WAVE FIELD SYNTHESIS
USER MANUAL

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Welcome.

Here is some information to help you get started with this wave field synthesis system built essentially with *Max*.

Today wave field synthesis (WFS) is possible with affordable tools. All you need is a rather powerful computer and multichannel digital audio interfaces.
This technique allows the whole audience to hear each sound source in the PA as if coming from the same location independently from their listening point.
It also opens new fields in sound design for stage productions since you can play with sound illusions.
This system's functionalities have been designed specifically for live stage work:
- sound reinforcement of live sources present on stage (voices, Foley sound and musical instruments);
- playback of recorded audio tracks, effects (reverbs) or sound synthesis.
The system's algorithm was designed for a frontal or circular stage with the speakers at the edge of the stage.
Since 2018/02/21 there can be speakers all around the audience and have only sources coming through speakers that are in between them and the listening area.

Notes regarding the license: all the tools presented here are under a BSD license. It allows you to copy and give the different files as long as you keep the license file with them and you cite the names of the authors.

Liability: These tools are made available for free. The authors are not liable for any problems in their use or in their compatibility.
You may contact the authors for a paid training in wave field synthesis and the present tools if you wish.

You will need a copy of *Max7* or *Max8* by *Cycling74* on a computer running *Windows* or *MacOS*. You have a one month free trial period for this software. After this if you don't have a license the main limitation is that you will not be able to save your patches after modification.
You will also need *Java "JRE"* (64bit if you are running a 64bit copy of *Max*).

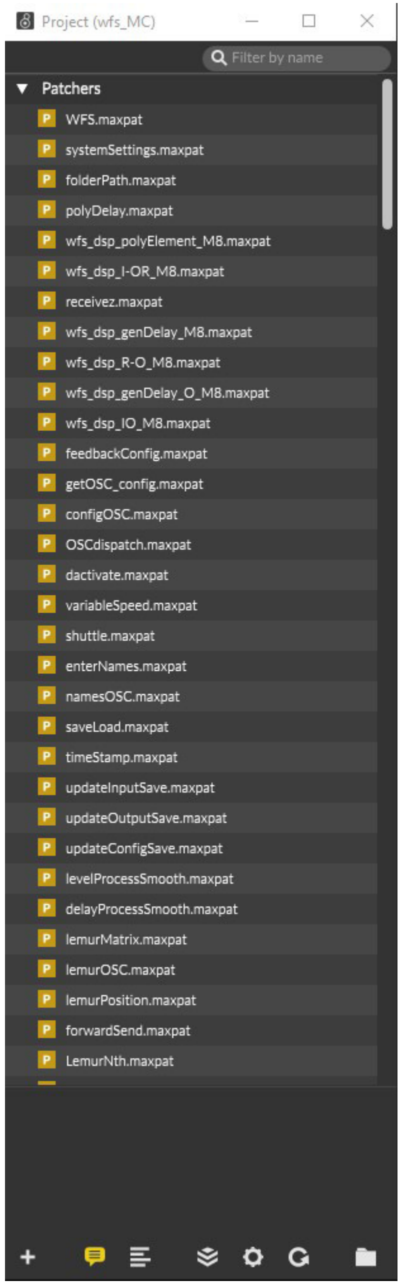
It is recommended you have *Lemur* by *Liine* for manual control of the sources' position. This application is available on *iPad* and *Android* tablets. Your computer will have to be on the same network as the tablet.

Available are also some macros to create OSC cues in *QLab* by *Figure53* to control the wave field synthesis system with cues. These macros come also as applications that can be triggered by a programmable keypad such a the *Stream Deck* by *Elgato*.
You will also find a *Pure-Data* patch to interface the system with a *ShuttleXpress* hardware jog/shuttle by *Contour* and a *Max* patch to convert *UDP* to *TCP* if required.

Once all these programs are installed and configured on your computer you will need to load either *WFS.maxproj* or *WFS.maxzip* projects in *Max*.
After this you can launch in the list to the right *WFS.maxpat*.
It may take a little while to open the patch since the program will need to rebuild itself according to the number of channels specified.

Notes on the configuration of *Max*: In the program's preferences you will have to push *Poll Throttle*, *Queue Throttle* and *Redraw Queue Throttle* to rather high values (2000 for instance) so you will not suffer from dropped control information.
If you get audio drop-outs when you change values in the interface it is possible to split the interface from the audio rendering on separate computers with a network connection.

The next page is an overview of the interface presenting quickly the different parts. The following pages are a more in-depth description of the general configuration, output (speakers) and input settings (microphones, synthesis and recorded soundtracks).
After this you will have an exhaustive list of all OSC methods to remote control the system through the network.



INTERFACE OVERVIEW

Settings for the selected input: specific delay, attenuation, wave front curvature only (minimal delay), control mode, position, height factor, maximum speed, full bandwidth distance attenuation, level maps (level, height, high frequency damping), live source damping when close to an output.

Delay, HF attenuation and level representations for the selected source. These values can also be edited manually.

Mute for each individual output for each source. Macros for quick mute settings (all, invert, even or odd channels, first or second half).

Processing configuration: input and output channel counts, number of computing threads (CPU cores), interpolation algorithm for variable delay lines. Speed of sound, Haas effect, global system latency, master level. Stage dimensions and origin.

Select the input you want to view and modify its settings.

Select the reverb you want to view and modify its settings.

Select the output you want to view and modify its settings.

Movement generator for each input: offset position, single straight/curved movement, periodic or random movements, spinning.
To the right speed, stop, pause/resume controls.

Store and recall settings (general configuration, output and input settings).
Revert to the last settings from the auto-save back-up system.

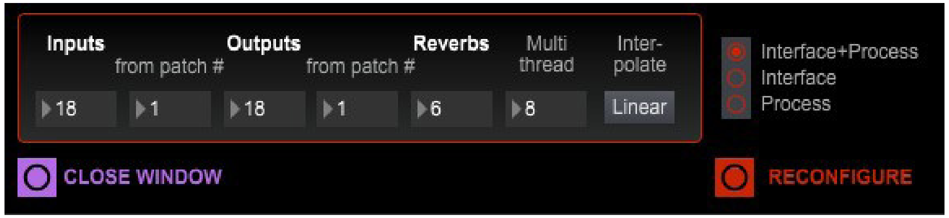
Name editing interface for outputs and inputs.

Control mode for each input: direct setting of delays and levels in the matrix, manual position setting, corresponding number of the *Lemur* interface marker. More than one input can be linked to a single *Lemur* marker.
Flipping of input coordinates regarding stage origin.

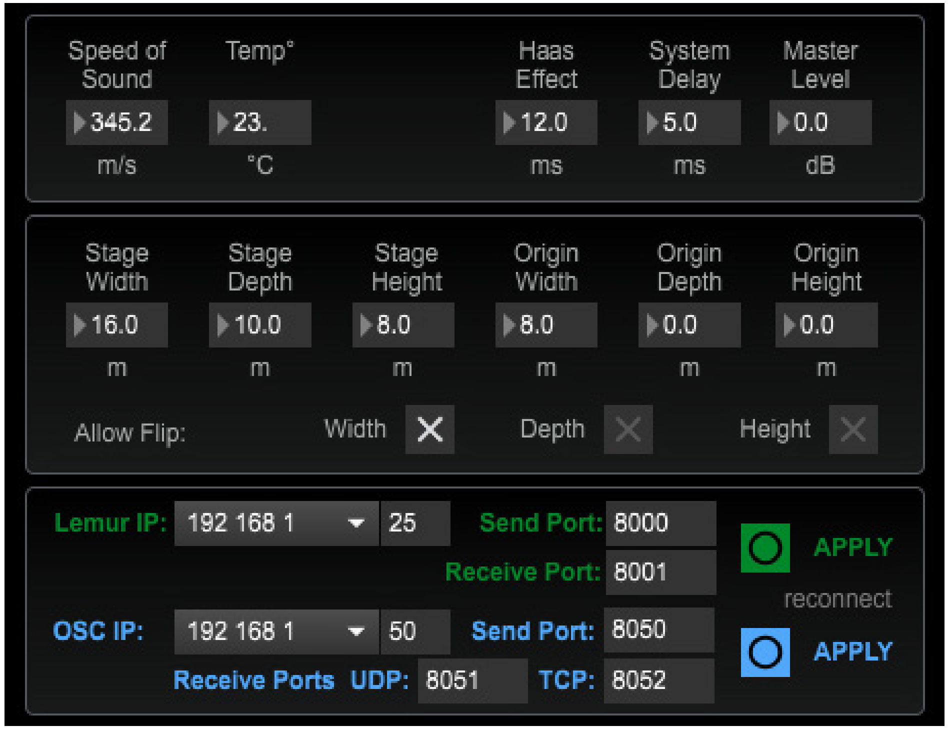
Settings for the selected output: specific delay, level attenuation, speaker position and high frequency damping per distance and eq/filter.

in gray are the corresponding page numbers

Settings for the process can only be edited by first clicking the orange button label *EDIT*. You will then get a pop-up window where you can make all the necessary changes.



