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Welcome to Sampling Master

What is Sampling Master

Sampling Master is a powerful tool designed for mapping samples, creating loops within samples, recording audio output from hardware and software instruments, and organizing your sample library.
With Sampling Master, creating sampled instruments is straightforward, efficient, and intuitive.

What Sampling Master is not

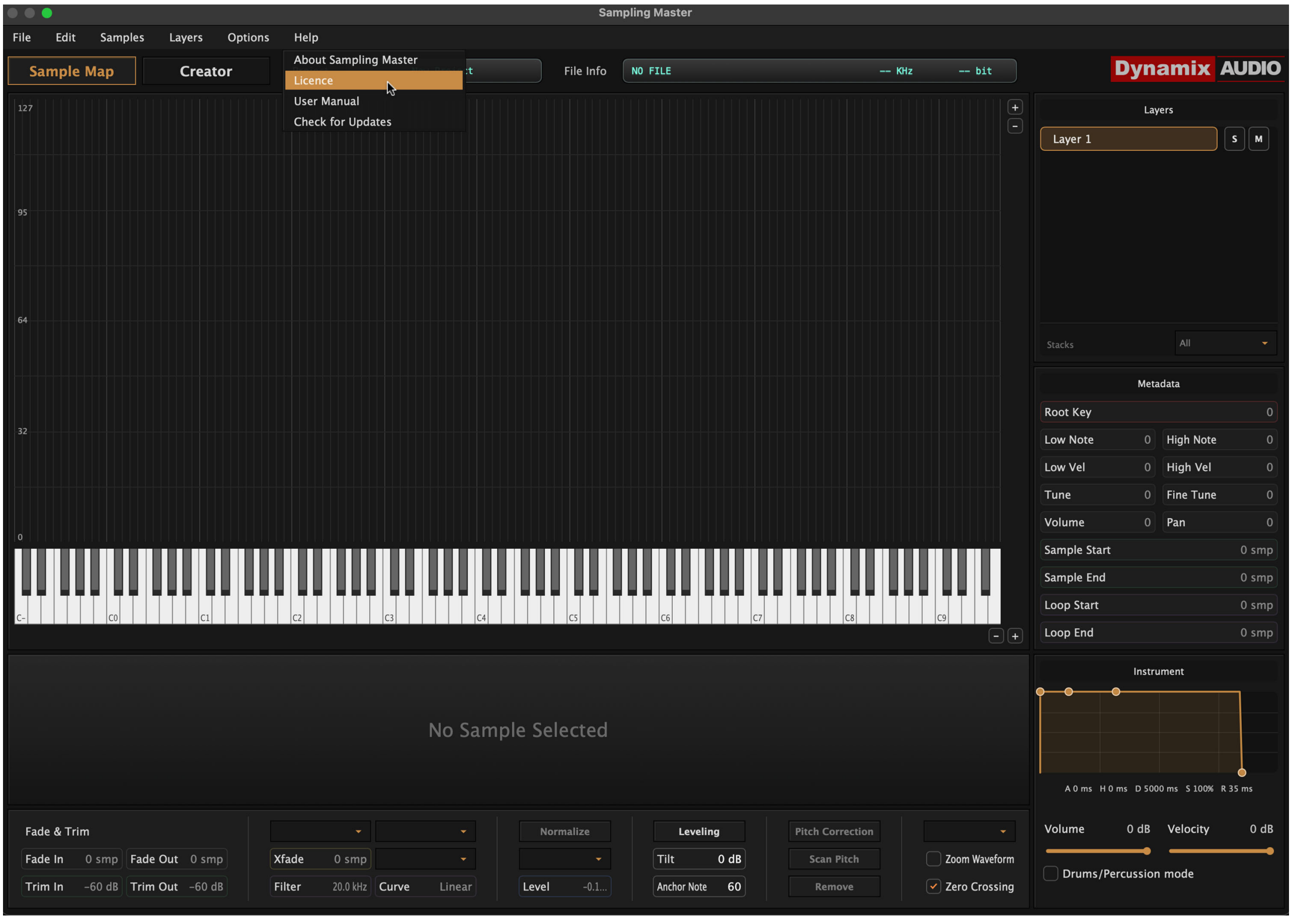
Sampling Master is not a sampler intended for live or studio performance. It is not a plugin, but a standalone application that can host plugins.
While creating sampled instruments, Sampling Master can play back samples and function as a basic playback engine, allowing preview during editing. However, it lacks the advanced processing and modulation capabilities required of a full-featured sampler.

Licencing

Sampling Master is licensed to a single user and may be used on one computer at a time.
You can move your licence between computers by deactivating it on one machine and activating it on another.

To activate your licence:

- Open the Help menu
- Select Licence



The activation process is quick and straightforward.

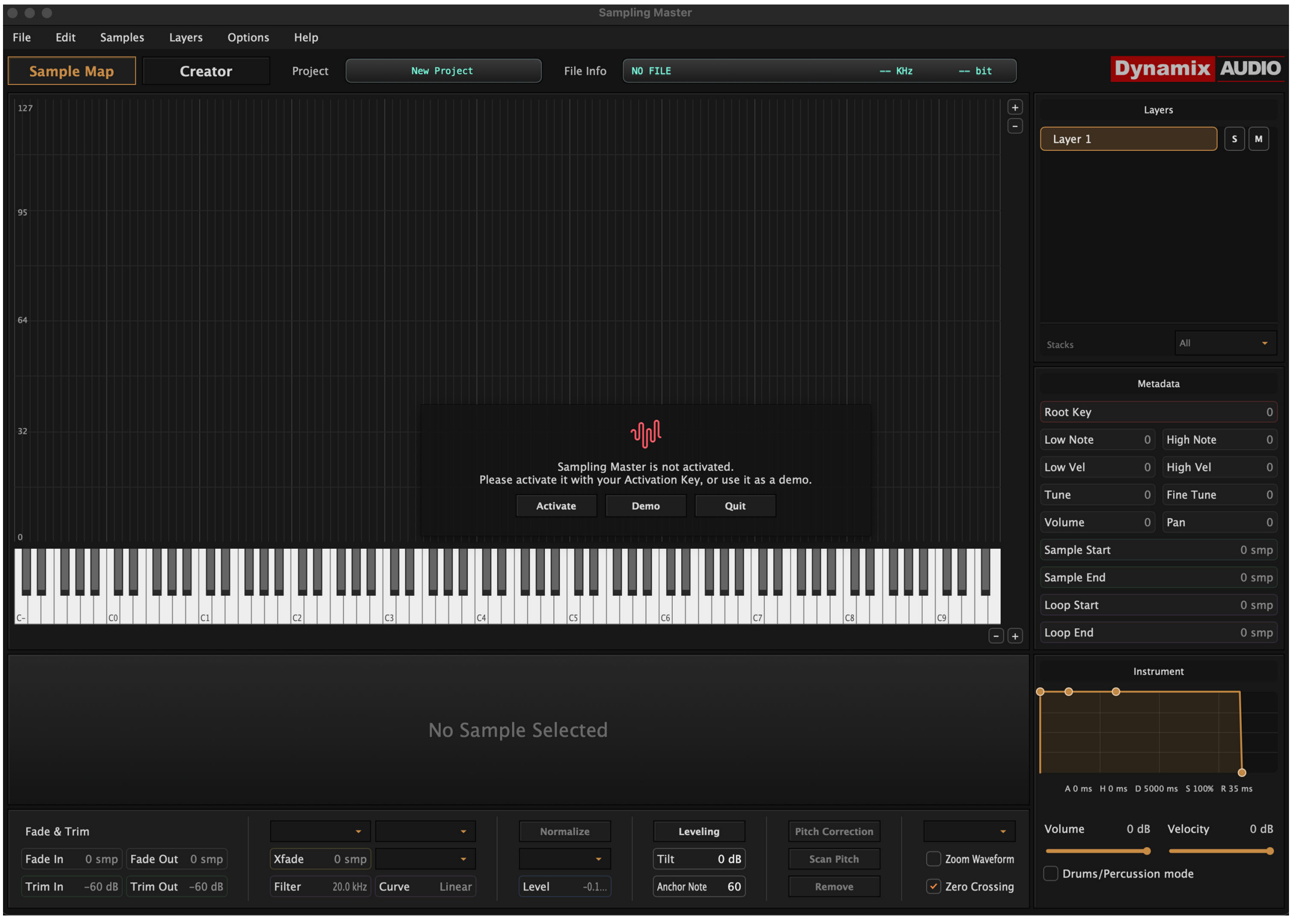
Important:

Before uninstalling Sampling Master, you must deactivate your licence.
If you uninstall without deactivating:

- The licence will remain marked as active on the server
- You will not be able to activate it again
- Only one active installation is allowed per licence

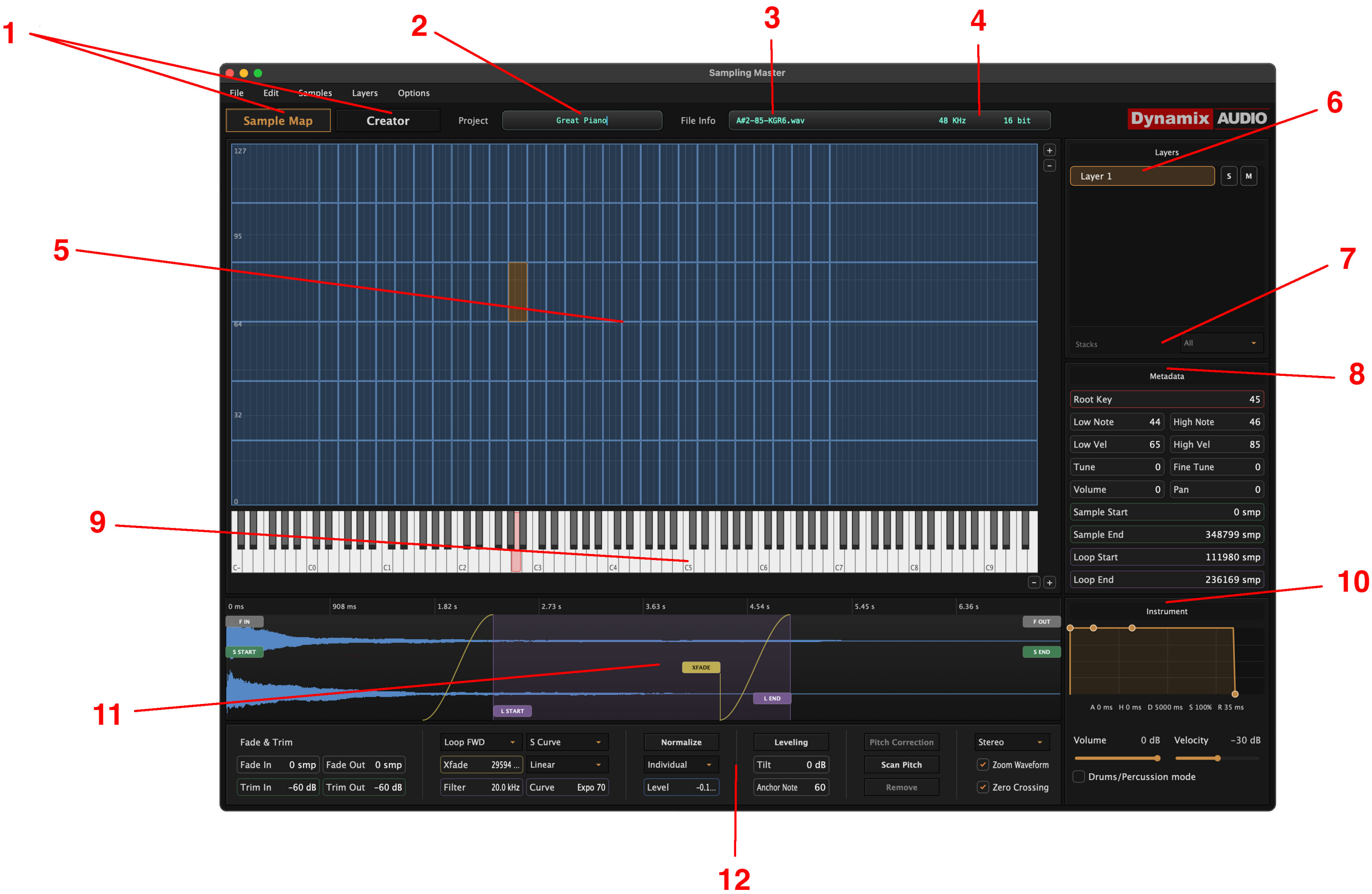
Demo mode

Sampling Master is available as a fully functional Demo version, allowing you to test all features before purchasing a licence.
The demo has no time restrictions, so you can evaluate the software at your own pace.
The only limitation is that audio export is disabled for the Demo version.



