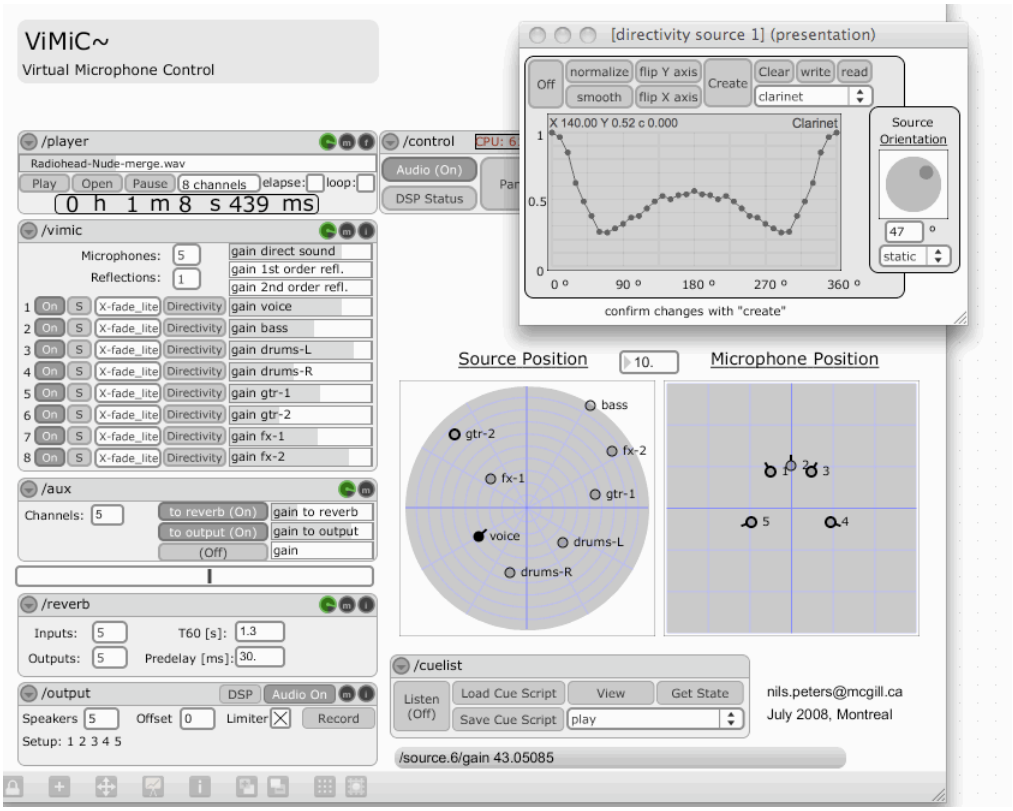
USER MANUAL

ViMiC - Virtual Microphone Control
for Jamoma and Max/MSP



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Chapter 1

Introduction

1.1 Acknowledgment

This work was funded by the Canadian Natural Sciences and Engineering Research Council (NSERC) and the Centre for Interdisciplinary Research in Music, Media and Technology (CIRMMT).

1.2 Requirements

- Max/MSP 5.1.4 or newer
- MacOS 10.5.8 or newer[1]

1.3 Installation

ViMiC for Max 5 is available in the github repository:
http://github.com/Nilson/ViMiC-and-friends

ViMiC for Max 4.6.3 is available in the Jamoma Max 4 maintenance repository:
https://jamoma.svn.sourceforge.net/svnroot/jamoma/branches/maintenance_max4_support

A tutorial to access and work with the Jamoma github repository is available at:
http://redmine.jamoma.org/wiki/jamoma/Working_with_GIT
To verify correct installation, load the help-patch *jmod.sur.vimic∼.maxhelp*, located at */ViMiC/sur.vimic∼/*. There should not be any errors in the max window.

[1] The external is also available for Windows OS (jcom.vimic∼.mxe) but has not been extensively tested.

1.4 Principles

ViMiC is a computer-generated virtual environment, where gains and delays between a virtual sound source and virtual microphones are calculated according to their distances, and the axis orientations of their microphone directivity patterns. Besides the direct sound component, a virtual microphone signal can also include early reflections and a late reverb tail, both dependent upon the sound absorbing and reflecting properties of the virtual surfaces.

1.4.1 Source - Microphone Relation

Sound sources and microphones can be placed and moved in 3D as desired. Figure 1.3 shows an example of one sound source recorded with three virtual microphones. A virtual microphone has five degrees of freedom: (X, Y, Z, yaw, pitch) and a sound source has four: (X, Y, Z, yaw). The propagation path between a sound source and each microphone is accordingly simulated. Depending on the speed-of-sound c and the distance d_i between a virtual sound source and the i-th microphone, time-of-arrival and attenuation due to distance are estimated. This attenuation function, seen in Eq. 1.1 can be greatly modified by changing the exponent q. Thus, the effect of distance attenuation can be boosted or softened. The minimum distance to a microphone is limited to 1 meter in order to avoid high amplification.

$$g_i = \frac{1}{d_i^q} \qquad d \geq 1 \tag{1.1}$$

As an alternative to this inverse proportional decrease, a second distance function is implemented. Here the decrease in dB per unit can be controlled by the k:

$$g_i = 10^{\frac{-k}{20} \cdot d_i} \tag{1.2}$$

Further attenuation happens through the chosen microphone characteristic and source directivity (see Fig. 1.4). For all common microphone characteristics, the directivity for a certain angle of incidence δ can be imitated by calculating Eq. 1.3 and applying a set of microphone coefficients from Table 1.4.1. By increasing the exponent w to a value greater than 1 will produce an artificially sharper directivity pattern. Unlike actual microphone characteristics, which vary with frequency, microphones in ViMiC are designed to apply the concept of microphone directivity without simulating undesirable frequency dependencies.

$$\Gamma = (a + (1 - a) \cdot \cos\delta)^w \qquad 0 \leq a \leq 1 \tag{1.3}$$

Characteristic	a	w
Omnidirectional	1	1
Subcardioid	0.7	1
Cardioid	0.5	1
Supercardioid	0.33	1
Hypercardioid	0.3	1
Figure-of-8	0	1

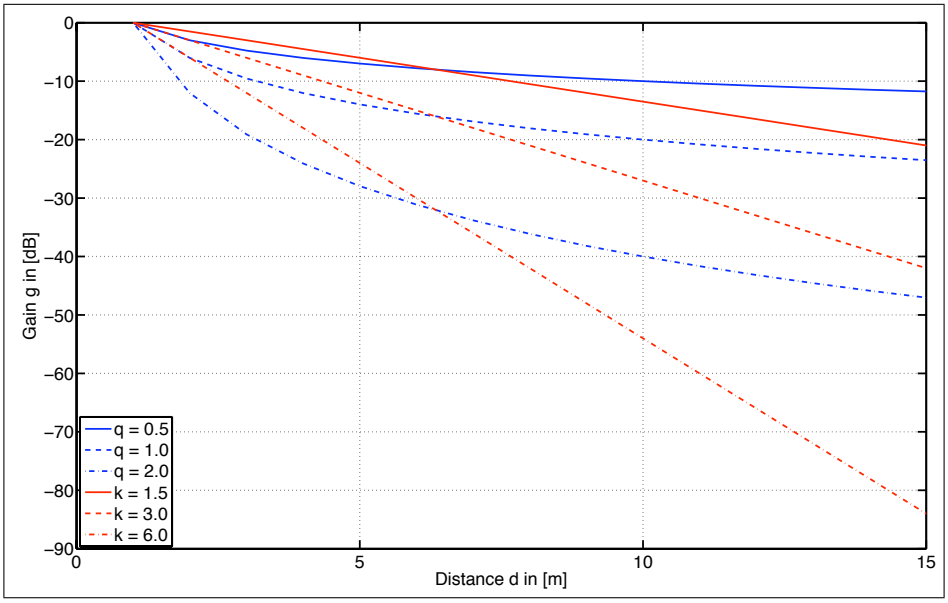


Figure 1.1: *Comparison of the distance attenuation functions with different settings: "inverse proportional decrease" model (blue lines, Equation 1.1) and "exponential decrease" model (red lines, Equation 1.2)*

Source directivity is known to contribute to immersion and presence. Therefore ViMiC is also equipped with a source directivity model. For the sake of simplicity, in a graphical control window, the source directivity can be modeled through a frequency independent gain factor for each radiation angle to a 1° accuracy.

