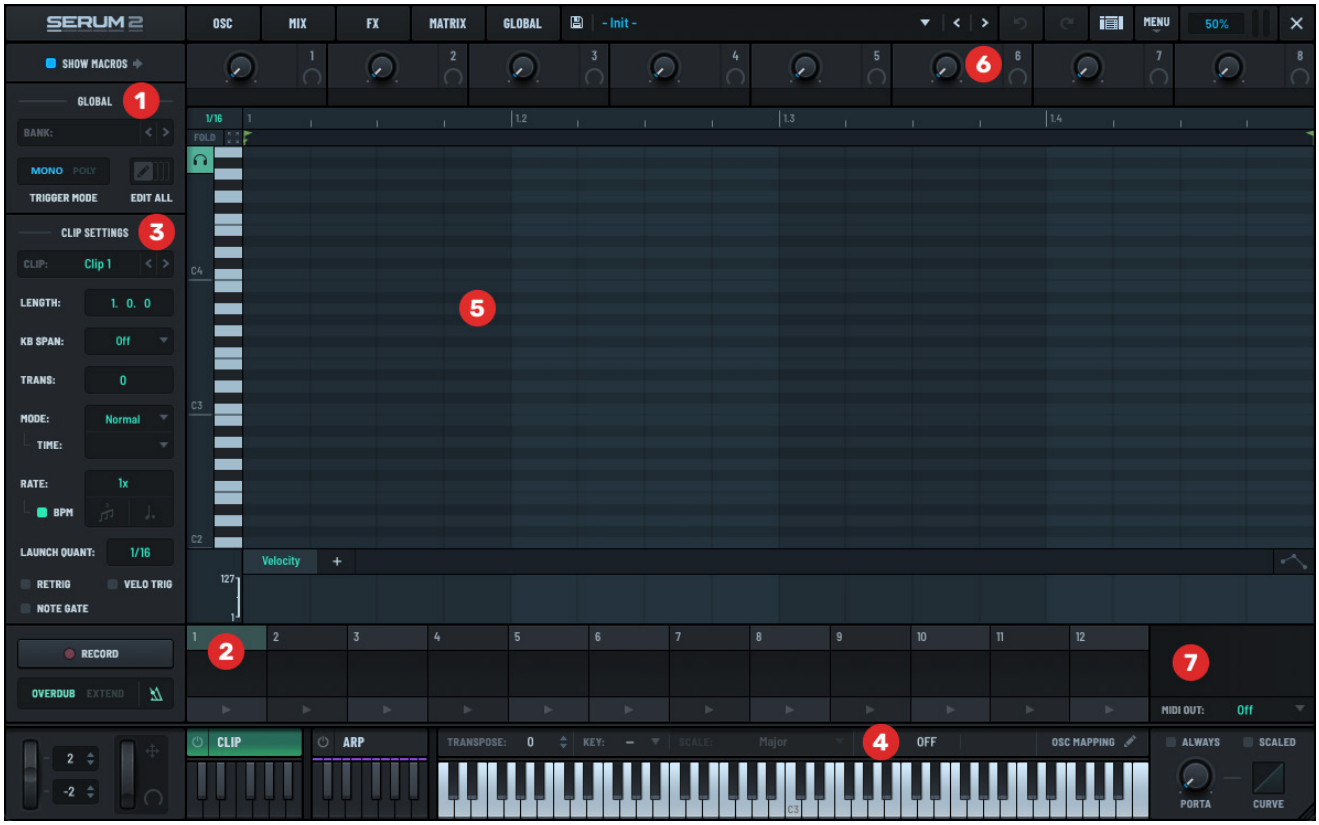
A MIDI clip contains notes and controller data for playing your current Serum patch. Each clip specifies the note pitch, length, position, and dynamics (velocity).

Click the CLIP button to access the **CLIP** module.

Serum Clip Mode

Exploring the CLIP Module

The **CLIP** module includes several panes to help you navigate and access the various features.



Clips Interface

Use the **GLOBAL** pane to load a clip bank preset. 1 Alternatively, if you already have clip slots already populated, you can initialize the **CLIP** module to start over.

Select a clip slot 2 from among the 12 available.

Use the **CLIP SETTINGS** pane 3 to set clip parameters, including the clip length, transposition, mode, rate, launch quantization, and more.

Optionally set the key and scale 4 to have Serum quantize the pitch of notes generated by the **CLIP** or **ARP** module, or from MIDI input..

Populate the piano roll 5 with appropriate notes. You can “click in” the notes or record a live performance.

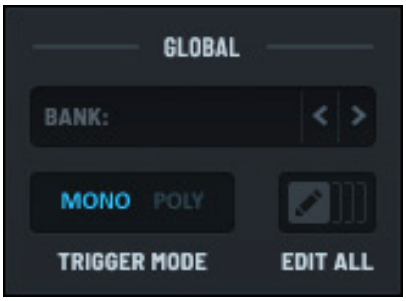
Optionally, assign macros 6 to a certain clip settings, allowing you to change one or more sound parameters using a single knob.

Specify whether Serum should output the MIDI data it generates internally. 7

Setting Global Parameters

You can use the **GLOBAL** pane to set parameters that affect the general operation of the **CLIP** module.

For example, you can use the **GLOBAL** pane to load a preset bank, set the trigger mode, and configure whether parameter edits apply to all clips.

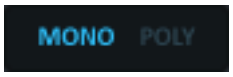


Global Clip Settings

To load a factory-supplied or user-defined bank, click the **BANK** field and choose a bank using the menu that appears.

The bank loads and populates the clips and associated settings.

Bank Menu

Use the **TRIGGER MODE** switch 

to set the **CLIP** mode to **MONO** or **POLY**. When set to **MONO**, only a single clip can play at one time. When set to **POLY**, you can trigger multiple clips to play simultaneously.

Enable the **EDIT ALL** switch  to have your parameter edits apply to all clips. When disabled, any changes to the clip settings (such as **TRANSPOSE** or **RATE**) apply only to the currently-selected clip.

When enabled, changes to any parameter become immediately effective for all clips.

Hold the Option (macOS) or Alt (Windows) key when editing a clip parameter to apply the change to all clips.

Bank Loaded

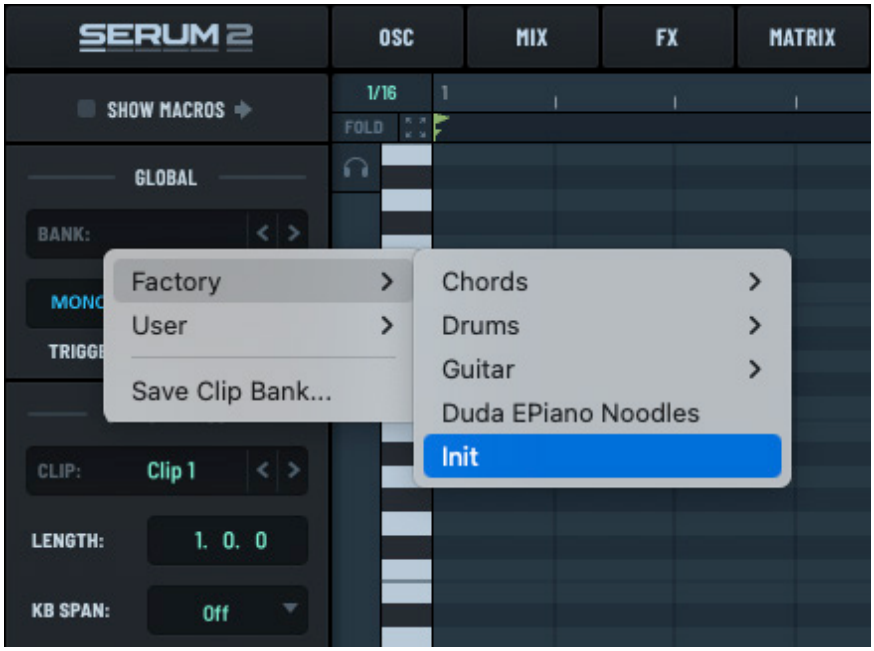
Creating a New Clip Bank

In addition to exploring some of the factory-supplied clip banks, you can initialize the clip bank to create your own from scratch.

Note: Initializing the clip bank only affects clip bank settings. This does not change any of the other sound design settings you've configured (including oscillators, filters, arpeggiators, and so on).

Click the **BANK** menu and choose **Init** in the menu.

This initializes the **CLIP** module to the default settings and sets the stage for you to create a custom clip bank.

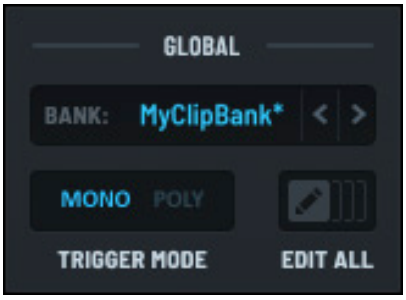


Creating a New Clip Bank

Saving Clip Banks

After creating a new clip bank, or editing an existing bank, you can save the entire clip configuration as a new bank.

Note that clip bank presets that have been modified display as asterisk (*) after the name.



Clip Bank Modified

Click the **BANK** field and choose **Save Clip Bank** in the menu that appears.

A dialog appears allowing you to type the clip bank name.

By default, the clip bank is saved in a standard location so that Serum can easily find it later.

Note that if you try to save a modified clip bank that has already been saved, a dialog appears allowing you to give your recently-modified clip bank a new name.

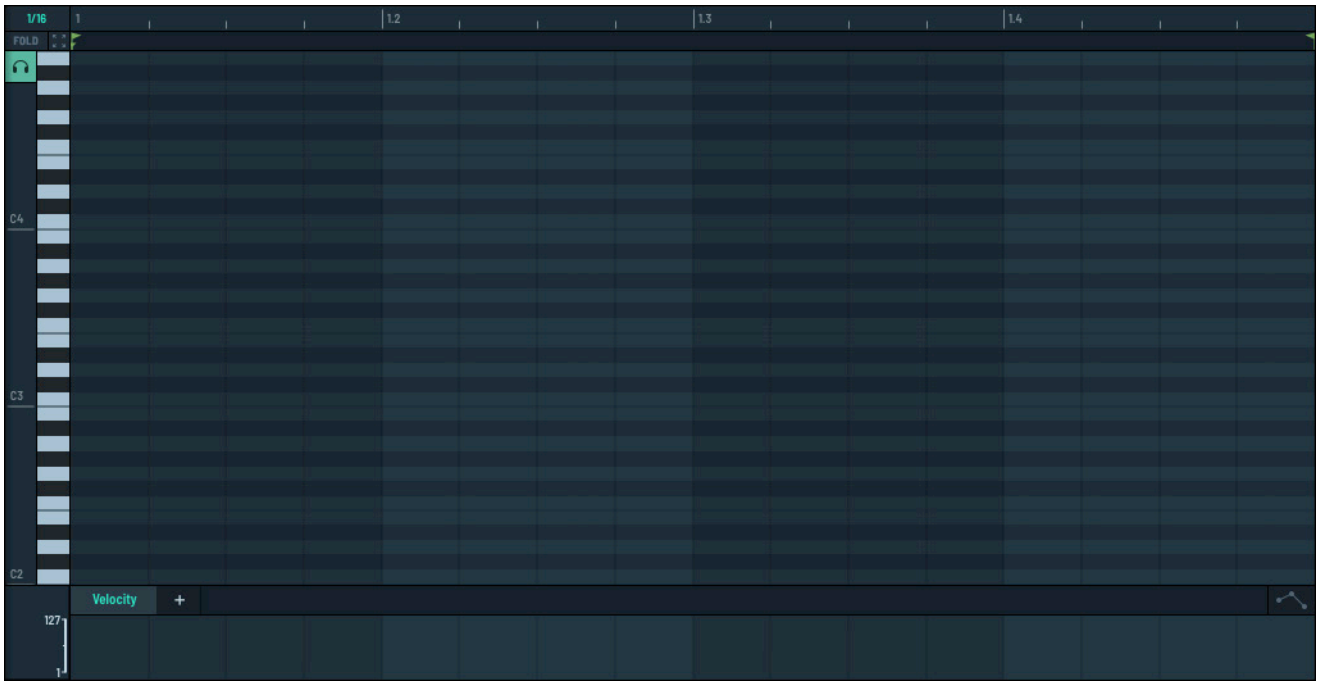
This allows you to freely experiment with changes, saving your work in steps as you progress.

Save Clip Bank

If you would instead prefer to overwrite the existing clip bank, you can choose the existing file name in the dialog.

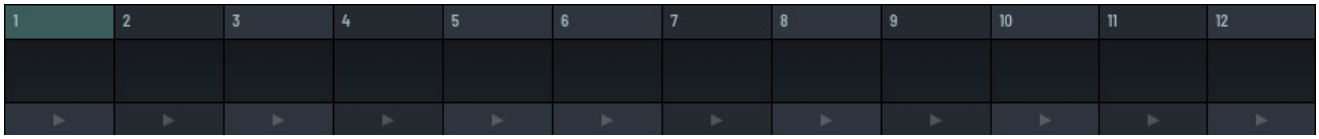
Working with the Piano Roll

The **CLIP** module offers a familiar piano roll that you can use to create MIDI clips. You can use the piano roll to create sequences to go with your sound presets, capturing context along with your sound design, and offering inspiration to others who might use your preset.



Piano Roll

Begin by selecting a clip from among the 12 available clip slots.



Clip Slots

Double-click to add a new note. The latest note you added appears in orange.

Adding Notes to the Piano Roll

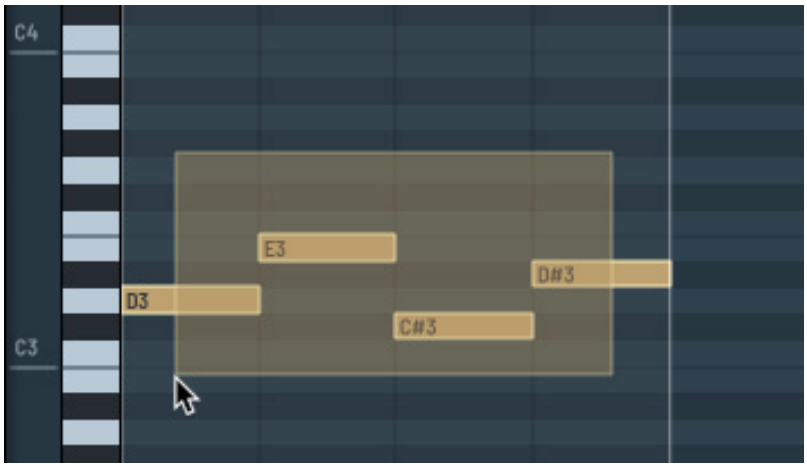
Click and drag across the piano roll to select notes.

Hold the Shift key while dragging to select additional notes.

You can also select multiple notes by Shift-clicking one note after the other.

Click and drag across the piano roll *without including any notes* to select a time range.

You can then copy/cut and paste the time range. Notes that overlap the edge of the time range are split.



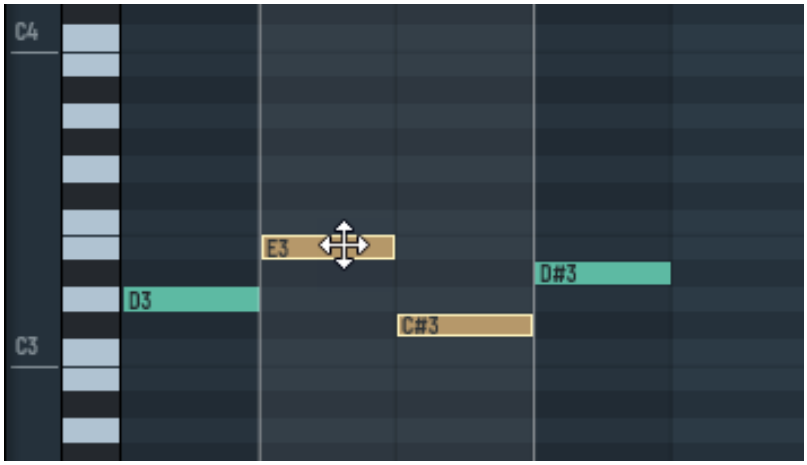
Selecting Notes

Important: When pasting a time range, unlike pasting notes, all existing notes in the destination range are overwritten.

Click and drag a note to move it to another location.

When multiple notes are selected, moving any note moves all selected notes.

You can also move and adjust selected notes using the arrow keys on your keyboard.



Moving a Note

Arrow	Description
↑	Move the selected note or notes up one semitone.
↓	Move the selected note or notes down one semitone.
←	Move the selected note or notes one position to the right in the grid.
→	Move the selected note or notes one position to the left in the grid.
Shift ↑	Move the selected note or notes up one octave.
Shift ↓	Move the selected note or notes down one octave.
Shift ←	Shorten the selected note or notes.
Shift →	Lengthen the selected note or notes.

You can perform a series of other operations on notes, as described in the following table. In all cases, select the appropriate notes and right-click to choose the operation. In addition, in most cases, there's a keyboard shortcut to go with the operation.

Action	Keyboard Shortcut	Description
Cut	Cmd-X/Ctrl-X	Cut the selected notes.
Copy	Cmd-C/Ctrl-C	Copy the selected notes.
Paste	Cmd-V/Ctrl-V	Paste the selected notes at the timeline.
Duplicate	Cmd-D/Ctrl-D	Duplicate the selected notes at the timeline. The timeline is set at the end of your selection. So, in practical terms, end your selection where you would like the duplicate notes to start.
Delete	Backspace	Delete the selected notes.
Chop	Cmd-U/Ctrl-U	Chop the selected notes.

Action	Keyboard Shortcut	Description
Conform to Scale	Cmd-K/Ctrl-K	Move the selected notes to the selected scale (if necessary). You can select the scale in the **Keyboard** pane.
Legato	Cmd-L/Ctrl-L	Apply legato to the selected notes, extending each note to smoothly connect to the start of the next.
Mute	0 (zero)	Mute the selected notes. The muted notes appear in gray.
Quantize	Cmd-Q/Ctrl-Q	Quantize the selected notes to the current grid setting.
Reverse	Cmd-R/Ctrl-R	Reverse the order of selected notes.
Scale Time 50%	/	Halve the time and duration of the selected notes without changing the clip length.
Scale Time 200%	*	Double the time and duration of the selected notes without changing the clip length.
Double Entire clip	Cmd-E/Ctrl-E	Append a duplicate of the clip, copying the notes and doubling the clip length.
Scale Entire clip 50%	(no keyboard shortcut)	Halve the clip length and halve each note's duration. For example, if the clip is currently two bars long with 8th notes, this decreases the clip to one bar and halves the note durations to 16th notes.
Scale Entire clip 200%	(no keyboard shortcut)	Double the clip length and double each note's duration. For example, if the clip is currently one bar long with 16th notes, this increases the clip to two bars and doubles the note durations to 8th notes.
Select All	Cmd-A/Ctrl-A	Select all notes in the clip.

To delete a single note, double-click on the note. To delete multiple notes, select the notes and press the **Delete** key (macOS) or **Backspace** key (Windows).

Normally, when you select a note, no sound is played. You can have the note sound when selected by toggling the headphones on.  The button changes to show that sound is enabled. 

Note that key commands on the piano roll only work if the note grid or automation lane has keyboard focus. You can give either focus by clicking on the corresponding background.

A keyboard icon appears near the top right to show which pane has keyboard focus. 

 The **Keyboard shortcuts** preference needs to be set to **ON** in the **GLOBAL** pane to enable key commands to work.

Setting the Grid Size

By default, Serum uses a grid size of 1/16 notes in the piano roll. You can change this by clicking the grid size button 

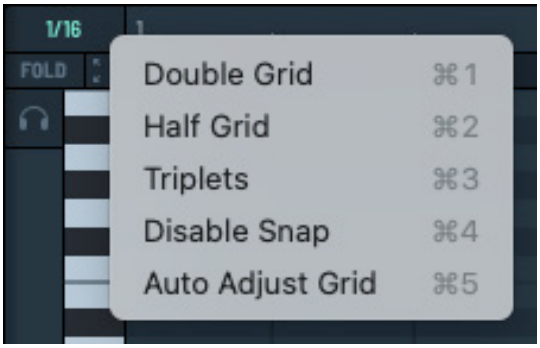
and dragging up or down to choose a new setting.

The grid updates as you drag the control. This gives you incredible creative flexibility in terms of rhythm and precision. A smaller grid size allows for more precise note placement, which is useful when fine-tuning timing or adding nuanced variations.

For example, you could use a larger grid size (such as quarter notes and eighth notes) to create standard rhythms like 4/4 beats or simple melodic lines. Smaller grid sizes (such as 16th notes, 32nd notes, or even smaller) enable more detailed rhythmic work, such as rapid hi-hat patterns, rolls, or syncopated melodies.

Right-click the grid size button and choose an option to quickly adjust the grid.

Alternatively, use the corresponding keyboard shortcut.

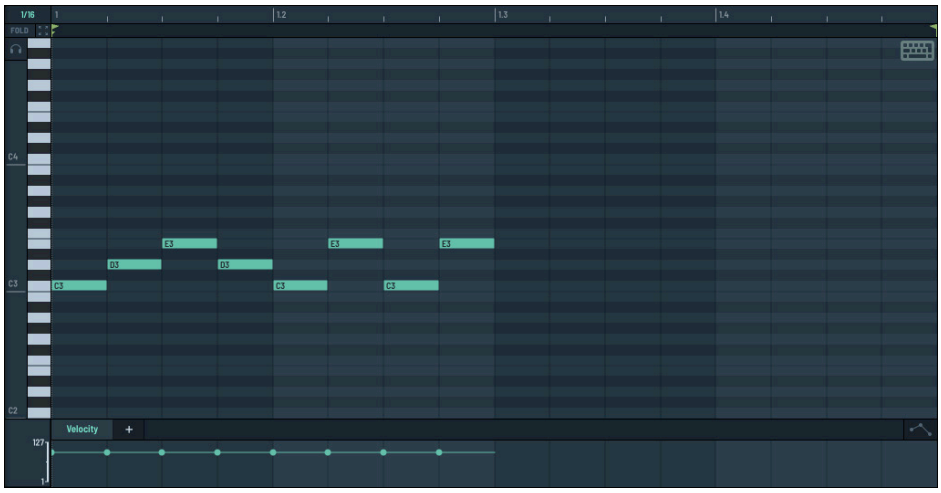


Grid Menu

Zooming the Piano Roll

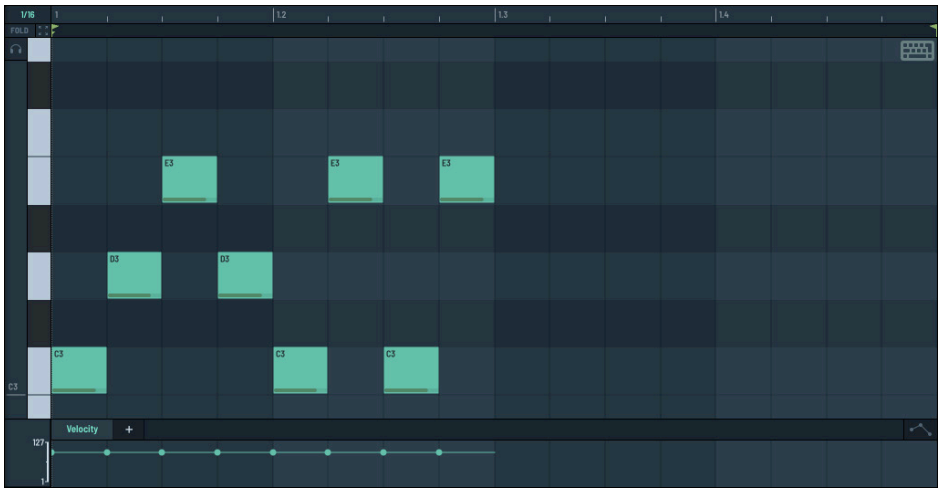
After adding notes to the piano roll, you can zoom the piano roll display to allow you to focus on the specific notes.

For example, suppose that you added the following notes.



Piano Roll (default zoom)

Clicking the  button zooms the display, allowing you to focus on the notes you entered.



Piano Roll Zoomed

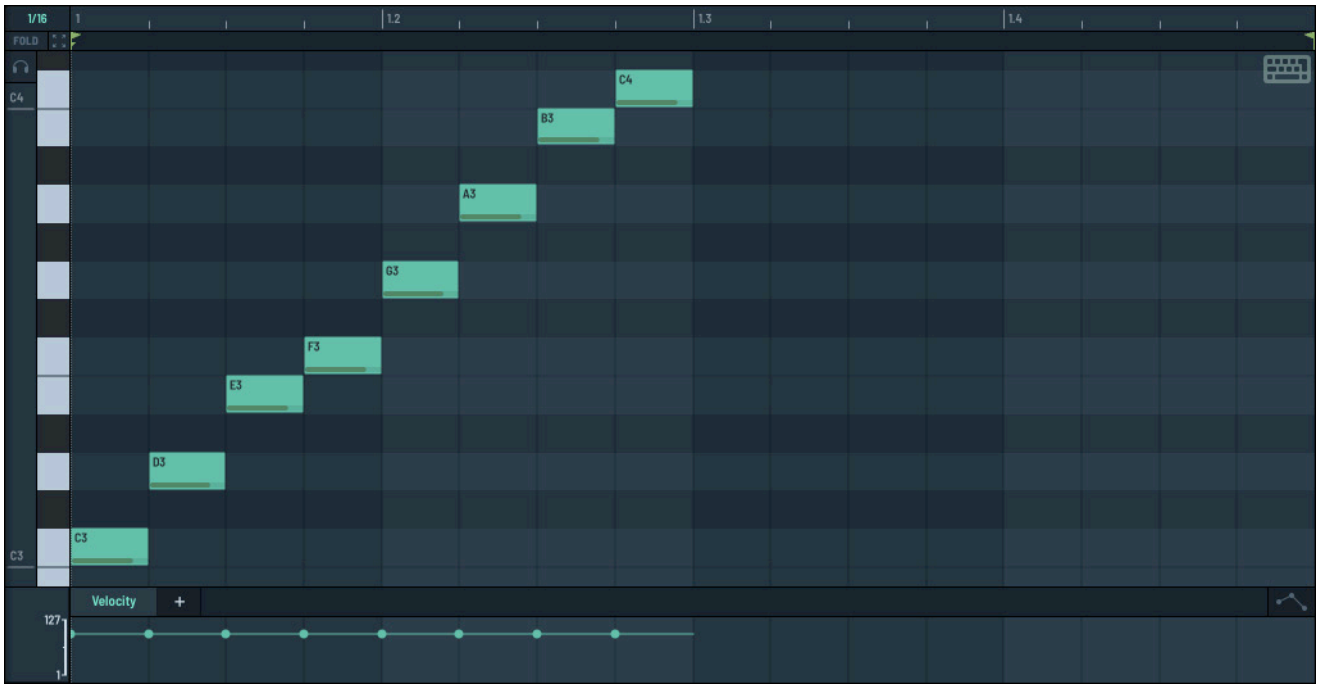
In addition, you can do the following:

- Hold Cmd-Option (macOS) or Ctrl-Alt (Windows) and drag the piano roll background to scroll in any direction.
- Hold Shift-Option or Shift-Alt and drag the piano roll background to zoom in any direction.
- Hold Shift-Cmd or Shift-Ctrl and drag on the note grid to drag a (black) marquee. The grid zooms to the selected area after releasing the mouse button.
- Use the mouse wheel to scroll the piano roll vertically or, with Shift held, horizontally.
- Hold the Option or Alt key and use the mouse wheel to zoom the piano roll vertically. Hold the Cmd or Ctrl key with the mouse wheel to zoom horizontally.

  You can also use the mouse wheel while hovering over the timeline (near the top) to zoom horizontally.

Folding the Piano Roll

By default, the piano roll displays all notes in the chromatic scale, giving you the flexibility to add any note to the clip.



Piano Roll with Notes Added

You can “fold” the notes on the piano roll to hide unused notes, making the piano roll appear cleaner and less cluttered.

Click the FOLD button to fold the piano roll to show only the notes that you are currently using.

The button highlights FOLD to indicate that the piano roll is folded.

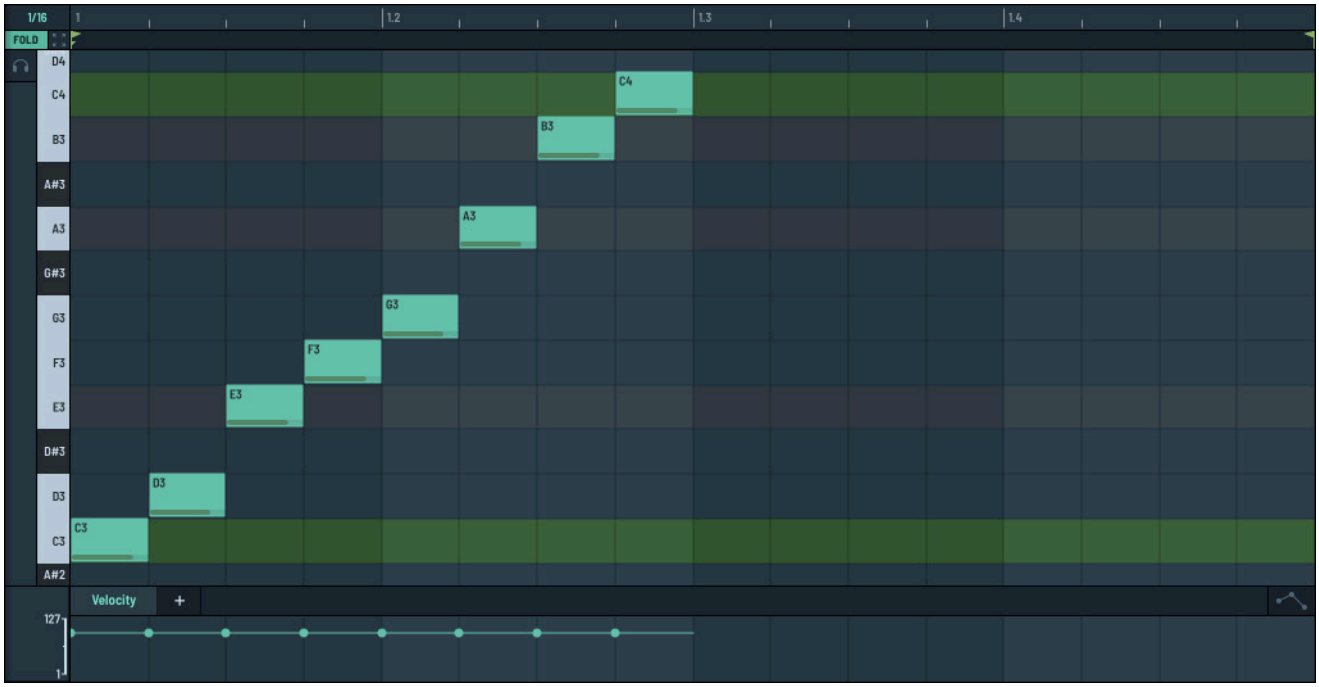
Piano Roll Folded

The fold feature becomes especially useful when a key and scale are selected. In this case, the piano roll collapses to display only the notes within the scale.

Hold the Cmd (macOS) or Ctrl (Windows) key when clicking the FOLD button to show notes in the scale as well as used notes.

This makes it easy to distinguish in-scale notes (blue background) from out-of-scale notes (gray background).

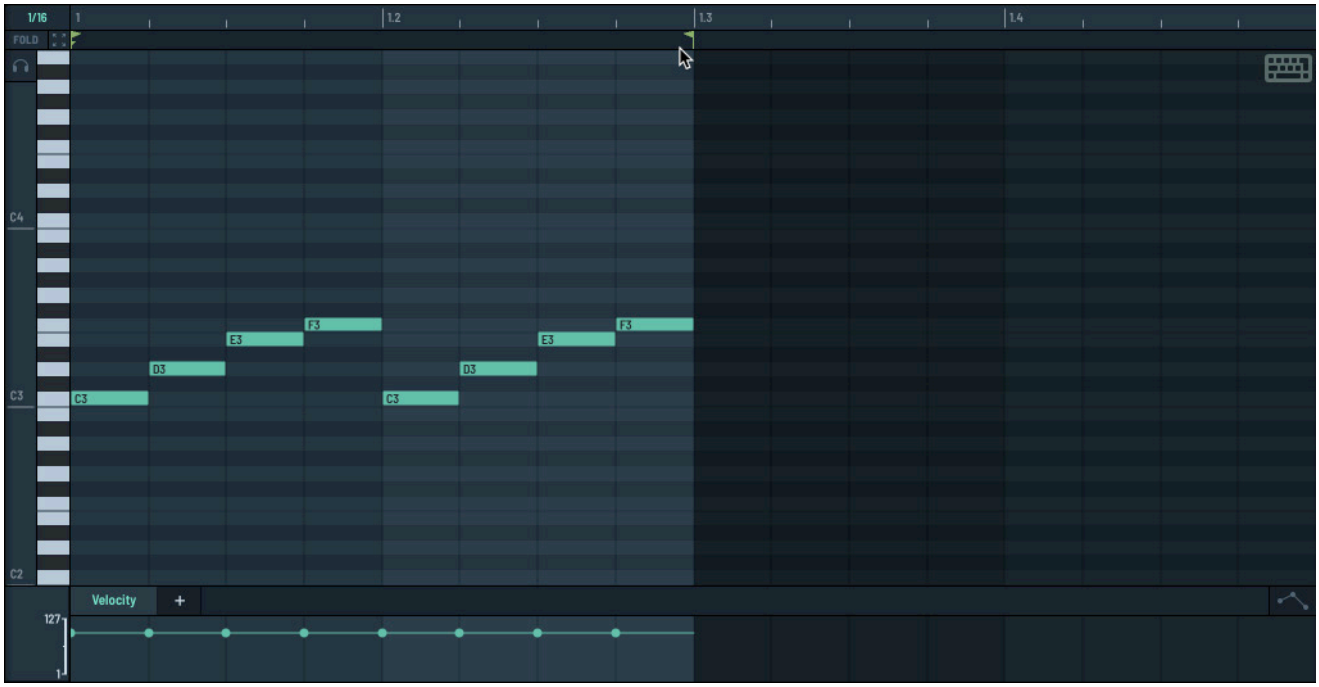
For example, the following shows notes in the C Major scale when the C Minor scale is selected and the piano roll is folded.



Piano Roll Folded (in relation to the selected scale)

Managing the Clip Length

To change the clip length in the piano roll, drag the right flag to the appropriate location.

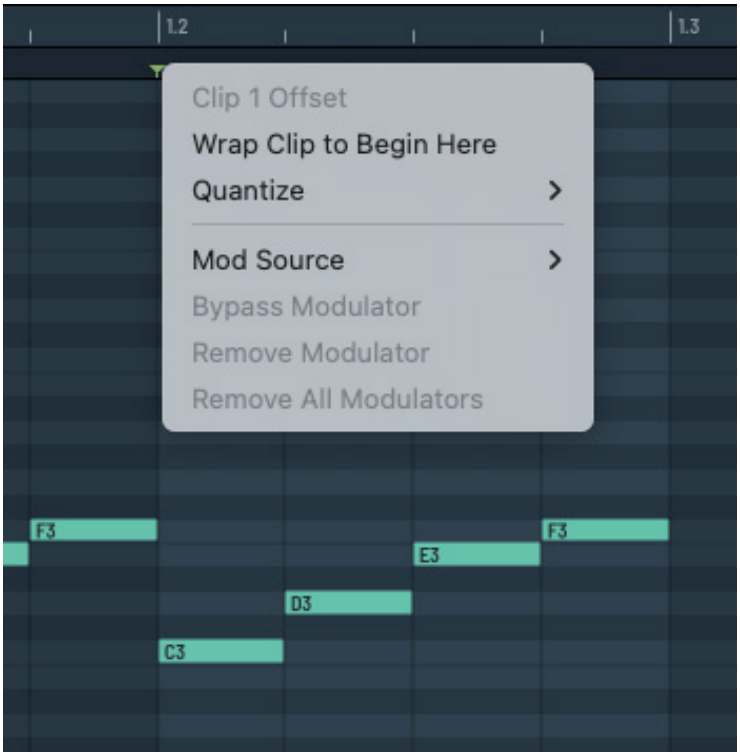


Changing the Clip Length

Similarly, to change the start of the clip, drag the left flag to the appropriate location.

To set the start offset marker, click and drag the marker from the left. The marker determines where playback starts when the clip is triggered.

Right-click the marker to display the context menu.



