

# TRANSFORMER

CREATIVE AUDIO MORPHING & PERFORMING TOOL

# USER GUIDE



# CONTENTS

1. INTRODUCTION
2. SETUP INSTRUCTIONS
3. INPUT
  - 3.1. AUDIO
  - 3.2. MICROPHONE
  - 3.3. VIRTUAL INSTRUMENT
  - 3.4. PROCEDURAL AUDIO
  - 3.5. SPECTRAL CROSSOVER
  - 3.6. EFFECT PLUGINS
4. CARRIER SIGNALS
  - 4.1. SAMPLES
    - 4.1.1. SAMPLE PLAYER
    - 4.1.2. SPECTRAL CROSSOVER
    - 4.1.3. EFFECT PLUGINS
    - 4.1.4. XY PAD
    - 4.1.5. INPUT DEPENDING
  - 4.2. VIRTUAL INSTRUMENT
    - 4.2.1. SPECTRAL CROSSOVER
    - 4.2.2. EFFECT PLUGINS
5. MASTER SECTION
  - 5.1. PITCH SHIFTING
  - 5.2. LFO
  - 5.3. PHASOR
  - 5.4. EFFECT PLUGINS
6. PROCESSING MODE
7. MASTER DRY/WET
8. MASTER DRY/WET + PROCESSING MODE
9. RECORDING
  - 9.1. SAMPLE PROCESSING
10. PRESET MANAGER
11. MIDI MATRIX
12. KEYBOARD SHORTCUTS
13. SETTINGS
14. INPUT & OUTPUT ROUTING
15. SYSTEM REQUIREMENTS
16. TROUBLESHOOTING

## 1 | INTRODUCTION

Transformer is a 64Bit standalone software developed to transform your audio samples into new sounds, using a microphone, pre-recorded audio files or a virtual instrument.

By using convolution processing you can add new layers of sound to existing audio files and performing them into many different ways depending your needs.

Transformer offers a new way of creative audio processing for sound designers and musicians.

**Includes 1.2 Gb of professional audio samples pack: foley, whooshes, explosions, footsteps, hits, creatures, rocks, sci-fi, drones, impacts, cinematic & vehicles.**

## 2 | SETUP INSTRUCTIONS

### MAC:

Open Transformer.mpkg and follow installation steps.

**Important:** Please read “**Important info for Mac users**” pdf before launching this software.

### WINDOWS:

Open Transformer Setup.msi and follow installation steps.

**We strongly recommend copying Transformer Extra Content folder in the same location.**

## 3 | INPUT (MODULATION SIGNAL)

Transformer allows using up to 4 different inputs.

- **Audio:** Audio Samples
- **Mic:** Microphone
- **Instrument:** Virtual Instrument
- **Procedural:** Procedural Audio

### 3.1 | AUDIO

Load any file from your computer (WAV, AIFF and MP3 files).

**Play.** Start/stop playback [“spacebar”]

**Audio Preview.** Turn on PRE on the input channel within the mixer section. [“w”]

**File.** Select your audio file. Drag&drop any audio file within the file name section.

**Folder.** Choose a folder to be shown within the menu. Drag&drop any folder within the folder selection window for faster folder selection.

**Loop.** Loop playback mode. When working in loop mode, carrier audio files will be played from the beginning on each modulation audio file cycle, even if carrier files are also in loop mode.

**Reverse.** Reverse playback.

**Waveform.** Shows the selected audio file waveform.

- Cmd+click on the waveform to zoom in (mouse up) / out (mouse down) and to scroll left (mouse left) /right (mouse right) when zoom is active.

**Waveform Range.** Select a specific region of the audio file length to be processed.

By default the whole file length is selected (all in blue)

Use the mouse to select a specific region. (Selection always in blue)

Left bottom corner shows the selected waveform region length.

- Ctrl + click on the selected waveform region to scroll left/right using the mouse.
- Ctrl + click on the selected waveform region to make the selection bigger (mouse up) or smaller (mouse down).
- Shift + Click on the selected waveform region to select a different selection start/end.

**Set.** Reset zoom and waveform selection range to the whole sample length.

**N.** Normalize file to 0dB. We recommend normalizing sounds for better performance.

**Speed.** Change playback speed from 10% to 265%. Normal speed is 100%.

Click over Speed label to reset playback speed to 100%.