Room model

ViMiC contains a shoe-box room model to generate time-accurate early reflections that increase the illusion of this virtual space and envelopment as described in the literature [9]. Early reflections are strong auditory cues in encoding the sound source distance. According to virtual room size and position of the microphones, adequate early reflections are rendered in 3D through the well-known image method [1]. For a general discussion of the image method, see, e.g. [4] and [3] for use in a rectangular room. Each image source is rendered according to the time of arrival, the distance attenuation, microphone characteristic and source directivity, as described in section 1.4.1. Virtual room dimensions (height, length, width) modified in real-time alter the refection pattern accordingly. The spectral influence of the wall properties are simulated through high-mid-low shelf-filters. Because larger propagation paths increase the audible effect of air absorption, early reflections in ViMiC are additionally filtered through a 2nd-order Butterworth lowpass filter with adjustable cut-off frequency.
Also, early reflections must be discretely rendered for each microphone, as propagation paths differ. For eight virtual microphones, 56 paths are rendered if the 1st-order

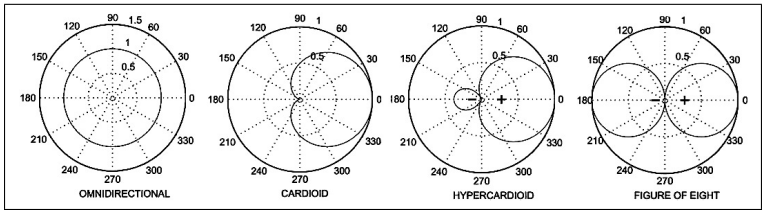


Figure 1.2: *Common microphone characteristics*

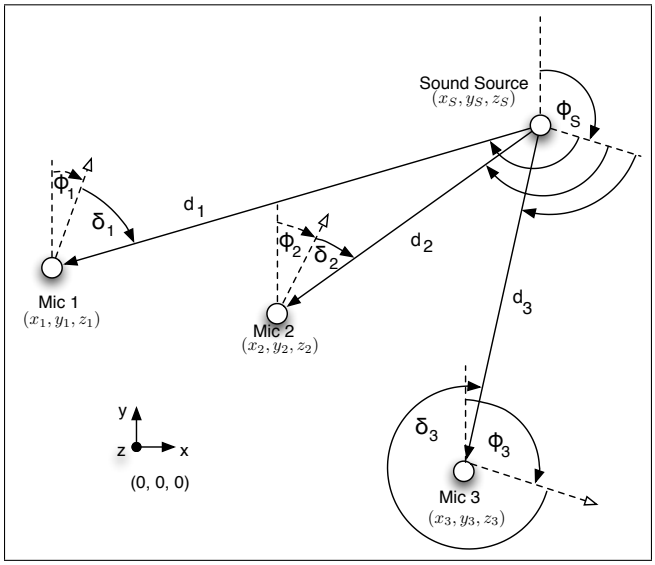


Figure 1.3: *Geometric example*

reflections are considered (8 microphones · [6 early reflections + 1 direct sound path]). Although time delays are efficiently implemented through a shared multi-tap delay line, this processing can be computationally intensive.
It has been shown that (in small rooms) increasing the level of individual early reflections in a simulated room environment the differences are first observed in sound coloration and then (after an increase of 2-4 dB) in spatial aspects, such as spatial impression [2]. Consequently in ViMiC the levels of direct sound and early reflections can be modified independently.

1.4.2 Late Reverb

The late reverberant field of a room is often considered nearly diffuse without directional information. Thus, an efficient late reverb model, based on a feedback delay network [6] with 16 modulated delay lines diffused by a Hadamard mixing matrix, is used. By feeding the outputs of the room model into the late reverb a diffused reverb tail is synthesized (see Fig. 1.4), for which timbral and temporal character can be mod-

ified. This late reverb can be efficiently shared across several rendered sound sources. For detailed information about the implementation of the Late reverb see chapter 2.5 on page 30.

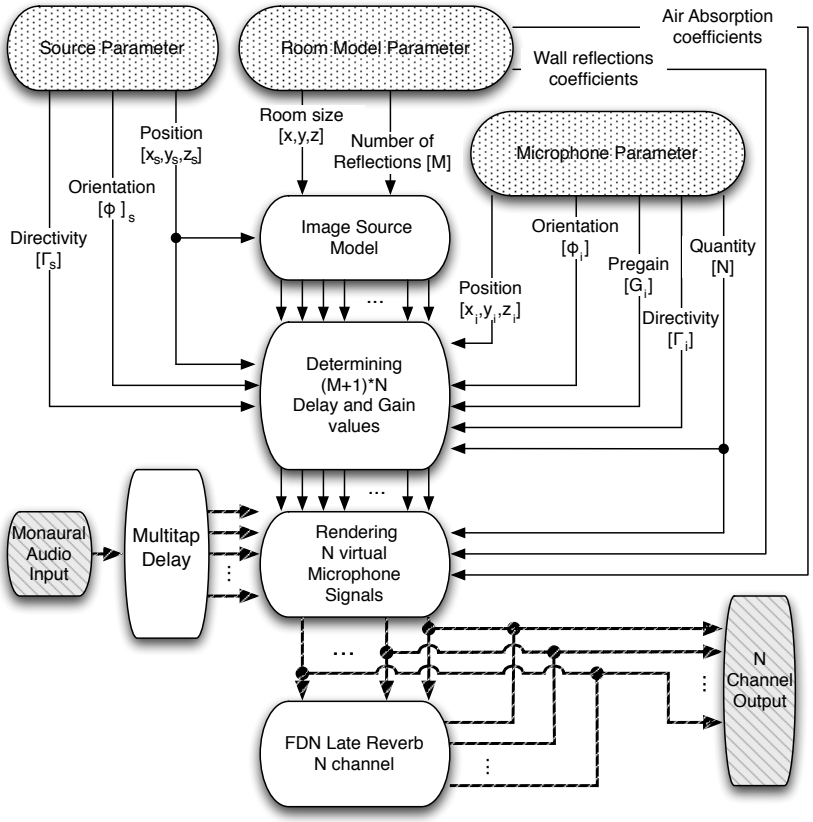


Figure 1.4: *Flowchart of the Max/MSP processing*

1.5 Moving Sources

In Figure 1.5 the sound source moved from (x, y, z) to (x', y', z'), changing the propagation paths to all microphones, and also, the time delay and attenuation. A continuous change in time delay engenders a pitch change (Doppler effect) that creates a very realistic impression of a moving sound source. Doppler effect might not always be desired. ViMiC accommodates both scenarios.

1.5.1 Rendering with Doppler effect

For each changed sound path, the change in time delay is addressed through a 4-pole interpolated delay-line, the perceived quality of which is significantly better than with an economical linear interpolation. To save resources, interpolation is only applied when moving the virtual sound source, otherwise the time delay is being rounded to the next non-fractional delay value. At $fs = 44.1$ kHz and a speed-of-sound of $c = 344$ m/s, the roundoff error is approximately 4 mm. Some discrete reflections might not be perceptually important due to the applied distance law, microphone characteristics, and source directivity. To minimize processor load, an amplitude threshold can be set to prevent the algorithm from rendering these reflections.

1.5.2 Rendering without Doppler effect

This render method works without interpolation: the time delays of the rendered sound paths remains static until one of the paths has been changed by more than a specified time delay. In this case, the sound paths of the old and the new sound position are cross-faded within 50 ms, in order to avoid strongly audible phase modulations.

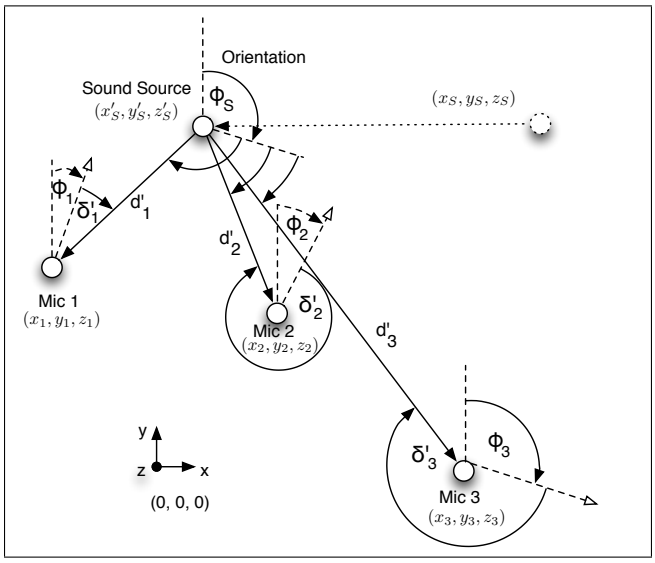


Figure 1.5: *Moving sources, geometric example*

1.6 Getting started

This manual explains the features of the ViMiC system developed for the Jamoma environment in Max/MSP. These four modules are entitled

- `jmod.sur.vimic~`, starting at page 11
- `jmod.sur.vimic8~`, starting at page 26
- `jmod.sur.vimic8poly~`, starting at page 26
- `jmod.sur.reverb~`, starting at page 30

All modules have a dedicated helpfile, e.g. jmod.sur.vimic~.maxhelp. Like all Jamoma modules, the ViMiC modules are fully controlled through a GUI and through external Open Sound Control messages [14]. In the following Sections, each GUI element is explained and its associated OSC message is provided.

For easier and more flexible controllability, ViMiC was structured as high-level modules using the Jamoma[1] framework for Max/MSP/Jitter [11]. Jamoma offers a clear advantage in its standardization of presets and parameter handling. ViMiC parameters have been sorted into three primary namespaces: *source*, *microphone* and *room*; and have specified attributes.
Jamoma modules can be creatd in Max/MSP in two ways, as explained below. For further information on Jamoma, please navigate to the growing tutorial section at jamoma.org.

