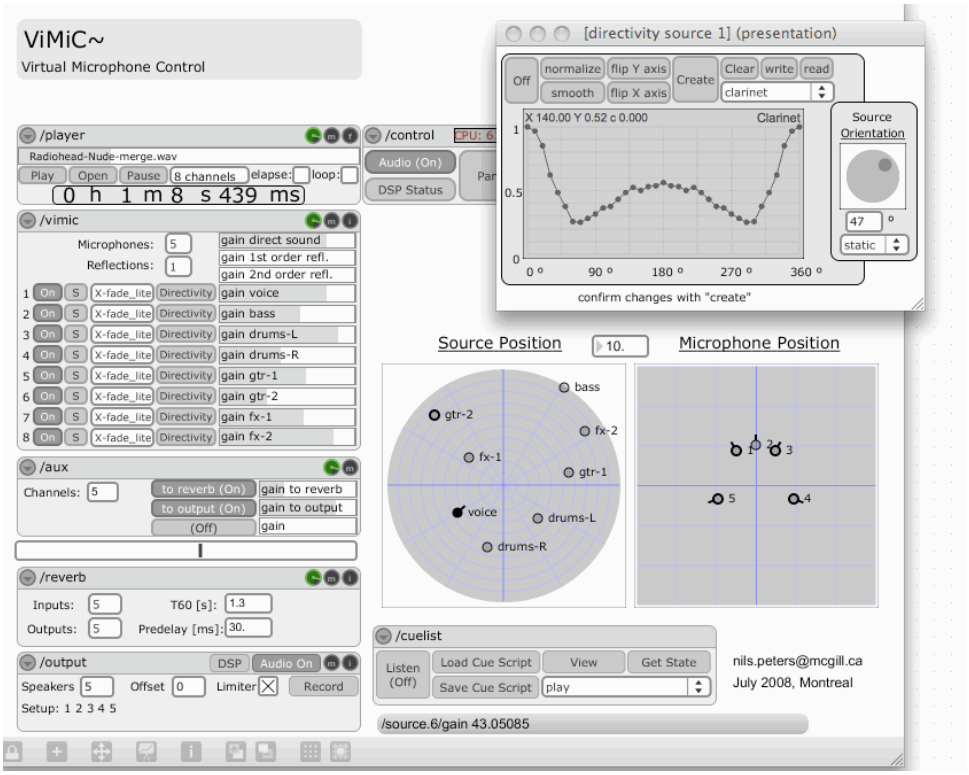
DOCUMENTATION

ViMiC - Virtual Microphone Control
for Max/MSP



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Draft Version, December 21, 2009

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Chapter 1

Introduction

This manual is not fully updated to ViMiC for Max 5 - a few pictures show the GUI used in Max 4.6.3. The functionality does not differ much, though.

1.1 Acknowledgment

This work has been 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

ViMiC requires:

- Max/MSP 5.0.7 or newer
- MacOS 10.5.6 or newer
- ej.function, part of ejies,
  freely available at `www.e--j.com/`

1.3 Installation

ViMiC for Max 5 is available in the github repository:

`http://github.com/Nilson/ViMiC-and-friends/tree/master`

ViMiC for Max 4.6.3 is available in the Jamoma Max 4 maintenance subversion repository:

`https://jamoma.svn.sourceforge.net/svnroot/jamoma/`

`branches/maintenance_max4_support`

A tutorial to accessed the Jamoma github repository (`http://github.com/tap/Jamoma/tree/master`) is posted at:

`http://redmine.jamoma.org/wiki/jamoma/Working_with_GIT`

For installing Jamoma after downloading, following the advises in the included Readme.html.
Unzip *ViMicMax∼.mxo.zip*, located in /UserLib/ViMiC/jmod.sur.vimic/.
The ViMiC toolbox is located in the folder /UserLib/ViMiC/. To test if everything is installed correctly, just load the help-patch *jmod.sur.vimic∼.maxhelp*, located at /UserLib/ViMiC/jmod.sur.vimic/.

Chapter 2

ViMiC

2.1 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.

