# Spatialisateur

## Ircam / Espaces Nouveaux

## User Manual

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# 1 Licence & Credits

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This manual originally was written by Jean-Marc Jot, and was produced under the editorial responsibility of Marc Battier.

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This document is a reference manual for version 4.x of **Spat**∼ , which runs on Max/MSP® version 5.x.

**Spat**∼ Version 4.x design and implementation was ensured by Thibaut Carpentier and Rémy Muller.

In this manual, you will be introduced to the general principles and conventions adopted in the design of **Spat**∼ . It is expected that you be familiar with Max/MSP® , and somewhat familiar with signal processing and acoustics. This introduction manual can, however, be read without extensive knowledge of these topics, and is divided as follows:

# Contents

This introduction manual does not go into detailed operational description of specific **Spat∼** objects in Max/MSP® . Such information is contained in the help-patch associated to each object, which is accessible through the standard on-line help mechanism provided in Max/MSP® (the help patch pops up whenever you alt-click on the corresponding box in a patch).

For a new user, we suggest reading this introduction manual first, and then trying the tutorial patches (found in the « tutorials » directory). Before starting to work with **Spat∼** , it may also be useful to open some of the main help-patches.

# 2 Overview

## 2.1 The Spatialisateur project

The Spatialisateur project started in 1991 as a collaboration between Espaces Nouveaux and Ircam. Its goal is to propose a virtual acoustics processor which allows composers, performers or sound engineers to control the diffusion of sounds in a real or virtual space. This project stems from research carried out within the Ircam room acoustics laboratory on the objective and perceptive characterization of room acoustic quality. It also incorporates research done at Télécom Paris on digital signal processing algorithms for the spatialization and artificial reverberation of sounds. The Spatialisateur is integrated into the Max/MSP® environment and runs on Mac OSX (Universal Binary) and Windows XP/Vista platforms.
**Spat∼** is an effort to organize and optimize the experimental patches developped in the Spatialisateur project, in order to make them accessible to musicians and researchers who work with Max/MSP® . The current release allows reproduction on multi-channel loudspeaker systems in studios or concert halls. It also integrates 3D stereo reproduction modes for headphones (binaural) or 2 loudspeakers (transaural), as well as Vector Based Amplitude Panning (VBAP, [Pul97]) and Ambisonics.

## 2.2 Features of Spat∼

**Spat∼** is a configurable real-time spatial processor integrating the localization of sound events with room acoustic quality. Since it is based on a modular organization, it provides both a complete application and a library of Max/MSP® objects for real-time spatial processing of sounds. The processor receives sounds from instrumental or synthetic sources, adds spatialization effects in real time, and ouputs signals for reproduction on an electroacoustic system (loudspeakers or headphones). The general approach taken in **Spat∼** can be characterized by the fact that it gives the user the possibility of specifying the desired effect from the point of view of the listener rather than from the point of view of the device or process used to generate that effect. Practically, this results in the following three general features:

- The system can be configured according to the reproduction setup. **Spat∼** is not designed to work in a specific reproduction format. The number of input/output channels of some processing modules can be configured to fit various reproduction setups (a multichannel system, a pair of loudspeakers or headphones). The desired effect is specified independently from the reproduction setup and is, as much as possible, preserved from a reproduction mode or listening room to another. When the listening room is not acoustically neutral, **Spat∼** can take into account measurements made at a reference listening position in order to automatically perform the necessary correction of the processed signal.
- To allow for a global description of the reproduced effect, the temporal aspects (artificial reverberation) and the directional aspects (localization of sound sources and spatial content of the room effect) are integrated in a single processor. This allows to overcome the limitations of heterogeneous systems in which the localization of sound sources and the reverberation effect are generated with separate devices. It allows, for instance, to control more precisely and more intuitively the distance or proximity of sound events. From this standpoint, **Spat∼** can be seen as an extension of the system designed by John Chowning in the seventies [Cho71].

- A control interface is proposed which allows to specify the desired effect using perceptual terms rather than technical terms. The artificial room effect can be controlled in terms of independent perceptual attributes derived from psychacoustic research carried out at Ircam. This method does not suffer from the constraints that would inevitably result from a control strategy based on a geometrical and physical description of the enclosure. Since each perceptual attribute is linked to an objectively measurable criterion of the transformation of the sound, this control interface allows to imitate the acoustics of an existing room. It then allows to interpolate or extrapolate continuously towards a different acoustic quality, going through natural-sounding transformations.

Finally, since **Spat**∼ is designed in an object-oriented programming environment (Max/MSP® ), it can be considered as a library of elementary modules which can be used individually (for instance: artificial reverberator, multi-channel panpot, parametric equalizer). This modularity allows one to build versions of the spatial processor for different applications or with different computational costs, depending on the desired flexibility in controlling the reproduced effect and the available digital signal processing resources.

# 3 General description of Spat∼ modules

The library of Max/MSP® objects which compose **Spat**∼ is divided in two main categories of objects: DSP objects and control objects.