Loading ViMiC as a subpatcher

1. Create a new empty Max patch.
2. Create a new object[2] and type `jmod.sur.vimic~ test`. This should create an object with two inlets and two outlets, similar to Figure 1.6. The module's OSC name assigned to this object is `/test`.
3. Double-clicking the object opens a small pop-up window, offering the user interface for this object.
4. Click on the upper left circle to close the interface.

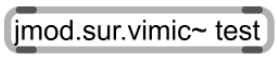


Figure 1.6: *Loading ViMiC as a subpatcher, the GUI is hidden and appears when double-clicking on it*

Loading ViMiC in a bpatcher context

If we instead want to embed the interface in the patch, we can load the ViMiC module as a bpatcher:

[1] http://www.jamoma.org
[2] A new object box can be created by using Max5's shortcut `n`.

1. Create a new empty Max patch.
2. Create a new bpatcher object.
3. Open the inspector, and set Patcher File to `jmod.sur.vimic~`
4. Assign a unique OSC name to the module by giving an argument, e.g. `test` (see Figure 1.7).
5. Close the bpatcher inspector.
6. The bpatcher should now have two inlets and two outlets.

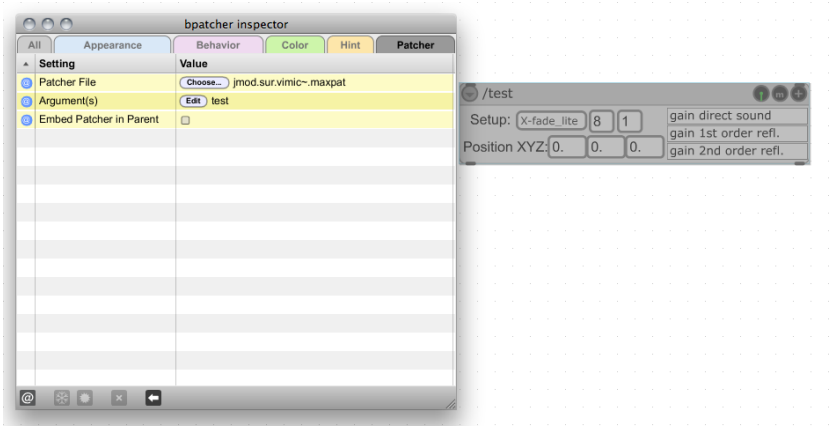


Figure 1.7: *Loading ViMiC in a bpatcher*

Alternatively to the steps 2 - 5, one can create a new object box and type `bpatcher @name jmod.sur.vimic~ @args test` (see Figure 1.8) for an example. This creation is facilitated by using Jamoma's keyboard shortcut `shift` + `B` which provides a pre-configured bpatcher object. Both display options provide the

Figure 1.8: *creating a bpatcher for ViMiC, using the object box*

same kind of functionality, it is just a matter of taste or space in the Max patch what display mode is preferred.

Connecting ViMiC

Audio connection The ViMiC module has two inlets and two outlets. The right inlet feeds the audio signal for processing in ViMiC, and the right outlet delivers the processed multichannel audio in the form of a Jamoma multicable. Jamoma multicable is an audio connection that can contains up to 32 discrete audio channels in one connection. This facilitates especially the work in context with with spatializaiton because the connection of multichannel audio modules is facilitated. For further information,

see [12]. The multicable output contains up to 24 discrete audio channels. Rather than manually connecting these 24 discrete audio channels to the desired destination (e.g. to the `dac~`), the Jamoma multicables provide more convenient solutions. In Figure 1.9 a simple noise signal is spatialized with ViMiC, and the processed multichannel audio signals are connected to the Jamoma module `jmod.sur.output~`, which distributes the multicable audio signal to the physical outputs of the sound card.

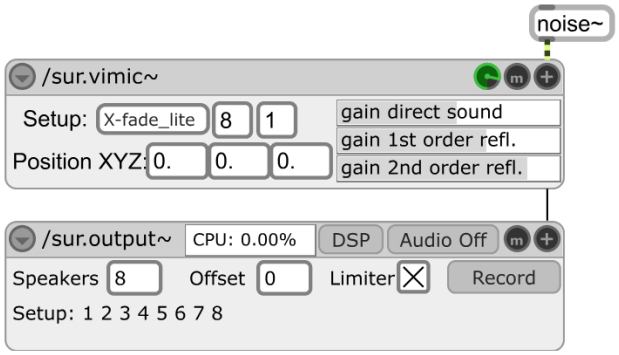


Figure 1.9: *Here the multicable output is connected to jmod.sur.output~*

OSC-message connection As usual in Jamoma modules, the left inlet and the left outlet are reserved for OSC control messages to and from the object. Every parameter change, either due to a received control message or due to a manipulation in the Interface, will be reflected in the left outlet. In Figure 1.9, the left outlet is connected to a message box in order to display the last parameter change.

Chapter 2

The Jamoma Modules

2.1 Parameter & controlling of jmod.sur.vimic~

2.1.1 Front-end

Identifier	Name	Description
A	Message inlet for OSC-messages	p. 10
B	Message outlet for OSC-messages	p. 10
C	Signal inlet	p. 10
D	Multichannel signal outlet	p. 10
1	Module pop-up menu	p. 12
2	Modules OSC name	p. 12
3	Rendermodes pop up menu	p. 12
4	Number of pre-configured microphones	p. 12
5	Order of rendered early reflections	p. 12
6	Sound Source Position	p. 13
7	Mute button	p. 13
8	Inspector button	p. 13, p. 14
9	Gain control for the direct sound component	p. 13
10	Gain control for the 1st-order early reflections	p. 14
11	Gain control for the 2nd-order early reflections	p. 14
12	Gain control for the output volume	p. 14

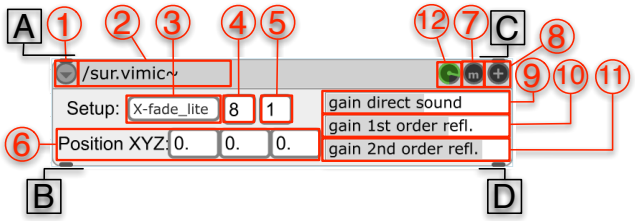


Figure 2.1: *jmod.sur.vimic~ main control interface*

In Figure 2.1 the front end of the module is shown. Here the most often used parameter are shown, with the most often used parameters shown. Information about each controllable parameter follow, including its equivalent OSC-message

(1) - The module pop-up menu Provides several Jamoma specific option. For more information, the reader is directed to the Jamoma documentation. This menu also gives access to a variety of modules presets, holding popular multichannel microphone setups.

(2) - The module's OSC name represents the first OSC branch in a parameter namespace. The uniqueness of the module's OSC name becomes crucial if an OSC-message is not sent when the module is controlled remotely e.g. through *jmod.cuelist* as opposed to being sent directly to the module's left inlet. Jamoma will rename the modules to provide a unique identifier by adding a number at the end of the name. e.g. a duplicated name `\test` becomes `\test.1` and so on.

(3) - The Rendermode From a drop down menu, different rendermodes can be chosen. The rendermodes differ in the quality of calculation of early reflections and the handling of the Doppler effect for moving sound sources. At the moment the following methods, can be selected:

- `ViMiC_XL`
- `ViMiC_lite`
- `X_fade_XL`
- `X_fade_lite`
- `Panning`
- `Static`

For detailed information of the different render methods refer to Section 2.4 on page 27.

Equivalent OSC-message

```
/rendermode <string>
```

(4) - Number of pre-configured microphones In a drop down menu, the maximum number of desired virtual microphones inside the ViMiC system can be selected. A change is performed only if the DSP is turned off.

Equivalent OSC-message

```
/microphones/amount <int> [1 .. 24]
```

(5) - Order of rendered early reflections In a drop down menu, the quantity of early reflections is specified by selecting the reflection order. The higher the order number, the more early reflections are rendered:

Order	Rendered reflections
0	0
1	6
2	18

Equivalent OSC-message

```
/room/reflection/order <int> [0, 1, 2]
```

(6) - Sound Source Position The coordinate triplet defines the position of the sound source in the virtual room.

According to the SpatDIF initiative [10] a position can be defined either in spherical or cartesian coordinates through external OSC-messages. Convention of the coordinate system are listed on page 37.

Equivalent OSC-message

```
/source/position <float float float> [x y z]
/source/position/aed <float float float> [azimuth elevation distance]
```

(7) - The mute button If muted, the ViMiC stops rendering, but still calculates the geometric model, so that when the module is unmuted, the virtual sound source appears at the most updated position.

Equivalent OSC-message

```
/audio/mute <boolean> [0, 1]
```

(8) - The inspector button By clicking on the ⊕ button in the upper right corner, ViMiC's inspector window (Figure 2.2) opens to give control over other parameters described in Section 2.1.2.

Equivalent OSC-message

```
/view/panel
```

(9) - Gain control for the direct sound component This horizontal fader controls the gain of the direct sound component at each virtual microphone and follows Jamoma's MIDI gain convention. A midi value 100.0 is similar to 0.0 dB which refers to a linear gain of 1.0 (see Appendix 3.1 on page 36). The SHIFT-key enables fine tuning of the slider values. the gain value can also be relatively increased/decreased as shown below.