Clip Marker Menu

The following table describes the available options:

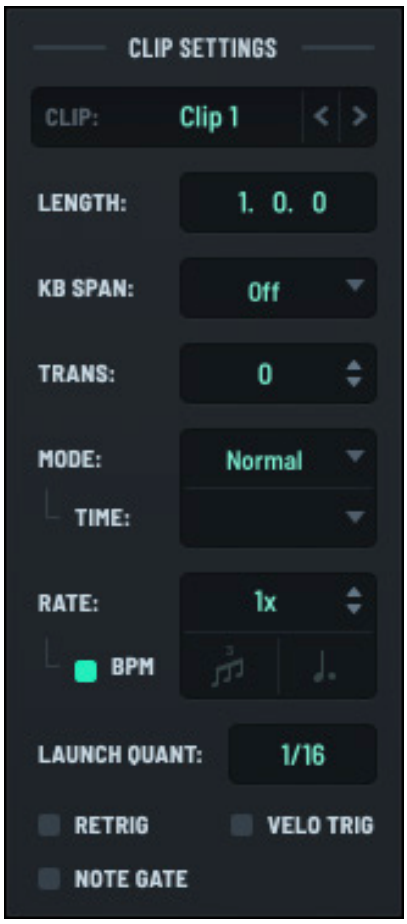
Option	Description
Wrap Clip to Begin Here	Reset the start offset marker to the same position as the start marker moving all notes and automation with it. Any events that came before the start offset marker are wrapped around to the end of the clip.
Quantize	Quantize the position of the start offset marker when modulation is applied to the marker. Also, determine the time division by which each successive key will offset the playback start position when **KB Span** is set to **Offset** (see Configuring Clip Settings).

Configuring Clip Settings

In addition to adding notes to the piano roll and defining note properties (such as velocity and expression), you can also configure a variety of parameters associated with a clip.

Use the **CLIP** field to load, save, or rename a clip.

CLIP Field



Clip Settings

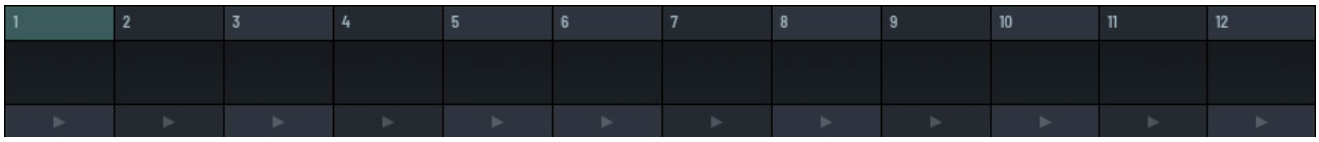
The following table describes the remaining parameters you can set:

Field	Description
LENGTH	The clip length in bars, beats, and 16th notes.
KB SPAN	Specifies whether the clip is triggered by MIDI notes, from among the following options: • Mono and Poly — Play the clip transposed according to the played note (relative to C3). • Offset — Play the clip starting from a different position for each note, determined by the offset quantize setting (accessed by right-clicking the start offset marker). • Off — Turn the setting off.
TRANS	The transpose amount for the clip.
MODE	The play mode for the clip, specifying how the playhead moves. Note the following about the available options: • Random — Playback position jumps between slices at random, • Rand.No Dup — As above, but never play the same slice twice in a row. • Rand.Start — The playback position jumps back to a slice each time the playhead reaches the end of the clip. • Rand.End — The clip plays for a random number of slices before jumping back to the start, • Static — When the clip is triggered, only events at the start position are sent and the playhead does not move. You can use this to set each clip to trigger a different chord or a different set of macro values, for example. **Note:** For the random settings, you can also set the associated time, which divides the clip into "slices" of the selected length.
RATE	The speed at which the clip plays.
LAUNCH QUANT	The interval at which a triggered clip syncs to the clock/DAW.
RETRIG	Specifies whether a clip restarts when triggered, or if the playhead immediately syncs to the clock/DAW.
VELO TRIG	Specifies whether note velocities in a triggered clip are scaled by the incoming note velocity.
NOTE GATE	Specifies whether a "note off" message stops playing a clip when triggered by input MIDI notes.

Note: If you enabled the **EDIT ALL** switch  in the **GLOBAL** pane, any changes you make to the clip settings apply to all clips. When disabled, any changes to the clip settings apply only to the currently-selected clip.

Managing Clips

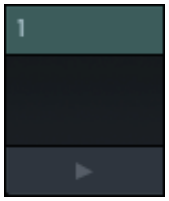
Serum features 12 clips per clip bank. As you'll see in the next section, this makes it particularly easy and convenient to trigger these clips using either a MIDI keyboard or even your computer keyboard.



Clip Slots

Click a clip to select it. The current clip is highlighted, and the contents of the clip appear in the piano roll.

To perform operations on the clip, right-click the clip and choose an option in the menu that appears.



Current Clip (Highlighted)

The following table describes the available operations:

Operation	Description
Rename Clip	By default, clips are named **1** to **12** (from left to right). You can rename any clip to better match the clip contents. After choosing **Rename Clip** in the menu, a text area appears in the clip header. Type the new clip name and press the **Enter** key.
Copy Clip	Copy the clip contents (notes) and settings (including the clip name) to Serum's internal clipboard. Right-click the source clip and choose **Copy Clip** in the menu. Right-click the destination clip and choose **Paste Clip** in the menu. Note that it is not possible to copy a clip from one instance of Serum to another.
Paste Clip	Paste the most-recently copied clip into the target clip.
Erase Clip	Erase the contents of the selected clip.
Set as Preview Clip	Set the clip as the MIDI sequence to play when someone previews the saved preset in the Serum presets browser.

When designing sounds, both for yourself and others, it's helpful to set a preview clip. This gives preset users a quick sense of the intent of the preset. Otherwise, Serum plays a designated "fallback" clip, which can quickly get repetitive when previewing a lot of presets.

Plus, by including one or more clips with your preset, you offer yourself and others a bonus when you choose to use the sound in the future!

Triggering Clips

You can trigger your clips in multiple ways, allowing clips to be a versatile part of your sound design process.

To trigger a clip while in **CLIP** mode, click the  button for the corresponding slot.

The clip starts playing and the button changes to a stop button to indicate which clip is currently playing.

To stop a clip, click the  button (for the corresponding clip).

Clip Slot with Play Button

You can make changes to any aspect of the clip while it is playing, including modifying notes, changing clip settings, and more. You can also change the underlying sound design as the clip plays, providing a versatile way to hear your sound evolve in context.

You can also trigger clips using the **CLIP** keyboard.

This is helpful when you are not in **CLIP** mode and instead tweaking the oscillators, mixing, applying effects, or adjusting the modulation matrix.

Clip Keyboard

Notice that the clip slots (in CLIP mode) are arranged similar to a keyboard. This makes it easy to logically map the clips slots to the CLIP keyboard.

Press a key to (continuously) play the corresponding clip. The key highlights in green.

Notice a small green indicator in the lower right corner. This flashes in rhythm with the notes played when you launch a clip.

Clip Keyboard (Playing)

If you launch a clip without notes, the key highlights in green but the rhythm indicator is not present.

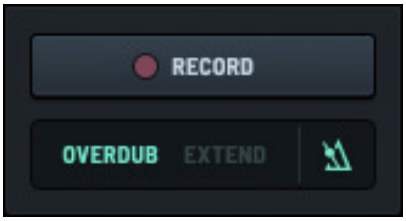
Press the same key to stop the clip, or press another key to launch the corresponding clip.

Important: If you launch a clip that you know has notes but don't hear any sound, verify that clip playback  is enabled.

Recording Your Clips

In addition to “clicking in” notes using a mouse, you can also create clips by recording a performance using a MIDI controller.

Begin by choosing a destination clip slot and then select either **OVERDUB** or **EXTEND** as the recording mode.



Recording Clips

OVERDUB loops continuously using the start and end markers that you set, allowing you to add new notes during each pass. **EXTEND**, in contrast, extends the piano roll timeline until you either stop recording or switch to **OVERDUB** mode. You can change between the two modes at any time.

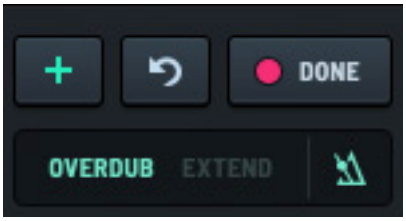
You can activate or deactivate the metronome  as needed. Click the 

button to start recording.

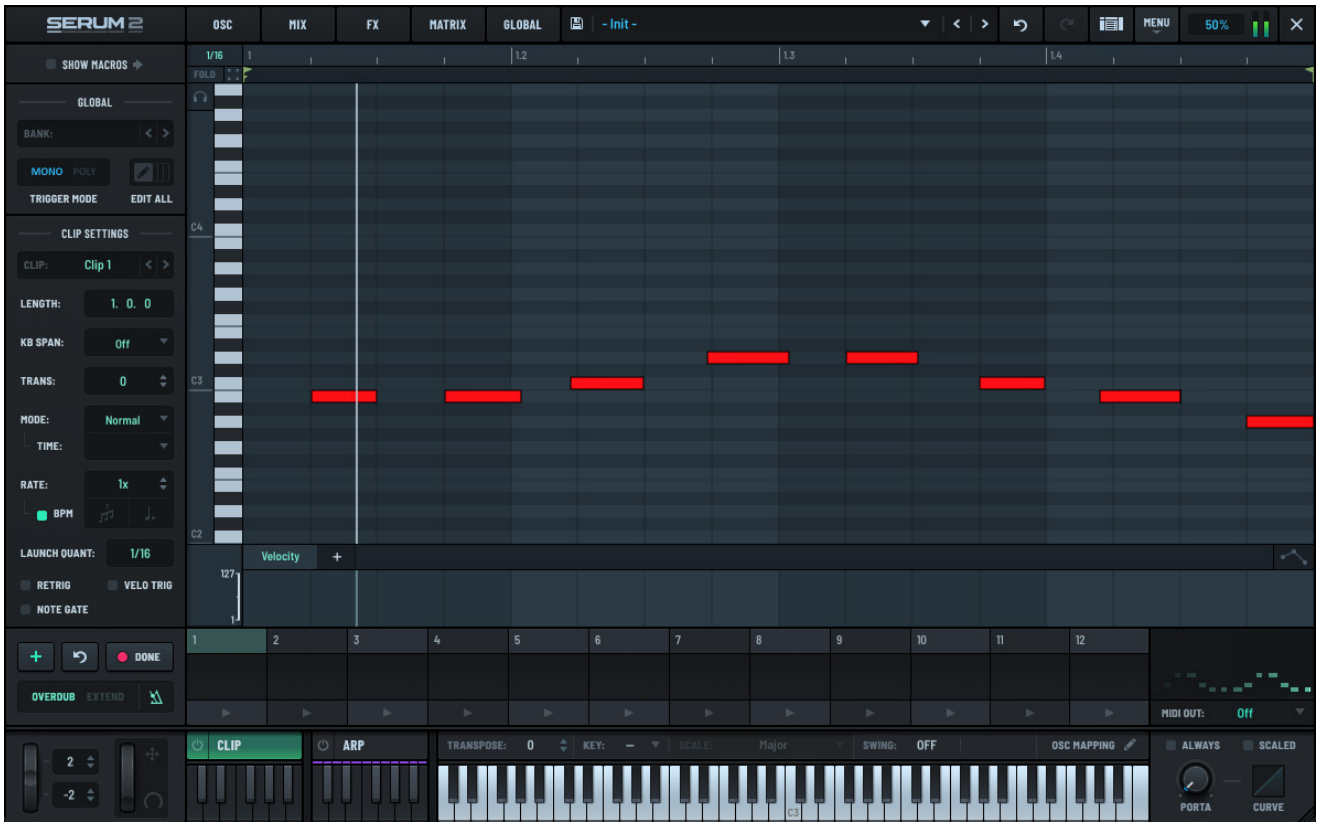
The **RECORD** pane changes to offer you additional controls to manage the recording process.

Play the appropriate notes using a MIDI controller.

The notes you play appear in red.



Recording In Progress



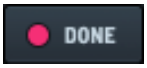
Recorded Notes Appear in Red

Click the  button to commit your most recent performance to the clip. The notes change from red to green to indicate the committed status.

Committed Notes (in green)

After committing, any new notes that you play appear in red (while the committed notes remain green).

Click the  button to undo your last performance (notes not committed to the clip).

When you've finished, click the 

button. All notes become "committed" and appear in green. Note, however, that the clip continues to play.

To stop playback, click the stop button for the corresponding clip.

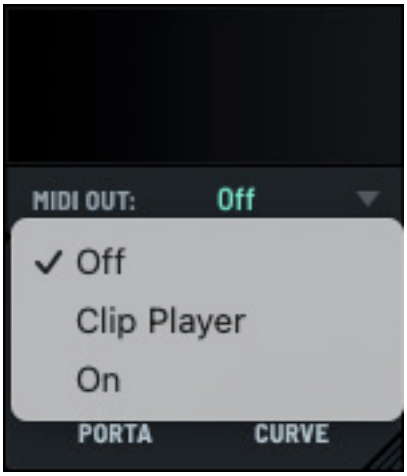
Managing MIDI Out

You can specify how Serum should output MIDI data.

Right-click the **MIDI OUT** field to choose an option using the menu. You can choose from among the following:

- Off — The default. Do not output MIDI data to the host.
- Clip Player — Output MIDI data from only the **CLIP** module, with no key/scale pitch quantizing applied.
- On — Output MIDI data from the **CLIP** module, through the **ARP** module (if enabled), with key/scale pitch quantizing applied.

Independent of the setting, the display shows the generated notes that get routed to the synth engine in Serum.



MIDI Out Menu

MIDI Out Data

Routing MIDI to Another Instrument

With **MIDI OUT** enabled (set to either **On** or **Clip Player**), you can route the MIDI output from Serum to another instrument (track) in your DAW.

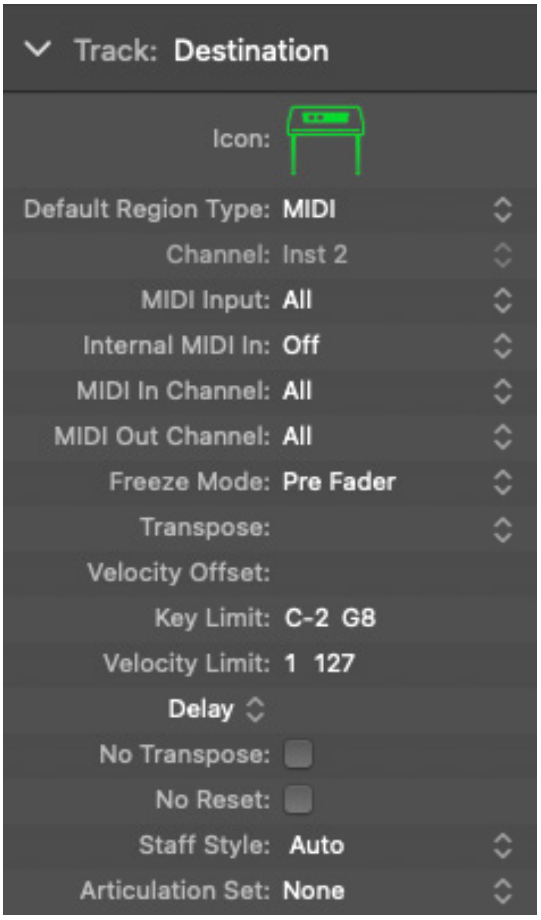
The specific setup to do this depends on your DAW. The following describes how to configure this using Apple Logic Pro X.

Begin by creating a **New Software Instrument Track** and adding Serum 2 as the instrument. This track is called **Source** in this example.

Then create a second software instrument track (called **Destination**) and add any instrument (in this example, Classic Electric Piano).

Logic Pro X Instrument Tracks

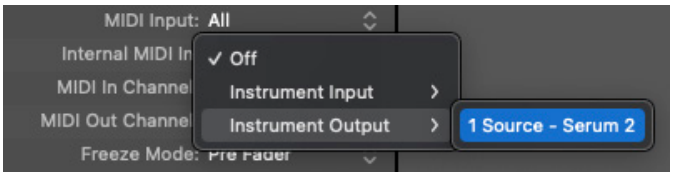
Expand the **Track** information for the **Destination**.



Expanded Track Information

Click the **Internal MIDI In** field, expand the **Instrument Output** menu, and choose the source (**Source - Serum 2** in this example).

The MIDI from the Serum CLIP player and/or arpeggiator is now routed to the **Destination** instrument.



Logic Pro X Instrument Tracks

Important: Remember that **MIDI OUT** in Serum needs to be enabled (set to either **On** or **Clip Player**).

Configuring the Clip Module

You can configure the **CLIP** module using the main context menu.

CLIP Menu

The following describes the options:

Operation	Description
Lock Module	Enable this option to lock the entire **CLIPS** module, including all loaded clips, when changing (sound) presets.
MIDI Input Trigger Octave	Select the octave from which notes trigger clips. Alternatively, set to **Off** if no triggering is required. By default, this is set to the lowest octave of the MIDI note range.

Using Macros

You can assign macros to a number of controls within the **CLIP** module, thereby allowing you to change one or more sound parameters using a single knob.

In addition, you can use any existing macro assignments to tweak the sound without having to leave **CLIP** view. Furthermore, you can record macro automation to a clip by dragging the macro knobs.

Click the 

button. The macros pane appears.

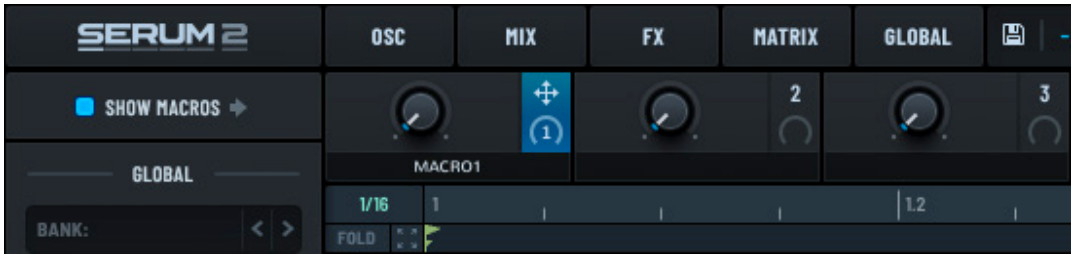
Macros Pane (in the CLIPS module)

To assign a macro to a control, drag and drop the macro selector to the appropriate control. The **+** sign indicates that you are over a valid modulation destination. Note that not all controls are valid destinations.

When you release the mouse button, Serum automatically makes the connection causing the macro to now affect the destination control.

Assigning a Macro to a Control

After setting the macro, notice that a number 1 now appears next to **MACRO 1**. This indicates that **MACRO 1** has one destination.



Macro with One Destination Assigned

Hover the mouse pointer over the macro to display a small tooltip showing the destinations.

Hovering to Show Destinations

In this case, **MACRO 1** displays **Clip 1 Rate** as the assigned destination.

You can repeat this process and assign the same macro to multiple controls. You can then manipulate the macro (perhaps assigned to a knob or slider on a physical controller) as you would a mod wheel.

Using the Arpeggiator

Serum features a versatile, full-featured arpeggiator designed to effortlessly sequence the notes of a chord into rhythmic patterns, enabling you to create dynamic, flowing musical phrases with ease.

Serum offers a broad array of patterns, fully customizable through parameters such as transposition, offsets, repeats, and chance, among others, giving you precise creative freedom.

Using the arpeggiator, you can swiftly craft intricate, evolving musical expressions that transform simple chords into captivating, complex melodies.

Click the ARP button to access the Serum arpeggiator.

Serum Arpeggiator

Exploring the Arpeggiator

The **ARP** module includes several panes to help you navigate and access the various features.

Arpeggiator Module

Use the **GLOBAL** pane to set parameters that affect all aspects of the arpeggiator, including the launch quantization and whether parameter edits affect all arp slots. You can also use the **GLOBAL** pane to load arpeggiator presets, create a new arp bank, and save your own arp bank as a user-defined preset.

Use the **PATTERN** pane to set the shape of the currently-selected arp slot (or all arp slots if you selected that **EDIT ALL** option in the **GLOBAL** pane). You can also use this pane to set the arpeggiator rate (in either beats per minute or Hertz), and enable triplets and dotted notes.

Use the **TRANSPOSE** pane to set the transpose shift (positive or negative) and range for the arpeggiator (which specifies how many times the pattern is transposed by the shift setting). You can also set the shape of the transpose range, which further expands your creative possibilities.

Use the **PLAYBACK** pane to specify the playback settings for the arpeggiator, including offsets and repeats, as well as the length of the gate and the probability of a note being played. You can also enable a latch to continue playing notes without needing to keep the keys pressed down, as well as pass incoming notes to the output (similar to a MIDI THRU port on a device) .

Use the **RETRIGGER** pane to specify how the arpeggiator retriggers the arp shape/pattern. You can set to retrigger the arp when the slot is launched or on a incoming note (first note or otherwise). You can also specify the rate at which the arpeggiator is retriggered.

Use the **VELOCITY** pane to configure the arpeggiator to raise or lower the arpeggiator note output velocities over time. You can set the speed at which the velocities are changed (decay) as well as the target velocity towards which the decay setting moves.

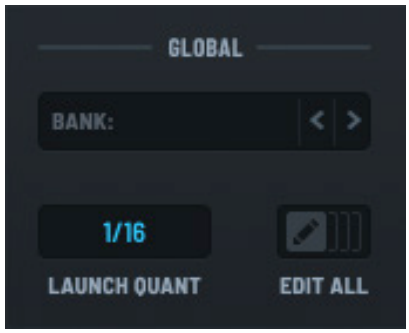
The **ARP** module further features 12 arpeggiators per arp bank. This offers a rich playground in which to explore and define evolving rhythmic patterns and dynamic musical phrases.

Setting Global Parameters

You can use the **GLOBAL** pane to set parameters that affect all aspects of the arpeggiator.

For example, you can use the **GLOBAL** pane to load an arp bank, set the launch quantization, and configure whether parameter edits apply to all arp slots.

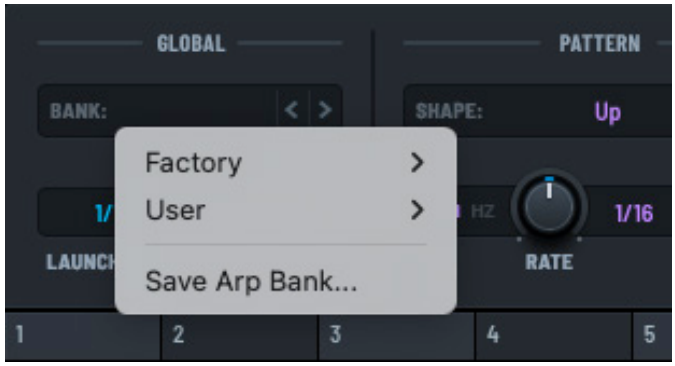
You can also create a new arp bank and save your own arp bank as a user-defined preset.



Global Arpeggiator Settings

To load a factory-supplied or user-defined bank, click the **BANK** field and choose a bank using the menu that appears.

The bank loads and populates the arpeggiator slots and associated settings.



Bank Menu

Set the **LAUNCH QUANT** by clicking in the field and dragging up or down.

This specifies the interval at which a triggered arp syncs to the clock/DAW.

Launch Quant Setting

Enable the **EDIT ALL** switch  to have your parameter edits apply to all arp slots.

When disabled, any changes to the arp settings (such as **TRANSPOSE** or **CHANCE**) apply only to the currently-selected arpeggiator slot.

When enabled, changes to any parameter become immediately effective for all arp slots.

Hold the Option (macOS) or Alt (Windows) key when editing an arp parameter to apply the change to all arps.

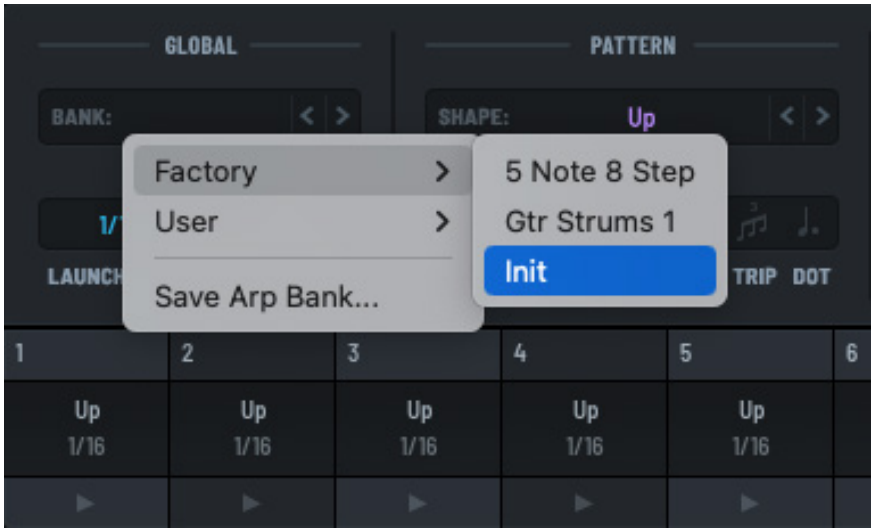
Creating a New Arp Bank

In addition to exploring the factory-supplied arp banks, you can initialize the bank and create your own arpeggiator bank.

Note: Initializing the arpeggiator bank only affects arp bank settings. This does not change any of the other sound design settings you've configured (including oscillators, filters, clips, and so on).

Click the **BANK** menu and choose **Init** in the menu.

This initializes the **ARP** module to the default settings and sets the stage for you to create a custom arpeggiator bank.



Creating a New Arp Bank

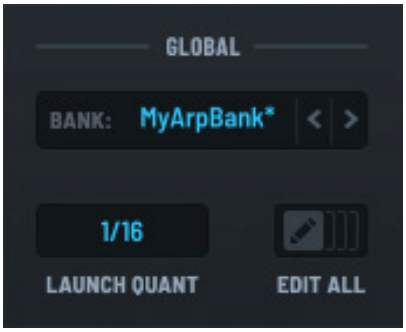
Saving Arp Banks

After creating a new arp bank, or editing an existing bank, you can save the entire arpeggiator configuration as a new preset.

Note that arp bank presets that have been modified display as asterisk (*) after the name.

Click the **BANK** field and choose **Save Arp Bank** in the menu that appears.

A dialog appears allowing you to type the arp bank name.

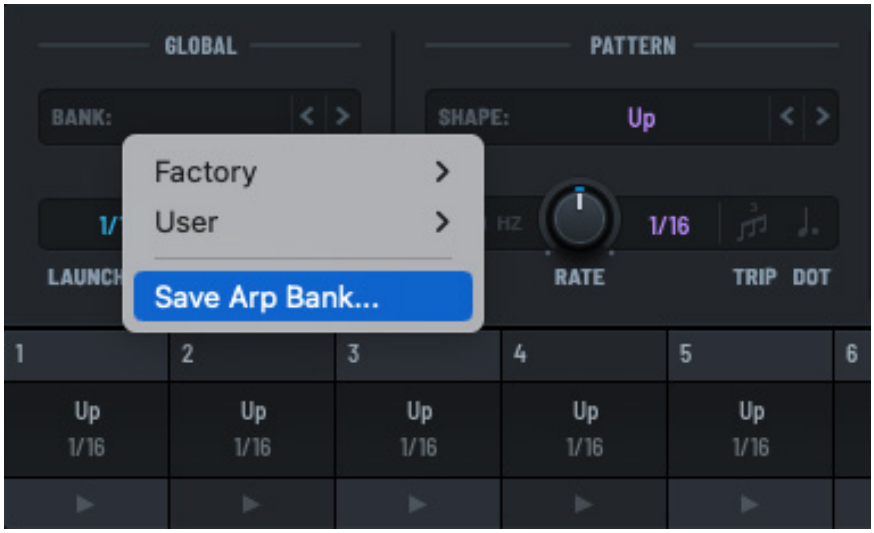


Arp Bank Modified

By default, the arp bank is saved in a standard location so that Serum can easily find it later.

Note that if you try to save a modified arp bank that has already been saved, a dialog appears allowing you to give your recently-modified arp bank a new name.

This allows you to freely experiment with changes, saving your work in increments as you progress.



Saving an Arp Bank

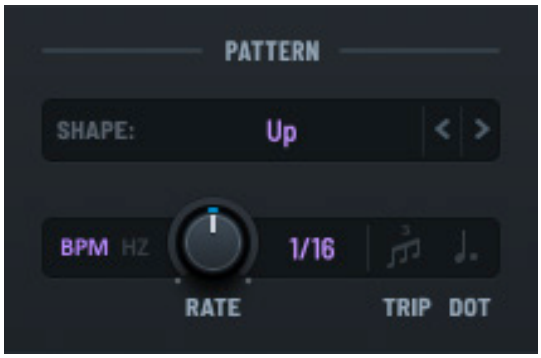
If you would instead prefer to overwrite the existing arp bank, you can choose the existing file name in the dialog.

Setting the Arp Pattern

You can set the shape of the arp pattern, as well as rate, triplets, and dotted note settings.

In the **SHAPE** field, click and choose an option using the menu that appears.

Serum offers a wide range of arpeggiator shapes, including standard up/down patterns as well as less familiar shapes such as **Converge** and **Diverge**, together with a collection of random shapes.



Pattern Settings

To set the arpeggiator rate, begin by selecting either **BPM** (beats per minute) or **HZ** (Hertz). Then click and drag either the **RATE** knob or the value field directly to the right of the knob (both work identically). To set a specific value, double-click either the knob or the field and type the appropriate value.

Click the **TRIP** button  to enable triplets. Click the **DOT** button  to enable dotted notes.

Important: The parameters you set in this and the other panes only affect the currently-selected arp slot unless you enabled **EDIT ALL** in the **GLOBAL** pane..

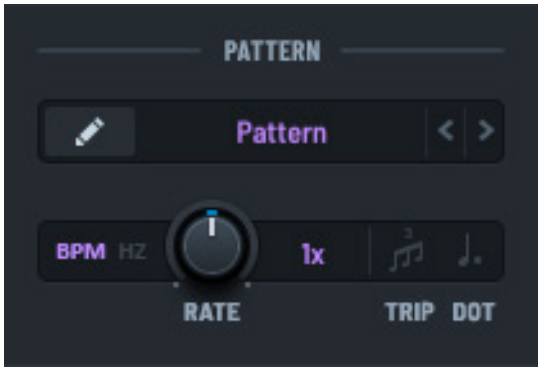
One way to try all the various shapes is to enable the **LATCH** button (in the **PLAYBACK** pane), play a chord on your MIDI controller, and then cycle through all the shapes one by one to hear the difference. You can also watch the Serum keyboard to get a visual sense of how the shape arpeggiates the chord.

Creating a Custom Pattern

You can define a custom arpeggiator using Serum's advanced pattern editor.

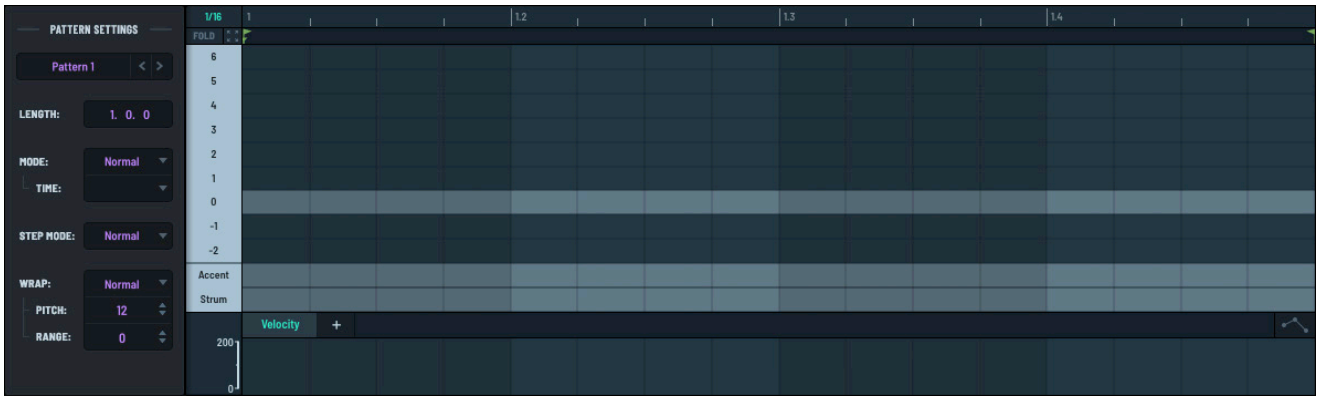
Choose **Pattern** in the **SHAPE** field, and click the  button.

The button highlights  and the arp editor appears in the top half of the interface.



Pattern Option

Use the left panel to specify the pattern settings and the right panel to configure the arpeggiator graph.



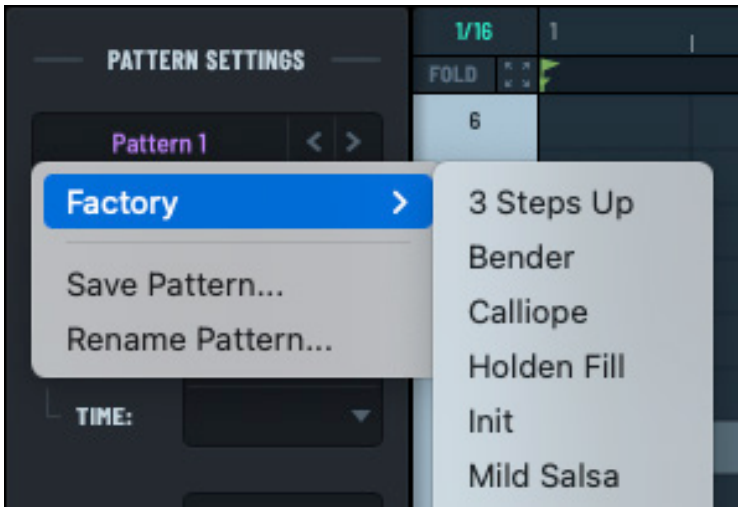
Arp Editor

Loading a Pattern

You can quickly load a factory-supplied or user-defined pattern.

Click the pattern name (topmost) field and choose a pattern using the menu that appears.

The pattern loads and populates the arp graph and associated settings.



Creating an Arp Pattern

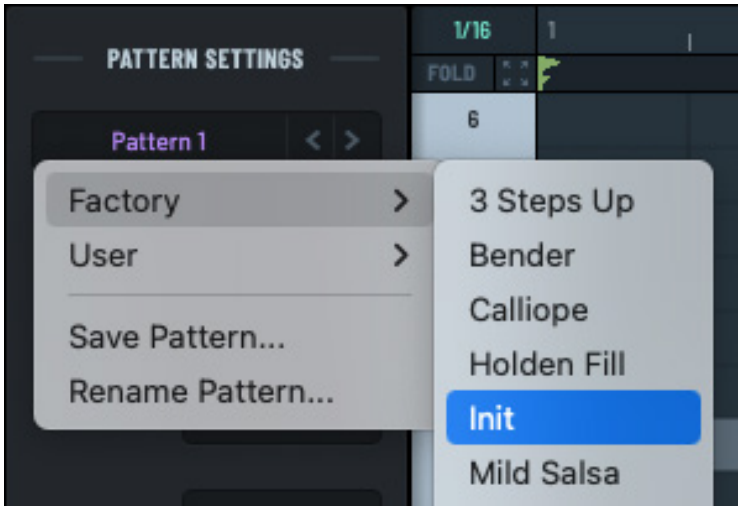
Creating a New Pattern

You can create a new pattern, or reinitialize the pattern editor, at any time.

Note: Initializing the pattern only affects current arp pattern settings. This does not change any of the other arp bank settings.

Click the pattern name field and choose **Init** in the menu.

This initializes the pattern editor to the default settings and provides an opportunity for you to create a new pattern.

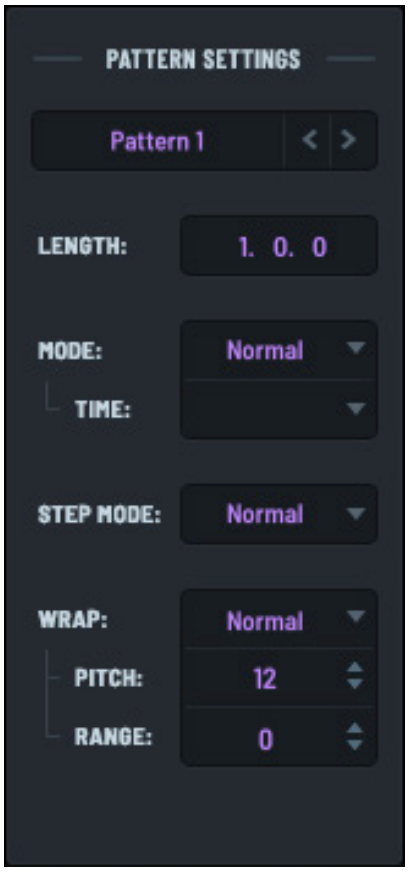


Creating an Arp Pattern

Configuring the Pattern Settings

You can use the **PATTERN SETTINGS** pane to specify parameters that affect the entire arp pattern.