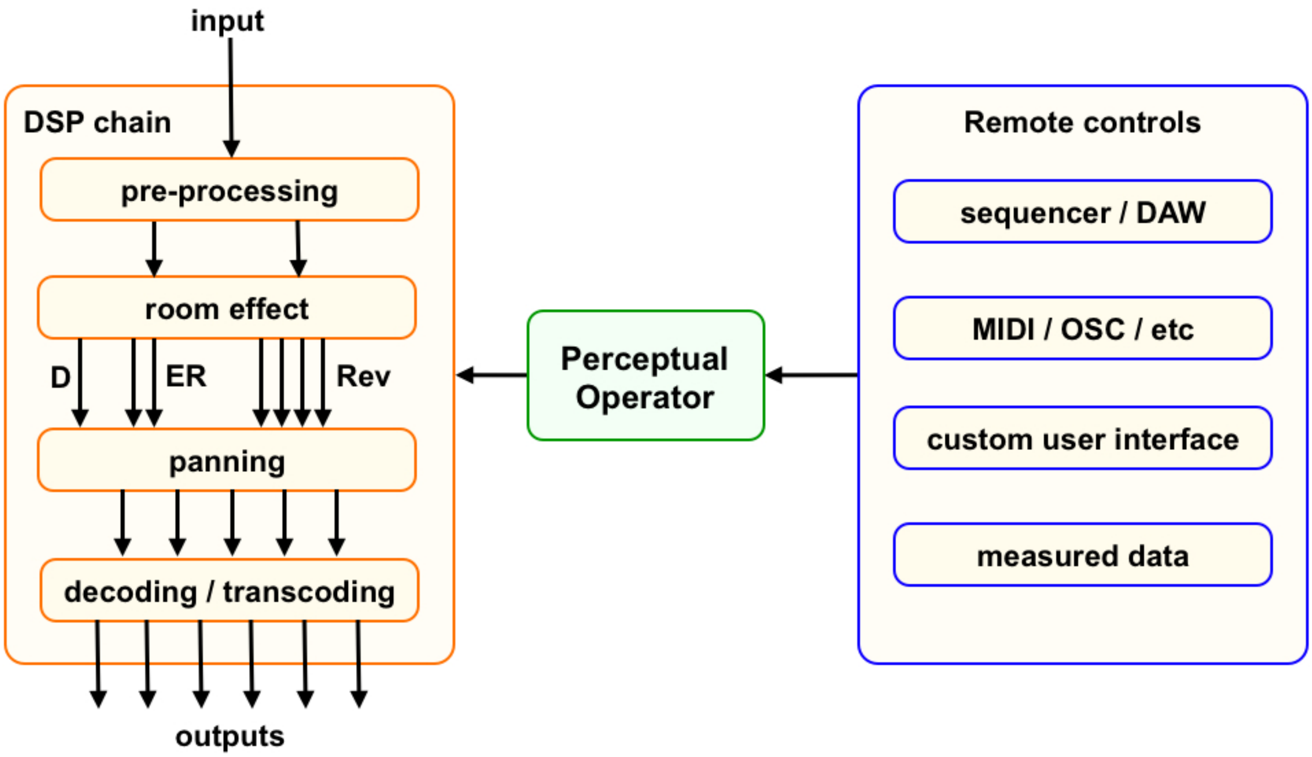


Figure 1: General structure of **Spat**∼

## 3.1 DSP objects

The signal processing in **Spat**∼ is organized in four stages corresponding to four main DSP objects (whose names end with the "∼" symbol, following the convention for signal objects in Max/MSP® ). Each DSP object receives and transmits one or more signals and receives control messages on its leftmost inlet. These messages are lists of symbols or numbers, according to a given syntax.

## 3.2 High-level control objects

These objects allow to define a selection of controls for modifying several DSP parameters simultaneously (possibly in different DSP objects). As before, a high-level control can be actuated through sliders or number boxes, or updated by an incoming control message, according to a given syntax. The Max/MSP® programming environment allows the user to easily build a sequencer, automatic process or higher-level (remote) control interface which can send orders to **Spat**∼ .

# 4 Signal Processing

The signal processing in **Spat**∼ is divided in four successive stages, separating directional effects from temporal effects :

- Pre-processing of input signals (Source) 4.1
- Room effects module (reverberator) (Room) 4.2
- Directional distribution module (Panning) 4.3
- Output equalization module (Decoding) 4.4

The reunion of these four modules constitutes a full processing chain from sound pickup to the ouput channels, for one source or sound event. Each one of these four modules works independently from the others (they have distinct control syntaxes) and can be used individually. Each module has a number of attributes that allow to vary its configuration (for instance, varying complexities for the room effect module or different output channel configurations for the directional distribution module). This modularity allows easy configuration of **Spat**∼ according to the reproduction format, to the nature of input signals, or to hardware constraints (e.g. available processing power).

## 4.1 The source module: pre-processing of input signals

This optional module receives the input signal(s) of **Spat**∼ and provides the two signals received by the Room module, which describe the virtual source:

- the face signal contains the acoustic information scattered by the source in the direction of the listener, which is used by Room to reproduce the direct sound;
- the omni signal contains the average acoustic information scattered by the source in all directions, which is used by Room to feed the artificial reverberation algorithm.

### 4.1.1 Pre-delay and Doppler effect

The main role of the Source module is to generate a "pre-delay" in order to reproduce, if necessary, time lags existing between the signals coming from several sound sources situated at different distances from the listener. A continuous variation of this pre-delay naturally reproduces the Doppler effect (apparent pitch shift) associated to the movement of a particular source. Low-pass filtering to reproduce the effect of air absorption can also be included in Source.

### 4.1.2 Pre-equalization and directivity of sound sources

If necessary, the Source module can incorporate additional processing. For instance, a spectral correction of the face and omni signals can be included, if it is made necessary by the position of the microphone(s) relative to the instrument, or by the nature of the input signal(s).

Nota Bene: From a general point of view, it is almost always necessary to apply a spectral correction to the two input signals of Room, to account approximately for the radiation characteristics (directivity and orientation) of the virtual sound source [[Moo83], [War90]]. Since the directivity of a sound source is usually frequency-dependent, spectral corrections are necessary. This is why two input equalizers are already incorporated in Room (see below), but it can be useful, in some cases, to add a specific pre-equalization in Source.

## 4.2 The Room module: room effect synthesis

The Room module of **Spat**∼ is an artificial reverberator allowing room effect synthesis and control in real time, based on digital signal processing algorithms under license from France Télécom [Jot92]. The Room module can be made in several versions having different complexities. This allows the user, according to the application, to make the best use of the available processing resources.

The Room module receives two signals and outputs several signals. The two input signals are the direct signal and the room signal. If a Source module is connected to the inputs of the Room module, the direct and room signal inlets should respectively receive the face and omni outputs from the Source module (see section 4.1).

The ouput signals are divided into three groups:

- center: the signal containing the direct sound
- sides: two signals, left and right, containing the early (oriented) room effect
- surround: N signals containing the later (diffuse) room effect (N is called the number of internal channels)

This format allows to control the directional aspects of the artificial room effect irrespective of the reproduction setup (see description of the Panning module below). This output format is directly compatible with the 3/2-Stereo format, derived from the video and motion-picture industry, comprising three front channels and two surround channels [[The93]]. Assuming a frontal direct sound, the outputs of the Room module can be directly reproduced on a 3/2-Stereo (or even 3/4-Stereo) loudspeaker setup, with no additionnal processing. The 3/2-Stereo format is downward compatible with loudspeaker setups comprising a lower number of channels, including conventional (2/0) stereo or 4-channel systems (2/2- or 3/1-Stereo) [[STN$^+$93]]. The output signals of Room can be readily downmixed for reproduction on these loudspeaker setups. The role of the Panning module is to provide dynamic matrixing of these output signals, allowing non-frontal localization of the sound source or sound event.

