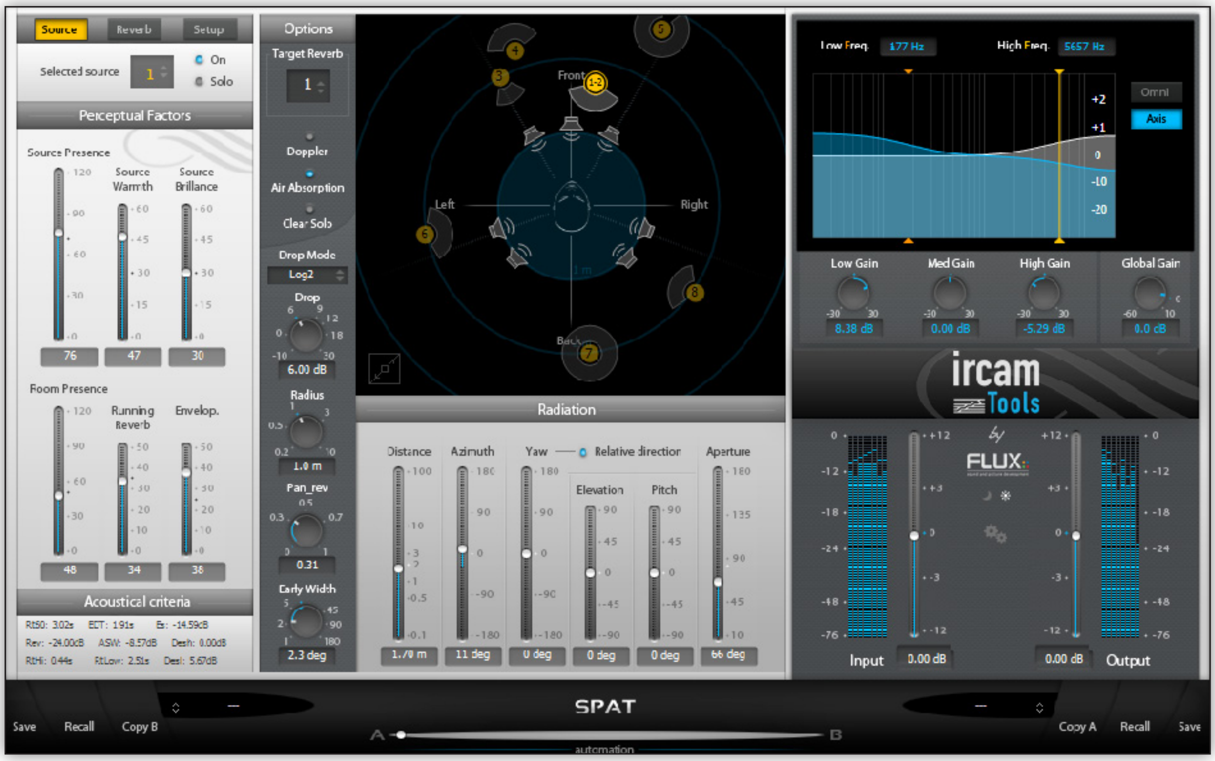
ircam Tools by FLUX:: sound and picture development

SPAT

Multiformat Room Acoustic Simulation & Localization Processor



Introduction

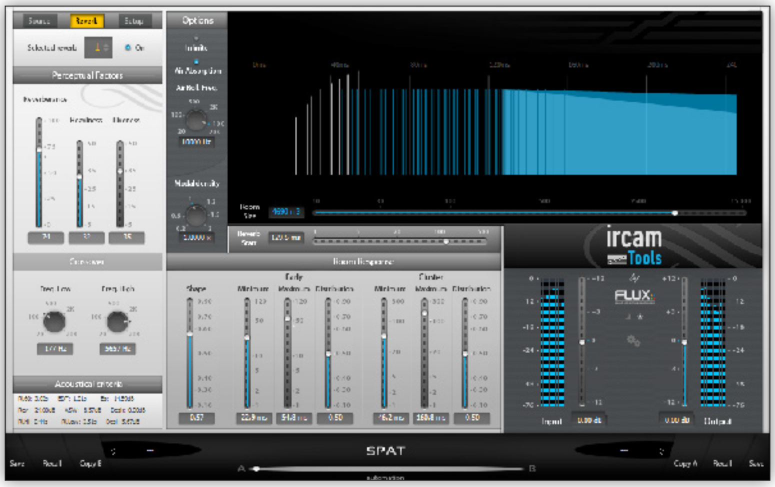
The Spat is a very powerful tool for spatialization. It can handle both aspects of spatialization (source localization and room acoustic simulation) in a very homogeneous and coherent way. A set of sources (up to eight in this plug-in, may be linked in stereo pairs) are localized in a 3D space (in the Source control tab). Each of these source is connected to a room (also called reverb). Up to 3 rooms are available in parallel, in order to simulate complex spaces (coupled room acoustics...).

Perceptual factors Vs. Acoustical criteria

In both sources edition and reverb edition, controls are available as a set of “perceptual factors”. Each of these parameters corresponds to a real-world criterion that actually has proper significance in terms of hearing perception, and not some obscure algorithm-specific parameter translating more-or-less to some kind of aesthetic interpretation. This is a feature unique to the Spat, which is the result of substantial research and development work at IRCAM and Flux, providing a relationship between these parameters and the plug-in internal parameters. Internally, the plug-in perceptual mapper translates the GUI parameters into a much larger set of parameters required for the actual reverb engine computations.
This way the complexities of the reverb algorithm internals are hidden from the user so one can make predictable and guided adjustments, instead of having to resort to a “poke in the dark” approach, adjusting settings at semi-random trying to find some “magical” combination of parameters to obtain the desired result.
In parallel, a display is detailing the “acoustical criteria”, these parameters are more familiar for those who are well acquainted with acoustics, and here instead will find the more commonly used terms.