For example, you can use the pane to set the pattern length, the mode (normal, reverse, and so on), the step mode, and the wrap settings.



Pattern Option

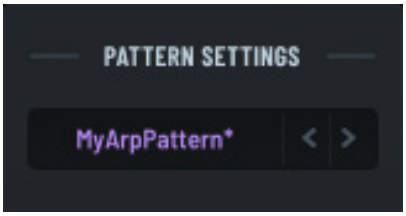
The following table describes the parameters you can set:

Field	Description
LENGTH	The pattern length in bars, beats, and 16th notes.
MODE	The mode of the pattern specifying how the playhead moves, from among the following: • Normal • Reverse • Pendulum • Random • Rand Start • Rand End • One Shot • Static
TIME	For random play modes, specifies how often the playhead jumps to a new random position. You can choose from values between 1/16th note and four bars.
STEP MODE	Specifies how each step is played, from among the following: • Normal • New Only • Chord • Chord (new) The chord triggers all held notes on each step with voicing determined by step numbers. Note that new modes trigger a step only if the input note is received exactly at the same time as the step triggers.
WRAP	Specifies how pattern steps less than, or greater than, the number of held keys are treated.
PITCH	The wrap transpose, from 0 to 24 semitones.
RANGE	Specifies how pattern steps less than, or great than, the number of held keys are transposed.

Saving a Pattern

After creating a new pattern, or modifying an existing pattern, you can save the configuration as a new pattern.

Note that pattern presets that have been modified display as asterisk (*) after the name.



Pattern Modified

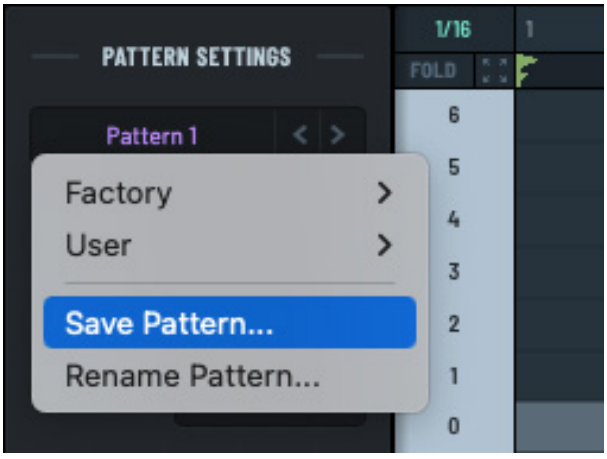
Click the pattern name (topmost) field and choose **Save Pattern** in the menu that appears.

A dialog appears allowing you to type the arp pattern name.

By default, the pattern is saved in a standard location so that Serum can easily find it later.

Note that if you try to save a modified pattern that has already been saved, a dialog appears allowing you to give your recently-modified arp pattern a new name.

This allows you to freely experiment with changes, saving your work in steps as you progress.



Saving a Pattern

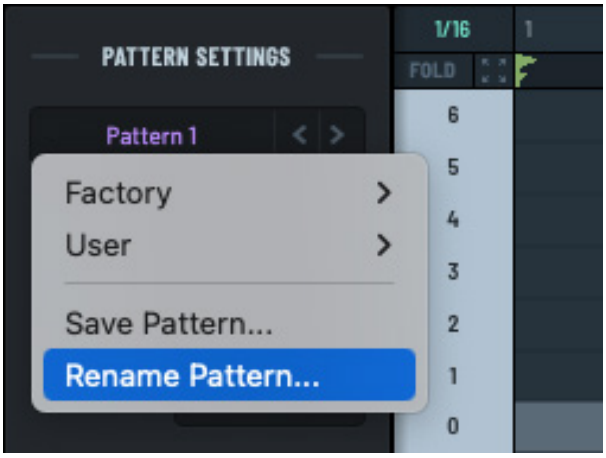
If you would instead prefer to overwrite the existing arp pattern, you can choose the existing file name in the dialog.

Renaming a Pattern

You can rename a pattern, as needed.

Click the pattern name field and choose **Rename Pattern** in the menu that appears.

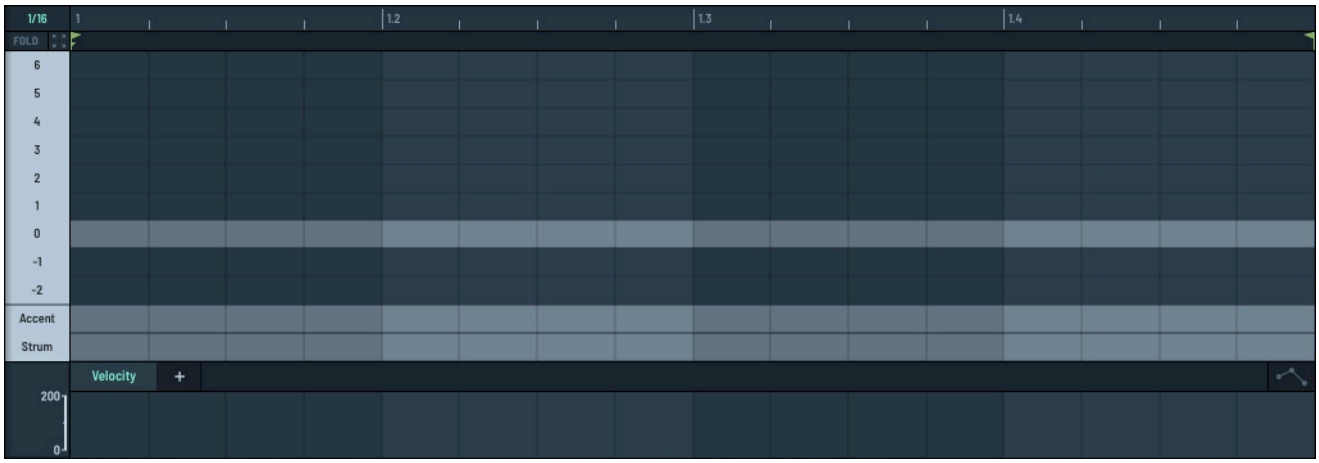
Type the new name of the pattern.



Renaming a Pattern

Using the Arp Graph Editor

The pattern editor features an advanced graph editor that you can use to create the arp graph. You can use the graph editor to interactively create complete patterns in the context of your current patch.



Graph Editor

Click the grid size 1 and drag up or down to change the default grid setting.

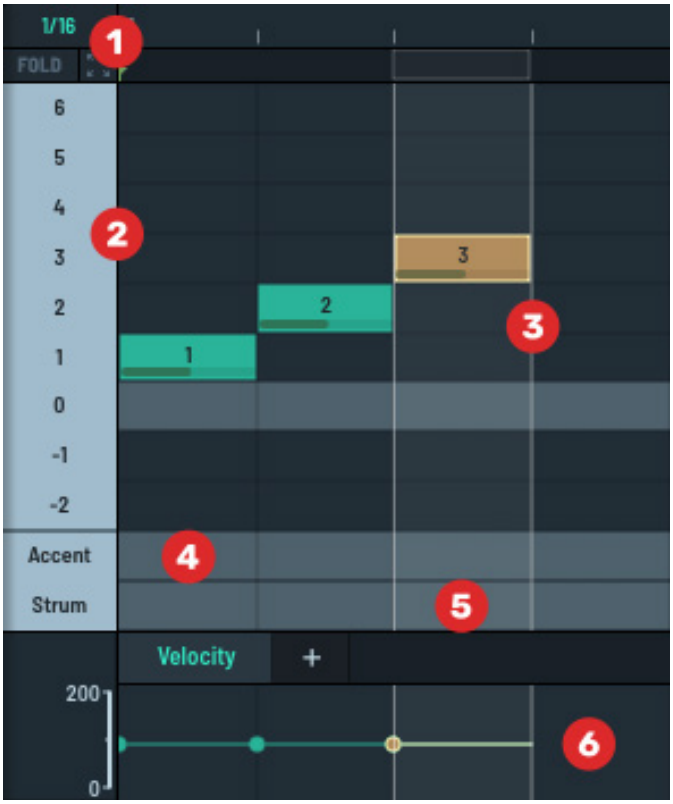
Scroll the grid up or down by clicking and dragging in the **Step** area. You can also scroll up or down using the mouse wheel. 2

Double-click to add a new *note event* in the arp graph. The latest note event you added appears in orange. 3

The arp graph offers capabilities very similarly to the piano roll in the **CLIP** module. For example, you can click and drag across the graph to select note events, and copy/cut and paste events to a new area in the graph.

You can also select and move note events around the graph. See “Working with the Piano Roll” on page 224 for more information.

Double-click in the **Accent** lane to add an accent to the corresponding note event. 4 Double-click in the **Strum** lane to add strumming to the corresponding area. 5



Saving a Pattern

Use the automation lanes 6 to control various parameters over time. Right-click in various places in the arp grid to display a context menu for the corresponding element.

Transpose Settings

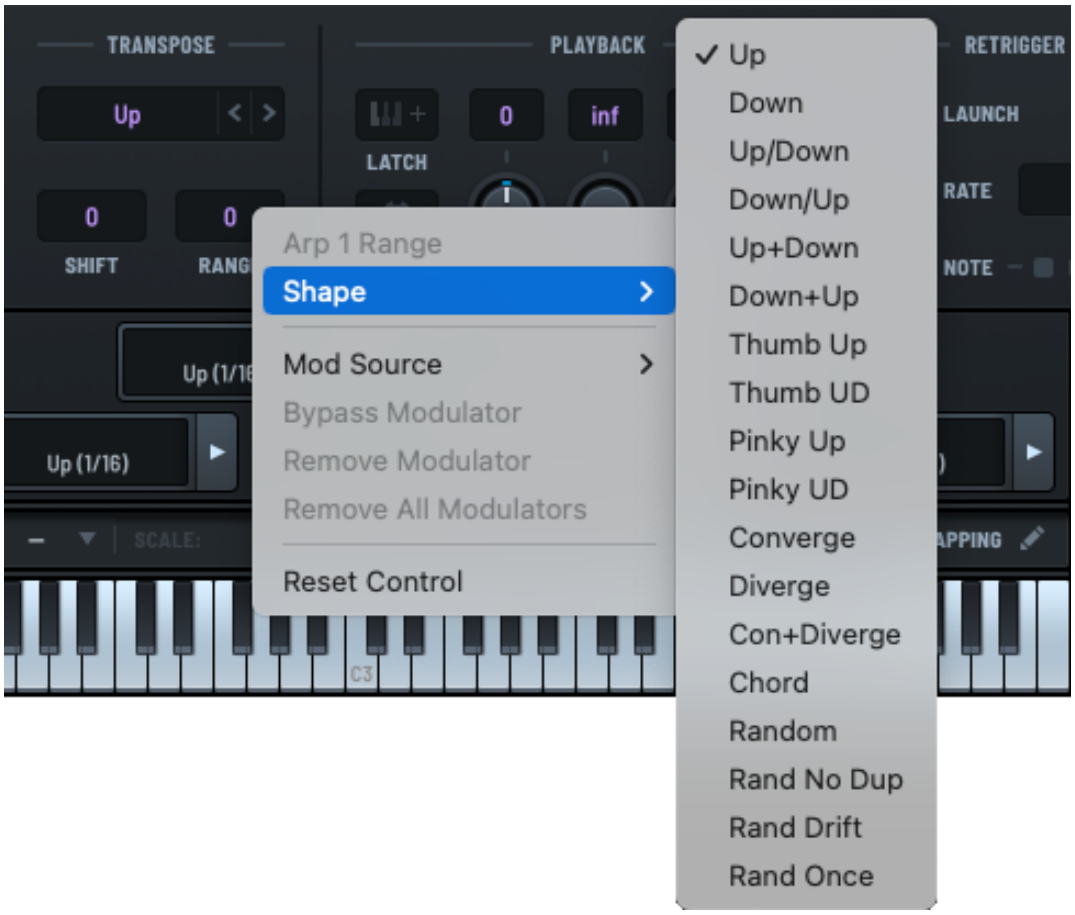
You can set the transpose shift and range for the arpeggiator.

Click and drag the **SHIFT** knob to set the amount that each repetition of the pattern is transposed. This can be a positive or negative value.

Click and drag the **RANGE** knob to specify how many times the pattern is transposed by the **SHIFT** setting.

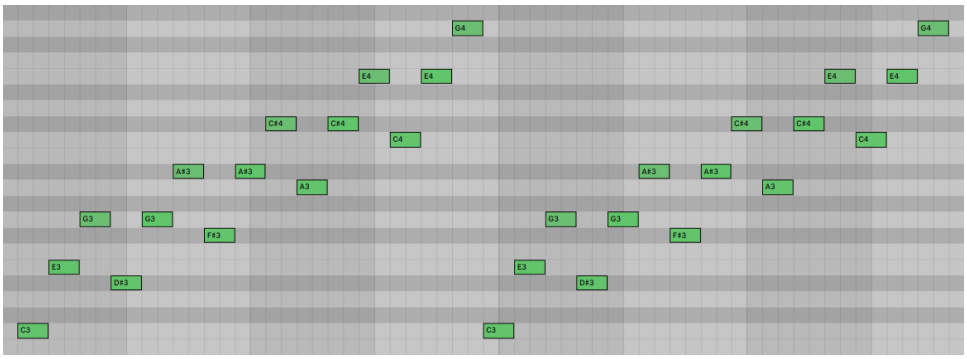
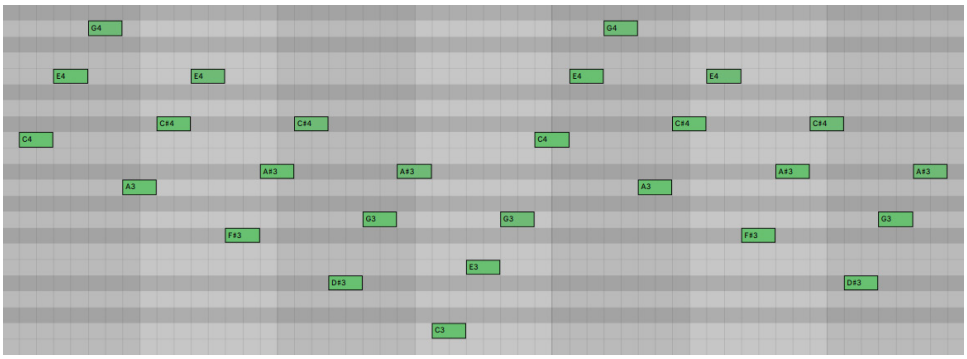
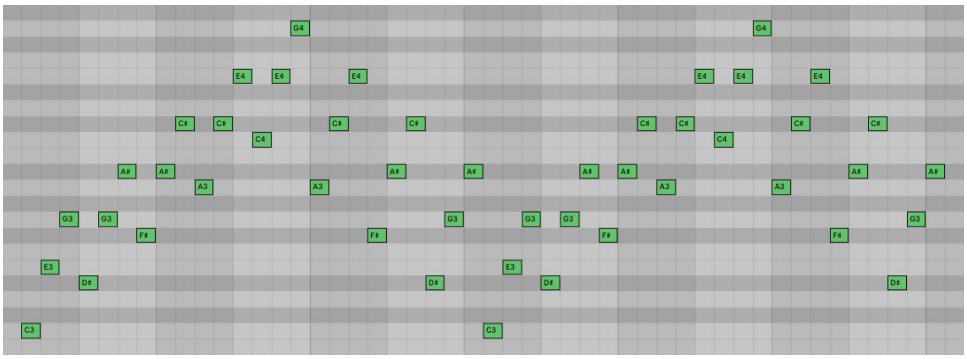
To set the shape of the transpose range, right-click the **RANGE** knob and choose an option in the menu that appears.

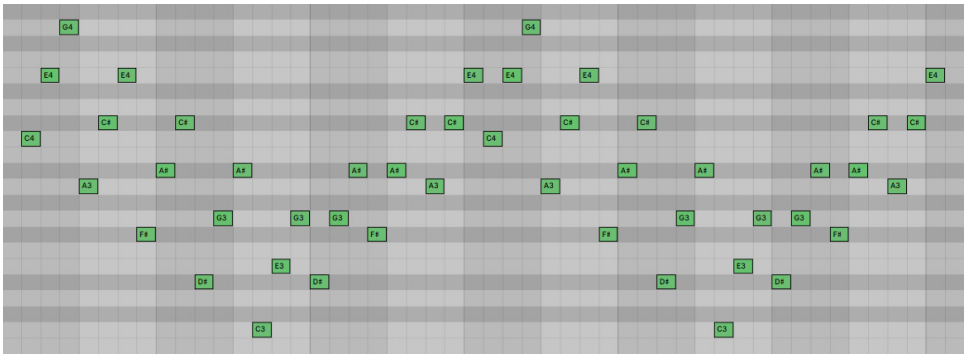
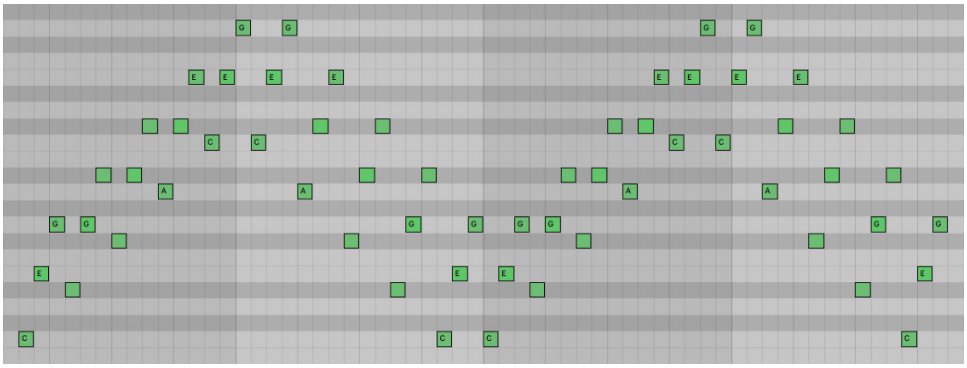
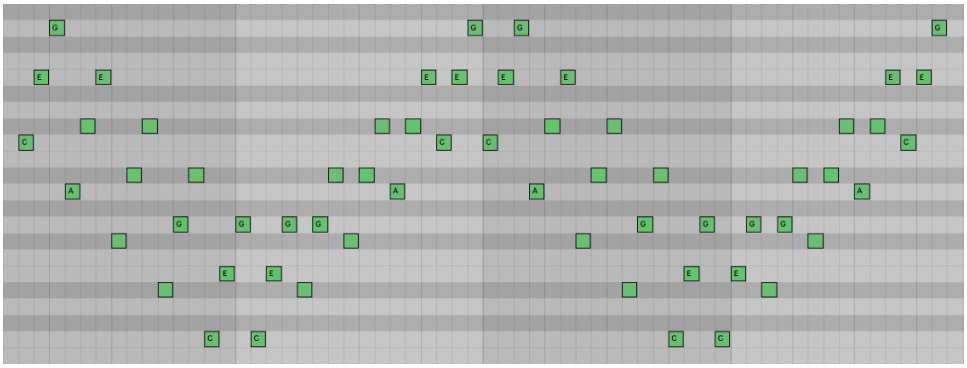
Transpose Settings

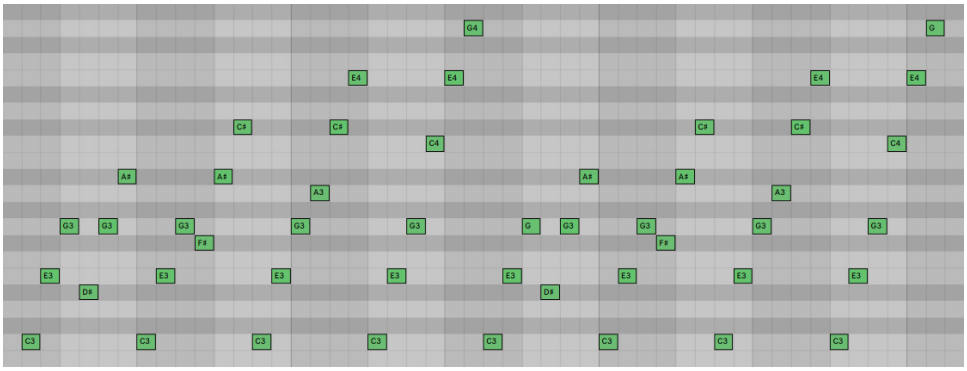
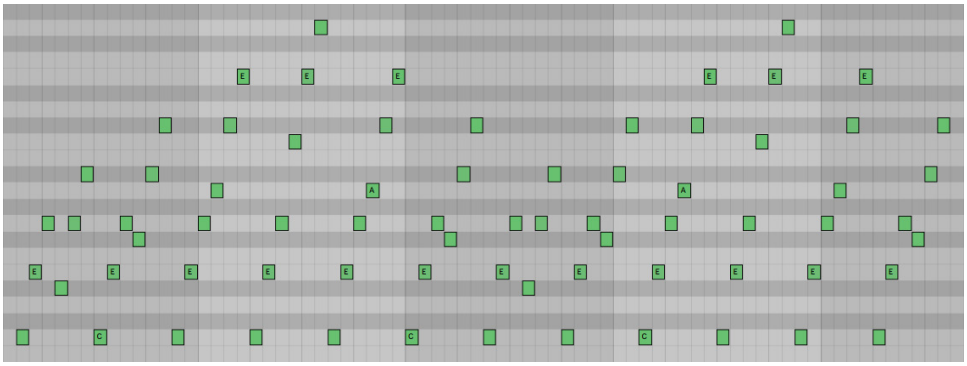


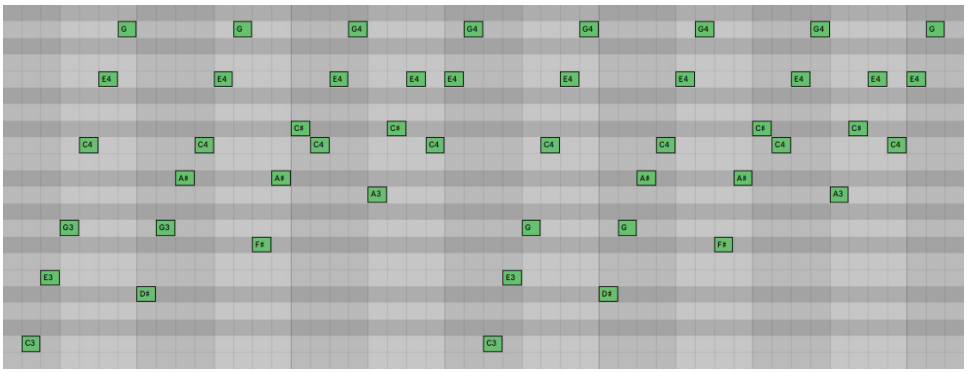
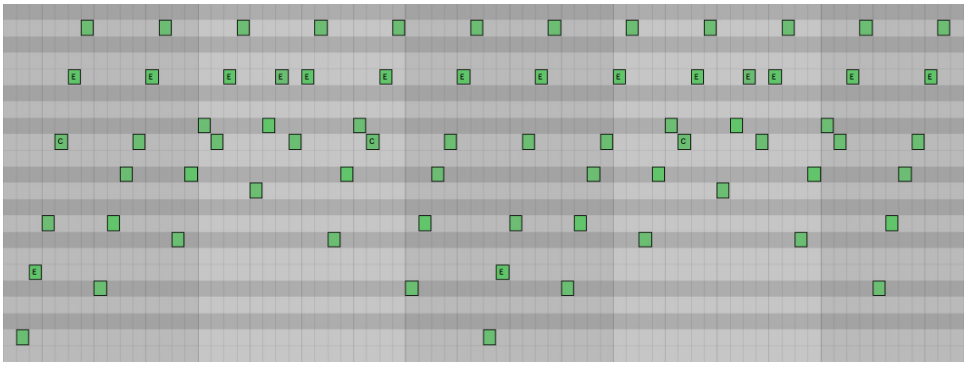
Transpose Range Shape Menu

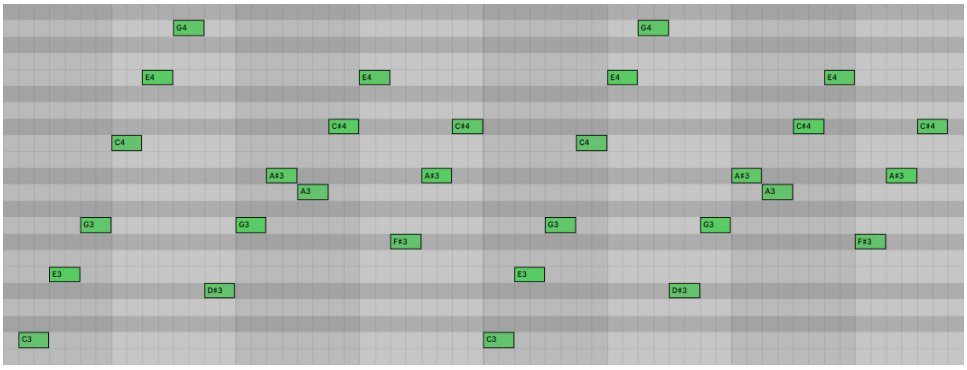
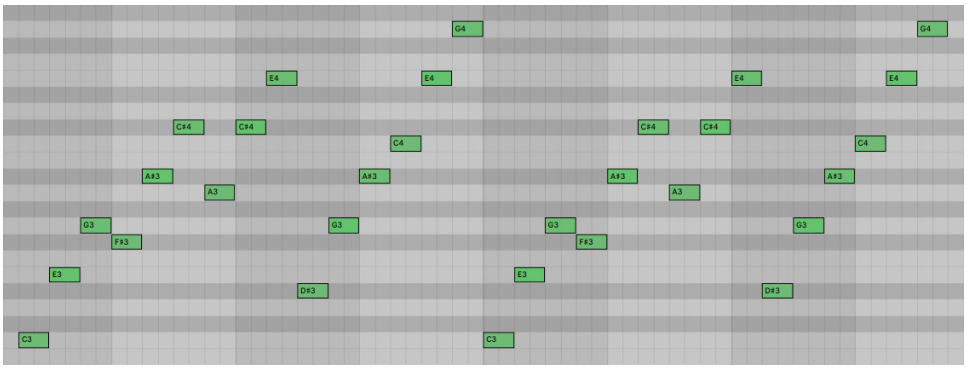
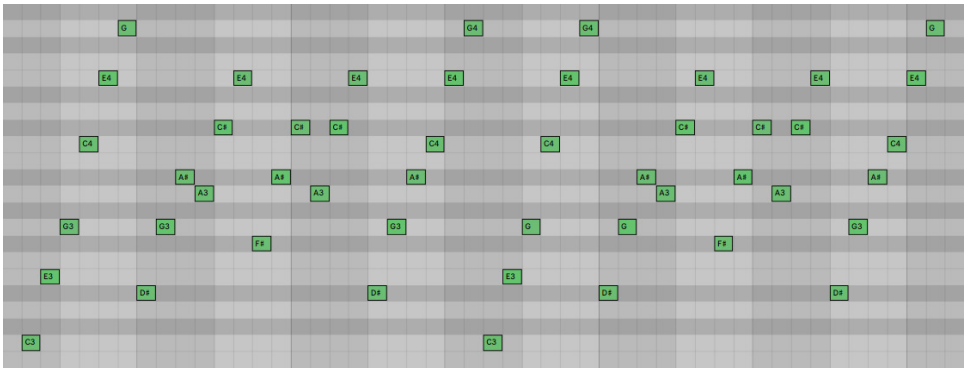
The following table describes the range shape options available. Note that each example uses a **SHIFT** setting of 3 and a **RANGE** setting of 4.

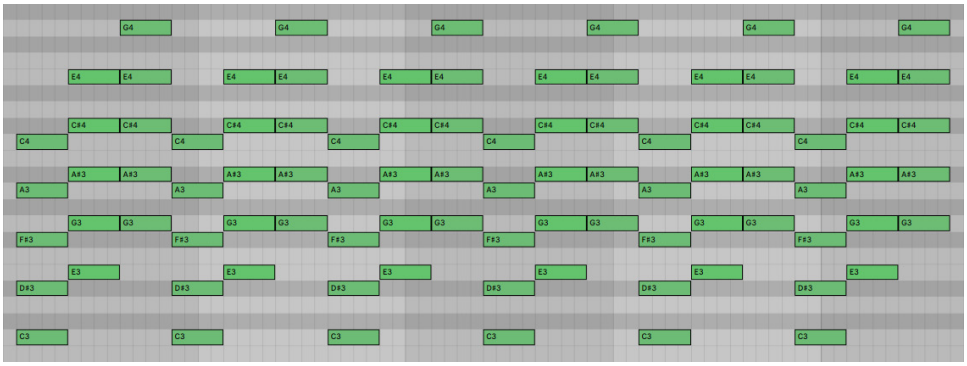
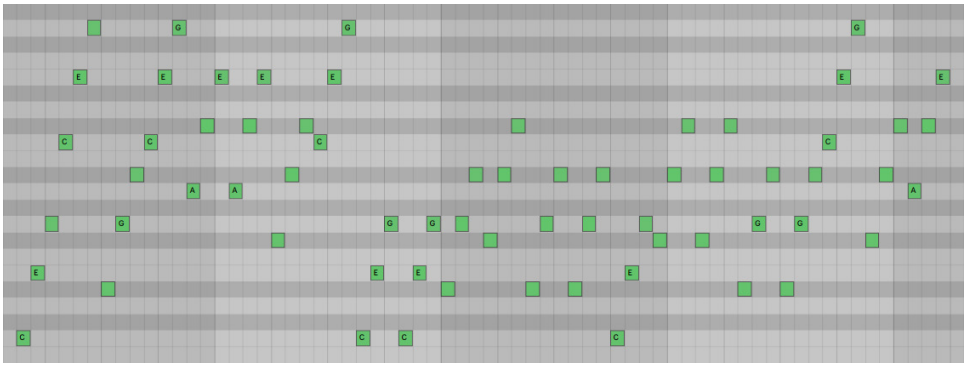
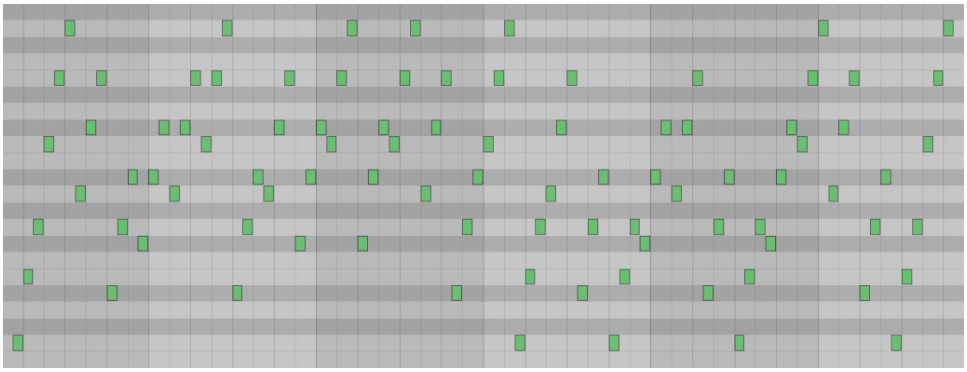
Shape	Description
Up	Transpose the arpeggiated notes by the **SHIFT** setting and repeat the number of times specified by the **RANGE** setting. 
Down	Transpose the arpeggiated notes by the **SHIFT** setting and repeat the number of times specified by the **RANGE** setting. 
Up/Down	Transpose the arpeggiated notes by the **SHIFT** setting and repeat the number of times as specified by the **RANGE** setting. Then immediately reverse (transpose the arpeggiated notes in the reverse direction and repeat using the same settings). 

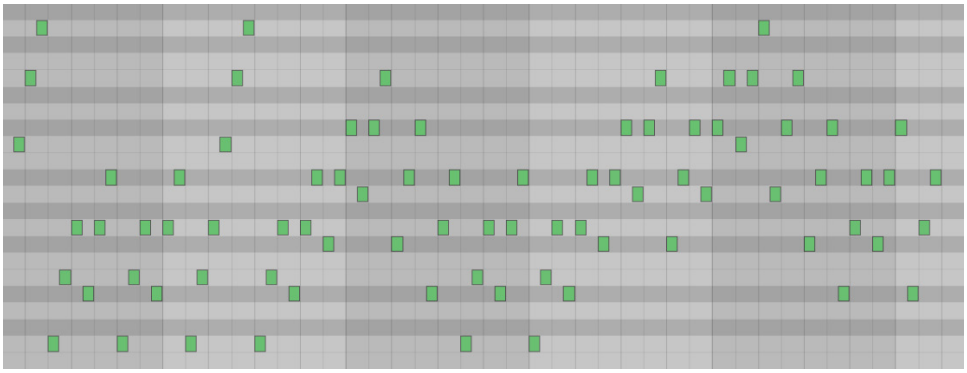
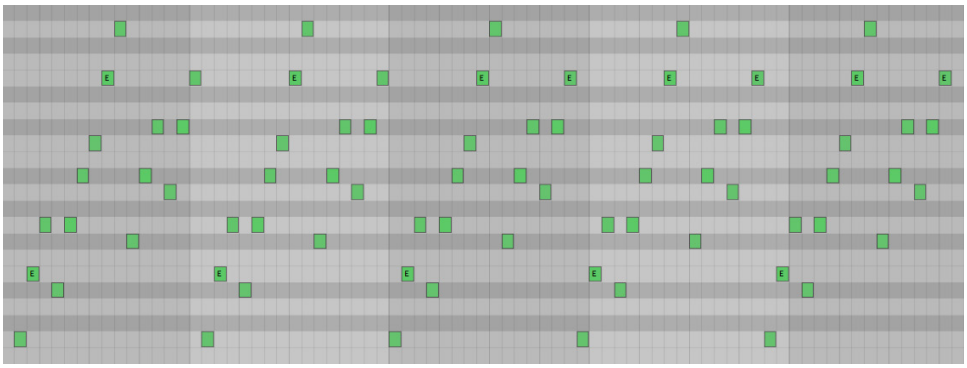
Shape	Description
Down/Up	Transpose the arpeggiated notes by the **SHIFT** setting and repeat the number of times specified by the **RANGE** setting. Then immediately reverse (transpose the arpeggiated notes in the reverse direction and repeat using the same settings). 
Up+Down	Transpose the arpeggiated notes by the **SHIFT** setting and repeat the number of times specified by the **RANGE** setting. Then start again in the reverse direction. 
Down+Up	Transpose the arpeggiated notes by the **SHIFT** setting and repeat the number of times specified by the **RANGE** setting. Then start again in the reverse direction. 

Shape	Description
Thumb Up	Transpose the arpeggiated notes by the **SHIFT** setting. After each transposition, play the original arpeggio followed by another transposition. Repeat the number of times specified by the **RANGE** setting. 
Thumb UD	Transpose the arpeggiated notes by the **SHIFT** setting. After each transposition, play the original arpeggio followed by another transposition. Repeat the number of times specified by the **RANGE** setting. Then immediately reverse (transpose the arpeggiated notes in the reverse direction and repeat using the same settings). 

Shape	Description
Pinky Up	Transpose the arpeggiated notes to the maximum (as specified by the **SHIFT** and **RANGE** settings). After each transposition, play the maximum transposition followed by the original arpeggio transposed. Repeat the number of times specified by the **RANGE** setting. 
Pinky UD	Transpose the arpeggiated notes to the maximum (as specified by the **SHIFT** and **RANGE** settings). After each transposition, play the maximum transposition followed by the original arpeggio transposed. Repeat the number of times specified by the **RANGE** setting. Then immediately reverse (transpose the arpeggiated notes in the reverse direction and repeat using the same settings). 

Shape	Description
Converge	Transpose the arpeggiated notes to the maximum (as specified by the **SHIFT** and **RANGE** settings). After each transposition, converge the transpositions between original to maximum. 
Diverge	Transpose the arpeggiated notes as specified by the **SHIFT** setting. After each transposition, diverge the transpositions further. 
Con+Diverge	Transpose the arpeggiated notes first using the converge pattern followed by the diverge pattern (see above). 

Shape	Description
Chord	Use the **SHIFT** and **RANGE** settings to play the chord instead of the arpeggio. 
Random	Randomly transpose the arpeggiated notes based on the **SHIFT** setting and repeating the number of times specified by the **RANGE** setting. 
Rnd.NoDup	Randomly transpose the arpeggiated notes (without duplicates) based on the **SHIFT** setting and repeating the number of times specified by the **RANGE** setting. 