The functional description below follows the division of the graphical control interface Room into three main parts: time structure, equalization, and decay time.

### 4.2.1 Time structure

The response of the Room module is divided into four time sections:

- direct : The direct sound is taken as the time reference (0 ms) for the description of the artificial room effect that follows it.
- early : This section contains the discrete early reflections, shared between the two sides signals of Room. The date and intensity of each reflection can be controlled individually.

- cluster: This section contains a denser pattern of diffuse later reflections which are equally shared between the surround signals.
- reverb: This section contains the late diffuse reverberation, divided into uncorrelated signals of equal energy sent to the surround outputs. The late reverberation decays exponentially with time according to the decay settings (see section 4.2.2).

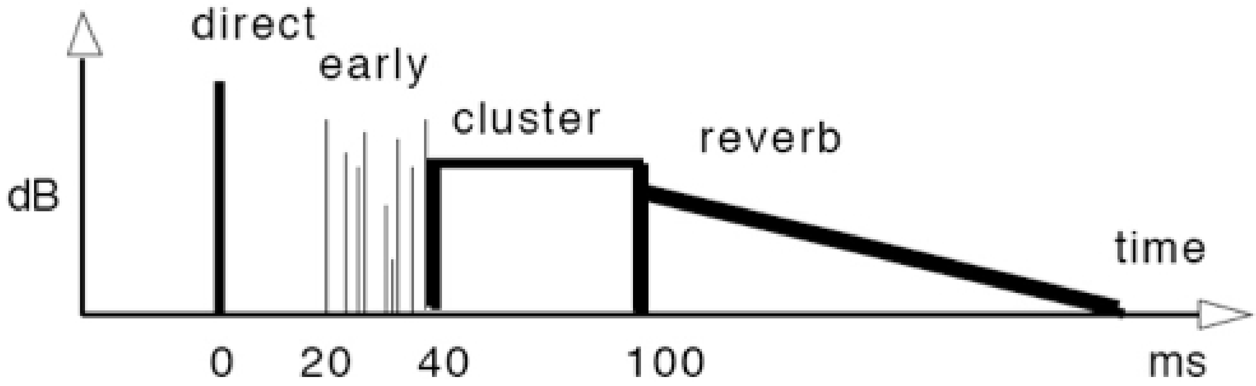


Figure 2: Time structure of generic room effect

The energies (in dB) and time limits (in ms) of these four sections can be controlled independently, which can be exploited to imitate the acoustics of rooms of various sizes. Modifying the limits of the early section sets default values for the dates of the early reflections, which can then be tuned individually. The three sections of the room effect can overlap, but the algorithm imposes some constraints (the lower limits must follow the chronological order given above, and the cluster section cannot be shorter than the early section).

This is a conventional way of describing the room effect [[Moo83], [Jot92]], except for the separation of the early room effect into two sections, resulting in the intermediate packet of reflections "cluster". The default time limits are roughly set to 20-40 ms for early reflections, 40-100 ms for later reflections, 100 ms and beyond for late reverberation. This generic room effect model is derived from studies of the perceptual characterization of the room acoustic quality of concert halls [JKWW92].

These psychoacoustic studies led to the conception of a user interface where each control is more directly related to the listeners's perception than the energies of the different time sections. A "high-level" user interface is described which includes a perceptual control panel for describing the artificial room effect.

Nota Bene: The complexity (and, consequently, the processing cost) of the artificial reverberation algorithm can be reduced by simplifying the time structure described above, at the expense of reduced flexibility in controlling the synthesized room effect. The most natural simplification consists in dropping the intermediate cluster section, in which case the reverberation starts decaying exponentially right after the early reflections. Further simplification is obtained by dropping the early section, for applications where controlling the early reflections is not of interest. Of course, dropping the reverb section to keep only the early and cluster sections is possible too, as well as keeping only the early section.

### 4.2.2 Decay time

The decay time control sets the exponential decay rate of the late reverberation (reverb section) as a function of frequency. The decay time is measured in seconds and defined as the time it takes for the late reverberation to drop 60 dB below its initial level, after an interruption of the input signal. The decay time is controlled in the same way as an equalizer curve, either globally or in three separate frequency bands with controllable transition frequencies, and can be varied from 0.1 s to 10 s. A toggle switch allows to momentarily set the decay time to infinite, then back to its initial setting.

## 4.3 The panning module

This module receives signals according to the output format of the Room module: one center channel, two sides channels, and N surround channels (containing respectively the direct sound, the early reflections and the diffuse reverberation). Panning can be configured to deliver signals for feeding the loudspeaker system, and allows dynamic control of the apparent source localization with respect to the listener. From a more general point of view, the Panning module can be considered as a conversion matrix which receives a 3/2-Stereo (or 3/4-Stereo) signal and outputs loudspeaker signals for systems of 2 to 64 channels. The control interface of the Panning module is divided in two sections: source localization and loudspeaker system configuration.

### 4.3.1 Source localization

Modifying the source position affects the distribution of the intensity of the center channel (direct sound) among the loudspeakers. The method used is derived from Chowning's algorithm [[Cho71], [Bos90]]. The distribution of the surround channels (containing the diffuse reverberation) is not affected by the source localization control. However, the Panning module extends Chowning's method by allowing for the two side channels (containing the early reflections) to rotate along with the center channel, according to the azimuth control.

This method of distributing early reflections improves the reproduction of the room effect and of the apparent distance of the source, yet without involving a geometrical description of the virtual room. Geometrical methods as proposed in [[Moo83], [Jot92], [Gar92]] could readily be implemented in Room module by allowing independent panning and continuous variation of the dates of the early reflections, but this additional complexity can be avoided by the panning method used here, while preserving the main perceptual effects.

Nota Bene: Modifying the source distance in the Panning module only affects the distribution of signals to the different loudspeakers when this distance becomes shorter than that of the loudspeakers, but does not affect the total intensity of the sound. Larger distances should be reproduced in connection with the room effect. This can be done in **Spat**∼ by combining the effects of the Panning, Room, and Source modules. A "high-level" control method for this purpose is described in section 5.

## 4.4 The Decoding module

This module can be used to apply spectral and temporal corrections to the output signals of the Panning module, before sending these signals to the loudspeakers. Each channel undergoes an adjustable time delay and a parametric equalization.