The interface

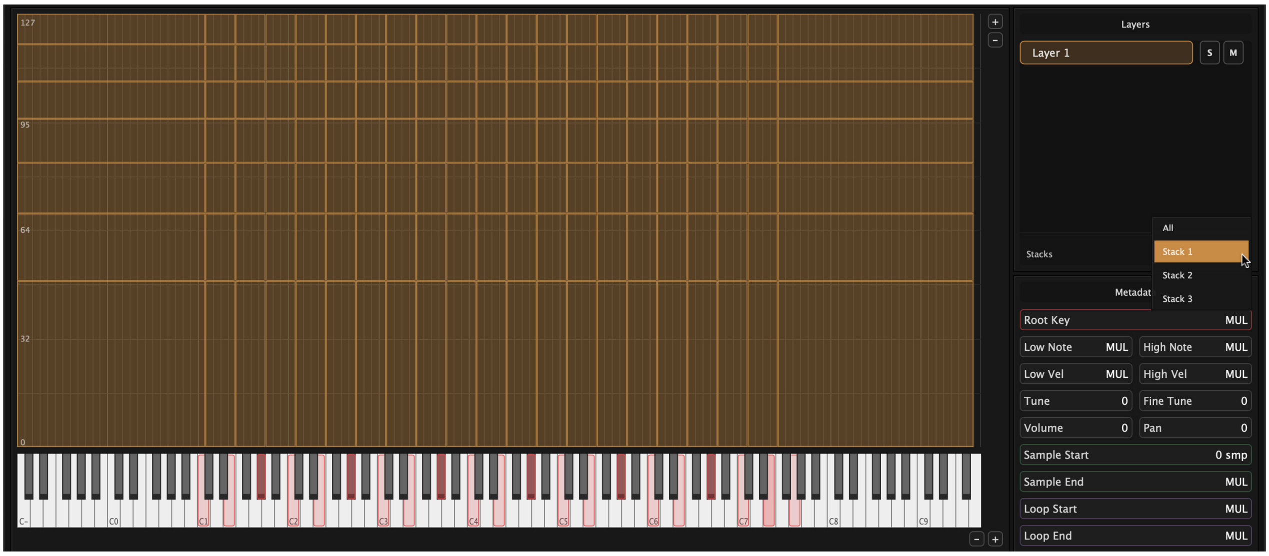
Sample Map



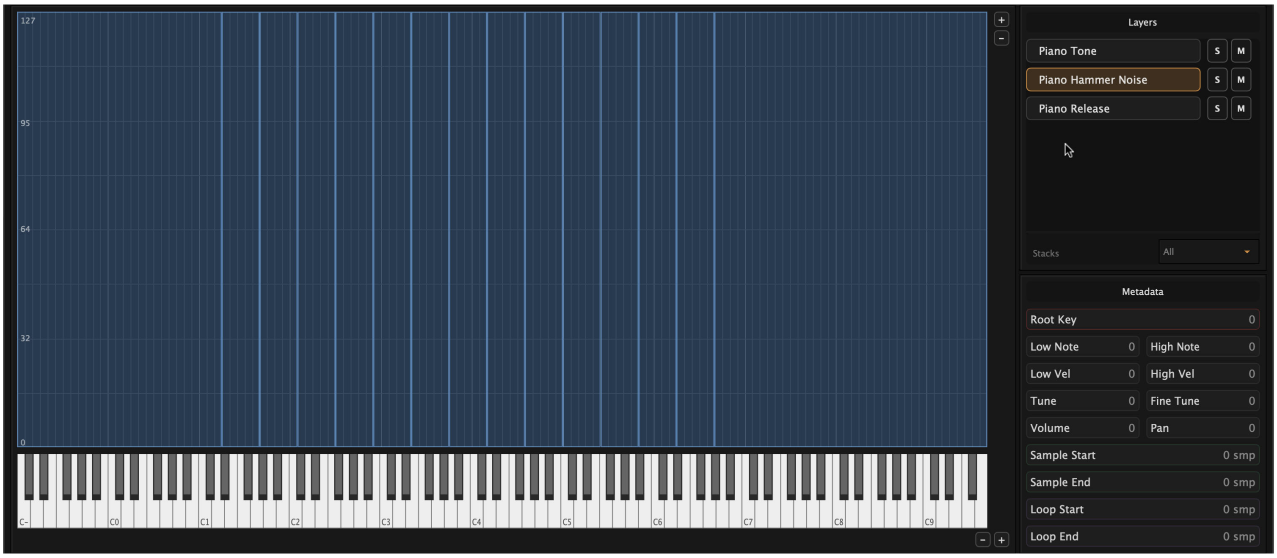
1. Mode
2. Project name
3. Selected sample name
4. Selected sample properties
5. Mapping panel
6. Layers panel
7. Stacks control
8. Sample metadata panel
9. Virtual keyboard
10. Basic instrument settings
11. Sample waveform panel
12. Sample render panel

Sample Map Mode allows you to import samples and map them as needed, either manually or using one of the included auto-mapping algorithms. You can edit samples by normalizing audio, applying fade-ins and fade-outs, detecting loop points manually or using automatic algorithms, and adjusting pitch and level as required.

Samples are mapped across two dimensions: horizontally by note and vertically by velocity. When samples overlap, they are automatically grouped into stacks. You can control which stack is currently visible, while all stacks belong to the same layer.



A Layer contains a single sample map along with its associated samples. Terminology varies across different samplers: in SFZ, it is referred to as a Group; in SF2, as an Instrument; in Native Instruments Kontakt, it is also called a Group. In the Giglad native sampler, this concept is referred to as a Layer.

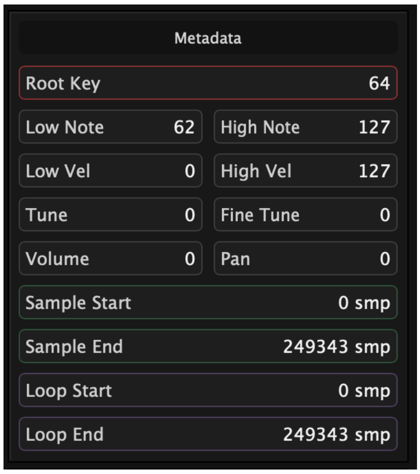


The Metadata panel contains all information related to a sample and its properties within the sample map. This includes the sample's root key, start and end positions, loop start and end points, as well as parameters such as volume, tuning, and the high and low ranges for velocity and note mapping.

Metadata is preserved in two places: within the sample file itself and within the sample map.

Metadata does not alter the audio data. It exists as descriptive information stored in the file and must be read—either from the sample or from the sample map—by the sampler in order to be applied.

Most samplers read metadata from the sample map; however, Sampling Master also embeds this metadata directly into the files, enabling easier management and manipulation later in the workflow.



The Processing panel applies structural changes directly to the audio file after the file is rendered. These changes are rendered into the file itself, ensuring consistent playback across all sample players.

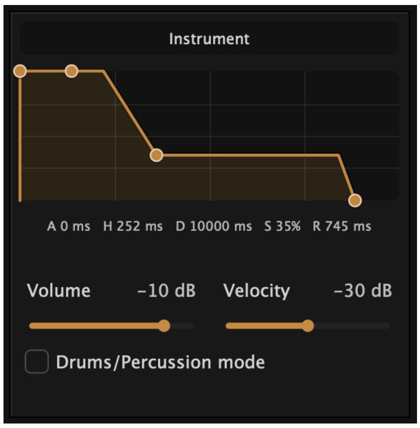
Within the Processing panel, you can apply fade-ins and fade-outs, as well as define accurate start and end points using trim functions.
Creating loops is also handled within the Processing panel, as loops are fully rendered into the file. The sampler then only reads the loop start and end positions. However, advanced loop creation features provided by Sampling Master—such as loop crossfade, loop crossfade filter, normalization, and pitch correction—are all rendered directly into the file.

Waveform panel displays a visual representation of the sample, including its start and end positions, loop region, crossfade, and curve shapes, allowing precise editing and accurate loop point placement.
The waveform can be zoomed so that its full height is visible regardless of the actual normalization level. When a sample is normalized, its level corresponds to the fully zoomed waveform.