Equivalent OSC-message

```
/room/reflection/gain.0 <float> [0.0 .. 127.0]
/room/reflection/gain.0:/value/inc
/room/reflection/gain.0:/value/dec
```

(10) - Gain control for the 1st-order early reflections If the pop up menu (5) is set to the rendered reflection order of one or higher, this horizontal fader controls the gain of the 1st-order reflections at each virtual microphone. If no early reflections are rendered (when the parameter described in section (5) is set to zero), the fader is disabled.

Equivalent OSC-message

```
/room/reflection/gain.1 <midi value> [0.0 .. 127.0]
```

(11) - Gain control for the 2nd-order early reflections If the pop up menu (5) is set to the rendered reflection order of two, this horizontal fader controls the gain of the 2nd-order reflections at each virtual microphone. If no 2nd-order reflections are rendered (when the parameter described in section (5) on page 12 lower than two), the fader is disabled.

Equivalent OSC-message

```
/room/reflection/gain.2 <midi value> [0.0 .. 127.0]
```

(12) - Gain control for the output volume The green knob controls the output volume of the module. parameter handling is equivalent to gain controller Nr. (9), (10) and (11). When changing its state, the current gain value is displayed in the area of the OSC module name (see (2) in Figure 2.1). Gain values can be higher than 0 dB, see MIDI gain convention on Section 3.1.

Equivalent OSC-message

```
/audio/gain <midi value> [0.0 .. 127.0]
```

2.1.2 Inspector Interface

By clicking on the inspector button, or by sending the OSC message `/view/panel` to the module, the inspector interface appears (see Figure 2.2). The biggest part of the inspector window is filled with the channel strip for each virtual microphone (I), for which functionality is explained in the following.

Identifier	Name	Description
I	Channel strips for each microphone	p. 15
II	Room model parameters	p. 18
III	Source directivity settings	p. 20
IV	Settings for X-Fade Render-method	p. 22
V	Miscellaneous parameters	p. 22

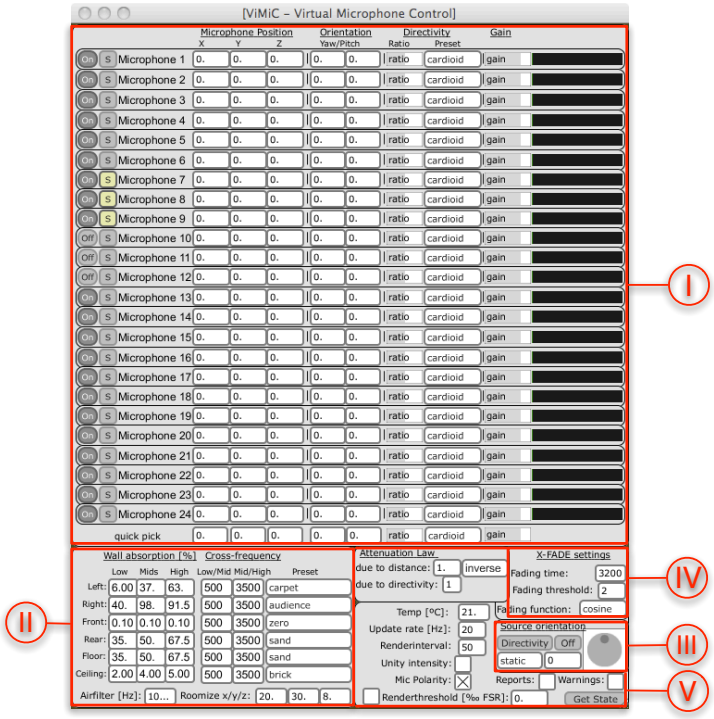


Figure 2.2: *ViMiC inspector window*

I - Microphone configuration

Figure 2.3 shows a part of the microphone configuration area. Like a channel strip in a mixing console, each strip has the same functionality. Therefore only one channel strip example is discussed. To access a specific microphone channel with an OSC-message, the displayed # in the OSC-namespace has to be replaced by the number of the virtual microphone (1-24). The setting of (4) in the front-end (see page 12) determines the highest number a virtual microphone can have, e.g. if (4) is set to eight, only the first eight microphones are displayed. However, it is possible to change setting of higher numbered microphones through OSC messages without loosing these settings. If (4) is changed later to use more virtual microphones, these settings will be recalled.

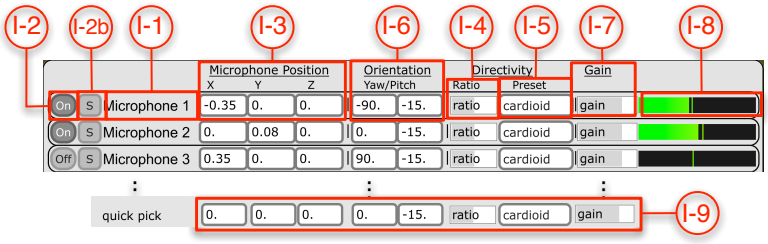


Figure 2.3: *Channel Strip of Virtual Microphone*

(I-1) - Channel number This displays the number of the related virtual microphone. The number of the virtual microphone is directly related to the output channel number; a parameter change in the channel strip of microphone Nr. 2 will affect the output signal Nr. 2 in the multicable.

(I-2) - The active button If active button is set to 0, the microphone will be muted. If the active button is set to 1, the microphone will be unmuted:

Equivalent OSC-message

```
/microphone.#/active <boolean> [0, 1]
```

Solo button The solo button acts as you might expect—it will mute all other microphone channels. Of course, it is possible to have more than one microphone in solo mode.

Equivalent OSC-message

```
/microphone.#/solo <int> [0, 1]
```

(I-3) - Microphone position The position of a microphone can either be set in cartesian or in spherical coordinates.

Equivalent OSC-message

```
/microphone.#/position <float float float> [x-coord. y-coord. z-coord.]
/microphone.#/position/x <float> [x-coord.]
/microphone.#/position/y <float> [z-coord.]
/microphone.#/position/z <float> [z-coord.]
/microphone.#/position/aed <float float float> [azimuth elevation distance]
```

(I-4) - Microphone directivity ratio As described in Section 1.4.1 on page 3, the microphone directivity is defined by the equation:

$$\Gamma = (a + (1 - a) \cdot \cos\delta)^w \qquad 0 \leq a \leq 1 \tag{2.1}$$

The horizontal slider controls the ratio of the parameter a of this equation.

Equivalent OSC-message

```
/microphone.#/directivity/ratio <float> [0.0 .. 1.0]
```

(I-5) - Microphone directivity presets Rather than manually setting the directivity of a virtual microphone, this drop down menu serves several predefined microphone directivity patterns. The following presets are available: `omni`, `subcardioid`, `cardioid`, `supercardioid`, `hypercardioid`, or `figure-eight`.

Equivalent OSC-message

```
/microphone.#/directivity/preset <string>
```

(I-6) - Microphone orientation A microphone can point in a certain direction. This feature becomes important when a virtual microphone is designed with a specified directivity. Yaw-angle ($-180.0°..180.0°$) and pitch-angle ($-90.0°..90.0°$) can be set. See also Jamoma's coordinate system convention in Appendix 3.2 on page 37.

Equivalent OSC-message

```
/microphone.#/orientation <float float>[yaw-angle pitch-angle]
/microphone.#/orientation/yaw <float> [yaw-angle]
/microphone.#/orientation/pitch <float> [pitch-angle]
```

(I-7) - Microphone pre-gain This option can be used to modify the sensitivity of each microphone.

Equivalent OSC-message

```
/microphone.#/gain <float> [0.0 .. 127.0]
```

(I-8) - VU-meter The VU-meter is a classical AFL-VU-meter in that it shows the signal energy after the output volume control described on page 14. The integration time is around 300 ms.

To save some processing power, the VU-meter can be switched off by ticking the entry *Defeat Signal Meters* in the modules pop-up menu (see (1), on page 12).

(I-9) - Quick pick configuration For those situations when uniform parameters for all microphones are desired, rather than configuring each microphone individually, a quick pick configuration will apply desired parameter to all microphones.

For example, if all microphones should have a cardioid directivity, just select the cardioid directivity from the quick pick area.

II - Room model parameter

When early reflections are selected in the front-end menu (see (5) on page 12), the room model becomes audible. Early reflections are calculated according to the position of the sound source, the microphones and the room model (see 1.4.1 on page 4). Below the controllable room parameter are briefly explained.

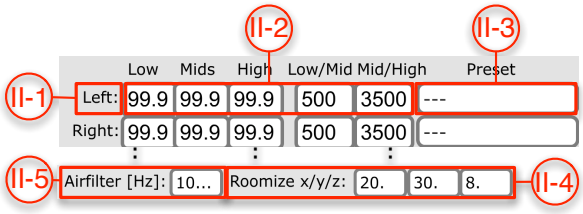


Figure 2.4: *Room model parameter*

Identifier	Name	Description
II-1	Wall identifier	p. 18
II-2	Absorption parameter	p. 18
II-3	Absorption preset	p. 19
II-4	Room size	p. 19
II-5	Air absorption filter	p. 19

(II-1) - Wall identifier Indicates the name of the wall to which the parameter on the right belongs.