The filters can be used to equalize the frequency response of each loudspeaker separately (note that more elaborate equalizers could be used). The delays can be used to make the signal propagation delays of all channels identical. Differences between these propagation delays can be introduced both by the geometry of the loudspeaker setup and the hardware configuration of the host computer.

Nota Bene: These corrections should be made so that all loudspeakers should be perceived, from the reference listening position, as being situated at the same distance and having approximately the same frequency response. The necessary delay and filter adjustments can be automatically derived from impulse response measurements made at the reference position for each loudspeaker (a software allowing to do this exists at Ircam but has not yet been implemented in the **Spat**∼ library).

# 5 High-level control : the Spat Oper

By associating the signal processing modules described in the previous chapters, a complete spatial processing chain can be constructed, starting from the captured or synthesized sounds to the distribution of the processed sounds to the loudspeakers.

For the musician or the sound engineer, it is preferable to use a control interface that is not simply the reunion of the low-level control interfaces of the different signal processing objects, but rather is made of a selection of high-level command controls.

The Max graphical programming environment allows the user to build his/her own custom control interface, capable of sending control messages to the DSP objects, according to each object's control syntax.

spat.oper is a high-level control interface of this kind. Its role is to provide a reduced set of controls which describe the reproduced effect through quantities that are intuitive to the user and perceptually relevant from the point of view of the listener. The core of spat.oper is a perceptual control module based on research carried out in the Ircam room acoustics team on the objective and perceptual characterization of room acoustic quality [[JKWW92], [Lav89]]. This control interface can be set to imitate the acoustics of an existing room and allows to interpolate or extrapolate naturally towards a different acoustic quality.

The graphical user-interface in spat.oper is divided into three kind of tabs:

- Source tabs which let you control the perceptual description of the acoustic quality of the source, localization of the virtual source, radiation of the virtual source (directivity and orientation),
- Reverb tabs which let you control the perceptual description of the acoustic quality of the reverb,
- Output tab which let you configure of the loudspeaker system (position, equalization, time alignment).