Input and output channel counts with the number of the first channel in the audio preferences of Max to ease the patching on audio I/O. The number for feed and return channels for reverbs.
Multithread : number of logical cores of the CPU available for multithreaded processing.
Interpolate : type of interpolation for the variable delays (linear or cubic).
In case of modifications you will need to click the red button labeled *RECONFIGURE* to apply the changes. If you wish you may off-load the interface from the computer running the audio process. In this case the interface will run on a separate computer connected over the network.
Choose either *Interface+Process* for an all-in-one system ; *Interface* for the computer running the user interface and send controls over the network according to the sources' position; *Process* for the computer running the audio rendering which will receive the commands over the network. The interface computer will find the process computer by itself.
In case of modifications you will need to restructure the patch by clicking the red button labeled *RECONFIGURE*.
Values are stored and recalled next time you open *wfs.maxpat*.



Speed of Sound and *Temperature* will set the speed of sound. It is mostly important when dealing with sound reinforcement of acoustic sources on stage (voices, musical instruments...).

Haas Effect will set the delay you wish to apply to keep the amplified sound always behind the acoustic sound to take advantage of the precedence effect to help localization. It also will also allow to have negative delays on some inputs or outputs that you wish to delay less than the rest of the system.

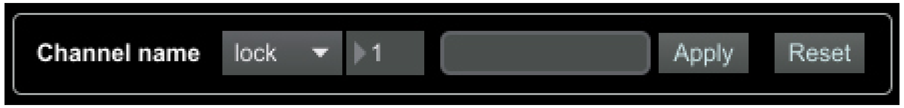
System Delay will allow to take into account in the delay calculations of the latency of the process: console, sound card and WFS process latencies mostly.

Master Level will set the audio output level.

Settings for stage dimensions and origin of the stage. All input and output positions will be given from this point.
Allow Flip tick boxes let you decide if you globally allow sources to have a symmetrical position regarding the point of origin. This can be handy for instance to handle linked inputs on a single *Lemur* marker but in mirror positions on either side of the stage.

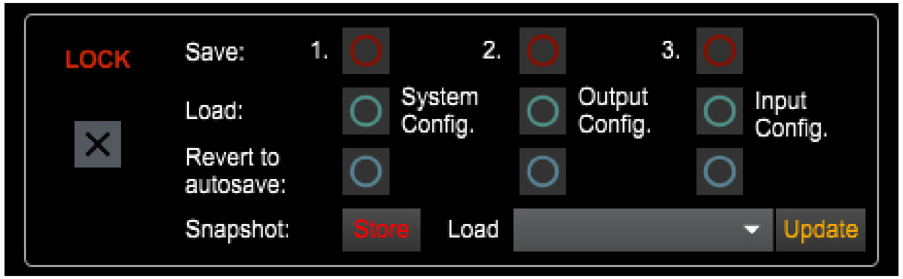
Settings for the *Lemur IP* and send and receive ports.
Settings for the *OSC IP* and send and receive ports (UDP and TCP).
In case of a lost connection press *Apply* again to reconnect.

You can set the names of the outputs with the interface below the general configuration and *Store* and *Recall* controls.



To give a channel a name, start by clicking *lock* and choose whether you want to name an *input* or an *output* Then select the channel number. You will see the current name of the selected channel in the text box. Make the necessary changes and click *Apply*.

To revert to the default name select *input/output* then the channel number and click *Reset*.



Store and recall settings for system configuration, outputs and inputs.

Save and *Load* will store and recall base settings.

A back-up system will store automatically all settings after each modification.

Revert to autosave will recall these settings.

Snapshot will *Store* and *Load* time-stamped input settings. This can be done through OSC commands too.

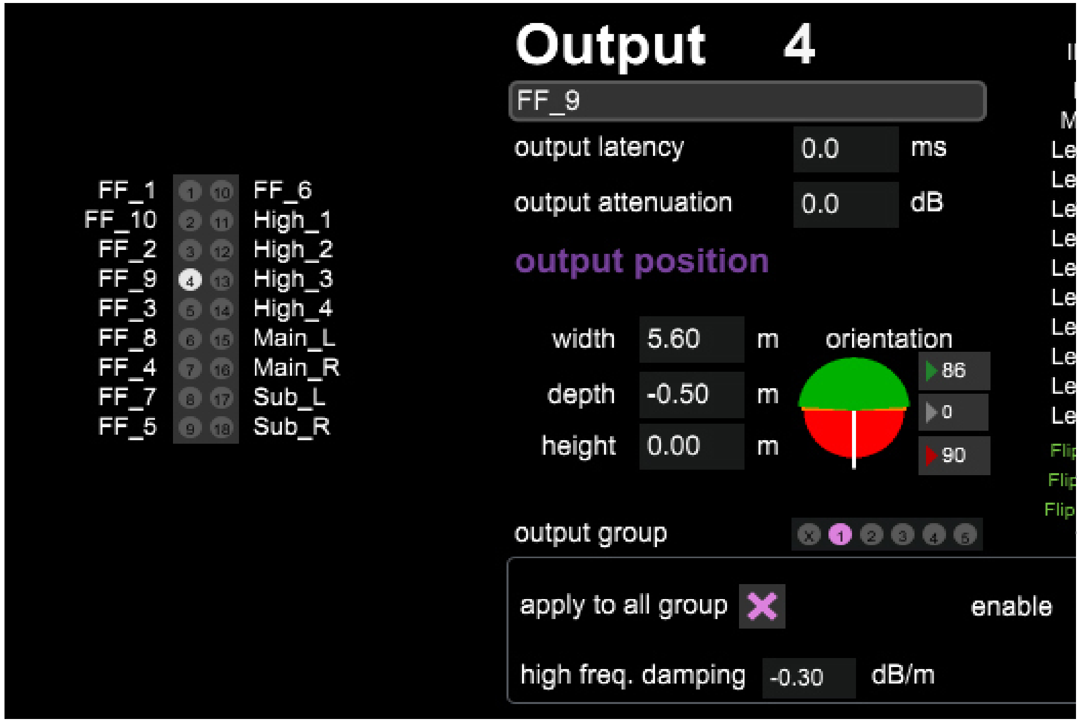
Update will store the input settings in the last loaded or created snapshot file.

Lock will disable saving and recalling.

Activates audio processing and puts the interface in fullscreen. The button will be locked. You need to click *EDIT* to unlock it to turn off the audio engine and come out of fullscreen. To avoid having the interface in fullscreen you can start the audio processing from the *Max* menu: *options/Audio Status...*

OUT PUT SETTINGS

Select the output for which you would like to see the settings and eventually change them.
The first half of outputs are in the left-hand side column and the rest on the right. The first channel at the top and then down.
You can jump to any input by pressing **o** and typing the channel number then **Enter**.



Specific setting for output channel latency. This can be used for example to compensate for the latency of a digital amplifier.

A *negative latency* can be used to <u>increase</u> the delay for the selected output. This can be useful to time align a speaker regarding other speakers of the PA.

Level attenuation of selected output.

Position of the speaker in relation to the stage origin as defined in the general configuration.

White mark: orientation of the speaker in the horizontal plane.

Green "On" sector: sources located in this sector will contribute to this output.

Red "Off" sector: sources located in this sector will **not** contribute to this output.

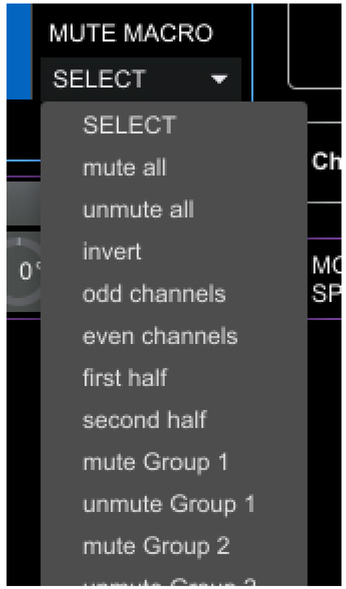
Orange sector: fade in between on and off sectors.