SPAT

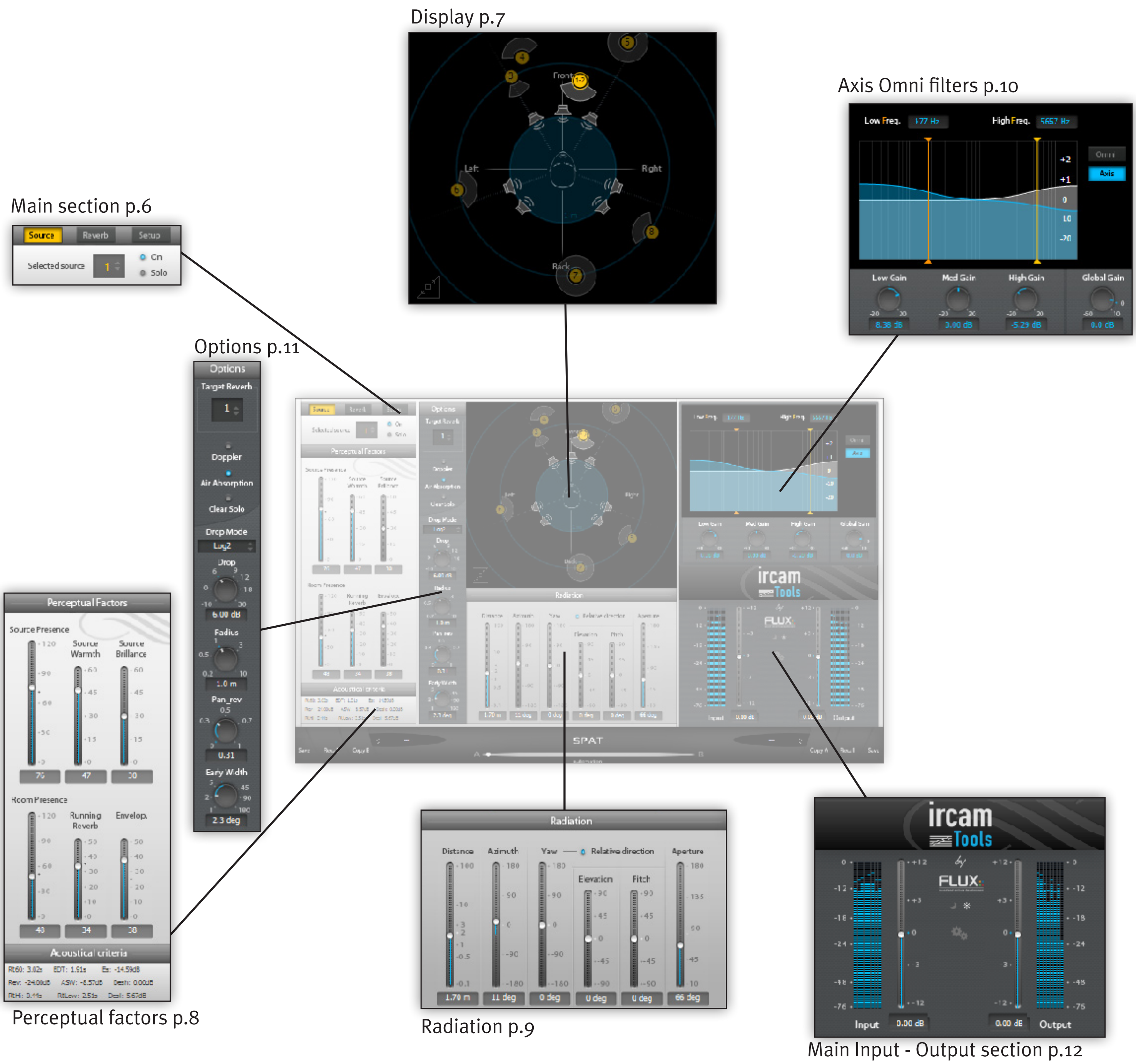

Options
Radiation
ircam
Tools
FLUX
SPAT


Options
ircam
Tools
FLUX
SPAT


ircam
Tools
FLUX
SPAT

SPAT

Sources


Display p.7
Axis Omni filters p.10
Main section p.6
Options p.11
Perceptual factors p.8
Radiation p.9
Main Input - Output section p.12

1 - Main Section

The source tab is where sources parameters are set and visualized.
In essence, input channels are actually fed to one or several virtual sources that are placed inside a virtual 3D acoustic space. The user can precisely control the location of the sources in the virtual acoustic space, and adjust a number of parameters separately for each source.

Note: a variation of this sub-panel is displayed when viewing the reverb tab.

(1) Source
Activates the source tab, where source localization, perceptual and acoustical parameters are set.

(2) Reverb
Activates the reverb tab, where the reverberation characteristics of the simulated acoustic space are adjusted.

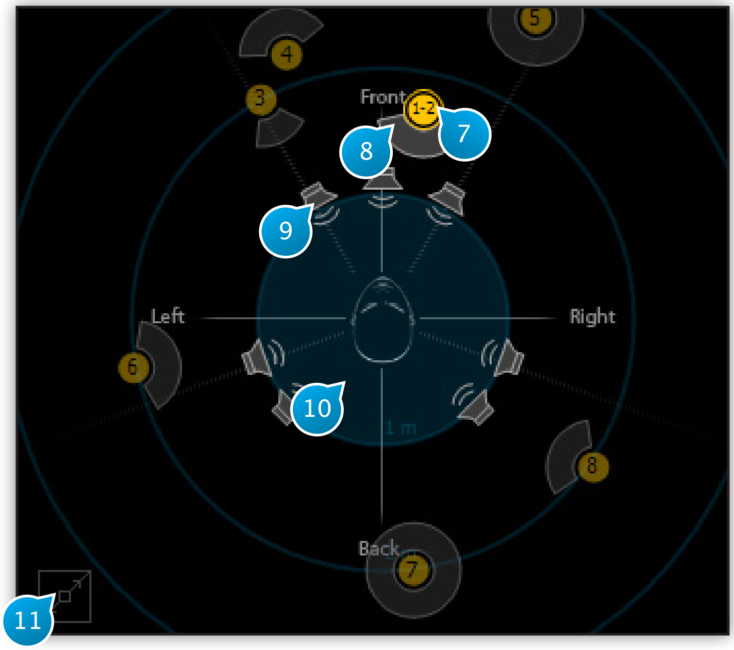
(3) Setup
Activates the setup tab, where input, source and output mappings are determined.

(4) Selected Source
Selects the source currently active for editing, which is highlighted accordingly in the visual space representation.
Each source has its own set of parameters which is completely independent of others, while sharing the same reverberation engine(s) that depict the acoustic response of the simulated space these sources are place into.
Note: The number of sources and mapping thereof is handled in the Setup tab.

(5) Source On
Toggles the source on and off in the global output mix. Used in conjunction with the solo button below, this allows to switch rapidly between listening to each source in isolation, for precise adjustments or tweaking settings, and listening to all sources simultaneously, which is essential to getting a good balance between the various sources and achieve a coherent mix.

(6) Source Solo
Isolates the currently active source, effectively muting all other sources.

2 - Display



This sub-panel displays a to-scale representation of the sources and speakers spatial localization settings, projected onto a 2D flat space. Assessing the location of a point is easy with the aid of the circular "grid" scale drawn onto the background as a guide, using a polar coordinate system.

The spherical coordinate system used here requires the following three parameters to describe the position of a point in 3D-space, relative to the listener:

* Distance to the centre reference point, expressed in meters on a logarithmic scale
* Azimuth, or the angle with respect to a vertical line (zero is in front), in degrees.
* Elevation, in degrees.

Also, the source orientation is available as yaw an pitch.

Zooming in and out of the display is achieved using the scroll wheel present on most available mice.

Note that when the current source is not at abase plane (z-axis), a little display appears on the right, explaining this z-elevation for this 2D view context.

(7) Source
A source represented by a yellow disk with its corresponding number. Clicking on this disk makes this source active and highlighted, allowing it to be moved and edited.
Alt-clicking (MacOS: Alt/option key) and dragging on the yellow source disk changes its aperture (See 21)
Ctrl-clicking (MacOS: Apple/command key) and dragging on the yellow source disk changes its yaw (See 20)

When two sources are associated in a stereo pair, this yellow disc is surrounded by a yellow circle.

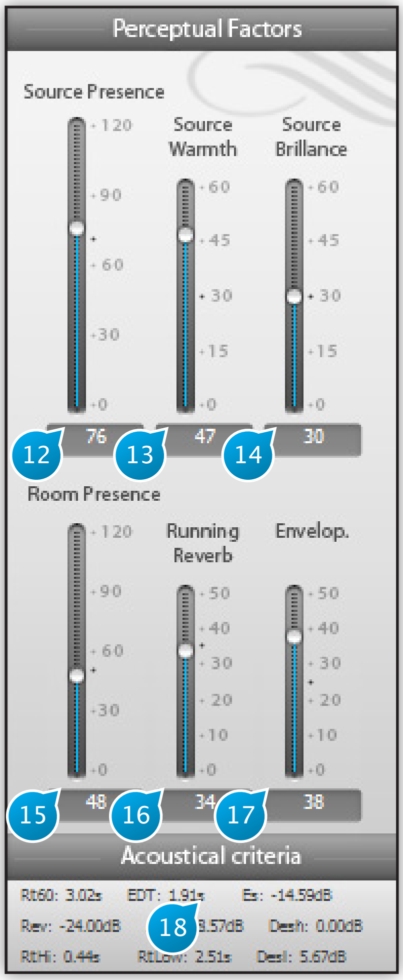
(8) Aperture
Represents spatial width and direction of the source signal part, which determines the preferred directivity. See 25.