2.1.1 Source - Microphone Relation

Sound sources and microphones can be placed and moved in 3D as desired. Figure 2.3 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. 2.1 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.

$$g_i = (d_i + 1)^{-q} \tag{2.1}$$

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:

$$g_i = 10^{\frac{-k}{20} \cdot d_i} \tag{2.2}$$

Further attenuation happens through the chosen microphone characteristic and source directivity (see Fig. 2.4). For all common microphone characteristics, the directivity

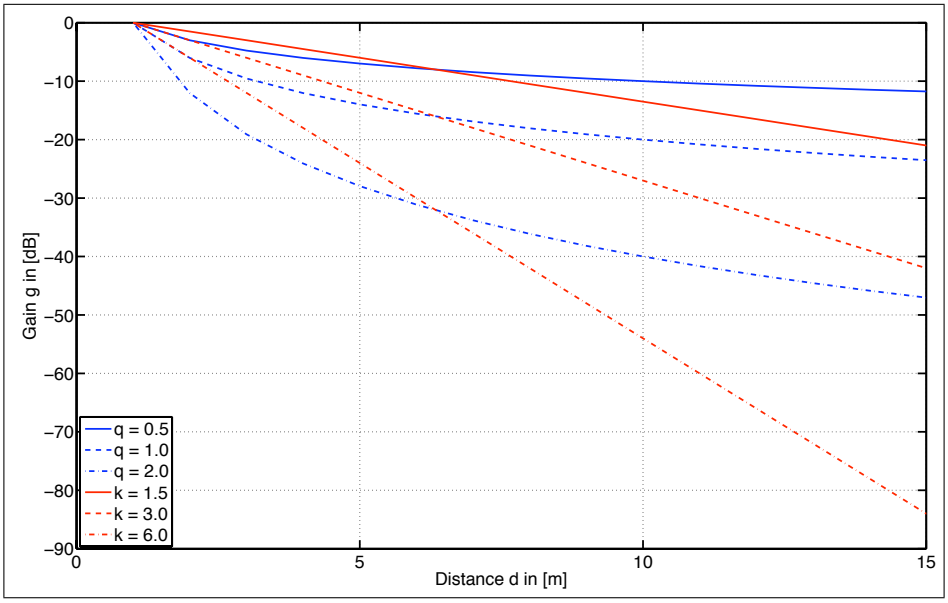


Figure 2.1: Comparison of the attenuation functions with different settings: "inverse proportional decrease" model (blue lines, eq. 2.1) and "exponential decrease" model (red lines, eq. 2.2)

for a certain angle of incidence δ can be imitated by calculating Eq. 2.3 and applying a set of microphone coefficients from Table 2.1.1. 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.

$$\Gamma = (a + (1 - a) \cdot \cos \delta)^w \qquad 0 \leq a \leq 1 \tag{2.3}$$

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

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.

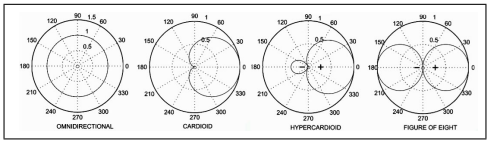


Figure 2.2: Common microphone characteristics

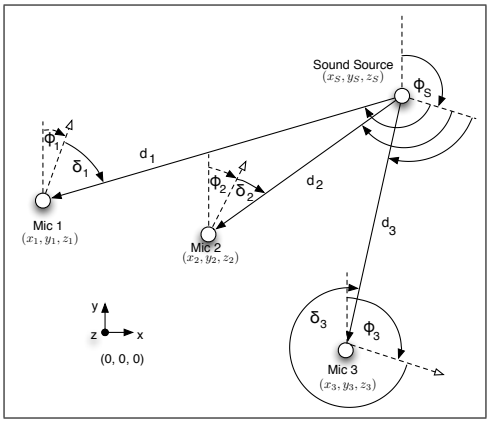


Figure 2.3: Geometric example

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 [Pellegrini, 2002]. 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 [Allen & Berkley, 1979]. For a general discussion of the image method, see, e.g. [Cremer & Müller, 1982] and [Berman, 1975] 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 2.1.1. 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 reflections are considered (8 microphones $\cdot$ [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 [Bech, 1998]. Consequently in ViMiC the levels of direct sound and early reflections can be modified independently.