High Freq. Damping: This mimics the attenuation of high frequency by air.

This setting has been placed in the output speakers for the reason that if you used a general or per input settings it would be counter productive for delay speakers that are placed far from the stage to try and relay the high frequency since you would have needed to boost the highs after the high frequency filter.

However it is possible to work on high frequency filtering on a specific level map in relation to the position of the source on stage.

Speaker groups for fast muting and unmuting.



Enable Minimal Latency: the output will be part of the pool of channels that will be polled for shortest delay and have this delay subtracted for each input with *Curvature only* engaged.

Enable Live Source Attenuation: the output will be affected by the local attenuation for inputs in range when their corresponding setting is engaged.

% Distance Attenuation: the calculation of the attenuation based on the distance to each input has a specific ratio between 0% (no level attenuation) to 200% (twice the nominal attenuation).

You can set the names of the outputs with the interface below the general configuration and *Store* and *Recall* controls.

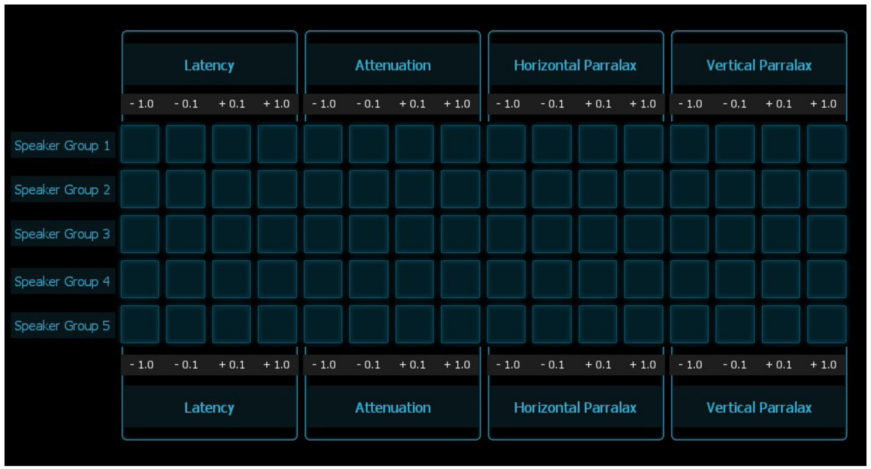
PARRALAX CORRECTION

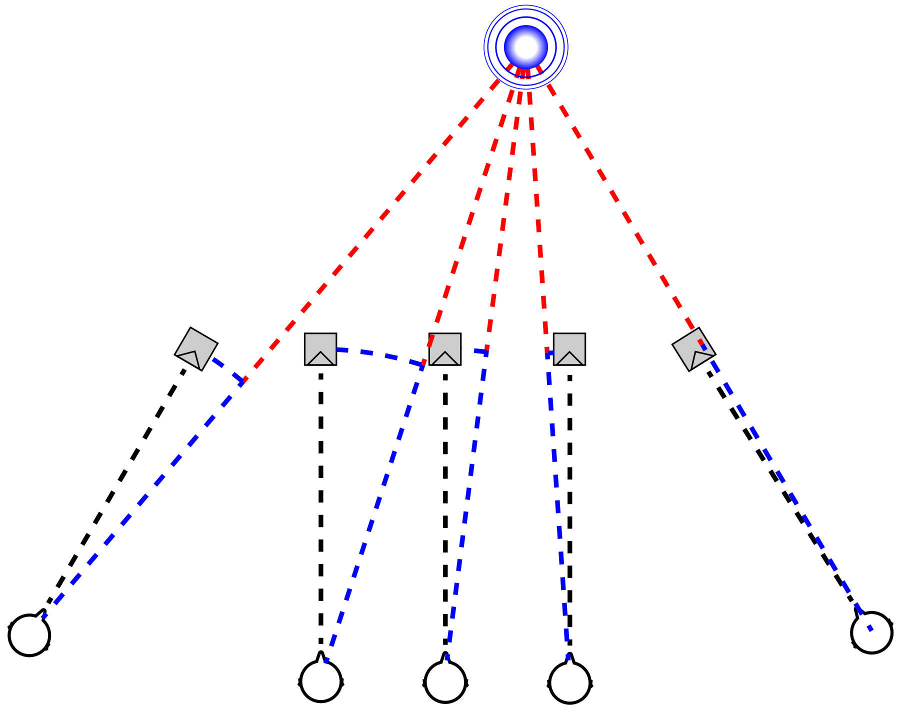
If the delay calculation only took in consideration the position of the speaker it would mean that the sources, speakers and listeners ere aligned. This is very hardly the case in a regular venue with the width of the rows and the tired seats or a large attendance which requires a flow array of speakers above the stage.
In order to compensate for the difference of time the (virtual or real) direct sound has to travel from the source to the listener and the position of the speakers, we have implemented a set of calculations: for each speaker we define a *first listener* on axis in the vertical plane. We calculate for each speaker the distance from each source to this *first listener* position. We then subtract the distance from the speaker to the *first listener* (black and blue dashed lines). This gives the delay time for this source through that speaker (red dashed line). The vertical offset will be affected by the *height ratio* of the input.

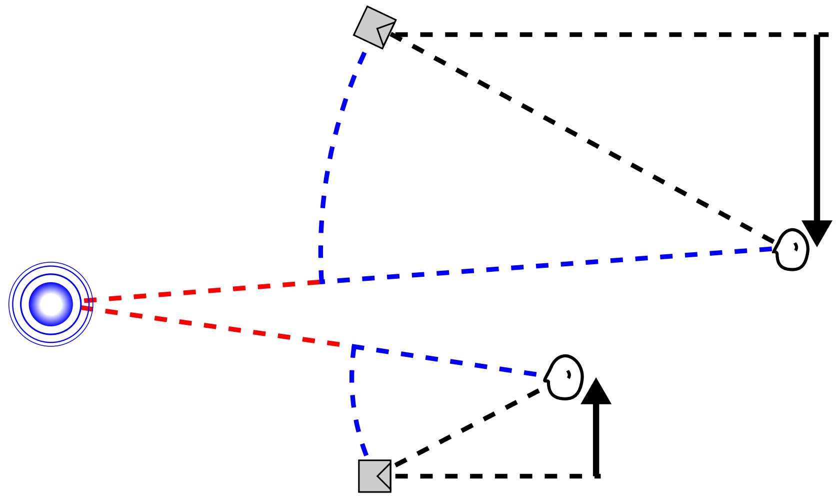
This should limit the parallax for the listeners. But in certain situations the resulting delays are much less than without this compensation. This may result in some coupling with strong high frequency boost when in line with the speakers and a blur in the localisation of the sources. Always check with your ears and adapt the *height factor* and the *parallax correction*!

Horizontal distance is always positive and should be taken on axis from the speaker.
Vertical distance is positive when the speaker is below the listener and negative when above the listener.

