

PODCASTER

PODCASTING. STREAMING. BROADCASTING MIXER

USER GUIDE



CONTENTS

1. INTRODUCTION
2. SETUP INSTRUCTIONS
3. SIGNAL PATH
4. INPUTS
 - 4.1. INPUT CHANNELS
 - 4.2. PLAYER
 - 4.3. SAMPLER
5. VOICE CHANGER ENGINE
6. MASTER RACK
 - 6.1. COMPRESSOR
 - 6.2. PLUGINS
7. RECORDING
8. METERS
9. INPUT & OUTPUT ROUTING
10. PRESETS
11. MIDI MAPPING
12. KEYBOARD SHORTCUTS
13. SETTINGS
14. SYSTEM REQUIREMENTS
15. TROUBLESHOOTING

1 | INTRODUCTION

Podcaster is a 64 Bits standalone software developed for streamers, podcasters, online radios, talk shows, dialogue recordings and small broadcasting events.

The perfect all-in-one solution for a professional recording & streaming session. It is designed to offer superb audio quality through a solid user-friendly interface.

2 | SETUP INSTRUCTIONS

MAC:

Open Podcaster.dmg. Drag and drop Podcaster to your applications folder. Double-click Podcaster.app.

Important: Please check “**Important info for Mac users**” pdf before launching Podcaster.

WINDOWS:

Drag and drop Podcaster to your Program Files folder. Double-click Podcaster.exe

3 | SIGNAL PATH



4| INPUTS

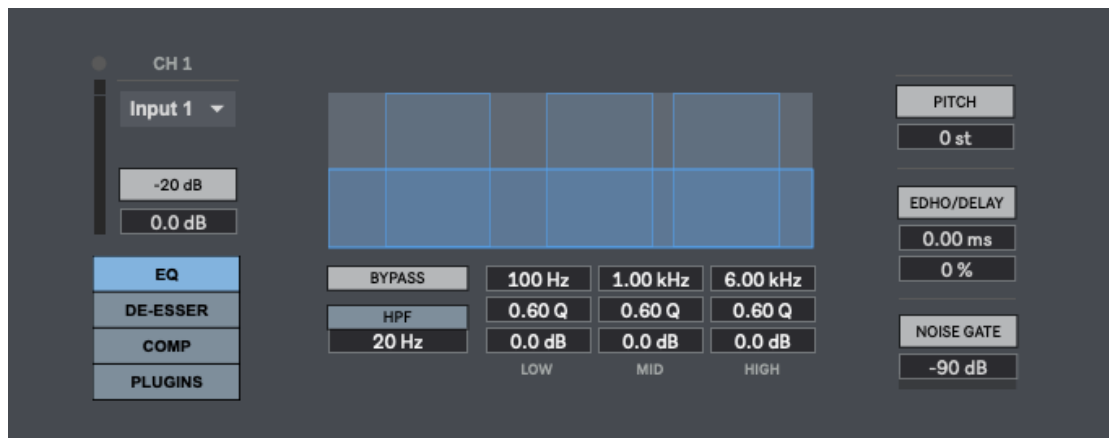
Podcaster is a 7-Channels mixer, featuring 5 mono inputs & 2 stereo inputs channels.

By default, channels 6 and 7 are used for Player and Sampler sections, but can also be used as stereo input channels by receiving audio from your soundcard (external source) instead of using Player and Sampler sections. Just turn on "External In" within these sections and select which soundcard inputs you want to use.

4.1 | INPUT CHANNELS

Up to 5 mono input channels including -20dB Pad, Input Gain, Brickwall-Limiter, 3-Band EQ, High Pass Filter, De-Esser, Compressor, 2 vst/au Plugins, Pitch, Echo-Delay, Noise Gate & Panorama.

Click on the channel name to access to the channel settings.



-20dB Pad: Reduce the level (amplitude) of the input signal by 20 decibels (dB). The purpose of this pad is to prevent an excessively strong signal from overloading or distorting.

Input Gain: Adjust input signal gain.