2.1.2 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 [Jot & Chaigne, 1991] 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. 2.4), for which timbral and temporal character can be modified. This late reverb can be efficiently shared across several rendered sound sources. For detailed information about the implementation of the Late reverb see chapter 5 on page 32.

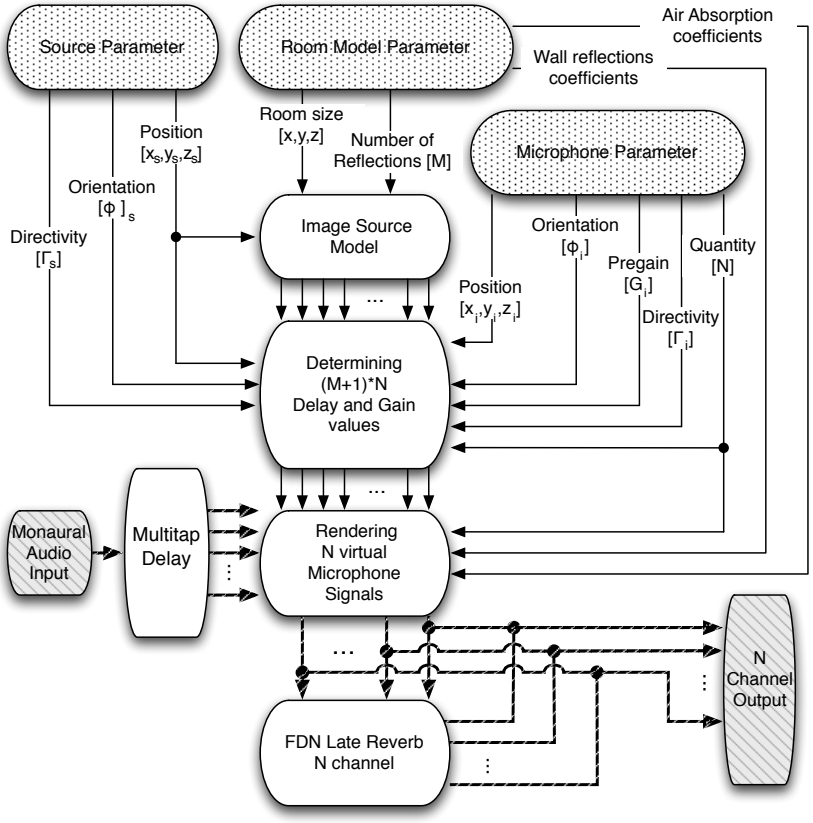


Figure 2.4: Flowchart of the Max/MSP processing

2.2 Moving Sources

In Figure 2.5 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.

2.2.1 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.

2.2.2 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.