**Pitch.** Set the pitch from -24 to 24 semitones. Click over Pitch label to reset Pitch.

**ENVELOPE.** Amplitude envelope designer allows creating your own envelope.

Audio waveform on this section will only show the selected audio length so you can create a more detailed envelope.

- Click in empty space to add a new envelope point.
  - Shift-click on a point to remove it.
  - Click R to reset envelope.
- T** – Envelope window stays on top.

### 3.2 | MIC

Use a microphone connected to your soundcard as input. You can set the microphone input threshold to trigger carrier signals (**shortcut “a”**), by default is set at -70dB.

### 3.3 | INSTRUMENT

Load and play any VST/AU Instrument from your computer through Transmorph.

**Scan.** Scan your midi devices.

**Menu.** Click on the selected midi device.

**Browse.** Select your vst/au instrument.

**View Instrument.** Click over the instrument name to show it on screen.

**S.** Save the plug-in internal preset. (.fxp file)

**L.** Load a plug-in internal preset (.fxp file)

**X.** Remove instrument.

### 3.4 | SPECTRAL CROSSOVER

Allows selecting a specific spectral frequency range to be processed. Use the mouse to make a spectral frequency range selection.

Left/right bottom corners show the selected frequency range start/end.

- Cmd+click on the selected range to scroll the selection left (mouse left) /right (mouse right)

**Inner/Outer.** Process the selected frequency range (INNER) or the frequency ranges below and above the selection (OUTER).

### 3.5 | EFFECT PLUGINS

Process audio input through up to 2 effect plugins. Use only 64Bit plugins.

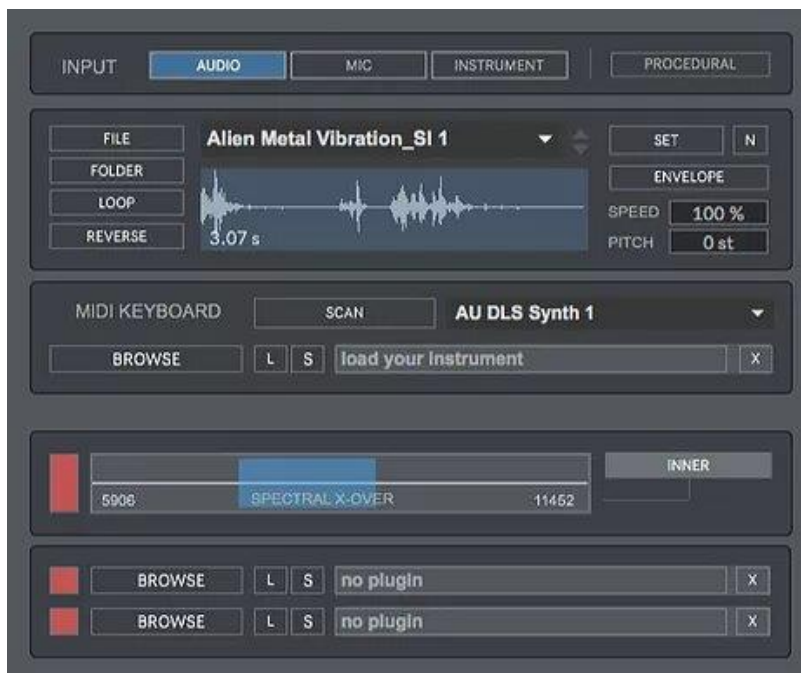
**Browse.** Select your vst/au plugin.

**View Plugin.** Click over the plugin name to show it on screen.

**S.** Save the plugin internal preset. (.fxp file)

**L.** Load a plugin internal preset (.fxp file)

**X.** Remove effect plugin.



### 3.6 | PROCEDURAL (keyboard shortcut "P")

This section allows using procedural audio to modulate the carrier signals (S1, S2, S3, S4 & Ins). You don't need to load any audio file at input section to modulate the carrier signals.

This section generates an audio signal covering the full audio spectrum range (20-20.000Hz) which amplitude is determined by the movement of the amplitude fader, from no sound to maximum amplitude.

**Filter.** Select which audio filter you want to apply to the procedural audio output.

Low Pass Filter (LPF), High Pass Filter (HPF), Band Pass Filter (BPF) or Band Stop Filter (BSF).