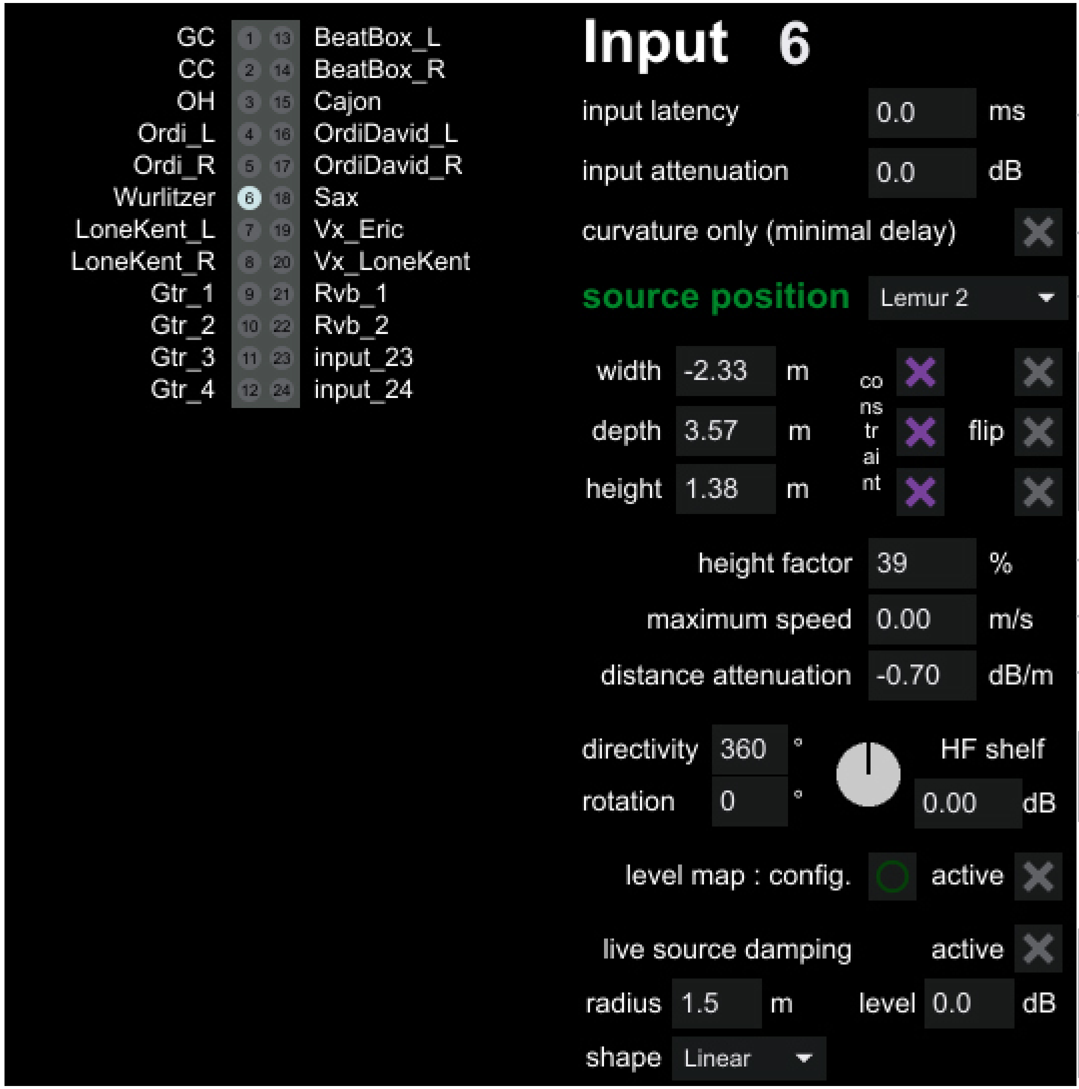
Output latency, attenuation, horizontal and vertical parralax can be adjusted for each speaker group from the Lemur interface to make it easier to adjust the relative settings of the different speaker arrays while listening in the house.







Select the input for which you would like to see the settings and eventually change them.
The first half of inputs are in the left-hand side column and the rest on the right. The first channel at the top and then down.
You can jump to any input by pressing **I** and typing the channel number then **Enter**.
You can scroll through inputs with the following keys **Space bar** and [shift]+**Space bar**.



You can set the names of the inputs with the interface below the general configuration and *Store* and *Recall* controls.

Specific setting for input channel latency. This can be used for example to compensate for the latency of a digital wireless microphone or some other specific processing.

A *negative latency* can be used to increase the delay for the selected input.

Level attenuation of selected input.

Curvature only : Normally the delays are calculated from the distance between a source and each output speaker. Alternatively you can decide to only work with the curvature. The smallest delay will be subtracted from all others.

Source position sets the control mode for the selected channel. *see next page*

Source position. You can use keyboard arrows (*left*, *right*, *up*, *down*) to move the source in width and depth and *page up* and *page down* to move the source in height. [shift] for larger steps (1m) and [ctrl] for smaller steps (1cm).

Constraint bounds the source to the stage.

Flip places the source in the symmetrical position relative to the stage origin.

Height factor sets how much weight height has in the distance calculations. This is to avoid disrupting the PA setup since distances between a source and high and low speakers will change differently when a source moves upstage at a constant height.

Maximum Speed sets a speed limit for the selected source. Above this limit the algorithm will keep the source moving as long as it takes to reach the target position with a smooth movements to avoid sharp accelerations and limit changes in the *Doppler* effect as much as possible.

Distance Attenuation sets the attenuation in dB/m. The idea here is not to match the physical reality since if the source is far from a speaker it will probably need some reinforcement. This settings helps focus a source in the WFS system and give a sense of depth when the source moves away from a speaker.

Directivity, rotation and *HF shelf* enable to fade the HF of a source facing away from the speakers such as when a comedian turns his back to the audience.

Level Map gives a control over the audio level, height and high frequency filtering in relation to the position of the selected source on stage. *see page Levelmaps*

Live Source Damping allows to lower the amplification of a live source on stage as it gets closer to a speaker to avoid over amplifying a loud source or to avoid feedback.
radius sets the influence radius for the selected source. Speakers located at less than this distance will be attenuated. Height is taken into account for the distance calculation whatever the *Height factor* setting.
level sets the maximum attenuation for the selected input when the positions of the source and speaker match exactly.
shape sets the profile for the attenuation when the source gets close to a speaker. Select either linear, log, square x², sine.

The *source position* menu defines the control mode for each source:
Matrix, *Manual*, *Lemur 1 - 0*.

You can assign the control mode directly from the keyboard:
- numbers from **1** à **0** assign the corresponding *Lemur* marker.
- **²** (to the left of the numbers) assign *Manual* mode.
- **°** (to the right of the numbers) to assign the previously assigned *Lemur* marker.

You can change the selected channel with keys **Space bar**(next input) and **[shift]+Space bar** (previous input).

The table shown below summarizes all the assigned control modes. It can be edited directly by clicking the appropriate cells.
You have the mirror position settings there too and you can change them there as well.

Matrix mode means that you use the delays and levels from the yellow and blue sliders in the input settings.

Modifying these sliders will assign the *Matrix* mode to the selected input. You will need to reassign any *Lemur* marker manually after this.

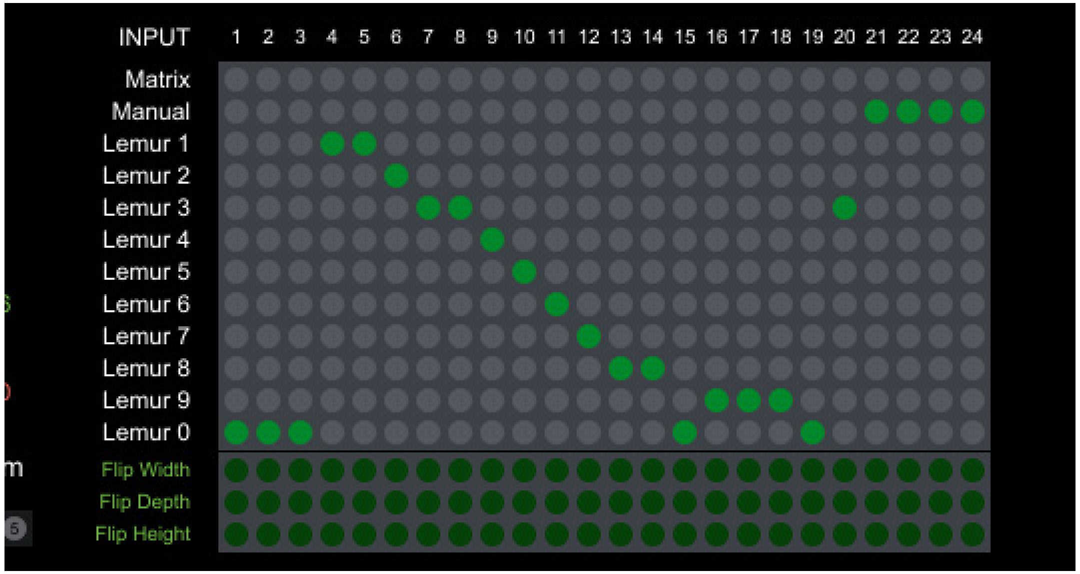
Manual will use the position entered through the user interface to calculate delays and levels. It will not react to any *Lemur* marker position.

When you assign a *Lemur* marker not yet assigned to any other channel, the marker will move to the currently selected input position.

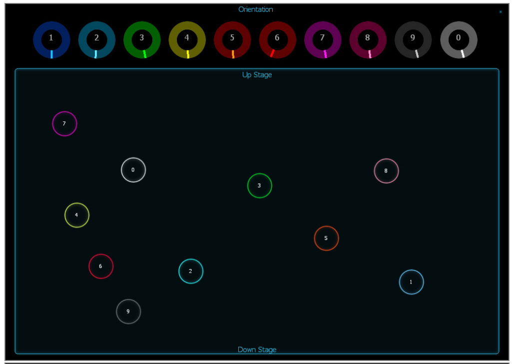
When you assign a *Lemur* marker already assigned to another channel the source's position will be updated to the position of the marker and other linked inputs.