(9) Speaker
Speaker location and direction. Each speaker can individually be moved by clicking and dragging, and reset to its default position with a right click.
Speakers can also be hidden by right-clicking anywhere in an empty space.
Please note the number and characteristics speakers are respectively adjusted in the Setup sub-menu and tab

(10) Radius
Represents the the no-drop zone (See 40, Drop) as a transparent blue disk of corresponding radius.

(11) Full Size Display
Toggles between standard display as a sub-panel and a full-size display, which is more convenient when very precise settings are need, or when doing intensive source localization automation for example.

3 - Perceptual Factors



This parameter group holds settings affecting the way the reverb sound is perceived by the listener. As explained previously, these are not simply names stuck onto any internal parameter dictated by the inner workings of the algorithm.
Instead, a true perceptually-oriented approach was used in the design of the plugin, where a test panel of listeners was presented with a test-set of sounds, constructed from several different variations of the reverb engine inner parameters. The listeners were then asked to rate each set onto a few different scales with perceptually and aesthetically meaningful names. Using principal components analysis (PCA) and optimization techniques, we then built an algorithm which reverses the process and automatically maps a given set of perceptual factor values to the many internal reverb engine parameters.

As a general guideline, we encourage you to learn the meaning of these parameters by carefully listening to the audible variations when adjusting them. We do provide a short explanation of each of them below, but training your ears is really the best way to be able to use these in context.

(12) Source Presence
Source presence refers to the prominence of the direct sound with respect to the reverberated sound. It is not just equivalent to a dry/wet ratio, and is influenced by other settings such as distance, radius and drop-factor.

(13) Source Warmth
Presence of the low frequency content part of the source.

(14) Source Brillance
Presence of the high frequency content part of the source.

(15) Room Presence
Prominence of the reverberation with respect to the source, or in other words, how much the room sound dominates the overall sound.

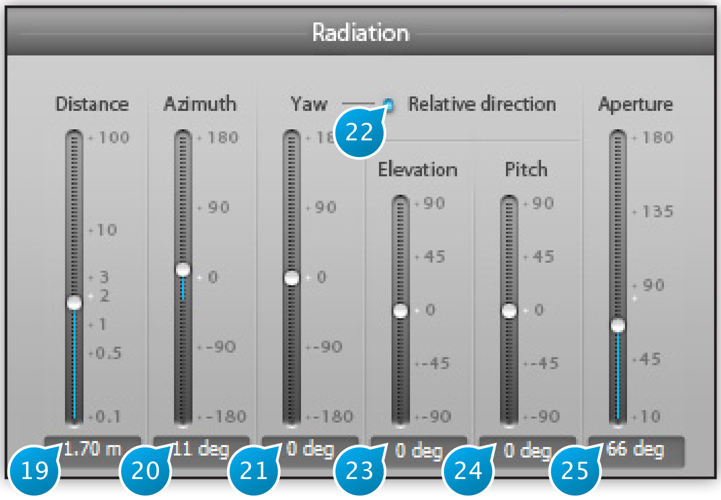
(16) Running Reverb
This parameter controls the amount of perceived reverb when feeding a continuous music message, where the overall sound is a tight blend of the dry and wet signals and the reverb part cannot be mentally separated. It is linked to the early reflections decay time.
Note: this setting is distinct from ‘reverberance’ in the reverb tab.

(17) Envelop.
Envelopment corresponds to the the perceived notion of how much the listener feels being surrounded by the ambient sound. In a multichannel configuration, one could describe this as the feeling of being wrapped inside the imaginary “sound sphere” that the listener pictures in his mind. It can also be described as the energy of the early room effect with respect to direct sound.

(18) Accoustical Criteria
This sub-panel displays the acoustical parameter values associated with the current settings. They are provided for the more technically inclined users who already know how they can make use of them; you can safely ignore them otherwise. Please note this is an indicative feature only, as these settings can not directly be changed here.

* RT60 (Reverberation Time, in seconds): measure of reverberation decay time, namely the time taken by the reverb to drop below -60dB of attenuation.
* EDT (Early Decay Time, in seconds): time to decay to -60dB, derived from a linear fit of the decay curve between 0 and -10dB. It explains the very early evolution of a reverberation
* Es (dB): Early sound : power of direct sound and early room effect
* Rev (dB): Late sound : power of later reverberation
* ASW (Apparent Source Width, in dB).
* Desh (dB): Relative early sound at high frequencies
* RtHi (dB): Same as RT60, but in high frequency.
* RtLow (dB) Same as RT60, but in low frequency.
* Desl (dB):

4 - Radiations



(19) Distance
Distance from the source to the center reference point (listener position), in meters.

(20) Azimuth
Angle between the source location and the listener-front reference axis, in degrees.

(21) Yaw
Angle of the source direction orientation relative to the listener-source axis, in degrees.

(22) Relative direction
When engaged, the source movement done with the mouse in the display are done in a way that the source always keep the same source-listener orientation.
If not, the source direction is constant.

(23) Elevation
Elevation angle, in degrees.

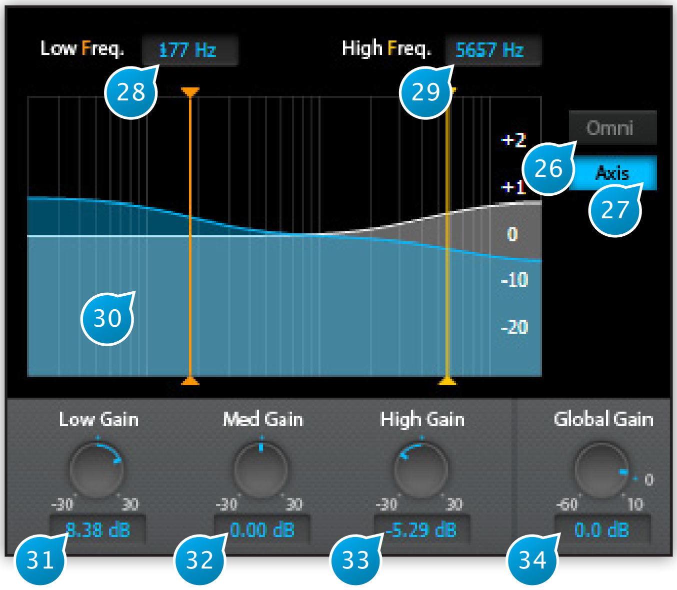
Please note that when the source is not located onto the base horizontal plane, at listener ear level (z = 0), a small gauge is displayed on the right edge of the preview, which gives the distance of the source to listener projected onto the Z axis, for easier handling of elevation.

(24) Pitch
Source direction orientation pitch angle, in degrees.
Please pay attention to the fact that the graphic preview cannot not display pitch movements because of the 2D projection constraints.

(25) Aperture
The aperture parameter relates to the “sound cone” projected by the virtual source in the acoustic space, and is measured in degrees. It determines wether the source will be very directive (small aperture), or omnidirectional (large aperture) inside the reverberant environment. You can use this parameter to simulate the how a source will interact with its acoustical environment.

Each source has a set of two three-band equalizers associated with it, as described below.

5 - Axis Omni Filters



(26) Omni
Toggles the equalizer associated with the omnidirectional part of the sound radiated by the virtual source. This equalizer mimics the global frequency-response of the source, just as a loudspeaker colors the sound.

(27) Axis
Toggles the equalizer associated with the on-axis part of the sound radiated by the virtual source.
Most, if not all commercially available loudspeakers do exhibit a radically different frequency response whether you're listening right in front or on the sides. Setting a rather flat on-axis equalizer curve, and maybe cutting the treble and mids for the omni response would be a good starting point to emulate a real-world speaker.

(28) Low Freq.
Low-pass filter frequency cutoff of the currently selected equalizer. Value can be quickly adjusted using the mouse by clicking and holding the value box or the vertical bar on the graphical frequency-response display, or exactly by clicking the value box and entering a value with the keyboard.