(II-2) - Absorption parameter The absorptive behavior of the room boundaries is modeled through a high-mid-low shelf filter. The wall absorption coefficient (α in architectural acoustics) in each band, the is defined in percent. The bigger the number, the higher the absorptive character. The crossover frequencies between the low, mid, and high band can be set as well.

Equivalent OSC-message

```
/room/absorption.##/low <float> [0.1 .. 99.9]
/room/absorption.##/mid <float> [0.1 .. 99.9]
/room/absorption.##/high <float> [0.1 .. 99.9]
/room/absorption.##/low_mid_frequency <int> [10 .. 9999]
/room/absorption.##/mid_high_frequency <int> [10 .. 9999]
```

The placeholder `##` in the OSC-messages must replaced with the wall identifiers `left`, `right`, `front`, `rear`, `floor`, or `ceiling`.

(II-3) - Absorption preset As explained above, several surface presets are available to set the Absorption parameter. a preset can be accessed with the message:

```
/room/absorption.left/preset <string>
```

Located in the ViMiC folder, the presets are stored in the textfile entitled `VimicSurfaceProperties.txt`. This file can be extended. A preset is stored as follows.

Absorption preset

```
<string>, <LF absorption> <MF absorption> <HF absorption>
                                <L-M cross freq> <M-H cross freq>;
```

Example: The preset *audience* is stored as:

```
audience, 40. 98. 91.5 500 3500;
```

(II-4) - Room size The size of the virtual room is set in width, length and height. The unit is in meters. Each dimension is limited to 40 meters.

Equivalent OSC-message

```
/room/size/xyz <float float float> [width length height]
```

(II-5) - Air absorption filter As explained in Section 1.4.1 on page 4, the early reflections are filtered to simulate the damping of the air. This 2nd-order lowpass filters all early reflections in the same way.

Equivalent OSC-message

```
/room/reflection/airfilter <int> [500..19000]
```

III - Source Directivity

Identifier	Name	Description
III-1	Source directivity activator	p. 20
III-2	Source directivity inspector button	p. 20
III-3	Directivity mode	p. 20
III-4	Orientation wheel	p. 21
III-5	Directivity editor	p. 21
III-6	Editor tools	p. 21
III-7	Directivity presets	p. 21

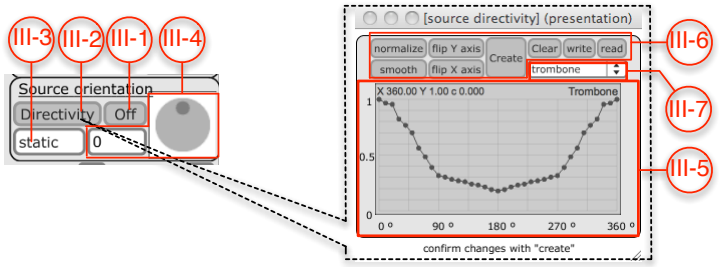


Figure 2.5: *Source directivity options*

(III-1) - Source directivity activator With the toggle, the source directivity calculation can be switched on/off.

Equivalent OSC-message

```
/source/orientation/active <boolean> [0,1]
```

(III-2) - Source directivity inspector button This button opens the source directivity editor, explained in (see III-5 and follow).

Equivalent OSC-message

```
/source/directivity/openEditor
```

(III-3) - Directivity mode Three directivity modes are implemented:
static: The source orientation remains static when the sound source moves. In this case, the orientation can be set via the Orientation wheel or the equivalent OSC-message (see Section 2.1.2).
center: During movement, the source is always oriented towards the center of origin.
follow: During movement, the source is automatically oriented in the direction of the movement.

Equivalent OSC-message

```
/source/orientation/mode <string> [static, center, follow]
```

(III-4) - Orientation wheel If the Orientation mode is set to *static*, this wheel and the connected number box can be used to control the orientation of the sound source. The knob of the wheel shows the direction where the front (the $0°$ direction) of the sound source is pointing. The Orientation is defined in the range of $[-180°$ to $180°]$. Values outside this range are automatically wrapped.

Equivalent OSC-message

```
/source/orientation/yaw <int> [-180 .. 180]
```

(III-5) - Directivity editor Using the mouse, the directivity of a sound source can be freely designed. It works like a breakpoint function editor. A line can be transformed into a curve by pressing the ALT-key while dragging the line with the mouse. Selecting a breakpoint while holding SHIFT will delete this point. To confirm an edited source directivity, press the *create* button (see III-6).

(III-6) - Editor tools These tools for creative editing of the source directivity:
smooth: smoothes out hard corners.
normalize: normalizes the directivity between the values 0.0 and 1.0.
flip Y axis: horizontally flip at the 0.5 point of the attenuation axis.
flip X axis: vertically flip at the 180 degree point.
clear: resets the editor display to a straight line at the 1.0 value.
create: confirms the edited curve.
write: saves a curve in a separate file.
read: loads a directivity from an external file. If a directivity function was stored by using the *write* button, it can be recalled either through the interface by using the *read* button, or via OSC.

Equivalent OSC-message

```
/source/directivity/loadFile <string>
```

(III-7) - Directivity presets Some presets are included which can be either loaded through the pop-up menu or by using the following OSC-message with the name of the preset as the argument.

Equivalent OSC-message

```
/source/directivity/preset <string>
```

IV - X-fade setting

The following parameters are specific for the `x_fade` render methods. More information for the `x_fade` render methods can be found in section 1.5 and in section 2.4. These parameters can only be changed if the sound source is currently not in motion.

(IV-1) Fading threshold This parameter sets the distance in samples to the extend to which one of the propagation paths has to change, in order to trigger a cross-fade from the current position to a new position.

Equivalent OSC-message

```
/rendermode/xfade/threshold <int> [1 .. 512]
```

(IV-2) Fading time These parameter controls the length of the cross-fade, which is triggered when one of the propagation paths changes more than the sample value set in IV-1. The unit of this parameter is samples.

Equivalent OSC-message

```
/rendermode/xfade/fadelength <int> [10 .. 9999]
```

(IV-3) Fading function The cross-fades can have different envelopes; the following envelope functions are implemented:

1. `cosine` - Cosine envelope, (equal intensity)
2. `cosine_squared` - Cosine squared envelope
3. `linear` - Linear envelope, (equal amplitude)
4. `tanh` - Tangent hyperbolic envelope
5. `sqrt` - Square-root envelope

Equivalent OSC-message

```
/rendermode/xfade/fadefunction <string> [cosine, cosine_squared, linear, tanh, sqrt]
```

V - Miscellaneous parameters

Identifier	Name	Description
V-1	Distance law settings	p. 23
V-2	Microphone directivity exponent	p. 23
V-2a	Microphone polarity	p. 24
V-3	“Get State” button	p. 24
V-4	ViMiC update rate	p. 24
V-5	ViMiC renderinterval	p. 24
V-6	Renderthreshold	p. 24
V-7	Intensity normalization	p. 25
V-8	Warning button	p. 25
V-9	Report button	p. 25
V-10	Room temperature	p. 25

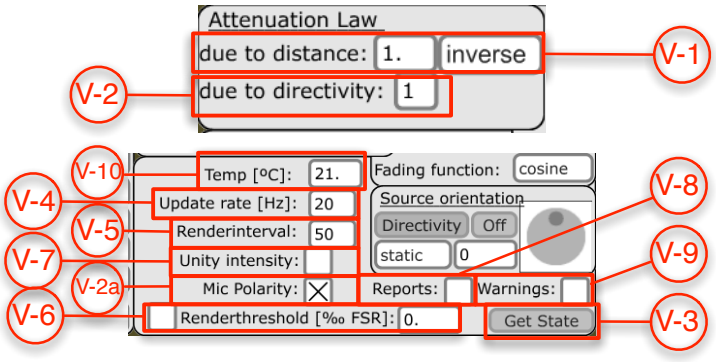


Figure 2.6: *Miscellaneous parameter options*

(V-1) - Distance law exponent Two distance functions are implemented, see section 1.4.1 for a comparison of the two distance functions.

Equivalent OSC-message

```
/room/distance/mode <string> [inverse, exponential]
```

Inverse proportional decrease: The parameter sets the exponent q in the attenuation function described in equation 1.1 on page 3. Thus, the effect of distance attenuation can be boosted or softened. According to the radiation of a point source in the free field, this factor has usually a value of 1.0.

Equivalent OSC-message

```
/room/distance/power <float> [0.0 .. 9.0]
```

Exponential decrease: The parameter k from equation 1.2 can be modified.

Equivalent OSC-message

```
/room/distance/dbUnit <float> [0.0 .. 60.0]
```

(V-2) - Microphone directivity exponent This parameter sets the exponent w in the attenuation function described in Equation 1.3 on page 3. Increasing the exponent w to a value greater than 1 will produce an artificially sharper directivity pattern. Furthermore, by applying even numbered parameters, anti-phased gain values are avoided. Note that only integer values are allowed for this exponent.