The rectangle on the *Lemur* interface corresponds to the stage dimensions.

You can change the position of a source through the patch interface or keyboard arrows even if it is assigned to a *Lemur* marker. The *Lemur* marker position will be updated as well as any other source linked to the same *Lemur* marker.

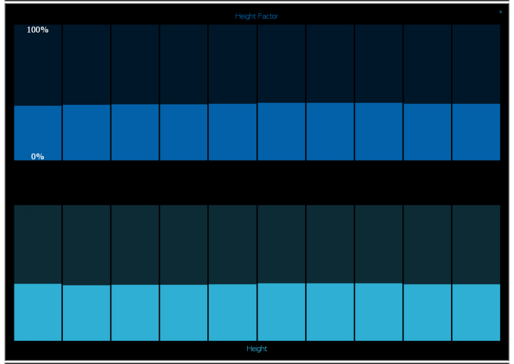


Lemur interface on the tab for width and depth position of the markers according to the full width and depth of the stage as defined in the configuration. Orientation of the sources on top or on the side depending on the aspect ratio of the tablet.

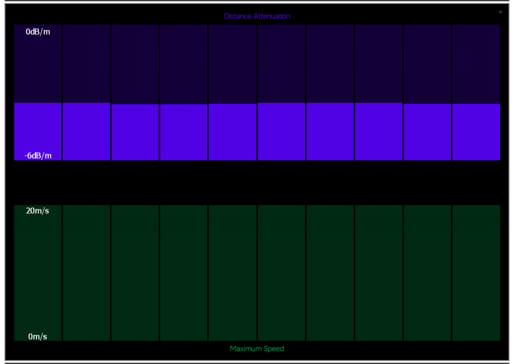


Lemur interface on the tab for marker height (bottom) ranging from 0 to the full height of the stage as defined in the configuration.

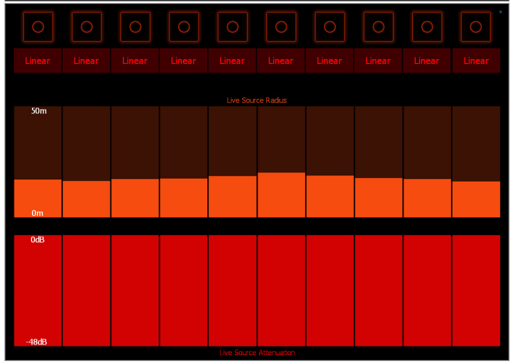
The parameter at the top are the height factor in the distance calculations (ranging from 0 to 100%).



Lemur interface on the tab for distance attenuation and maximum speed. Settings range from 0dB/m at the top to -6dB/m for distance attenuation and 0m/s (unlimited) and 20m/s for the maximum speed.



Lemur interface on the tab for live source damping. Settings are on/off; shape of attenuation profile; radius from 0m to 50m; attenuation range from 0dB at the top to -48dB.



You can use the position of a source on stage to change certain parameters automatically: audio level, height or high frequency damping.

These parameters will be respectively combined with the calculated audio level, height and high frequency damping proportional to distance set for each output.

Application examples:

For sound level you can use this feature to cut out a microphone when a comedian wearing a wireless microphone goes offstage.

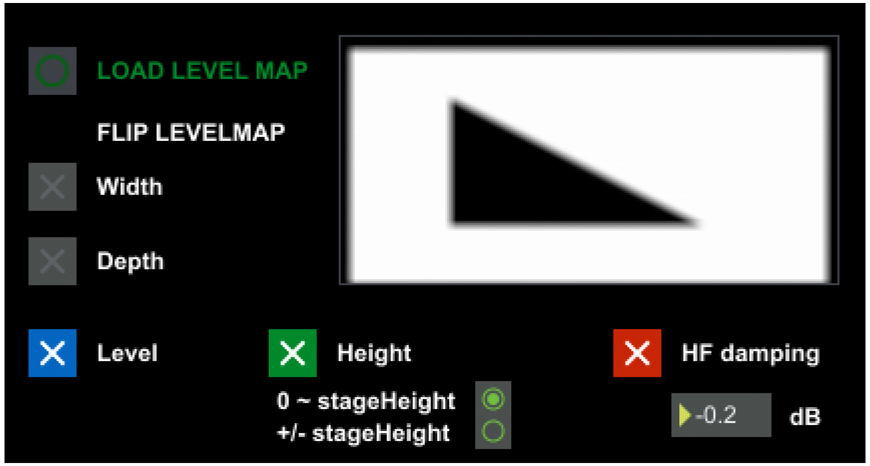
It is also possible to define different zones on several channels to mute and unmute different channels. This can be used for instance to mimic the acoustic characteristics in different rooms of a house when a comedian moves around the stage.

Height can be used to compensate the impression of sources rising when going upstage if using a high *height factor*.

Height can also be used to match automatically the different heights of a set when the source moves around.

High frequency damping can be used to emphasize the impression of distance when a source moves away upstage.

The green button labeled *config.* will open a specific pop-up window for further settings.
The green cross labeled *active* will enable or disable all levelmaps on the selected channel.



LOAD LEVEL MAP will load a picture with the different map each in a colour.

Flip levelmap Width/Depth will allow you to have a mirror image of the level map image in width or depth. Each has an independent toggle.

The blue layer will be used for audio level. Blue will be maximum attenuation (-inf) and white is unity (0dB).

The green layer is used for height. There are two options: either height ranges from ground green) to full stage height (white) or from plus or minus the whole stage height (from green to black).

The red layer is used for high frequency damping. There is a setting for maxi-mum attenuation assigned to white. Red will be no attenuation.

LIVE SOURCE DAMPING

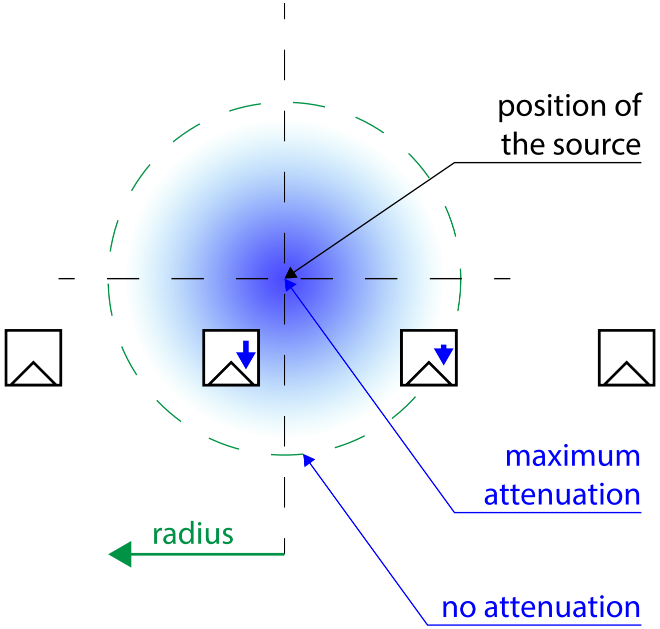
Live Source Damping affects the contribution of an input to outputs located near it.

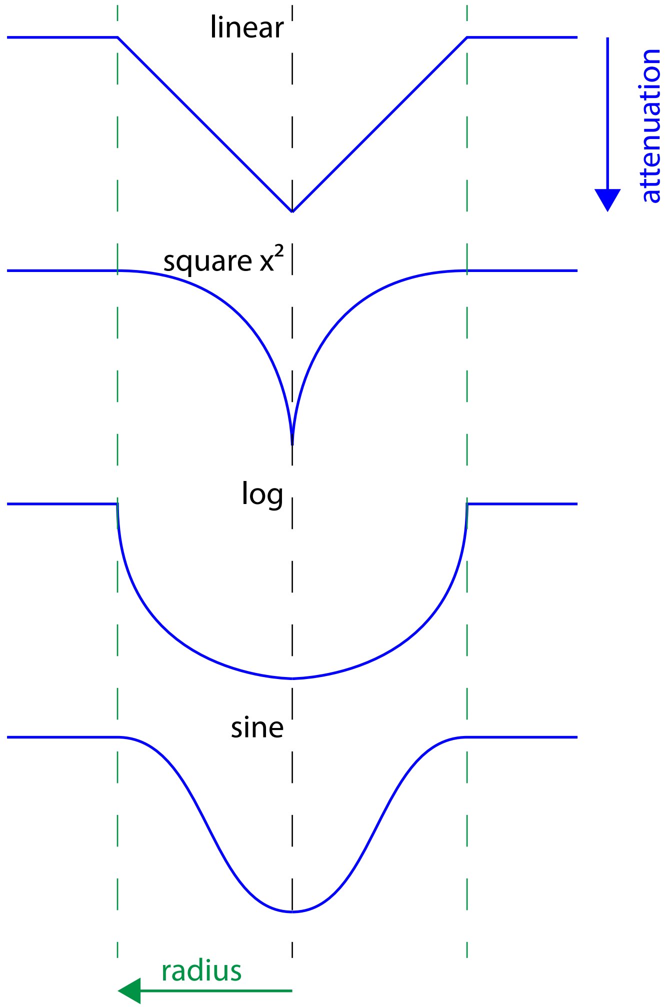
The *radius* determines the influence in terms of distance. When Live Source Damping is active on an input and if an output is located at less than the radius from it, the level sent to this output will be attenuated.
Distance between the input and output in this case will always take elevation in its calculations.

