ReSample Manual
v1.0.0

www.2ndsenseaudio.com

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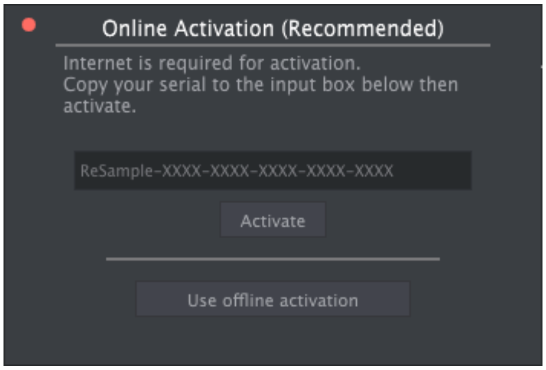
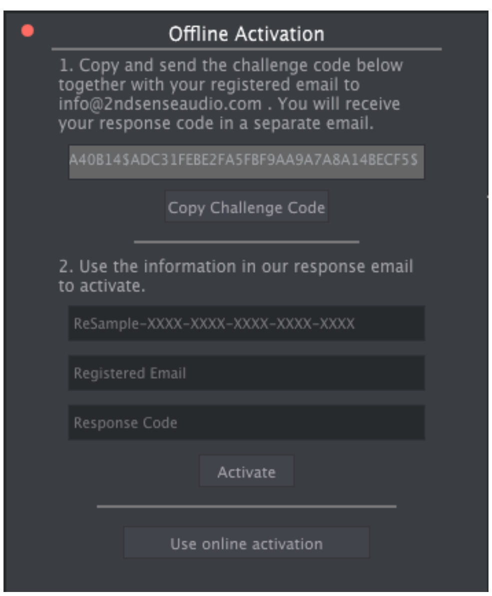
Activation

1. Online Activation



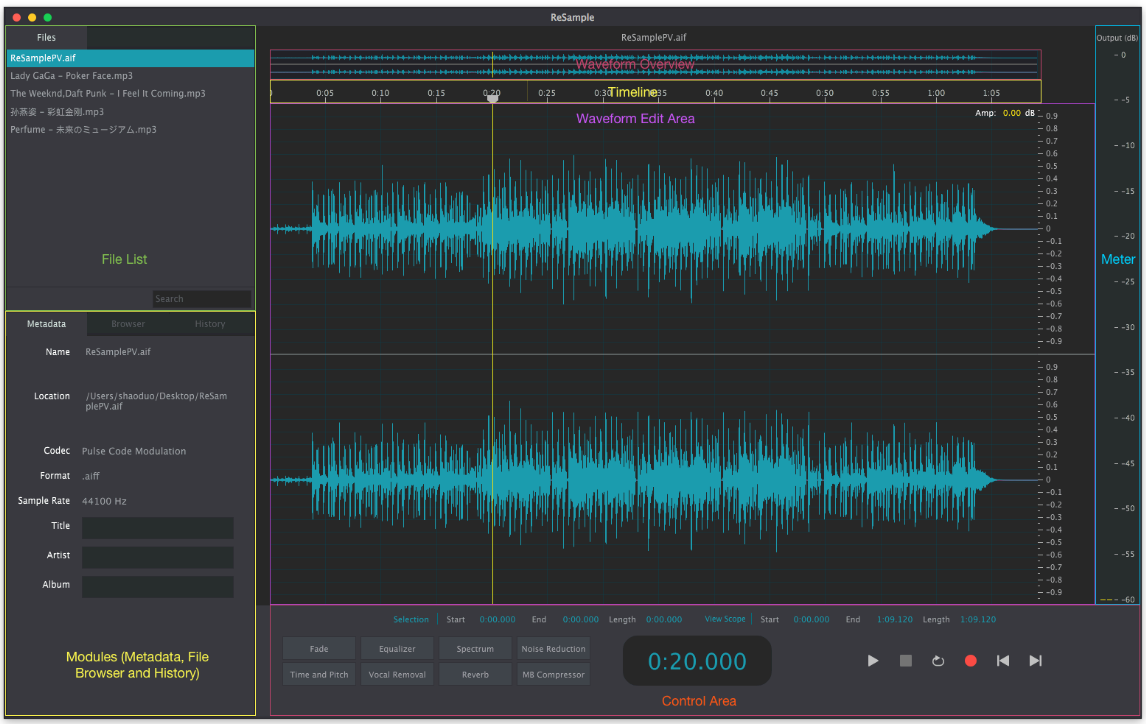
We recommend using online activation since it is easy and fast. Simply copy and paste the Serial number (starting with ReSample-) you received after your purchase. Then press the "Activate" button. The software should respond with a successful activation message. Please make sure your machine is connected to the Internet during activation. (Internet is not required for normal usage of the software after activation.)

2. Offline Activation



If your computer cannot connect to the Internet, we still have a manual offline activation method. Please follow the steps prompted by the software. In short, you will **send us your Challenge code by email**. And we will reply to you within 24 hours with a Response code which will then be used to activate your software.

Main Interface



- File List: The list of files that have been loaded and ready to be browsed and edited.
- Information Module: An area of integrated information, which has file property page, file browser page, and operation history page.
- Entire Waveform Overview: The waveform overview of the currently opened file. The content will not be zoomed and always show the global overview.
- Timeline: the metric of the time information of the file.
- Waveform Edit Area: The most important area of the software. Users browse, zoom, edit and process audio in this area.

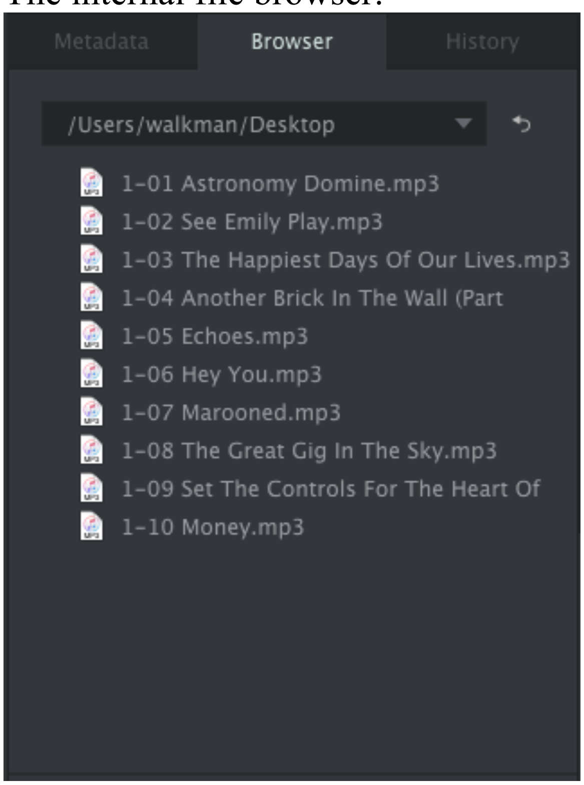
- Control Area: Users can control the playback, quickly open selected effects/processors, and check other detailed information in this area.
- Meter: A true peak meter monitoring the output audio signal.

File Operations

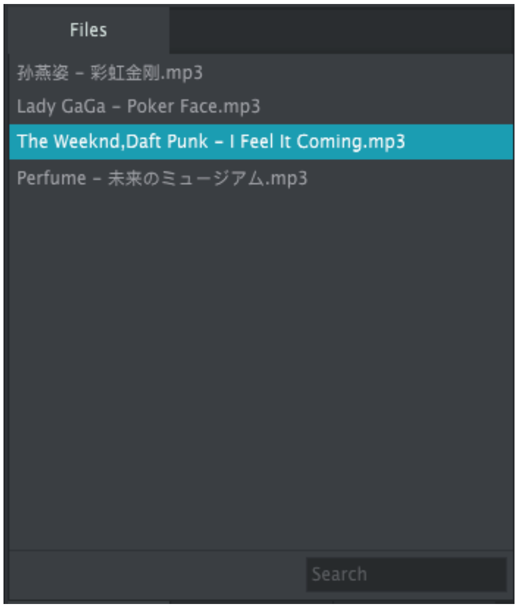
1. Open File

Files can be opened in various ways:
Use menu "File" or shortcut Ctrl + O (Mac: Cmd + O) to open the operating system's file browser window;
Drag the file directly into the software. When dragged into the waveform area, current active file will be switched to the newly opened file. When dragged into the file list area, the new file will be added to the list.
Use the software's internal file browser: double click or drag file to open.

The internal file browser:



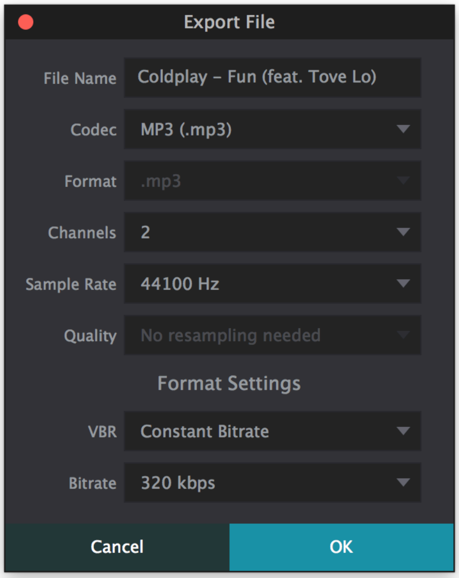
2. Save File



Once the file has been edited or processed, the file name will be marked with *, and the file can be saved.