Equivalent OSC-message

```
/microphones/directivity/power <int> [0 .. 9]
```

(V-2a) Microphone polarity / In-phase restriction If a microphone directivity ratio is smaller than 0.5 (e.g. a figure-of-eight microphone setting), for certain angles of incidence, a negative attenuation factor will be calculated. In order to avoid these anti-phase components, this toggle can be disabled to keep the allowed microphone attenuation between zero and one and set all negative factors to zero.
By restricting the microphone's polarity and with an appropriate microphone arrangement, the "Polarity-restricted cosine" panning function can be achieved as described in [8].

Equivalent OSC-message

```
/microphones/polarity <int> [0, 1]
```

(V-3) - "Get State" button By pressing this button, the inner state of the ViMiC-external is posted into the Max window. This is especially helpful for finding bugs.

Equivalent OSC-message

```
/getState
```

(V-4) - ViMiC update rate To save resources, not every parameter change causes ViMiC to update (re-render) its virtual microphone signals. This parameter sets the frequency for how often ViMiC is forced to re-render its virtual microphone signals.

Equivalent OSC-message

```
/updaterate <int> [0 .. 100]
```

Alternatively to this update rate, ViMiC can be forced to re-render the virtual microphone signals from an external clock through the OSC command:

```
/update
```

(V-5) - ViMiC renderinterval This parameter is used for the `ViMiC` Rendermodes and determines how many signal blocks are used in order to fulfill the delayline interpolatation. The higher the number, the more time it takes for the interpolation. If the number of blocks is too small, audible artifacts appear.

Equivalent OSC-message

```
/rendermode/interval <int> [1 .. 200]
```

(V-6) - ViMiC renderthreshold Some discrete reflections might not be perceptually important due to the applied distance law, microphone characteristics, and source directivity. To minimize processor load, an amplitude threshold can be set to prevent the algorithm from rendering these reflections. This amplitude threshold can be set very precisely in one-tenth of a percent of the full scale range. This is an experimental parameter.

Equivalent OSC-message

```
/rendermode/threshold/active <boolean> [0, 1]
/rendermode/threshold <float> [0.0 .. 1000.0]
```

(V-7) - Intensity normalization By activating, all calculated gain values g_i are normalized in order to have unity intensity according to equation 2.2. This is an experimental option.

$$I = \sum_{i=1}^{n} g_i^2 = 1 \tag{2.2}$$

Equivalent OSC-message

```
/rendermode/normalization/active <boolean> [0, 1]
```

(V-8) - The warning button If this button is ticked, different postings appear in the Max window. For example, if the sound source position exceeds the virtual room, ViMiC will post a warning message, but will not stop the rendering.

Equivalent OSC-message

```
/warning <int> [0, 1]
```

(V-9) - The report button If this button is ticked, every parameter change will cause postings to the Max Window.

Equivalent OSC-message

```
/report <int> [0, 1]
```

(V-10) - Room temperature As described in Section 1.4.1, the inter-channel time differences, depending on the spatial relation of the source and each microphone, are calculated for a given speed of sound. The speed of sound depends on the medium, and for air it primarily depends on the temperature, e.g. [13]. By default the temperature unit of ViMiC's virtual room is assumed to be in degree Celsius. However, Jamoma's dataspace feature allows the value to be set also in degree Fahrenheit or Kelvin.

Equivalent OSC-message

```
/room/temperature <float> [-20 .. 30]
```

2.2 The module jmod.sur.vimic8∼

This module contains basically 8 jmod.sur.vimic∼ modules in one module. This has the advantage that a lot of settings, such as the microphone position and the orientation have to be made only once–rather than eight times. Because the module renders 8 sources, it has a lot more inputs than the regular jmod.sur.vimic8∼. As usual in Jamoma, the first inlet is reserved for communication via OSC. The next eight inlets are conventional signal inlets to connect audio sources directly to the rendering algorithm. The rightmost inlet is reserved for Jamoma's multicables whereas its first eight signals are internally separated and routed to the rendering algorithm. Because there are now eight sources to be controlled, the OSC namespace is slightly extended for the sources, compared to the namespace in jmod.sur.vimic∼. All messages related to `/source` are extended with an index starting from 1 to 8, e.g:

OSC-message example

```
/source.1/gain 100.0 /source.5/orientation 23.0 0.0
```

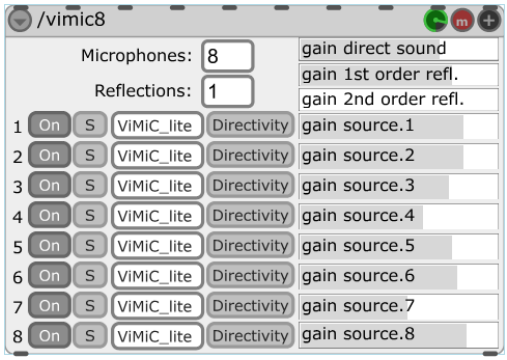


Figure 2.7: *jmod.sur.vimic8∼*

2.2.1 Description message

In order to make the GUI more informative, the text displayed in the gain slider of each source can be modified with the following OSC message:

OSC-message to set a description string

```
/source.#/name <string> /source.1/name piano
```

This string will also appear in the window that controls the source directivity.

2.3 The module jmod.sur.vimic8poly∼

The module jmod.sur.vimic8poly∼ is basically the same as jmod.sur.vimic8∼, but uses the multithreading functionalities of Max5 offered by the poly∼ external. If you are

using a multi-core computer, the rendering of the eight sound sources can be distributed amongst different CPUs. There are two new parameters to control the multi-threading:

multithreading OSC-message

```
/parallelization/active <boolean> [0, 1]
/parallelization/numThreads <int> [0 .. 8]
```

If `/parallelization/numThreads` is set to 0, then Max/MSP will automatically use as many cores as available. An integer number higher than 0 specifies explicitly how many cores are going to be used. These two parameter can also be manipulated over the panel (Figure 2.8).

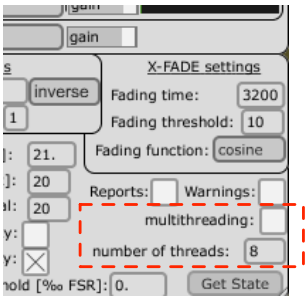


Figure 2.8: *Multithreading options in jmod.sur.vimic8poly~*

2.4 Rendermodes

ViMiC holds several DSP rendermodes that one can chose from according to the artistic aspiration or limitation of the DSP resources (see item (3) on page 12). The following section describes each of these modes.

ViMiC_XL If a sound moves, `ViMiC_XL` renders the sound rays using delay-line interpolation, which causes an audible Doppler effect (see section 1.5). Each wall of the virtual room can be simulated with a separate surface properties by setting the filter coefficients for the wall as described in section 2.1.2.

X_fade_XL This mode renders moving sound sources with the cross-fade method, described in section 1.5. The special X-Fade settings in the ViMiC inspector (see section 2.1.2 now become relevant. As the suffix `XL` in the name `X_fade_XL` states, the early reflections are rendered with respect to separate surface properties for each wall.

ViMiC_lite & X_fade_lite The render methods `ViMiC_lite` and `X_fade_lite` differ from `ViMiC_XL` and `X_fade_XL` in the quality of the room model: Instead of having the choice of individual wall absorption parameters for each wall, the entire

virtual room is being modelled with the wall absorption of the “left” wall, the first wall in the ViMiC Room settings in the ViMiC inspector panel (see Section 2.1.2).

Static This rendermode is similar to `ViMiC_lite`, but no delay-line interpolation is performed. This setting is a “cheap and dirty” rendering method. It’s worth a try if one is short in DSP resources or if sources are not moving.

Panning This mode does not apply the calculated time delays to the signals. It just takes the gain values according to distance, source directivity and microphone characteristic into account. No early reflections are rendered, regardless of the settings in in (5), page 12, because the room reflections would appear without delay. If the gain normalization option is active (V-7), and microphone characteristic is set to `omni`, Distance Based Amplitude Panning (DBAP) [7] is created.