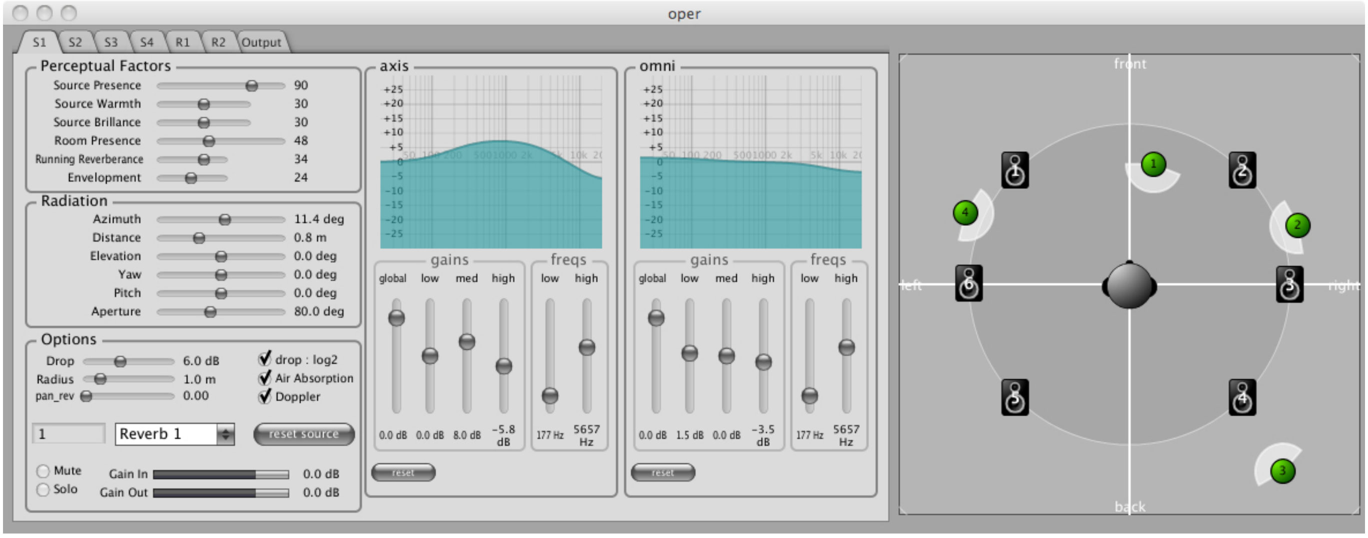


Figure 3: Spat Oper : Source tab

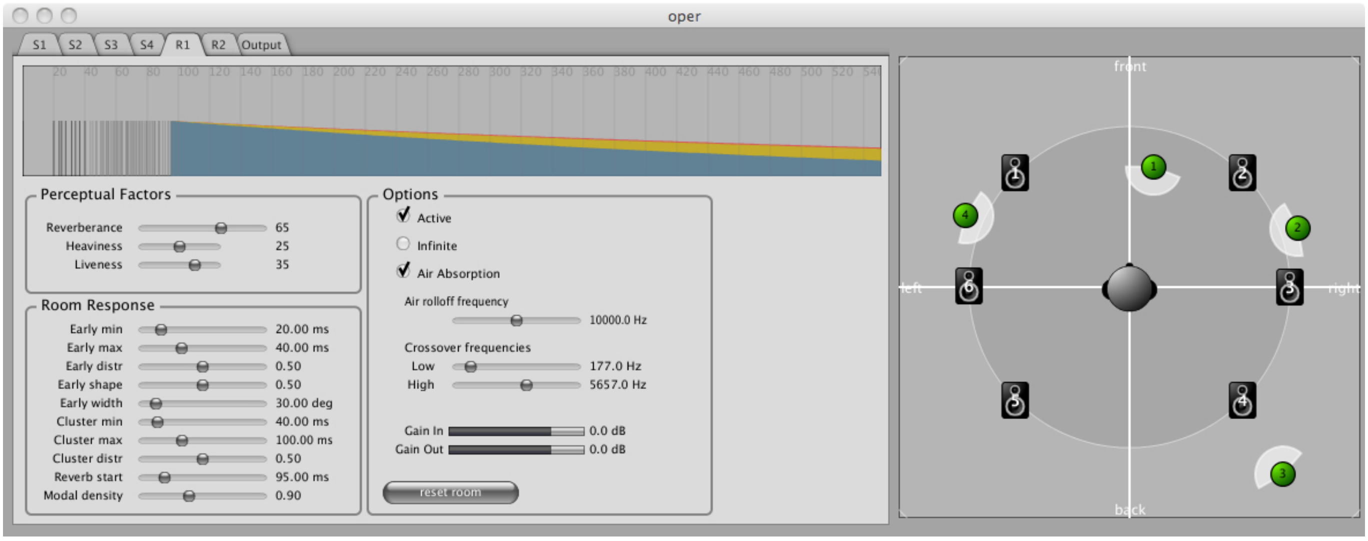


Figure 4: Spat Oper : Reverb tab

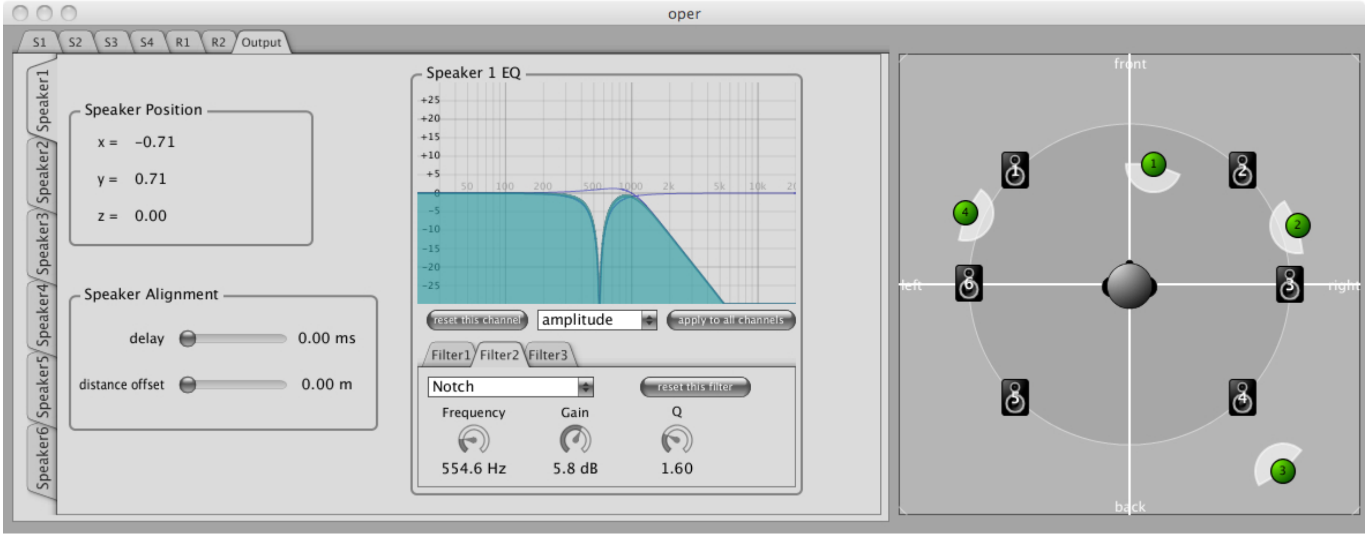


Figure 5: Spat Oper : Output tab

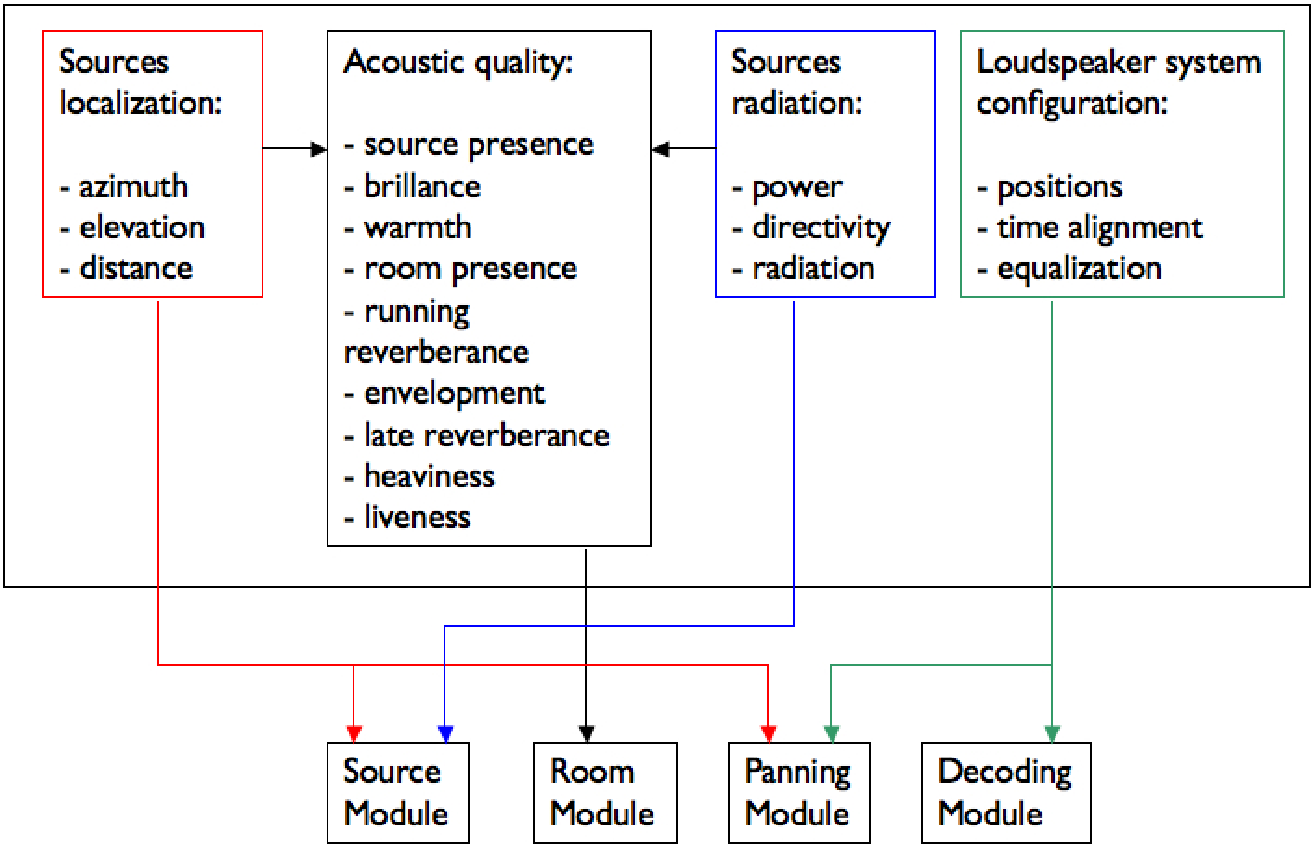


Figure 6: Functional description of the Spat Oper

## 5.1 Perceptual control of the acoustic quality

### 5.1.1 Definition of acoustic quality

The term “acoustic quality” is used in spat.oper to describe globally the transformations undergone by the message radiated by a sound source before it reaches the listener. In a natural situation with a sound source and a listener in a room, the acoustic quality is influenced by:

- the geometry and acoustic properties of the listening room and obstacles,
- the positions of the listener and the sound source in the room,
- the orientation and directivity of the sound source.