(29) High Freq.
High-pass filter frequency cutoff of the currently selected equalizer.

(30) Axis - Omni Curves
Actual frequency response curve of the currently selected equalizer, expressed in dB (deciBel).
Click and drag one of the three square points with your mouse to quickly change the gain of the corresponding band, or one of the two vertical bars to adjust the transition frequencies of the bands.

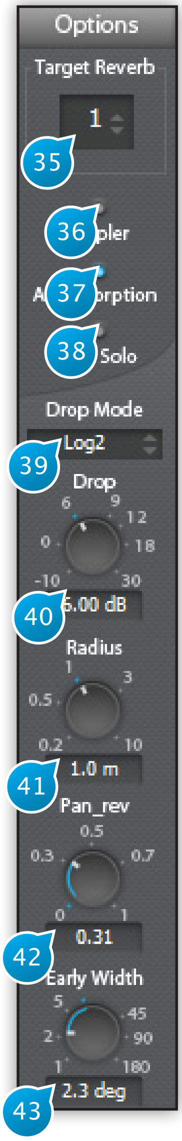
(31) Low Gain
Gain applied to the low frequency band of the currently selected equalizer.

(32) Med Gain
Gain applied to the medium frequency band of the currently selected equalizer.

(33) High Gain
Gain applied to the high frequency band of the currently selected equalizer.

(34) Global Gain
Overall gain applied to the currently selected equalizer.

6 - Options



<u>(35) Target Reverb</u>
Drop-down menu indicating the currently selected reverb engine. Clicking this menu allows you to choose between the three available engines, each one having its one independent set of parameters. For example, you could combine two or three reverbs in order to simulate a complex room architecture, made up of different spaces with completely different sizes and materials, or to achieve special effects.

<u>(36) Doppler</u>
Toggles Doppler effect handling on and off. The Doppler effect is a well-known wave propagation phenomenon where the height of a sound perceived from a listener standpoint rises when the source is accelerating, and falls when decelerating. This is the ambulance or police siren pitch going up then down when passing you by effect you are familiar with.
It will only be heard if you rapidly move the sources locations quite fast, but thanks to the virtual nature of the Spat, you can bypass Physics' laws and manually inhibit it using this switch, should it be a nuisance for the particular application you're dealing with.

<u>(37) Air Absorption</u>
Simulates the frequency-dependent absorption of air, where high frequencies roll-off quicker than low-frequencies with respect to distance. You've most probably noticed this other real-world phenomenon when you're far away from a concert venue and only able to hear the bass, and gradually start to hear the whole mix as you get closer.

<u>(38) Clear Solo</u>
Clear all solo that have been previously engaged on sources.

<u>(39) Drop Mode</u>
Owing to a fundamental law of acoustics and geometry, namely energy conservation, sound pressure drops in level as one moves away from the source. Choose "Log2" for an acoustically accurate setting, which corresponds to a "Drop" value attenuation every time the distance from the source is doubled (logarithmic behaviour), or "Linear", for a less radical drop law.

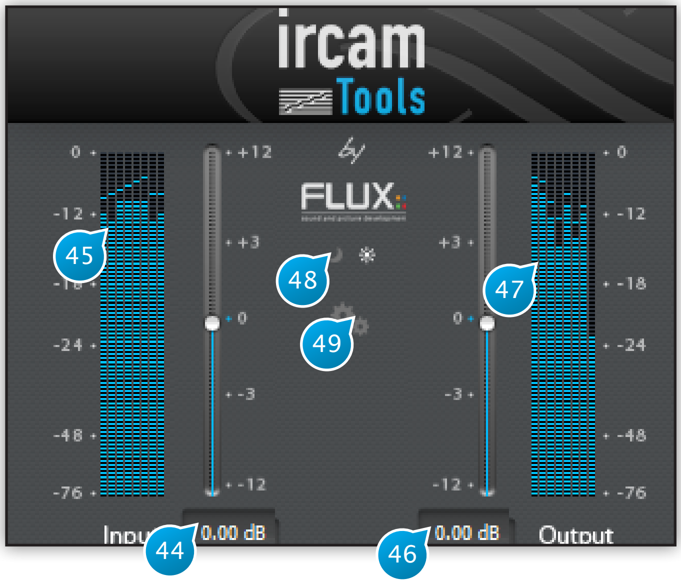
<u>(40) Drop</u>
This value determines the value of the level attenuation in dB (deciBels), according to the selected drop mode. Default is 6dB, which is the physically accurate value.

<u>(41) Radius</u>
Specifies the radius of a disk, in meters, centered around the listener's head, where the drop attenuation is not taken into account, and the sound level is kept constant with regards to distance. This is not only useful to prevent any dramatic sound level peak when placing a source too close to the listener, it also reflects real-world behaviour quite accurately, where sources do have a certain physical size, unlike point sources that are commonly used to model far-field acoustics.
This "no-drop" zone is displayed as a transparent-blue disk of matching radius on the preview.

(42) Pan_rev
By default, only early reflections are panned, and the cluster reflections, which form the diffuse part of the early reverberation, are panned dead center. Pan_rev allows you to modify cluster panning, thus imparting some directionality or perceived direction to the diffuse part of the sound.

(43) Early Width
Controls the width of the sound projection lobe of the early reflections from a source in the virtual acoustic space, in

7 - Main Input - Output Section



degrees. The minimum setting, 1°, gives a very directional source, whereas 180° makes it omnidirectional.

(44) Input Gain
Adjust the level of the signal fed to the plug-in, in dB increments.

(45) Input level meter
Shows the current level of the input signal after applying input gain, in dB FS (deciBel Full Scale).

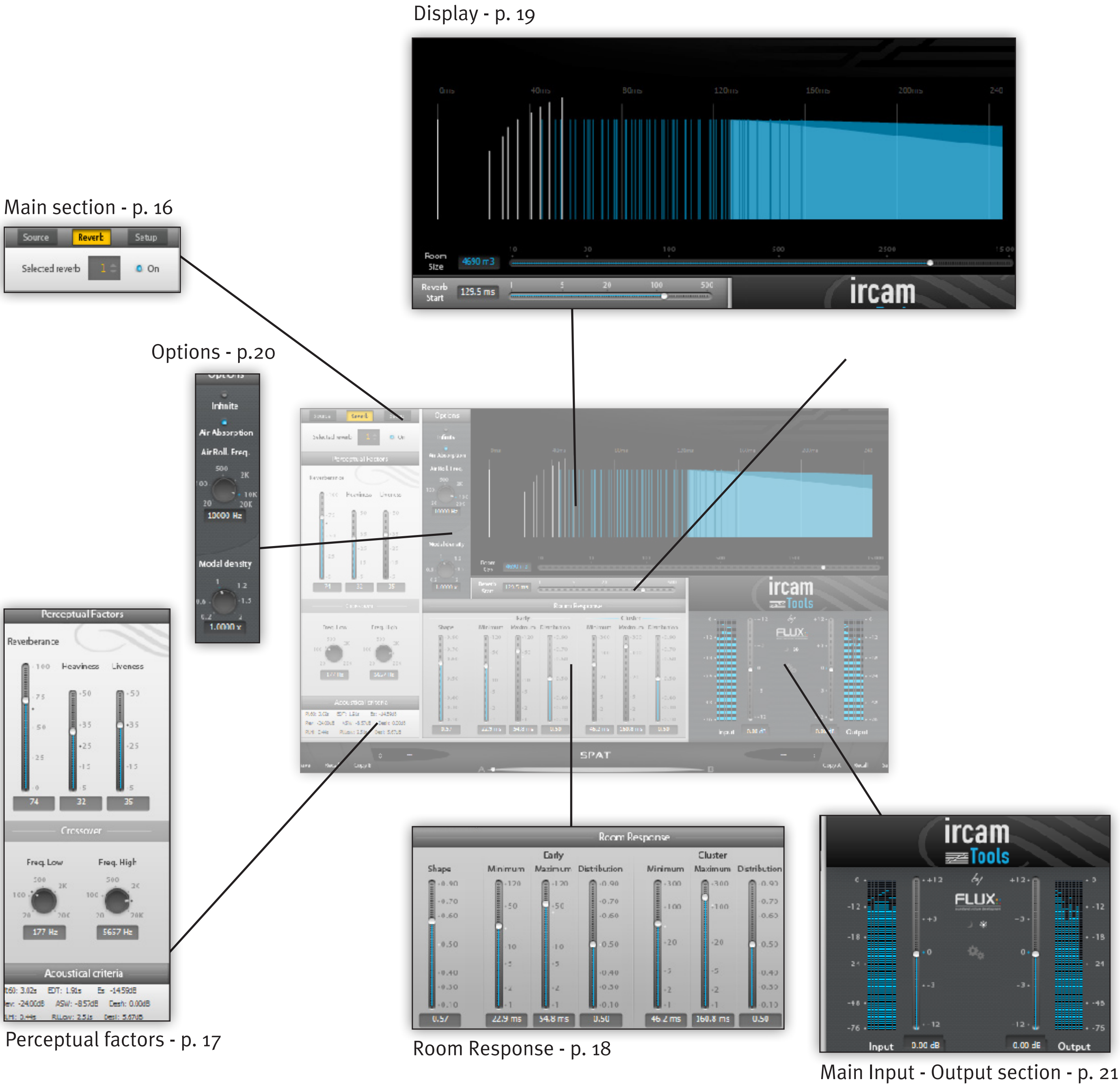
(46) Output Gain
Used to trim the output signal and possibly avoid any overloading of the signal in the rest of the signal-chain.