The *attenuation* setting is the maximum attenuation when the speaker is exactly at the speaker position. Speaker directivity may cut out the sound too in this case.

The *shape* setting determines the profile of the level attenuation with distance:

- *linear* is very progressive;
- *square (x^2)* has a pronounced level dip very close to the source;
- *log* has a pronounced level drop as soon as the output enters the attenuation zone;
- *sine* is close to linear but with smoother changes close to the radius and center.





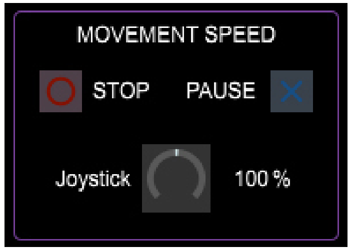
Application examples:

Live source damping is helpful when a source on stage is already loud enough and doesn't need to have any reinforcement in its vicinity. Imagine you have an opera singer close to the edge of the stage. He's probably more than loud enough for the people sitting nearby. In this case it would start to get really uncomfortable for the audience to have even more of the singer's voice in the front fills.

An other situation where this can prove handy is if the musicians complain that there is too much sound coming from the rear of the speakers sitting in front of them at the edge of the stage.

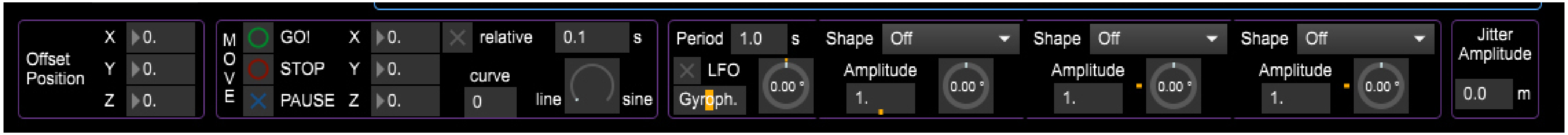
It is possible to move a source from one point to an other or to give them periodic or random trajectories.
The controls are located in the middle of the interface. They correspond to the currently selected input.
The functions can be remote controlled by OSC commands sent by an other software such as *QLab* by *Figure53*.

Movement Speed
STOP will halt all sources in their current position.
PAUSE will temporarily halt all sources where they are until they resume their movement.
Joystick allows to speed up or slow down the movements of all sources currently moving. When all sources have finished moving this dial comes back to nominal speed (100%).
These actions have no incidence on *LFO, Jitter* or manual movements

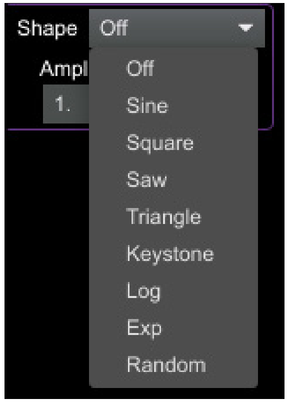


Offset
Will shift the position of the input by a certain amount regarding the position.
This can be useful if several sources are assigned to the same Lemur marker to move together but have to maintain constant relative positions and simplify controls to only one marker for all the linked sources.

Jitter
Gives a random movement to the selected source according to the amplitude setting.
The source will move fast in a random fashion in all three dimensions. These quick movements will create increasingly more Doppler effect as the amplitude is increased.



Move
This will make a source follow a trajectory in space.
GO will start th movement for this source, *STOP* will stop it where it is, *PAUSE* will halt it until resumed.
X, Y and *Z* set the destination coordinates. If *relative* is marked then it defines the amount to move relative to the position at the beginning of the movement. If *relative* is marked then it they define the absolute position relative to the stage origin.
Curve defines the curvature of the movement. When at 0 then the source will move in a straight line. For positive values between 1 and 100 the trajectory will curve upstage. For negative values between -1 and -100 the trajectory will curve downstage.
Then you have the time in seconds for he movement.
The *line/sine* setting allows you to choose between constant speed (*line*) and a smooth start and finish (*sine*). In this case the maximum speed will be greater than at constant speed, but there will be no sudden change in speed at the start and stop that gives noticeable Doppler effect.
You can dial any intermediate value depending on the type of audio material and how sensitive it might be to changes in pitch.



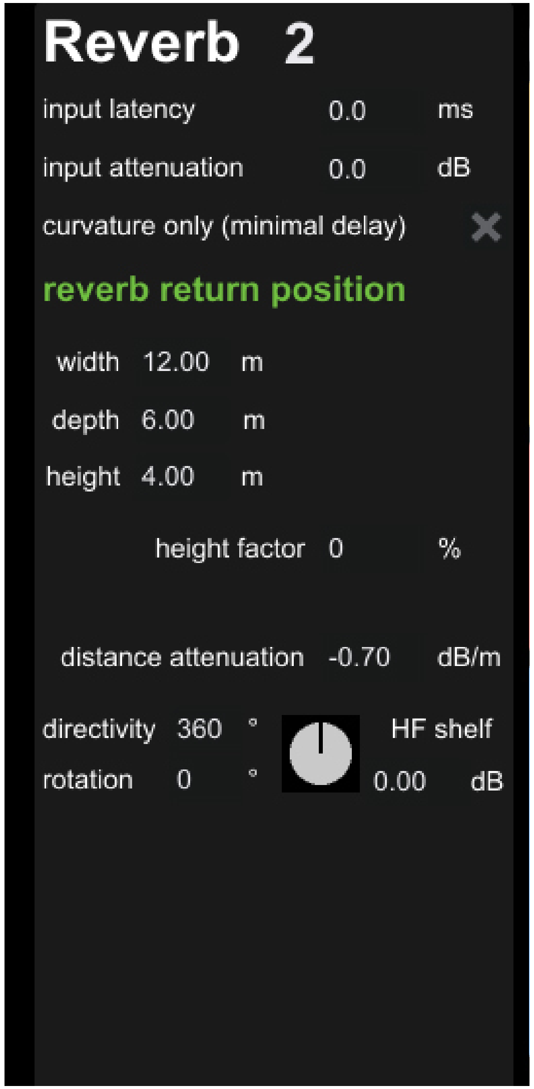
LFO (Low Frequency Oscillator)
This will give periodic movement to a source. There is a main oscillator principal that will be used to create the different shapes for each dimension. This oscillator has two settings: its period in seconds and its phase (from 0° to 359°) in case you have several sources at the same speed but shifted.
Gyroph. will make a source's directivity spin like the horn of a Leslie.
For each dimension there are several choices of wave shapes, amplitude and phase settings.
The wave shapes are:
Off no oscillation;
Sine standard sine curve (used to draw circles or ellipsoids);
Square alternates between two values;
Saw will rise progressively and the return suddenly to the original value and rise again;
Triangle rises and falls progressively between two values;
Keystone rises and falls progressively between two values but maintains each value before rising or falling again (used to draw squares and rectangles) ;
Log rounded off saw tooth;
Exp rounded off saw tooth;
Random changes randomly value for each cycle of the main oscillator.