Using menu "File - Save" (Ctrl + S, Mac: Cmd + S) will save and replace current file.

3. Export File

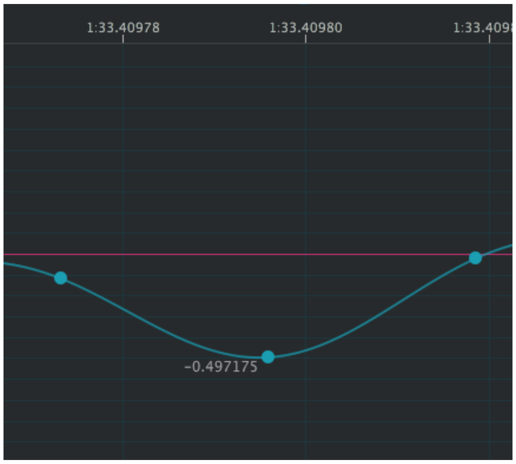


Exporting file can save the file as a new one without replacing the original file. When exporting file, users can select several parameters that will be applied to the exported file, such as sample rate, bit rate, resample quality (when target sample rate is different from the original). Different encoding formats will have different sets of parameters.

View Operations

1. Zoom Waveform

Use shortcut =, -, or mouse scroll wheel or touch pad, when the cursor is focused on the waveform area to zoom. The waveform can be zoomed into sample level. At sample level, users can drag the sample up and down to change the value.



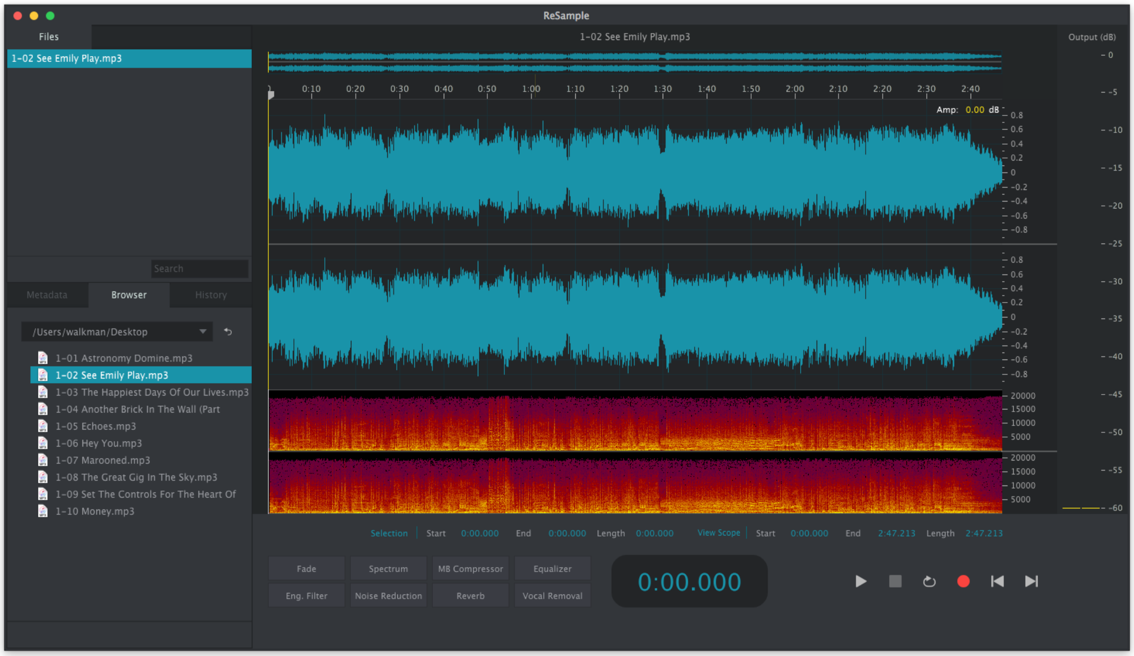
2. Shift Waveform to Left and Right

Drag the rectangle frame in the preview area above the waveform, or use the mouse scroll wheel, the touch pad, or shortcut Ctrl + ←， Ctrl + → (Mac： Cmd + ←， Cmd + →) to shift the waveform to left or right.

Preview area:

3. Show/Hide Spectrogram

Use menu "View - Show/Hide Spectrogram" to switch spectrogram on and off. The spectrogram will be updated according to current waveform.



4. Playback Follow: Turn Page

Use menu "View - Playback Follow: Turn Page" to switch on and off "Turn Page" mode for playback following. When switched on, if the playhead moves out of the waveform area, the waveform will be updated following the playhead.

5. Playback Follow: Scroll

Use menu "View - Playback Follow: Scroll" to switch on and off "Scroll" mode for playback follow. When switched on, if the waveform is zoomed in, the playhead will stay at the center of the area, while the waveform being scrolling during playback.

6. Timeline Unit

Use menu "View - Time Unit", or right click the timeline area, to change the unit of the timeline. "Time" unit shows hour, minute, second. "Samples" unit shows the current number of samples instead of time.

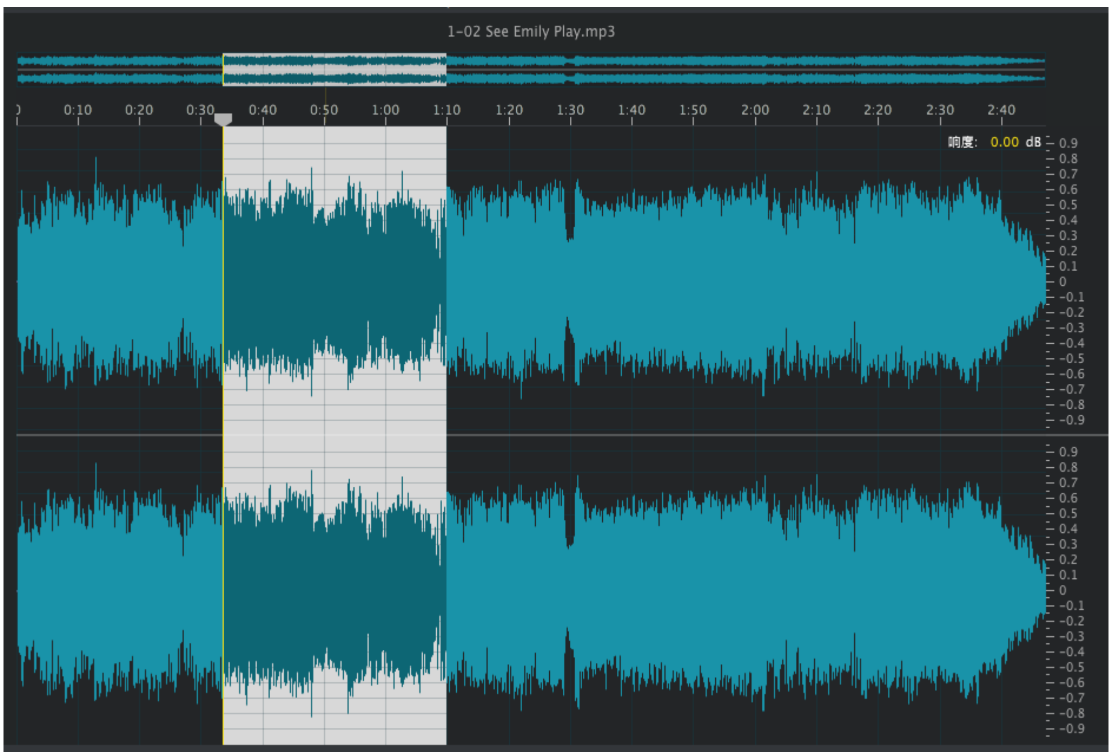
7. Amplitude Meter Unit

Use menu “View - Amplitude Unit”, or right click the amplitude meter area, to change the unit of amplitude. “Normalized” unit shows the actual value of the samples. “Decibels” unit shows the decibel measure of the normalized value of the samples.

Edit Operations

1. Select Section

To select an audio section, drag the cursor along the wanted section in the waveform area. Double click selects the whole audio. Selected section will be shown in inverted color.



2. Delete

First select a section, then right click and choose “Delete” in the pop-up menu, or use shortcut Delete, or menu “Edit - Delete” to delete section.

3. Copy

First select a section, then right click and choose “Copy” in the pop-up menu, or use shortcut Ctrl + C (Mac：Cmd + C), or menu “Edit - Copy” to copy section. If no section has been selected, the whole audio will be copied.