**Freq.** Set central frequency for audio filters.

**Q.** Set bandwidth for audio filters, narrow or wide.

**Tone.** Modify the pitch of the procedural audio output, from lower to higher pitch.

**Hold.** When activated it generates a continuous audio signal based on the amplitude settings, you can still increase or decrease the amplitude when Hold is active.

**Amplitude.** Set amplitude for the procedural audio signal.

**T.** Procedural window stays on top.

**IMPORTANT:** When using Procedural Audio input spectral X-Over and plugins are disabled. If you don't move the amplitude fader, procedural audio section will not generate any sound.



## 4 | CARRIER SIGNALS

Transformer allows modulating up to 4 samples and a virtual instrument as a synth or sampler.

This effect uses samples as the carrier files for a convolution processing. Load up to 4 samples from your computer to be modulated simultaneously by the input signal.

The resulting output signal will retain some, or most of the original frequency content of the input signal, while its spectral envelope will be largely determined by the selected samples spectral envelope.

### 4.1 | SAMPLES

Load any file from your computer (WAV, AIFF and MP3 files).

#### 4.1.1 | SAMPLE PLAYERS (Keyboard shortcuts “1,2,3,4”)

**Audio Preview.** Click & hold PREVIEW on the selected sample player.

Click & hold ALL to preview all sample players at same time.

**File.** Select your audio file. Drag&drop any audio file within the file name section.

**Folder.** Choose a folder to be shown within the menu. Drag&drop any folder within the folder selection window for faster folder selection.

**Loop.** Loop playback mode.

**Reverse.** Reverse playback.

**Waveform.** Shows the selected audio file waveform.

- Cmd+click on the waveform to zoom in (mouse up) / out (mouse down) and to scroll left (mouse left) /right (mouse right) when zoom is active.

**Waveform Range.** Select a specific region of the audio file length to be processed.

By default the whole file length is selected (all in blue)

Use the mouse to select a specific region. (selection always in blue)

Left bottom corner shows the selected waveform region length.

- Ctrl + click on the selected waveform region to scroll left/right using the mouse.
- Ctrl + click on the selected waveform region to make the selection bigger (mouse up) or smaller (mouse down).
- Shift + Click on the selected waveform region to select a different selection start/end.

**Set.** Reset zoom and waveform selection range to the whole sample length.

**N.** Normalize sample volume to 0dB. We recommend normalizing sounds for better performance.

**Speed.** Change playback speed from 10% to 265%. Normal speed is 100%.

Click over Speed label to reset playback speed to 100%.

**Pitch.** Set the pitch from -24 to 24 semitones. Click over Pitch label to reset Pitch.

**Delay.** Add a delay to the sampler player output. Click over Delay label to reset Delay.

**Panorama.** Set sampler player output left/right/center.

**ENVELOPE.** Amplitude envelope designer allows creating your own envelope.

Audio waveform on this section will only show the selected audio length so you can create a more detailed envelope.

- Click in empty space to add a new envelope point.
- Shift-click on a point to remove it.
- Click R to reset envelope.
- T – Envelope window stays on top.

#### 4.1.2 | SPECTRAL CROSSOVER

Allows selecting a specific spectral frequency range to be processed. Use the mouse to make a spectral frequency range selection.

Left/right bottom corners show the selected frequency range start/end.

- Cmd+click on the selected range to scroll the selection left (mouse left) /right (mouse right)

**Inner/Outer.** Process the selected frequency range (INNER) or the frequency ranges below and above the selection (OUTER).

#### 4.1.3 | EFFECT PLUGINS

Process sample player output through up to 2 effect plugins. Use only 64Bit plugins.

**Panorama.** Set instrument output left/right/center.

**Browse.** Select your vst/au plugin.

**View Plugin.** Click over the plugin name to show it on screen.

**S.** Save the plugin internal preset. (.fxp file)

**L.** Load a plugin internal preset (.fxp file)

**X.** Remove effect plugin.





#### 4.1.4 | XY PAD

XY Pad allows you to mix the output levels of sample players before to be processed. It works as a virtual mixing step that take place before the convolution processing and before the mixer section.

This way you can blend between the outputs from all sample players to find the right balance using the mouse. If you need a more accurate balance between sounds just use the volume faders at mixer section. It is very useful to create mode dynamic sounds where the level from each audio sample used changes constantly.

