MAIR Rendering Tool

USER DOCUMENTATION

APPLIED PSYCHOACOUSTICS LABORATORY

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Quick Start Guide

1. Download and install Max 7
2. Load *MAIR_RenderingTool_v1.mxf*
3. Press the 'MAIR Folder' button in the patch and find the *MAIR_Lib2017* folder on your Mac.
4. Press the 'BRIR Folder' button in the patch and find your BRIR folder (e.g. SADIE_BRIRs) folder on your Mac.
5. Load a sound source into the patch.
6. Select a Microphone array.
7. Choose output type (binaural or speaker).
8. Turn on the DSP in Max.

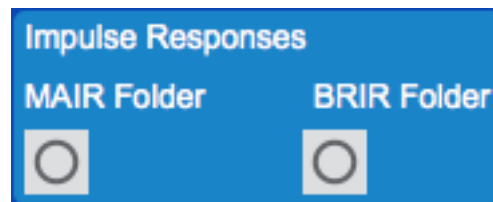## Impulse Responses Section

Image 1a.

This section of the patch deals with loading the Microphone Array Impulse Response (MAIR) Library and the Binaural Room Impulse Responses (BRIRs).

Once the 'MAIR Folder' button is pressed a dialogue box opens, in which you must locate the main folder for the MAIR Library (the folder that holds all the mic array subfolders). Image 1b shows an example of this.
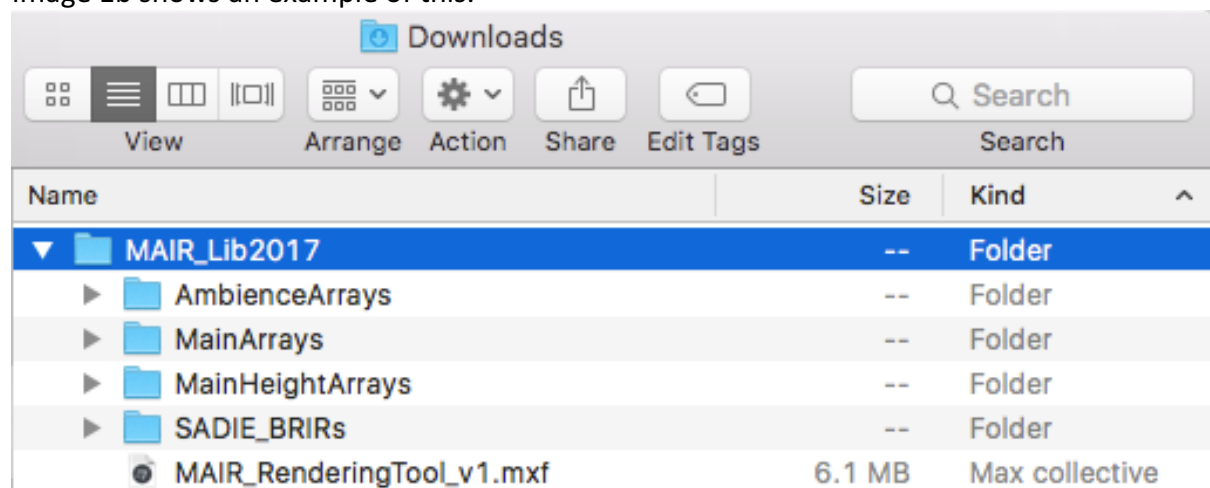

Image 1b.

The 'BRIR Folder' button works in the same way. You are asked to find the folder which contains the BRIRs you wish to use. Users of the rendering tool can use other BRIRs than the ones provided, so long as the order and positions of the impulses are the same as what is shown in Table 1a. The BRIRs also need to be stereo and have a sampling frequency of 44.1kHz.