(47) Output level meter
Shows the current level of the input signal after applying output gain, in dB FS(deciBel Full Scale).

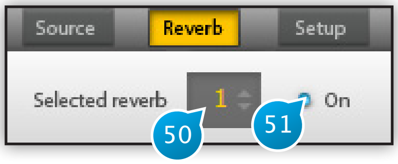
(48) Day - Night
Toggles between two interface schemes, which, as the name implies, are best suited to high or low light environments respectively. In a dimly-lit studio environment, switching to the nigthtime scheme with its darker color palette and lower contrast will minimize eye-fatigue when doing long sessions.

(49) Setting
Gives access to a sub-menu where you can either select the I/O configuration, namely the input channel count followed by the output channel count, for hosts that support dynamic I/O configuration, or display the credits page.
The exact I/O combinations available depend on your actual audio hardware and host configuration.

REVERB


Display - p. 19
Main section - p. 16
Options - p.20
Perceptual factors - p. 17
Room Response - p. 18
Main Input - Output section - p. 21

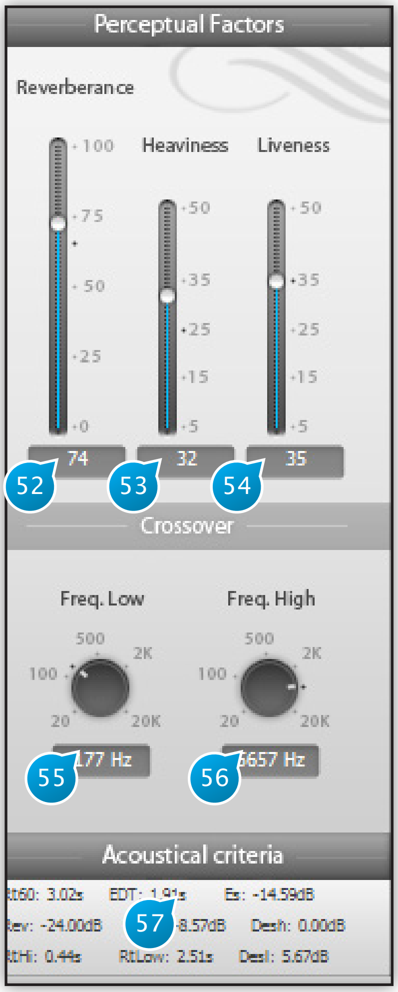
8 - Main Section



(50) Selected Reverb
Selects and indicates the reverb engine currently being edited. Three engines, working in parallel and independently, are available

(51) Reverb On
Toggles the currently selected reverb engine on or off. Use this switch at design time, to listen to one reverb in isolation.

9 - Perceptual Factors



(52) Reverberance
Reverberance affects the amount by which the listener perceives the music to be prolonged by the reverb, when the musical message suddenly stops. The effect of this setting is also obvious when the source material is of percussive nature. Reverberance is tightly related to overall decay time of the mid-frequencies, which in turn is the time taken by the late reflections to vanish into silence.

(53) Heaviness
Relative decay time of low-frequency content, relative to the reverberance.

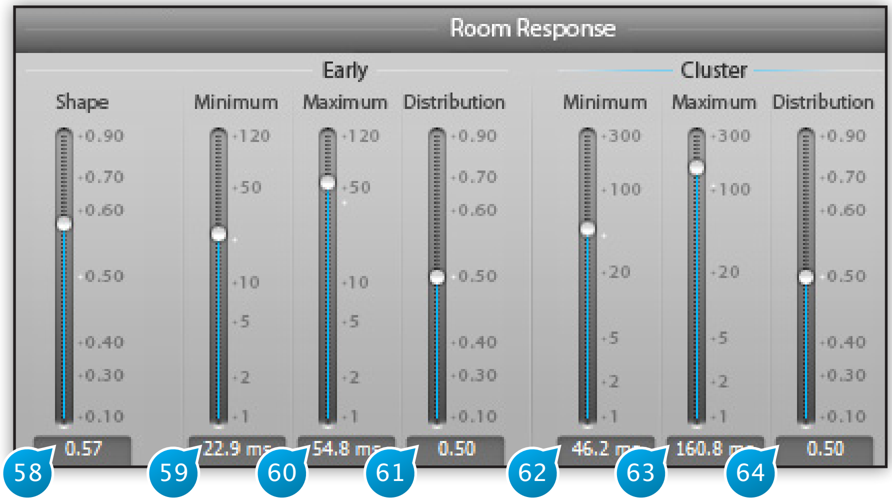
(54) Liveness
Relative decay time of high-frequency content, relative to the reverberance. Describes the liveliness and movement associated with the reverb tail (late reflections).

(55) Freq. Low
Sets the frequency below which decay time is determined by the Heaviness setting, expressed in Hertz(Hz). Default value: 180 Hz

(56) Freq. High
Sets the frequency above which decay time is determined by the Liveliness setting, expressed in Hertz(Hz). Default value: 5657 Hz

(57) Acoustical criteria
See 18.

10 - Room Response



(58) Early Shape
Governs the amplitude rise or fall of early reflections. The default setting of 0.5 corresponds to early reflections all having the same level. This mimics an acoustical space where reflective surfaces are all located at roughly the same distance to the listener. Below 0.5 early reflections decay with time, above 0.5 they rise with time. Early reflections of decreasing level would be typical of a space where most of the reflective surfaces are grouped at a range closest to the listener.

(59) Early Minimum
Early reflections minimum time, i.e. the time at which the early reflections start to appear, in milliseconds. This is the analogous of the ubiquitous “pre-delay” setting found on most reverberation processors. It represent the time between the direct sound and the first early reflection.

(60) Early Maximum
Early reflections maximum time, i.e. the time at which these cease to appear.

(61) Early Distribution
Early reflections distribution. Determines the way early reflections are scattered in time, inside the Early Min. / Early Max. interval. The default setting of 0.5 corresponds to regularly spaced reflections, above these are more grouped towards the Early Max. value, and vice-versa.