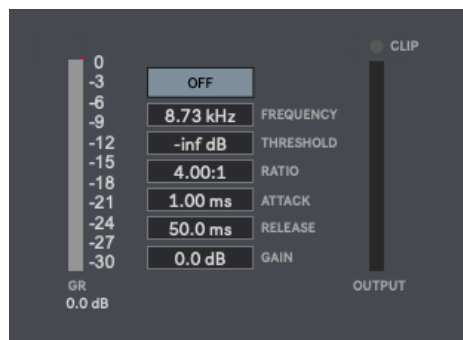
Brickwall-Limiter: The primary purpose of a brickwall-limiter is to control and prevent peak levels from exceeding 0dB, ensuring that the input signal does not go into distortion or clipping. A small led located above the input meter will show you when it's working so you'll need to lower the input gain to avoid distorting the audio signal.

3-Band EQ: A 3-band parametric equalizer allows you to adjust the level (amplitude) of three different frequency bands in an audio signal. Each band can be individually controlled for frequency, gain, and bandwidth, providing precise control over specific portions of the frequency spectrum.

HPF: It is an audio filter that allows frequencies above a certain cutoff point to pass through while attenuating frequencies below that point. The HPF is commonly used to remove low-frequency content, such as rumble, wind noise, or unwanted subsonic frequencies, from an audio signal.

De-Esser: Audio processor used to reduce or control excessive sibilance in vocal recordings or other audio sources. Sibilance refers to the harsh, hissing, or "ss" and "sh" sounds that can be pronounced when certain consonants are pronounced. These high-frequency sounds can be distracting or unpleasant to the listener. A de-esser works by selectively reducing the level of frequencies in the sibilant range, typically around 2 kHz to 10 kHz. When the de-esser detects a sibilant sound above a certain threshold, it applies gain reduction to those frequencies, effectively attenuating the sibilance and taming its prominence in the audio signal.

- **Frequency:** The frequency control allows you to select the central frequency that the de-esser will target for gain reduction. It determines the frequency band where the sibilant sounds, such as "s" and "sh," are most prominent.
- **Threshold:** This determines the level at which the de-esser begins to activate and reduce the sibilant frequencies. When the input signal surpasses the threshold, the de-esser engages its gain reduction process.
- **Attack:** The attack time refers to the speed at which the gain reduction is applied when the sibilant frequencies exceed the threshold. It determines how quickly the de-esser engages and reduces the level of the targeted frequencies. For voice processing is recommended to use fast attack settings from 1/15 milliseconds (ms)
- **Release:** The release time refers to how long it takes for the gain reduction to recover or return to normal after the sibilant frequencies fall below the threshold. It determines the speed at which the de-esser stops applying gain reduction once the sibilance subsides. To avoid noticeable pumping artifacts is recommended to use medium-slow release times from 50/300ms.
- **Gain:** Adjust the output level of the de-esser after the gain reduction process.



Compressor: A compressor is an audio effect used to reduce the dynamic range of a signal by decreasing the volume of louder sounds and increasing the volume of quieter sounds.

- **Threshold:** When the audio signal exceeds the compressor threshold, the compressor activates and reduces the volume of the signal according to its compression ratio and other settings. The threshold acts as a reference point, determining when the compressor kicks in.
- **Ratio:** The compressor ratio is expressed as a numerical value, such as 2:1, 4:1, or 10:1, and represents the relationship between the input level and the output level. For example, a ratio of 2:1 means that for every 2 dB the input signal goes above the threshold, the output signal will only increase by 1 dB. For voiceover processing a 4:1 ratio is recommended.
- **Attack:** The compressor attack time refers to the time it takes for the compressor to start reducing the gain of the audio signal after it exceeds the threshold. It is a parameter that determines how quickly the compressor responds to changes in the signal level. For voice processing is recommended to use fast compressor attack settings from 1/15 milliseconds (ms)
- **Release:** The compressor release time refers to the time it takes for the compressor to stop reducing the gain of the audio signal after it falls below the threshold. It determines how quickly the compressor returns the gain to its normal level once the input level drops below the threshold. To avoid noticeable pumping artifacts is recommended to use medium-slow release times from 50/300ms.
- **Gain:** The compressor gain refers to the amplification applied to the audio signal after compression to restore its overall level. When a compressor reduces the gain of the audio signal above the threshold, the make-up gain compensates for the reduction to achieve the desired output level.



Plugins: Each input channel allows 2 vst/au audio plugins. Click “Browse” to select your plugins and click over plugin name to see it on screen. We strongly recommend using zero-latency plugins.

Pitch: Change the pitch of incoming audio in semitones (+/- 24 st).