Two remarks should be made regarding the definition of the acoustic quality used here:

- If several sound sources are present in the same room at different positions or with different orientations or directivity patterns, the acoustic quality is generally different for each one of them.
- In this definition, it is assumed that the source is in front of the listener. This means that, although the distance from the source to the listener influences the acoustic quality, the direction where the source is located with respect to the listener is not considered part of the acoustic quality.

In general, the acoustic quality, as defined here, changes when the source rotates around the listener. Reproducing such changes with a spatial processor requires manipulating a geometrical and physical description of the virtual room, the virtual source and the receiver. Such a description can be stored in a computer model which computes the DSP parameters of the spatial processor whenever a physical or geometrical parameter changes (e.g. the position of the source or the dimensions of the room).

Although this approach is possible [[Moo83], [Gar92]], it has a number of disadvantages in the context of real-time musical applications:

- The control parameters are not perceptually relevant: the perceived effect of varying a geometrical or physical parameter may often be unpredictable (sometimes non-existant).
- This control method is limited to reproducing physically feasible situations. Even if the modelled room is imaginary, the laws of physics limit the range of feasible effects. For instance, in a room of a given shape, modifying wall absorption coefficients to modify the decay time will cause a change in the level of the room effect at the same time.
- Updating the DSP parameters involves a complex control process (usually involving the computation of a source image distribution to compute the dates and energies of room reflections [[Moo83], [Gar92]]).

The approach adopted in the Spatialisateur project allows to design a spatial processor which does not rely on a physical and geometrical description of the virtual environment for synthesizing the room effect. Instead, the proposed user-interface is directly related to the perception of the reproduced sound by the listener. In a musical context, this approach allows to immediately take the acoustic quality into account at the composition stage (by integrating perceptual attributes in the score, for example), without referring to a particular electroacoustic setup or to the place where the work will actually be performed. Additionally, the real-time computational efficiency is maximized since processing is focussed on the reproduction and control of perceptually relevant attributes.

### 5.1.2 The perceptual factors

In the proposed control interface, the acoustic quality is described in terms of mutually independent perceptual factors. These perceptual factors were derived from psychoacoustic research on the characterization of the acoustic quality of concert halls [[JKWW92], [Lav89]]. As a result of these studies, each perceptual factor is related to a measurable objective criterion. This allows for the translation of the perceptual factors into DSP parameters, and allows to reproduce the acoustic quality of an existing room. The perceptual factors form the most relevant basis for controlling interpolation processes between different acoustic qualities.

The perceptual factors are manipulated by means of sliders which are scaled to account for the average sensitivity of listeners with respect to the different factors. They were given names by the research team and can be categorized as follows.

A group of 6 perceptual factors describing effects which depend of the position, directivity and orientation of the source. The first 3 are perceived as characteristics of the source. The next 3 are perceptually associated to the room:

- source presence: early sound (energy of direct sound and early room effect)
- source brilliance: variation of early sound at high frequencies
- source warmth: variation of early sound at low frequencies
- running reverberance: early decay time
- room presence: late sound: energy of later reflections and reverberation
- envelopment: energy of early room effect relative to direct sound

A group of 3 perceptual factors describing effects which are characteristics of the room:

- late reverberance: mid-frequency decay time
- liveness : relative decay time at high frequencies
- heaviness : relative decay time at low frequencies

A variation of the source presence creates a convincing effect of proximity or remoteness of the sound source. The term "reverberance" refers to the sensation that sounds are prolonged by the room reverberation. Late reverberance differs from running reverberance by the fact that it is essentially perceived during interruptions of the message radiated by the source. Running reverberance, on the contrary, remains perceived during continuous music.

Nota Bene: This perceptual control interface only affects the DSP parameters of the Room module. When spat.oper is used, the high and low transition frequencies are given fixed values everywhere within Room, for all spectral corrections and for the decay time as well.

## 5.2 Localization of the sound source and configuration of the loudspeaker system

The "localization" section gives access to the corresponding parameters of the Panning module (section 4.3), describing the direction and distance of the source. However, in spat.oper, the distance control can have two additional effects :

- It is linked to the duration of the pre-delay in the Source module (causing a Doppler effect and a high-frequency boost or cut reproducing air absorption).
- It is linked to the perceptual factor source presence (see section 5.1.2 above). This link allows to reproduce movements of a sound source in space with no limitation but those introduced by the reproduction setup (for instance, controlling the elevation of the sound source is not available with a horizontal loudspeaker setup).

The "output" section contains all the parameters describing the positions of the loudspeakers with respect to the reference listening position. These are the configuration parameters of the Panning module (section 4.3). This description is completed by the spectral and temporal correction to be applied to each output channel (the parameters of the Decoder module described in section 4.4).

## 5.3 Radiation of the sound source

As mentioned in section 4.1, the main effects due to the directivity and orientation of the sound source can be reproduced by spectral corrections of the direct sound and of the signal that feeds the artificial reverberation algorithm. These corrections are interpreted as a modification of the acoustic quality perceived by the listener, which causes an update of the displayed values of the perceptual factors.

The control of these effects takes the following form in spat.oper:

- source directivity index (as a "double shelving" curve),
- source power spectrum (as a "double shelving" curve),
- angular orientation of the source.

## 5.4 Source radius

One very important parameter of the Spat Oper interface is the "radius" associated to a given source.