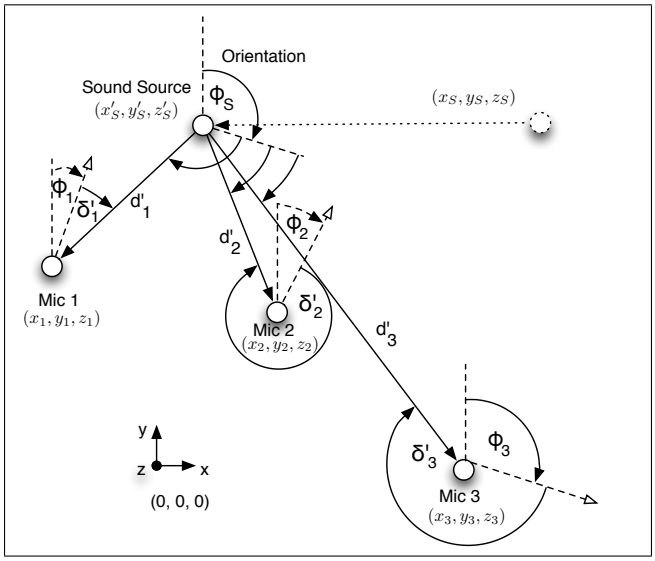


Figure 2.5: Moving sources, geometric example

Chapter 3

The Jamoma module `jmod.sur.vimic~`

3.1 Introduction

For easier and more flexible controllability, ViMiC was structured as high-level modules using the Jamoma[1] framework for Max/MSP/Jitter [Place & Lossius, 2006]. Jamoma offers a clear advantage in its standardization of presets and parameter handling. Each ViMiC parameter has been sorted into three primary namespaces: source, microphone and room; and has a specified data range. ViMiC is fully controlled through a GUI and through external OSC-messages. [Wright & Freed, 1997].
Jamoma modules can be loaded in two ways.

Loading ViMiC as an ordinary object

1. Create a new empty Max patch.
2. Create a new object box, and type in
   `jmod.sur.vimic~.maxpat /test`
   This should create an object with two inlets and two outlets, similar to Figure 3.1. The module's OSC name assigned to this object is `/test`.
3. Double-click the object, to open a small window, offering an interface for the object.
4. Clik on the upper left circle to close the interface.

Loading ViMiC in a bpatcher

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`

Figure 3.1: Loading ViMiC as an ordinary object

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~.maxpat`
4. Assign a unique OSC name to the module by giving an argument, e.g. `test`, see Figure 3.2.
5. Close the bpatcher inspector.
6. The created bpatcher has two inlets and two outlets.

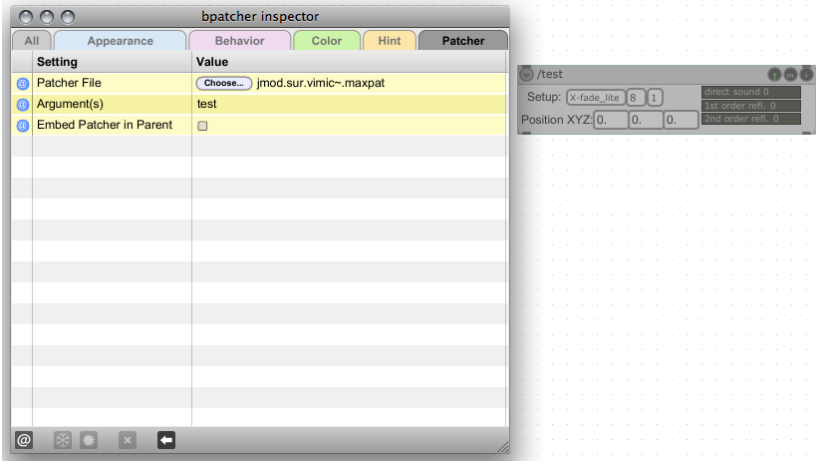


Figure 3.2: Loading ViMiC in a bpatcher

Alternatively, by clicking `shift` + `B` a Jamoma-template is created where you just have to define the Jamoma module (@name) and its OSC-name (@args) as shown in Figure 3.3.

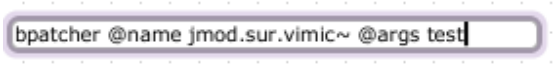


Figure 3.3: Loading ViMiC as an ordinary object

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

3.1.1 Connecting ViMiC

Audio connection

The ViMiC module has two inlets and two outlets. The right inlet takes the audio signal to be processed in ViMiC, and the right outlet delivers the processed multichannel audio in form of a multicable. 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 3.4 a simple noise signal is spatialized with ViMiC, and the processed multichannel audio signals are connected to the Jamoma module `jmod.sur.output~.maxpat`. The latter distributes of a multicable signal to multiple dac∼ channels.