Echo/Delay: Audio effect that creates the perception of repeated, decaying sounds that occur after the original sound.

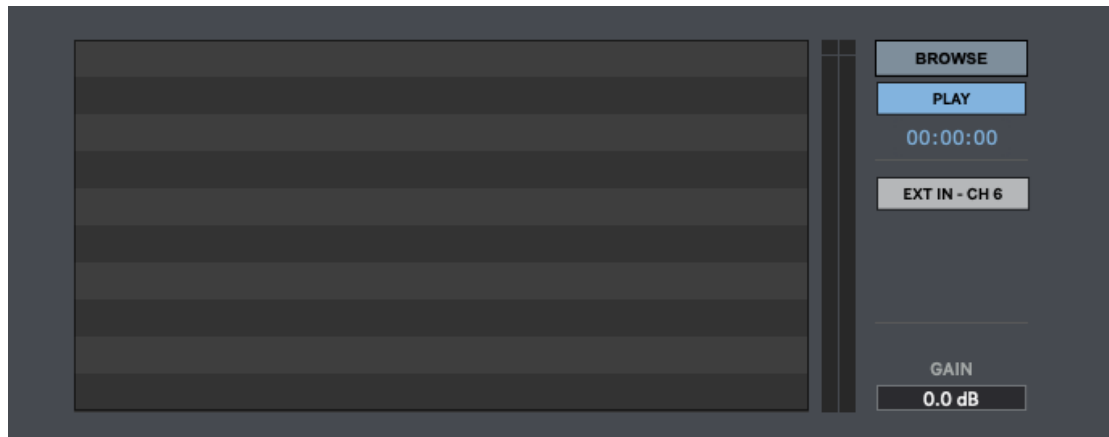
Noise Gate: A noise gate processor for cleaning up audio that has an excess of background noise, such as traffic rumble, air conditioners, chatter in other rooms. It's particularly handy if you are recording in a space that is not professionally sound treated.

4.2 | PLAYER

Player section allows you to play wav/aiff/mp3 audio files from your computer through Podcaster.

Click “Browse” to load any audio folder from your computer within the player library. You can play the audio files by double-clicking them or using spacebar to start/stop the playback (click outside the player library before using spacebar to start the playback).

Additionally, Player section can be used as stereo input channel. Just turn on “External In” and select which soundcard inputs you want to use. When using external in mode, the player section cannot be used.

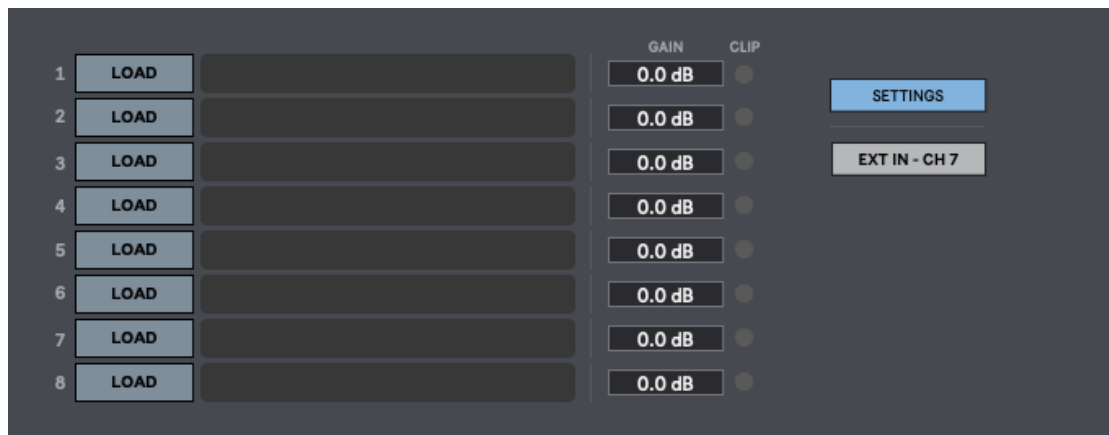
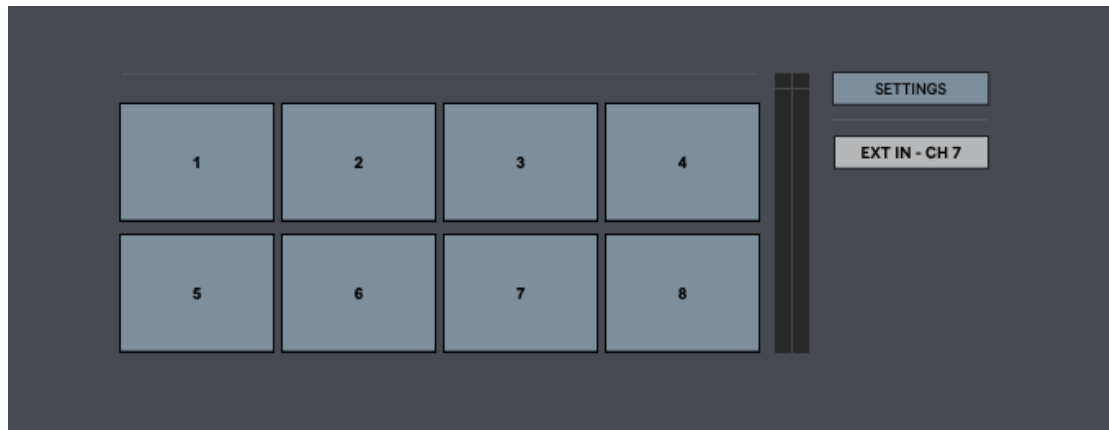


