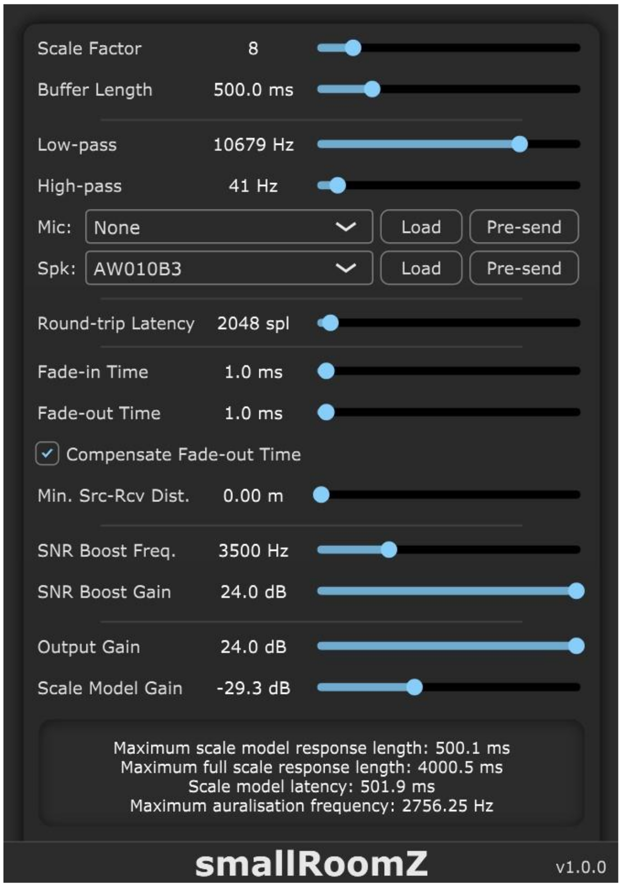
smallRoomZ User Manual v1.0.0



27 SEPTEMBER 2023

Authored by: Peter Stitt

smallRoomZ Plug-in

smallRoomZ is a VST® 3 plug-in that aims to help make working with scale models for real-time auralisation accessible to acousticians and engineers working in DAW environments.

smallRoomZ enables the user to make changes to their scale model and hear the changes to the acoustics in real-time, providing instant feedback on the influence of a change of materials or positioning. The plug-in allows two sources to be played simultaneously.

It also allows recording with two microphones, allowing for binaural rendering through the use of a scale model dummy head. The processing includes several useful DSP tools to improve the quality of the auralisation, such as custom correction filters for the speakers and microphones used.

Signal Routing

The plug-in can be used in any DAW that supports tracks with 4 input and output channels. Inputs 1 and 2 are supplied with the live input of the signal to be played in the