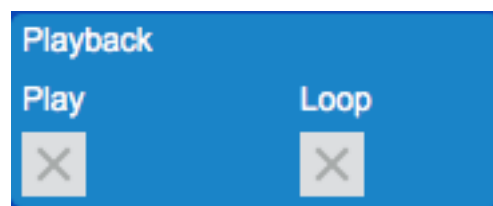| Order number	Speaker position
1	Front Left (FL)
2	Front Right (FR)
3	Front Centre (FC)
4	Rear Left (RL)
5	Rear Right (RR)
6	Front Left Height (FLh)
7	Front Right Height (FRh)
8	Rear Left Height (RLh)
9	Rear Right Height (RRh)

Table 1a.

Playback Section

Image 2a.

This section will control the playback of the sound sources loaded directly into the rendering tool.

NOTE: Once a new sound source has been loaded, you must repress the play button to restart playback.
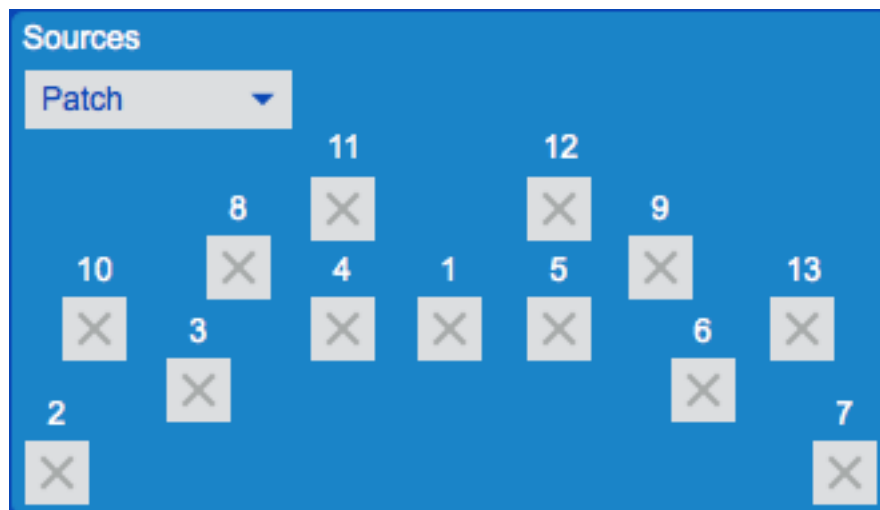
Source Section

Image 3a.

In this section, you can load any .wav and .au files with a sampling frequency of 44.1kHz to any of the 13 source positions when the 'Patch' mode is selected in the dropdown menu. Once one of source positions is clicked, a dialogue box opens which allows you to find your chosen audio file. To clear the audio file, simply click the source position it was loaded into. This will however mean you must reload the audio file if you want to hear it again in that position.

Another option is to stream audio to the rendering tool through a DAW. To do this you must use a virtual audio interface (e.g. SoundFlower for mac) to send the audio into the inputs of the rendering tool, which are inputs 1-13.
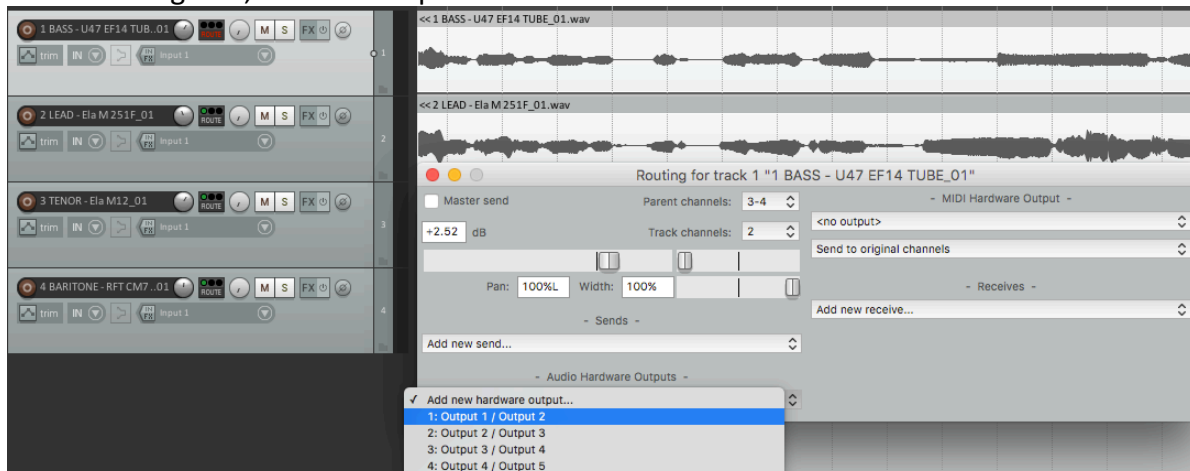