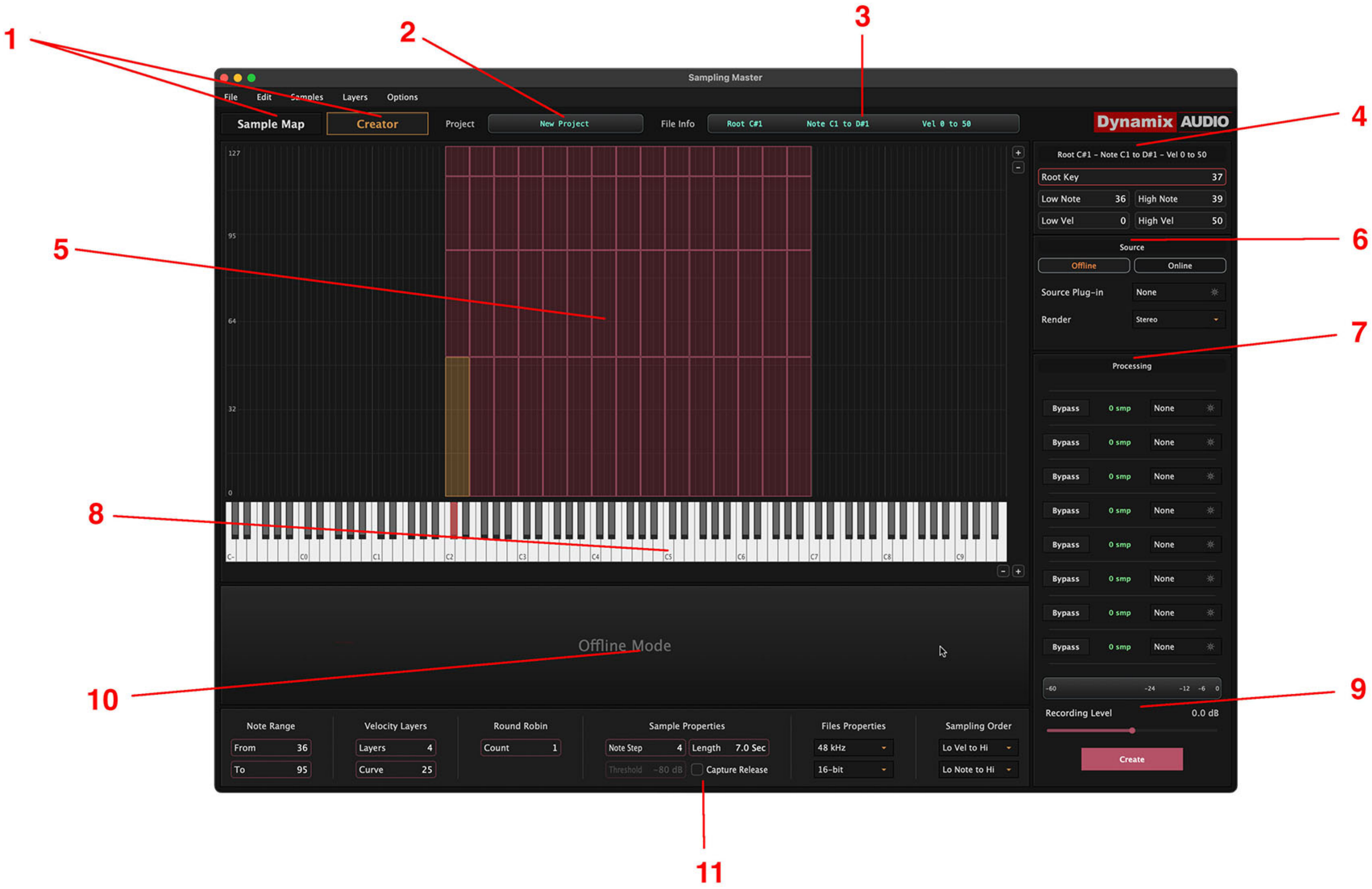
Zero Crossing option allows you to automatically align sample start and end points to the nearest zero crossing. This ensures that any adjustment of sample start, end, loop start, or loop end positions searches for the optimal zero crossing nearby.

Instrument panel provides internal control over playback within the current project. The envelope included here is intended only for the Sampling Master playback engine and is deliberately basic. Your target sampler will typically offer more advanced envelopes, filters, and modulators. This panel exists solely to make the sample map playable during the editing process.

Drums/Percussion mode is relevant for SF2 and MSAMP formats used by the Giglad native sampler, as these formats distinguish between tonal and drum instruments.

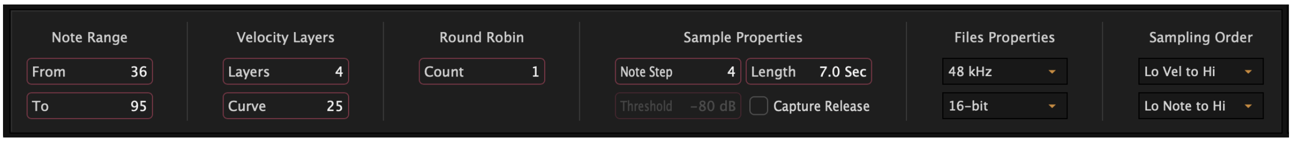


Creator



1. Mode
2. Project name
3. Selected region properties
4. Selected region metadata
5. Regions ready for sampling
6. Source for sampling
7. Processing for sampling
8. Virtual keyboard
9. Recording level
10. Sampled region waveform
11. Sampling parameters

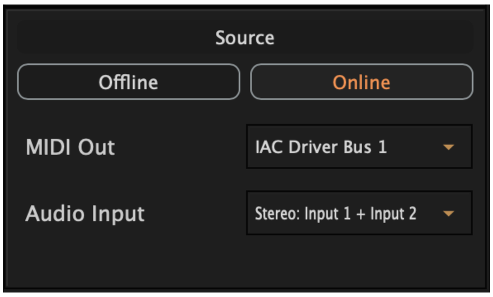
Creator Mode allows you to automatically record and sample audio output from hardware and software instruments, providing a fast and efficient way to consolidate your sound library into a single sampler.
This workflow can be used to reduce CPU load in studio environments by replacing real-time synthesis processing with sampled instruments, or to prepare sounds for LIVE performance by organizing them into a unified, playable library. You can make sampled sounds for hardware workstations such as Korg, Roland, and Yamaha workstations and arrangers, or for the Giglad software arranger.



The Sampling Parameters panel allows you to define the exact settings for newly created samples. This includes the note range, velocity layers, and velocity mapping type, such as exponential or logarithmic, and round-robin.
It also controls per-sample properties such as sample length, the number of notes to be sampled within the selected range, file settings, and the sampling order.
When Capture Release is enabled, Creator captures the release portion of the sound. The release samples are then placed onto a separate layer, which is automatically set to trigger on a Note Off MIDI message.

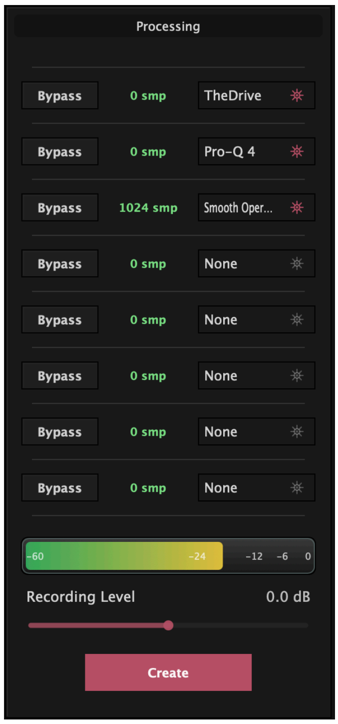
Round Robin creates a new layer for each round-robin pass. In Sample Map mode, Sampling Master treats these layers as regular layers. Your sampler is responsible for playing them as round-robin layers.

You can sample patches from both hardware and software instruments.



Online mode sends MIDI data from Sampling Master to your hardware instrument while simultaneously recording the audio input from the selected channels.

Offline mode allows sampling of virtual instruments by instantiating a plugin instrument directly, similar to a DAW workflow.
Right-click the instrument name to open its window.



For both Offline and Online modes, samples can be processed during recording using up to eight plugin inserts.
Each plugin slot displays the latency of the loaded plugin in samples. Care should be taken with plugins that introduce latency or use oversampling, as this delay will be printed into the recorded audio.
This can later be corrected using the Trim In function in Sample Map Mode.

Attention should also be given to recording levels and non-linear processing such as compressors, saturators, and other dynamic effects, as their behaviour depends on the input level fed into the plugin chain.

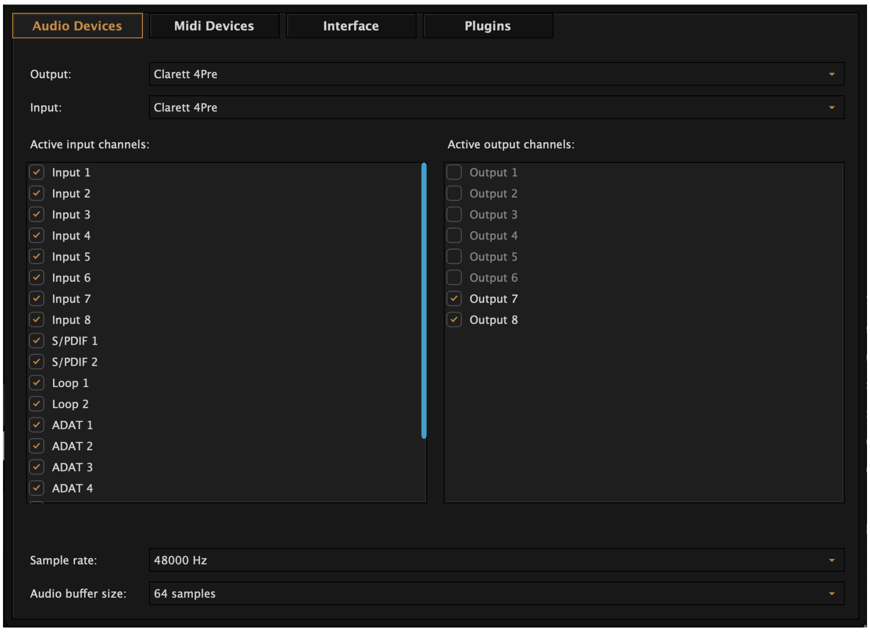
Right-click the plugin name also opens its window.

Please respect all third-party EULAs and copyright laws when sampling from hardware or software instruments.

Settings

Audio Devices

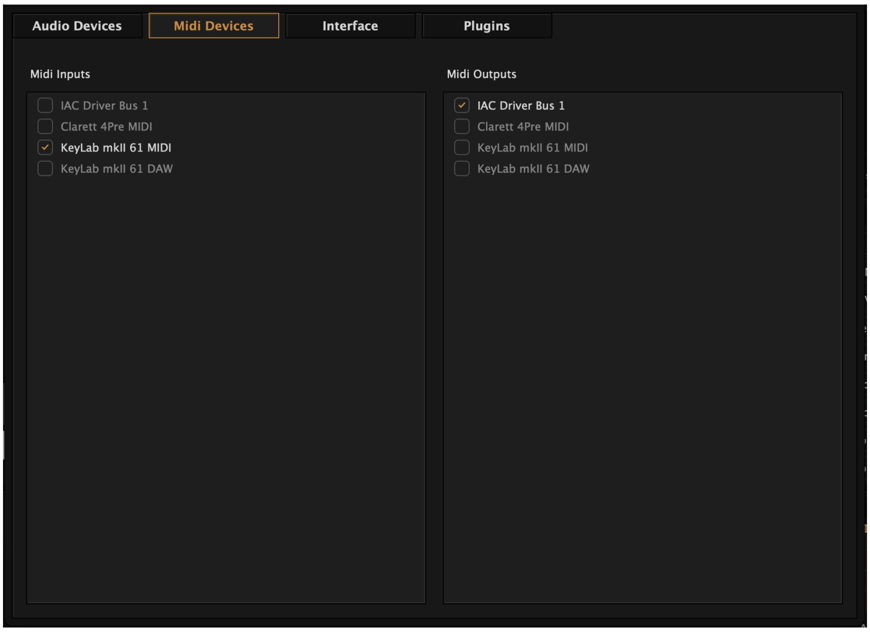
Select the audio device used for both playback and recording.



On Windows, it is recommended to use an ASIO-compatible audio device to ensure low-latency playback and recording.