Sample players 1 & 2 are paired vertically and sample players 3 & 4 are paired horizontally, it means that if you move the pointer (yellow circle) up, S2 output will be decreased so S1 output will be more present, and viceversa if you do it the opposite way. It works on the same way for S3 & S4. When the pointer is placed in the center of the xy axis all outputs are balanced at same volume.

You can always activate/deactivate sample player outputs. Deactivated sample players will not be processed by this virtual mixing section and will be bypassed to the next stage at its original volume. This way you can select which sample player outputs you want to mix using xy pad while not processing deactivated outputs through xy pad.

Press R to reset xy pad to central position.



#### 4.1.5 | INPUT DEPENDING

Activate or deactivate sample player outputs depending on the input level.

It allows creating more dynamic sounds by activating sample players outputs only when the input level is above the selected threshold for sample player. For example, gradually activating sample players outputs when the input level is increased.

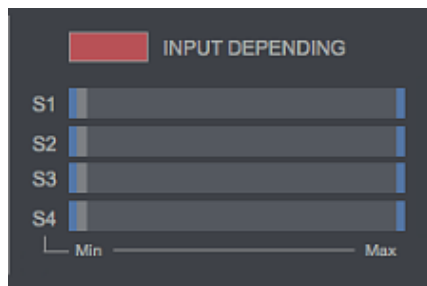
Select the threshold level for sample player individually.

A small led indicates when the sample player output is inactive (red)/active (green).

**Min** = Minimum threshold level. Sample player outputs always active.

**Max** = Maximum threshold level (0 dBfs)

**Led.** When Input Depending is activated a small led will appear next to each fader.



## 4.2 | VIRTUAL INSTRUMENT (Keyboard shortcuts “5”)

Load and play any VST/AU Instrument from your computer. Up to 2 instruments.

**Keyboard mode.** Use the keyboard to play the instrument.

**Auto Key mode.** Instrument will be played using the selected auto key. You don't need to play the midi keyboard to generate sound. The selected key will be hold as long as input signal is detected.

**Browse.** Select your vst/au instrument.

**View Instrument.** Click over the instrument name to show it on screen.

**S.** Save the plug-in internal preset. (.fxp file)

**L.** Load a plug-in internal preset (.fxp file)

**X.** Remove instrument.

**Panorama.** Set instrument output left/right/center.

**Preview.** Turn on PREVIEW to preview virtual instrument output.

### 4.2.1 | SPECTRAL CROSSOVER

Allows selecting a specific spectral frequency range to be processed. Use the mouse to make a spectral frequency range selection.

Left/right bottom corners show the selected frequency range start/end.

- Cmd+click on the selected range to scroll the selection left (mouse left) /right (mouse right)

**Inner/Outer.** Process the selected frequency range (INNER) or the frequency ranges below and above the selection (OUTER).

### 4.2.2 | EFFECT PLUGINS

Process virtual instruments output through up to 2 effect plugins. Use only 64Bit plugins.

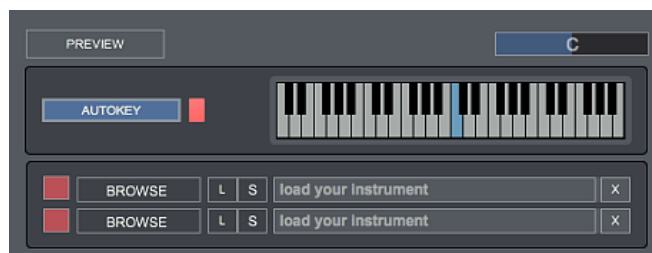
**Browse.** Select your vst/au plugin.

**View Plugin.** Click over the plugin name to show it on screen.

**S.** Save the plugin internal preset. (.fxp file)

**L.** Load a plugin internal preset (.fxp file)

**X.** Remove effect plugin.



## 5 | MASTER SECTION (Keyboard shortcut "6")

Master section features:

- 5-Band Parametric EQ (Click over EQ label to reset EQ)
- Pitch Shifter
- LFO
- Phasor
- 4 vst/au plugins

### 5.1 | PITCH

The pitch-shifting stage can work in 2 modes:

- **FIXED:** Normal mode. +/- 24semitones.
- **DYNAMIC:** This mode automatically changes up/down the pitch of the incoming audio signal. Semitones are increased/decreased faster or slower depending the user-defined modulation speed.