Shape	Description
Rnd.Drift	Randomly transpose (with a bias towards drifting) the arpeggiated notes based on the **SHIFT** setting and repeating the number of times specified by the **RANGE** setting. 
Rnd.Once	Randomly transpose the arpeggiated notes once based on the **SHIFT** setting and repeating the number of times specified by the **RANGE** setting. Then continue using this pattern. 

Playback Settings

You can specify the playback settings for the arpeggiator, including offsets and repeats, as well as the length of the gate and the probability of a note being played.

Click the **LATCH** button  to continue playing notes without needing to keep the keys pressed down. The button highlights  when enabled.

Click the **THRU** button  to pass the incoming notes to the output (in a conceptually similar way to a physical MIDI THRU port on a device).

Playback Settings

When this setting is disabled, the arpeggiator consumes the input; when enabled, the notes are passed through and played.

Click and drag the following knobs to perform the associated action:

Knob	Description
OFFSET	Offset the order of the arpeggiator notes.
REPEATS	Set the number of times the pattern and transpose range are repeated.
GATE	Set the length of the arpeggiator notes, relative to the **RATE** setting.
CHANCE	Set the probability of a note being played. Right-click the knob and choose **Pre** in the menu to apply the **CHANCE** setting before the **SHAPE** pattern is advanced. You can use this to guarantee that the next note played is always next in the sequence.

When the arp is enabled, MIDI CC64 sustain messages sent to Serum will control the latch instead of regular note sustain.

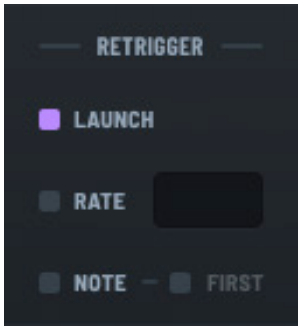
Retrigger Settings

You can specify how the arpeggiator retriggers the arp shape/pattern. Enable the **LAUNCH** setting to retrigger the arp when the slot is launched.

Enable the **RATE** setting and set an appropriate rate by clicking and dragging the field. The rate sets the interval at which the arpeggiator is retriggered.

Enable the **NOTE** setting to retrigger on an incoming note.

Similarly, enable the **FIRST** setting to retrigger only on a first incoming note, as would be the case for held notes in a chord.



Retrigger Settings

Velocity Settings

You can set the arpeggiator to raise or lower the arpeggiator note output velocities over time. Enable the **VELOCITY** setting to activate the velocity settings (in this section).

Enable the **RETRIG** setting to reset the velocity decay value (back to incoming velocities) whenever the **RETRIGGER** feature is activated. (See the previous section for more information about retrigger settings.)

Click and drag the **DECAY** knob to set the speed at which the velocities are changed.

Click and drag the **TARGET** knob to set the target velocity towards which the **DECAY** setting moves.

Velocity Settings

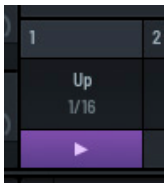
Managing Arpeggiators

The **ARP** module features 12 arpeggiators per arp bank (similar to the **CLIPS** module). This makes it easy and convenient to trigger these arps using either a MIDI keyboard or your computer keyboard.

Arpeggiator Slots

Click an arp slot to select it. The current arpeggiator is highlighted and the play button turns purple. The arp settings appear in the various sections.

To perform operations on the arpeggiator, right-click the arp and choose an option in the menu that appears.



Current Arp (Highlighted)

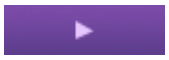
The following table describes the available operations:

Operation	Description
Rename Pattern	By default, arpeggiators are named after the pattern shape, such as Up (1/16) or Down (1/32). When you define the pattern yourself (by choosing **Pattern** in the **SHAPE** field and then using the pattern editor), you can rename the arp to better describe the pattern you created. Right-click in the arp and choose **Rename Pattern** in the context menu. A text area appears in the clip header. Type the new pattern name and press the **Enter** key.
Copy Pattern	Copy the arp settings to Serum's internal clipboard. Right-click the source arp and choose **Copy Pattern** in the context menu. Right-click the destination arp and choose **Paste Pattern** in the menu. Note that it is not possible to copy a pattern from one instance of Serum to another.
Paste Pattern	Paste the most-recently copied arp into the target arp.
Erase Pattern	Erase the contents of the selected arp.
Show Pattern Names	Show the arp names in the slots.

Triggering the Arpeggiator

You can trigger an arpeggiator using the play button or the arpeggiator keyboard.

To trigger an arpeggiator while in **ARP** mode, click the  button for the corresponding slot.

The arp starts playing and the button changes  to indicate which arp is currently playing.

To stop an arpeggiator, stop the MIDI input to the arpeggiator either by releasing the keys, stopping a playing clip, or disabling the latch (if enabled).

Arpeggiator Slot with Play Button

You can make changes to any aspect of the arp while it is playing, including changing patterns, modifying arp settings, and more. You can also change the underlying sound design as the arp plays, providing a versatile way to hear your sound evolve, in context.

You can also trigger clips using the **ARP** keyboard.

This is helpful when you are not in **ARP** mode and instead tweaking the oscillators, mixing, applying effects, or adjusting the modulation matrix.

Arpeggiator Keyboard

Notice that the arp slots (in **ARP** mode) are arranged similar to a keyboard. This makes it easy to logically map the arp slots to the **ARP** keyboard. Press a key to play the corresponding arp. The key highlights in purple.

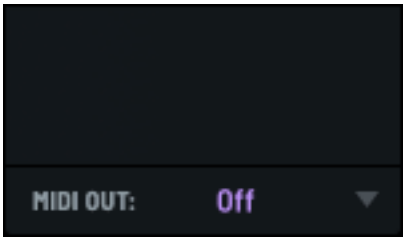
Press the same key to relaunch the arp, or press another key to launch that arp.

Important: If you launch an arp but don't hear the sequence that you expect, verify that arp playback is enabled. 

Managing MIDI Out

You can specify whether Serum should output MIDI data from the **CLIP** player, the **ARP** (arpeggiator), or neither.

By default, **MIDI OUT** is set to **Off**, which means that Serum does not output MIDI data to the host (DAW).



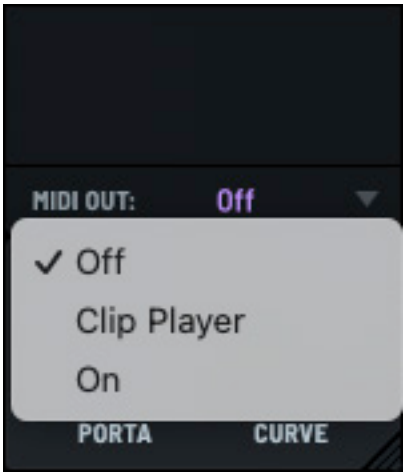
MIDI Out Pane

Otherwise, the display shows the corresponding MIDI data stream.

MIDI Out Data

Right-click the **MIDI OUT** field to choose an option using the menu. You can choose from among the following:

- Off — The default. Do not output MIDI data to the host.
- Clip Player — Output MIDI data from only the **CLIP** player.
- On — Output MIDI data from both the **CLIP** player and the arpeggiator.



MIDI Out Menu

Configuring the Arp Module

You can configure the **ARP** module using the main context menu.

ARP Menu

The following describes the options:

Operation	Description
Lock Module	Enable this option to lock the entire **ARP** module, including all arps, when changing (sound) presets.
MIDI Input Trigger Octave	Select the octave from which notes trigger arps. Alternatively, set to **Off** if no triggering is required. By default, this is set to the lowest octave of the MIDI note range.

Using the Keyboard

Serum features an on-screen keyboard that you can use to play notes as well as set a range of parameters that affect MIDI input and output.

Serum Keyboard

Use the **TRANSPOSE** field 1 to transpose the keyboard within a range of two octaves (set in semitones).

Use the **KEY** and **SCALE** fields 2 to choose appropriate options. Serum offers a very thorough list of scale options from which to choose.

Use the **SWING** field 3 to set the swing applied to certain notes to add groove.

And finally, use the **OSC MAPPING** feature 4 to edit the note ranges of the oscillators and arpeggiator.

Transpose

You can transpose the keyboard within a range of -24 to +24 semitones.

Keyboard Transpose

Click the **TRANSPOSE** field and drag up or down. Alternatively, for even finer control, click the up and down arrows to adjust the setting by a single semitone.

You can also hover over the field and adjust the setting by 12 semitone increments using the mouse wheel.

Setting the Key and Scale

You can set the key and scale within Serum. These settings, together with the **TRANSPOSE** setting (described above), are applied to both MIDI input and the output of both the **CLIP** and **ARP** modules.

Key and Scale

Serum offers a comprehensive selection of scales.

Scale Options

After selecting the key and scale, the piano roll in the **CLIP** module highlights the root note of the key. In addition, notes within the scale are highlighted with a gray background, while notes outside the scale appear with a darker gray background.

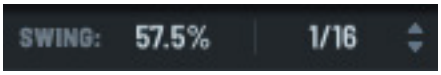
Setting Swing

You can set the swing applied to certain notes to add groove, thereby imparting a more human feel instead of a rigid, machine-like rhythm.

Swing

Click the **SWING** field and drag up or down. The range of swing values depends on the host DAW that you are using (see the note below). You can also hover over the field and adjust the setting using the mouse wheel.

By default, swing is set to **OFF**. As soon as you move away this setting, a second field appears showing the swing division.



Click and drag to set the appropriate division. You can also use a mouse wheel or the arrow buttons.

Serum attempts to match the swing value displayed to the convention used in the host DAW. This means that setting the same values in Serum and the host should result in the same amount of swing.

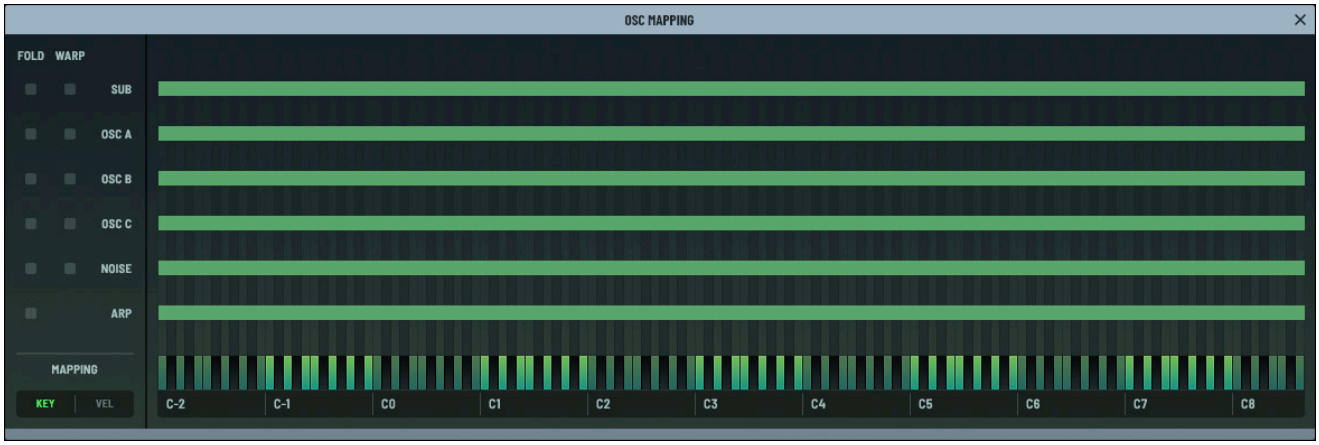
For example, when using Ableton Live, swing is displayed as a range between 12.5% and 87.5%. In contrast, FL Studio displays swing with a range of -150% to 150%.

Oscillator Mapping

You can edit the note ranges of the oscillators and arpeggiator to define and limit the range of notes to which the oscillator or arpeggiator responds.

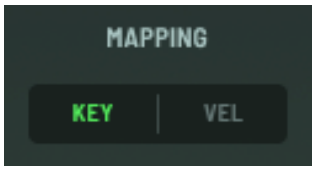
Oscillator Mapping

Click the **OSC MAPPING** button. The oscillator mapping dialog appears.



Oscillator Mapping Editor

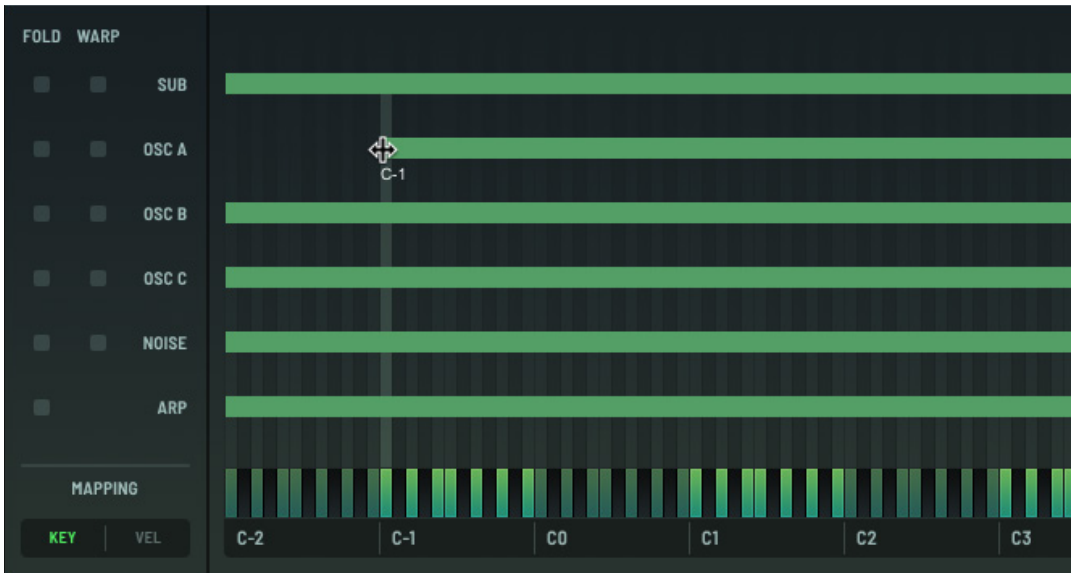
To set the key mapping, select **KEY** in the **MAPPING** section (lower left).



Key Mapping

Click and drag the left or right edge of the note range for the corresponding oscillator or arpeggiator.

The current note appears below the cursor, and a highlight shows the note in the context of the keyboard.

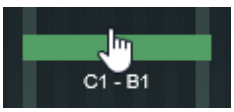


Setting the Note Range

To move a note range, drag the range to the new location.

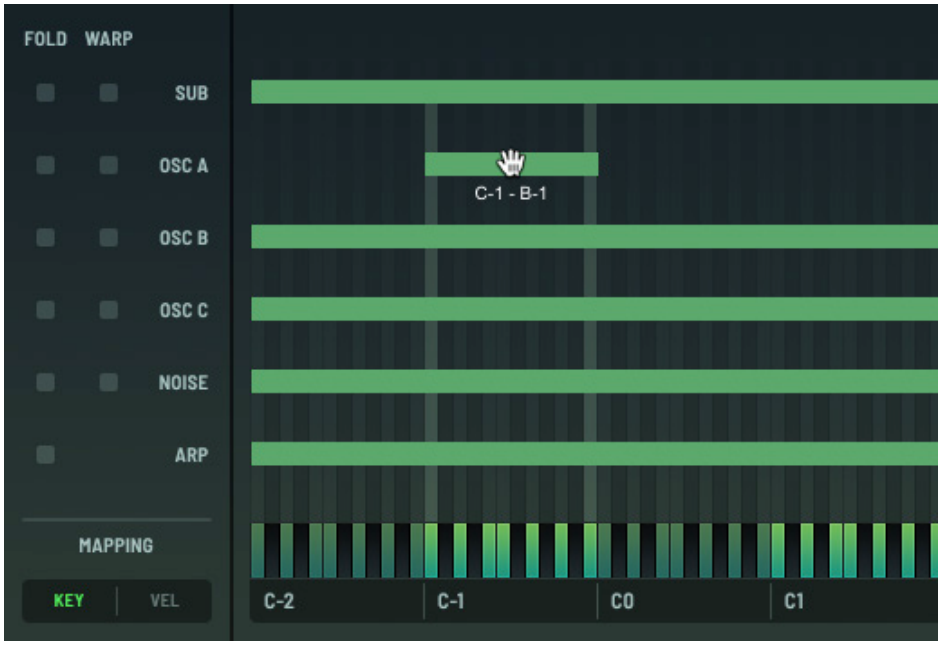
macOS

Windows



Note that the “hand” looks slightly different when moving a note range on macOS and Windows.

The current note boundaries appear below the cursor (hand), and highlights show the note range in the context of the keyboard.



Moving a Note Range

You can double-click a range to type specific values into the text box that appears.

Use the same notation that appears on the keyboard below the ranges.

Use the **FOLD** checkbox to specify what happens to notes outside the selected range. When unselected, the notes are ignored. When selected, the notes get "folded" into the selected range. You can use this, for example, to restrict the range of an oscillator used for sub-bass to a single low octave, regardless of the octave the incoming note was played.

Select the **WARP** checkbox to set the corresponding oscillator to warp other oscillators when keys are played outside of its note range. For example, consider the case when the **OSC A** warp is set to **FM from B**. If you want the warp to apply regardless of the **OSC B** note ranges, select the **WARP** checkbox for **OSC B**. This allows you to restrict the **OSC B** note range without losing the warp on **OSC A** for notes outside of the **OSC B** range.

You can define note ranges to do the following:

- Create splits and layers

  You can assign different oscillators to specific note ranges to create multi-layered sounds or keyboard splits. For example, you might create bass and lead layers, assigning one oscillator to low notes for a bass sound and another to high notes for a lead.

  Similarly, you could create complex pads by layering different oscillators with unique timbres across the keyboard range for rich, evolving textures.

- Emulate acoustic instruments

  Many acoustic instruments have distinct tonal characteristics across their range. Assigning oscillators to specific note ranges can mimic this behavior.

  For example, you could consider simulating how low strings sound warmer while higher strings sound brighter on a string instrument.

- Add specific harmonic content to note ranges

  You could create timbral variation by assigning unique waveforms and effects to different note ranges. For example, you could consider using a sine wave oscillator in the lower range for pure bass tones and a saw wave in higher ranges for harmonic richness.

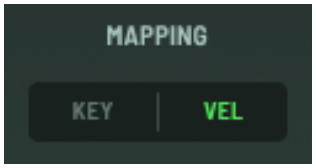
  This could thereby enhance expressiveness by varying the oscillator's contribution based on the note range.

Mapping Velocity Ranges

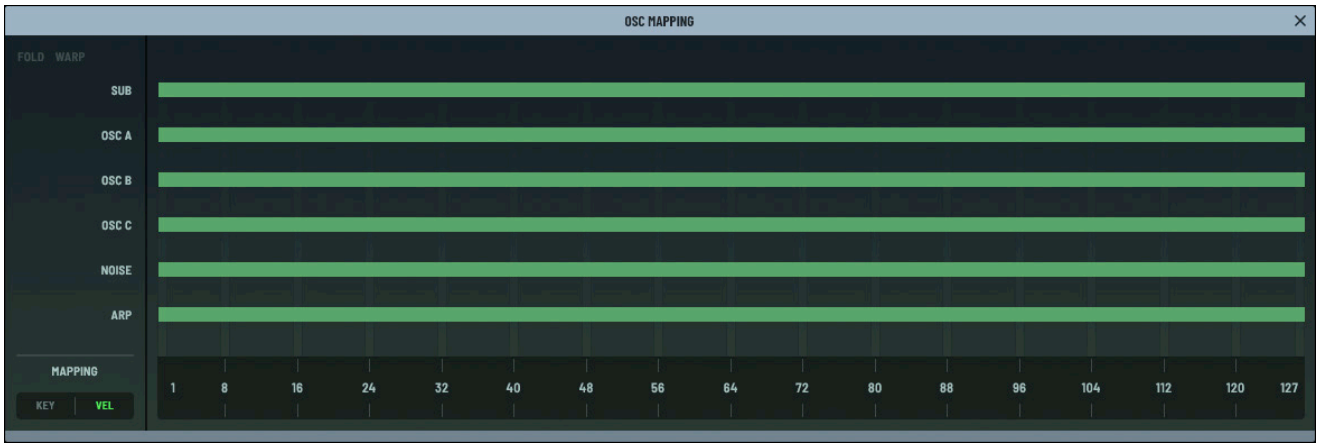
You can edit the velocity ranges of the oscillators and arpeggiator to define and limit the range of velocities that the oscillator or arpeggiator can produce.

Right-click any oscillator header and choose **Edit Note Ranges** in the menu. The **Mapping Editor** appears.

To set the velocity mapping, select **VEL** in the **MAPPING** section (lower left).



Velocity Mapping



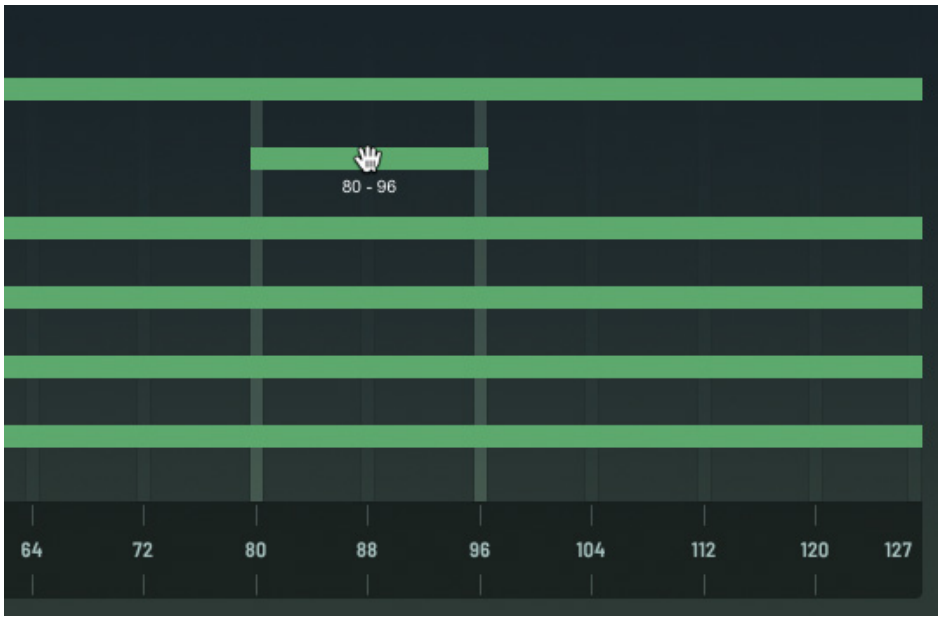
Velocity Mapping Editor

Click and drag the left or right edge of the velocity range for the corresponding oscillator or arpeggiator. The current velocity appears below the cursor, and a highlight shows the velocity in the context of the complete range (1 to 127).

Setting the Velocity Range

To move a velocity range, drag the range to the new location.

The current velocity boundaries appear below the cursor (hand), and highlights show the velocity range in the context of the complete range (1 to 127).



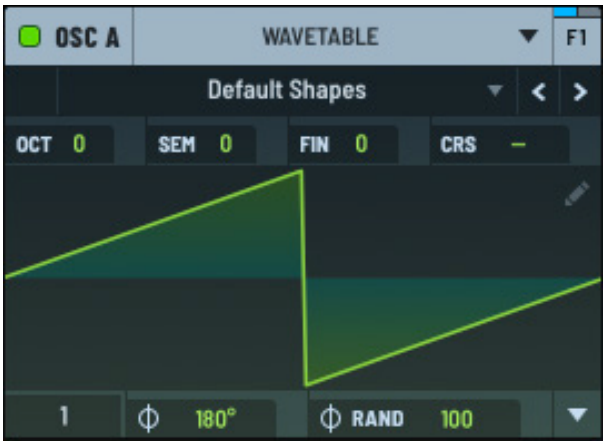
Moving a Velocity Range

Using the Wavetable Editor

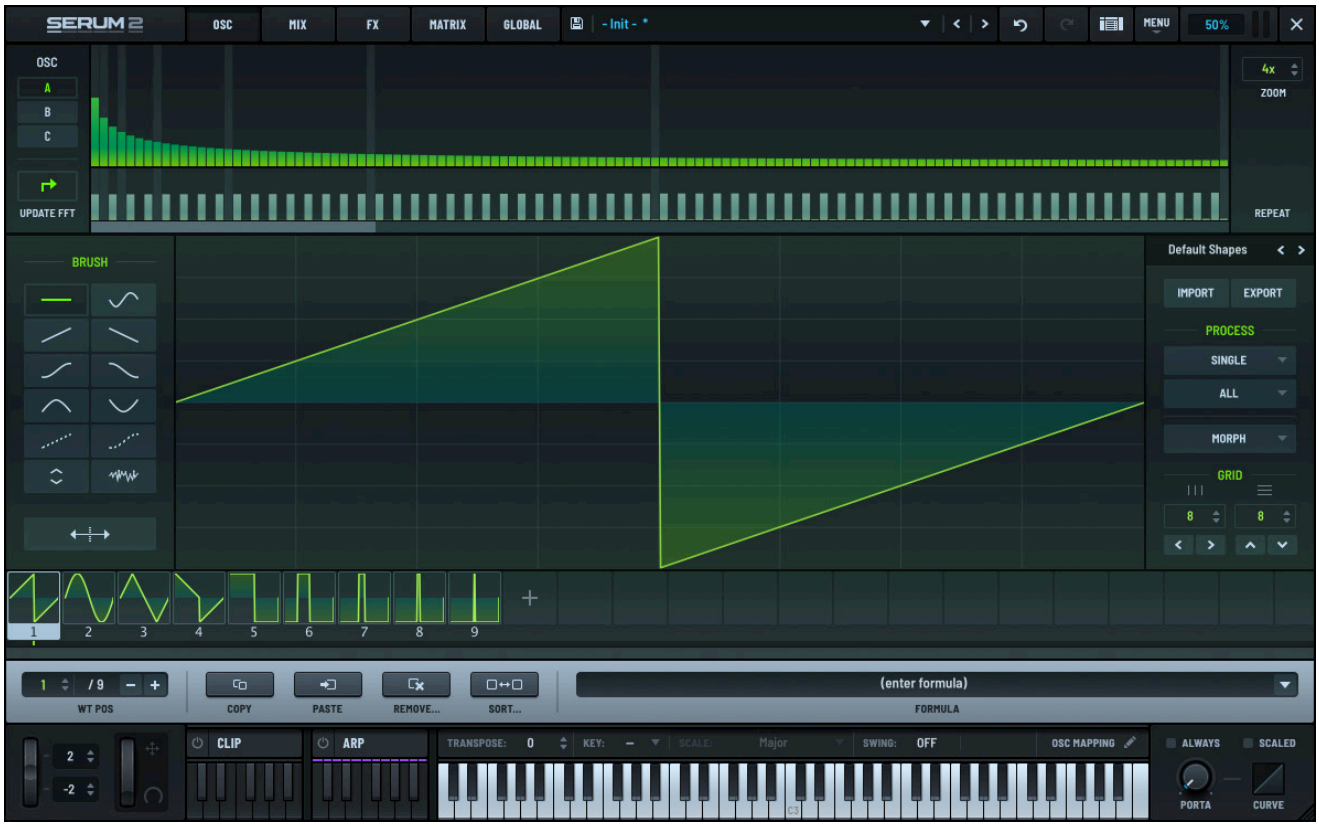
Serum features an advanced Wavetable Editor that makes it fun and easy to create and edit wavetables.

To access the Wavetable Editor, click the  button in the **OSC A**, **OSC B**, or **OSC C** pane.

The Wavetable Editor appears.



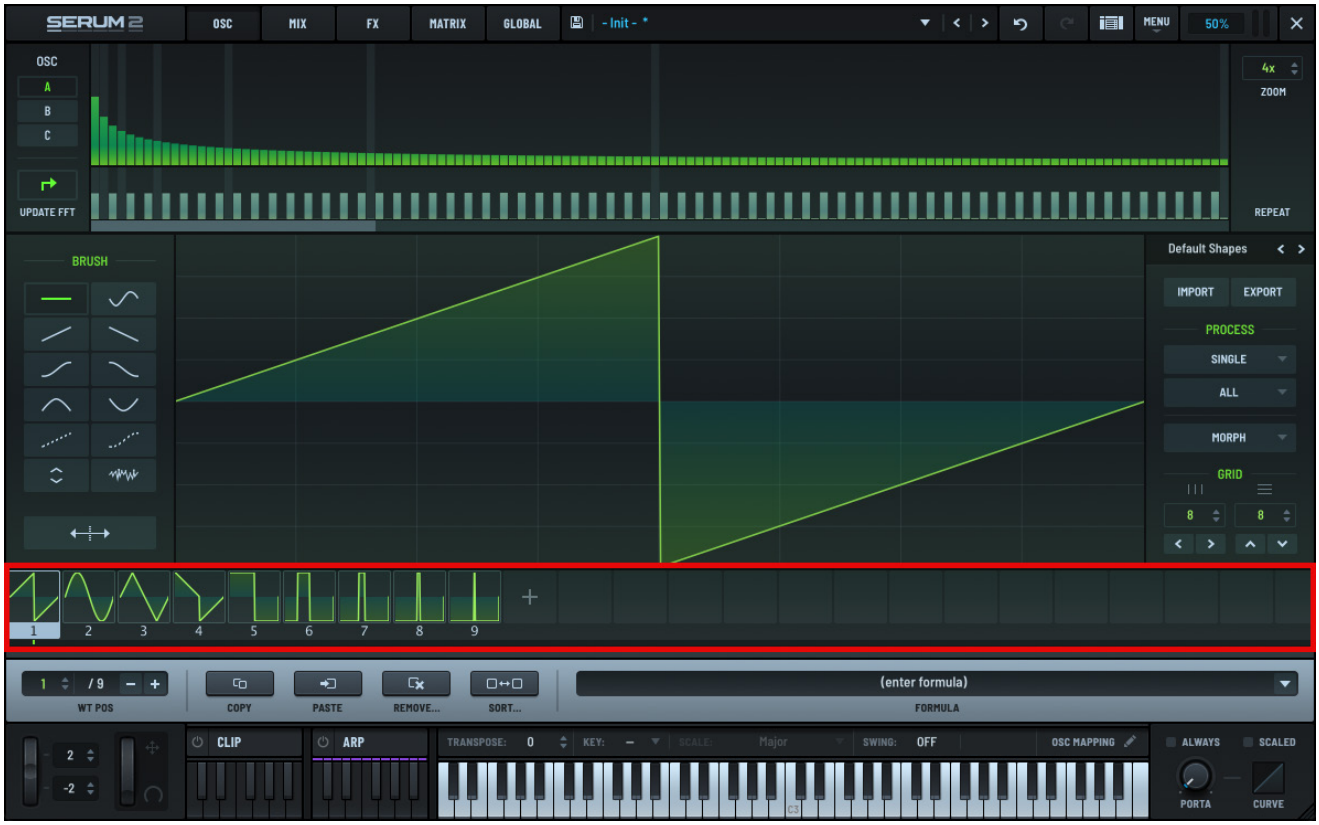
Accessing the Editor



Wavetable Editor

Using the Thumbnails

Thumbnails appear along the bottom of the Wavetable Editor.



Wavetable Editor Thumbnails

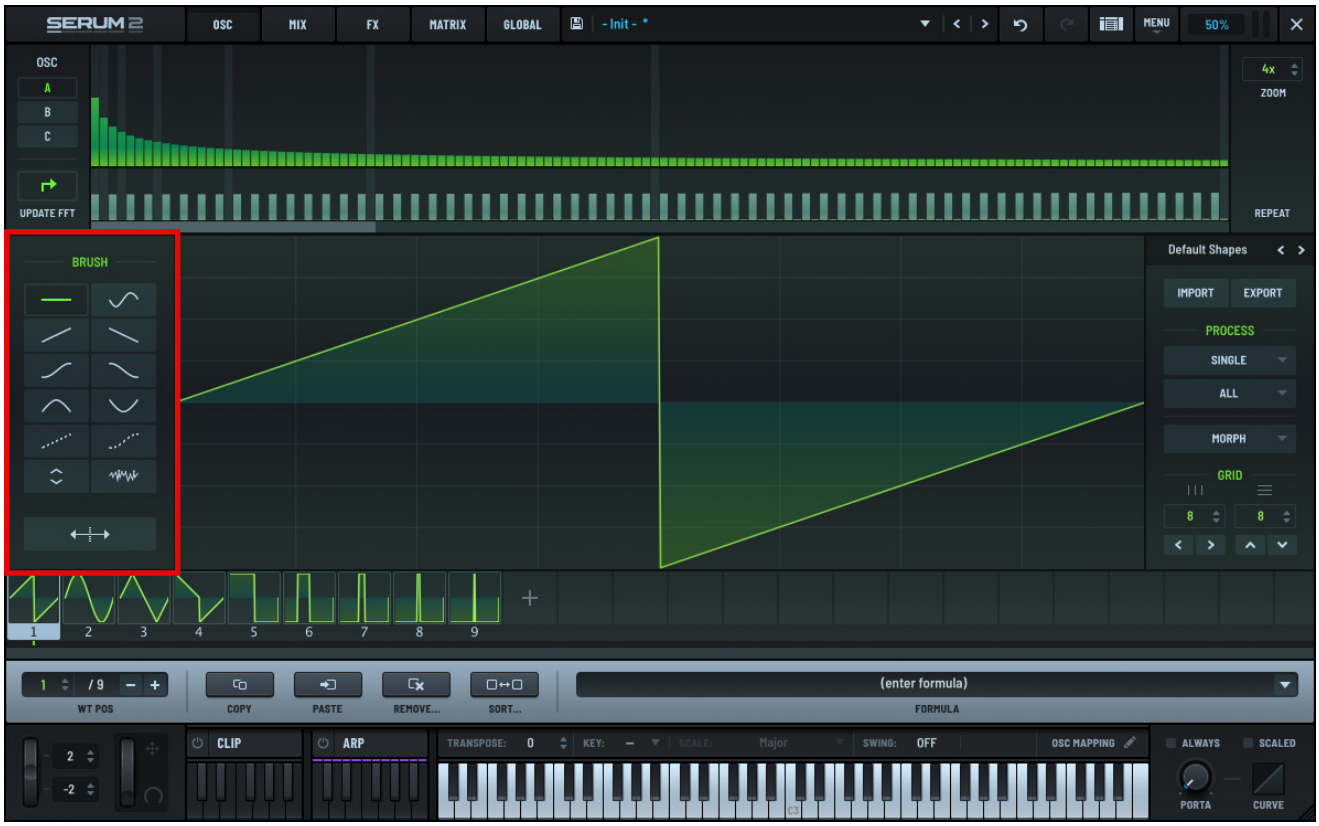
Thumbnail overviews provide an easy way to view, select, and reorder the subtables that make up a wavetable.

You can use the thumbnails to perform a variety of functions:

Thumbnail Action	Description
Click	View and edit the specific frame (subtable).
Click-drag	Move a frame to a new location. As you drag left or right, a yellow vertical cursor appears showing where the frame will relocate after you release the mouse.
Shift-click	Select a range of frames. After selecting a range, you can perform operations such as **Remove: Multiselection** and **Remove All (Except Selected)**, among others.

Using the Drawing Tools

The Wavetable Editor offers a toolbar that you can use to draw in the main display. Each tool operates in relation to the grid size (set in the lower-right corner of the main waveform).