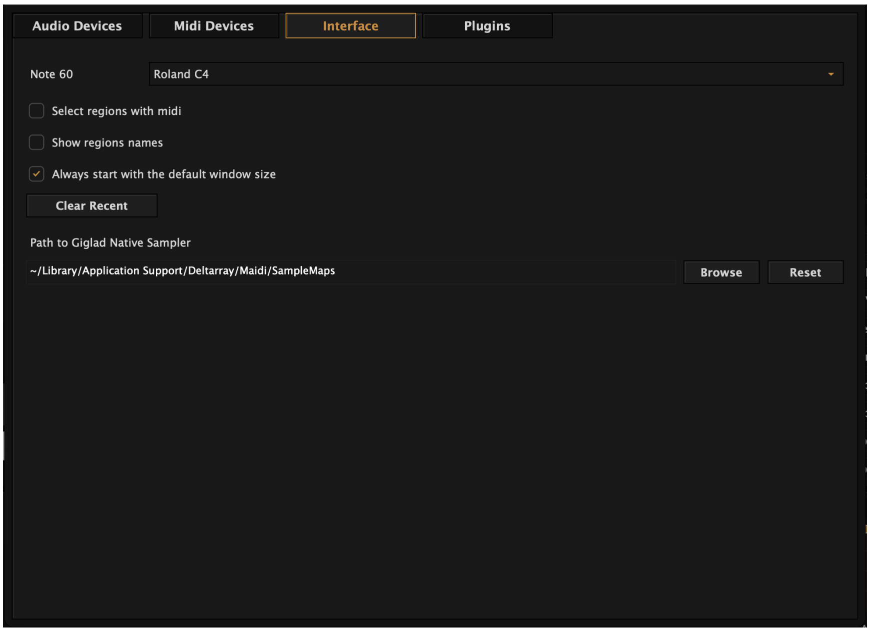
Midi Devices

Select and enable your MIDI device, input and output devices for playing and controlling external hardware.



Interface

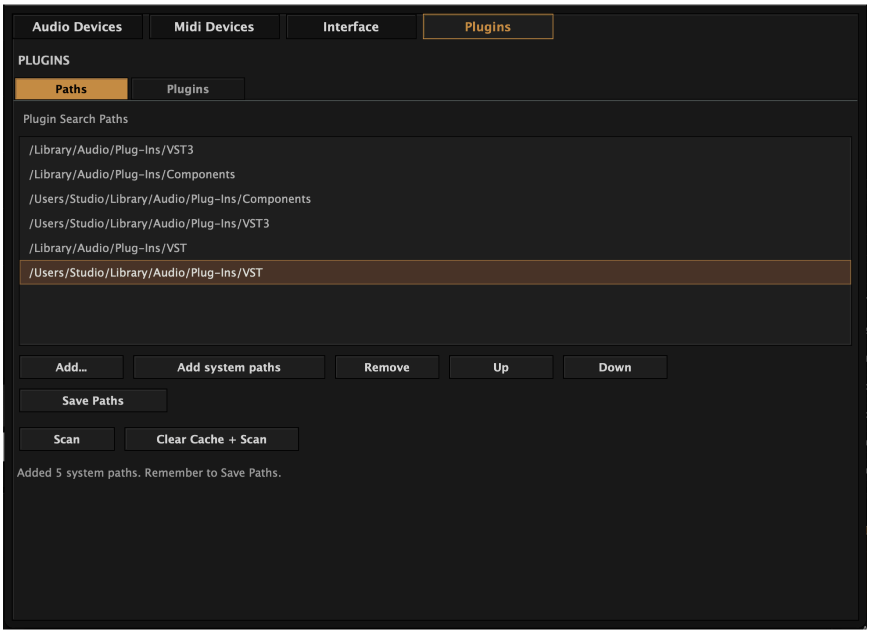
In the Interface section of the settings, you can customize the behavior of Sampling Master to suit your workflow and preferences.



Depending on the standard you choose, you can set MIDI note 60 (middle C) to either Roland C4 or Yamaha C3.

Plugins

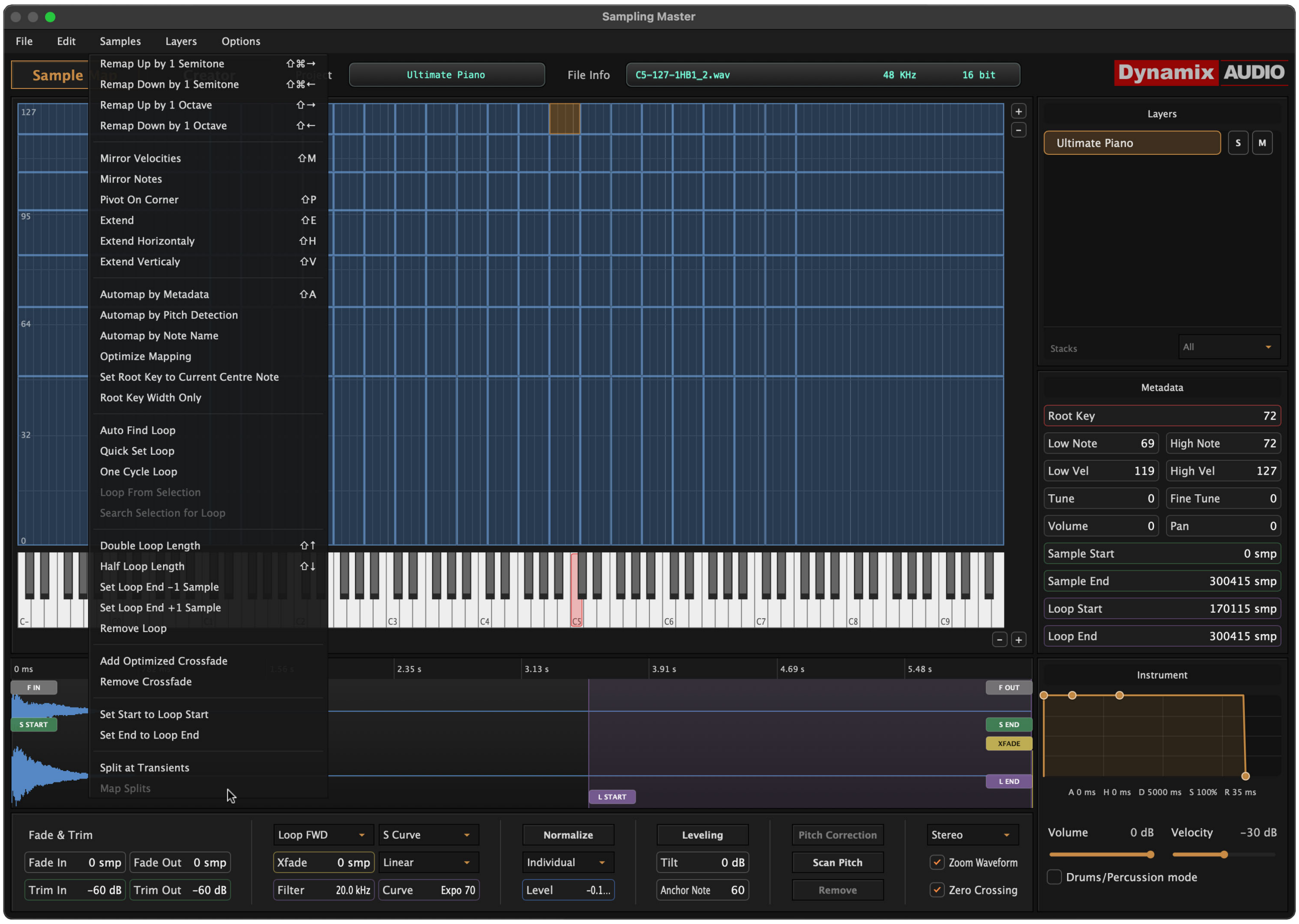
From this settings page, you can also scan all installed plugins. Plugin scanning paths can be customized, or you can restore them to the default system locations.



Selecting Scan will add newly installed plugins to the database. Selecting Clear Cache + Scan will remove all previously scanned plugins and rebuild the plugin list from scratch.

Samples

Accessing Samples entry in the menu provides a complete list of features available in Sampling Master.
This menu can be opened either from the top interface menu or by right-clicking a sample in the Mapping panel.
The Samples menu is clearly organized with separators between functional groups, grouping related operations into distinct categories for easier navigation and quicker access.



- Remap functions shift samples up or down by one semitone or one octave.
- Mirror Velocities and Mirror Notes reverse the order of selected samples across velocity and note ranges.
- Pivot on Corner remaps all selected samples to a single note but different velocity.
- Extend automatically fills empty space between mapped samples.
- Automap functions organize samples automatically based on metadata, pitch detection, or file naming.
- Optimize Mappings ensures even distribution of samples across the key range.
- Auto Find Loop searches for optimal loop points based on defined parameters.

• Quick Set Loop applies loop points using preset parameters.

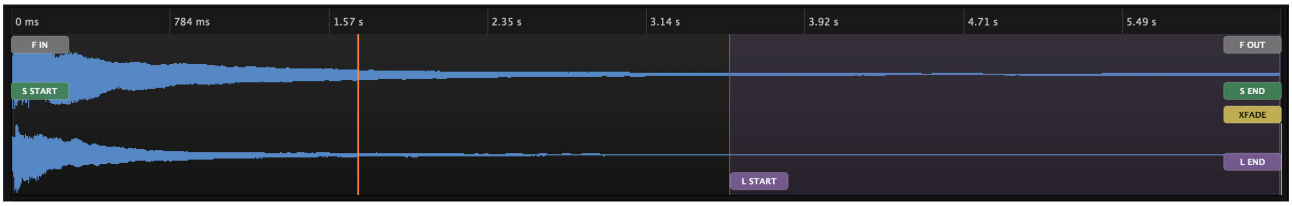
• One Cycle Loop enables creation of very short loops, typically one to ten cycles in length.

• Add Optimized Crossfades calculates appropriate crossfade lengths for each sample.

• Split at Transients divides a single audio file into individual samples based on transient detection.

• Map Splits assigns sliced audio segments across the key range for further editing.

The selected sample can be previewed by pressing the Space bar on the keyboard. Playback starts from the Sample Start position.



The orange playhead begins moving from the start of the sample and will loop indefinitely if a loop is defined.