### 5.4.1 Definition

The "radius" parameter defines the radius of a virtual sphere surrounding the listener.
When a sound source comes closer to the listener, its energy progressively increases. When the source reaches the "radius sphere", its energy no longer increases. It is clipped.
The "radius" parameter hence represents the minimum radius under which the sound level is limited. This provides a "safety area" around the listener in order to avoid over-amplified sounds for nearby sources.

### 5.4.2 Radius in 3D

The "radius" parameter is also associated to another phenomenon for 3D panning techniques (such as VBAP, 3D-Ambisonic, and binaural or transaural) :
When sound sources comes close enough to reach the "radius sphere", they will smoothly slide over the sphere surface (and thus pass over the listener head). This guarantees a smooth sound trajectory for sources that "crosses" the sphere.

### 5.4.3 Radius and distance attenuation law

When a sound source comes closer (resp. further) to the listener, its energy (i.e. its source presence) increases (resp. decreases). The attenuation law quantitatively characterizes this phenomenon.
The attenuation law is governed by several parameters :

- the drop model represents the type of attenuation (linear or logarithmic)
- the drop value represents the slope of the attenuation law. For instance a 6dB drop means that the energy of the source is reduced by 6dB when the distance of the source is multiplied by 2.
- the actual distance of the source
- the "radius" parameter as described above
- the energy of the source (i.e. the "source presence" perceptual factor)

As a matter of fact all these parameters are inter-dependent. It is thus very important to properly initialize them, and in the proper sequence, so that the distance attenuation law can be correctly applied :

- first set the value of the "radius" parameter (this value depends on what kind of proximity effects you want to create)
- then set the drop (i.e. the type of drop model, and the drop value). By default **Spat**~ uses a logarithmic 6dB attenuation law which corresponds to a natural attenuation law in free-field conditions
- move the source so that its distance is equal to the "radius"
- now, set the source presence value. This value will then correspond to the maximum reachable sound level

One have to keep in mind that the source presence parameter has a limited range : it spans from 0 to 120 (which corresponds to an energy range from -40dB to 0dB). It is thus important to carefully choose the maximum source presence (i.e. the source presence on the "radius sphere") otherwise the useful variation range may be limited.

# 6 Spectral equalizers in Spat∼

All spectral corrections in **Spat**∼ are performed with the same signal object, called spat.hlshelf∼. This is a "double shelving filter" which was designed specifically for **Spat**∼ , and allows to boost or cut three frequency bands separately and control the high and low transition frequencies. spat.hlshelf∼ is made of a second-order recursive filter with a particular method of computing the coefficients, in order to achieve symmetrical boost and cut curves (on a dB scale), as well as symmetrical response curves on a log frequency scale. This control method was first implemented in a Max object called coefhlshelf1∼, written by Zack Settel and included in the "jimmies" library.

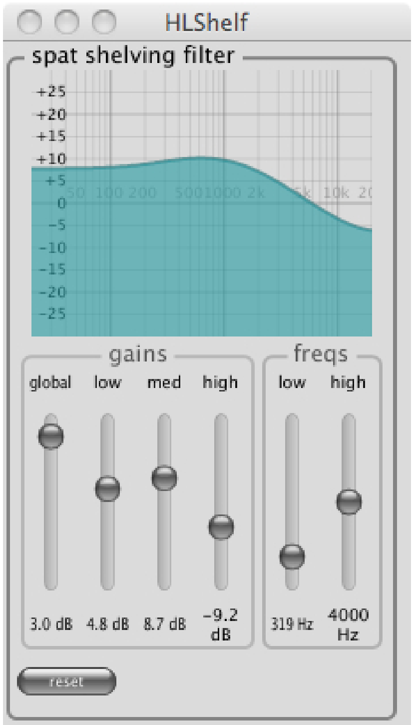


Figure 7: **Spat**∼ double shelving filters

Note that more elaborate parametric equalizers can be built, for instance by cascading a double shelving filter with parametric band-pass second-order equalizers. **Spat**∼ 4 does not make use of equalizers more elaborate than the double shelving filter, so this is left up to the user.

# 7 Coordinates system

**Spat**∼ 4 implements several coordinate systems, all defined from the listener's point of view (i.e. the listener is the origin of the coordinate system).

In Max/MSP® these different coordinate systems are :

- xyz : 3D Cartesian coordinates
- xy : 2D Cartesian coordinate within xy plane
- ade : azimuth, distance, elevation
- aed : azimuth, elevation, distance
- ad : azimuth, distance (2D within xy plane)
- az : azimuth only (distance = 1 m)
- spat3 : azimuth, distance, elevation format for backward compatibility with **Spat**∼ 3.x convention

**Spat**∼ 4 external objects can interpret messages formatted in any of these coordinate formats. Furthermore utilities objects are provided to perform conversions between the different coordinate systems.

## 7.1 Cartesian coordinates (xyz)

X axis on the right of the listener
Y axis in front
Z axis to the top

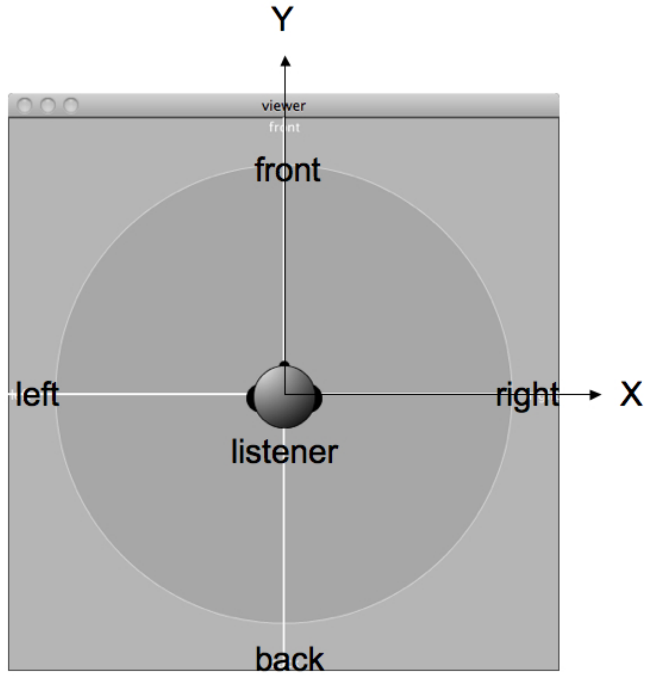


Figure 8: Top view (xy plane)