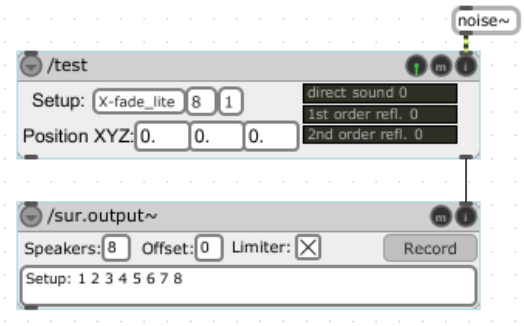


Figure 3.4: Here the multicable output is connected to jmod.sur.output∼.maxpat

OSC-message connection

As usual in Jamoma modules, the left inlet and the left outlet are assigned 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 at the left outlet. In Figure 3.4 the left outlet is connected to a message box in order to display the last parameter change.

3.2 Parameter & Controlling

3.2.1 Front-end

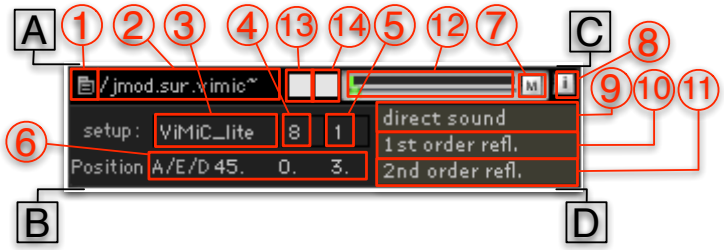


Figure 3.5: ViMiC main control interface

Identifier	Name	Description
A	Message inlet for OSC-commands	p.11
B	Message outlet for OSC-commands	p.11
C	Signal inlet	p.11
D	Multichannel signal outlet	p.11
1	Module pop-up menu	p.12
2	Modules OSC name	p.13
3	Rendermodes pop up menu	p. 13
4	Number of pre-configured microphones	p.13
5	Order of rendered early reflections	p.13
6	Sound Source Position	p.14
7	Mute button	p.14
8	Inspector button	p.14, p.16
9	Gain control for the direct sound component	p.14
10	Gain control for the 1st-order early reflections	p.15
11	Gain control for the 2nd-order early reflections	p.15
12	Gain control for the output volume	p.15
13	Warning button	p.15
14	Report button	p.15

In Figure 3.5 the front end of the module is shown. Here the most often used parameter are shown. The following section gives information about each controllable parameter.

(1) - The module pop-up menu

The module pop-up menu provides several Jamoma specific option. Reader are pointed to the Jamoma documentation.
related OSC-messages:

```
ui/refresh
ui/freeze <1,0>
```

(2) - The modules OSC name

Here the given modules OSC-name appears. It identifies the module to its namespace. This name becomes necessary if a OSC-message is not send directly to the module's left inlet, but the module is controlled remotely e.g. through the `jmod.cuelist.maxpat` module. The module OSC name has to be unique. In case of multiple used module name, Jamoma will rename the modules to a unique identification 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`

Detailed information to the different render methods refer to section 3.3 on page 27.
equivalent OSC-message:

```
/rendermode <string>
```

(4) - Number of pre-configured microphones

In a drop down menu the maximum number of virtual microphones can be selected. A change is only performed 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 the rendered early reflections can be determined. The higher the chosen number, the more early reflections are rendered. A change is only performed if the DSP is turned off. Per microphone the following number of reflections are processed:

Order	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 standardization [Peters et al., 2007] a position can be defined either in spherical or cartesian coordinates through external OSC-messages. Convention for the coordinate system can be found in appendix A.
related OSC-messages:

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

(7) - The mute button

If mute is active, the ViMiC stops rendering, but still calculates the geometric model, so if 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 opens to give control over other parameter described in 3.2.2.
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. The gain controller were programmed by using Jamoma's gain dataspace, see also Appendix B on page 41. As every slider in Jamoma, the SHIFT-key enables fine tuning of the slider value. the gain value can also relatively increased/decreased as shown below:
related OSC-message:

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

A midi value 100.0 is similar to 0.0 dB which refers to a linear gain of 1.0.