4. Paste

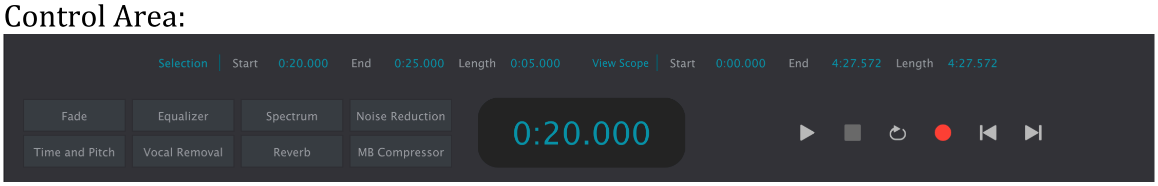
First copy a section, then move the playhead to the place where the section will be pasted, right click and choose “Paste” in the pop-up menu, or use shortcut Ctrl + V (Mac：Cmd + V), or menu “Edit - Paste” to paste section.

5. Cut

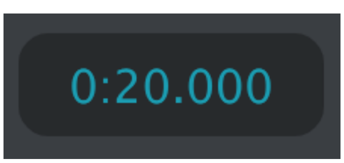
First select a section, then right click and choose “Cut” in the pop-up menu, or shortcut Ctrl + X (Mac：Cmd + X), or menu “Edit - Cut” to cut section.

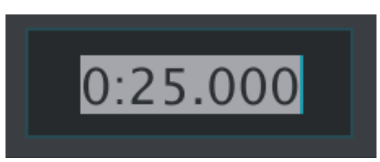
Control Area Operations

Control Area:



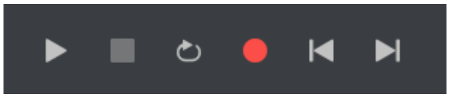
1. Playhead Position





Display current playhead position. Click to set position manually.

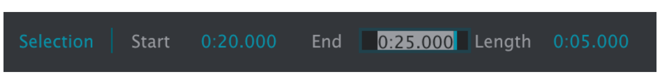
2. Playback Control



- Play: Play current file.
- Stop: Stop playing current file.
- Loop: loop the selected section.
- Record: start recording and insert into the playhead position.
- Move backward: move playhead backward (to the beginning of the selected section or entire file)
- Move forward: move playhead forward (to the end of the selected section or entire file)

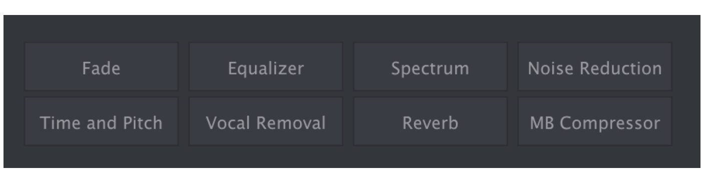
3. Selection Area and View Area Information

Show the information of the selected section and view area. As shown above, the audio/waveform between 0:20 and 0:25 has been selected; Current view area is displaying the waveform between 0:00 and 4:27.572.



Click the numbers to set the time manually. The waveform in the view area will be updated accordingly.

4. Selected Processors Panel

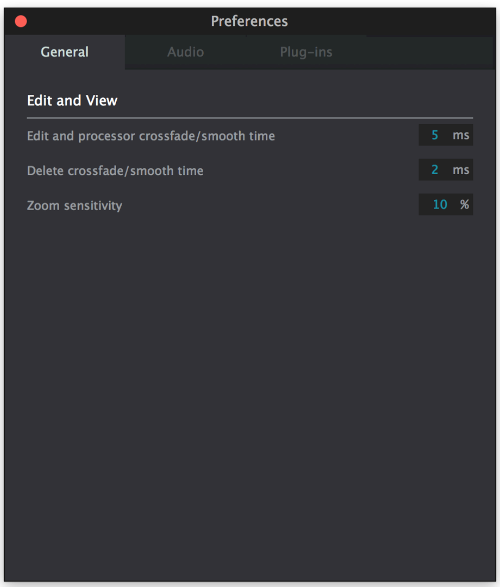


Click the buttons in this panel to quickly open selected processors.

Preference Settings

Using menu “ReSample - Preferences” or shortcut Ctrl + , (Mac: Cmd + ,) to open preferences panel. Preferences panel consists of General page, Audio page, and Plug-ins page.

1. General



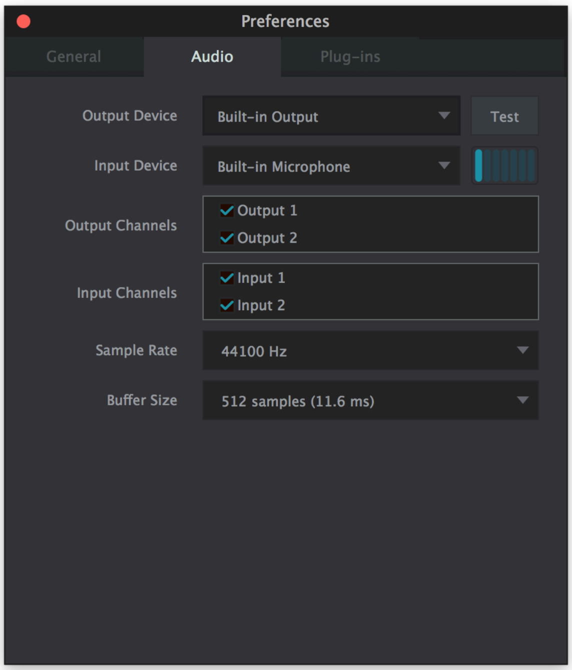
In this panel users can change edit and view settings.

Edit and processor crossfade/smooth time: When doing editing operations, sometimes new audio section needs to be concatenated with current section: For example, when copying and pasting a new audio section into current section, there might be an impulse signal at the crossing of the two sections, which will cause a click sound perceptually. A few milliseconds of crossfading/smoothing could overcome this problem. The default setting is 5ms, which means 5ms of crossfading at the crossing of the two sections.

Delete crossfade/smooth time: Similarly, this process is used for the same purpose as above, but applied to delete and cut operations.

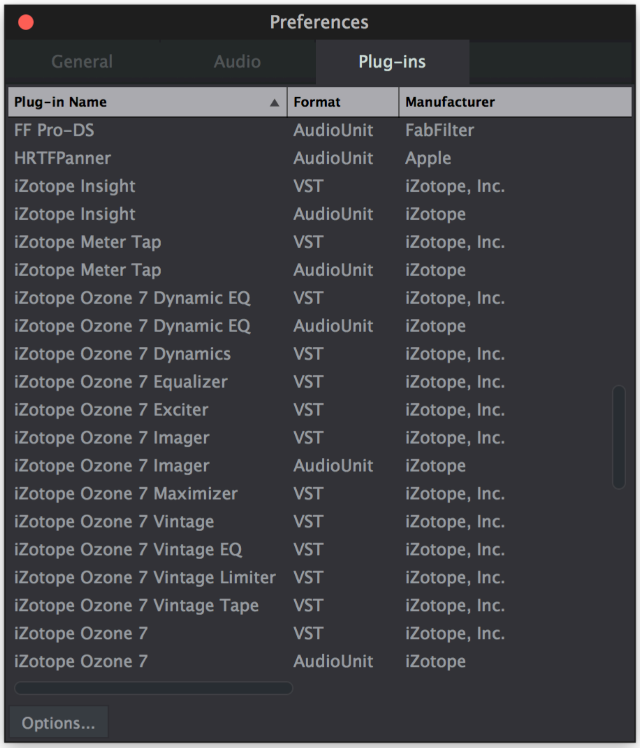
Zoom sensitivity: This sets how much the waveform would be zoomed for each zooming operation: For example, 10% means one zooming operation (pressing the shortcut once or moving the scrolling wheel with one minimum step) will zoom in or out 10% of current length.

2. Audio



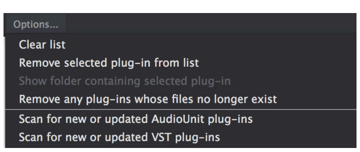
In this panel users can set input and output devices (sound card), input and output channels, sample rate, and buffer size. In ReSample, the sample rate of input and output device must be the same.

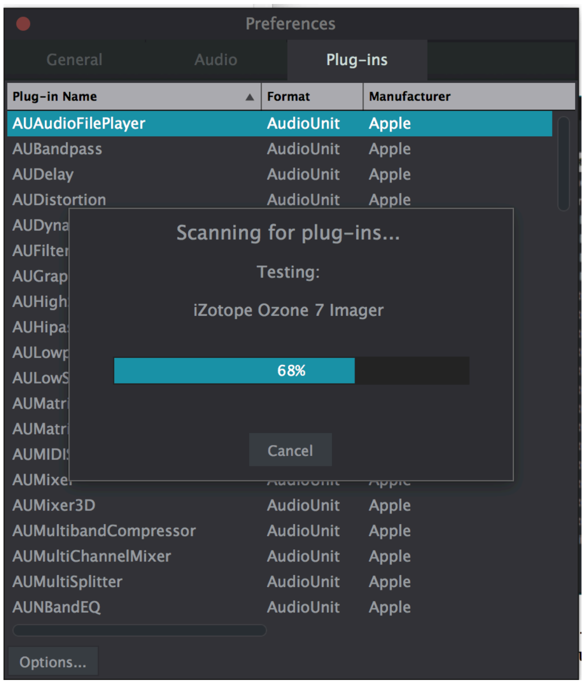
3. Plug-ins



ReSample is able to run third-party plug-ins of open frameworks (VST and AudioUnit). To use such plug-ins, users need to scan installed plug-ins first.