4.3 | SAMPLER

Sampler section allows to play any audio samples through Podcaster using the computer keyboard. Load up to 8 audio samples.

Go to “Settings” window to load audio samples within each sample player. Adjust gain for each audio sampler player. Clip indicator will show you if the player is clipping so you can lower the volume to avoid distortion.

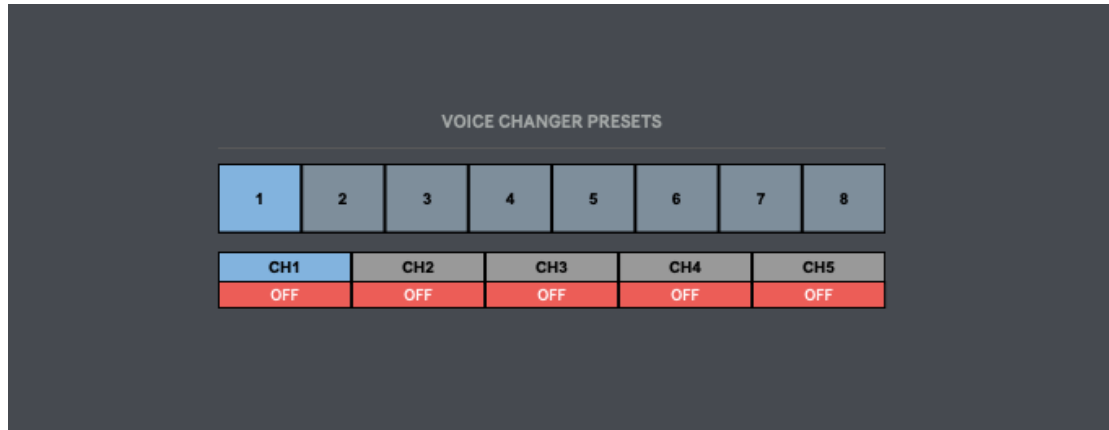
Additionally, Sampler section can be used as stereo input channel. Just turn on “External In” and select which soundcard inputs you want to use. When using external in mode, the sampler section cannot be used.



5 | VOICE CHANGER ENGINE

Podcaster includes a voice changer engine with 8 pre-designed voice effects.

It allows assigning a different voice changer preset to each channel (ch1 to ch5). The stereo channels cannot be processed by voice changer engine.