**Up/Down.** Choose pitch-shifting processing from lower to higher values (Up) or from higher to lower values (Down). Pitch shifting always works in Palindrome mode, it means as soon as the semitone value reach the minimum or maximum value selected it will start decreasing or increasing again for as long as input is detected.

**Min.** Lower semitone value. (Min. -24 semitones)

**Max.** Higher semitone value. (Max. 24 semitones)

**Mod. Speed.** Set the amount of time between each semitone increase/decrease.

**L.** Turn it on to link Pitch and LFO modulation speed.

**To work properly minimum parameter must to be always lower than the maximum parameter.**

### 5.2 | LFO

The LFO stage can work in 2 modes:

- **FIXED:** Normal mode. 1-40Hz.
- **DYNAMIC:** This mode automatically modulates up/down the LFO Rate. LFO Rate is increased/decreased faster or slower depending the user-defined modulation speed.

**Up/Down.** Change LFO Rate from lower to higher values (Up) or from higher to lower values (Down). LFO Rate Modulation always works in Palindrome mode, it means as soon as the LFO Rate reach the minimum or maximum value selected it will start decreasing or increasing again for as long as input is detected.

**Min.** Lower LFO Rate value. (Min. 1 Hz)

**Max.** Higher LFO Rate value. (Max. 40 Hz)

**Mod. speed.** Set the amount of time between each LFO Rate increase/decrease.

**L.** Turn it on to link Pitch and LFO modulation speed.

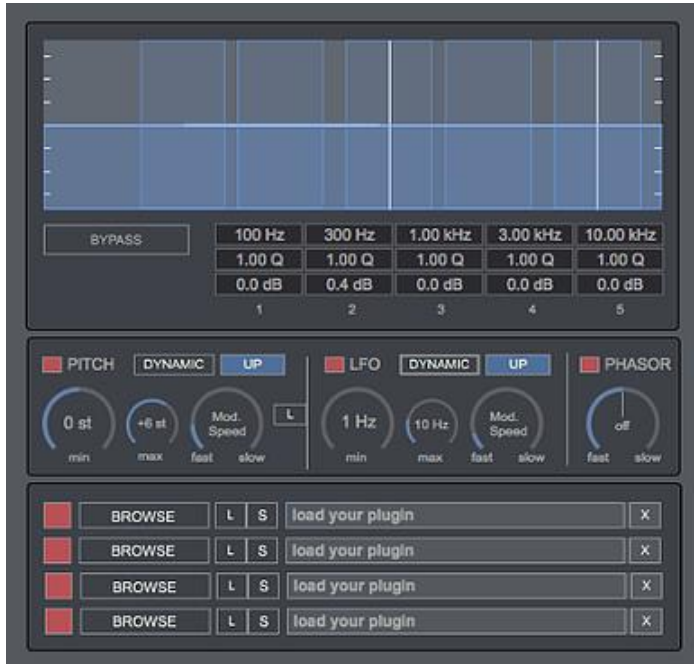
**To work properly minimum parameter must to be always lower than the maximum parameter.**

### 5.3 | PHASOR

Process stereo output through a phasor, from faster to slower processing.

### 5.4 | PLUGINS

Load up to 4 vst/au plugins at output stage. Use only 64Bit plugins.

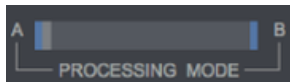


## 6 | PROCESSING MODE

Set the convolution processing mode.

- **Mode A:** Normal mode. Input as modulation signal and sample players/Instrument as carrier signal.
- **Mode B:** Reverse mode. Input as carrier signal and sample players/instrument as modulation signal.

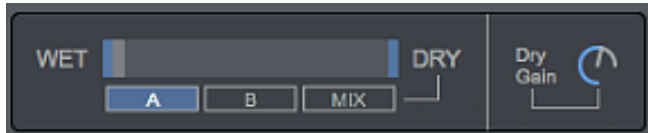
Processing mode fader allows setting the processing mode, from A to B, or mixing both modes.



## 7 | MASTER WET/DRY

The Dry/Wet control allows you balancing the processed signal with the original unprocessed signal.