When scanning for the first time, use the scan button in the menu to start scanning.





In the scanning process, due to possible crash of some third-party plug-ins (which might not be quite robust), ReSample will inform the crash and users can restart the scanning process to continue.

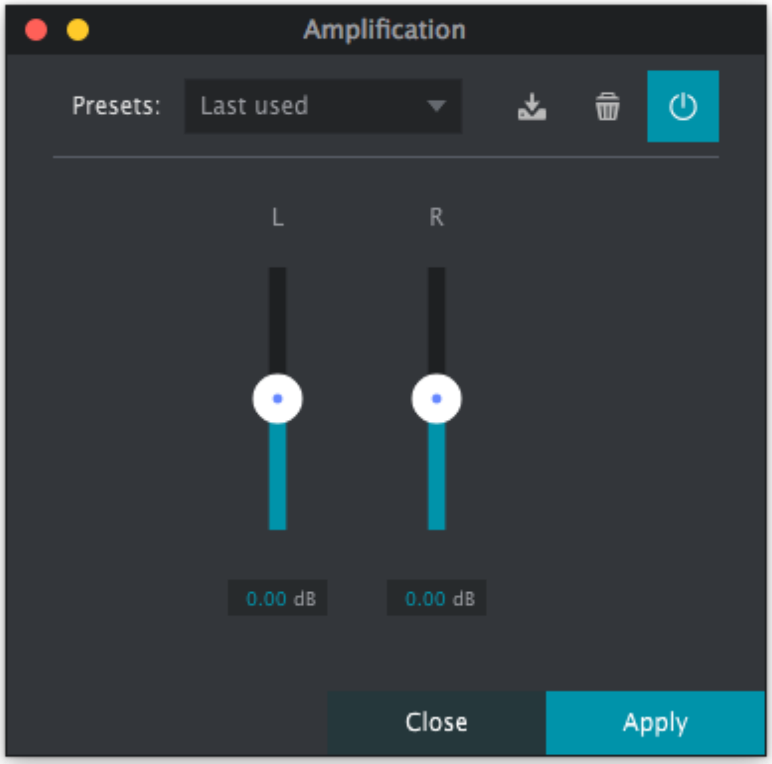
Once scanned successfully, the plug-ins will show in the “Processors” menu.

Processor (Effect)

“Processor” is a collection of tools for audio processing and analysis. These tools utilize DSP (Digital Signal Processing) technologies to process audio to generate a wide range of sound effects, or to analyze audio for various practical purposes. When the processor is on, the output of playback is the live preview of the processed audio, while original audio being intact. To apply the processing to the original audio, click “Apply” button in the processor to apply offline processing. (The live preview during playback is aimed for tuning processor parameters. Its sound quality might not be as good as the final offline result in some cases.)

1. Amplification

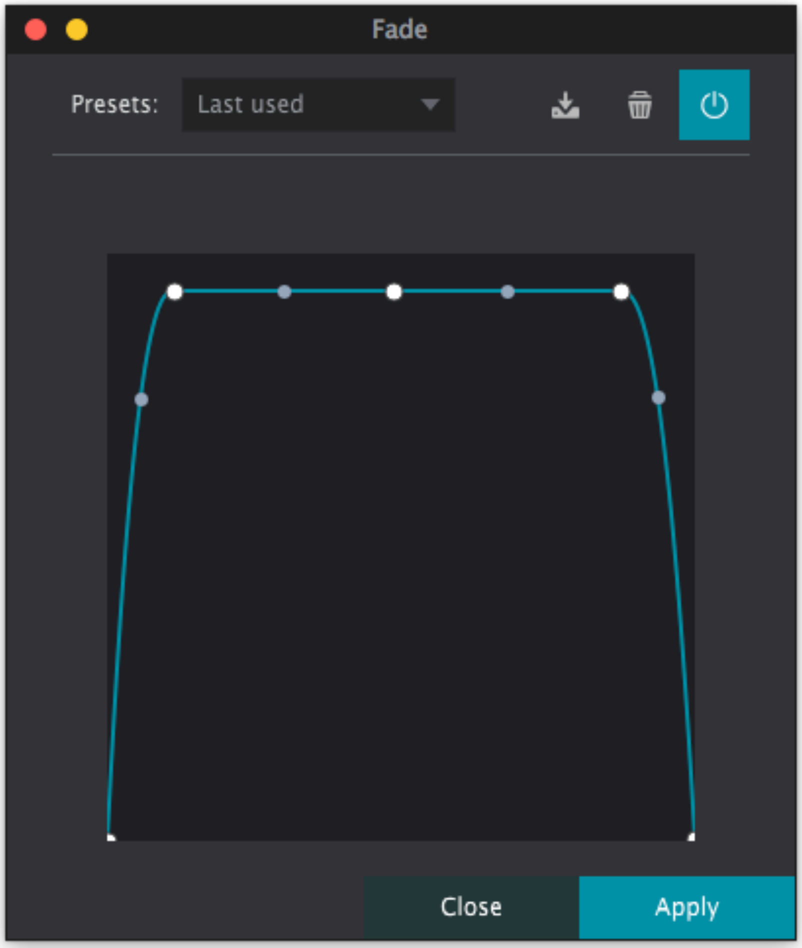
Amplification is used to amplify or attenuate audio signal of each channel in real time.



- *Amplification Slider* – adjust the loudness of each channel.

2. Fade

Fade is used to adjust the volume of audio according to an arbitrary envelope along time. A section of audio needs to be selected first, which the fade envelope will be applied to.



- *Fade Curve* – Control the volume change over time. Click the effective area to add drag point. Change the fade envelope by dragging the curve point.

3. Compressor

Compressor adjusts the dynamic range in real time, which would make the level of audio to stay in a stable range.

- *Scrollscope Button* –Show Scrollscope and compressor gain curve in real time. The gain curve indicates how much the audio is being compressed. The lower the curve, more compressed the audio gets.
- *Spectrum Button* – Show real time spectrum.
- *Attack Number Slider* – Adjust the attack time when the level of audio surpasses threshold.
- *Release Number Slider* – Adjust the release time when the level of audio drops below threshold.
- *Threshold Knob* – Adjust the threshold of the compressor. When the level of audio surpasses the threshold, the compressor begins to compress. When the level of audio drops below the threshold, the compressor stops to compress.
- *Ratio Knob* – Adjust the compressing ratio. The bigger the ratio, more compressed the audio gets, and the smaller the output level.
- *Gain Knob* – Adjust output gain.

4. MultiBand Compressor

MultiBand Compressor is similar to Compressor. But the whole frequency range is divided into 4 bands. Each band is assigned to an independent compressor.

- *Scrollscope Button* –Show Scrollscope and compressor gain curve in real time. The gain curve indicates how much the audio is being compressed. The lower the curve, more compressed the audio gets.
- *Spectrum Button* – Show real time spectrum.
- *Attack Number Slider* – Adjust the attack time when the level of audio surpasses threshold.
- *Release Number Slider* – Adjust the release time when the level of audio drops below threshold.
- *Threshold Knob* – Adjust the threshold of the compressor. When the level of audio surpasses the threshold, the compressor begins to compress. When the level of audio drops below the threshold, the compressor stops to compress.
- *Ratio Knob* – Adjust the compressing ratio. The bigger the ratio, more compressed the audio gets, and the smaller the output level.
- *Gain Knob* – Adjust output gain.
- *Crossover Number Slider* – Adjust the crossover cutoff of adjacent

bands (filters). Therefore the frequency range of the 4 bands are: [0, cutoff1], [cutoff1, cutoff2], [cutoff2, cutoff3], [cutoff3, upper limit (20k)](Hz).

5. Equalizer

Equalizer adjusts the gain of the audio in different frequency ranges. It is mainly comprised of 9 filters, resulting into 9 frequency bands. Each column of control knobs is assigned to one filter. From left to right they are: low-cut filter, low-shelf filter, 5 peak filters, high-shelf filter, and high-cut filter. The integrated frequency response of all the filters is shown as the red line. Each filter has a drag point, which can be used to adjust the cutoff frequency and gain of that filter, whose coordinate is defined by these two parameters.

- *Show Spectrum Button* – Show real time spectrum.
- *Output Gain Slider* – Adjust output gain.
- *Drag Points* – Adjust the cutoff and gain of assigned filter, corresponding to its coordinate.

- *Cutoff Knob* – Adjust the cutoff of assigned filter.
- *Q/Bandwidth Know* – Adjust Q/Bandwidth of assigned filter.
- *Gain/Order Knob* – For high-cut and low-cut filters, adjust order. For other filters, adjust gain.
- *Switch Button* – Choose whether to bypass specific filter.