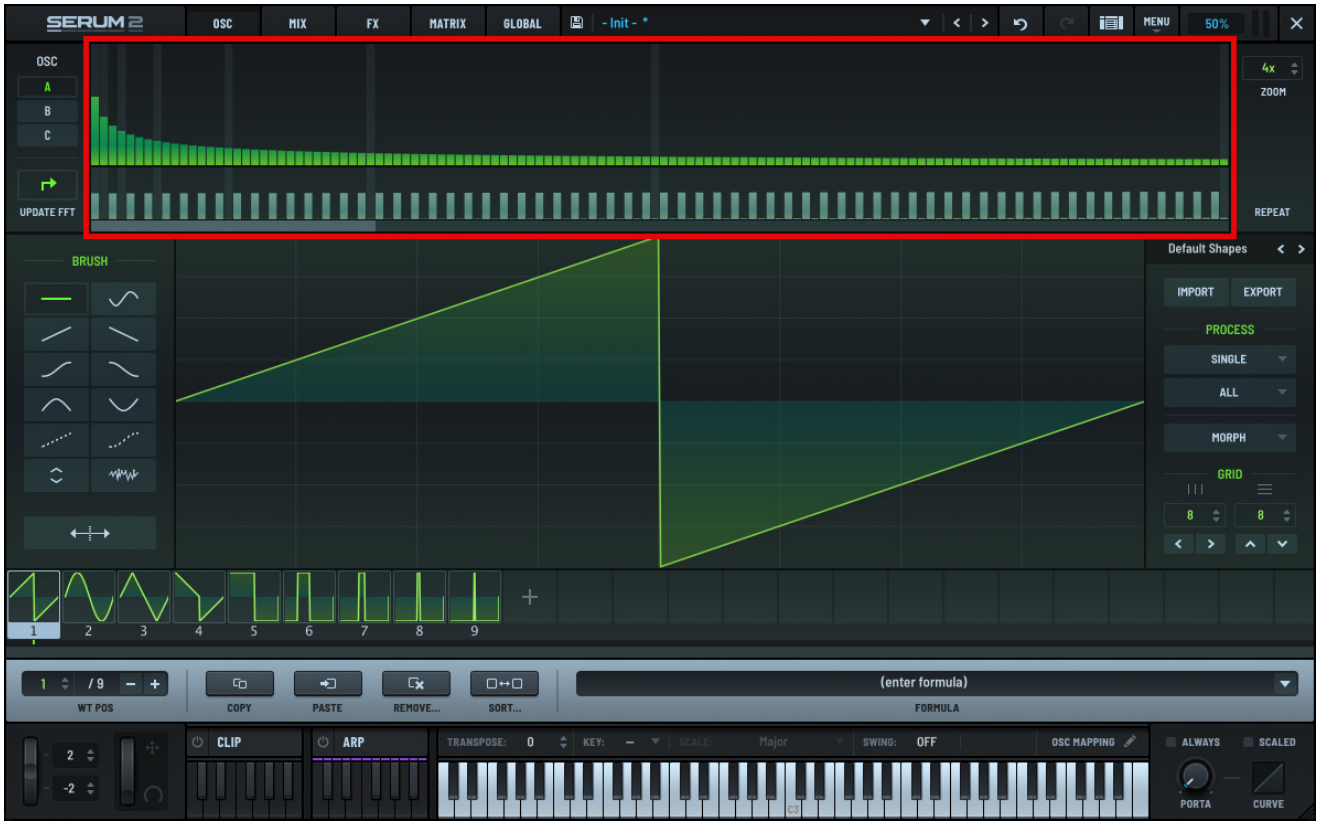
Wavetable Editor Toolbar and Grid Size

Tool	Name	Description
	Flat Line	The default tool. Draw a flat line in the grid step.
	Sine	Draw a sine wave. This is useful for adding a pure harmonic at the grid size. For example, if the grid is set to 4, this tool adds an overtone two octaves above the fundamental.

Tool	Name	Description
	Slope Up	Draw a diagonal line upwards. This is useful for drawing small saws to give your waveform a buzz.
	Slope Down	Draw a diagonal line downwards. This is useful for drawing small saws to give your waveform a buzz.
	Curve up	Draw a quarter-sine waveform.
	Curve down	Draw a quarter-sine waveform.
	Half sine (peak)	Draw a half-sine waveform.
	Half sine (valley)	Draw a half-sine waveform.
	Interpolate linear	Rather than drawing, this tool connects endpoints to draw a straight line across the grid segment. This can be useful for smoothing out a waveform (LPF).
	Interpolate curved	Similar to interpolate linear, however with a more gradual transition (not as much harmonics as linear; even more low pass/smoothed).
	Nudge	Move a portion of the waveform up or down. Use this for clipping (if you click-drag vertically far enough, the audio begins to clip/flat-line).
	Noise	Add noise to the waveform. Dragging vertically up adds noise, while dragging down reduces noise.
	Mirror	Create a symmetrical wavetable drawing by mirroring from the center.

FFT Area

Use the FFT area to draw the waveform harmonics and their relative phases to one another.



Wavetable Editor FFT Area

The top section shows the frequency bins. The left-most bin represents the fundamental or single sine wave at the oscillator's base frequency (the note you play).

The vertical bars to the right represent the harmonics that make up the sound, (2:1 is the octave, 3:1 is +19 semitones, 4:1 equals 24 semitones, and so on).

The bottom section shows the phase offset for each harmonic. Right-click to display the context menu.

Shift-Cmd-click (macOS) or Shift-Ctrl-click (Windows) in the bottom section to nudge the phase offset to the right.

Wavetable Editor FFT Menu

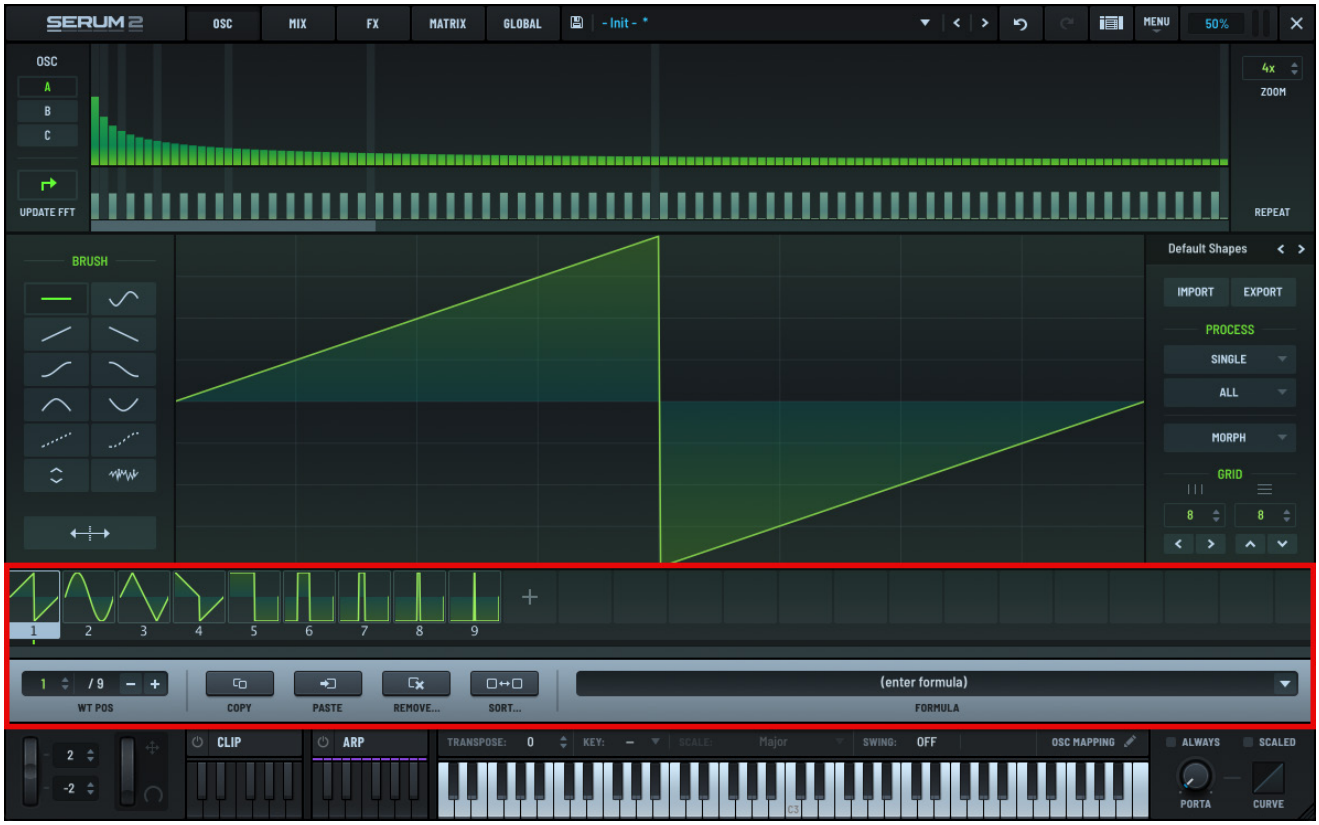
There is a separate menu for the frequency and phase bins.

Menu Item	Description
Clear All	Zero out all bins (silence).
Clear HF (Bin 0 to End)	Zero all bins (silence) to the right of the clicked bin. Since the high harmonics are silenced, you can think of this as a low-pass filter.
Clear LF (Start to 0)	Zero all bins (silence) to the left of the clicked bin. Since the low harmonics are silenced, you can think of this as a high-pass filter
Generate Saw	Set all harmonic bins to relative amplitudes creating a standard saw waveform.
Randomize Low x Bins	Insert random values for the left-most number of bins. This changes the base tone of the sound while leaving the higher-frequencies (“buzz”) intact.

Menu Item	Description
Randomize Low x Bins (with Half)	Insert random values for the left-most number of bins, but allows 50% amplitude for harmonics. This provides more tonal options.
Randomize All	Randomize all bins. This results in a noisy/buzzy tone. This is useful as a starting point for further destruction with FFT drawing or other menu commands.
Create Random Series Gaps	Randomly remove harmonics.
Progressive Fade	Reduce the high frequency content gradually. This is useful for bringing back a natural-sounding tonal balance to an overly bright or buzzy wavetable without resorting to removing the highs altogether.
Shift Octave Up	Spread all bins (1->2, 2->4, and so on) so that the resultant waveform is doubled in frequency.
Shift Octave Down	Same as above, but in reverse.
Repeat Bin Group (Start to Cursor Pos)	Create copies of the harmonics from the start to the current cursor location, repeating all the way through the spectrum.
Draw Even Harmonics Only	Prevent the mouse from operating on odd-numbered bins.
Draw Odd Harmonics Only	Prevent the mouse from operating on even-numbered bins.
Snap Vertical Draw to Quarters	Snap vertical mouse drawing to exactly 0%, 25%, 50%, 75% and 100%.
Scale Freq Values by Bin Index	Amplitude draw/react proportionally to the selected bin, exaggerating higher harmonics visually. Using this feature, a proper sawtooth will be a horizontal line draw instead of an exponential curve across the harmonics.

Managing Frames (Subtables)

You can use the Wavetable Editor to perform a wide range of operations on frames within a wavetable.



Managing Frame (Subtables)

Exploring Common Frame Operations

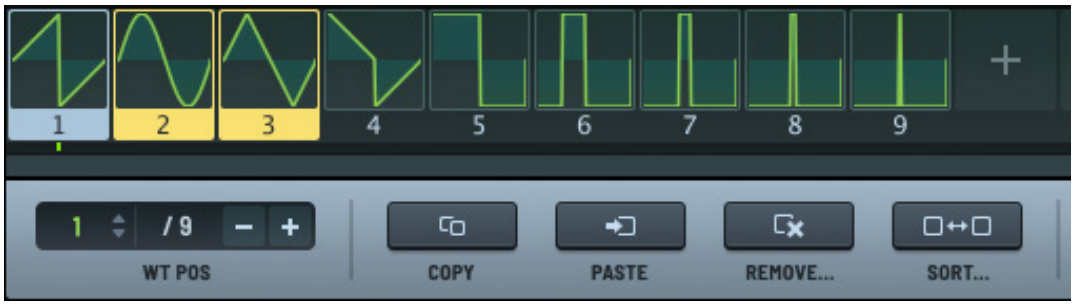
You can access the most commonly used frame operations through a series of buttons and menus directly below the frame display.

Frame (Subtable) Buttons and Menus

Copying and Pasting Frames

You can copy one or more frames and paste the contents to a new location in the wavetable.

Start by selecting the frames you want to copy. You can click to select a frame, and Shift-click to extend your selection. Your selection needs to be contiguous.

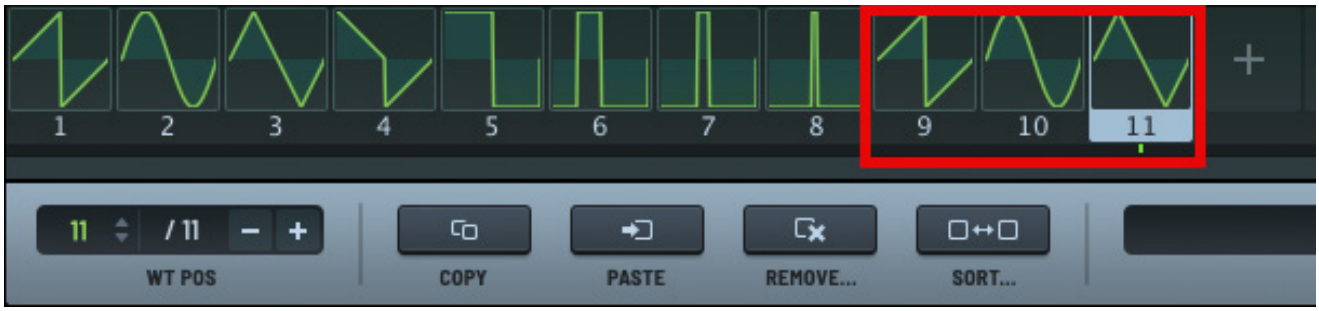


Selecting Frames to Copy

Click the **COPY** button. 

Select the starting frame of the destination and click the **PASTE** button. 

The frames appear starting in the target frame.



Pasting Frames into the Wavetable

Inserting and Removing Frames

You can use the **REMOVE** menu to insert or delete frames. You can also initialize all frames using this menu.

Click the  button and choose an option using the menu that appears. The following table describes the menu options:

Menu Item	Description
Init All (Default)	Clear all frames/interpolations. The wavetable is brought to its default state (a single saw frame).

Menu Item	Description
Insert (at current index)	Add an additional frame to the wavetable set, inserted after the currently-selected frame. This has the same effect as pressing the + button below the waveform.
Remove (current index)	Remove the currently selected frame. This is the same function as pressing the - button (below the waveform).
Remove: multiselection	Remove multi-selected frames (Shift-click in the thumbnail area to multi-select a range of frames).
Remove: beginning-> selected	Remove the frames from the beginning (1) through the selected frame.
Remove: selected-> end	Remove the frames from the selected frame through to the end.
Remove All (except Selected)	A “crop” feature, useful when you decide that a single frame or certain range (multiselect) is all you want to keep.
Reduce to	Thins the number of frames; helpful to easily keep only every nth frame.

Sorting Frames

You can use the **SORT** menu to reorder existing frames based on a spectral property. Keep in mind that you can also manually sort frames by dragging the frames left and right within the thumbnail display.

Click the  button and choose an option using the menu that appears. The following table describes the menu options:

Menu Item	Description
Sort by spectrum (Peak Spect)	Sort frames based on which frame has the highest peak frequency bin.
Sort by spectrum (Average Spect)	Sort frames based on where the average spectral content exists (sum of all frequencies).
Sort by spectrum (Peak Amount)	Sort frames by the highest overall peak (concentrated frequency energy).
Sort by spectrum (Num w/Spect)	Sort frames by how many frequency bins contain energy (spectral complexity).
Sort by spectrum (Highest w/Spect)	Sort frames by the highest frequency bin to contain spectra. This should work well for filter sweeps, for instance.
Sort by spectrum (Fundamental Amt.)	Sort frames by the amount of energy in the fundamental. When in doubt, try this sort first.
Sort Randomize	Randomize the order of the table indices.
Reverse Entire Table Order	Reverse all frames in the table. This is useful if you decide that you want the table order to become bright-to-dull instead of dull-to-bright, for instance.

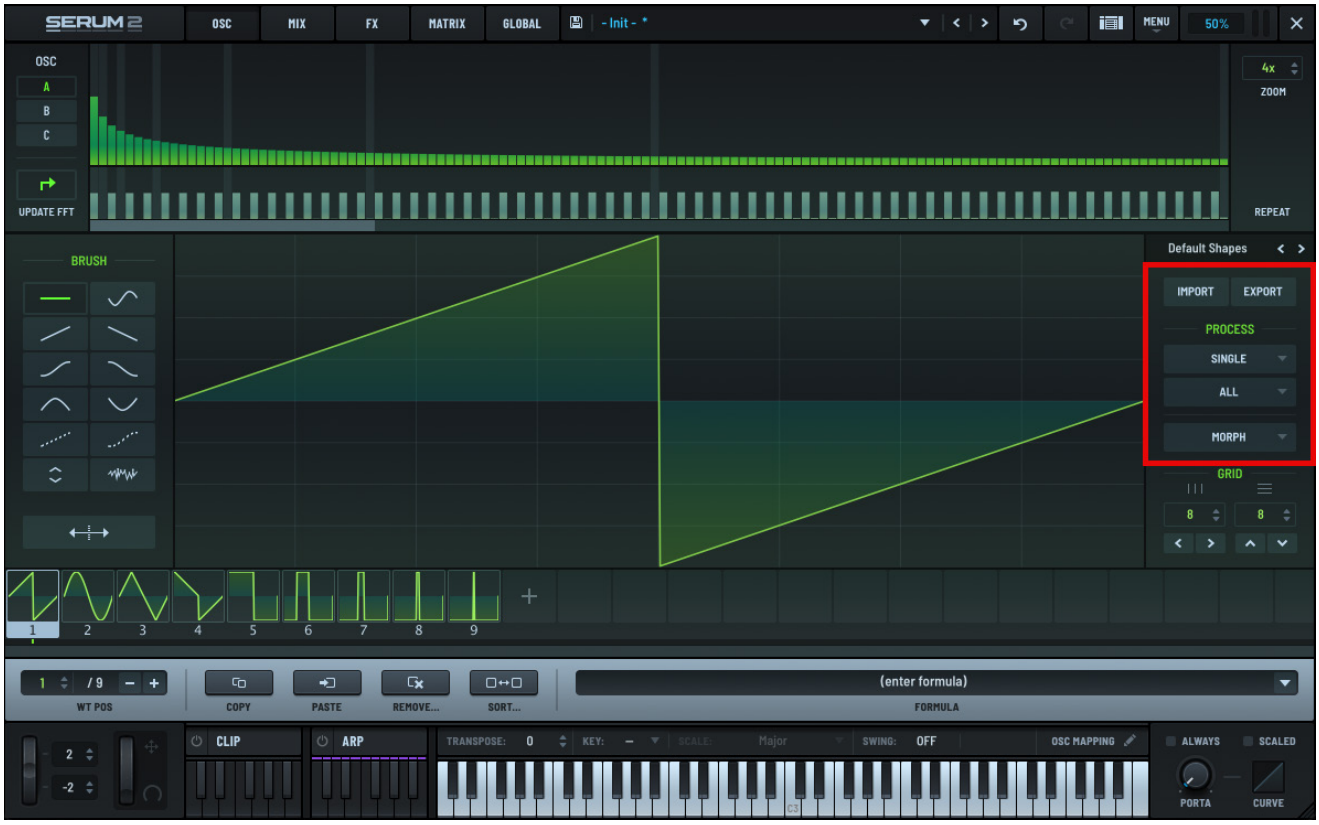
Using the Formula Parser

Use the formula parser to generate a wavetable (single frame or entire wavetable set) from a mathematical formula. You can use the **Formula** presets menu containing multiple examples to get you started.

See “Using the Formula Parser” chapter for more information.

Exploring the Menu Commands

The Wavetable Editor features a set of menus that allow you to quickly and easily manipulate the waveforms.



Wavetable Editor Menu

Importing a Wavetable

You can import a wavetable into the Wavetable Editor.

Click the IMPORT button, and choose an import option in the menu that appears.

A dialog appears allowing you to select a file to load (which overwrites the wavetable set you are currently editing). Refer to the "Importing Audio" chapter for a detailed description of the options in this menu.

Keep in mind that you can also drag-and-drop waveforms directly from the macOS Finder or Windows Explorer. This is often a more convenient way to import sounds.

Exporting a Wavetable

You can export a wavetable from the Wavetable Editor.

Click the EXPORT button, and choose an export option in the menu that appears.

The following table describes the available options:

Menu Item	Description
Export All As 8-bit (.256)	Export the wavetable in 8-bit format. This saves the wavetable using a file format compatible with certain hardware modular synthesizers (such as Wiard and Piston Honda).
Export All As 16-bit (.wav)	Export the wavetable in 16-bit format. This saves the file in WAV format, mono, 16-bit, 44100 Hz, 2048 samples per subtable. Note that additional header information for interpolation mode and interpolation tables are not saved as part of the WAV data.
Export All As 32-bit (.wav)	Export the wavetable in 32-bit format. This saves the file in WAV file format, mono, 32-bit, 44100 Hz, 2048 samples per subtable. Note that additional header information for interpolation mode and interpolation tables are not saved as part of the WAV data.
Export All As Single-Cycle Waves	Export the wavetable as a set of individual single-cycle WAV files.
Export Selection	Export and save a wavetable using the Serum file format (a WAV file with extra header information). This is similar to the other save commands, except that only the selected frames are included in the file.

If you don't see a format compatible with your needs, you can convert WAV files to an alternate format using a third party application or utility.

Single Menu

The **SINGLE** menu contains process functions that alter the currently selected (visible) frames.

Menu Item	Description
Normalize	Adjust the waveform to maximum-scale amplitude.
Remove DC Offset	Remove any DC (up/down bias to the waveform). This is typically not needed.
Flip Vertical (Polarity Invert)	Perform a polarity invert operation (often somewhat mistakenly called "phase invert").
Flip Horizontal (Reverse)	Reverse the audio. As a short looping wave, it probably won't sound reversed in most situations.
Shift Horizontal to Zero Crossing	Nudges/wrap the wave so the edges fall on a zero crossing. This isn't essential, but might be useful for applying fades or matching phase among multiple waveforms.
Init (Silence)	Replace the current table with silence.
Fade Edges (Grid Size)	Create a fade in at the left edge and a fade out at the right edge based on the width of the horizontal grid size (set in the lower right).
X-Fade Edges (Grid Size)	Similar to above, but instead of fading to the center line, the edges fade to each other.
Filter (Grid Size)	Completely remove upper harmonics using a low pass filter. A lower grid size means more low pass filtering. You can achieve the same low pass filter effect using the FFT area (Ctrl/right-click > Clear HF). Since this uses FFT, it is infinitely steep; you might notice some DC shift as all DC offset is removed.
Sample Redux at Grid Size	Perform a sample rate reduction for a lofi sound. The grid size is not used in a literal way but as a means for specifying an amount (larger number offers more sample redux).
Send to Noise Oscillator	Convert the currently selected frame to a single cycle waveform and import the waveform into the noise oscillator.

All Menu

The **ALL** menu offers process functions, similar to the **SINGLE** menu, but applying to all frames (1-256) in the wavetable.

Menu Item	Description
Normalize Each (Gained Separately)	Individually normalize every frame to its own peak maximum.
Normalize Same (Max From All Frames)	The entire wavetable set is scanned for a peak level, and the same gain amount is applied to each frame.
Remove DC Offset	Remove any DC (up/down bias to the waveform). Noet that this is typically not needed.
Flip Vertical (Polarity Invert)	Perform a polarity invert operation (often somewhat mistakenly called "phase invert").
Flip Horizontal (Reverse)	Reverse the audio. As a short looping wave, it probably won't sound reversed in most situations.
Fade Edges (Grid Size)	Create a fade in at the left edge and a fade out at the right edge based on the width of the horizontal grid size (set in the lower right).
X-Fade Edges (Grid Size)	Similar to above, but instead of fading to the center line, the edges fade to each other.
X-Fade Edges (16 Samples)	Same as above, but only a micro-fade on the edges (16 samples is small!).
Filter (Grid Size)	Completely remove upper harmonics using a low pass filter. A lower grid size means more low pass filtering. You can achieve the same low pass filter effect using the FFT area (Ctrl/right-click > Clear HF). Since this uses FFT, it is infinitely steep; you might notice some DC shift as all DC offset is removed.
Remove Fundamental (HPF)	Remove the lowest frequency (pitch), effectively akin to zeroing the top-left bar of the FFT display (fundamental) but for all tables. The formula "z=(q>1)?in:0" would also yield the same result, or simply "(q>1)?in:0" if you want to only apply to the currently visible frame.
Remove Low Spectra (Grid Size)	Remove the lowest frequency bins, somewhat similar to a high-pass-filter.
Remove Low Phases (Grid Size)	Zero the phases on the lowest frequency bins.
Sample Redux at Grid Size	Perform a sample rate reduction for a lo-fi sound. The grid size is not used in a literal way but as a means for specifying an amount (larger number offers more sample redux).

Menu Item	Description
Resize Tables to be Half (2x Total Number)	Divide every frame in half to become two frames. This doubles the number of frames (so you shouldn't use this if you have more than 128 frames).
Resize Tables to be Double (½ Total Number)	Every frame is grouped with the following frame. This is useful if you notice every second frame similar after a WAV import (the frequency was detected an octave too high).
Create PWM from This Table to All	Remove all frames except the current frame and create a PWM shift across all 256 frames. There is a real time PWM effect using the WARP knob (main panel), but this way you can perform pulse width modulation and use another warp effect.
Nudge All Phases for Fundamental to 50%	Preserve the phase of the lowest frequency (fundamental) across various waveforms.
Set Spectra (This Frame to All)	Apply FFT bins (spectrum) from the current frame to all frames.
Set Phases (This Frame to All)	Apply the FFT phase information from the current frame to all frames. This is useful to make all frames match in phases beyond just the fundamentals, for example, solid/consistent sound during morphing.
Set Spectra from Other Osc	Remove the relative amount of harmonic content contained in the other oscillator.
Set Phases from Other Osc	Apply the FFT phase information from the other oscillator frames (A<>B) to the frames of the visible oscillator.
Subtract Spectra from Other Osc	Subtract harmonic content based on the harmonic content of the other oscillator's wavetable.
Blur Spectra - Adjacent Bins (Grid Size)	Interpolate (smooth) the harmonic content between adjacent harmonics. The grid size value determines how many neighboring harmonics are factored into the smooth operation.
Blur Phases - Adjacent Bins (Grid Size)	Interpolate (smooth) the phase content between adjacent harmonics. The grid size value determines how many neighboring harmonics are factored into the smooth operation.
Blur Spectra - Adjacent Frames (Grid Size)	Interpolate (smooth) the frequency content between adjacent frames. The grid size value determines how many neighboring harmonics are factored into the smooth operation.
Blur Phases - Adjacent Frames (Grid Size)	Interpolate (smooth) the phase content between adjacent frames. The grid size value determines how many neighboring harmonics are factored into the smooth operation.
Shift Horizontal to Zero-Crossing	Move the audio data to the right to have the left edge of the frame begin at a zero crossing (where the waveform polarity changes from negative to positive).

Morph Menu

Use the **MORPH** menu to create or remove interpolation frames between existing frames. Note that you must have more than 1 and less than 256 frames.

After selecting a morph item from the menu, you'll notice that there are now 256 frames in the **WT POS** selector. This is because all empty wavetable slots are now filled with interpolated (blended) waveforms of the neighboring tables.

You'll also notice that the thumbnails no longer display "1, 2, 3", and so on, but rather "1, 17, 33" (or similar). This is because the intermediate tables (for example, 2-16 in this example) contain the newly-created morph tables.

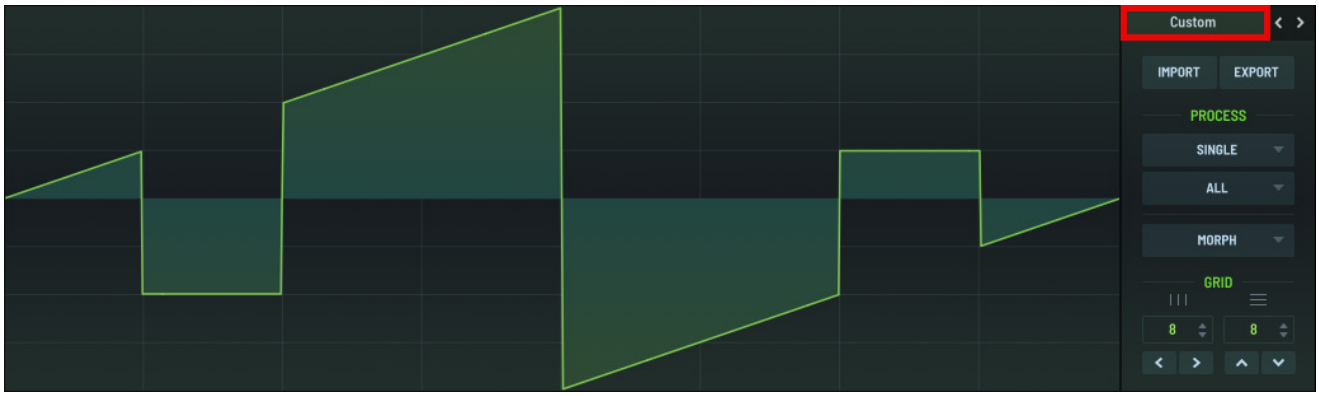
You'll see these intermediate tables in the waveform area as **WT POS** is adjusted, but they don't exist as thumbnails.

After you exit the Wavetable Editor and return to the main panel, you'll also notice the interpolated frames appear gray in the 3D wave overview (whereas green shows the "real" frames, and yellow shows the currently-selected frame).

Menu Item	Description
Morph - Crossfade	Create interpolated frames by crossfading the neighboring frames together. This is the recommended default, and what traditional wavetable synths do.
Morph - Spectral	Use the spectral and phase content of neighboring frames to re-synthesize the interpolated frames. This is what additive synthesizers do.
Morph – Spectral (Zero Fund. Phase)	Same as Morph - Spectral, but the phase content of the fundamental is zeroed for all source frames. This way the lowest frequency does not shift/rotate between frames.
Morph – Spectral (Zero All Phases)	Same as Morph - Spectral, but all phase content is discarded. This might alter the sound of the source content drastically, and therefore can sometimes be undesirable. However this option also creates the smoothest transitions between frames since no frequencies need to shift phase.
Remove Morph Tables	Revert back to how things were before interpolation was applied. Note that clicking "Undo" might be the better choice for reversing the morph since, in the case of the spectral (zero-phase) modes, the two zero-phase choices destructively alter the source tables.

Saving Wavetables

If you have modified one or more frames in the Wavetable Editor, the wavetable name changes to a tinted background indicating that the wavetable has been modified but not yet saved to storage.



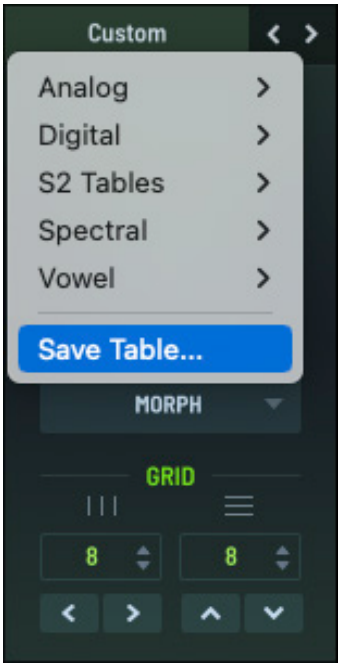
Wavetable Modified

Right-click **Custom** and choose **Save Table** in the menu that appears. Choose the file name and location in the dialog that appears.

Save your wavetables in the **User** folder and, if at all possible, do not overwrite factory wavetables (or else the presets may sound different). Also, when in doubt, always pick a new name.

Note that you do not need to save your wavetables explicitly. Serum always saves changes you make to wavetable data inside a preset (your song) unless it is a factory wavetable. This uses some hard disk space — how much depends on how many frames you use in the wavetable, from 8k to 4 Megabytes

The benefit, however, is that you can exchange presets with others, or open your song in the future, without having to worry about table file management. The only reason to explicitly save a wavetable is to have it appear in the Wavetable menu.



Saving a Wavetable

Importing Audio as Wavetables

A great way to obtain high-quality source material for wavetables is by importing audio data. This can be done using various methods, which can generally be categorized as either single-cycle and multi-cycle imports.

It's important to realize that wavetables do not behave like samples in a sampler. When importing audio to create a wavetable, you shouldn't expect to create ultra-realistic reproductions of acoustic instruments. Use **Sample** mode (in any of the oscillators) to work directly with source samples.

Understanding Multi-cycle Waveforms

Most real sounds you come across (such as speech) consist of multiple waveform cycles. In a human voice waveform, you can see the repetitive nature of sound, which becomes the pitch we hear. You've likely seen this before when manipulating audio in a DAW.

Samplers typically play audio as a single stream of data. In contrast, when importing a sound into Serum, it attempts to slice the sound into individual single-cycles. Because of the nature of audio, it's generally best to select source sounds that are monophonic (that is, sounds that contain a single pitch). This means that, in most cases, a single note is better than a chord.

Because these single cycles become the basis of the oscillator, pitch information is effectively removed. In other words, if you load a sound that has a pitch bend, the pitch bend will no longer exist. However, wavetables have their own set of advantages, which one could describe as a "solid" or "fixed" sound that lends itself well to unison and wave manipulation (sync, FM, and more) without sounding flimsy.

Serum therefore does a fantastic job of importing the waveform of other sounds. Fixed-pitch, one-shot (monophonic) sounds, such as a one-shot sample of a synthesizer, are among the best choices for importing into Serum. Speech and other complex sounds can however yield some pretty interesting results if you're open to some experimentation.

Importing Multi-Cycle Waveforms

To import a multi-cycle WAV file, drag the audio file from the macOS Finder, Windows Explorer, or host DAW file browser to the waveform display on Serum's main window.

Note: Dragging files directly from the host arrangement window or "region bins" is not possible in most hosts, but using the host's standard file-browser should work. In addition, many hosts are able to show the (parent) sound file in the host file browser, and you should be able to drag from there.

When importing stereo files as a wavetable, Serum uses the left channel in **OSC A** and **OSC C**, and uses the right channel in **OSC B**.

Import Audio Options

As you drag the waveform, you'll see several options appear. The location where you release the mouse determines the import method. Serum then analyzes the sound and creates a new wavetable in memory. The analysis specifics depend on the import method you choose.

The following describes the import options:

- DYNAMIC PITCH - ZERO SNAP

  Serum scans the audio file and builds a pitch map. Serum then attempts to locate zero-crossings within the pitch map. While this works well with simple sounds, complex sounds typically don't adhere to sensible zero crossings, so you'll end up with at least some glitches at best.

  Use this mode when you have a sound with a non-fixed fundamental (pitch bend or vibrato) and the sound is pretty simple, for example a sawtooth wave with little filter sweep/resonance.

- DYNAMIC PITCH - FOLLOW

  Similar to the previous option, Serum builds a pitch map and imports a varying-sized segment of audio for each frame (subtable) based on the analyzed pitch.

  Unlike the previous option, pitch follow import does not attempt to locate zero crossings. This option is therefore better suited for complex sounds, such as a source sample that might have a touch of chorus/unison, resonance, or background noise/notes.

- FREQUENCY ESTIMATION

  Serum analyzes the incoming waveform to determine its dominant frequencies and harmonic content, and uses this information to convert the audio into a wavetable. This involves analyzing the audio signal to identify the fundamental frequency, and then determining the harmonics (integer multiples of the fundamental frequency).

  By matching the detected fundamental frequency to a musical pitch (for example, C4 or A3), Serum can align the wavetable content to the correct pitch. Serum then uses the estimated frequencies to guide the conversion of the audio signal into a series of wavetable frames. This ensures that the harmonic structure and frequency content of the original audio are preserved within the wavetable, maintaining the character of the original sound.

  Frequency estimation is particularly beneficial for audio samples with well-defined pitches and harmonic structures. These types of samples allow Serum to accurately extract fundamental frequencies and harmonics, making it easier to convert the sounds into expressive and playable wavetables. Examples include monophonic synth leads, bass sounds, vocals with sustained notes, plucked and struck instruments, and FM (frequency modulation) generated sounds.

- Constant framesize (PITCH AVERAGE)

  If in doubt, try this one first!

  This option is typically the best choice when a sound has a fixed frequency, such as a one-shot from a synthesizer (in other words you hear it as a perfect or near-perfect constant pitch, with essentially no pitch bend or vibrato).

  In this mode, Serum analyzes the entire file for an average pitch, and then uses this number of samples as the import length. Because some sounds contain half-cycles, silence, multiple notes, and other artifacts, Serum might not correctly guess the desired pitch.

  Fortunately, Serum displays the number of samples it is using per frame in the Wavetable Editor formula area and switches over to this “fixed” value found during analysis (unless changed or cleared from the Formula field).

- FFT 256/512/1024/2048

  FFT, or Fast Fourier Transform, is a method that converts a signal from the time domain to the frequency domain, revealing its frequency components. It efficiently breaks down a complex waveform into individual sinusoidal frequencies, enabling analysis of amplitude and phase at different frequencies.

  Unlike the other import modes, which import and divide (and possibly stretch) chunks of the original waveform, the FFT modes are a spectral import. This means that these import modes divide the source audio into small snippets of time, and analyze the spectral content.

  One way of thinking of this is a “blurred averaging of the frequency content”. This can be very useful for sounds such as drum loops, speech, and other material where you want the flavor of the sound for abstract purposes.