Image 3b

An example of sending audio to the rendering tool through a DAW is shown in Image 3a. As the tool accepts audio from inputs 1-13, all that needs to be done is the DAW is change the audio device to the virtual audio interface and send the audio through one of the hardware outputs from 1-13. The virtual audio interface also needs to be set as the rendering tool's input device in Max 7. The output channel number will correspond with the source position in the tool.
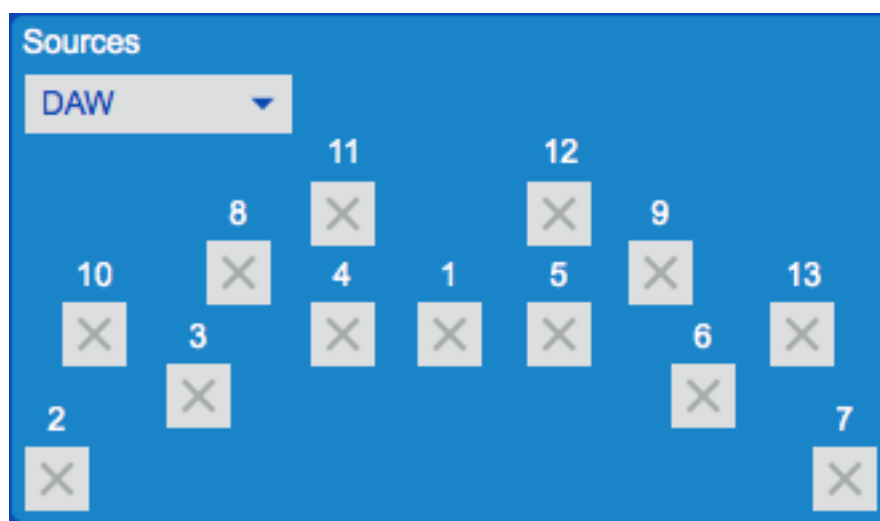

Image 3c.

When the Source section's dropdown menu is set DAW the function of the source positions changes. The toggles on the source positions now have a mute and unmute function. By default, all positions are muted at first so once audio has been sent by the DAW, you must unmute the chosen source position, e.g. if you have sent audio to channel 1, you must unmute source position 1.