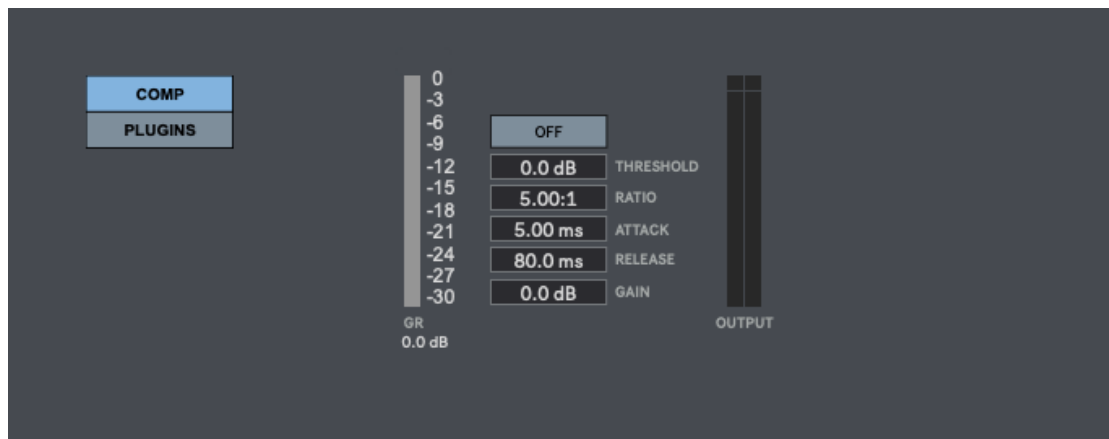
6 | MASTER RACK

Podcaster includes a stereo compressor and 3 audio plugin slots at master rack section. This is the last section before the output stage. Load any vst/au audio plugin.

6.1 | COMPRESSOR

A compressor is an audio effect used to reduce the dynamic range of a signal by decreasing the volume of louder sounds and increasing the volume of quieter sounds.

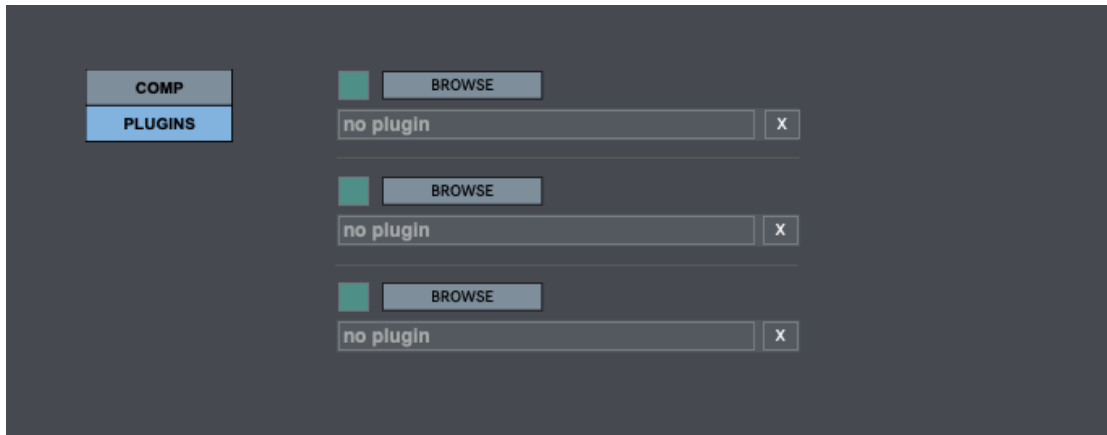
- **Threshold:** When the audio signal exceeds the compressor threshold, the compressor activates and reduces the volume of the signal according to its compression ratio and other settings. The threshold acts as a reference point, determining when the compressor kicks in.
- **Ratio:** The compressor ratio is expressed as a numerical value, such as 2:1, 4:1, or 10:1, and represents the relationship between the input level and the output level. For example, a ratio of 2:1 means that for every 2 dB the input signal goes above the threshold, the output signal will only increase by 1 dB. For voiceover processing a 4:1 ratio is recommended.
- **Attack:** The compressor attack time refers to the time it takes for the compressor to start reducing the gain of the audio signal after it exceeds the threshold. It is a parameter that determines how quickly the compressor responds to changes in the signal level. For voice processing is recommended to use fast compressor attack settings from 1/15 milliseconds (ms)
- **Release:** The compressor release time refers to the time it takes for the compressor to stop reducing the gain of the audio signal after it falls below the threshold. It determines how quickly the compressor returns the gain to its normal level once the input level drops below the threshold. To avoid noticeable pumping artifacts is recommended to use medium-slow release times from 50/300ms.
- **Gain:** The compressor gain refers to the amplification applied to the audio signal after compression to restore its overall level. When a compressor reduces the gain of the audio signal above the threshold, the make-up gain compensates for the reduction to achieve the desired output level.