2.4.1 Comparison & CPU load

Rendermode	Source is...	No Reflections			1st-order Reflections			2nd-order Reflections		
		8 mics	16 mics	24 mics	8 mics	16 mics	24 mics	8 mics	16 mics	24 mics
ViMiC_XL	stationary	7.4%	12.3%	14.6%	15.5%	28%	40.7%	30.6%	59.2%	+++
	moving	8.6%	14.5%	18.3%	23.4%	42.5%	63.4%	50.6%	+++	+++
X_fade_XL	stationary	7.6%	12.5%	15.6%	17.3%	31%	45.5%	35.8%	68%	+++
	moving	8.2%	14%	16.4%	20.9%	37.6%	54.6%	43.4%	86%	+++
ViMiC_lite	stationary	7.2%	12%	214.4%	10.8%	26.5%	25.5%	15.4%	28%	40.0%
	moving	8.3%	13.8	17.6%	17.4%	31%	44.9%	33.2%	60%	+++
X_fade_lite	stationary	8%	13.2%	16.5%	13.5%	26%	34.0%	22.5%	42%	60.9%
	moving	8.2%	14%	17.0%	14.0%	25.8 %	34.6%	22.2%	41.2%	58.3%
Panning	stationary	6.4%	10.5%	12.2%	-	-	-	-	-	-
	moving	6.5%	10.5%	12.4%	-	-	-	-	-	-

Table 2.1: *Max/MSP Processor load for one sound source at different render settings, Max/MSP overdrive option disabled, $f_s = 44.1kHz@16Bit$, measured on a mac mini 1.66 GHz intel core duo*

+++ indicates processor overload

Max/MSP Version 4.6.3

Values are taken from the internal Max/MSP "CPU Utilization" information

2.5 The module jmod.sur.reverb∼

2.5.1 Introduction

The late reverb is implemented as a feedback delay network (FDN) with 16 delay lines that models later reflections as an exponentially decaying random gaussian process, characterized by a spectral envelope. A flowchart of the algorithm is displayed in Figure 2.9. Because late reverberation decays can be approximated with a stochastic model, the use of FDNs for artificial reverberation is legitimate, assuming that the acoustic model has sufficient density in the frequency domain and sufficient number of reflections in the time domain. An FDN late reverb cannot guarantee the same accuracy as a convolution-reverb using a measured impulse response, but provides a more efficient parametrization for dynamic and flexible control of the reverberation effect, such as reverberation time, modal density or reverb coloration.

Any FDN can be represented as a set of digital delay lines for which the inputs and outputs are connected by a feedback/mixing matrix. This matrix provides diffusion by mixing the energy of every input to every output. The matrix is supposed to be unitary (or orthogonal). A system is said to be unitary if its matrix transfer function is unitary for any complex variable z on the unit circle. Another requirement is that the matrix has to be lossless. For this implementation, a 16x16 Hadamard matrix is used. The advantage of this matrix is that it can be efficiently implemented with a butterfly algorithm using $N \log 2N = 64$ additions (Figure 2.10). As proposed in [5], the delay-lines in the FDN can be modulated by low frequency oscillators in terms of delay-length and feedback attenuation coefficient.

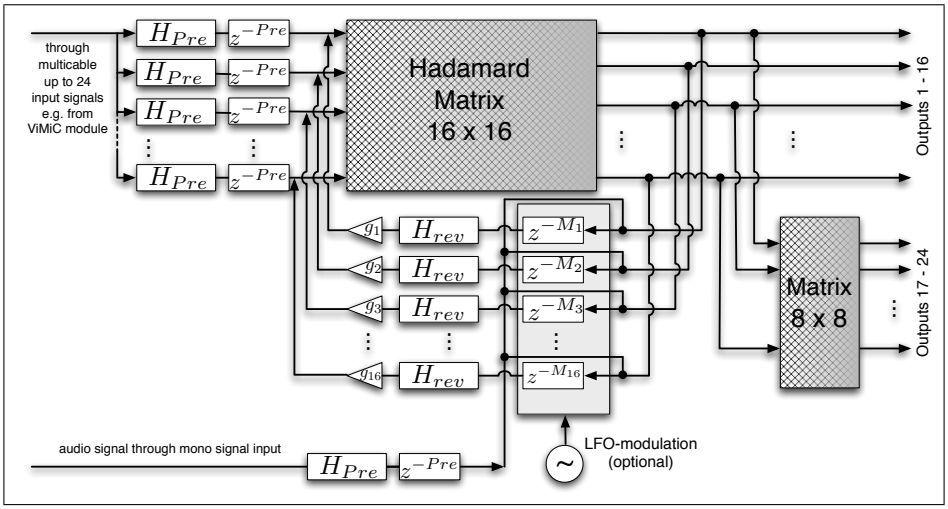


Figure 2.9: *FDN-reverb flowchart*

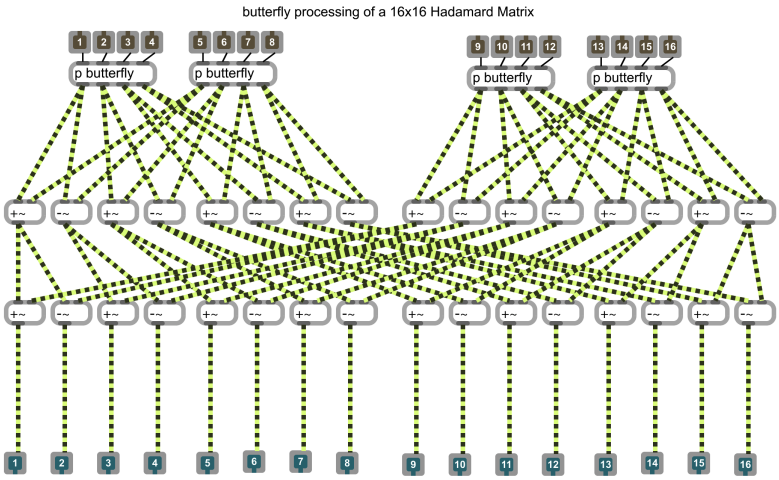


Figure 2.10: *Butterfly implementation of the FDN 16x16 Hadamard matrix*

2.5.2 Parameter & Controlling

Front-end

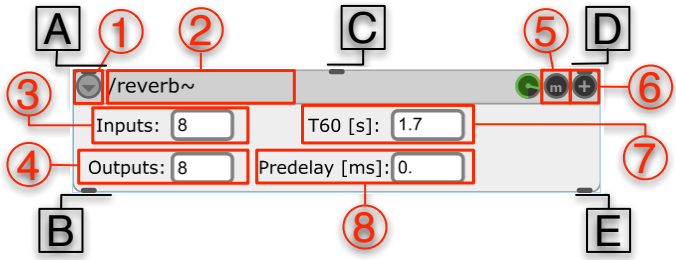


Figure 2.11: *jmod.sur.reverb∼ main interface*

Identifier	Name	Description
A	Message inlet for OSC-messages	p. 32
B	Message outlet for OSC-messages	p. 32
C	Multichannel signal inlet	p. 32
D	Mono signal inlet	p. 32
E	Multicable signal outlet	p. 32
1	Module pop-up menu	p. 32
2	Modules OSC name	p. 32
3	Number of input channels	p. 32
4	Number of output channels	p. 32
5	Mute button	p. 33
6	Inspector button	p. 33
7	Reverb time T_{60}	p. 33
8	Pre-delay	p. 33

Reverb connections As depicted in Figure 2.11, the reverb module has three inlets and two outlets. The connections A and B are used for OSC communication, whereas the others are meant to process audio signals. There are two audio inlets, so that a mono audio signal (inlet D) coming from a dry sound source, or preprocessed multi-cable signals coming from ViMiC modules can be reverberated. Both inlets can be used in parallel as well.

Module pop-up menu This provides several Jamoma specific option. Reader are pointed to the Jamoma documentation.

Modules OSC name This field shows the OSC-name of the reverb module, a unique name that becomes important if an OSC-message is not send directly to the module's left inlet, but is controlled remotely e.g. through the `jmod.cuelist` module. No modules can have the same name.

Number of input channels The number of used inputs should be defined here. According to this value, some filters and delays from unused input will be disabled (z^{-pre} and H_{pre} in Figure 2.9). Thus DSP resources are optimized.

Equivalent OSC-message

```
/numInputs <int> [1 .. 16]
```

Number of output channels Determines how many audio channels the multi-cable contains at the output. A change in the number of outputs will only be effective when the DSP is switched off.

Equivalent OSC-message

```
/numOutputs <int> [1 .. 24]
```

Mute button If mute is activated, the reverb stops outputting the audio and internal DSP processes.

Equivalent OSC-message

```
/audio/mute <boolean> [0, 1]
```

Inspector button By clicking on the ⊕ button in the upper right corner, the inspector window opens to give control over extended parameters described in 2.5.2.

Equivalent OSC-message

```
/view/panel
```

Reverb time T_{60} Reverberation time in ms. Technically speaking, it influences the gain values g_1 - g_{16}, depicted in Figure 2.9.

Equivalent OSC-message

```
/t60 <float> [0 .. 60.0]
```

Pre-delay This parameter delays the incoming signals to the reverb processing by the specified time in ms (depicted as z^{-pre} in Figure 2.9). This parameter is important to temporally align the late reverb correctly with the direct sound and early reflections rendered with ViMiC.