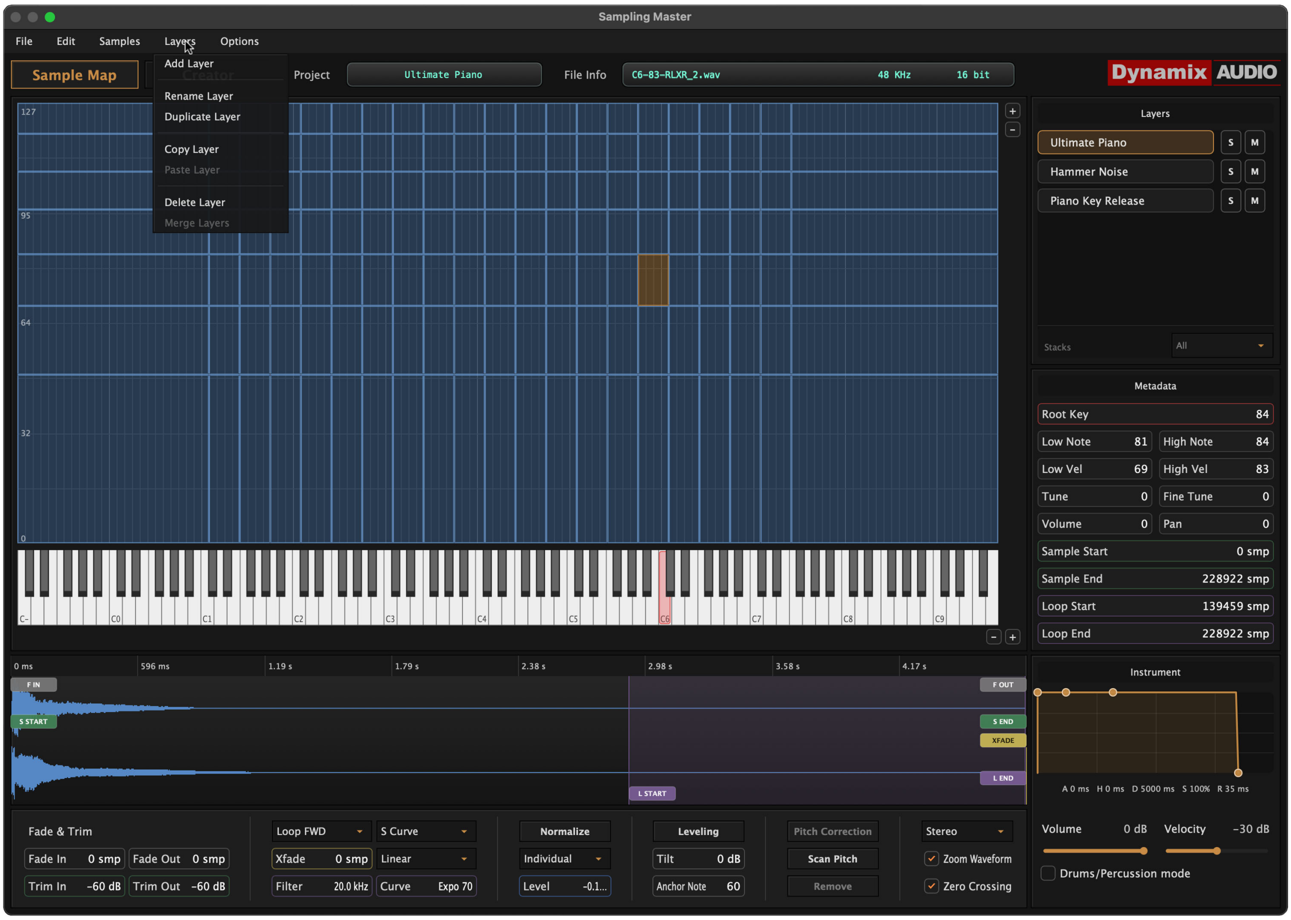
Using Shift + Space starts playback from the Loop Start position and loops the sample indefinitely.

Selecting another sample—either by clicking it in the Mapping panel or using the arrow keys—restarts playback from the previously defined position, either Sample Start or Loop Start.

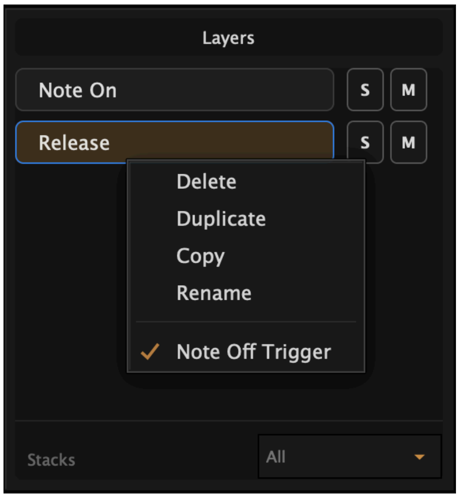
Layers

Sampling Master allows you to create an unlimited number of layers. Each layer contains a single sample map with its associated set of samples.

Layers can be created, duplicated, deleted, or merged either from the menu or via right-clicking the layer.
Two layers can be merged only if their samples do not overlap.



The Layer can be set to trigger on a Note Off MIDI message instead of a Note On. In this mode, the samples within the layer are triggered only when the key is released.
To enable this, right-click the layer and select Note Off Trigger.
Once activated, the layer outline will turn blue.
The layer is automatically set to Note Off Trigger when the Capture Release option is enabled in Creator mode during the sampling process. Release samples are then captured and placed onto a separate layer.



Please note that while previewing Release layers in Sampling Master, you may hear abrupt transitions between layers. This is expected behavior, as the smooth continuation is intended to be handled by the final sampler/player.

File management

Project

Sampling Master projects are saved as .SMP files. The project stores all edits and references to samples in their original locations for later reopening.

Projects can be opened via the File menu or by dragging and dropping the project file directly onto the interface.

All edits within a project are non-destructive. Processing changes are only written into the audio files after rendering.

Import

Sampling Master handles three formats at the time of writing this documentation:

- **SFZ**
- **SF2**
- **MSAMP**

SF2 and SFZ are free and open standards compatible with many software and hardware samplers.

MSAMP is the proprietary sample map format used by the Giglad native sampler.

Sampling Master imports these formats and automatically maps samples based on metadata contained in the sample map.

Export

Saving an SMP project creates only the project file, while exporting to SFZ, SF2, or MSAMP generates the corresponding sample map files and renders the audio samples into a folder alongside the sample map.

The export dialog provides several configuration options.

Some formats, such as SFZ and MSAMP, support FLAC-compressed audio. FLAC is lossless, so audio quality is preserved while file sizes are reduced by approximately two to four times.

Unity Guru SFZ checkbox formats SFZ export for Unity Guru sampler compatibility.
Exporting layers as separate instruments can be useful when working with round-robin layers.

Formats that do not support FLAC, such as SF2, will disable this option in the export dialog when FLAC is selected.

Multiple output formats can be exported simultaneously.

You can choose to preserve source file names; however, it is recommended to rename files during rendering so they include proper metadata-based naming plus a four-character unique suffix. This improves file identification in browsers and prevents filename collisions.

Alternatively, you can export only the samples. In this mode, samples are rendered with all processing and metadata intact for later import and automatic mapping

Loops

Creating a seamless loop within a sample remains one of the most critical—and increasingly overlooked—skills in sampling. Modern software samplers provide effectively unlimited memory, which has led many sample libraries to include full sustain recordings (e.g., a piano note lasting up to 20 seconds or more).

While this approach can enhance realism, it is often impractical in musical contexts, where notes are rarely sustained for such durations. In addition, the perceptual difference between a natural sustain after extended time (e.g., 20 seconds) and a well-crafted loop shaped with AMP and FILTER envelopes is, in many cases, negligible.

Sampling Master provides a set of tools designed to help identify optimal loop points within samples, enabling reduced file sizes and more efficient library management without materially compromising perceived audio quality.

Auto Find Loop

In an ideal scenario, solving complex tasks with a single action is highly desirable. Sampling Master aims to approximate this when identifying loop points within samples.
Auto Find Loop allows you to define key algorithm parameters so the system can locate loop points automatically.

The dialog provides three key parameters that allow you to control the algorithm:

- Search Start % – Defines the approximate starting point of the loop search region.
- Search End % – Defines the approximate ending point of the loop search region.
- Min Length % – Specifies the minimum loop length; this parameter may also be set to AUTO.

The result is that all selected samples will be processed, and the algorithm will attempt to identify the most suitable loop points for each.
While this feature generally produces strong results and significantly speeds up the workflow, it is advisable to review and audition the generated loops. This allows you to identify any suboptimal results and refine them manually where necessary.

This algorithm is particularly effective for material with repeating patterns, such as vibrato, or for sustained sounds that maintain relatively constant amplitude over time—such as woodwinds, brass instruments, and organs. Those sounds do not decay naturally like piano or guitar.

Quick Set Loop

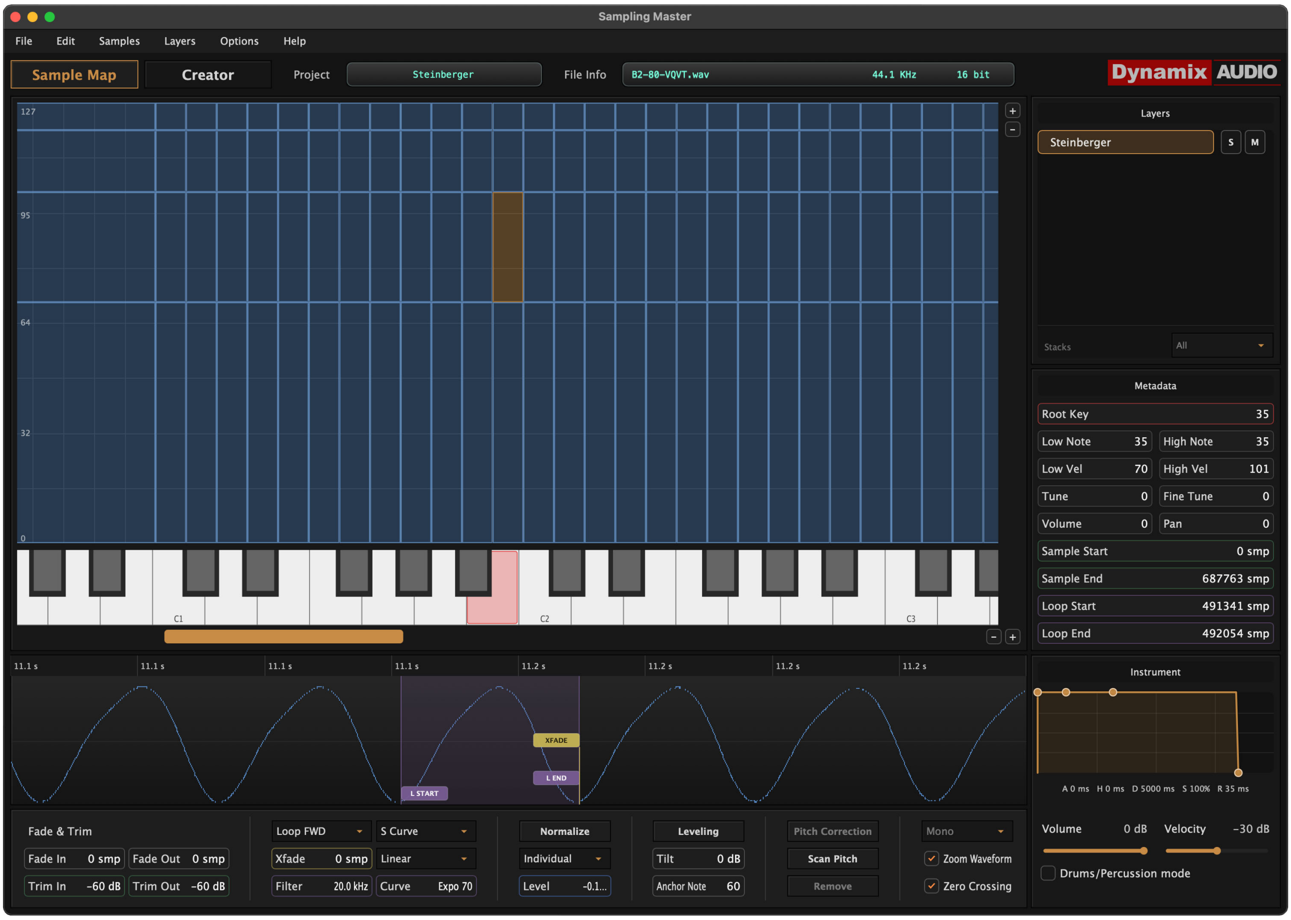
This dialog is similar to the previous one; however, the key difference is that this algorithm does not search for loop points. Instead, it places them precisely at the positions defined by the specified parameters.

This algorithm is particularly effective for sounds with a natural decay in their sustain, such as pianos and guitars, where loop points can be set reliably using this approach.
A practical method is to move the loop region further into the sample—typically around 4 to 5 seconds—until the initial brightness has diminished. At that point, applying a full crossfade can help achieve a smoother and less perceptible loop.