6.2 | PLUGINS

Podcaster allows using up to 3 vst/au plugins at master rack section.

Click "Browse" to select your plugins and click over plugin name to see it on screen. We strongly recommend using zero-latency plugins.

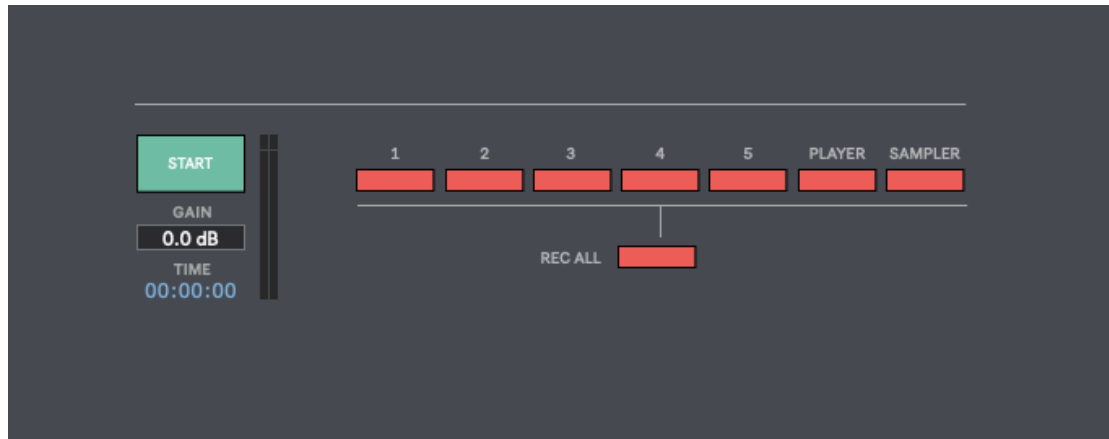


7 | RECORDING

Podcaster allows recording the main audio output in wav (16/24Bits) and up to 192kHz sampling rate (depending your soundcard).

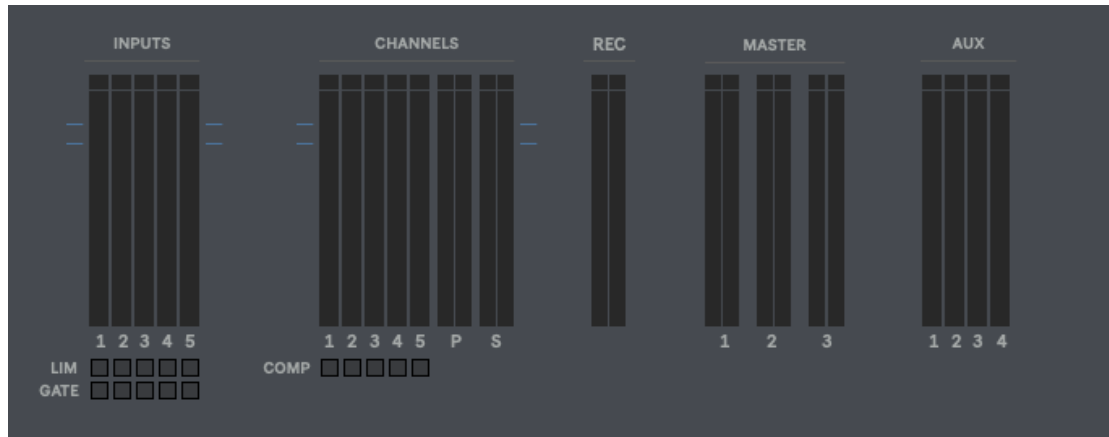
The recording bit depth and sampling rate can be set at Audio Settings window.

You can also record each mixer channel in a separate audio file. This way you will get an individual audio track for each channel if needed for editing purposes. Just select which channels you want to record before pressing “Start”.



8 | METERS

This section allows monitoring all input/output levels for Podcaster.