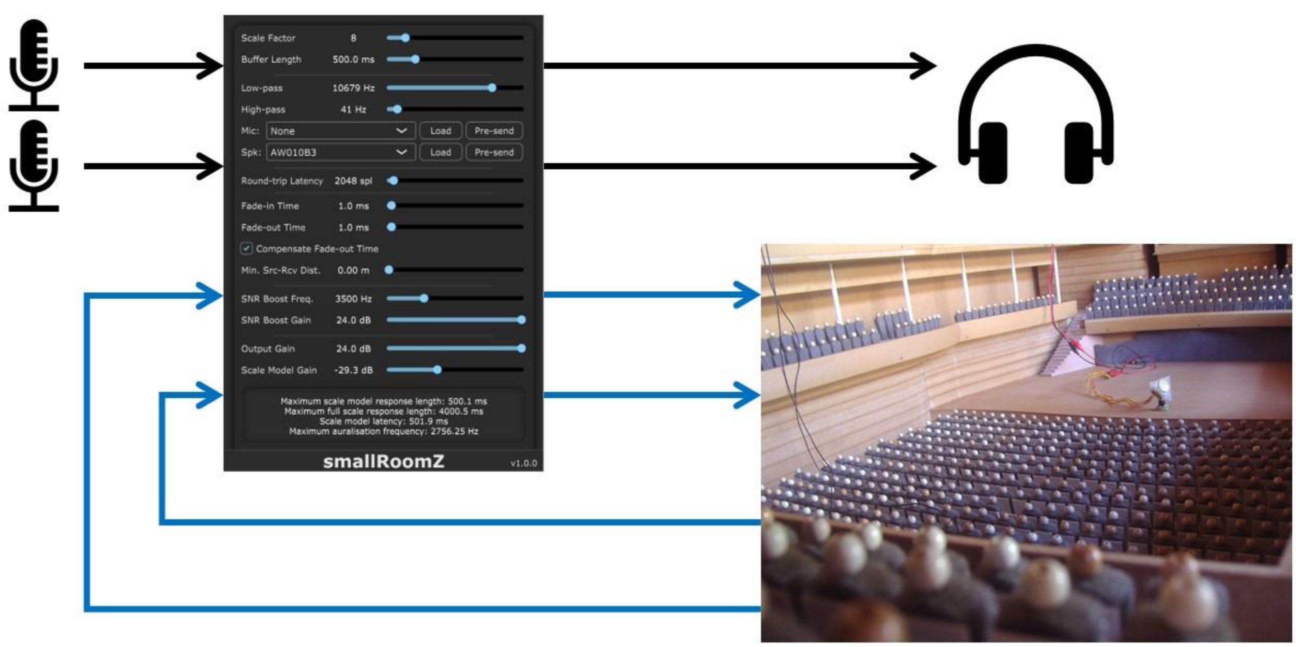


Figure 1 - The signal routing for stereo source recording with stereo recording in the scale model.

scale model. Inputs 3 and 4 are the returns from scale model. Outputs 1 and 2 are the final auralisation while outputs 3 and 4 are the signal to be sent to the scale model. The signal routing for a stereo recording and playback is shown in Figure 1.

Note: If only a single source and microphone are being used in the scale model these should be routed to the inputs 1 and 3 respectively. Similarly, output 1 will be the auralisation while output 3 is the scale model send signal.

User Interface Overview

The user interface, shown in Figure 2, is divided into several sections to group the parameters into different sections depending on their use. This section provides a brief description of these sections. The next section describes each parameter in more detail.

- The top section contains parameters related to the scale model itself (specifically its scale factor and its reverb time).
- The next section provides some EQ controls – low- and high-pass for band-limiting the signal and correction filters for the microphone and speaker.
- The third second contains on the round-trip latency parameter.
- The fourth section contains parameters related to the fade-in and -out windows that are used to reduce clicks when playback is in the presence of on background noise.
- The fifth section provides controls for shelf filtering that can be useful to boost the gain of the output to the scale model and then the corresponding inverse filter on the return from the scale model. As long as clipping is avoided (digital or at the speaker) then this can lead to a better signal-to-noise ratio and reduce noise related to e.g. the audio interface.
- The bottom section contains parameters that control the gain of the auralisation output and the output to the scale model.

The bottom of the user interface has an information box that details the latency of the system, the maximum reverberation times, and the maximum reproduction frequency.

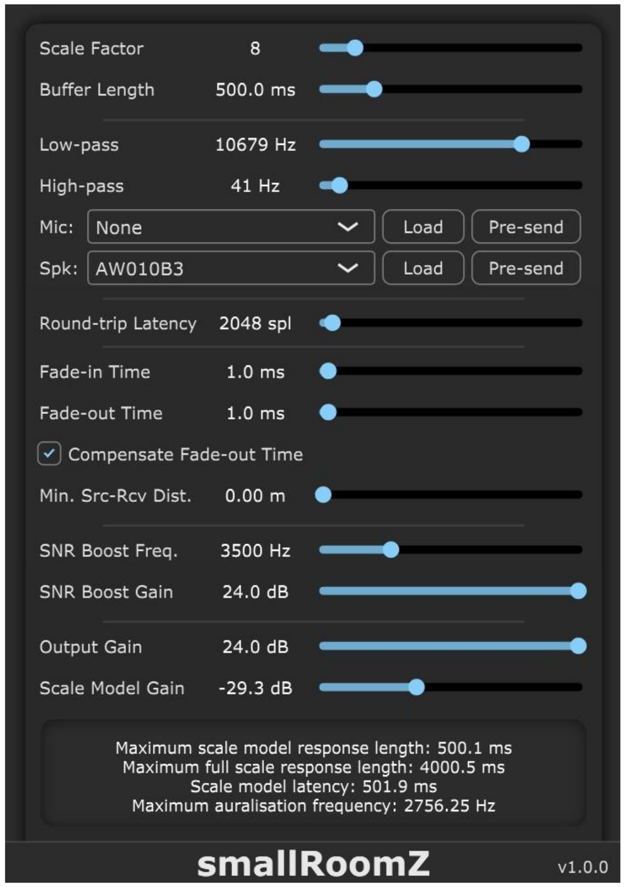
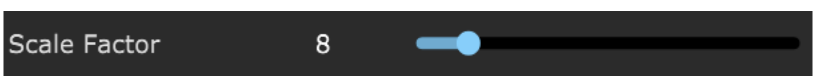
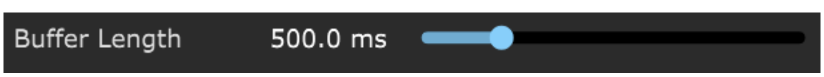


Figure 2 - The user interface of the smallRoomZ plug-in.