Experimentation with different source material is recommended to determine the most suitable parameter settings. In addition, avoid applying identical settings across the entire key range. Lower notes generally sustain longer, while higher notes decay more quickly; using the same percentage-based loop start for all samples can result in disproportionately short loops in higher registers.

One Cycle Loop

One-cycle loops were the primary looping technique in early ROMplers from the 1980s and 1990s, when memory resources were extremely limited. Minimizing sample length was essential, so loops were placed as early as possible within the waveform.
For example, instruments such as the Korg M1 operated with only a few megabytes of sample memory. This was achievable because the recorded samples were very short—the loop would begin just milliseconds after the initial transient, and the sustained portion was reduced to a single waveform cycle repeated continuously.

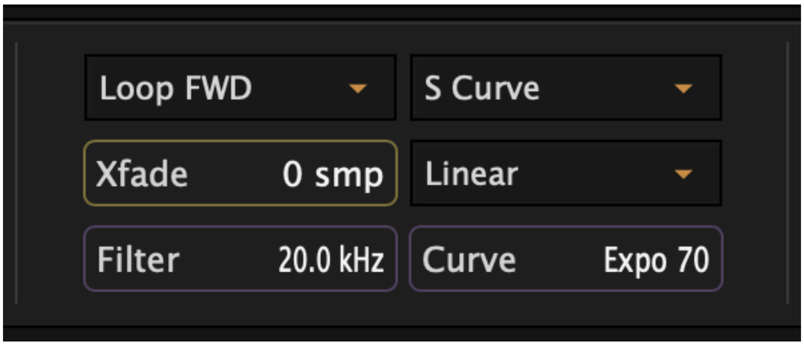


As shown in the screenshot, the loop spans a single cycle of the waveform—one half below the center line and one half above it.

One-cycle loops are often slightly out of tune, which means the target platform must support fine loop tuning to compensate. Sampling Master addresses this limitation by allowing you to define loops of up to 10 cycles. In practice, using 2–3 cycles typically results in a loop that remains very short and efficient, while avoiding tuning issues and preserving a more natural sound.
This is why two menu options are provided: Double Loop Length and Half Loop Length. They allow you to quickly adjust the loop size if you notice tuning instability in a one-cycle loop scenario.

Loop settings allow you to define parameters in a way that produces the best possible loop for a given file.
This enables the creation of different types of loops:

- Loop FWD
- Loop BCK
- Loop Bi-Di



Loop FWD is a standard forward loop. The playhead moves from the loop start point to the loop end, then jumps back to the loop start and continues cycling in the same direction.

Loop BCK is a reverse loop. The playhead moves toward the loop end, then reverses direction back to the loop start and continues looping in reverse.

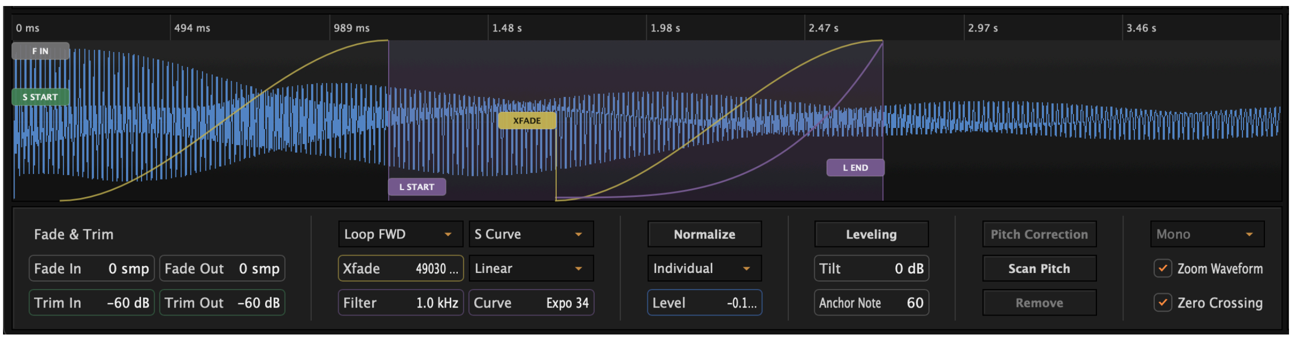
Loop Bi-Di combines both behaviors. The playhead moves from start to end, then reverses back from end to start, alternating direction continuously in a ping-pong manner.

There are three types of the crossfade curves:

- Linear
- S Curve
- Equal Power

Finding the optimal curve depends on the source material. Experimentation is required to determine which option produces the least audible artifacts in the resulting loops.

A filter is also available to reduce artifacts within loops. It attenuates high-frequency content when the loop start position is close to the beginning of the sample. This helps control the initial brightness that can persist into the loop region, since the crossfade may extend back toward the start of the sample.



Mapping

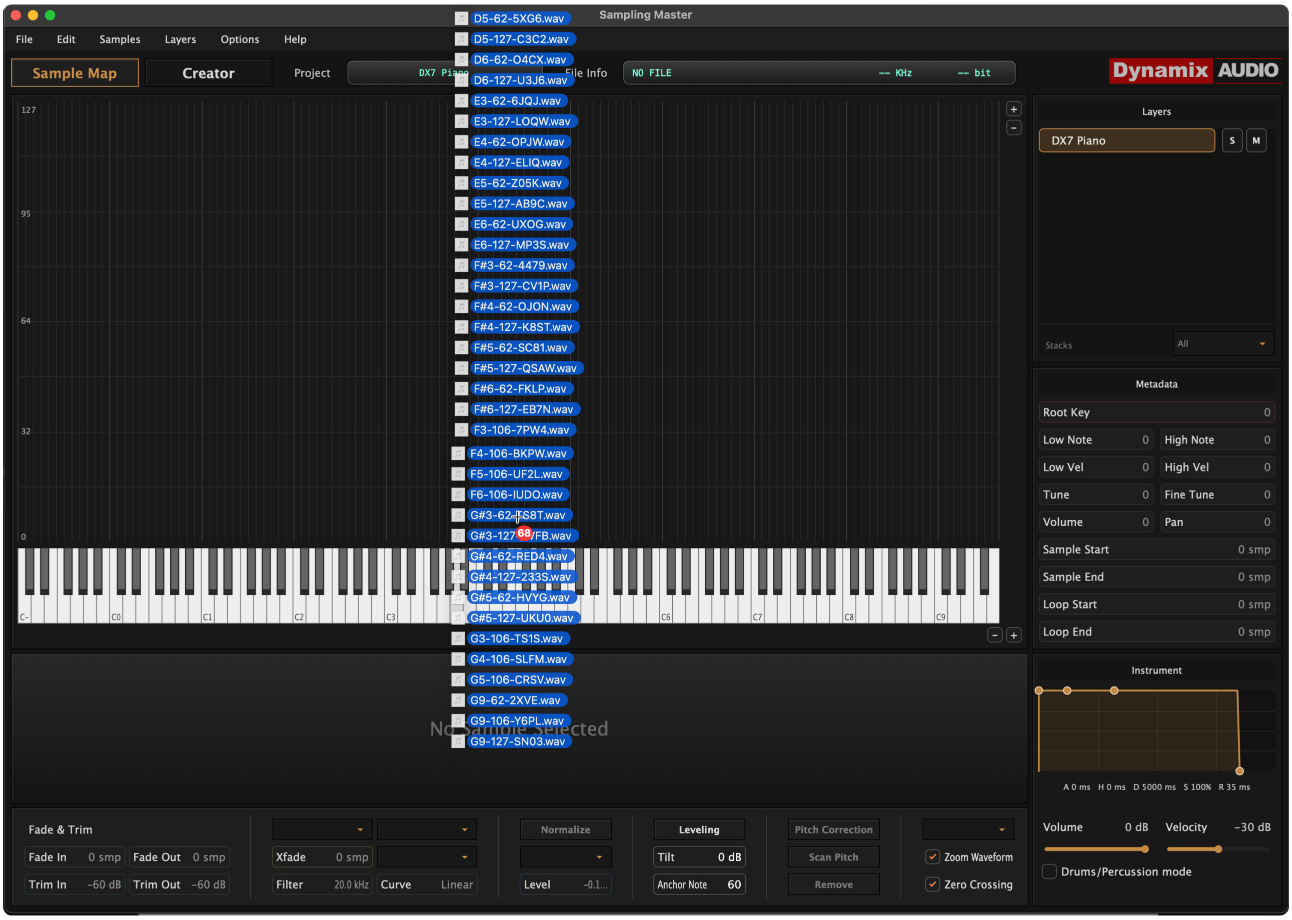
Automap

After looping your samples, correctly and efficiently mapping them is the second major challenge in sampling workflows, particularly when working without dedicated tools. Sampling Master provides solutions for both issues.
Sampling Master can automatically map your samples in three primary ways:

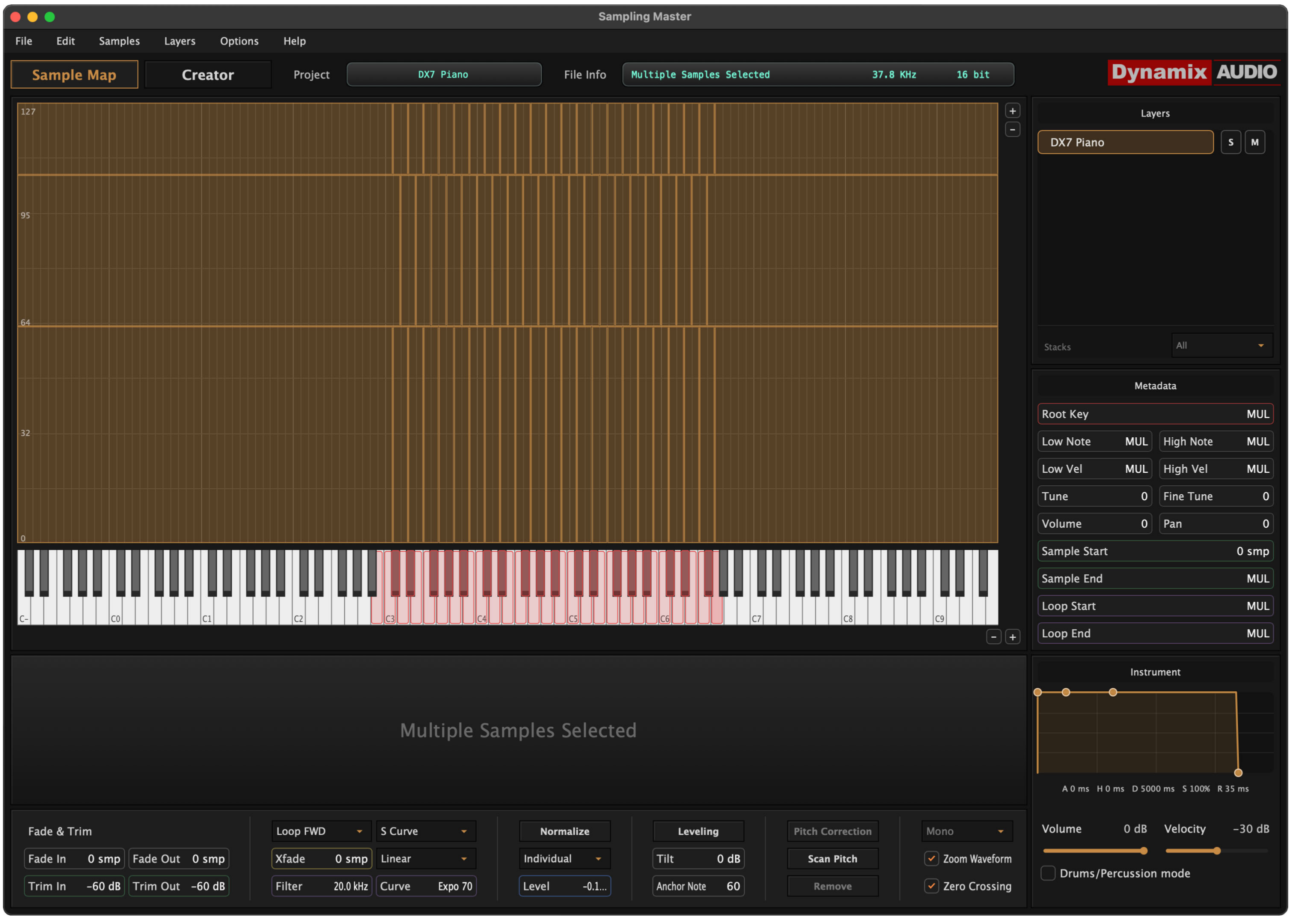
- Automap by Metadata
- Automap by Pitch detection
- Automap by Name

The preferable way is to automap with meatadata, as this is done automatically when you import the files.
If your samples have metadata writen inside, Sampling Master will place them to their correct places both vertically and horizontally.