6. Engineering Filter

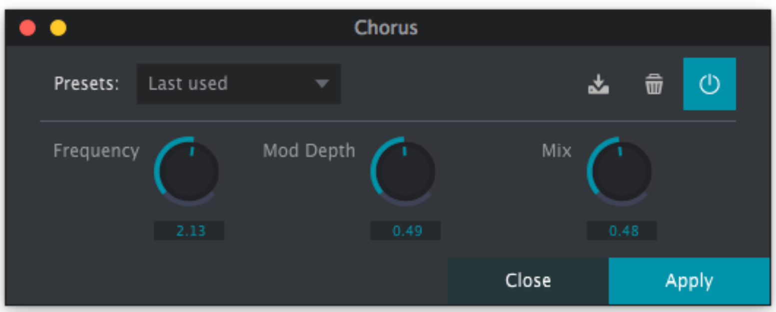
Engineering Filter provides a variety of filters that are commonly used in a wide range of tasks, which can be used for create different audio effects. The frequency and phase response are shown as red lines, which indicate the characteristics of current filter. Real time spectrum shows the effects the filter has on current audio.

- *Frequency Response Button* – Show frequency response.
- *Phase Response Button* – Show phase response.
- *Output Gain Slider* –Adjust output gain.
- *Filter Family Box* –Select filter family.
- *Filter Type* – Select filter type.

- *Order Number Slider* – Adjust order of filter.
- *Transition Width Number Slider* – Adjust transition width (which is related to order. Only one of them can be adjusted at a time, where the other one will be changed accordingly).
- *Pass Ripple Number Slider* – Adjust pass ripple.
- *Stop Ripple Number Slider* – Adjust stop ripple.
- *Cutoff Knob* – Adjust cutoff. Band-pass or band-stop filters will have 2 cutoffs referring to the upper and lower one.
- *Show Spectrum Button* – Choose whether to show real time spectrum.
- *Disable Realtime Cutoff Change Button* – Choose whether the disable real time cutoff change when the audio is playing. If the order of filter is high, real time change of cutoff might cause the audio to be distorted. Thus it is recommended to disable real time cutoff change when order is over 16.

7. Chorus

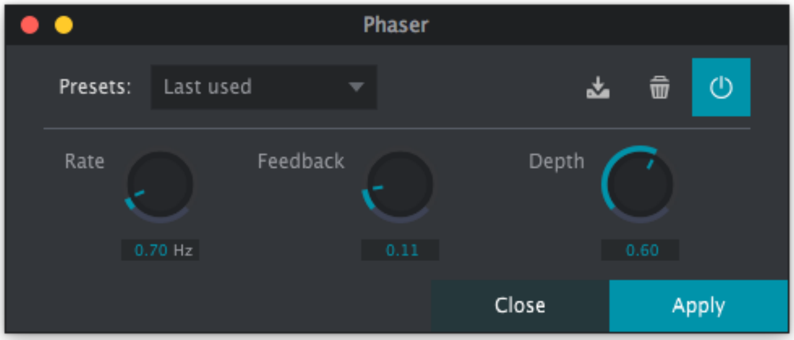
Chorus modifies the timing and frequency of the audio to generate variations of original sound to create a richer sound. It is commonly used for human voice.



- *Frequency Knob* – Adjust the amount of frequency variation.
- *Mod Depth Knob* – Adjust mod depth, which is related to the richness of sound.
- *Mix Knob* – Adjust the ratio between wet and dry signal.

8. Phaser

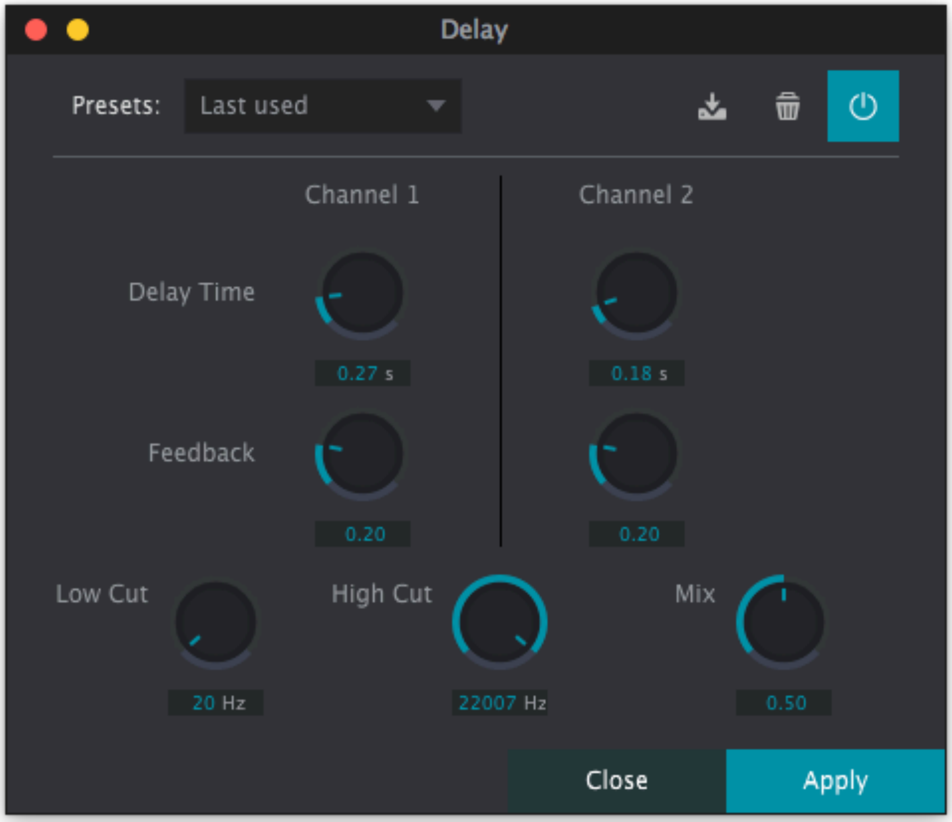
Phaser changes the phase of audio. Combining phase processed audio and the original can create special sound effect.



- *Rate Knob* –Adjust modulation rate, which affects how fast the phase is changed.
- *Feedback Knob* – Adjust feedback.
- *Depth Knob* – Adjust modulation depth.

9. Delay

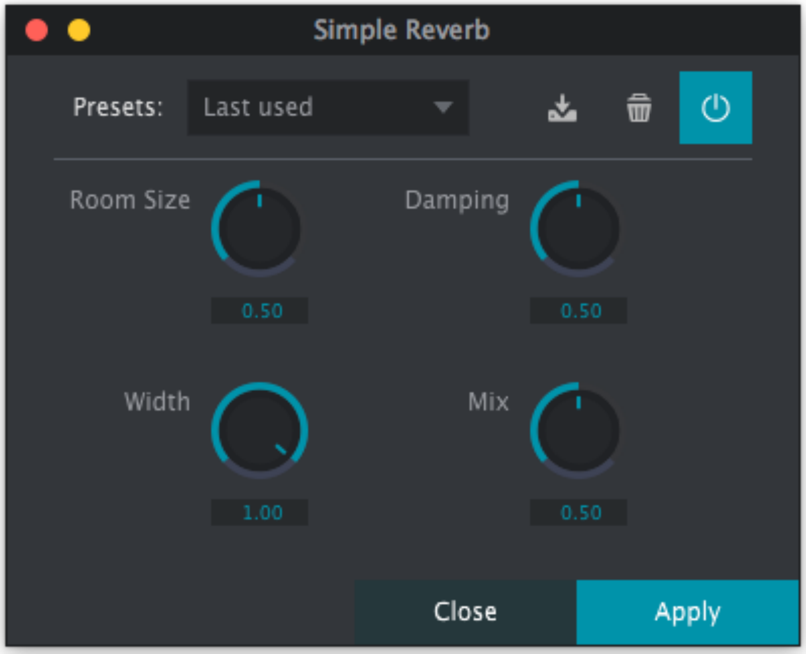
Delay adds delay to arbitrary channel. Though it seems to be a simple process, with which a variety of highly demanded effects can be achieved, such as chorus, echo, localization and so on. Each channel has its on set of control knobs. If the there are more than 2 channels, there will be a combo box to select the channel that the knobs are assigned to.



- *Delay Time Knob* – Adjust delay time.
- *Feedback Knob* – Adjust feedback.
- *Low Cut Knob* – Adjust cutoff of the low cut filter, applied to all channels.
- *High Cut Knob* – Adjust cutoff of the high cut filter, applied to all channels.
- *Mix Knob* – Adjust the ratio between wet and dry signal.

10. Simple Reverb

Reverb simulates the reflections by walls and obstacles in real life, making the binaural impression that the sound is located in a room environment.



- *Room Size Knob* – Adjust room size.
- *Damping Knob* – Adjust damping.
- *Width Knob* – Adjust room width.
- *Mix Knob* – Adjust the ratio between wet and dry signal.