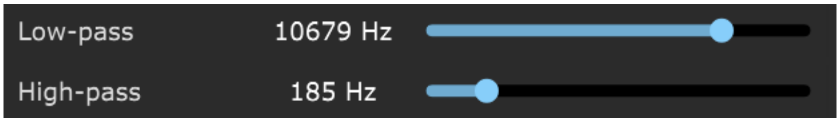
User Controls



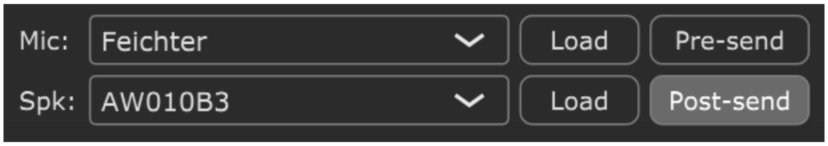
Scale Factor: Set the (integer) scale factor of the scale model being used.



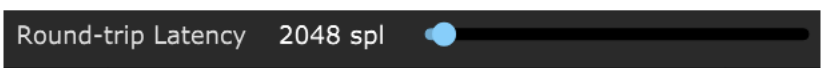
Buffer Length: The length of the zero-padding applied to the down sampled audio buffer. This should be long enough to capture the entire reverberation tail of the scale model, along with source-to-receiver propagation delay. To minimize the latency of the auralisation, this parameter should be as low as possible while still capturing the full reverb tail of the scale model.



Low-/High-Pass: Low- and high-pass filters applied to the final auralisation to remove unwanted frequency regions.



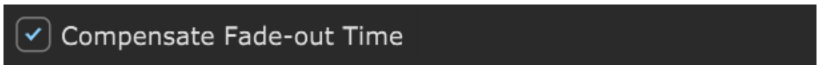
Microphone/Speaker EQ presets: Selection of the microphone and speaker EQ presets to be applied to the signal returning from the scale model. The *Load* button will open a window that allows you to select an FIR correction filter stored in a mono .wav file. The *Pre-send/Post-send* option sets when the correction filter is applied. Depending on the equipment being used it might be better to use one or the other. For example, if the audio interface introduces noise in the high frequency range, then it might be better to apply the correction filters pre-send, so that this noise is not boosted. On the other hand, if the correction filters lead to large boosts in the signal level at certain frequencies, leading to clipping or low SNR, the post-send option might be the preferred option.



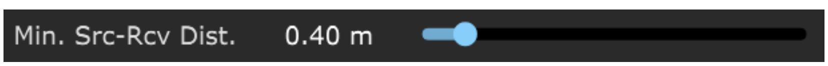
Round-trip Latency: The delay for a signal to output and then return introduced by the audio interface. It is primarily governed by the I/O buffer size B specified in the audio driver settings, but can include additional hardware-related latency beyond waiting for the buffers to be filled. It should be measured using a loop-back technique and supplied in samples. Regardless of the user input a value of $2B$ (theoretical round-trip latency excluding hardware-related delay) is used internally. The round-trip latency is used to initialise the read-heads of the overlap-add processing such that a read-head reads the return from the scale model at the correct time, ensuring minimum latency. RTL can be measured using the free RTL Utility program: https://oblique-audio.com/rtl-utility.php.

Fade-In Time: The length of the fade-in window used to suppress transient noise due to the presence of the background noise.

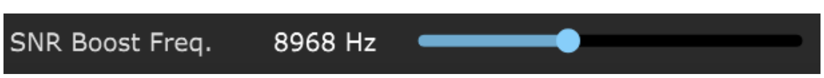
Fade-Out Time: As above, but for the fade-out time.



Compensate Fade-out Time: If this parameter is activated then the length of the fade-out is added to internal buffer used for recording the scale model signal. This is to ensure that the fade-out is not applied to the desired portion of the reverberation tail specified by **Buffer Length**, causing a truncation of the reverberation. Disabling this parameter reduces the latency but may introduce audio artefacts due to the premature fading of the reverb tail.



Minimum Source-Receiver Distance: This specifies the closest distance the source will have to the receiver in the scale model in metres. If this distance is known, then latency is minimised by applying as much of the fade-in window during the propagation time as possible. If a source-receiver distance greater than zero is specified and the source is brought closer than this distance, the resulting fade-in window will be applied to the direct sound, creating artefacts in the auralisation.



SNR Boost Freq.: The shelf-frequency of the filter used to boost the output to the scale model and to cut the return. This can be used to increase the SNR in frequency ranges where the audio interface has significant noise.

SNR Boost Gain: The amount of boost and cut applied by the shelf filters above the SNR Boost Freq. for the scale model send and return signals respectively.

Output Gain: The amount of gain applied to the final auralisation.

Scale Model Gain: The amount of gain applied to the signal sent to the scale model.