(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 (the numberbox described in section (5) on page 13 is set to zero), the fader is disabled.
related 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 second order reflections are rendered (the numberbox described in section (5) on page 13 is set to zero or to one), the fader is disabled.
related OSC-message:

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

(12) - Gain control for the output volume

The horizontal slider controls the output volume of the module. parameter handling is equivalent to gain controller Nr. (9), (10) and (11). In order to boost the output, gain values can be bigger than 0 dB, see also Appendix B on page 41.
related OSC-message:

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

(13) - 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]
```

(14) - the report button

If this button is ticked, every parameter change will cause postings to the Max Window.

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

3.2.2 Inspector Interface

Through clicking on the inspector button, or by sending the OSC message `/view/panel` to the module the Inspector interface appears (see Figure 3.6). The biggest part of the inspector window is filled with the channel strip for each virtual microphone (I), which functionality is explained in section 3.2.3 on the following page.

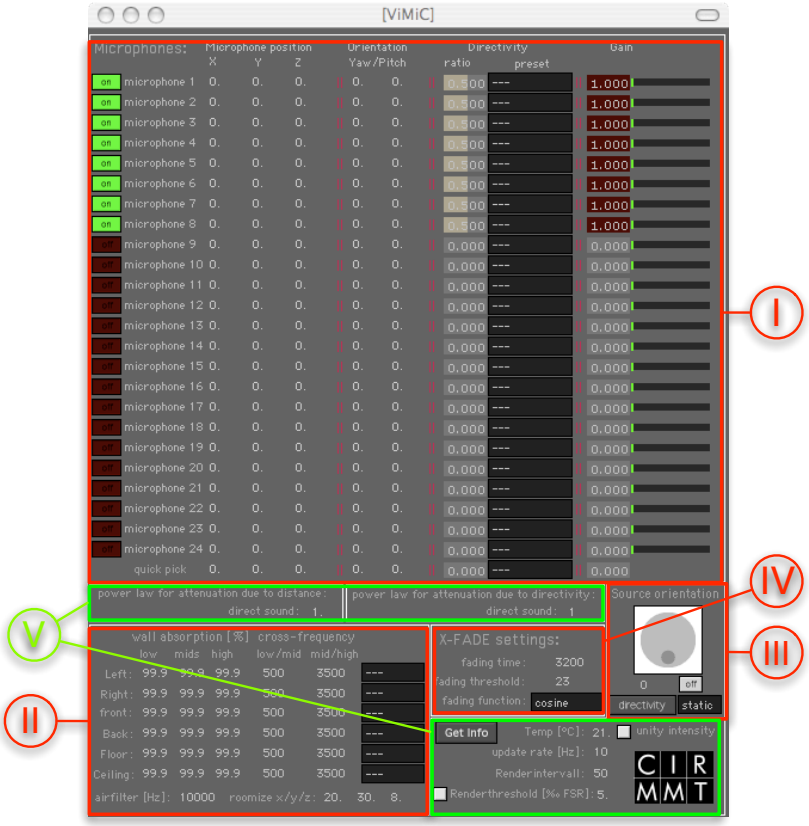


Figure 3.6: ViMiC inspector window

Identifier	Name	Description
I	Channel strips for each microphone	p.17
II	Room model parameters	p.20
III	Source directivity settings	p.22
IV	Settings for X-Fade Render-method	p.24
V	Miscellaneous parameters	p.25

3.2.3 I - Microphone configuration

Figure 3.7 shows a part of the microphone configuration area. Like a channel strip in a mixing board, each strip has the same functionality. Therefore only one channels strip example is discussed here. In order to access a specific microphone channel with an OSC-message, 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 13) determines the highest number a virtual microphone can have. So If (4) is set to eight, only the first eight microphones are displayed. It is possible to change setting of higher numbered microphones through OSC messages. These settings don't get lost! If (4) is changed later in order to render more virtual microphones, this settings will be activated.

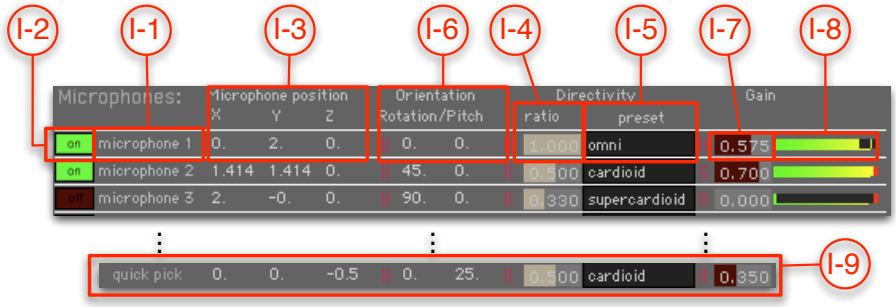


Figure 3.7: Channel Strip of Virtual Microphone

Identifier	Name	Description
I-1	Channel number	p.17
I-2	Active button	p.18
I-3	Microphone position	p.18
I-4	Microphone directivity ratio	p.18
I-5	Microphone directivity presets	p.18
I-6	Microphone orientation	p.19
I-7	Microphone pre-gain	p.19
I-8	VU-meter	p.19
I-9	Quick pick configuration	p.19

(I-1) - Channel number

This displays the number of the related virtual microphone. There is nothing to edit here. The number of the virtual microphone is directly related to the output channel number. So 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 off, the microphone will be muted. If the active button is set to on, the microphone will be unmuted.related OSC-messages [1]:

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

Solo button

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

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

(I-3) - Microphone position

The position of a microphone can either be set in cartesian coordinates or in spherical coordinates.
related OSC-messages:

```
/microphones.#/position <x-coordinate y-coordinate z-coordinate>
/microphones.#/position/z <z-coordinate>
/microphones.#/position/aed <azimuth-angle elevation-angle distance-value>
```

(I-4) - Microphone directivity ratio

As described in section 2.1.1 on page 4, the microphone directivity is defined by the equation:

$$\Gamma = (a + (1 - a) \cdot \cos\delta)^w \qquad 0 \leq a \leq 1 \tag{3.1}$$

The horizontal slider controls the ratio of the parameter a of this equation.
equivalent OSC-message:

```
/microphones.#/directivity/ratio <float>  [0..1]
```

(I-5) - Microphone directivity presets

Rather than manually setting the directivity of a virtual microphone, this drop down menu serves several predefined microphone directivity pattern. The following pattern are available:

- `omni`
- `subcardioid`
- `cardioid`
- `supercardioid`

[1] the # in the OSC-namespace has to be replaced with the number of the virtual microphone (1-24) e.g. for the microphone 5: `/microphones.5/active 1`

- `hypercardioid`
- `figure-eight`

equivalent OSC-message:

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

(I-6) - Microphone orientation

A microphone can point towards 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 A on page 40.

`/microphones.#/orientation <yaw-angle pitch-angle> [-180.0 .. 180.0]`
`/microphones.#/orientation/yaw <yaw-angle> [-180.0 .. 180.0]`
`/microphones.#/orientation/pitch <pitch-angle> [-180.0 .. 180.0]`

Note: /microphones.#/orientation requires float numbers as arguments.

(I-7) - Microphone pre-gain

This option can be used to modify the sensitivity of the microphones separately from each other.
related OSC-message:

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

(I-8) - VU-meter

The VU-meter shows the signal energy after the output volume control. In other words, it is a classical AFL-VU-meter. The integration time is around 300 ms. To save some DSP, the VU-meter can be switched off either due to ticking the entry *Defeat Signal Meters* in the modules pop-up menu (see (1), on page 12).

(I-9) - Quick pick configuration

Sometimes it is not necessary to configure certain microphone parameter differently for each microphones. In this case the quick pick configuration eases the microphone set up: Each modification of one of these quick pick parameter affects all microphones in the same way. For example, if all microphones should have a cardioid directivity, rather than configuring each microphone separately, just select the cardioid directivity from the quick pick area.

3.2.4 II - Room Model Parameter

If early reflections were selected in the front-end menu (see (5) on p.13), the room model becomes visible and also active. Early reflections are calculated according the the position of the sound source, the microphones and the room model (see 2.1.1 on page 6). Here the controllable room parameter are explained briefly.

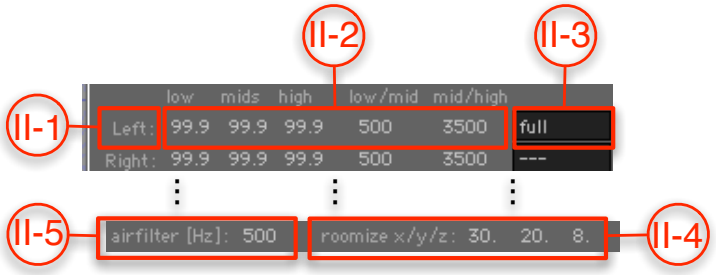


Figure 3.8: Room model parameter

Identifier	Name	Description
II-1	Wall Identifier	p.20
II-2	Absorption parameter	p.20
II-3	Absorption preset	p.21
II-4	Room size	p.21
II-5	Air absorption filter	p.21

(II-1) - Wall Identifier

Tells the name of the wall where the parameter on the right belong to.

(II-2) - Absorption parameter

The absorptive behavior of the room boundaries is modeled through a high-mid-low shelf filter. In each band the wall absorption is defined in %. The bigger the number, the higher the absorptive character. The crossover frequencies between the low mid and high band can be set as well.