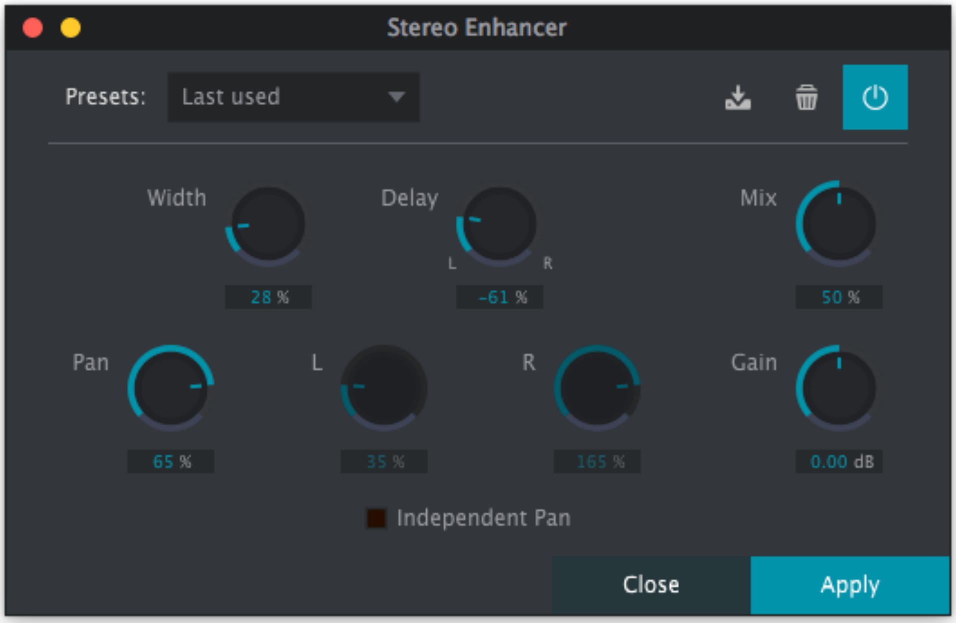
11. Reverb

Reverb is similar to Simple Reverb, but has more parameters and better quality.

- *Predelay Knob* – Adjust predelay before the reverberation process.
- *Early Ref. Knob* – Adjust the amount the early reflection.
- *Damping Knob* – Adjust damping.
- *Decay* – Adjust decay.
- *Size Knob* – Adjust room size.
- *Mix Knob* – Adjust the ratio between wet and dry signal.
- *Gain Knob* – Adjust output gain.
- *Advanced Button* – Choose whether to show advanced options.
- *High Cut Knob* – Adjust cutoff of the high cut filter.
- *Density Knob* – Adjust density.
- *Mod Depth Knob* – Adjust modulation depth.
- *Mod Rate Knob* – Adjust modulation rate.
- *Diffusser 1 Knob* – Adjust feedback of diffuser 1.
- *Diffusser 2 Knob* – Adjust feedback of diffuser 2.

12. Stereo Enhancer

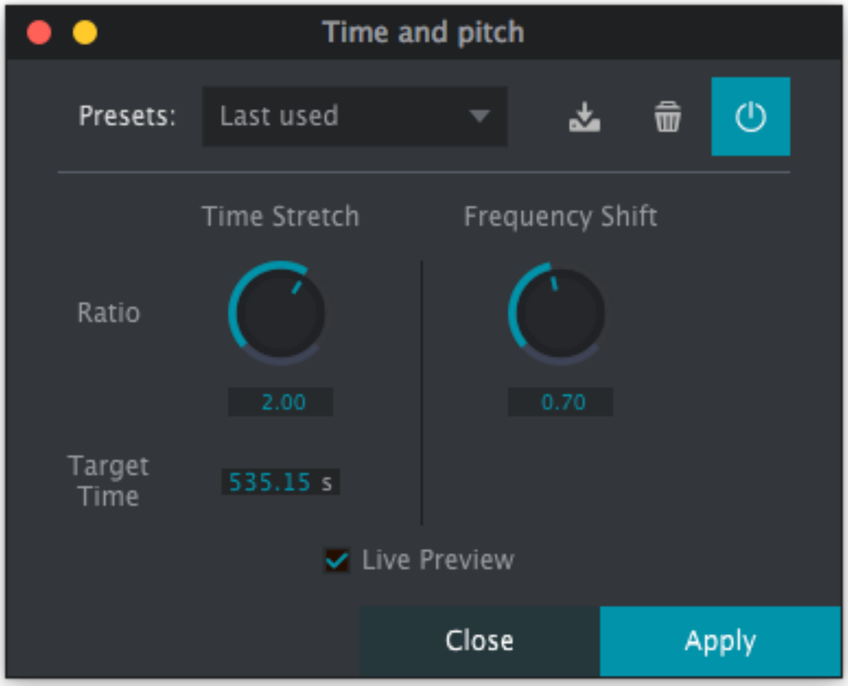
Stereo Enhancer adds variation to left and right channel to enhance the stereo image.



- *Width Knob* – Adjust width of stereo image.
- *Delay Knob* – Adjust the relative delay between left and right channel (only one channel is delayed).
- *Pan Knob* – Adjust panning.
- *Independent Pan Button* – Choose whether to adjust level of each channel independently.
- *L Knob* – Adjust the level of left channel.
- *R Knob* – Adjust the level of right channel.
- *Mix Knob* – Adjust the ratio between wet and dry signal.
- *Gain Knob* – Adjust output gain.

13. Time and Pitch

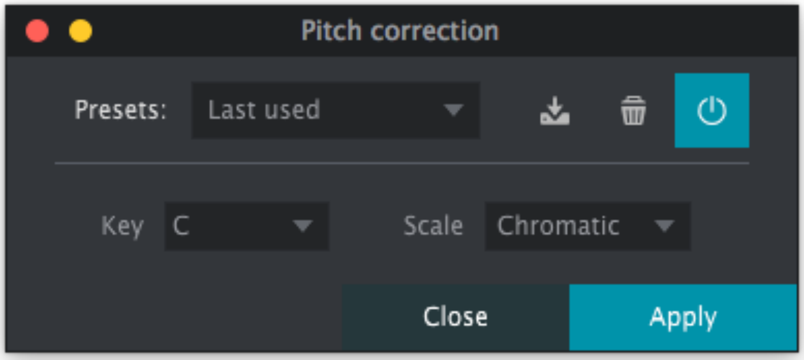
Time and Pitch provides time stretch and frequency shift using a phase vocoder. Time stretch and pitch shift can be done independently, which means time stretch will not change pitch of audio, while pitch shift won't affect timing of audio.



- *Ratio (Time Stretch) Knob* – Adjust the ratio between stretched and original audio. Target time will be calculated automatically.
- *Target Time Number Slider* – Adjust the time length of target audio. Ratio (Time Stretch) will be calculated automatically.
- *Ratio (Frequency Shift) Knob* – Adjust the ratio between shifted and original audio.
- *Live Preview Button* – Choose whether to apply real time processing when playing audio. Live preview quality is not as good as the offline process (click apply button). It is use to preview the effect and help tuning the parameters.

14. Pitch Correction

Pitch Correction detects the pitch of sound and maps it to the closet note in selected scale. It can be used to correct out-of-tune notes, or other special effects. **This processor is currently in experimental phase. The result is not optimal.**



- *Key Box* – Select key.
- *Scale Box* – Select scale.

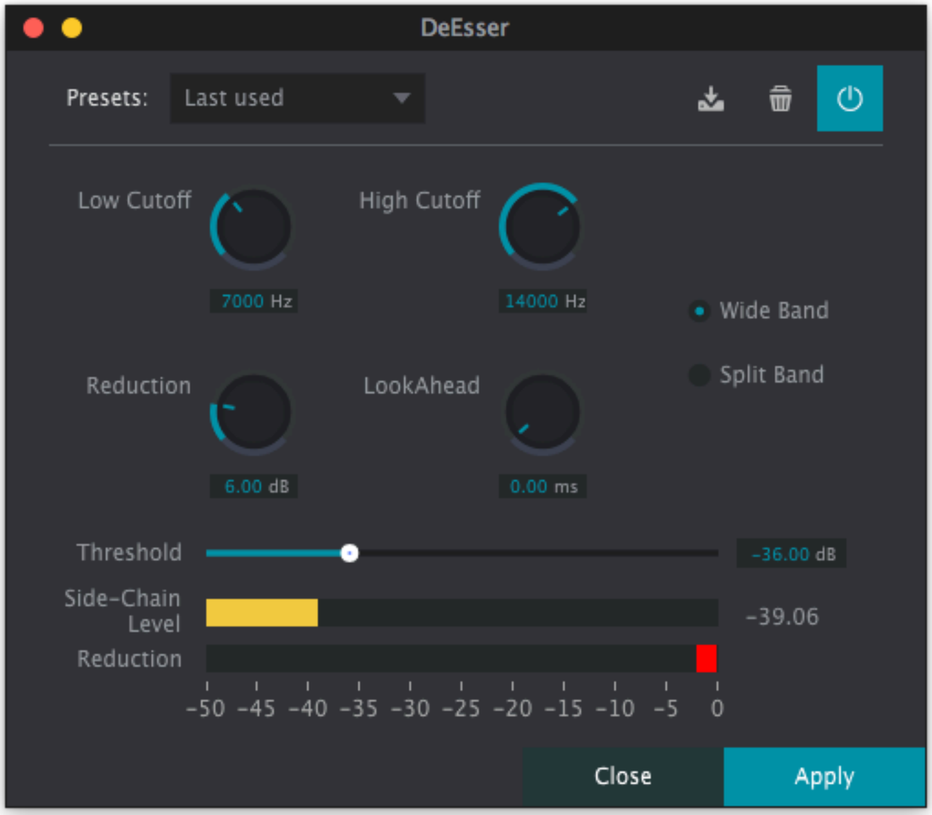
15. Noise Reduction

Noise Reduction analyzes the noise to generate a noise profile, which is used to reduce similar noise from the audio. First, select a section of audio (ideally pure noise) and click "Estimate Noise Profile", then the section of audio will be analyzed to generate noise profile. The spectrum of the resulted noise profile will be shown. Then select another section of audio that has noise and click "Apply". The similar noise in the section will be reduced.