For more information on the source positions, please refer to the *Microphone Array Impulse Response (MAIR) Library for Spatial Audio Research* paper.
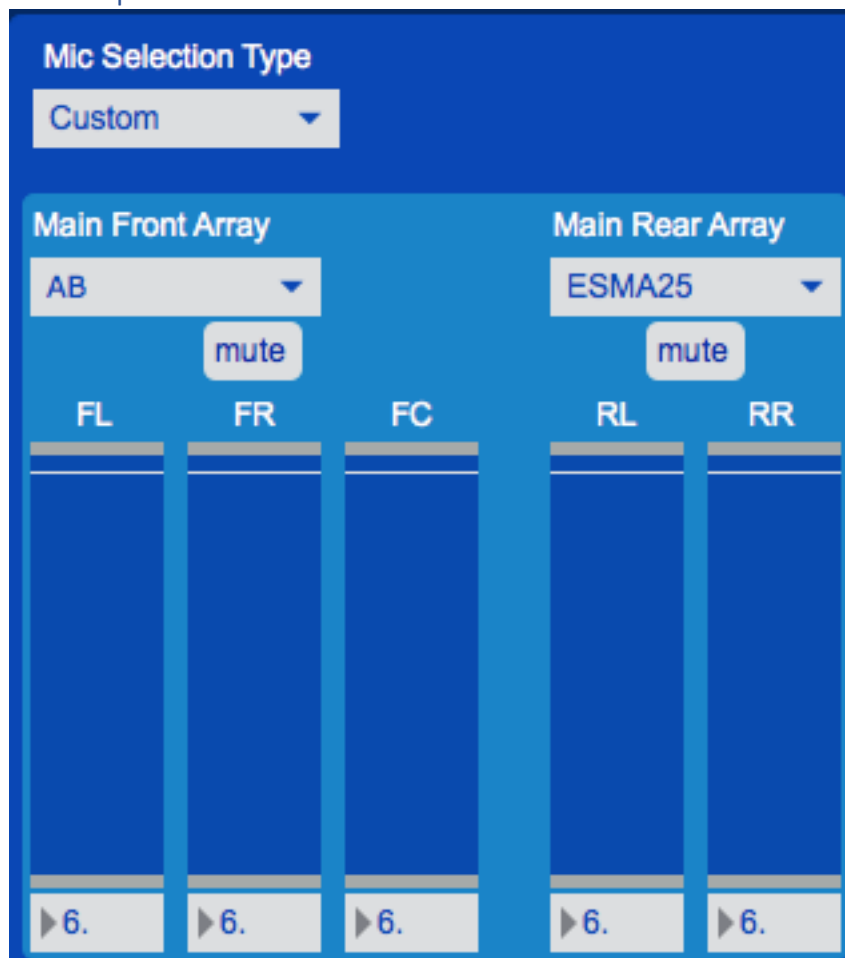
Mixer and Microphone Section

Image 4a.

This section allows you select which microphone array is used to capture the sound source. The 'Mic Selection Type' has two options in the dropdown menu: 'Preset' and 'Custom'. 'Preset' will allow you select complete microphone arrays, whereas 'Custom' allows you to change what microphone array is used for the front pair/triplet and which microphone array is used for the back pair.

The mute function allows you to silence the audio for the front and back of an array individually. This mute also can help reduce CPU load as it mutes some DSP functions in Max 7.

The sliders control the level for the virtual microphone signals. They have up to +6 dB of gain.
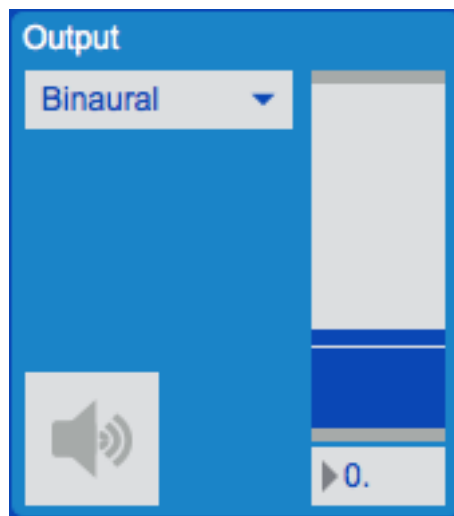
Output Section

Image 5a.

The Output section controls the level of the output signal and whether the signal is sent straight to a multichannel loud speaker array or binauralised for headphone use. The dropdown menu will set the output type and slider has a maximum of +12 dB. The speaker icon turns the DSP in Max 7 on and off. When using the speaker output the audio is routed to the channels shown in Table 2a.

Output Channel	Speaker
1	Front Left (FL)
2	Front Right (FR)
3	Front Centre (FC)
4	NOT USED – No LFE
5	Rear Left (RL)
6	Rear Right (RR)
7	Front Left Height (FLh)
8	Front Right Height (FRh)
9	Rear Left Height (RLh)
10	Rear Right Height (RRh)

Table 2a.

To report any bugs or ask any query, please contact Connor Millns on this email: connor.millns@hud.ac.uk