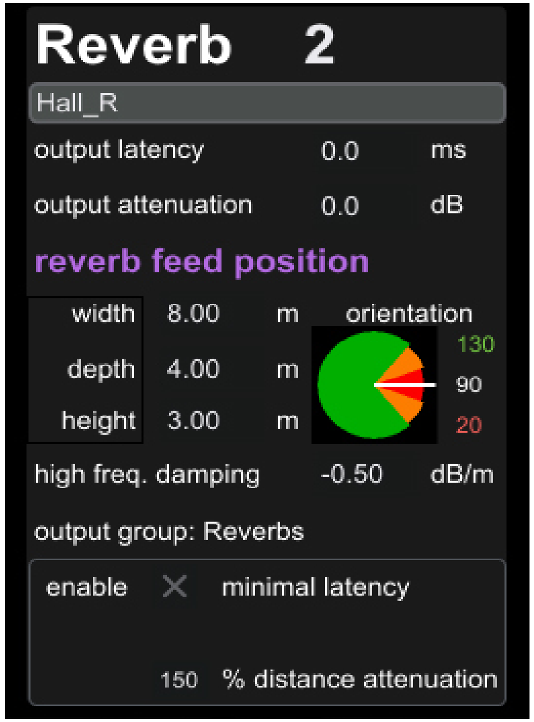
13 REVERB FEEDS AND RETURNS

The WFS processor can be used to mix the reverb feeds depending on the position of inputs.
The reverb return can also be played back through the WFS system.
Currently the system does not generate the reverb effects internally and will be sent to a separate processor (in console effects, out board effects unit or other computer application). This may be included in a future version.

Our hearing is very sensitive to early reflections and the ratio between the the direct sound and the reverberation to locate a sound source. With the intensity of the reverb and the different timings it is possible to give a more tangible presence to playback tracks or synth sounds.

Reverb feeds have similar settings to normal outputs. There are latency and attenuation, position and orientation, high frequency damping. You can adjust the distance attenuation factor and enable the feed to be part of the minimal latency outputs (this setting is off by default).
Reverb feeds are all in mute group 6 and cannot be taken out of this group nor other regular output channels added to it.





Reverb returns behave like other inputs, but with limited settings.
They cannot be assigned to any reverb feed by design to avoid any feedback issue.
There are latency, attenuation, position and directivity settings as well as height factor and distance attenuation. Minimal delay can be engaged if necessary.

default receive port: UDP 8051 / TCP 8052
default send port: 8050

it is possible to change to IP address and port directly through OSC. A confirmation will be necessary within one second after the new setting:

/wfs/config/OSChost [i i i i] configures the destination IP address where to send the OSC commands from the WFS system.

/wfs/config/OSChost/confirmHost confirms the new IP address

/wfs/config/OSCport [i] configures the destination port where to send the OSC commands from the WFS system.

/wfs/config/OSCport/confirmPort confirms the new port.

get: request to send back the current value once.
stream: request to send a continuous stream of the value.

input or output channel or *all* for all inputs or outputs
[i] integer number
[f] floating point number
[0/1] 0 : stop/off / 1 : start/on
[string] string of characters

CONFIGURATION

/wfs/config/stageWidth [f]
/wfs/config/stageDepth [f]
/wfs/config/stageHeight [f]
/wfs/config/stageDimensions [f] [f] [f]
/wfs/config/originWidth [f]
/wfs/config/originDepth [f]
/wfs/config/originHeight [f]
/wfs/config/originPosition [f] [f] [f]
/wfs/config/flipX [i]
/wfs/config/flipY [i]
/wfs/config/flipZ [i]
/wfs/config/flipXYZ [i] [i] [i]
/wfs/config/speedOfSound [f]
/wfs/config/temperature [f]
/wfs/config/HaasEffect [f]
/wfs/config/globalLatency [f]
/wfs/config/masterLevel [f]
/wfs/config/OSChost [i i i i]
/wfs/config/OSChost/confirmHost
/wfs/config/OSCport [i]
/wfs/config/OSCport/confirmPort

/wfs/config/get/all
/wfs/config/get/stageWidth
/wfs/config/get/stageDepth
/wfs/config/get/stageHeight
/wfs/config/get/stageDimensions
/wfs/config/get/originWidth
/wfs/config/get/originDepth
/wfs/config/get/originHeight
/wfs/config/get/originPosition
/wfs/config/get/flipX
/wfs/config/get/flipY
/wfs/config/get/flipZ
/wfs/config/get/flipXYZ
/wfs/config/get/speedOfSound
/wfs/config/get/temperature
/wfs/config/get/HaasEffect
/wfs/config/get/globalLatency
/wfs/config/get/masterLevel

NAMES

/wfs/names/input/label [i] [string]
/wfs/names/input/reset [i]
/wfs/names/output/label [i] [string]
/wfs/names/output/reset [i]
/wfs/names/reverb/label [i] [string]
/wfs/names/reverb/reset [i]

/wfs/names/input/get all
/wfs/names/input/get [i]
/wfs/names/output/get all
/wfs/names/output/get [i]
/wfs/names/reverb/get all
/wfs/names/reverb/get [i]

SNAPSHOTS

/wfs/saveLoad/snapshot/store [string: date_time]
/wfs/saveLoad/snapshot/recall [string: date_time]

OUTPUTS

/wfs/selectIO/output [i]

/wfs/output/#/latency [f] ms
/wfs/output/#/attenuation [f] dB
/wfs/output/#/positionX [f] m
/wfs/output/#/positionY [f] m
/wfs/output/#/positionZ [f] m
/wfs/output/#/positionXYZ [f] [f] [f] m m m
/wfs/output/#/orientation [i] -180°~180°
/wfs/output/#/HFdamping [f] dB/m
/wfs/output/#/group [i] 0: off / 1~5 / 6: reverb feeds
/wfs/output/#/miniLatencyEnable [0/1]
/wfs/output/#/liveSourceEnable [0/1]
/wfs/output/#/distanceAttenuationPercent [i] 0~200%
/wfs/output/#/Hparallax [f] m >0
/wfs/output/#/Vparallax [f] m

/wfs/output/#/get/all
/wfs/output/#/get/latency
/wfs/output/#/get/attenuation

/wfs/output/#/get/positionX
/wfs/output/#/get/positionY
/wfs/output/#/get/positionZ
/wfs/output/#/get/positionXYZ
/wfs/output/#/get/orientation
/wfs/output/#/get/HFdamping
/wfs/output/#/get/group
/wfs/output/#/get/miniLatencyEnable
/wfs/output/#/get/liveSourceEnable
/wfs/output/#/get/distanceAttenuationPercent
/wfs/output/#/get/Hparallax
/wfs/output/#/get/Vparallax

REVERB FEEDS

/wfs/selectIO/reverb [i]

/wfs/reverbFeed/#/latency [f] ms
/wfs/reverbFeed/#/attenuation [f] dB
/wfs/reverbFeed/#/positionX [f] m
/wfs/reverbFeed/#/positionY [f] m
/wfs/reverbFeed/#/positionZ [f] m
/wfs/reverbFeed/#/positionXYZ [f] [f] [f] m m m
/wfs/reverbFeed/#/orientation [i] -180°~180°
/wfs/reverbFeed/#/HFdamping [f] dB/m
/wfs/reverbFeed/#/miniLatencyEnable [0/1]
/wfs/reverbFeed/#/distanceAttenuationPercent [i] 0~200%

/wfs/reverbFeed/#/get/all
/wfs/reverbFeed/#/get/latency
/wfs/reverbFeed/#/get/attenuation
/wfs/reverbFeed/#/get/positionX
/wfs/reverbFeed/#/get/positionY
/wfs/reverbFeed/#/get/positionZ
/wfs/reverbFeed/#/get/positionXYZ
/wfs/reverbFeed/#/get/orientation
/wfs/reverbFeed/#/get/HFdamping
/wfs/reverbFeed/#/get/miniLatencyEnable
/wfs/reverbFeed/#/get/distanceAttenuationPercent

INPUTS

/wfs/selectIO/input [i]