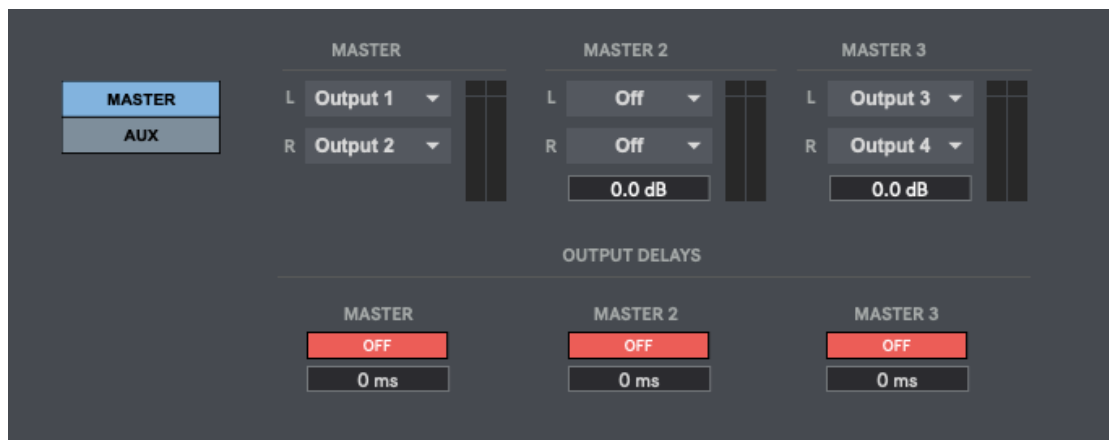
9 | INPUT/OUTPUT ROUTING

This section allows setting the soundcard output channels used for master & aux outputs. Podcaster features 3 master outputs (stereo) and 4 auxiliary outputs (mono).

For instance, you can use the master outputs to feed a loudspeaker system, for audio recording on an external device (recorder/computer) and for syncing to video signal on an external device (computer/cam/converter).

Each master/aux output offers a delay compensation (1/1000ms). This is useful feature when syncing to video signal, as video signal processing usually introduces some latency due to heavy processing needed for video. This way you can compensate the video latency by adding some delay to the audio output to perfectly synch video & audio in real-time.

For each Aux output you can select which channels you want to include in the selected auxiliary output channel.



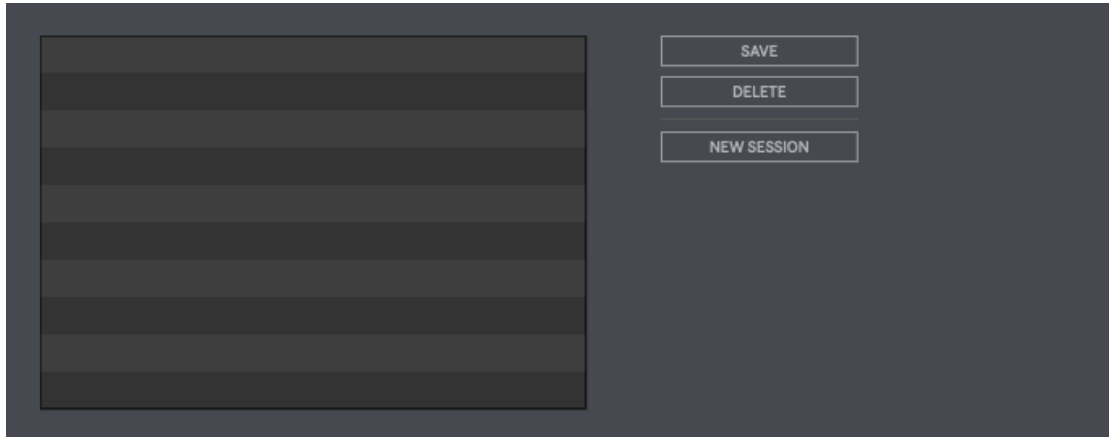
10 | PRESETS

Save and load the whole Podcaster session. By default, Podcaster will always reload the last session used before closing the app.

Click SAVE and introduce the desired preset name to save the whole session data. As soon as you save a preset it will appear within the presets list. Double-click on any preset name to load it.

If you want to delete a preset just select it and click DELETE.

To start a new session from scratch just click on NEW SESSION.



11 | MIDI MAPPING

You can control several controls using your midi device.

At Audio Settings window click on “**MIDI**” to activate midi mapping mode. To assign midi controls just select which control you want to assign and move the desired control within your midi device. To remove a midi assign, just activate again midi mapping mode, right-click on the selected control you want to remove midi assign and select “Delete Mapping”.

Press “**ESC**” to close midi mapping mode.

You can also Access to midi mapping mode using the shortcuts cmd+m (mac) and ctrl+m (windows).