The numbers 256, 512, 1024, and 2048 represent the number of samples used to perform the FFT analysis. FFT 256, for example, analyzes 256 samples at a time.

Larger FFT sizes, such as 1024 or 2048, provide higher frequency resolution because more samples are considered during the transform. This means you get a more detailed breakdown of frequency components, making it better for detecting specific tones and harmonics.

Smaller FFT sizes, including 256 and 512, provide lower frequency resolution, which results in a broader view of the frequency spectrum. This can be less detailed but faster in terms of processing.

- Switch OSC type

  You can choose to import the audio as a regular sample, automatically switching to the Sample, Granular, or Spectral oscillator modes. In these cases, the audio is imported directly without creating a wavetable.

Advanced Imports

Due to the complex nature of audio signals, using a pitch average might not always be perfect. Sometimes you might want to specify the exact number of samples for each cycle (that you can measure yourself in a sample editor).

The Wavetable Editor includes a formula parser (described in the next chapter) featuring a text box with the placeholder text "(enter formula)". Although the primary function of this field is to create waveforms from functions, you can also type the following:

- A one to four-digit number (such as 1024) to instruct Serum to split the sound file into segments with this number of samples.
- A MIDI "note name" (such as B0, C#2, D6, and so on).

  Serum converts the sound file into the appropriate nearest number of samples (rounding to the nearest whole number of samples).

  This MIDI note/sample conversion assumes a 44,100 Hz source sound file. If your source sound file is not 44,100 Hz, the number of samples is still held true, but you should disregard the displayed frequency/note information in the formula text area.

Note: Since musical note frequencies are not often perfectly divisible by a sample rate, many pitches will have some visible "drift" on import.

When creating a wavetable from imported audio, you might hear undesirable buzz from subharmonics or other factors caused by the beginning and end of the wave cycle not lining up perfectly.

Therefore, after importing using one of the above methods, try using a fade command from the Process menu. For example, **X-Fade Edges (16 Samples)** or **X-Fade Edges (Grid Size)** should give you a less buzzy-sounding result.

Text File Overrides for Specific Results (Advanced)

When dragging an audio file to import, Serum notes the location of the file you dragged and then looks for a file with the same name but with a **`.txt`** file extension containing audio file information.

For example, if your audio file is **`sound.wav`**, Serum looks for the **`sound.txt`** file. If Serum can't find the matching file, Serum then looks for the **`FolderInfo.txt`** file in the same folder. The **`FolderInfo.txt`** file contains information that applies to all sounds in the folder; this saves you from having to duplicate the same text file for every WAV file in the folder.

The format of the text file is very basic.

The first line specifies the number of samples-per-cycle of the audio file (in square brackets). Valid values are numbers between 32 and 9999. The second line specifies whether to create intermediate interpolated tables. The default is yes; use [no interp] to specify the opposite.

The following table shows some examples:

Samples per Second	Description
[512]	Specifies a 512 samples-per-cycle waveform with crossfade interpolation.
[768] [interp]	Specifies a 786 samples-per-cycle waveform with crossfade interpolation. Note that the second line is optional, since it is the default.
[1024] [no interp]	Specifies 1024 samples-per-cycle with no interpolation.

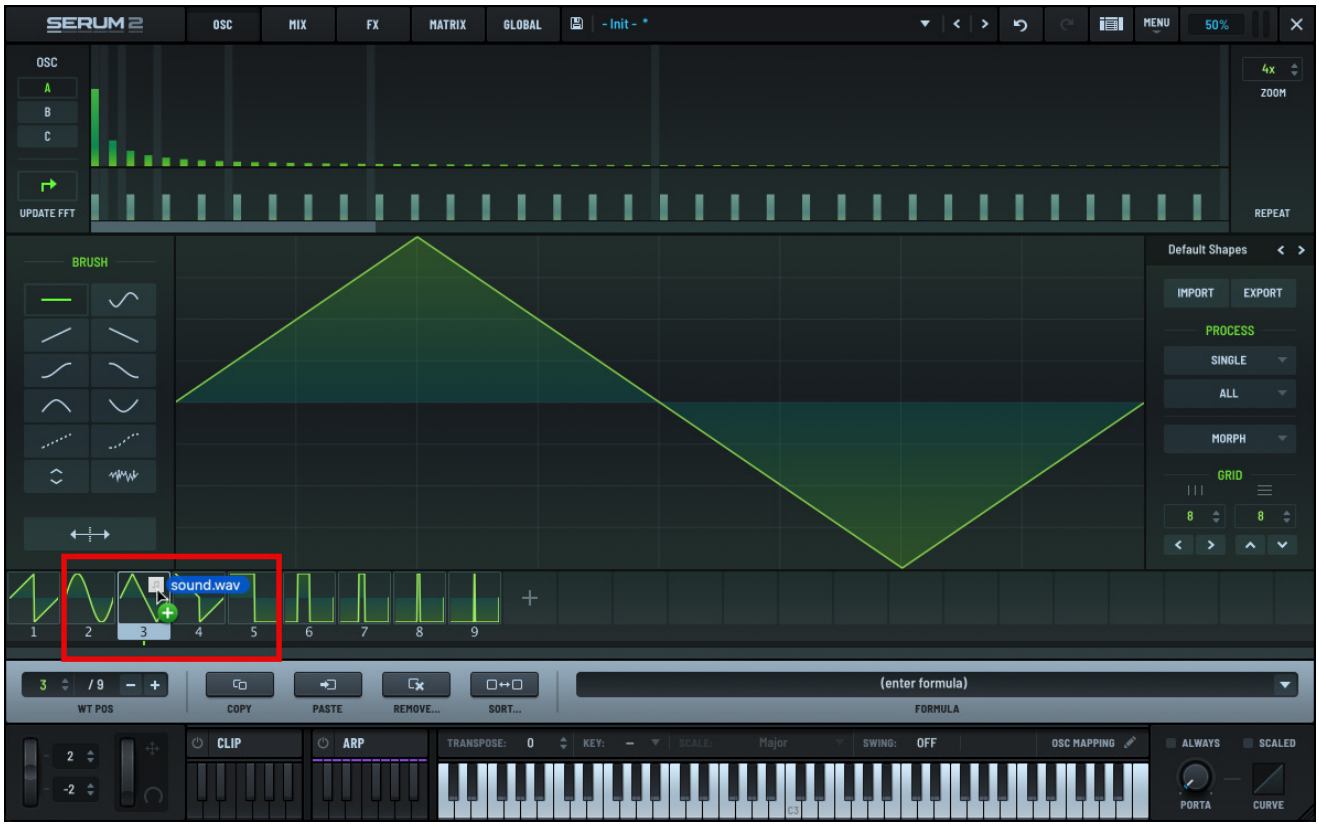
Internally, Serum uses 2048 samples for each single-cycle. This makes 2048 the ideal number of samples to use as a wave cycle if you plan on generating sounds from synthesizers, vocoders, and so on, to import into Serum (because no resampling of the source audio is needed).

This 2048 samples-per-cycle works out to 46.875 Hz at 96000 KHz sampling rate (96000/2048 = 46.875), which is F#0 +24 cents. Since the octave might differ in various hosts and synthesizers, render and measure in a program that allows you to select in samples, such as the freeware sound editor Audacity).

Type 2048 in the **Formula** field prior to dragging to import. Serum will then not have to alter the sample data at all.

Importing Single-Cycle Waveforms

To import a single-cycle waveform, (or import a short sound to be interpreted as such), drag an audio file from the macOS Finder, Windows Explorer, or the host DAW file browser to the corresponding frame thumbnail in the Wavetable Editor.



Importing a Single-Cycle Waveform

This replaces the waveform in the specific frame; the rest of the wavetable remains untouched.

Single-Cycle Waveform Imported

To import multiple single-cycle waveforms at once, drag them together to the oscillator waveform view.

Serum notices that you dragged multiple files and treats them as single-cycle waveforms.

This replaces your current wavetable with a new one that consists of the files you dragged.

Importing Multiple Single-Cycle Waveforms

Using the Import Menu

The **Import** menu in the Wavetable Editor offers the same import options as the drag-and-drop operation. Use this method if you prefer a standard file browser dialog instead of dragging and dropping files.

Importing Audio using the Import Menu

Creating a Sound Specifically for Serum Import (Advanced)

Use this procedure when you want to import a waveform (sound) into Serum with maximum quality from another synthesizer to use as a wavetable.

Whenever possible, use low-frequency notes. Also, use 2048 samples-per-cycle, which precludes the need for Serum to interpolate the sound.

Alternatively, use low C at 44,100 Hz (33 Hz, 1349 samples per cycle). This is slightly easier to set up since no fine tuning of pitch is needed.

The following procedure shows an example of how to export a 2048 samples-per-cycle wave from another synthesizer and import it into Serum.

1. Determine the appropriate source note (pitch).

   Open the Serum Wavetable Editor and type 2048 into the Formula text box. Serum displays the following in the field:

   `split at: 2048 samples (22 Hz, note: F1 and -22 cents)`

   This suggests that you play a low F with -22 cents to create the proper length of 2048 samples per cycle.

2. Render a file playing this pitch from the other synthesizer.

   Do the following:

   a. Create the F1 note in your host sequencer piano roll on the synthesizer track.

   Note that octave naming might be different since, unfortunately, there is no standard. You should probably use a spectrum analyzer to verify ~22 Hz is the fundamental, not ~11 Hz or ~44 Hz.

   As for the duration of the note, since Serum can handle 256 wave cycles, you can render this 22 Hz file for up to 11.6 seconds, which is about 8 bars at 165 BPM. If, on the other hand, the sound has very little modulation/change, there is no need for such a long note. You can probably determine by ear when the source sound has stopped changing; there is no need for a longer note.

   b. Set the fine tuning to -22 cents.

   How you do this varies depending on the synthesizer. Typically, this is set per-oscillator, but some synths have a global tuning.

   If you can't do this in the source synth, consider adding the appropriate amount of "pitch bend" down as a MIDI message (22% down if pitch bend range is 1 on the source synth). A=440 to approximately A=434.5 Hz is another potential option.

   c. Render (export) this note from the synth as audio. Alternatively, you can freeze and flatten to produce the synth note as an audio file.

3. Drag the rendered audio file into Serum.

   If the value 2048 is no longer visible in the Formula text field (for example, if you closed the Serum window after starting this procedure), type 2048 into the field.

   When you release the mouse, the audio is imported. The source audio file (render) is no longer needed at this point unless you wish to re-import it into Serum with different settings.

Note the following:

- You will likely have blank or unnecessary tables at the end of the wavetable.

  Using the thumbnails, find the first table you don't want to include. Click the  button and choose **Remove: Selected->End** in the menu that appears.

- You might want to apply fades to the tables (use one of the **Fade** or **X Fade** options using the Wavetable Editor **Process** menus) to eliminate any buzzing artifacts from frame edges.

  If your source sound has no subharmonic content, this shouldn't be necessary.

- Listen to the entire wavetable set by playing a note and dragging the **WT POS** number box in the Wavetable Editor (or, similarly using the **WT POS** knob on the main panel for the oscillator).

  Alternatively, for automatic playback, you can use a mod source (for example, an envelope with long attack or an LFO with an upward saw shape) to modulate the wavetable position.

Importing an Image File as a Wavetable

One of the most experimental ways to create a wavetable in Serum is to import an image file. When importing the file, Serum automatically maps the pixel luminance to the amplitude, creating largely unpredictable results that can sometimes lead to happy sonic accidents.

The process is extremely simple.

Drag a PNG file (8-bit depth) from the Finder (macOS) or Explorer (Windows) to a wavetable oscillator.

This can be either a color or grayscale image.

The width of the image (in pixels) determines the number of frames in the wavetable (to a maximum of 256).

In a grayscale image, black (luminance value 0) represents silence (no amplitude). White (luminance value 255) represents maximum amplitude.

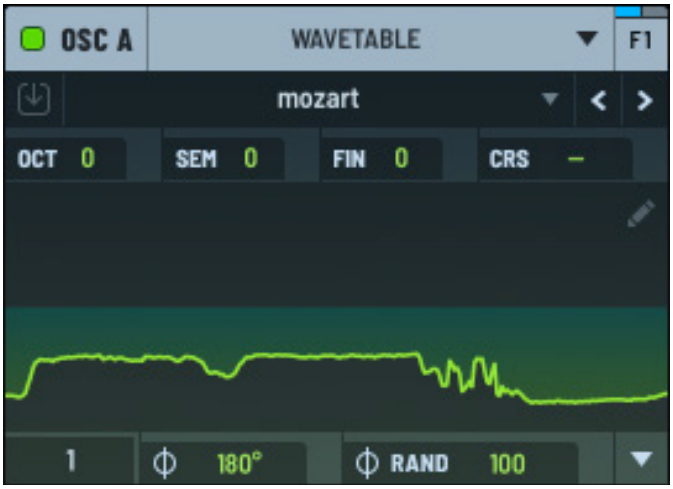
Shades of gray interpolate between these extremes.

In general, gentle gradients create smoother waveform transitions. Unique shapes produce genuinely experimental sounds.

Importing a PNG File

The first frame of the wavetable appears in two dimensions.

As expected, the waveform has no recognizable relationship to the source image.

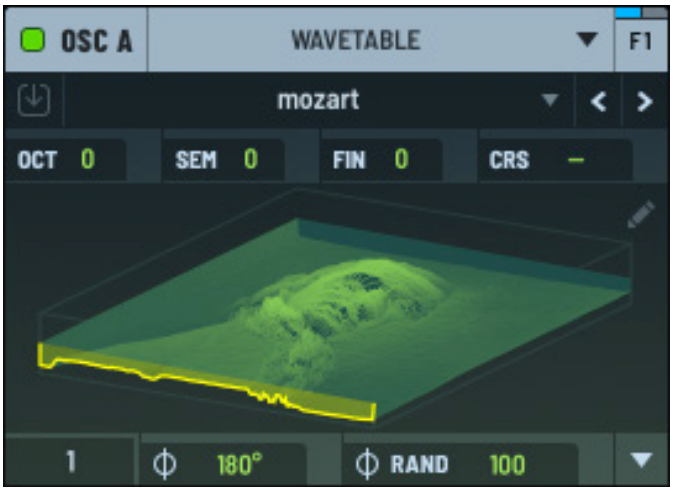


Imported Wavetable (2D View)

Switching to 3D however can sometimes show a slight semblance.

Open the Wavetable Editor to further process the wavetable. You can normalize the frames (to ensure that the frames are balanced), crossfade edges (to smooth transitions between frames), and filter the frames, among other operations.

See “Using the Wavetable Editor” on page 274 for detailed information.



Imported Wavetable (3D View)

To automate a sweep through the wavetable, draw an LFO ramp and drag the **LFO** tab to the **WT POS** knob.

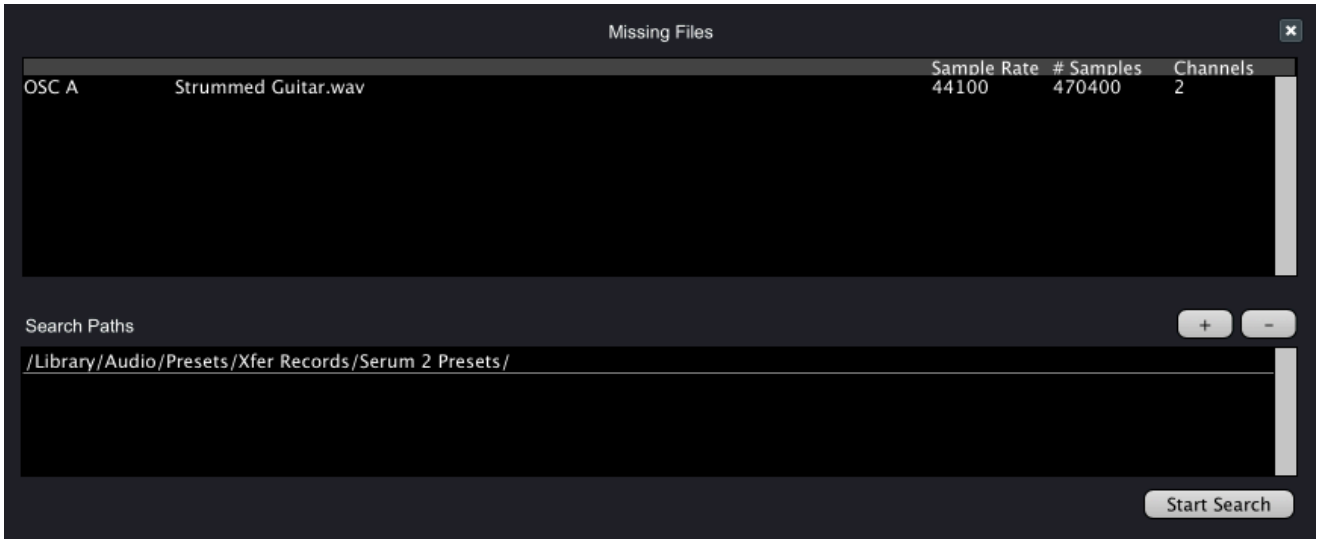
This modulates the wavetable position moving from frame 1 to 256.

LFO Ramp

Embedding Wavetables When Saving a Preset

When saving a preset that includes a custom wavetable created from imported audio, Serum saves the file path of the audio sample along with the preset. By default, Serum does not store the audio sample (and corresponding wavetable) within the preset file itself.

This means that if you move the audio sample to a different location on your drive, Serum will not be able to locate the file automatically. When you try to load a preset that has missing audio files, Serum displays a dialog similar to the following:



Missing Files Dialog

You have two options. Do either of the following:

- Copy the missing audio file to a folder within the current search path and click the Start Search button.
- Click the + button to include the folder containing the audio file in the search path, then click Start Search

To simplify file management, Serum also allows you to embed the wavetable generated from the imported audio into the preset when saving.

After importing the audio file and creating the custom wavetable, close the Wavetable Editor and return to the oscillator display.

Click the  button (near the top left of the oscillator display).

The button becomes enabled. 

Alternatively, click the menu and choose **Embed in Preset**.

This has the same effect and enables the  button.

Save the preset.

The custom wavetable is now saved with the preset. You no longer need to manage the location of the audio file.

Embed in Preset

Using the Formula Parser

You can use the Formula parser to make a sound using math!

Specifically, the Formula parser field allows you to enter math functions to plot tables in a generative fashion. This is a fairly advanced feature; don't worry if you don't follow all the details. You can still explore this capability through experimentation.

However if you want to generate waveforms from thin air and enjoy this kind of approach, you'll be glad that you paid attention in math class!

The good news is that Serum includes many formula presets, and it's easy to paste formulas created by other users.

Basic Functions

The following table lists the basic functions supported by the parser.

Name	Arguments	Description
sin	1	sine function
cos	1	cosine function
tan	1	tangent function
asin	1	arcus sine function
acos	1	arcus cosine function
atan	1	arcus tangent function
sinh	1	hyperbolic sine function
cosh	1	hyperbolic cosine function
tanh	1	hyperbolic tangent function
asinh	1	hyperbolic arcus sine function
acosh	1	hyperbolic arcus cosine function
atanh	1	hyperbolic arcus tangent function
log2	1	logarithm base 2
log10	1	logarithm base 10
log	1	logarithm base 10
ln	1	logarithm base e (2.71828...)
exp	1	e raised to the power of x
sqrt	1	square root of a value

Name	Arguments	Description
sign	1	sign function -1 if x<0; 1 if x>0
rint	1	round to nearest integer
abs	1	absolute value
min	var.	minimum of all arguments
max	var.	maximum of all arguments
sum	var.	sum of all arguments
avg	var.	mean values of all arguments

Built-in Binary Operators

The following table lists the default binary operators supported by the parser.

Operator	Meaning	Priority
&&	logical and	1
||	logical or	2
<=	less than or equal	4
>=	greater than or equal	4
!=	not equal	4
==	equal	4
>	greater than	4
<	less than	4
+	addition	5
-	subtraction	5
*	multiplication	6
/	division	6
^	raise x to the power of y	7

Constants and Variables

The following table lists the constants and variables supported by the parser.

Note the following:

- sel refers to the current (existing) waveform value of the selected table

  This does not change for each table when using a multi-table formula. In other words, the formula contains y or z, and refers only to the selected table when formula processing begins.

- rand is a random number from -1.0 to 1.0

  This value stays the same for all tables.

Symbol	Description
pi	3.14159265897932384626433832
e	2.718281828182818281828
w	The current time-value getting plotted, from 0.0 to 1.0. This is the same as (x+1)/2.
x	The current time-value being plotted, from -1.0 to 1.0.
y	The current table number, from 0.0 to 1.0.
z	The current table number, from -1.0 to 1.0. This is equivalent to (y*2)-1.
q	When a q is present in the formula, the function plots to the FFT bins instead of plotting to the waveform display. Note that q iterates from 1 to 512 for the respective harmonics/bins.
in	The current (old) visible waveform value of the plotting table. This changes to each old table, if using y or z, which plots all tables.
sel	Similar to in, but only the currently-selected wavetable (does not change to each table; uses the selected table when formula processing begins).
rand	A random number from -1.0 to 1.0 that stays the same for all tables (precalculated for every time position).

Note the following:

- y and z refer to the current table number. Therefore, when y or z is in the expression, all tables for the current oscillator are regenerated.

- q renders the function to FFT instead of to the waveform display. There is no reason to include x or y in a formula containing q.

Exploring the Formula Presets Menu

Serum offers a formula presets menu that you can use to get started with preset examples. This provides a good way to learn the capabilities, as well as to create your own presets.

Formula Presets Menu

As soon as you select a formula, it appears in the **Formula** field automatically and the waveform is calculated.

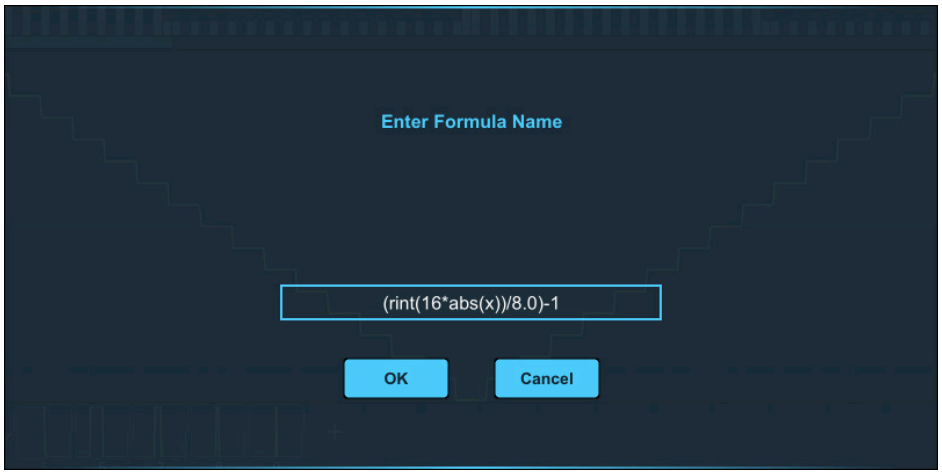
Menu Item	Description
Singles	These are single-cycle formulas that do not contain y or z variables. When you select an option, only the currently-selected table is replaced/generated with the formula.
Multis	These are multi-cycle formulas that contain y or z variables. When you select an option, the entire wavetable is replaced.
User Singles	These are user-defined single-cycle formulas.
User Multis	These are user-defined multi-cycle formulas.

Saving Your Own Formulas

You can save your own formulas for quick access.

1. Type the formula in the **Formula** field and press the **Enter** key.

2. Click the formula presets menu and choose **Save Formula** in the menu that appears. A dialog appears allowing you to save the formula.



Save Formula Dialog

By default, the text box displays the formula.

3. Type the name of the formula and click the OK button.

The formula is saved in the corresponding formula sub-menu (either **User Singles** or **User Multis**, as appropriate). You can now access your formula using the formula presets menu.

Managing Formula Files Manually (Advanced)

Serum uses two files to store the user formula presets:

- **`FormulaUserMultis.txt`**
- **`FormulaUserSingles.txt`**

You can find these files in the **`Serum 2 Presets/System/`** folder. The text files have the following format:

```
[formula1][formula1-name]

[formula2][formula2-name]
```

You can find existing examples in the **`FormulaFactorySingles.txt`** or **`FormulaFactoryMultis.txt`** files.

If you need to remove a formula preset that you saved earlier, do the following:

1. Open the appropriate user formula preset file using a text editor.
2. Delete the line containing the formula you want to remove.
3. Save the text file.

Important: While you can also edit the **`FormulaFactorySingles.txt`** and **`FormulaFactoryMultis.txt`** files, it's better only to use the user files listed in this section instead. This prevents your personal formulas from being overwritten when you install a Serum update.

Exploring Formula Parser Examples

This section shows a set of examples of using the q variable in formulas. When you include the q variable in a formula, Serum processes the formula using the FFT area instead of directly "as audio" in the wavetable display (time domain).

This means you can do signal generation or processing of a wavetable in the frequency domain. In this case, the q value represents the bin number in the FFT area at the top of the Wavetable Editor, from 1 to 512.

For example, consider the following formula:

```
q<17
```

The formula creates a wavetable frame with only the first 16 harmonics.

Formula as q<17

As another example, consider the following formula:

```
q<z*256
```

This formula creates an entire table set of 256 frames, with one additional harmonic in each consecutive frame.

Next consider the following formula:

```
(1/q)^0.25
```

This creates a sawtooth wave with higher harmonics than a normal saw (-3 dB/oct). Replace the 0.25 value with a different number to alter the spectral decay (for instance, 0.5 produces a normal saw, 0.75 creates a duller saw).

Alternatively, you could replace 0.25 with z, as shown in the following:

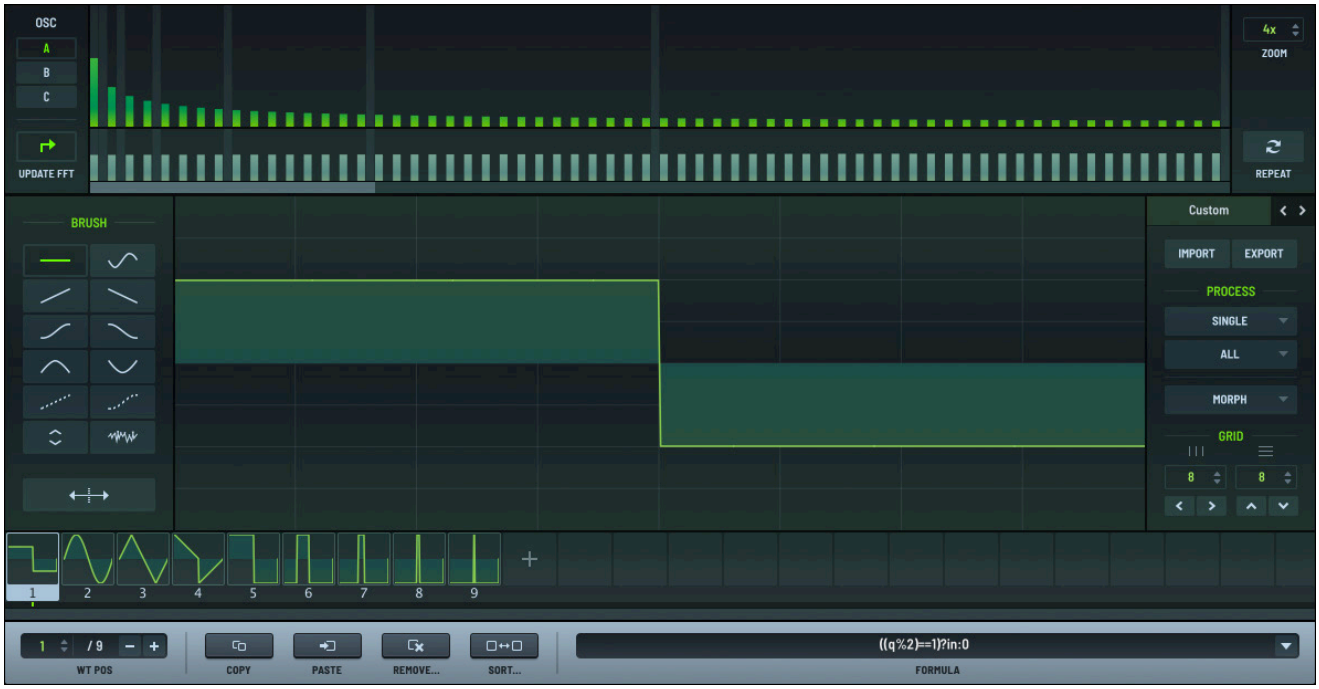
```
(1/q)^z
```

This causes Serum to create 256 frames in the wavetable with the harmonic scale.

Finally, consider the following formula:

```
((q%2)==1)?in:0
```

This formula removes every second harmonic (even harmonics) from the current wavetable frame, leaving odd harmonics only (like a square wave).



Formula as ((q%2)==1)?in:0

Using this on a (default) sawtooth wave results in a square wave. This is a popular formula, particularly in Dubstep bass sound design, to make tables sound more “hollow” (taking up less of the spectrum).

Note: This formula appears as an included preset called **Proc:Squarify**.

Exploring Global Settings

You can use the **GLOBAL** module to configure global settings in Serum.

There are two types of global settings: those saved with your preset, song, or patch, and those saved as part of Serum, accessible to all presets.

Serum Global Settings

Using the Global Module

Click the **GLOBAL** tab to access the module.

Accessing Serum Global Settings

The global settings page appears.

The page is divided into the following panes to help you quickly find the appropriate settings:

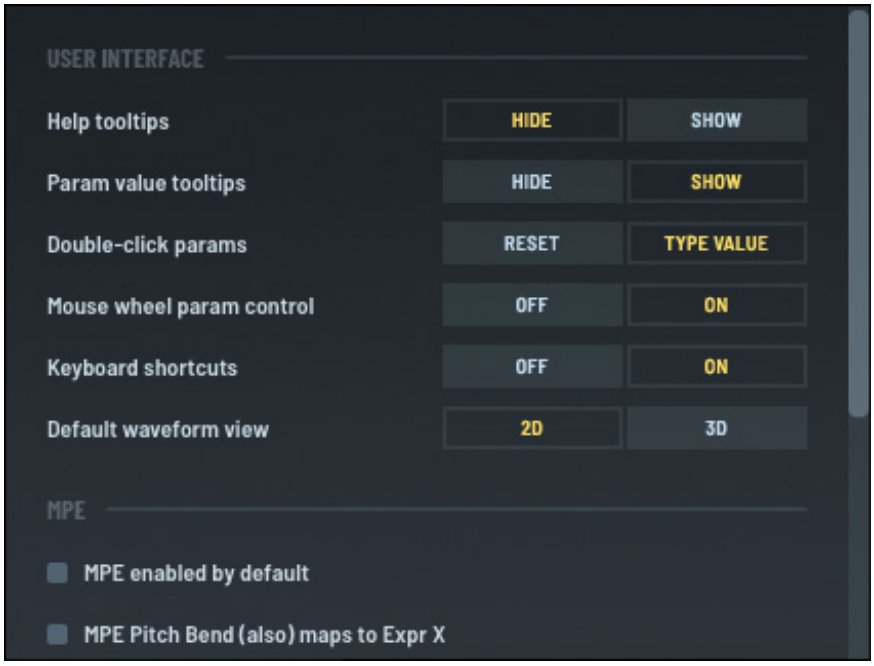
- Preferences — Specify global preferences including user interface and MPE settings
- Voice Control — Define the behavior of each voice across the available oscillators
- Quality — Set the render quality and Serum 1 compatibility
- Tuning — Set the concert pitch and manage Serum tuning

The settings page also displays the current Serum version and build date.

Preferences

Use the **PREFERENCES** pane to configure your global preferences in Serum.

This includes default waveform display, user interface options, and double-click behavior, among other settings.



Serum Global Preferences

The following table describes the preferences you can set.

Category	Preference	Description
User Interface	Help Tooltips	Display tooltips after hovering the mouse pointer over a control for a moment.
	Param value tooltips	Display numeric values (as a small pop-up) when modifying a control.
	Double-click params	Specify whether double-clicking a control resets the control to the default (init) value or whether double-clicking displays a pop-up text box allowing you to type a specific value.

Category	Preference	Description
User Interface (cont.)	Mouse wheel param control	By default, moving the mouse wheel adjusts the knob that the cursor is currently hovering over. If you don't want this capability, enable this setting to deactivate the feature.
	Keyboard shortcuts	Set whether Serum should respond to input from the computer keyboard. Disable this to prevent Serum from stealing keyboard focus from your DAW.
	Default waveform view	The default waveform view in the OSC panels, either 2D (default) or 3D.
MPE	MPE enabled by default	Specify whether MPE mode is enabled when a new instance of Serum is added to your project.
	MPE Pitch Bend (also) maps to Expr X	By default, Serum maps MIDI pitch bend messages to per-note pitch bend and CC10 (pan) messages to note expression X. Some MPE controllers use the X axis (left/right movements) to control pitch and will therefore transmit pitch bend messages. Select this option to have these gestures mapped to note expression X in Serum, as well as to per-note pitch bend.
	MPE Expr Y acts bi-directional	Specify whether note expression Y is treated as unipolar or bipolar. This allows you to match how your DAW or MPE controller treats note expression Y.
General	Limit Mod depth on drop (based on knob value)	When a modulator source is dropped on a knob, reduce the range amount (if needed) so that the modulation does not exceed the maximum value for the range. For instance, if you drag a modulator to an oscillator **LEVEL** knob set at 75%, the assigned modulation depth will only be 25% (so that a maximum modulator will have volume reach exactly 100%).

Category	Preference	Description
General (cont.)	Mod Wheel -> WT Pos (when WT Editor is open)	When the Wavetable Editor is active, the mod wheel scans the indices from 1-256.
	Silence note + FX tails when host transport stops	Mute effects and any sustaining notes when the host DAW is stopped.
	Load MIDI Map from Presets	Normally, Serum loads a default MIDI CC map if the **Serum 2 Presets/System/ MIDI CC Maps/Default.SerumMIDIMap** is found, in the following cases: • When creating a new instance of Serum • When selecting the **Init** preset • When loading a preset When the **Load MIDI Map from Presets** option is enabled, loading a preset causes Serum to load the MIDI CC mapping that was saved with the preset instead of the default. You might want to enable this option if you have a specific mapping for FX parameters that you want to recall, for instance.
	Use Ultra quality when rendering	Instruct Serum to perform an offline render (bounce) using ultra quality mode. This results in the highest quality playback of the rendered sound. Since the rendering is performed offline, the performance trade off in using ultra quality mode for renders generally makes sense.
	Automatically check for updates	Enable to ensure that you are notified when new versions of Serum become available.

Setting the Voice Control

You can specify voice control settings for Serum to define the behavior of each voice across the available oscillators.

Select the oscillators to which the voice control settings apply. By default, all oscillators are selected (green).

Select the Oscillators

Voice Control Settings

Loading a Preset

You can optionally load a factory supplied or user-defined preset.

Click the voice control (topmost) field and choose a preset using the menu that appears.

The configuration loads and populates the relevant settings.

Voice Control Menu

Creating a New Configuration

You can create a new configuration, or reinitialize the voice control, at any time.

Click the voice control field and choose **Init** in the menu.

This initializes the voice control to the default settings and provides an opportunity for you to create a new configuration.

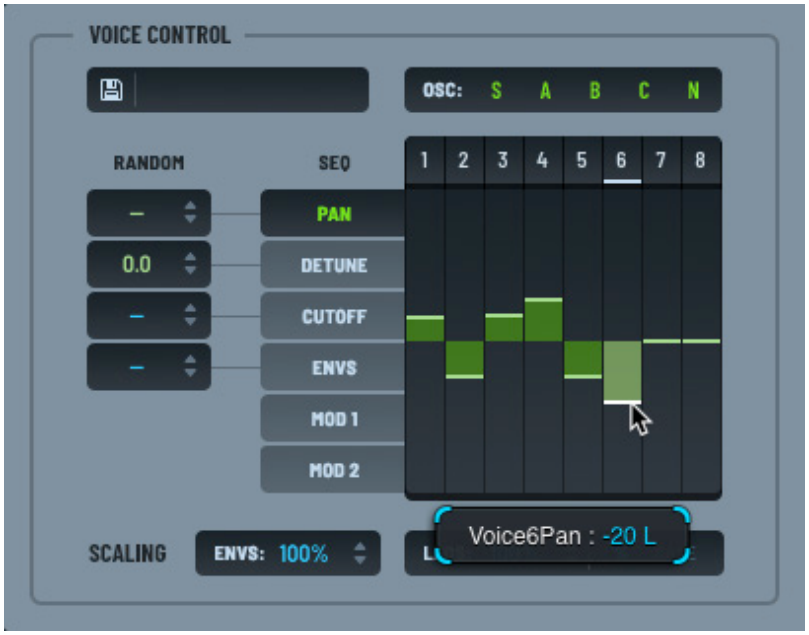
Initializing the Configuration

Configuring the Voice Control

Click one of the numbers, from 1 to 8, to set the sequence length.