- **Dry signal selector:** Allows selecting which signal is used as dry signal (unprocessed).
  - A. Input signal as dry signal.
  - B. Sample players/instrument signal as dry signal.
  - Mix.** Both audio signals as dry signal. In this mode both signals are summed.
  - Dry Gain.** Set dry signal gain.



## 8 | MASTER WET/DRY + PROCESSING MODE

This section allows you changing from dry to wet signal and from processing mode A to processing mode B using only one control.



## 9 | RECORDING (keyboard shortcut "R")

Stereo output recording in WAV or AIFF (16/24/32Bits).  
Up to 192kHz (depending your soundcard).

- Stereo/Mono.** Select to export Stereo or Mono files.
- Destination.** Select the destination folder and file name.
- Rec.** Press to start recording.
- Stop.** Press to stop recording.
- Volume.** Set the recording volume

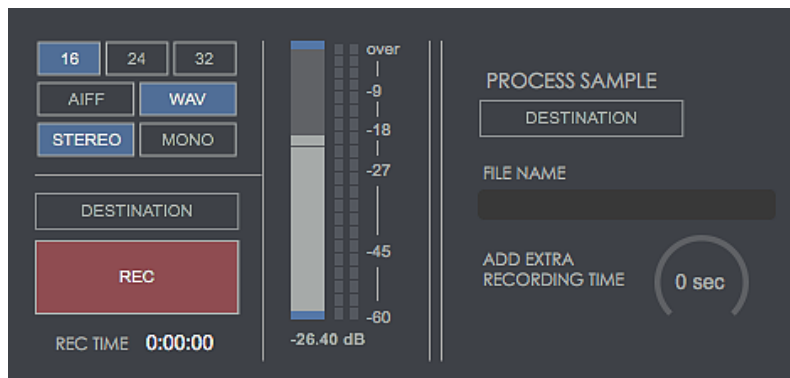
**IMPORTANT:** You'll need to set recording destination folder and file name for each recording take.

## 9.1| SAMPLE PROCESSING

Process the selected audio file within the input audio player in one-click.

Recording file properties will be the same as in Recording section, bit depth, file type and stereo/mono. You need to set destination for sample processed files

- Set process sample destination. Every time you process a file it will be saved within the same destination folder.
- Add time to the recording files. In case you used time modification plugins as delays or reverbs.
- Choose file name. Files are stored in destination folder with the same name but adding the selected name at the end of the file name. (Ex: door.wav will be saved as door.xxx.wav). Do not press **Enter** after typing in the file name.
- Press **PROCESS** on the main UI. Audio rendering will stop automatically when the file has been processed. Process button will turn red when its working and green when it's ready again for a new sample processing.

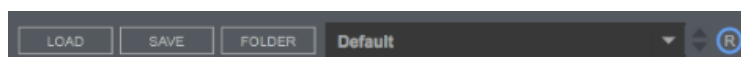


## 10 | PRESET MANAGER

Save and load your own presets. Including midi configuration.

To save your own presets, we strongly recommend creating a new folder in your computer with the same name of the preset and save it inside this folder. Preset saving process automatically create a **.json** (Voxpat preset) file and some **.fxp** (plug-in preset) files associated to the plug-ins used. To restore any session you just need to open the **.json** file associated to this session and the whole session data will be restored. **Do not separate these files.**

- Controls:**
  - Load.** Load a preset from your computer.
  - Save.** Save a preset in your computer
  - Folder.** Select your preset folder to be shown in the menu. It makes quicker to change between presets.
  - Drag & Drop Folder.** Click FOLDER and drag/drop any folder within the dialog window.
  - R.** Reset to default setup. Reset will not affect your midi channels configuration.
  - +/- .** Select previous or next preset within the menu.