/wfs/input/#/latency [f (f)] ms / optional transfer time in seconds
/wfs/input/#/attenuation [f (f)] dB / optional transfer time in seconds
/wfs/input/#/curvature [0/1]
/wfs/input/#/control [i] 0 matrix, 1 manual, 2~11 Lemur 1~0
/wfs/input/#/positionX [f] m
/wfs/input/#/positionY [f] m
/wfs/input/#/positionZ [f] m
/wfs/input/#/positionXYZ [f] [f] [f] m m m
/wfs/input/#/constraintX [i]
/wfs/input/#/constraintY [i]
/wfs/input/#/constraintZ [i]
/wfs/input/#/constraintXYZ [i] [i] [i]
/wfs/input/#/flipX [i]
/wfs/input/#/flipY [i]
/wfs/input/#/flipZ [i]
/wfs/input/#/flipXYZ [i] [i] [i]
/wfs/input/#/heightFactor [i (f)] % / optional transfer time in seconds
/wfs/input/#/maxSpeed [f (f)] m/s / optional transfer time in seconds
/wfs/input/#/distanceAttenuation [f (f)] dB/m / optional transfer time in seconds
/wfs/input/#/directivity [i (f)] 2°~360° / optional transfer time in seconds
/wfs/input/#/rotation [i (f)] -180°~180° / optional transfer time in seconds
/wfs/input/#/HFshelf [f (f)] dB / optional transfer time in seconds
/wfs/input/#/levelMap [i:levelMapActive] [i: flipX] [i: flipY] [i:levelActive] [i:heightActive] [i:heightMode] [i:HFdampin-gActive] [f:HFdamping]
/wfs/input/#/liveSource [i f f i] active ; radius ; attenuation ; shape
/wfs/input/#/liveSourceActive [1/0]
/wfs/input/#/liveSourceRadius [f (f)] m / optional transfer time in seconds
/wfs/input/#/liveSourceAttenuation [f (f)] dB / optional transfer time in seconds
/wfs/input/#/liveSourceShape [i] 0:Linear, 1:Log, 2:Square x^2, 3:Sine
/wfs/input/#/mutes [i_list]
/wfs/input/#/delays [f_list] ms
/wfs/input/#/levels [f_list]
/wfs/input/#/HFdampings [f_list]
/wfs/input/#/muteMacro [i]
1: mute all, 2: unmute all,
3: invert,
4: odd channels, 5: even channels,
6: first half, 7: second half,
8: mute output group 1, 9: unmute output group 1,
10: mute output group 2, 11: unmute output group 2,
12: mute output group 3, 13: unmute output group 3,
14: mute output group 4, 15: unmute output group 4,
16: mute output group 5, 17: unmute output group 5,
18: mute output group 6 (reverb feeds), 19: mute output group 6 (reverb feeds)

/wfs/input/#/get/all
/wfs/input/#/get/latency
/wfs/input/#/get/attenuation
/wfs/input/#/get/curvature
/wfs/input/#/get/control
/wfs/input/#/get/positionX
/wfs/input/#/get/positionY
/wfs/input/#/get/positionZ
/wfs/input/#/get/positionXYZ
/wfs/input/#/get/constraintX
/wfs/input/#/get/constraintY
/wfs/input/#/get/constraintZ
/wfs/input/#/get/constraintXYZ
/wfs/input/#/get/flipX
/wfs/input/#/get/flipY
/wfs/input/#/get/flipZ
/wfs/input/#/get/flipXYZ
/wfs/input/#/get/heightFactor
/wfs/input/#/get/maxSpeed
/wfs/input/#/get/distanceAttenuation
/wfs/input/#/get/directivity
/wfs/input/#/get/rotation
/wfs/input/#/get/HFshelf
/wfs/input/#/get/levelMap
/wfs/input/#/get/liveSource
/wfs/input/#/get/mutes
/wfs/input/#/get/delays

/wfs/input/#/get/levels
/wfs/input/#/get/HFdampings

/wfs/input/#/stream/all [0/1]
/wfs/input/#/stream/latency [0/1]
/wfs/input/#/stream/attenuation [0/1]
/wfs/input/#/stream/curvature [0/1]
/wfs/input/#/stream/control [0/1]
/wfs/input/#/stream/positionX [0/1]
/wfs/input/#/stream/positionY [0/1]
/wfs/input/#/stream/positionZ [0/1]
/wfs/input/#/stream/positionXYZ [0/1]
/wfs/input/#/stream/constraintX [0/1]
/wfs/input/#/stream/constraintY [0/1]
/wfs/input/#/stream/constraintZ [0/1]
/wfs/input/#/stream/constraintXYZ [0/1]
/wfs/input/#/stream/flipX [0/1]
/wfs/input/#/stream/flipY [0/1]
/wfs/input/#/stream/flipZ [0/1]
/wfs/input/#/stream/flipXYZ [0/1]
/wfs/input/#/stream/heightFactor [0/1]
/wfs/input/#/stream/maxSpeed [0/1]
/wfs/input/#/stream/distanceAttenuation [0/1]
/wfs/input/#/stream/directivity [0/1]
/wfs/input/#/stream/rotation [0/1]
/wfs/input/#/get/HFshelf [0/1]
/wfs/input/#/stream/levelMap [0/1]
/wfs/input/#/stream/liveSource [0/1]
/wfs/input/#/stream/mutes [0/1]
/wfs/input/#/stream/delays [0/1]
/wfs/input/#/stream/levels [0/1]
/wfs/input/#/stream HFdampings [0/1]

INPUT MOVEMENTS

/wfs/input/#/curveXYZ [f: destination x] [f: destination y] [f: destination z] [0 absolute position/1 relative position] [f: curvature of trajectory: 0 straight line, -100< <0 curves downstage, 0> >100 curves upstage] [f: time in seconds] [f: 0 constant speed ~ 100 smooth start and stop]
/wfs/input/#/curveXYZ/pause [0/1]
/wfs/input/#/curveXYZ/stop
/wfs/input/all/curveXYZ/moveSpeed [i] 0 to 200 (%)

/wfs/input/#/lfo/active [0/1]
/wfs/input/#/lfo/gyrophone [0 off/1 clockwise/-1 anti-clockwise]
/wfs/input/#/lfo/lfo [f: period in seconds] [i: phase 0°~360°]
/wfs/input/#/lfo/x [i: 0~359° phase for X] [i: shape* for X] [f: amplitude for X]
/wfs/input/#/lfo/y [i: 0~359° phase for Y] [i: shape* for Y] [f: amplitude for Y]
/wfs/input/#/lfo/z [i: 0~359° phase for Z] [i: shape* for Z] [f: amplitude for Z]
/wfs/input/#/lfo/shapeXYZ [i] [i] [i] (shapes* for X Y Z)
/wfs/input/#/lfo/xyz [i] [i] [i] [i] [i] [i] [f] [f] [f] (0~359° phases for X Y Z ; shapes* for X Y Z ; amplitudes for X Y Z)
/wfs/input/#/lfo/lfoXYZ [f period of oscillator in seconds] [i: 0~359° phase of oscillator] [i] [i] [i] [i] [i] [i] [f] [f] [f] (0~359° phases for X Y Z ; shapes* for X Y Z ; amplitudes for X Y Z) [0/1 gyrophone]
* shapes: 0 Off / 1 Sine / 2 Square / 3 Saw / 4 Triangle / 5 Keystone / 6 Log / 7 Exponential / 8 Random

/wfs/input/#/jitter [f: amplitude]

/wfs/input/#/offset [f] [f] [f]

REVERB RETURNS

/wfs/selectIO/reverb [i]

/wfs/reverbReturn/#/latency [f] ms
/wfs/reverbReturn/#/attenuation [f]s
/wfs/reverbReturn/#/curvature [0/1]
/wfs/reverbReturn/#/positionX [f] m
/wfs/reverbReturn/#/positionY [f] m
/wfs/reverbReturn/#/positionZ [f] m
/wfs/reverbReturn/#/positionXYZ [f] [f] [f] m m m
/wfs/reverbReturn/#/heightFactor [i] %
/wfs/reverbReturn/#/distanceAttenuation [f] dB/m
/wfs/reverbReturn/#/directivity [i] 2°~360°
/wfs/reverbReturn/#/rotation [i] -180°~180°
/wfs/reverbReturn/#/HFshelf [f] dB
/wfs/reverbReturn/#/mutes [i_list]
/wfs/reverbReturn/#/muteMacro [i]
1: mute all, 2: unmute all,
3: invert,
4: odd channels, 5: even channels,
6: first half, 7: second half,
8: mute output group 1, 9: unmute output group 1,
10: mute output group 2, 11: unmute output group 2,
12: mute output group 3, 13: unmute output group 3,
14: mute output group 4, 15: unmute output group 4,
16: mute output group 5, 17: unmute output group 5

/wfs/reverbReturn/#/get/all
/wfs/reverbReturn/#/get/latency
/wfs/reverbReturn/#/get/attenuation
/wfs/reverbReturn/#/get/curvature
/wfs/reverbReturn/#/get/positionX
/wfs/reverbReturn/#/get/positionY
/wfs/reverbReturn/#/get/positionZ
/wfs/reverbReturn/#/get/positionXYZ
/wfs/reverbReturn/#/get/heightFactor
/wfs/reverbReturn/#/get/distanceAttenuation
/wfs/reverbReturn/#/get/directivity
/wfs/reverbReturn/#/get/rotation
/wfs/reverbReturn/#/get/HFshelf
/wfs/reverbReturn/#/get/mutes