12 | KEYBOARD SHORTCUTS

For better and faster user experience.

Spacebar – Play audio file from Player section

1,2,3,4,5,6,7,8 – Play audio files from Sampler Section

P – Player section

S – Sampler section

V – Voice Changer section

R – Recording section

M – Meters section

A – Audio Settings section

CMD (mac) / CTRL (Windows) + a – Audio Settings

CMD (mac) / CTRL (Windows) + i – IO Settings

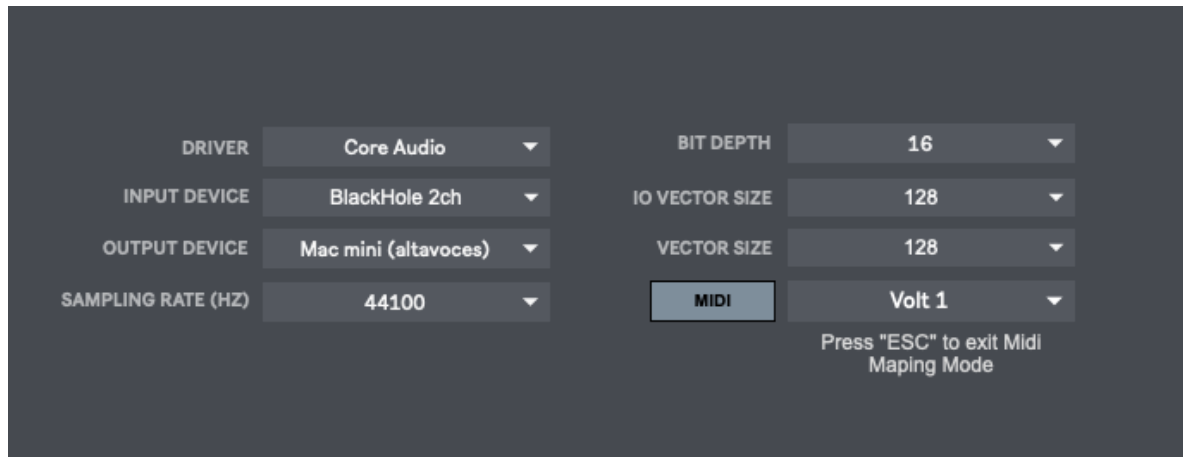
CMD (mac) / CTRL (Windows) + m – Midi Mapping Mode

Shift + Mouse – Holding Shift key allows changing values more accurately when tweaking a numerical display or controller.

CTRL + q (windows) / CMD + q (Mac) - Close Podcaster

13 | SETTINGS

In this section you can select your audio settings. Set up your own input/output configuration depending your soundcard and needs at any time.



- **Driver:** Select your audio driver.
- **Input Device:** Select you input device (soundcard)
- **Output device:** Select you output device (soundcard)
- **Sampling Rate:** Set the sampling rate for the session.
- **Bit Depth:** Set the bit depth for the session.
- **IO Vector Size:** Set the IO Vector Size. The *I/O Vector Size* (I/O stands for input/output) controls the number of samples that are transferred to and from the audio interface at one time. The *I/O Vector Size* may have an effect on latency and overall performance. A smaller vector size may reduce the inherent delay between audio input and audio output, because Podcaster has to perform calculations for a smaller chunk of time. We recommend working at 128.
- **Signal Vector Size:** Set the Signal Vector Size. The *Signal Vector Size* sets the number of samples that are calculated by Podcaster at one time. This can be less than or equal to the I/O Vector Size, but not more. We recommend working at 128.

14 | SYSTEM REQUIREMENTS

Mac

- Mac Intel machine running OS X 10.11.6 or later. 64 Bits. 4GB Ram or more recommended.
- Apple Silicon ARM machine running OS 11.0 or later. 64 Bits. 4GB Ram or more recommended.

Windows

- Windows 8 or later, multicore processor. 64 Bits. 4GB Ram or more recommended.

15 | TROUBLESHOOTING

If you have problems with the sound (clicks/distortion) try changing IO Vector Size and Signal Vector Size to lower values.

THANKS FOR YOUR SUPPORT

Technical support at
support@digitalbrain-instruments.com

DIGITAL BRAIN INSTRUMENTS | <http://www.digitalbrain-instruments.com>
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