## 11 | MIDI MATRIX (keyboard shortcut “CTRL+M/windows - CMD+M/Mac”)

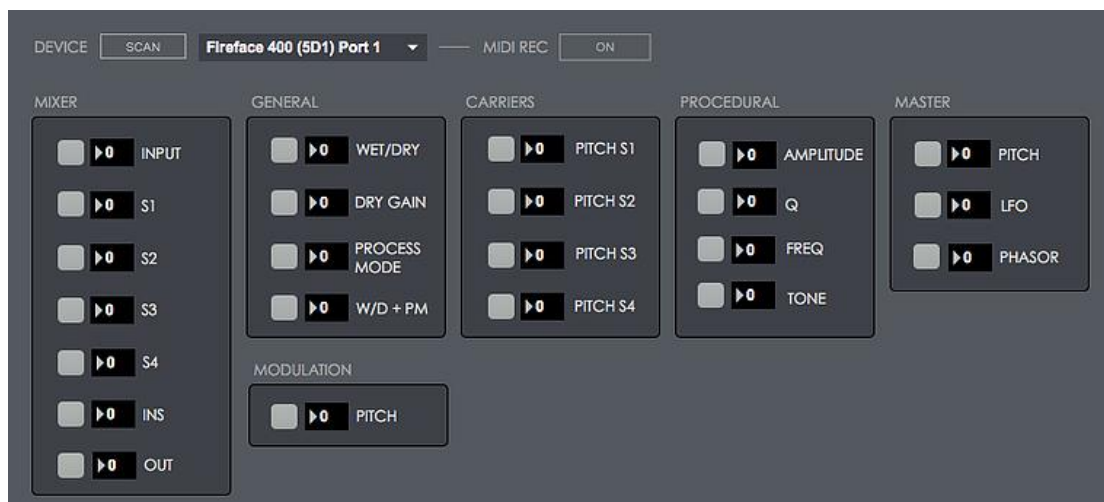
Control software parameters via your midi device.

**Scan.** Scan midi controllers in your computer. Click the selected controller.

**Play/Learn modes.** Click on the selected parameter to activate midi learn mode (red), touch the controller in your midi device that you want to associate to the selected Voxpat parameter. You can also manually select the midi channel assigned to each parameter control. Click again to change to Play mode (white).

**MIDI REC.** This section allows transferring your midi configuration from one preset to another preset automatically. This way you don't need to setup again the same midi configuration manually each time you change to another preset (Keep in mind that midi configuration is also stored when you save a preset).

*Turn MIDI REC “ON” (by default) and setup your midi configuration or load your midi map preset. Turn MIDI REC “OFF” before loading a new preset, once a new preset is loaded the last used midi configuration will be restored automatically. Then save and overwrite this preset in your presets folder. The new midi configuration will be now stored within this preset.*



## 12 | KEYBOARD SHORTCUTS

### GENERAL SECTION

- **p** – Procedural Audio
- **ctrl/cmd + m** – Midi Matrix
- **ctrl/cmd + s** – Settings
- **r** – Recording
- **a** – Microphone Input triggering threshold
- **CMD (mac)/CTRL (windows)** – Holding this key allows changing values more accurately when tweaking a numerical display or controller.
- **CTRL + Q (windows) / CMD + Q (Mac)** – Quit Transformer.

### INPUT SECTION

- **Spacebar** – Play/Stop input sample player
- **w** – Activate input preview
- **1,2,3,4** – Sample players (carrier signals)
- **5** – Virtual Instrument player (carrier signal)
- **6** – Master section



## 13 | SETTINGS (keyboard shortcut “CTRL+S/Windows – CMD+M/Mac”)

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

Click on **IO MAPPINGS** (bottom right) to edit the input/output configuration associated to your soundcard inputs/outputs.

Default Input/Output configuration:

**Mic Input - Soundcard Input 1**

**Main Out – Soundcard Outputs 1(L) & 2(R)**

## 14 | INPUT & OUTPUT ROUTING

Send audio from any DAW or audio editor to Transformer, or from Transformer to any other software for recording.

Via **BlackHole** (Mac) or **Jack Audio** (Windows/Mac) you can use Voxpat within any DAW or audio editor.

**BlackHole:**

<https://github.com/ExistentialAudio/BlackHole>

**Jack Audio:**

<https://jackaudio.org/downloads/>

## 15 | 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.

**Minimum screen resolution 1400x900. Recommended screen resolution 1920x1080.**

It can work on 1080x720 but the lowest interface section will be lost (not important, there is not any control missed on this lower section).

## 16 | TROUBLESHOOTING

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

### **THANKS FOR YOUR SUPPORT!**

1.2 GB Samples pack provided by **Mechanical Wave SFX**  
<https://www.mechanicalwavesfx.com>

Technical support at  
[support@digitalbrain-instruments.com](mailto:support@digitalbrain-instruments.com)

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