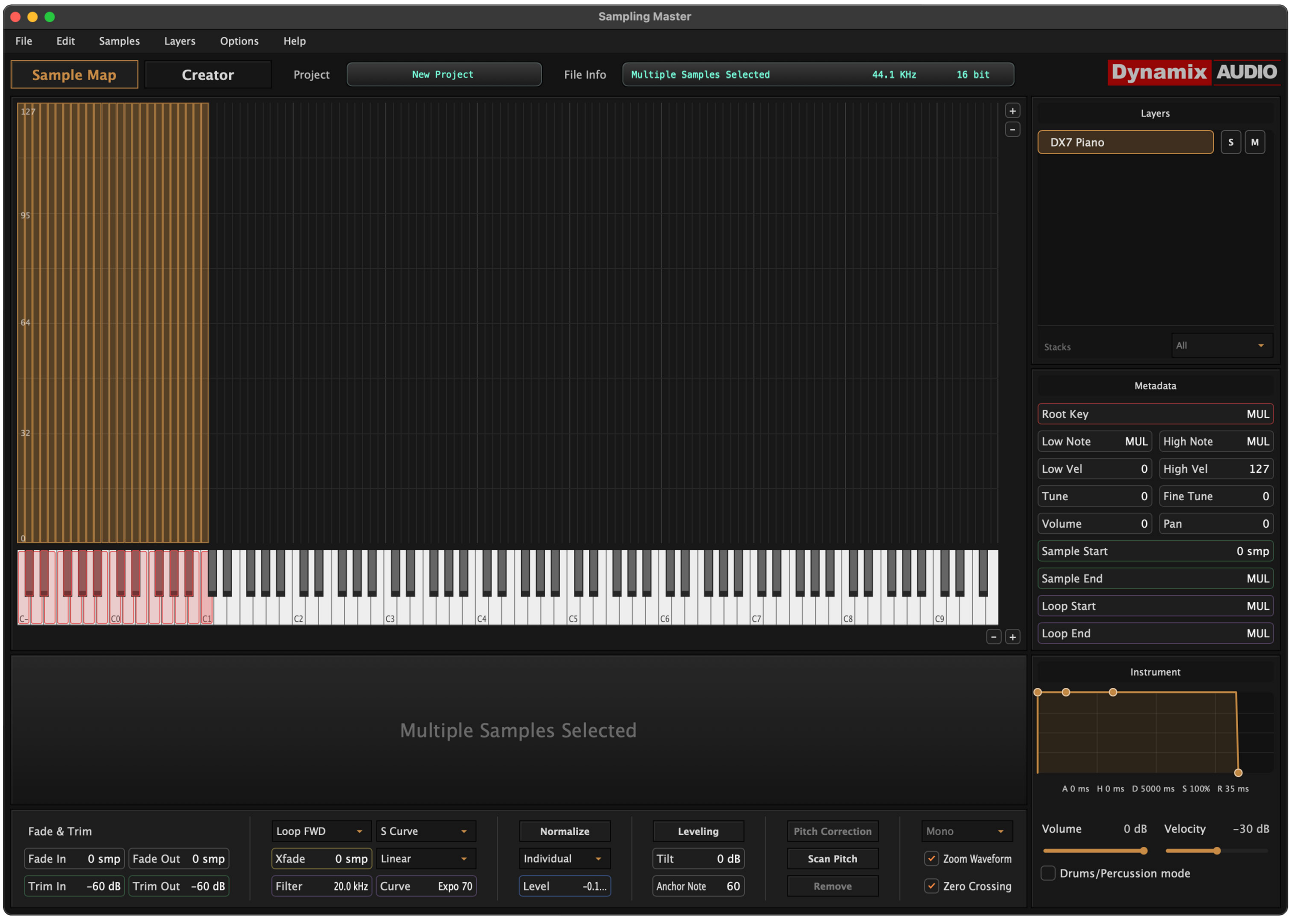
Very few software embed metadata into files; however, Sampling Master addresses this by writing that information into every rendered file, both in the filename and in the embedded metadata. As a result, samples created in Sampling Master can be mapped simply by dragging and dropping them onto the interface.



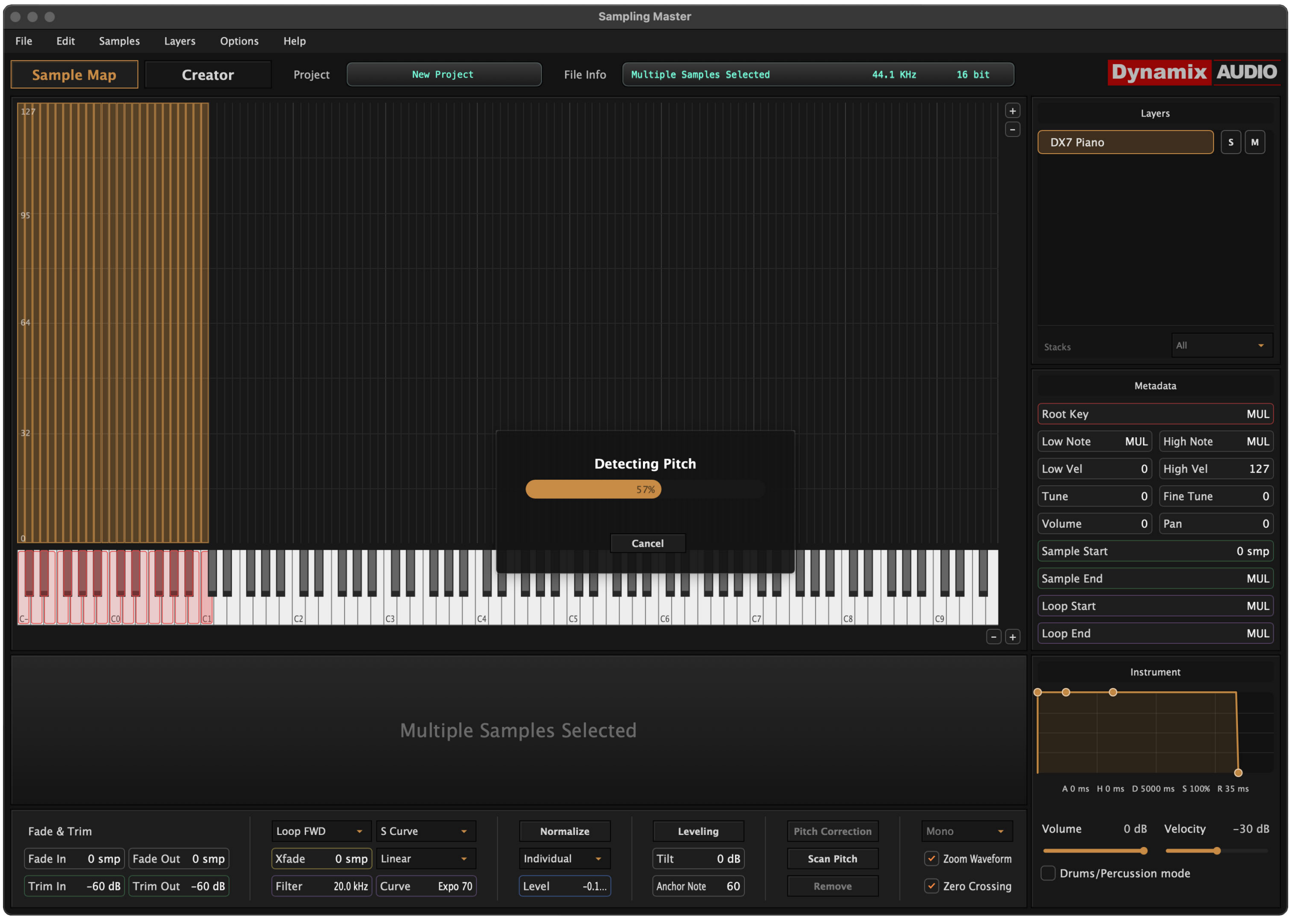
If the files contain metadata, they are mapped instantly, requiring no additional action.



However, if you import files that do not contain metadata, they will be mapped sequentially starting from the beginning of the note range.



In this particular case, you can use the two remaining algorithms: Automap by pitch detection and Automap by name.



Pitch detection generally works well in most situations; however, for sources with strong harmonic content, it may occasionally produce errors. This is typically manifested as a sample being mapped one octave too high or too low.

If this occurs, you can correct it using the available remap functions and their corresponding shortcuts:

- Remap Up by 1 Octave
- Remap Down by 1 Octave

Remap

As mentioned earlier, remapping functions allow you to quickly shift samples one octave up or down with a single button press. There are also options to move samples up or down by one semitone, all of those are straightforward and self-explanatory.

In addition, there are several other remapping functions that can be useful:

- Mirror Velocities
- Mirror Notes
- Pivot On Corner
- Extend
- Extend Vertically
- Extend Horizontaly

Mirror Velocities reverses the order of the selected samples. If velocities are arranged from hardest to softest (top to bottom), applying this function flips them so they run from softest to hardest instead.

The same principle applies to Mirror Notes, which reverses the note mapping so that the highest sample is assigned to the lowest note and vice versa.

Pivot on Corner reassigns sequentially mapped notes so they collapse onto a single note while being distributed across different velocity layers.

The Extend function expands the sample mapping region and fills the available space both horizontally and vertically.
When multiple samples are selected, the function distributes them across the expanded area.

You can choose to extend both directions simultaneously, or restrict the operation to horizontal or vertical expansion only.