- *Channel Box* – Select channel. Since the audio content of each channel might not be the same, distinct noise profile can be generated for each channel. The parameters can be adjusted independently.
- *Profile Tuning Slider* – Tune the noise profile between indicated range.
- *Subtract Strength Slider* – Adjust how much the noise is subtracted from audio. The higher the value, the more the noise will be subtracted, meanwhile the original audio will be affected more.
- *Wiener Gain Slider* – Adjust gain of the wiener filter.
- *Spectral Decay Slider* – Adjust spectral decay.
- *Advanced Button* – Choose whether to show advanced options.
- *Apply to all channels Button* – Choose whether to apply change of current parameter to all channels.
- *Smoothing Slider* – Adjust smoothing.
- *Filter Width Slider* – Adjust filter width.
- *Filter Factor Slider* – Adjust filter factor.
- *Subtract Iterations Slider* – Adjust how many times the noise profile will be subtracted from original audio.

16. DeEsser

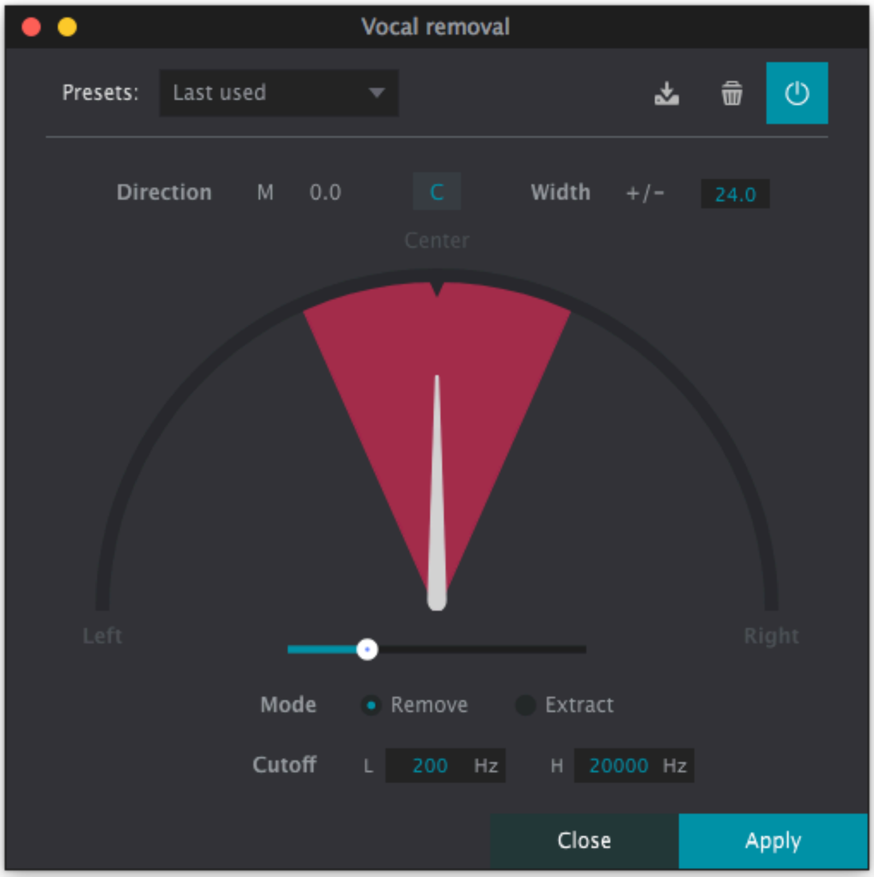
DeEsser reduces sibilance like “s” and “sh” in audio with a compressor.



- *Low Cutoff Knob* – Adjust the cutoff of the high pass filter.
- *High Cutoff Knob* – Adjust the cutoff of the low pass filter.
- *Reduction Knob* – Adjust the amount of reduction when the level of audio surpasses threshold.
- *LookAhead Knob* – Adjust look-ahead time, which affects how fast the processor reacts to audio change.
- *Wide Band Button* – Apply compression on the whole frequency range.
- *Split Band Button* – Only apply compression on the frequency range higher than the low cutoff.
- *Threshold Slider* – Adjust threshold of the compressor.
- *Side-Chain Level* – The level of the filtered audio that is used to detect sibilance.
- *Reduction* – Show the amount of reduction of current audio in real time.

17. Vocal Removal / Extraction

Vocal Removal / Extraction utilizes phase information to separate vocal part from original mix. **It is only suitable for stereo audio**. Move the pointer to adjust the location in the stereo image. If vocal is in the middle, then drag the pointer to the center.



- *Direction* – Display current direction.
- *C Button* – Reset direction to center.
- *Width Number Slider & Slider* – Adjust width.
- *Remove Button* – Remove vocal from mix.
- *Extract Button* – Extract vocal from mix.
- *Cutoff Number Slider* – Adjust the lower and upper of filter.

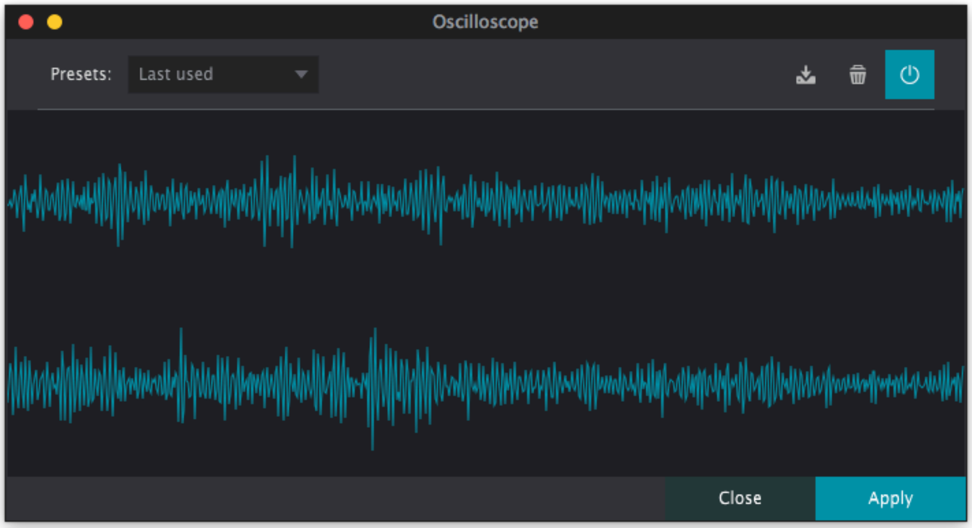
18. Doppler Effect

Doppler Effect modifies the sound according to the movement of the sound source to the reference point: when the sound source is moving towards the reference point, the pitch of the sound is perceived as higher than the actual pitch; when the sound source is moving away from the reference point, the perceived pitch is lower. The level of sound will also change according to the distance between the two points. To use this effect, a section of audio needs to be selected first. Then draw the movement curve of the sound source: click to add drag point, drag to move the curve. The reference point is located as the fixed red point in the center. The effect will be applied on the selected section of audio. When the selected audio is played, a blue point will appear to indicate the location of the moving sound source. The speed of the moving point is related to the duration of selected section.

- *Dist. Scale Knob* – Adjust the distance scale, which affects the relative distance between the sound source and reference point. Higher value represents further distance, representing a larger space.
- *Pitch Scale Knob* – Adjust the amount of pitch change of the moving sound source.
- *Bypass Button* – Choose whether to enable pitch change (or only use level change).
- *Gain Knob* – Adjust output gain.