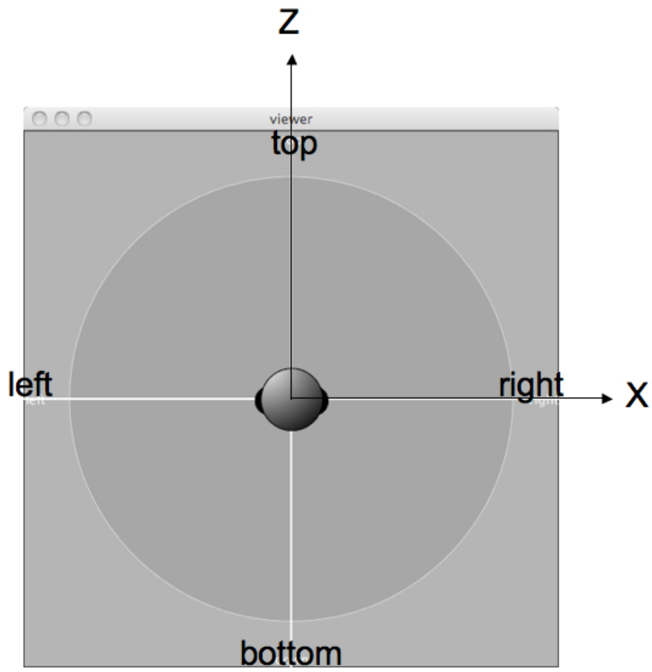


Figure 9: Front view (xz plane)

## 7.2 Navigational coordinates (ade)

The default coordinate system in **Spat**∼ 4 is called the navigational coordinate system. Positions are expressed in term of azimuth, elevation and distance (format "ade" in Max/MSP® ).

Azimuth is measured toward the y axis, with 0° in front of the listener.
Elevation is 0° within xy plane. +90° on top, -90° bottom.

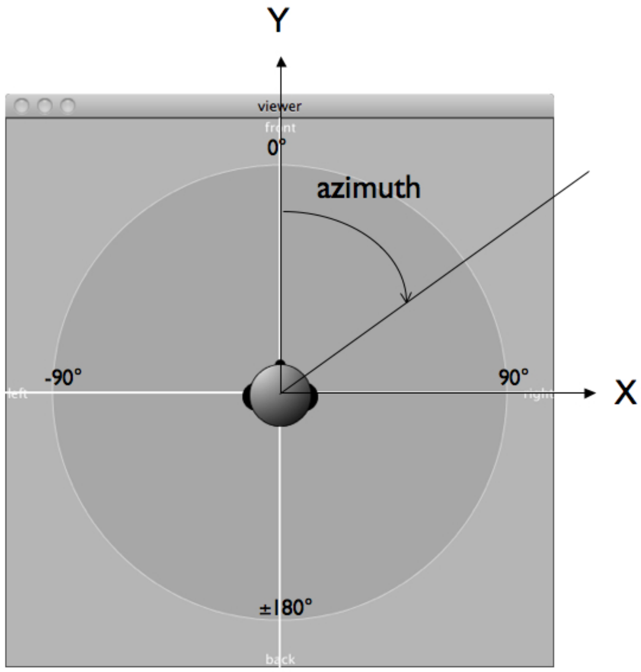


Figure 10: azimuth convention

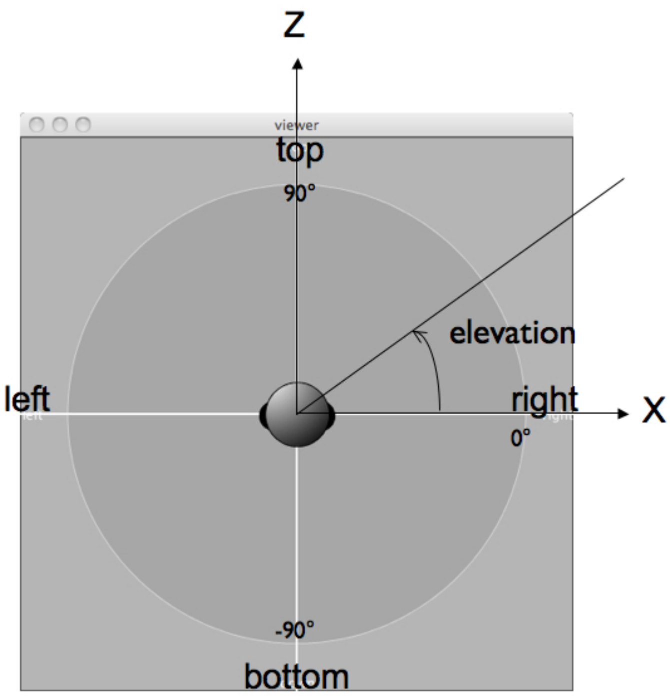


Figure 11: elevation convention

## 7.3 Spat∼ 3.x coordinate system (spat3)

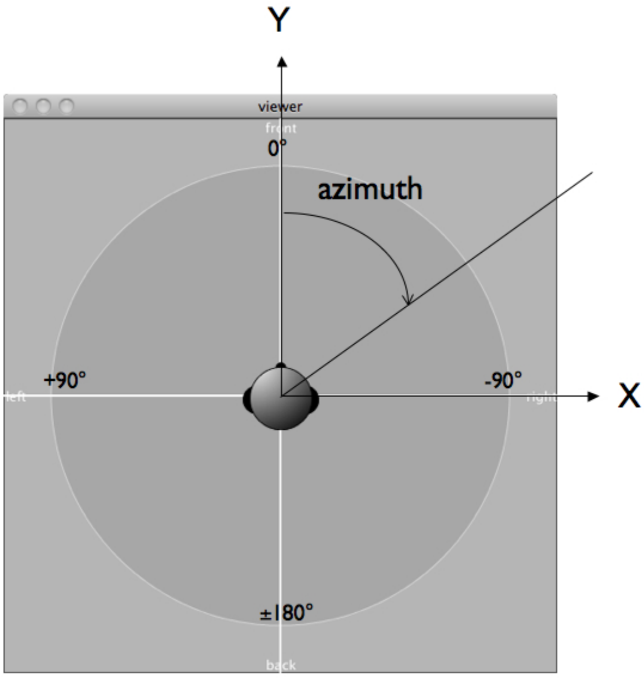


Figure 12: **Spat**∼ 3.x azimuth convention

# 8 Contacts

The use of this software and its documentation is restricted to members of the Ircam software users group.

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