(62) Cluster Minimum
See 59, Early Min.
Please keep in mind the cluster is fed with the input of the early reflections processor section, as is shown accordingly on the display.

(63) Cluster Maximum
See 60, Early Max.

(64) Cluster Distribution

11 - Display

See 61, Early Distribution.

(65) Direct signal

The grey bar at the start of the reverberation pictogram represents the direct sound send at the input of the plug-in. In the time structure of the reverberation, it is the first element that is heard.

(66) Early

Overall representation of the early reflections distribution. Vertical bars roughly indicate at what time locations (horizontally) and levels (bar height) these early reflections occur.

(67) Cluster

Overall representation of the cluster reflections distribution. Vertical bars roughly indicate at what time locations (horizontally) and levels (bar height) these early reflections occur.

(68) Reverb

Shows a graphical representation of the reverberation tail part of the engine. The decay curves of the high, mid and low bands, which are controlled by the decay time settings, are superimposed in different colors and can rapidly be assessed and checked.

Also see 12.

(69) Room Size

This is a meta-parameter that takes care of varying several other parameters in order to quickly set the equivalent size of the virtual room, adjusting early, cluster and tail reverb parameters to match the room characteristics.

(70) Reverb Start

Reverb start sets the duration between the direct, dry source signal, and the first late reflections, or start of the reverb tail. Please note its value can never go below that of the cluster minimum time as the reverb tail is fed with an signal derived from the cluster section.

12 - Options

(71) Infinite

When activated, decay time temporarily rises to infinity, making the signal recirculate indefinitely inside the reverberation engine. This is best suited for one-off special effects such as “deep-freezing” audio material, or if you’re looking to create something a little less conventional than a fade-out for the end of your track.

(72) Air Absorption

Simulates the frequency-dependent absorption of air, where high frequencies roll-off quicker than low-frequencies with respect to distance. You’ve most probably noticed this real-world phenomenon when you’re far away from a concert venue and only able to hear the bass, and gradually start to hear the whole mix as you get closer.

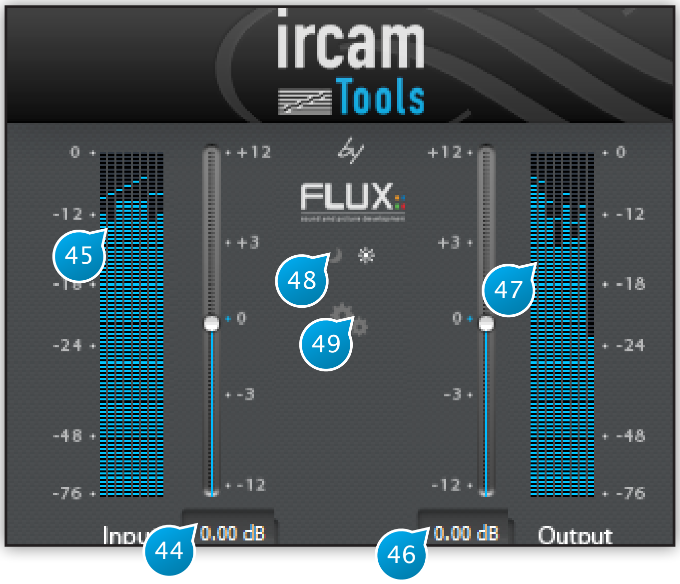
(73) Air Roll. Freq.

Roll-off frequency for the air absorption simulation. Signal content above this frequency vanishes faster.

(74) Modal Density

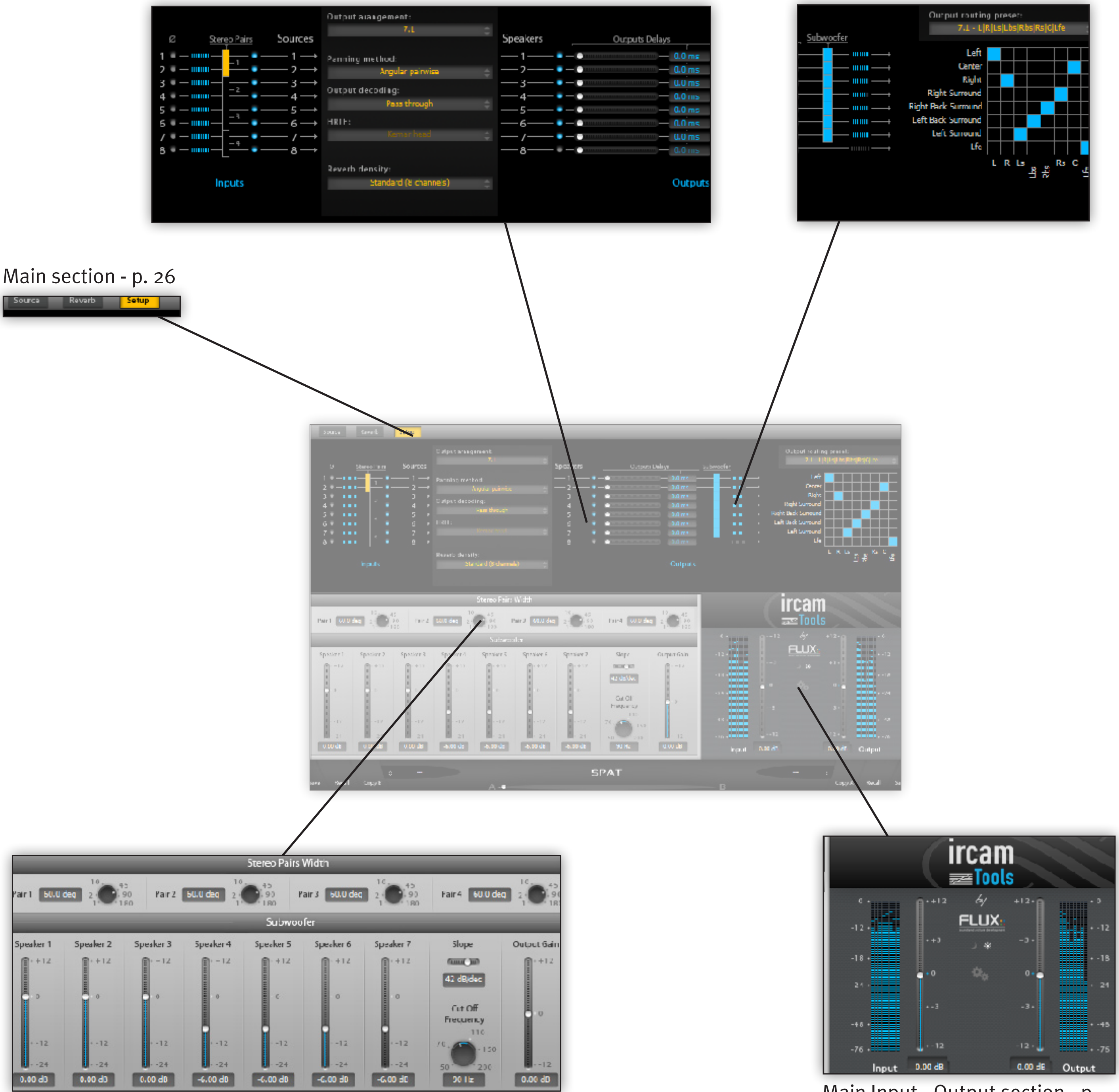
Scales the modal density with respect to the current setting, which is internal to the plug-in engine, and depends on other parameters such as reverberation time, etc.
The modal density governs the frequency “smoothness” of the verb engine. Increasing this setting reduces the graininess of the reverberation. Adjust to taste, depending on the source material and desired result.

13 - Main Input - Output Section



See page 12.