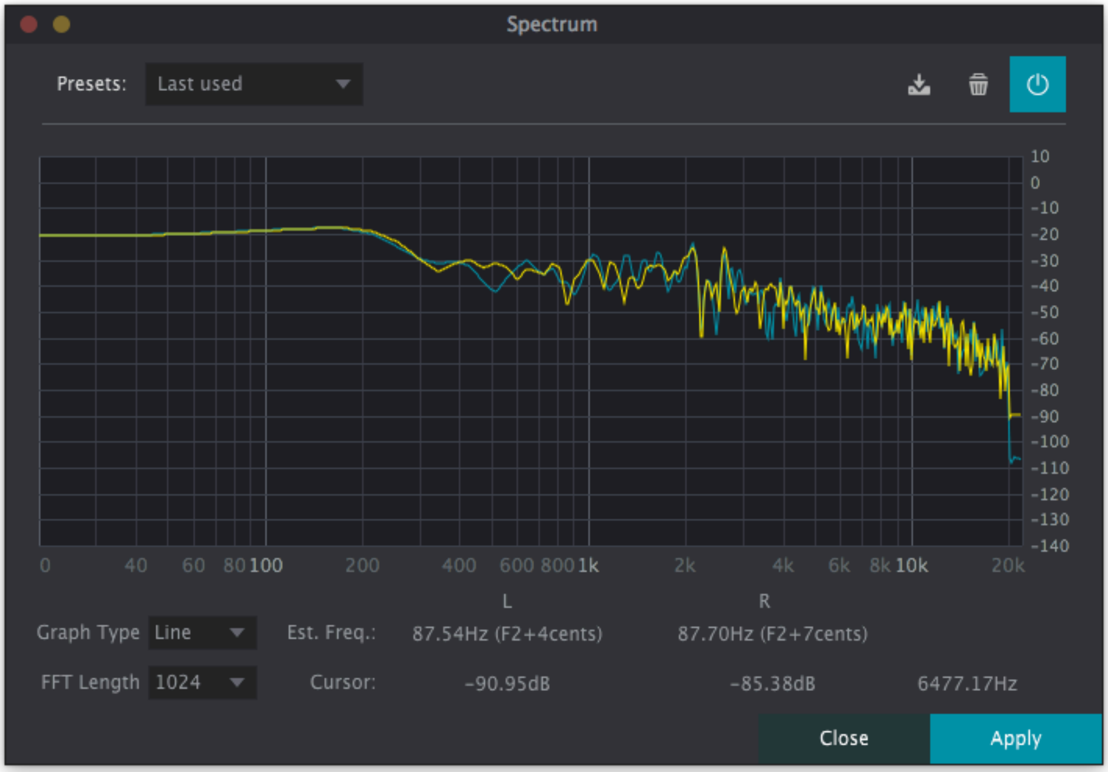
19. Oscilloscope

Oscilloscope shows the scanned waveform of audio.



20. Spectrum

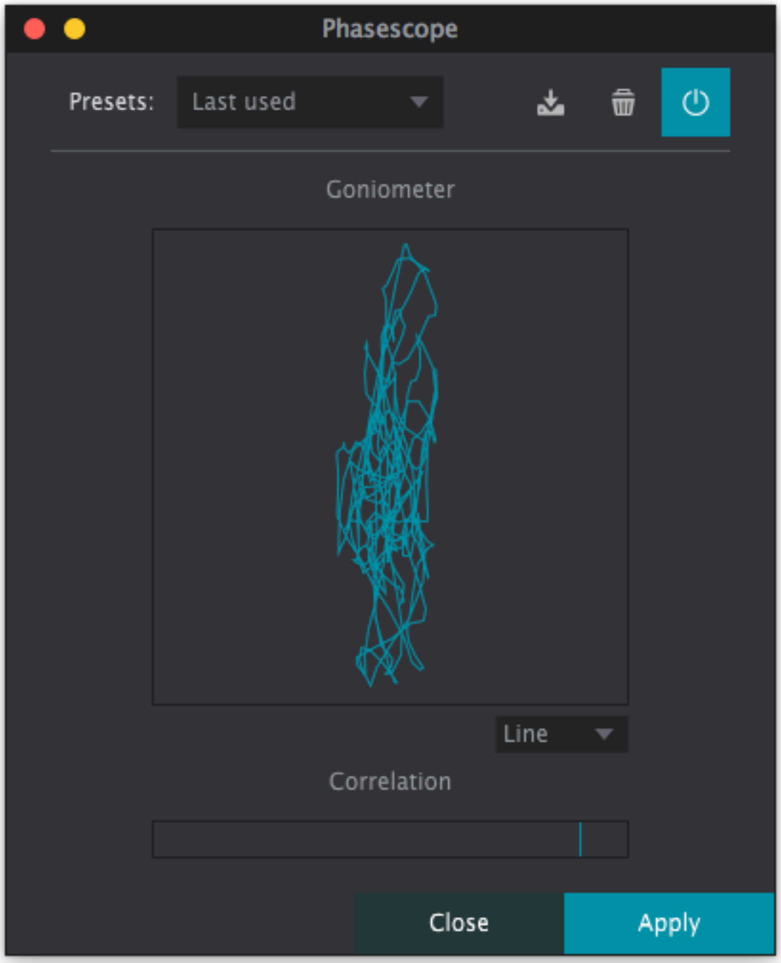
Spectrum shows the real time spectrum of each channel. If there are more than 2 channels, random colors will be used.



- *Graph Type Box* – Choose graph type.
- *FFT Length Box* – Adjust FFT length.
- *Est. Freq.* – Show estimated frequency (pitch) calculated from spectrum (**suitable for monophonic music**).
- *Cursor* – Show the value of spectrum at the frequency where the cursor is.

21. Phasescope

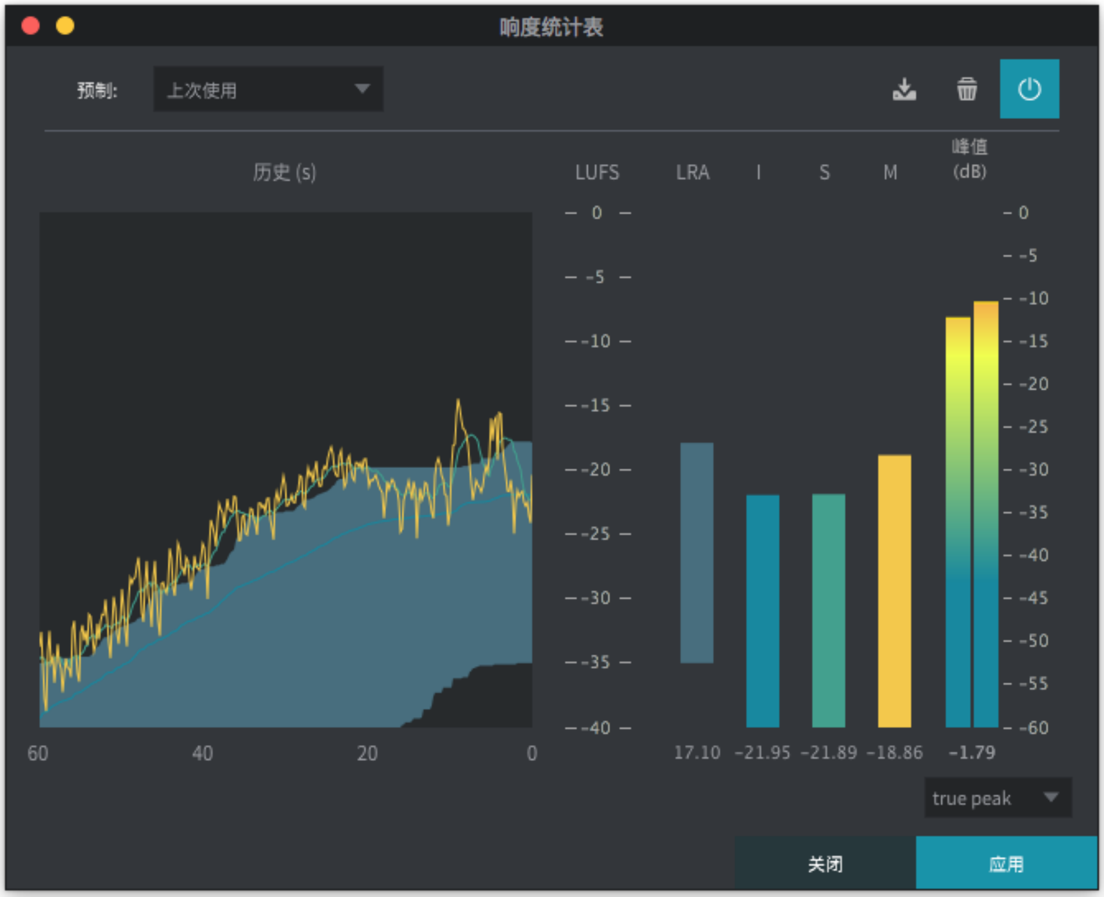
Phasescope is used to monitor the phase of stereo signal, which can indicate out-of-phase occurrences. The upper graph is a Lissajous Curve. If the curve looks like a vertical line, the stereo image is more narrow and close to the center. If the curve looks like a horizontal line, the stereo image is more wide. If the curve is completely a horizontal line, it means the audio is strictly out of phase. The lower meter indicates the correlation of left and right channel. The far right indicates high correlation and vice versa. Far left means strictly out of phase.



- *Graph Box* – Choose the graph type.

22. Loudness Meter

Loudness Meter calculates the loudness statistics according to ITU BS1770 and EBU R128 standards. The statistics include M (Momentary Loudness), S (Short-term Loudness), I (Integrated Loudness), and LRA (Loudness Range). The unit is LUFS. The history graph shows the 60-second history of the statistics. The momentary level of audio is shown at far right.



- *Peak Box* – Choose to use true peak or sample peak for calculating audio peaks.