Select one of the **SEQ** controls and adjust the per voice settings.

Continue setting the other controls, as appropriate.

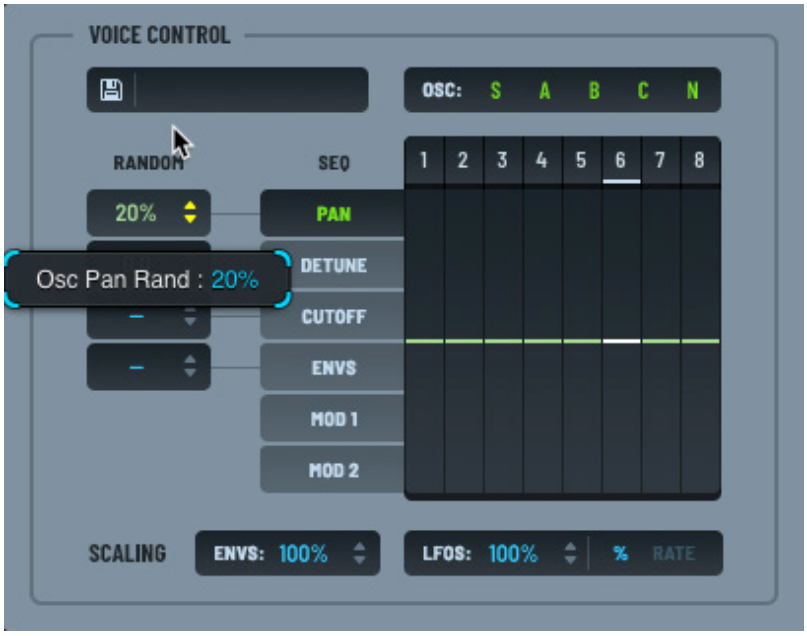


Per Voice Settings

Setting Randomization

You can set the randomization for the **PAN**, **DETUNE**, **CUTOFF**, and **ENVS** (envelopes).

Click the corresponding **RANDOM** field and drag to set the value. You can also double-click the field and type a value.



Setting Randomization

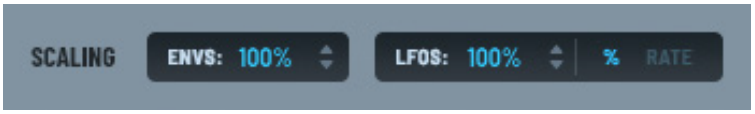
The following table outlines the randomization effect:

Field	Description
PAN	Randomizes the stereo position, per voice (as a percentage).
DETUNE	Randomizes the tuning offset, per voice (in cents).
CUTOFF	Randomizes the filter cutoff, per voice (as a percentage).
ENVS	Randomizes the envelope offset, per voice (as a percentage).

Setting the Scaling

You can set the scaling for all envelopes and LFOs.

To set the scaling for all envelopes, click the field and drag up or down.



Scaling Settings

You can use the arrow keys to fine tune the setting. You can also double-click the field and type a value.

This is useful if you change the BPM (beats per minute).

To set the LFO scaling, click the field and drag up or down. You can use the arrow keys to fine tune the setting. You can also double-click the field and type a value. Choose to set by percentage or rate.

This is helpful for creating many simultaneous pattern changes.

Saving the Voice Control Settings

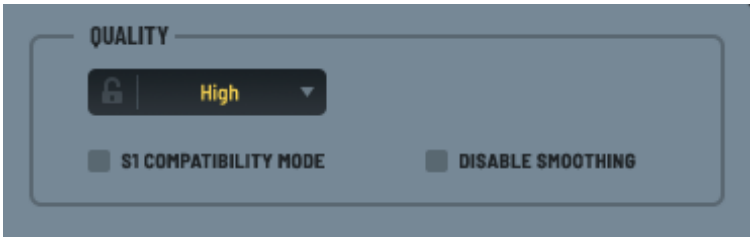
Click the  button to save the voice control settings. A dialog appears allowing you to specify a name.

Setting the Quality

You can specify the quality (oversampling) in Serum.

You can further disable smoothing, as needed, and enable Serum 1 preset compatibility.

Choose the oversampling quality settings using the drop-down menu.

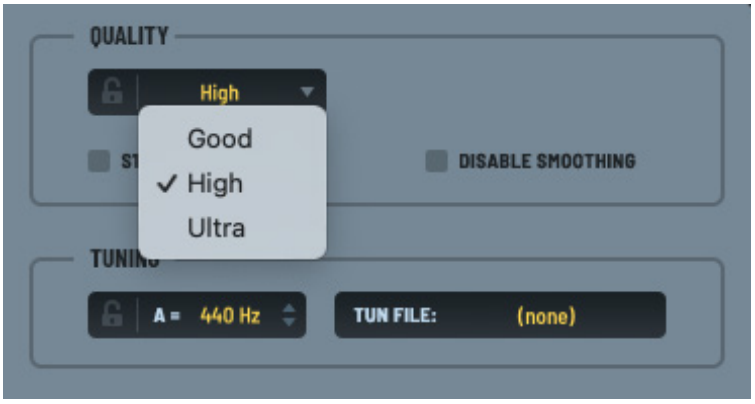


Quality Settings

Draft quality sets 1x oversampling (no oversampling). High quality sets 2x oversampling, while Ultra results in 4x oversampling.

Click the  button to lock the quality settings, even when you load a new preset.

This means that when locked, Serum ignores the quality configuration in the preset and uses the settings that you have locked.



Quality Settings

You can also set the following quality settings:

Setting	Description
S1 Compatibility Mode	Serum 2 features a completely rebuilt sound engine. However, when you load a Serum 1 preset, this option is automatically enabled to preserve maximum sonic similarity with Serum 1. Disable this option if you prefer that Serum 1 presets instead take advantage of the DSP updates available in the Serum 2 sound engine.

Setting	Description
Disable Smoothing	Disable automation parameter smoothing. While Serum is built to try to avoid clicks and jumps in the signal, there are times when you might want precision over parameter changes (for instance, during fast rhythmic automation). In this case, you can choose to disable smoothing (Serum supports sample-accurate automation). **Note:** Smoothing also applies to parameter changes effected through mouse actions on on-screen controls.

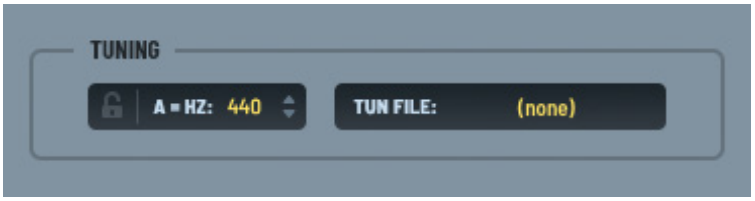
Setting the Tuning

You can specify the tuning for Serum, including setting the concert pitch and managing tuning using a tuning file.

Setting Concert Pitch

Concert pitch is the standard reference pitch used to tune musical instruments.

The most widely accepted concert pitch is A4 at 440 Hz, where the A4 refers to the A above middle C (the fourth A key on a piano).



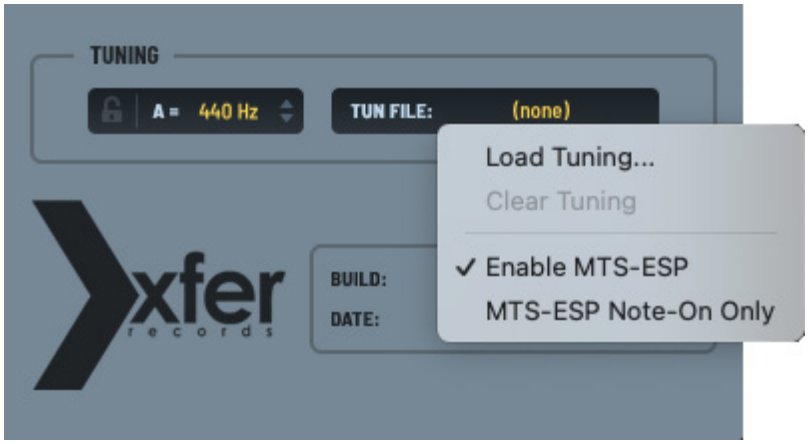
Tuning Settings

To set the concert pitch for Serum, click the left field and drag up or down. You can use the arrow keys to fine tune the setting. You can also double-click the field and type a value.

Using a Tuning File

You can also set the tuning using a tuning file. Serum offers two options.

You can load a tuning file for the current instance of Serum, or you can have the Serum instance follow the tuning specified elsewhere in your project using MTS-ESP.



Tuning Menu

To set the tuning file for the Serum instance, click the **TUN FILE** field and choose **Load Tuning** in the menu that appears. A dialog appears allowing you to locate the appropriate tuning (.tun) file. To clear the tuning file, use the same menu and choose Clear Tuning in the menu.

Using MTS-ESP to Set the Tuning

You can load a separate tuning file for each instance of Serum (using the procedure described in the previous section). However, if you would like to use a single main tuning file across all instances, Serum supports the MTS-ESP microtuning system developed by ODDSOUND (www.oddsound.com).

Using the free MTS-ESP MINI plugin, you can load **`.scl`**, **`.kbm`** or **`.tun`** files and automatically retune all connected MTS-ESP clients (including Serum). This allows you to retune any number of supported virtual instruments from a central location without requiring you to tune each instrument separately.

MTS-ESP support is enabled in Serum by default. You can disable this feature by unchecking the **Enable MTS-ESP** menu option.

Enable the **MTS-ESP Note-On Only** option to have the tuning set on **Note-On** MIDI events. This ensures that the tuning of a note will not change during its duration, even if the global MTS-ESP tuning updates.

Note the following about using MTS-ESP:

- Loading a .tun file always takes precedence over MTS-ESP. This allows you to tune any Serum instance differently from the global MTS-ESP tuning.
- If the oscillator pitch mode is set to **Steps** (set by right-clicking an oscillator **OCT** or **SEM** control), you can pitch oscillators up or down in “periods” and “steps” as defined by the active MTS-ESP tuning, rather than in octaves or semitones.

In addition to the free MTS-ESP MINI plugin, you can choose from among the following additional MTS-ESP plugin options:

- Wilsonic MTS-ESP
- Surge XT
- Entonal Studio
- Infinitone

Locking the Tuning Configuration

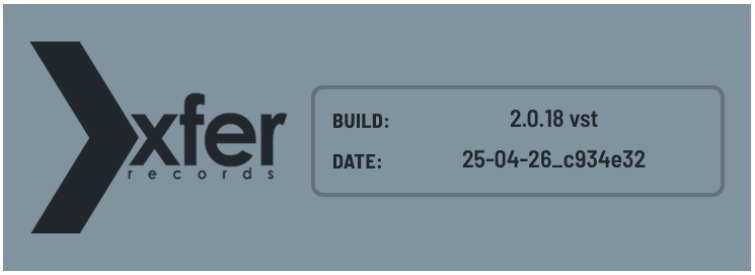
Click the  button to lock the tuning configuration, even when you load a new preset.

This means that when locked, Serum ignores the TUN file and the concert pitch setting in any new preset that you load.

Checking the Build Version and Date

You can quickly check the version of Serum that you're running using the Xfer pane.

Be sure to refer to this version number if you reach out to Xfer Records for technical support.

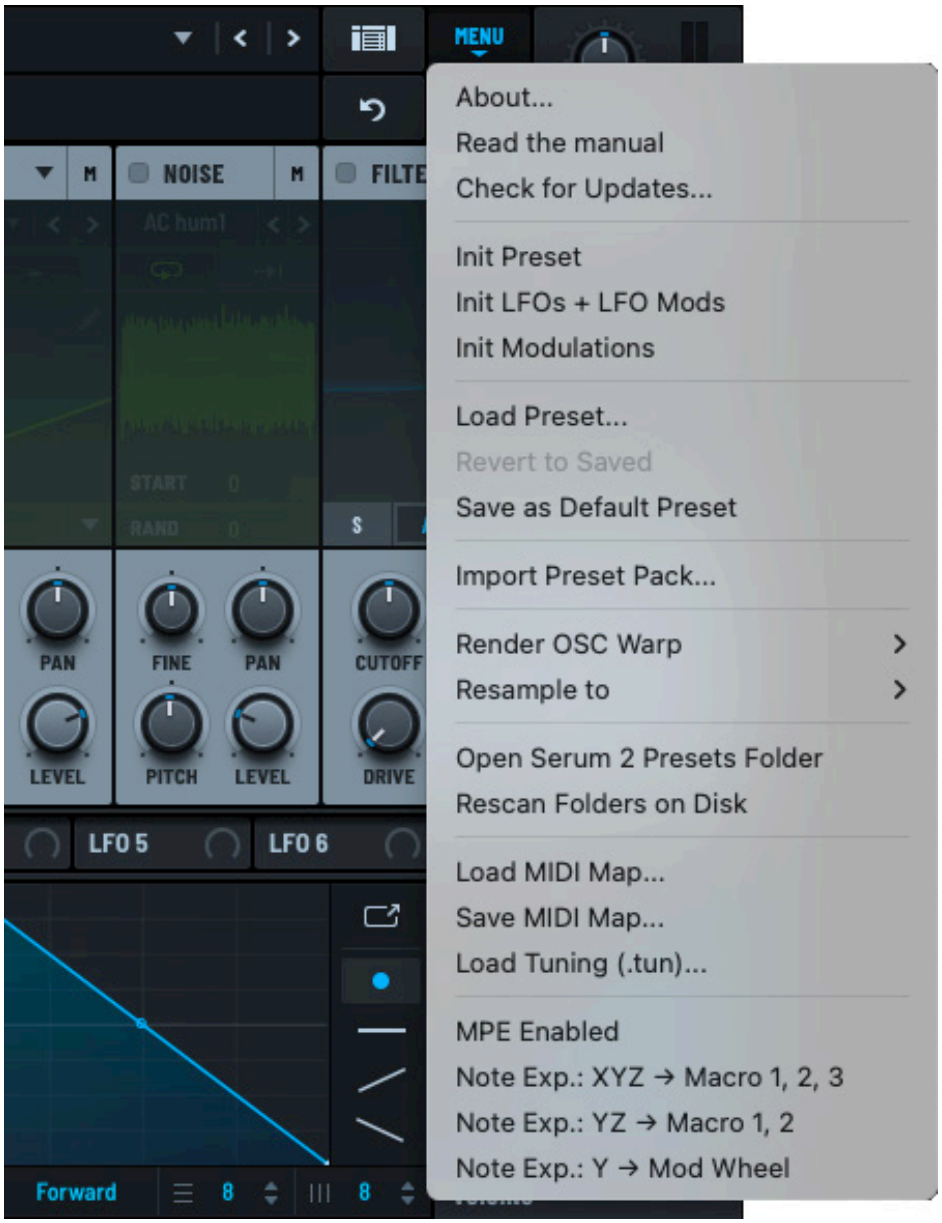


Serum Build and Date

Appendix A: Using the Main Menu

You can use the main menu to complete operations that affect the overall performance and capabilities of Serum. These operations include managing presets, initializing modules, rendering waveforms, configuring MPE settings, and more.

You can access the main menu MENU near the top right of the Serum window.



Serum Main Menu

The following table describes each menu item:

Category	Options	Description
General	About	Display Serum release information.
	Read the manual	Display this manual, the Serum User Guide (PDF).
	Check for updates	Display whether an update is available, with a link to the website.
Initialization	Init Preset	Initialize Serum to allow you to create a new patch.
	Init LFOs + LFO Mods	Initialize just the LFOs and LFO modulations
	Init Modulations	Initialize just the modulation assignments, leaving everything else untouched (including the LFOs).
Presets	Load Preset	Load a preset from storage. A system dialog appears allowing you to locate the preset file.
	Revert to Saved	After loading a preset and making changes, revert to the saved preset.
	Save as Default Preset	Save the current preset to the following file: **`Serum 2 Presets/Presets/User/default.SerumPreset`** Note that if you are running the Serum FX version, this saves to the **`defaultFX.SerumPreset`** file. This allows you to configure different default configurations for Serum and Serum FX. When you load Serum FX, it will look for the **`defaultFX.SerumPreset`** file. If the file isn't found, Serum FX instead looks for the **`default.SerumPreset`** file.
Import	Import Preset Pack	Import a Serum preset pack. A system dialog appears allowing you to locate the preset pack.
Rendering	Render OSC Warp	Use the current wavetable frame of the selected oscillator and create 256 frames (subtables) spanning 0 to 100% of the **WARP** knob.
	Resample to	Play a note of the preset for one bar and capture (render and import) the result as a wavetable in the selected oscillator (or **OSC A** and **OSC B**).
Folders	Open Serum 2 Presets Folder	Display the Serum 2 Presets folder using the Finder (macOS) or Explorer (Windows).
	Rescan Folders on Disk	Rescan the Serum 2 Presets folder. Do this when you make changes to the folders outside of Serum (using the Finder or Explorer).

Category	Options	Description
MIDI/Tuning	Load MIDI Map	Load a saved MIDI CC map from storage.
	Save MIDI Map	Save a MIDI CC map to storage. By default, MIDI maps are stored in the **Serum 2 Presets > System> MIDI CC Maps** folder. Saving a MIDI map as **default.SerumMIDIMap** in the default folder instructs Serum to load that MIDI map every time you add an instance of Serum or you load a preset.
	Load Tuning (.tun)	Load a tuning file for the current instance of Serum. A dialog appears allowing you to locate the appropriate tuning (.tun) file. See "Using a Tuning File" for more information.
MPE	MPE Enabled	Enable support for MIDI Polyphonic Expression (MPE). When enabled, Serum responds to MPE messages, allowing for more expressive and nuanced musical performances using compatible MPE controllers. **Important:** In the VST3 version of Serum (not the AU or AAX versions), when MPE is disabled, Serum will respond to VST3 Note Expression in cases when the host DAW supports this feature (this includes hosts such as Bitwig, Cubase, and Nuendo). This means that main menu options to map note expressions to macros remain available and applicable even when MPE is disabled. However, the **MPE Bend Range** option is not available since VST3 pitch note expression has a fixed range of +/-120 semitones.
	MPE: XYZ -> Macro 1,2,3	Map the X, Y, and Z axes of an MPE-compatible controller to Serum macros 1, 2, and 3 respectively. In this context, the x, y, and z axes represent the three dimensions of touch-sensitive control associated with MPE. Specifically: • X-axis: Horizontal movement, often used to control pitch bending. • Y-axis: Vertical movement, often assigned to parameters like filter cutoff or modulation depth. • Z-axis: Pressure or aftertouch, controlling intensity-related effects like volume or timbre.

Category	Options	Description
MPE (cont.)	MPE: YZ -> Macro 1,2	Map the X and Y axes of an MPE-compatible controller to Serum macros 1 and 2 respectively.
	MPE: Y -> Mod Wheel	Map the Y-axis movement of an MPE controller to the Modulation Wheel (Mod Wheel) control.
	MPE Bend Range: 48	The pitch bend range for MPE controllers, specified as the number of semitones above or below the original pitch. Choose this menu option to display a dialog allowing you change the current value. You can set any value from 1 to 96 semitones. A wider range allows for more expressive pitch variations and glides across the tonal spectrum.

Appendix B: Using the Presets Browser

Serum features an advanced presets browser that you can use to quickly access both factory-supplied presets as well as any presets that you have added.

Click the  button to access the presets browser. The browser appears.

Serum Presets Browser

This appendix explains how to use the presets browser to quickly locate the sounds you need and organize your favorites in a way that best suits your workflow.

Navigating the Folders

The presets browser includes a **Folders** pane that displays the presets in a hierarchical structure to help you quickly locate the right sound.

You can expand and collapse the hierarchy as needed.

Presets Folders

Select a folder in the hierarchy to have the folder contents (presets) appear in the **Presets** pane.

Presets in the Folder

Loading a Preset

Click an entry in the list to load the corresponding preset.

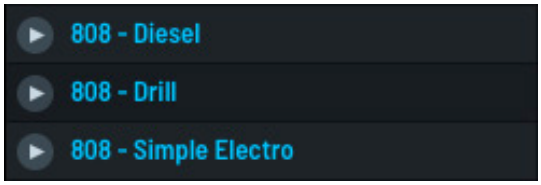
Most presets load immediately; presets with larger embedded samples, such as multisampled instruments, display a small green progress bar (directly beneath the preset name) when loading.

Previewing Presets

The browser makes it easy to preview presets. Click the corresponding  button to hear the preset play.

Serum plays an embedded preview clip (MIDI sequence) if the preset designer specified one. Otherwise, Serum plays a "fallback" clip to give you a sense of the preset.

You can choose the fallback clip from among three standard options.

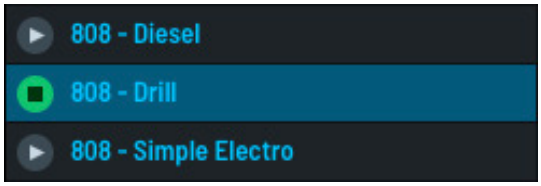


Presets Showing Play Buttons

Serum also allows you to auto-preview clips. See "Performing Standard Preset Operations" on page 334 for more information.

Click another play button to preview the corresponding preset.

Click the  button to stop the preview.



Preset Playing

After selecting a preset, press the right arrow key to start the preview and press the left arrow key to stop the preview. Use the up and down arrow keys to quickly change presets.

While previewing a preset, you can modify any macros that are defined as part of the preset and hear the effects immediately.

Preset Macros

Searching Presets

You can use the presets browser to help locate the right type of sound by name, category, tag, rating, and more.

Searching by Name

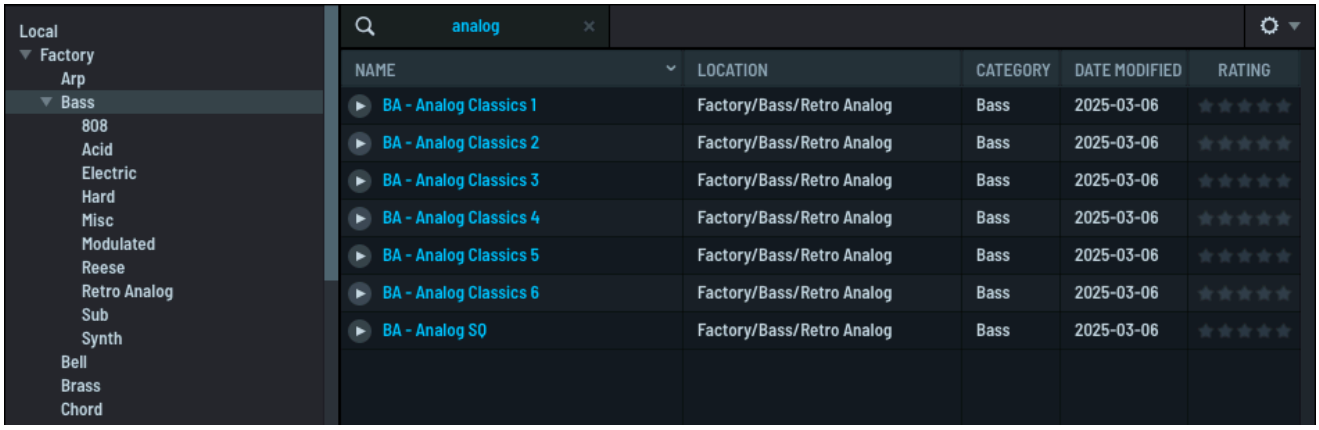
Perhaps the easiest way to find presets is to search by name. Preset names often contain important sonic attributes as part of their name. For example, if you're searching for variations of piano sounds, it's often the case that the preset name will include "piano" as part of its name.

Start by choosing a folder in the **Folders** pane. All searches in the presets browser are limited to the folder you select.

Search Field

Then type the search term and press the **Enter** key.

The search results appear. Click the **x** button to clear the search results.



Searching for Presets

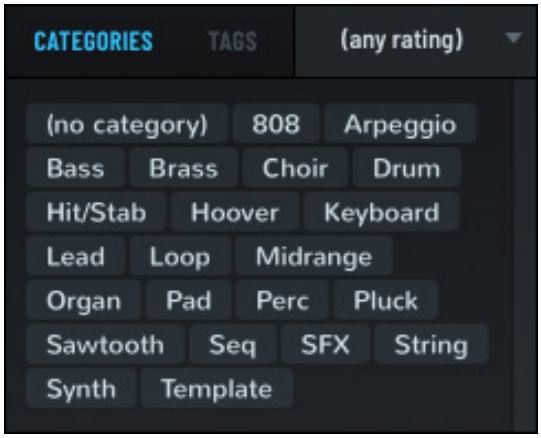
Press Cmd-F (macOS)/Ctrl-F (Windows) to position the cursor in the search field. Note that this only works when you are already in the presets browser.

Searching by Categories or Tags

You can also search for presets by categories or tags.

As with any search, start by choosing a folder in the **Folders** pane.

Ensure that the **CATEGORIES** tab is selected.



Searching Using Categories

Then select the appropriate categories in the pane. The search results appear.

To search by tags, select the **TAGS** tab.

Then select the appropriate tags in the pane. The search results appear.

Searching Using Tags

Searching by Ratings

You can search for presets based on the ratings you've assigned.

Note: Ratings are user-assigned; presets, by default, have no assigned rating. Ratings allow you to quickly score presets that you find interesting, allowing you to easily locate them later.

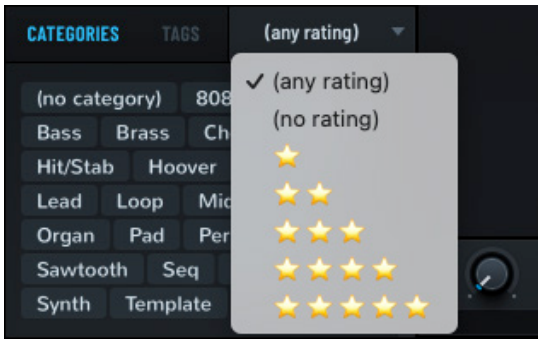
Click the **Ratings** menu and choose a score from the list that appears.

The search results appear.

Note that the ratings match exactly. For example, if you select four stars from the menu, only presets rated with four stars appear (not four stars or above).

Choose **(no rating)** using the menu to identify the presets that you haven't reviewed and rated.

To show all presets, choose **(any rating)** in the menu.



Searching Using Ratings

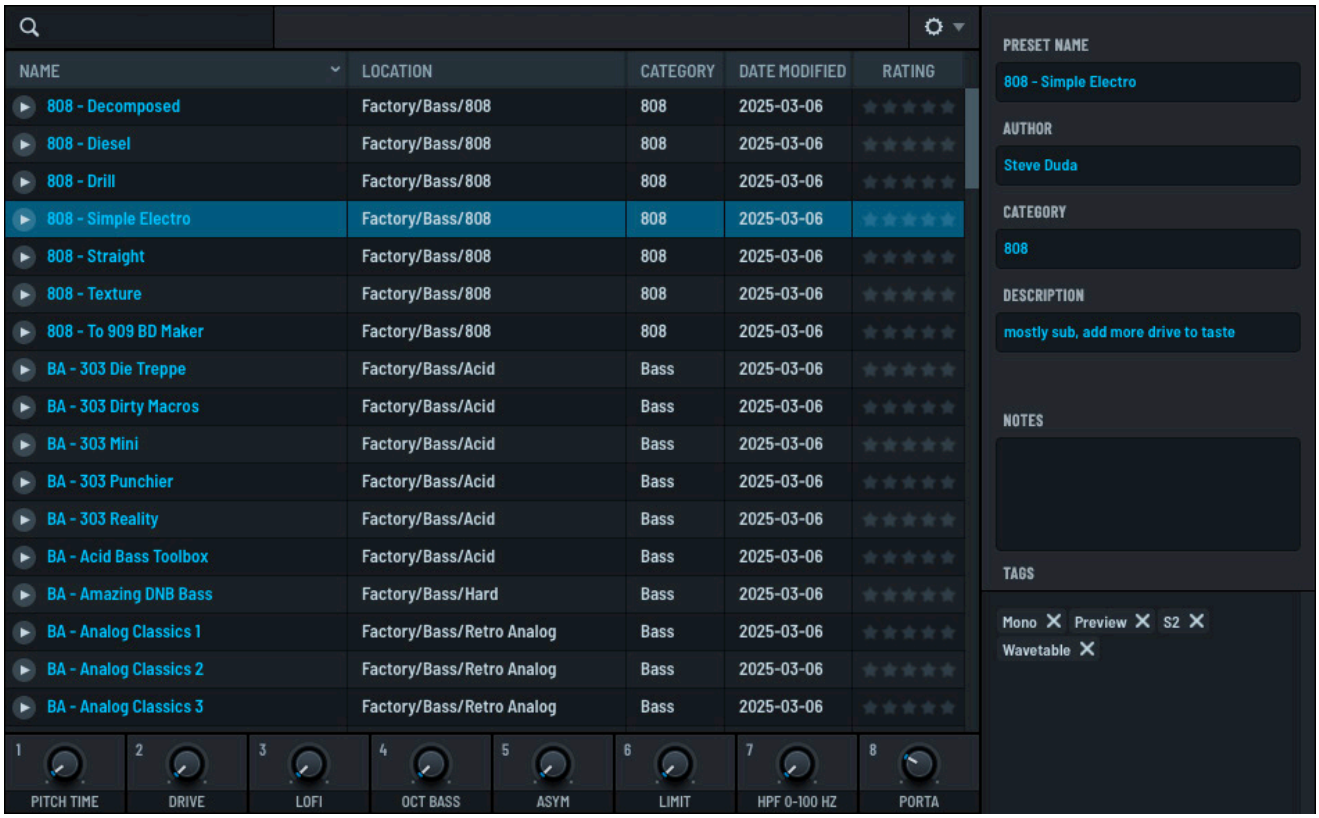
Managing Your Presets

The preset browser offers a versatile set of capabilities that allow you to display and edit preset metadata, rate presets, and perform a range of other operations including renaming, moving, and deleting your presets.

Displaying and Editing Preset Metadata

You can display the metadata associated with a preset, including the artist, category, description, comments, date on which the preset was created, and associated tags.

Click a preset in the list. The preset highlights and the metadata appears in the pane on the right.



Preset Metadata

Modify the information, as appropriate. In most fields, click in the field and edit the contents. Two fields are slightly different: **CATEGORY** and **TAGS**. You can read about those below.

Note that you cannot change the **DATE** field; the preset creation date is managed by the computer operating system and represents when the preset file was created (or last modified).

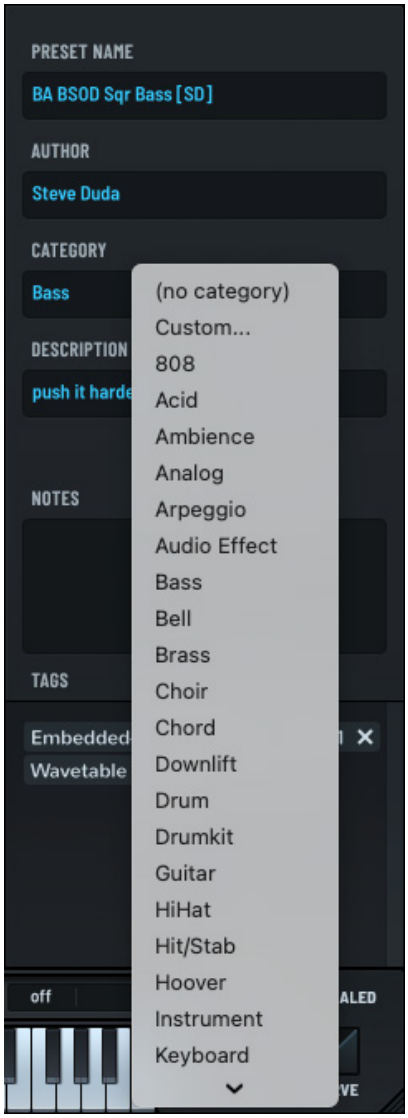
Specifying the Category

To set the preset category, click in the **CATEGORY** field and choose an option from the menu that appears.

If you don't see an appropriate category in the list, choose **Custom** to define a new category.

The **CATEGORY** field becomes editable. Type the new category name and press the **Enter** key.

The new category now appears in the list for you to use later, if needed.

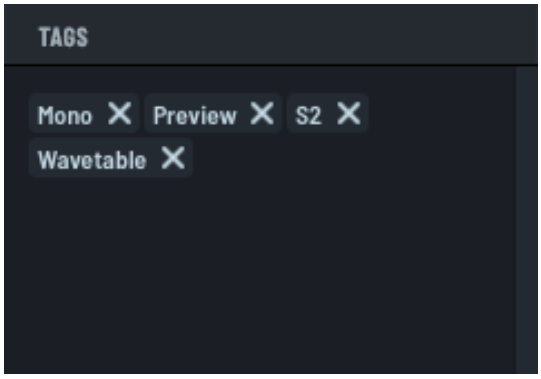


Preset Category

Managing Tags

You can manage the tags associated with a preset.

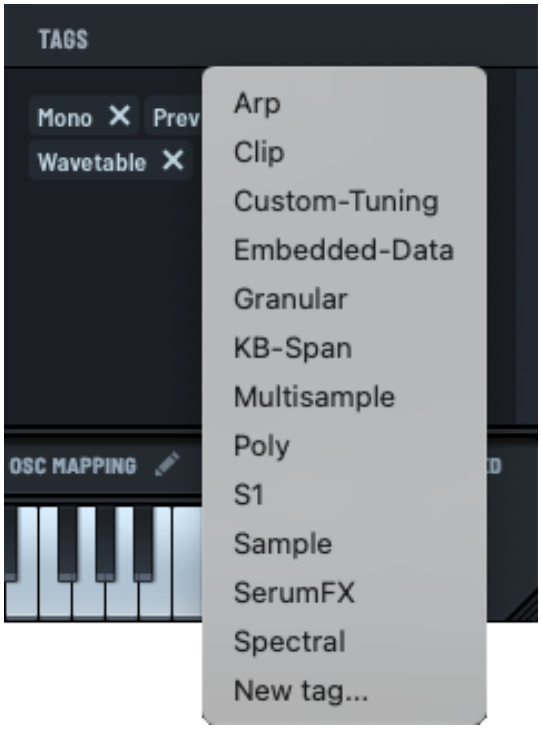
Click in the field to add a new tag. A menu appears allowing you to choose from the existing tags.



Preset Tags

Select a tag or choose **New tag** to create a new entry. A dialog appears allowing you to type the name of the tag.

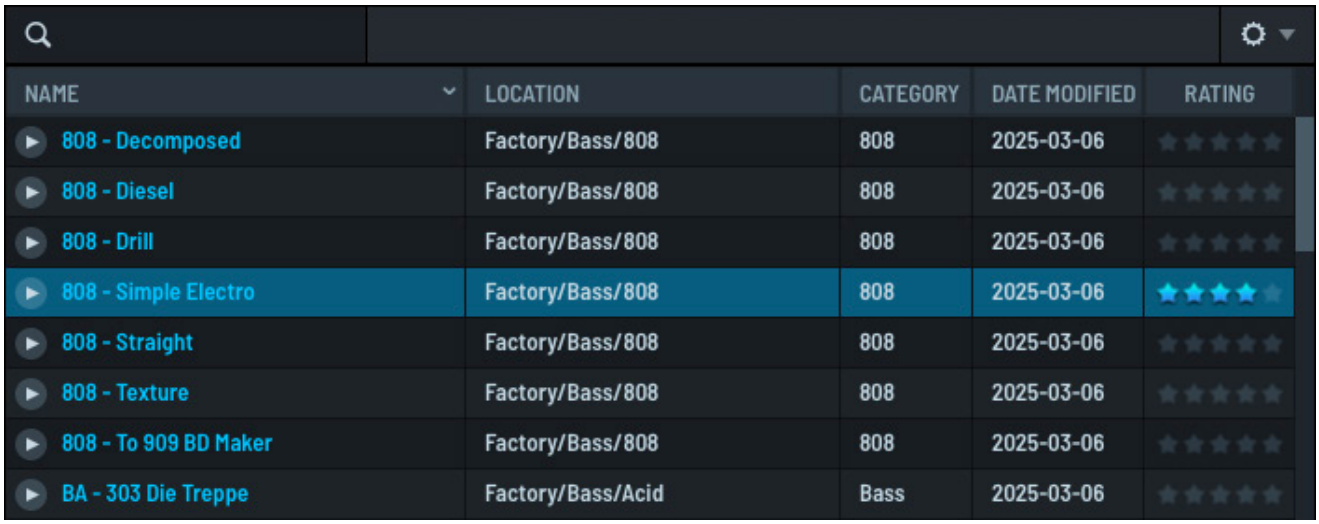
To remove a tag from the preset metadata, click the **X** button associated with the tag. The tag is removed.