Setup



Setup - p. 24

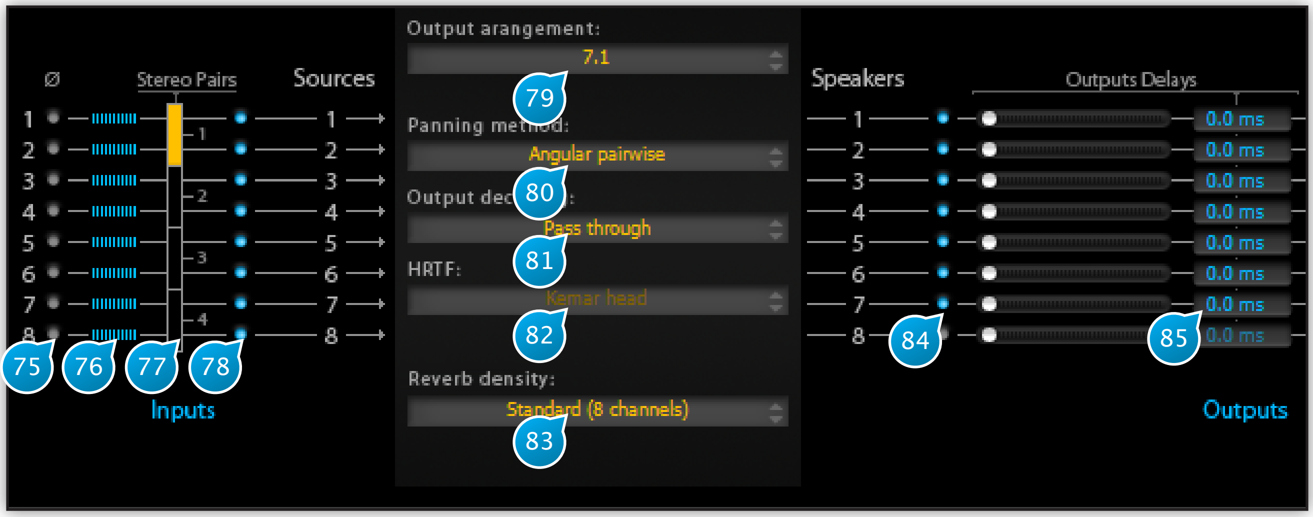
Routing - p. 26

Main section - p. 26

Stereo Pairs width - Subwoofer p. 27

Main Input - Output section - p.

14 - Setup



(75) Inv. Phase
This little button labelled “ ” inverts the phase of the source, applying a 180° phase shift.

(76) Input level meter
Gives an indication of the signal presence and peak level for the corresponding input source.

(77) Stereo Pair
Enable two successive inputs to be treated as a stereo pair, displayed by a yellow vertical block overlapping the two inputs, which make it possible to adjust their width in the stereo space.

(78) Source On
Toggles wether the corresponding source is active or not (muted).

(79) Output Arangement
Depending on the number of I/O channels you have selected in the setup menu, accessed via the cog icon between the I/O meters, this drop-down menu allows you to select the desired output arrangement amongst a choice of stereo and a variety of surround configurations, with or without a sub-woofer.
This setting is provided so you can operate the Spat on a subset of your surround channels, should your project require this.
For example, when mixing a film soundtrack, it is mandatory that the center channel, which is reserved for voice material, be left empty. The solution is to work with a so-called virtual center (equal energy in channels), disabling processing by the Spat for the physical center channel.

(80) Panning Method
Depending on the Output arrangement setting, choose the method to use for placement of the sources inside the audio space.
Possible choices are (some will be grayed out depending on your output configuration):

* Recursive
* Angular pairwise (default)
* Surround LRS
* Surround LRCS

* Binaural. Intended for stereo reproduction. Prefer transaural output encoding when loudspeakers are to be used.
* VBAP 2D (Vector Base Amplitude Panning).
  Allows for arbitrary placement of the speakers onto a horizontal plane.
  Refer to http://www.acoustics.hut.fi/~ville/ for more information on this technique.
* VBAP 3D
  Same as above, but for a 3D speaker setup.
* DBAP (Distance-Based Amplitude Panning).
  Does not rely on any assumptions regarding speaker array or listener position, which is good for avoiding sweet spots.
  Refer to http://jamoma.org/papers/icmc2009-dbap.pdf for more information.
* B-Format 2D.
  A basic variant of Ambisonics. Amongst Ambisonics strong points are a good stability of the audio image with respect to listener position, free speaker positioning, and independence towards the reproduction system used.
* B-Format 3D
* 2nd order Ambisonic 2D
* 2nd order Ambisonic 3D
* XY. Stereo only.
* M/S (Mid/Side). Stereo only.
* A-B. Stereo only.

<u>(81) Output Decoding</u>
Spat can encode its output in the following currently supported formats

* Pass-trough, i.e. no encoding (default)
* Transaural. A method to reproduce binaural audio when using loudspeakers.
* B-Format 2D. A variant of Ambisonic encoding.
* B-Format 3D
* 2nd order Ambisonic 2D
* 2nd order Ambisonic 3D
* Surround LRS
* Surround LRCS

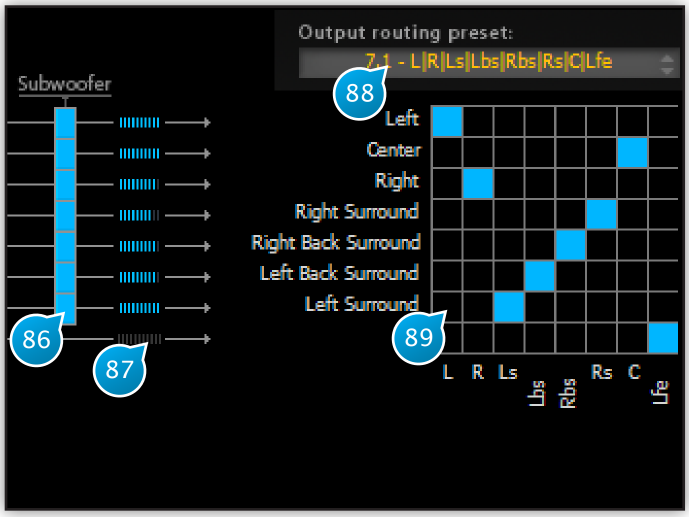
<u>(82) HRTF</u>
When the Binaural panning method is chosen, one can select between a variety of head HRTF profiles, which stands for Head-Related Transfer Function.
In essence, a HRTF models the way a listener's head modifies an incoming sound beam, depending on its direction. One can then measure and replicate the interaction of the head of a particular listener to recreate a sensation of 3D space. Please keep in mind this technique is only applicable to the stereo case. The default one available is the Kemar head, a standard Manikin head. Some others are available (choose the HRTF library at install time). Their numbers refers to the anonymous id of the listeners that have sampled their profile at IRCAM laboratories.

<u>(83) Reverb Density</u>
Internally, spatial variations are computed using a kind of 2D-network of reverbs, and this setting toggles between an 8x8 (standard) or 16x16 size (high). The choice of which sounds best is left up to you, as this depends on the source material at hand, although it must be emphasized that the high density consumes a little more CPU and that the color of the reverb can be altered by this setting, particularly at some extreme parameter setting combinations.

<u>(84) Speaker On</u>
Toggles the corresponding speaker output on and off, which can prove useful for checking purposes. In general it is advisable to use your DAW surround mixer capabilities instead, as these are global to your project.

(85) Output Delays
An output delay, expressed in milliseconds, can be inserted just before output to the speaker, for example to compensate for a non-standard speaker arrangement.
Please note that if you move a speaker in the source view, the associated delay is automatically applied on the corresponding output according to the speaker distance.

15 - Routing



(86) Subwoofer On
Activates send of a speaker output to the sub-woofer bus.

(87) Output level meter
Output peak level meter.

(88) Output routing preset
Gives quick access to commonly used output mappings.

(89) Output routing
Here you can map any Spat speaker output to a “physical” output. The plugin always ensures there is a one-to-one correspondence between Spats internal outputs and the plugin output channels, and remaps channels automatically when necessary.