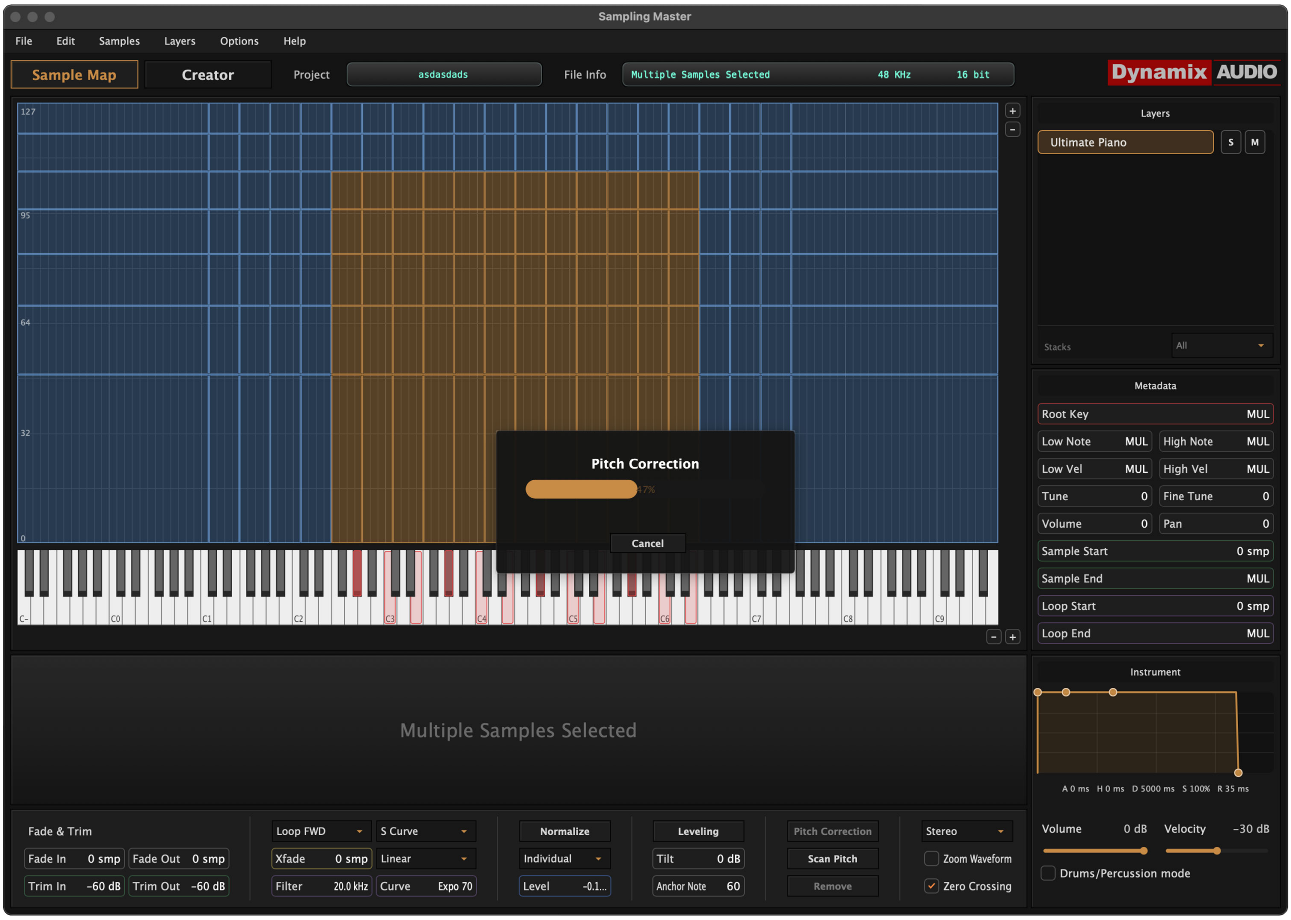
Processing

Pitch Correction

Although the interface is structured so that everything in the Metadata panel is stored only as metadata, while the Processing panel applies changes directly to the rendered files, there is one exception to this rule. One functions that does not strictly follow this rule is:

- Pitch Correction

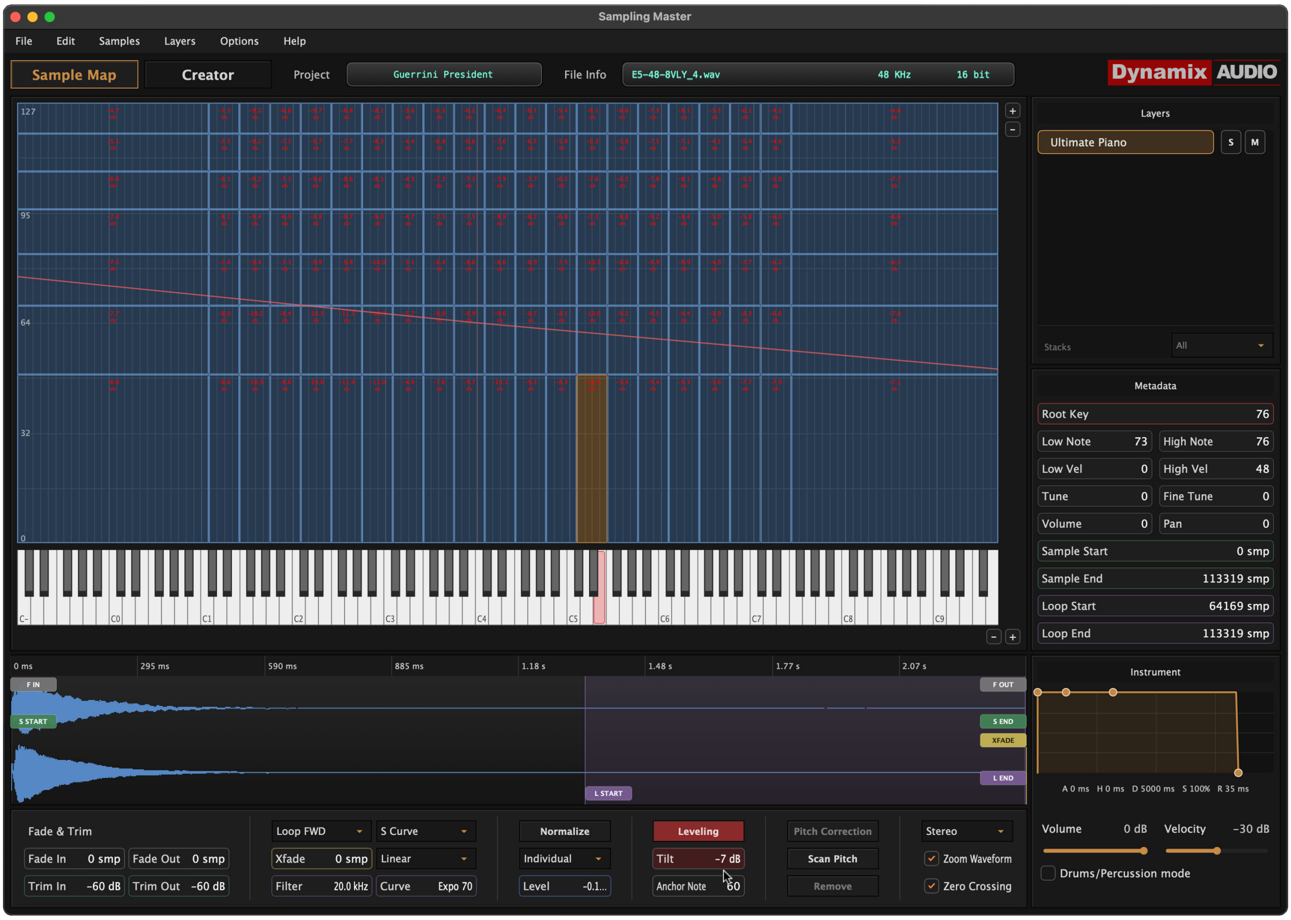
Pitch correction is used when the original material is not perfectly in tune. This is common with real instruments, so you can choose whether to preserve the natural tuning for realism, or correct it by placing each sample precisely in its intended position within the frequency range.
This is done by selecting the samples you want to process and pressing the Search Pitch button.



After detection is complete, the Pitch Correction button will light up in pink, indicating the pitch correction status for each selected sample. As an exception to the usual workflow rules, this correction is not rendered into the audio file itself. Instead, it is stored as metadata, allowing the sampler to apply pitch adjustment dynamically. This makes the process fully reversible.

Leveling

Leveling allows broad adjustments to your sample map, enabling you to reduce the level of higher regions or increase them, depending on the desired balance.
This is particularly useful for achieving realistic results in instruments where energy distribution varies across the keyboard range. A typical example is the accordion, where certain registers exhibit reduced energy in higher octaves. This effect can be lost if samples are normalized individually.
Leveling addresses this by allowing you to reduce the volume of higher octaves. It operates by selecting an anchor note, around which the volume distribution is tilted up or down according to the configured settings.



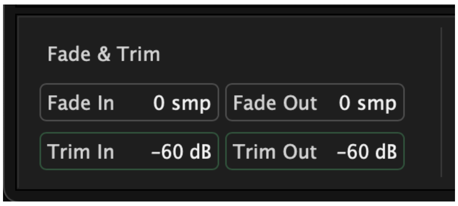
If leveling is applied, the sample regions will display red dB values indicating how much each sample's level has been increased or decreased according to the configured settings.

These adjustments are later written directly into the rendered file, as the audio is processed to reflect the resulting gain changes.

Stereo to Mono

Another important feature in the Processing panel is the ability to convert files from stereo into different mono variants. You can choose to retain only the left channel, only the right channel, or the summed combination of both left and right channels.
It is not possible to reconstruct stereo files from mono sources.

Trimming



Trimming allows you to accurately define sample start and sample end positions by simply adjusting a value.

This function respects the Zero Crossing option; when enabled, trimming will always snap to the nearest zero crossing point, reducing or eliminating clicks.

This is particularly useful for determining optimal start points in your samples. For example, if there is silence at the beginning and you do not want to cut into the initial attack, you can adjust the trim accordingly and the correct start position will be set automatically.

The same applies to the trim-out function. For instance, with a snare drum where you want to remove the inaudible tail, you can set the threshold to around -60 dB, and the system will locate the appropriate end point and place the sample end marker accordingly.

Normalize

Normalizing your samples can be done by selecting them and pressing the Normalize button. This adjusts the volume of the selected samples to the desired level.

Normalization can be performed in two modes: individual or group.
Individual normalization sets the level of each sample independently.

Group normalization takes the highest-level sample in the selected group of samples as a reference and applies a uniform gain adjustment to all samples, preserving their relative balance.

Group normalization is particularly useful when working with instruments that naturally lose energy across higher octaves, preventing unnatural volume inconsistencies across the keyboard range.

Advanced Synthesis

Advanced Synthesis is a feature in Sampling Master that enables the creation of new audio material using a high-quality pitch shifting and time stretching algorithm.
In practical terms, this allows you to generate up to two octaves of additional samples from a single original sample.

Unlike standard sampler processing, Advanced Synthesis preserves:

- the original formants
- the natural timing characteristics (including vibrato)
- the overall tonal character of the source

This results in significantly more natural and realistic sound compared to traditional linear pitch shifting and time stretching commonly used in samplers.
The key reason for this improved quality is processing context. Most samplers operate in real time and must avoid added latency, which limits the complexity and quality of their algorithms. Advanced Synthesis, on the other hand, operates offline, allowing it to use more advanced processing without latency constraints.

Support

If you encounter a problem, please contact Dynamix Audio via email at:

info@dynamix-audio.com

Alternatively, you can use the contact form on the Dynamix Audio website:

https://dynamix-audio.com/contact/