Equivalent OSC-message

```
/predelay <float> [0 .. 1000.0]
```

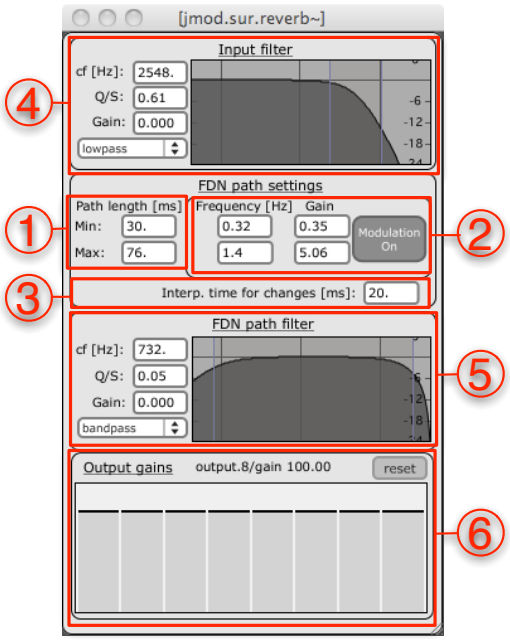
Inspector window

Identifier	Name	Description
1	Range of the FDN delay lengths	p. 33
2	FDN Modulation options	p. 34
3	Interpolation time	p. 34
4	Input filter settings	p. 34
5	FDN path filter settings	p. 35
6	Gain per output channel	p. 35

Range of the FDN delay lengths Minimal and maximal delay length of the feedback paths in ms. According to these settings, an algorithm will calculate orthogonal values for each of the 16 feedback paths internally.

Equivalent OSC-message

```
/fdn/delaylength <float float> [min-value max-value]
```



FDN Modulation options The feedback delay paths can be amplitude modulated to avoid ringing frequencies. For percussive sounds, this option sounds great. However, for sustained sounds such as guitar, or violins, the modulation might be audible and you might want to turn it off. According to the amplitude and frequency settings, an algorithm will calculate 16 orthogonal values within the given range to be used for the amplitude modulation.

Equivalent OSC-message

```
/modulation/active <boolean> [0,1]
/modulation/amplitude <float float> [min-value max-value]
/modulation/frequency <float float> [min-value max-value]
```

Interpolation time Interpolation time in ms for smooth interpolation of changes in the feedback path length (Section 2.5.2) and path modulation (Section 2.5.2).

Equivalent OSC-message

```
/interpolationtime <float> [0.0 .. 1000.0]
```

Pre-filter settings The input signal can be pre-filtered (H_{pre} in Figure 2.9) for instance to simulate air absorption.

Equivalent OSC-message

```
/input/frequency <float> [Hz]
/input/filtergain <float> [dB]
/input/filtertype <string> [lowpass, highpass, bandpass, bandstop,...
                                                peaknotch, lowshelf, highshelf]
/input/q <float> [Q or S value]
```

FDN Filter settings These settings define the filter inside the feedback delay network (H_{rev} in Figure 2.9). Gain values higher than 0 dB (amplifications) are prohibited to ensure a stable reverb.

Equivalent OSC-message

```
/fdn/q <float> [Q or S value]
```

Gain per output channel The gain for each of the outputs can be manipulated independently. For instance, one could add more reverb to the surround channels than to the frontal loudspeakers if desired. Clicking on the "reset" button will set all outputs back to equal gain. The number of gain sliders will adapt automatically according to the defined number of outputs.

Equivalent OSC-message

```
/output.#/gain <float> [midigain]
```

2.5.3 CPU-load

The CPU-load depends up to a certain degree on the number of input/output channels and whether the length of the FDN-paths are being modulated by the LFOs (see section 2.5.2 on the previous page).

Table 2.2: *Max/MSP Processor load for one sound source at different late reverb settings, Max/MSP overdrive option disabled, $f_s = 44.1kHz@16Bit$, measured on a mac mini 1.66 GHz intel core duo. Numbers are taken from the internal Max/MSP "CPU Utilization" information.*

	Number of outputs	8	16	24
Inputs	FDN-modulation			
1	off	10 %	10.0 %	10.5%
1	on	11.5%	11.5%	12.0 %
8	off	10.6%	10.6%	11.0 %
8	on	12 %	12.0 %	12.5%
16	off	11 %	11.0 %	12.3%
16	on	12.5%	12.5%	13.0 %

Chapter 3

Appendix

3.1 Jamoma's MIDI to gain conversion

All messages controlling gain properties are expecting values in the MIDI range (0.0 - 127.0). The following table shows how the MIDI values correspond to a linear gain value. A MIDI value of 100 corresponds to a gain of 1. All MIDI values higher than 100 will consequently cause an amplification, whereas MIDI values below 100 will cause an attenuation.

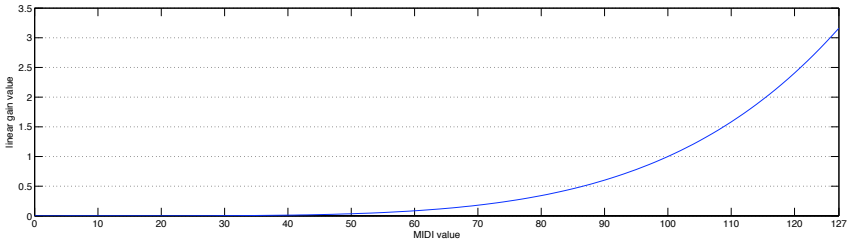


Figure 3.1: *Jamoma's MIDI to gain conversion*

3.2 Coordinate System

ViMiC uses a right-handed cartesian coordinate system, with limits defined by the virtual room size (see Section (II-4) on page 19).

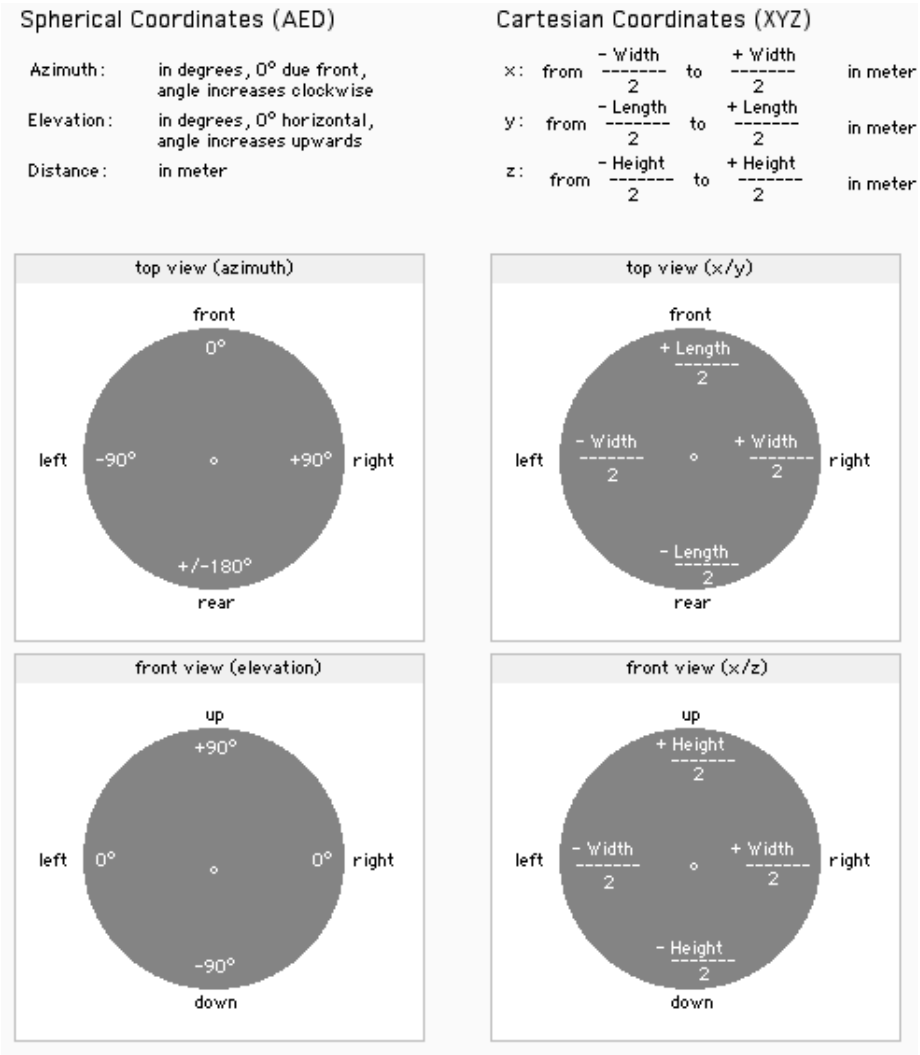


Figure 3.2: *Coordinate Systems: Spherical (left) and Cartesian (right)*

3.3 Known Issues

IF YOU SEE SOMETHING – SAY SOMETHING.

If you find something suspicious in this software, don't hesitate to inform the right person! Please send bug reports and suggestions to

nils.peters_AT_mail_DOT_mcgill.ca.

jmod.sur.vimic~

- ~~The first time the inspector opens takes quiet long time.~~
- If the activation button of a microphone channel strip is turned off, it can cause a click, because the gain of this microphone is set to zero without being ramped.
- ~~Quick pick configuration for the x-coordinate and y-coordinate is not implemented yet.~~
- ~~The "follow" option for the source directivity is not implemented yet.~~
- After loading ViMiC the "fading time" in X-Fade settings is disabled. It needs the DSP turned on to be manipulatable.
- "X_fade XL" produces nasty clicks when source moves.

jmod.sur.reverb~

- ~~After unmuting the module, the number of inputs are set to 16 and the number of outputs are set to 24 automatically.~~

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