16 - Stereo Pairs Width - Subwoofer

<u>(90) Stereo Pair Width</u>
Sets the width of a stereo pair, if previously defined, in degrees.

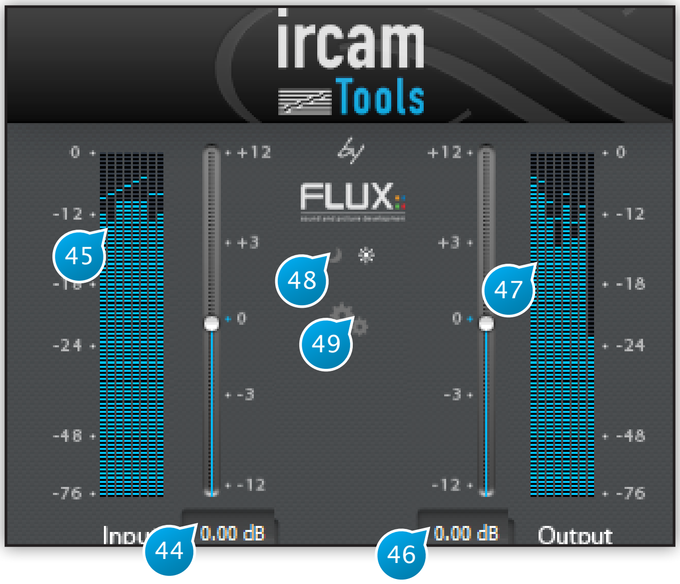
<u>(91) Subwoofer Speaker Gain</u>
This is the send level value from each of the Spat reverb outputs having the Subwoofer checkbock ticked, expressed in dB (deciBels).

<u>(92) Slope</u>
Slope of the low-pass filter section applied to the subwoofer buss, in dB per octave.

<u>(93) Cut Off Frequency</u>
Cutoff frequency of the low-pass subwoofer section, expressed in Hz (Hertz, default 110Hz). This filter ensures only the frequencies reproducible with a subwoofer are sent to the corresponding hardware output.

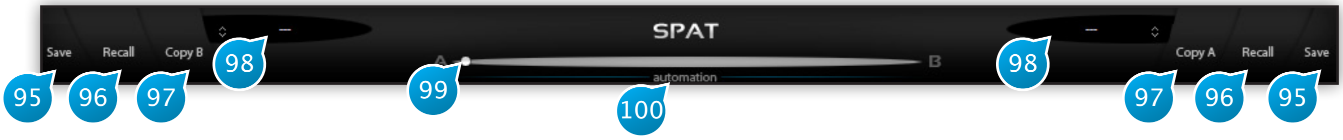
<u>(94) Subwoofer Global Output Gain</u>
Adjusts the overall gain of the subwoofer buss.

17 - Main Input - Output Section



See page 12.

18 - Presets Section



<u>(95) Save</u>
Saves a snapshot of the current settings for future use.
Short description and assorted comments can be provided, which comes in especially handy when sharing presets with other users, when the preset is part of a large preset bank, or to identify the author and source.
Entering a descriptive keyword is a good practice to be able to quickly sort your presets, according to character, the type of space they simulate (e.g. hall, room, etc.), and the intended usage (e.g voice, percussion, guitar, etc.)
A preset can be locked to prevent any further editing.
To re-save your preset under a new name, open the preset manager by clicking the corresponding (A/B) preset slot, then select New, enter a name for your preset, and finally press Save.

<u>(96) Recall</u>
Recall the settings from the currently selected preset, overwriting any current settings of the plug-in. The sub-menu which appears allows to recall at your choice:

* all parameters
* all parameters but setup: intended for when your particular speaker configuration is different from that of the preset's author (typically stereo)
* all parameters but setup and dry/wet mix: useful in a mix setting when comparing and choosing presets

<u>(97) Copy B</u>
Copy current settings to the second parameter slot (B). To try out a variation of the current settings without erasing the reference, press this button, switch to B and adjust your parameters of choice, then switch or morph between A and B. When copying a preset to a slot, the morphing slider will automatically fly to the corresponding slot.

<u>(98) Preset Name</u>
Displays the current preset name, if any. Clicking the associated button (up&down arrows) brings up the preset manager.

<u>(99) Morphing A B</u>
Gradually morphes parameters from A to B slots.
The parameter set associated with the current morphing slider position can be saved as a preset. In addition, when the morphing slider is in an intermediary position, any edit made to a parameter switches the slider back to slot A or B, whichever is closest to the current position.

<u>(100) Automation</u>
Enabling the Automation control switch makes the morphing slider exposed and available for automation read.

When engaged, keep in mind only the morphing slider value is used for automation, and other parameter values are ignored. This behavior is intended and necessary to prevent any parameter conflicts that would otherwise occur.

As a consequence of this, you need to make sure the Automation switch is engaged when mapping the morphing slider mapped to a control surface hardware knob or slider. On the opposite, when not engaged, the plug-in will listen for any parameter automation, except the morphing slider.

19 - Preset Management

From the Plug-in interface

A-B Sections

A plug-in features two preset sections : A & B. Clicking on the slot of a specific section reaches the shared preset bank. From the preset management window you can select the preset you want to recall in the specific preset section.

Save

Save replaces the selected preset by a new one under the same name featuring the current settings. If you want to keep an existing preset without your new modifications, just select an empty place into the preset list, enter a new name for this modified preset featuring the current settings and press Save.

Recall

Once a preset is selected from the preset list it must be explicitly loaded into the section A or the section B by using the recall button. A preset is effective only after it has been recalled.
Double-clicking on the preset name from the list, reloads the preset into the selected slot.

AB Slider

This horizontal slider has no unity nor specific value display. It allows to morph current settings between two loaded presets. A double-click on one side of the slider area toggles between full A and full B settings. The results of an in between setting can be save as a new preset.

From the Preset Management Window

The Preset Management Window features three preset banks:
• The Factory bank gathers presets that can't be edited by users.
• The User bank is dedicated to the users presets.
• The Global bank features presets for A, B and morphing sections. A single global preset includes A and B section content and the morphing slider position.

A Preset can directly be recalled into the preset section selected by the morphing slider position, by double-clicking on its name on the list. The preset lists can be filtered. This filter is applied to any preset information such as name, description, author, comments or key words.

<u>Recall A</u>

recalls the selected preset into the corresponding section.

<u>Recall B</u>

recalls the selected preset into the corresponding section.

<u>Copy A and Copy B</u>

buttons allow to easily create a variation around a preset.

<u>Update</u>
allows to save the current settings for the selected preset.

<u>New</u>

creates a new preset in the list.

<u>Duplicate</u>

creates a new preset in the list from the selected one.

<u>Edit</u>

gives access to the specific windows which allows to change preset name, description, key words.....

<u>Delete</u>

suppresses the selected preset.

<u>Export</u>

creates a file reflecting the content of the preset bank.

Import

adds existing presets into the preset bank.

Ordering arrows

orders the presets into the list.

The preset protection if engaged, allows only its original modification author to uncheck and edit. So you can protect your presets in a multi-user configuration. Protected presets can only be modified using the session of their creator. If used in another user session they can only be imported or deleted.

20 - Credits

Spatialisateur and Spat~ are trademarks of Ircam and Espaces Nouveaux.
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Ircam R&D Director : Hugues Vinet
Collection Manager: Frederick Rousseau

IRCAMTOOLS
Collection Manager for IRCAM: Frederick Rousseau
Collection Manager for Flux:: Gael Martinet

IRCAMTOOLS SPAT, VERB

Flux::
Head software engineering: Gael Martinet
Developpers: Gael Martinet, Samuel Tracol, Siegfried Hand
Designer: Nicolas Philippot
Contributions: Lorcan Mcdonagh