Preset Tags

Rating Presets

You can rate presets using a one to five-star system and easily search for presets based on their ratings, allowing you to quickly find presets that capture your interest.



Rating a Preset

Click the corresponding star for the preset you want to rate. To change a rating, click the new star rating.

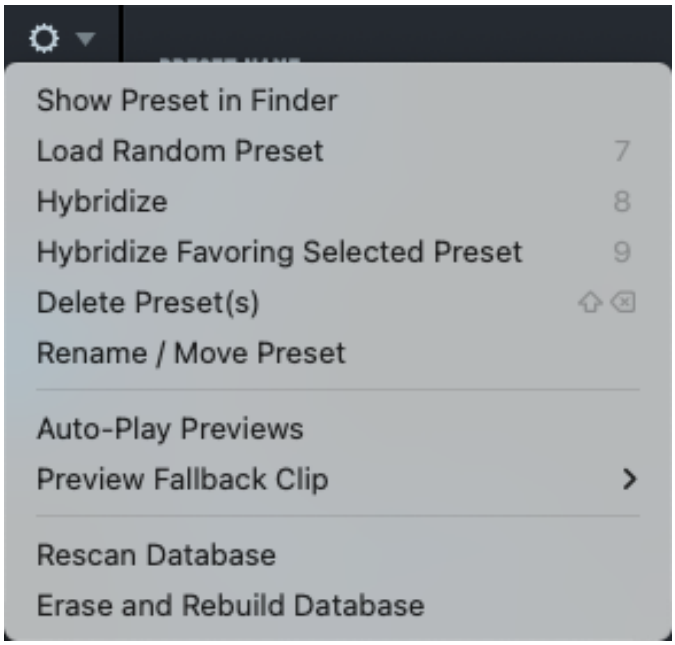
To remove a rating, click the first star to select it and then click the same star again to remove the rating.

Performing Standard Preset Operations

You can perform a range of other operations on presets, including loading random presets, renaming or moving presets, deleting presets, and more.

You can also rescan the presets database or erase and rebuild the database.

Click the  button. The presets menu appears.



Presets Menu

The following describes the operations available:

Category	Operation	Description
Presets	Show Preset in Finder (macOS) Show Preset in Folder (Windows)	Display the selected preset in the Finder (macOS) or Explorer (Windows).
	Load Random Preset	Load a random preset. To load another random preset, type 7
	Hybridize	Load a hybrid preset consisting of four randomly-selected presets. To load another hybrid preset, type 8. The generated preset name reflects the hybrid mix. 
	Hybridize Favoring Selected Preset	Load a hybrid preset consisting of the currently-selected preset and three other randomly-selected presets. To load another hybrid like this, type 9.

Category	Operation	Description
Presets (cont.)	Delete Preset(s)	Select one or more presets by clicking a preset or Shift-clicking a set of presets, and choose **Delete Preset(s)** from the menu. You can also type **Shift-Backspace** to delete the selected presets. **Important:** Serum does not ask for confirmation before deleting, and there is no way to undo the deletion from within Serum. Don't despair. See the tip below to recover any accidentally deleted presets.
	Rename/Move Preset	Rename or move a preset. Select a preset, and choose **Rename/Move Preset** from the menu. A macOS or Windows system dialog appears. Use the dialog to rename or move the preset, as appropriate. After returning to Serum, click the  button in the presets browser, and choose **Rescan Database** from the menu.
Previews	Auto-Play Previews	Toggle on to have preset previews play automatically. After selecting the first preset using the mouse, you can quickly preview other presets using the up and down arrow keys.
	Preview Fallback Clip	Use this option to choose the Serum "fallback" clip to play when previewing presets. When previewing presets, Serum plays the author-designated MIDI preview clip by default (if the preset author chose to specify a preview clip). Otherwise, Serum plays the selected fallback clip to give you a sense of the preset.
Database	Rescan Database	Rescan the Serum presets database. You can do this after adding new presets to the Serum 2 Presets folder. You can also do this after making other changes to the database, as described in this section.
	Erase and Rebuild Database	Do this if rescanning the database (above) unexpectedly fails to reflect all the presets in the Serum 2 Presets folder. Under normal circumstances, you will never likely need to do this.

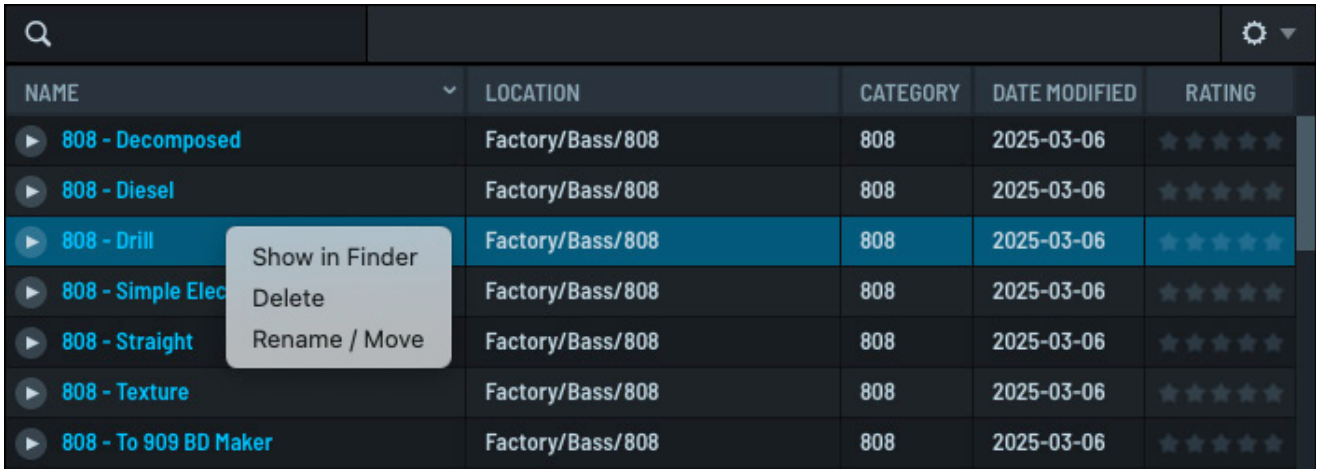
If you accidentally delete one or more presets, Serum moves the preset files to the Trash (macOS) or Recycle Bin (Windows).

Using macOS, open the Trash, find the preset file, right-click the file, and choose **Put Back** from the menu.

Using Windows, open the Recycle Bin, find the preset file, right-click the file, and choose **Restore** from the menu.

Return to Serum, click the button in the presets browser, and choose **Rescan Database** from the menu. The presets reappear.

You can also perform the most common preset operations by right-clicking a preset and choosing an option from the menu that appears.



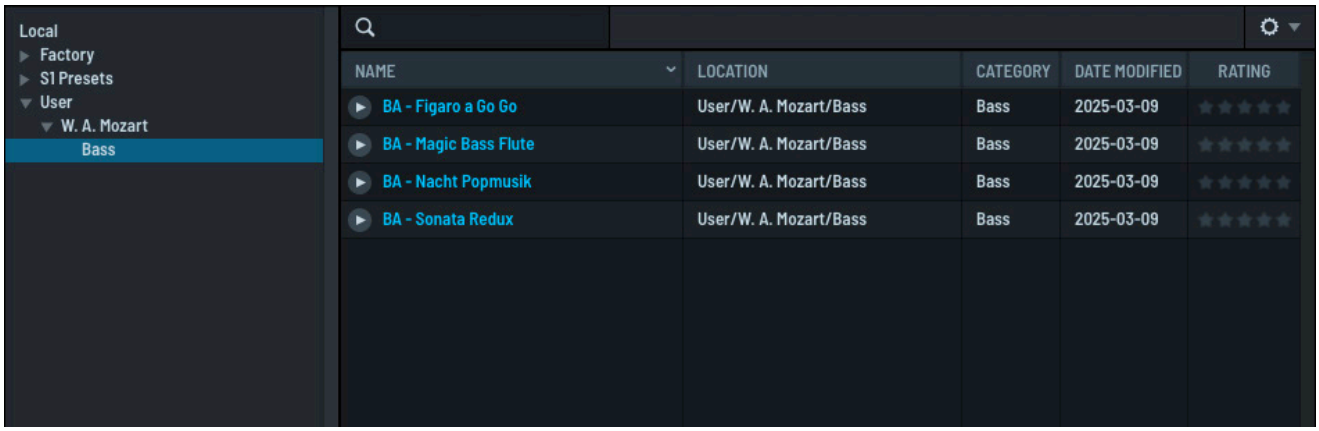
Preset Menu

The following table describes the operations you can perform:

Operation	Description
Show in Finder/Folder	Display the selected preset in the Finder (macOS) or Explorer (Windows).
Delete	Delete the selected preset. **Important:** Serum does not ask for confirmation before deleting, and there is no way to undo the deletion from within Serum. Don't despair. See the earlier tip to recover any accidentally deleted presets.
Rename/Move	Rename or move the selected preset. A macOS or Windows system dialog appears. Use the dialog to rename or move the preset, as appropriate. After returning to Serum, click the button in the presets browser, and choose **Rescan Database** from the menu.

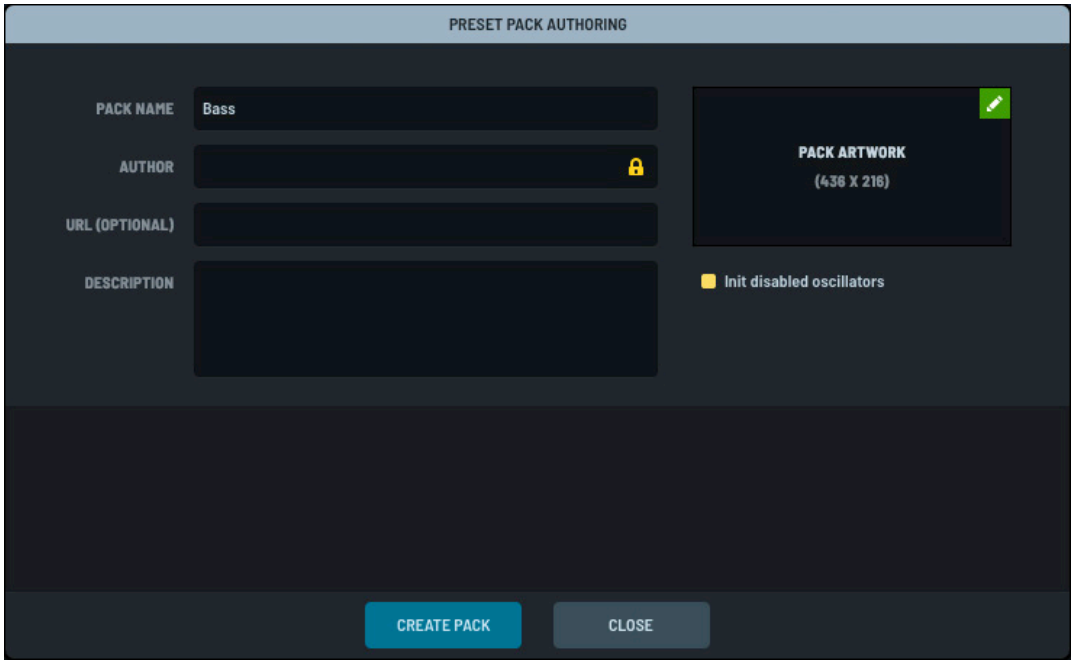
Creating and Exporting a Pack

You can create and export a preset pack from a folder of presets. Right-click a folder and choose **Create and Export Pack** in the menu that appears.



Creating and Exporting a Pack

A dialog appears allowing you to specify the pack information.

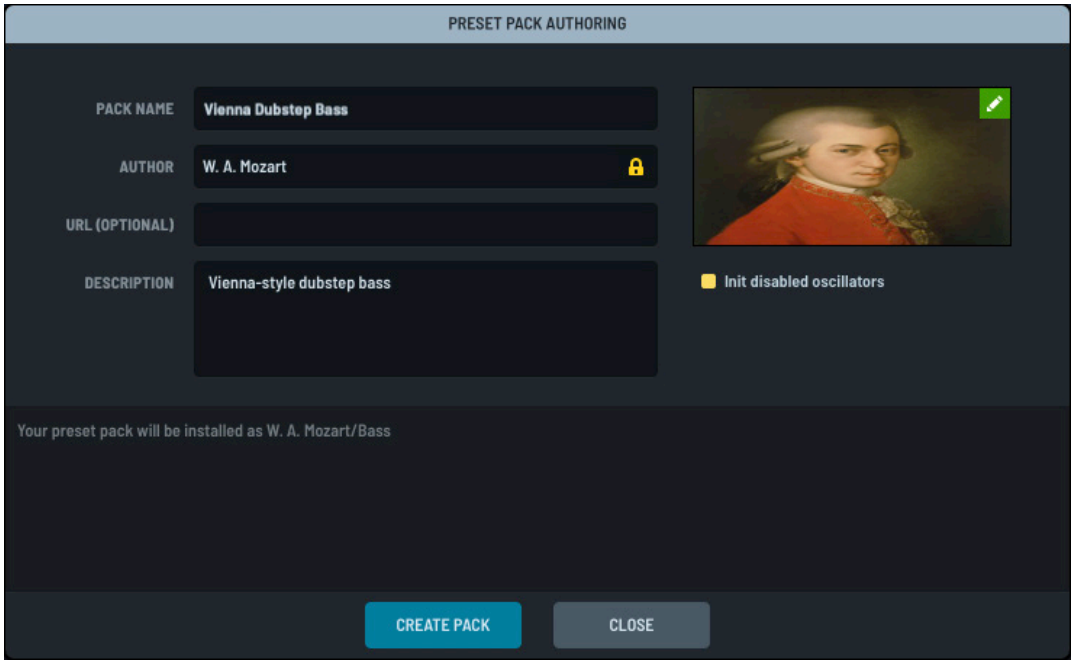


Preset Pack Dialog

The following describes the information you can specify:

Field	Description
PACK NAME	The name of the pack. This is used as the default file name for the preset pack.
AUTHOR	The author name. Make sure that the button is enabled to prevent users from modifying the **AUTHOR** field.
URL	The URL of your website.
DESCRIPTION	A description of the preset pack.
PACK ARTWORK	The artwork for the preset pack. The ideal image size is 436 x 216 pixels. Click the button. A dialog appears allowing you to locate the image file.
Init disabled oscillators	Reset disabled oscillators, removing associated parameteres and samples.

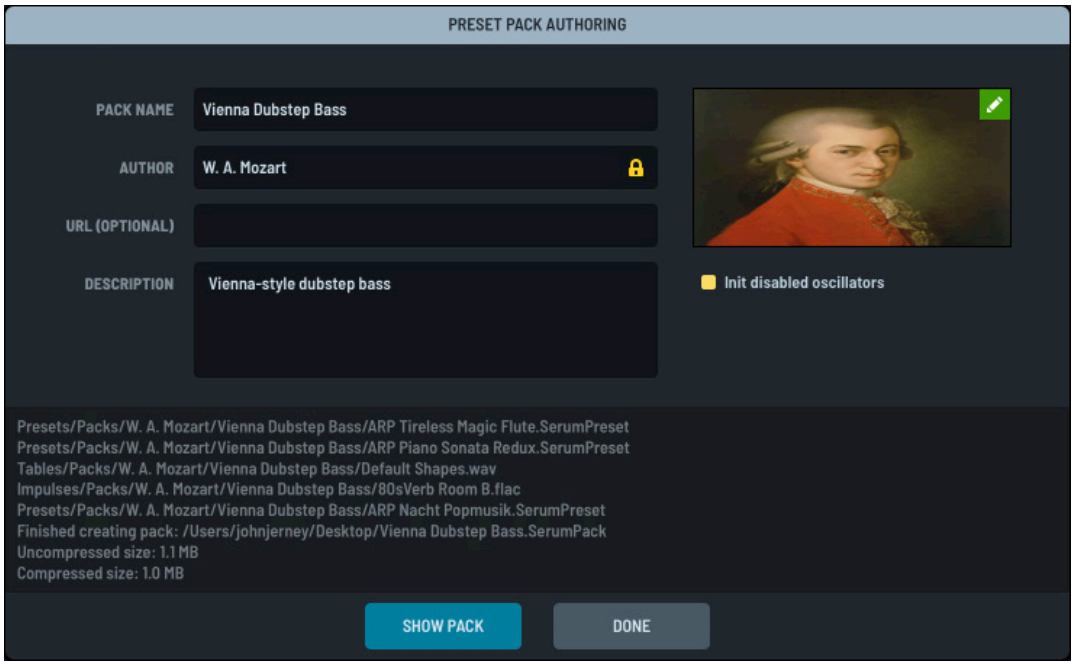
The following shows an example preset pack:



Sample Preset Pack

Click the CREATE PACK button. A dialog appears allowing you to specify the file name and folder.

The dialog displays details about how the pack was created.



Preset Pack Details

Click the SHOW PACK button to show the location of the artist pack. The name of the file is:

```
<pack-name>.SerumPack
```

Click the DONE button to dismiss the dialog.

Appendix D: Editing the Serum Preferences File

Serum stores application preferences in a special file called **`Serum2Prefs.json`**. You can find this file in the following locations:

- (macOS) **`~/Library/Preferences/`**
- (Windows) **`%APPDATA%\Xfer\Serum 2\`** (use Windows Key + R to access)

This file stores the preferences accessible on the **Preferences** page, the last known path to the Serum Presets folder, and (optionally) power user options, among other options.

Serum automatically recreates this file if it is missing, so you can reset your preferences to the factory defaults simply by deleting this file and restarting Serum.

You probably shouldn't change many of the settings. However, there are a few power user settings that you might want to explore.

Important: When editing the JSON file, use a text editor application such as TextEdit (macOS) or Notepad (Windows). Do not use an application that saves the file in any format other than text (the standard format for JSON files).

For each preference described below, do the following:

1. Edit the **`Serum2Prefs.json`** file using a text editor.
2. Make the changes, as described in the corresponding section below.

   Very Important: Each line in the configuration file ends with a comma (,). Ensure that you keep the comma at the end of each line when you edit the file.
3. Save the file.

Changing the Default Artist Name

You can specify the default artist name for the **Init** preset. This is the name that appears in the **ARTIST** field when you initialize a new preset using the main menu.

Locate the following line:

```
"Default Author": "  ",
```

Type the name you want to use between the quotation marks. For example:

```
"Default Author": "Wolfgang A. Mozart",
```

Any presets made from scratch will now have the name you specified listed as the artist.

Enabling Preset Changes using a MIDI Controller

You can configure Serum to allow you to step through presets using a physical controller (such as a button on a MIDI keyboard) by mapping the controller's MIDI CC number to the corresponding preset selection parameter.

In the Serum preferences file, locate the following line:

```
"Enable CCForRockers": 0,
```

Change the line to the following:

```
"Enable CCForRockers": 1,
```

Next, locate the following lines:

```
"CCForRocker Preset +": -1,

"CCForRocker Preset -": -1,
```

The first line maps the preset forward arrow (**>**) to a MIDI CC number; similarly, the second line maps the preset backward arrow (**<**) to a MIDI CC number.

Update the -1 values on each line to the appropriate MIDI CC number for your controller (button).

For example, if you would like to assign MIDI CC 21 to the preset forward arrow and MIDI CC 22 to the preset backward arrow, change the lines to the following:

```
"CCForRocker Preset +": 21,

"CCForRocker Preset -": 22,
```

For the settings discussed in this section, the value -1 means unassigned.

To remove the MIDI CC assignments that you configured in this section, edit the file and reset the corresponding values to -1.

A MIDI controller button mapped to a CC typically sends a value of 127 when pressed and a value of 0 when released. In Serum, a value of 64 or above triggers the action (changes the preset). Similarly, Serum needs to receive a value below 64 before the action can be triggered again.

Using Notes to Trigger Preset Changes

Alternatively, you can configure Serum to allow you to step through presets using MIDI notes. The configuration process is very similar to mapping controller MIDI CC numbers, as described in the previous section.

In the Serum preferences file, locate the following line:

```
"Enable CCForRockers": 0,
```

The value on this line might already be 1 if you followed the procedure in the previous section. Change the line to the following:

```
"Enable CCForRockers": 2,
```

Next, locate the following lines:

```
"CCForRocker Preset +": -1,

"CCForRocker Preset -": -1,
```

Again, the values might be different than -1. As before, the first line maps the preset forward arrow (**>**) to a MIDI note number; similarly, the second line maps the preset backward arrow (**<**) to a MIDI note number.

Update the values on each line to the appropriate MIDI note number.

Enabling Oscillator Preset Changes using a MIDI Controller

You can similarly configure Serum to allow you to step through oscillator presets using a physical controller by mapping the controller's MIDI CC value to the corresponding preset selection parameter.

In the Serum config file, locate the following line:

```
"Enable CCForRockers": 0,
```

Change the line to the following:

```
"Enable CCForRockers": 1,
```

Next, locate the following lines:

```
"CCForRocker OSC A +": -1,

"CCForRocker OSC A -": -1,
```

The first line maps the **OSC A** preset forward arrow (**>**) to a MIDI CC value; similarly, the second line maps the **OSC A** preset backward arrow (**<**) to a MIDI CC value.

Update the -1 values on each line to the appropriate MIDI CC value for your controller (knob).

For example, if you would like to assign MIDI CC 23 to the **OSC A** preset forward arrow and MIDI CC 24 to the **OSC A** preset backward arrow, change the lines to the following:

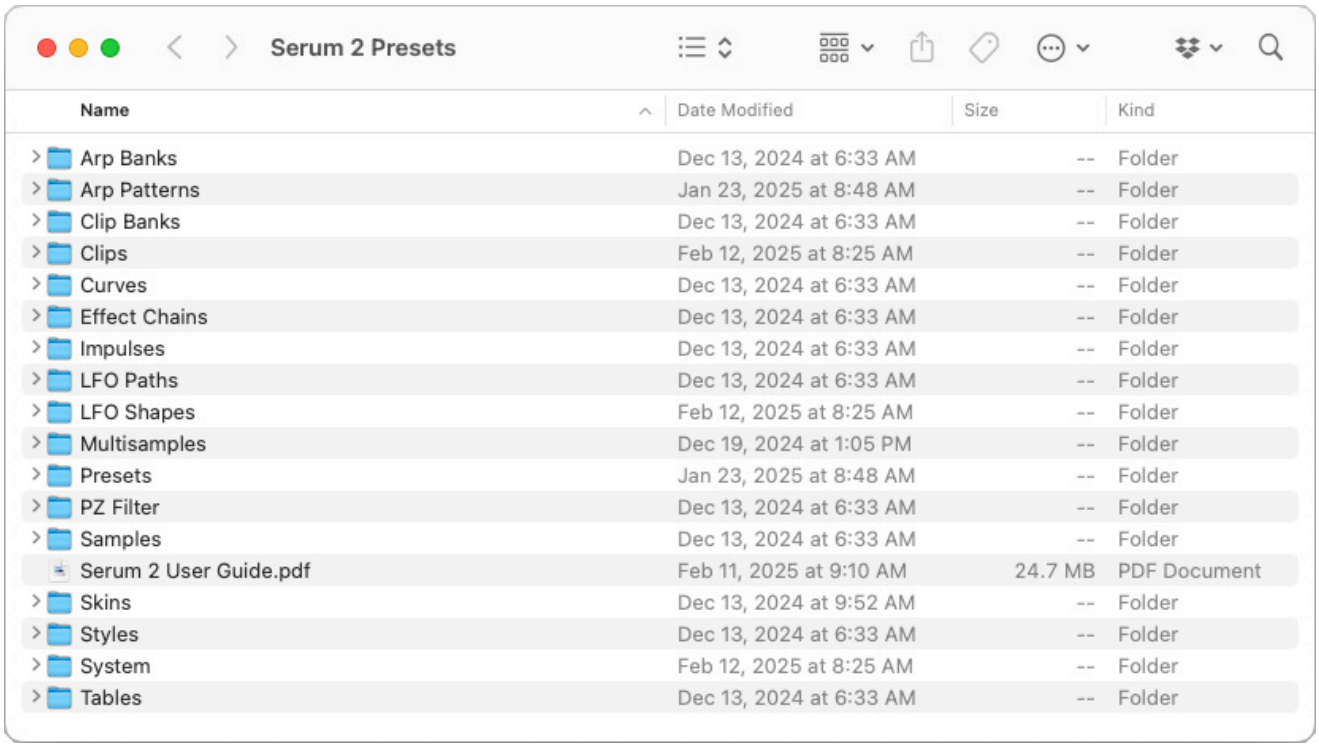
```
"CCForRocker OSC A +": 23,

"CCForRocker OSC A -": 24,
```

Remember to include the comma (,) at the end of each line. This is important. Also, recall that you can reset these settings by changing the values back to -1.

Note that **OSC B**, **OSC C**, and **OSC N** (the noise oscillator) have similar settings in the configuration file. Use the same procedure to assign those to appropriate MIDI CC values for your controllers (knobs).

Appendix C: Exploring the Serum File Structure

This appendix describes how Serum stores data on your computer drive.



Serum File Structure

The Serum Presets folder contains all the files that Serum reads and writes, except the preferences file. Serum installs this folder to the following locations by default:

- (macOS) **`/Library/Audio/Presets/Xfer Records/`**
- (Windows) **`/Documents/Xfer/`**

The following describes some of the more important folders in the file structure:

Folder	Description
Arp Banks	Contains factory-supplied and user-saved ARP (arpeggiator) banks.
Clip Banks	Contains factory-supplied and user-saved CLIP banks.
Multisamples	Contains factory-supplied and user-saved multisample instruments.

Folder	Description
LFO Shapes	Contains LFO shapes (**`.XferShape`** files) that are in the same file format as the shape files used in the LFOTool plug-in. LFO shapes appear in the following locations: • The LFO section of Serum • The waveshaper (in the FX section, when the X-Shaper effect type is selected and the waveshaper graph is displayed) • The Remap editor (when WARP mode is set to Remap and the graph is displayed)
Presets	Contains subfolders holding the presets that you see in the presets browser and menu. These include the factory presets together with any presets you save.
Samples	Contains factory-supplied (tonal and non-tonal) and user-saved samples.
System	Contains the four formula files: • **`FormulaFactoryMultis`** • **`FormulaFactorySingles`** • **`FormulaUserMultis`** • **`FormulaUserSingles`** You can edit any of these text files, but the intent and recommendation is to edit **User** files and leave **Factory** files untouched. See "Using the Formula Parser" on page 304 for more information about formulas. The **System** folder also contains a **MIDIccMaps** folder, which is where any MIDI CC maps you create are stored. Finally, the **`User.dat`** file holds your registration information.
Tables	Contains subfolders holding the wavetables that you see in Serum. These include the factory presets along with any wavetables you save. Wavetables are special Serum-saved WAV files. You can create your own subfolders in the Tables folder, but Serum does not scan deeper (no sub-subfolders are scanned).

Appendix E: Creating Wavetables

This appendix provides additional information about creating wavetables. Refer to “Using the Wavetable Editor” on page 274 for detailed information about how to create your own wavetables.

What Makes for a Good Wavetable?

Like a good song, creating a good wavetable is about getting that careful balance between consistency and contrast. Too much consistency or repetition and things won’t feel like much of a journey. Too much contrast and it’s difficult to draw a connection between random events (like turning the dial on the radio feels disconnected and hard to comprehend as intentional).

Good wavetables generally have the following basic characteristics:

- Contains many correlated frames

  This typically is a sound with a lot of frames, but ones that fluidly work together (think of a pluck string that decays across many cycles). In general, there might be a lot of individual cycles, but they all go together well and relate nicely with each other.

  Another example would be a synthesizer sample. You want many cycles so it doesn’t feel too “steppy,” however all the frames should feel as part of a collection (a similarity/sweep across them).

- Contains few frames

  Many of the factory presets in popular wavetable synths contain only four or five cycles. This provides some variety within the table, but not so many waveforms that it feels disconnected or crazy.

Table Ordering

It typically makes sense to have the frames progress from dull to bright, or vice-versa. You may have a situation where you want the spectrum to peak somewhere in the middle, but probably not.

You can drag the thumbnails at the bottom of the Waveform Editor to rearrange them. This way, when you move the **WT POS** knob, it feels as if you’re traveling in a straight line, instead of in a zig-zag fashion (spectrally speaking).

Interpolation

If you have four frames and automate the **WT POS** knob, you'll hear four discrete tones immediately jumping from one to the next. This is typically undesirable.

In the Wavetable Editor, you can select **MORPH > Morph - Crossfade**. This is what many wavetable synths do automatically in every preset wavetable. However, there are times when you might prefer discrete (non-interpolated) waveforms.

Creating Wavetables from Scratch

There are several ways to create source audio for a wavetable, including the following:

- Drawing

  The grid size in lower-right determines snapping. Try different sizes such as 6 or 12 to bring out +7 and +19 harmonics.

- Drawing in FFT bins

  Right-click in the bins for a pop-up menu with more options (including random, among others).

- Moving drawings up to FFT bins

  Use the curved arrow near the top left and add a new table (using the > button in the lower left). Then make some adjustments and repeat.

  About four frames seem to be a popular number in many other software synthesizers.

Appendix E: Optimizing Serum

This appendix describes a series of easy-to-adopt approaches to optimizing your sound design results in Serum.

Exploring CPU Optimization

Serum is designed to optimize audio processing and CPU performance. However, there certain approaches that you can incorporate to help your sounds achieve the very best performance and maintain the highest-quality in sound design.

Managing Unison

Unison is a powerful tool for enhancing the depth and richness of a sound by layering multiple voices slightly detuned or panned, creating a fuller, more powerful tonal presence. However, overuse of unison can potentially lead to quality and performance issues.

Consider doing the following:

- Keep unison counts low

  Using more than three to seven unisons per oscillator is often unnecessary and can potentially negatively impact efficiency. This is because higher unison counts not only significantly increase CPU usage, but can also introduce phasing issues, potentially degrading your sound quality.

- Use chorus instead of unison

  Instead of stacking unisons to create a thick, chorused sound, use a dedicated FX bus with a chorus effect. This approach offers two principal advantages. First, by apply the effect once instead of processing it for every voice, this approach is considerably more CPU friendly.

  Second, this provides greater flexibility by allowing easier tweaking and layering of effects without duplicating processing effort.

Appendix F: Keyboard Shortcuts

This appendix describes the keyboard shortcuts available in Serum, organized by category.

Presets and Presets Browser

Task	macOS	Windows
Position the cursor in the search field. Note that this only works when you are already in the presets browser.	Cmd-F	Ctrl-F
Move up a preset	↑	↑
Move down a preset	↓	↓
Play preset preview	→	→
Stop preview	←	←
Select multiple presets	Shift-click-presets	Shift-click-presets
Delete selected presets	Shift-Backspace	Shift-Backspace
Save preset using same name without dialog	Option-click save button	Alt-click save button

Controls (Knobs and Sliders)

Task	macOS	Windows
Fine tune setting	Shift-drag control	Shift-drag control
Reset control	Cmd-click control	Ctrl-click control

Modules (Oscillators and Filters)

Task	macOS	Windows
Copy module without modulations	Option-drag module label	Alt-drag module label
Copy module with modulations	Shift-Option-drag module label	Shift-Alt-drag module label

Task	macOS	Windows
Cycle through filter display modes	Option-click display	Alt-click display

The copy module shortcuts also apply to oscillators on the **MIXER** page.

Sample/Granular/Spectral

Task	macOS	Windows
Manually add a slice	Option-click	Alt-click
Remove an existing slice	Option-click slice	Alt-click slice
Change window amount and skew (Granular)	Option-click and drag button	Alt-click and drag button

Audio

Task	macOS	Windows
Display **`Renders`** folder	Option-click wave icon (next to logo)	Alt-click wave icon (next to logo)
Copy saved preset to computer	Shift-drag wave icon to Finder	Shift-drag wave icon to Explorer

FX

Task	macOS	Windows
Copy module without modulations	Option-drag module	Alt-drag module
Copy module with modulations	Shift-Option-drag module	Shift-Alt-drag module
Bypass all FX on a bus	Option-click any bypass button	Alt-click any bypass button
Expand/revert the size of the FX rack and list view	Option-F	Alt-F

Matrix

Task	macOS	Windows
Expand/revert the size of the matrix view	Option-F	Alt-F

LFOs

Task	macOS	Windows
Copy LFO settings	Option-drag LFO tab to another LFO tab	Alt-drag LFO tab to another LFO tab
Add/remove LFO point	Double-click	Double-click
Draw steps (at grid size)	Shift-click draw	Shift-click draw
Snap point to grid size	Option-drag point	Alt-drag point
Move all curve points at once	Option-drag any curve point	Alt-drag any curve point
Select multiple points	Click-drag on background	Click-drag on background
Select multiple points for relative movement	Cmd-click-drag a point	Ctrl-click-drag a point
Set point as loopback position	Shift-Cmd-click a point	Shift-Ctrl-click a point
Copy LFO shape to wavetable	Option-drag an LFO tab to a wavetable	Alt-drag an LFO tab to a wavetable

Modulation

Task	macOS	Windows
Change modulation type (directional or bidirectional)	Shift-Option-click modulated control	Shift-Alt-click modulated control

Clips

Task	macOS	Windows
Apply changes to parameters to all clips	Option-edit parameter	Alt-edit parameter
Scroll piano roll	Cmd-Option-drag background	Ctrl-Alt-drag background
Zoom piano roll	Shift-Option-drag background	Shift-Alt-drag background
Zoom to marque	Shift-Cmd-drag a marquee	Shift-Ctrl-drag a marquee
Scroll piano roll horizontally	Shift-mouse wheel	Shift-mouse wheel
Zoom piano roll vertically	Option-mouse wheel	Alt-mouse wheel
Zoom piano roll horizontally	Cmd-mouse wheel	Ctrl-mouse wheel

Clips (Note Movements)

Task	macOS	Windows
Move selected notes up one semitone	↑	↑
Move selected notes down one semitone	↓	↓
Move selected notes one position to the right	←	←
Move selected notes one position to the left	→	→
Move selected notes up one octave	Shift ↑	Shift ↑
Move selected notes down one octave	Shift ↓	Shift ↓
Shorten selected notes	Shift ←	Shift ←
Lengthen selected notes	Shift →	Shift →

Clips (Note Operations)

Task	macOS	Windows
Cut selected notes	Cmd-X	Ctrl-X
Copy selected notes	Cmd-C	Ctrl-C
Paste selected notes at timeline	Cmd-V	Ctrl-V
Duplicate selected notes at timeline	Cmd-D	Ctrl-D
Delete selected notes	Backspace	Backspace
Chop selected notes	Cmd-U	Ctrl-U
Conform selected notes to scale	Cmd-K	Ctrl-K
Apply legato to selected notes	Cmd-L	Ctrl-L
Mute selected notes	0 (zero)	0 (zero)
Quantize selected notes	Cmd-Q	Ctrl-Q
Reverse order of selected notes	Cmd-R	Ctrl-R
Scale time 50%	/	/
Scale time 200%	*	*
Double entire clip	Cmd-E	Ctrl-E
Select all notes	Cmd-A	Ctrl-A

Wavetable Editor

Task	macOS	Windows
Move frame to new location	Click-drag frame	Click-drag frame
Select range of frames	Shift-click a frame	Shift-click frame
Shift the phase offset to the right	Shift-Cmd-click phase offset	Shift-Ctrl-click phase offset

Additional Credits

The following producers, artists, and sound designers contributed to Serum.

Serum 2 Factory Multisamples

AnSolas
Brandon Seliga
Edward “shreddward” Braillif
Matt Aimonetti
Robin Tyndale
Rodrigo Montes
Steve Duda

Serum 2 Factory Wavetables and Samples

Alice Efe
Drumsound & Bassline Smith
Edward “shreddward” Braillif
ill.gates
Matt Aimonetti
Mr. Bill
Sharooz
Splice
Steve Duda
Virtual Riot

Serum 2 Factory Presets

7 SKIES
AGENT METHOD
Alice Efe
AnSolas
Audiotent
Beatdemon
Caster
CFA-Sound
Chord Shore
DATABROTH
DnBline Smith
Edward “shreddward” Braillif
Electric Himalaya
Endov Lane
ERB N DUB
Genji Siraisi
Gigantor
J. Scott G. / Libra Rising
Leo Lauretti
Level 8
LP24 Audio
Matt Aimonetti
Mr. Bill
NEST Acoustics
Oiko
Paul Laski
Scott Diaz
Skope
Specimen A
Splice
Steve Duda
SynthHacker
Tunecraft
Van Derand
Wisteria Motif