```
/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 has to be replaced with the identifier of the wall:

- left
- right
- front
- rear
- floor
- ceiling

(II-3) - Absorption preset

Several surface presets are available to set the Absorption parameter, as explained above. The presets are stored in the textfile entitled `VimicSurfaceProperties.txt`, located in the ViMiC folder. This file can be extended in order to have more presets available.
A preset is stored as following:

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

for example the preset "audience" is stored as:

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

a preset can be accessed with the OSC message:

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

(II-4) - Room size

Here the size of the virtual room is set in width, length and height. The unit is meter. Each dimension is limited to 40 meter.
related OSC-messages:

```
/room/size/xyz <width length height>
```

(II-5) - Air absorption filter

As explained in section 2.1.1 on page 6, the early reflections are filtered to simulate the damping of the air. This 2nd-order lowpass filters all early reflections in the same way. The cut-off frequency can be set through:

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

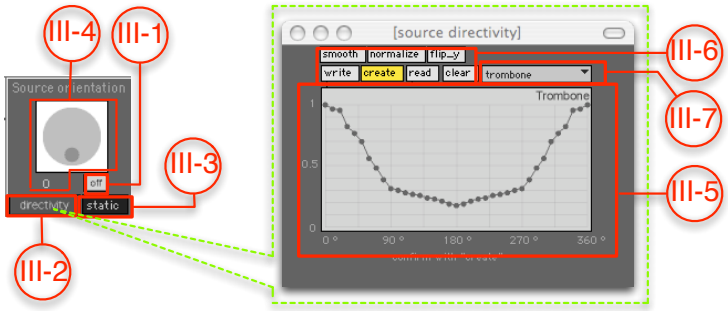


Figure 3.9: Source Directivity options

3.2.5 III - Source Directivity

Identifier	Name	Description
III-1	Source directivity activator	p.22
III-2	Source Directivity Inspector button	p.22
III-3	Directivity mode	p.22
III-4	Orientation wheel	p.23
III-5	Directivity editor	p.23
III-6	Editor tools	p.23
III-7	Directivity presets	p.24

(III-1) - Source directivity activator

With that toggle, the source directivity calculation can be switched on/off.

`/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).

`/source/directivity/openEditor`

(III-3) - Directivity mode

Three directivity modes are implemented:

1. `static`
2. `center`
3. `follow`

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 (III-4)).
center: During a movement, the source is oriented automatically towards the center.
follow: During a movement, the source is automatically oriented in the direction of the movement.

/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 (means the $0°$ direction) of the sound source is pointing. The Orientation is defined in the range of $[-180°..180°]$.

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

(III-5) - Directivity editor

Here, the directivity of a sound source can be freely designed with the help of the mouse. It works like a break point 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, the "create" button (see III-6) has to be clicked.

(III-6) - Editor tools

Some 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: has to be clicked in order to confirm the edited curve.
write: to save a curve in a separate text file.
read: to load a directivity from a extern 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 through the OSC command:

/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.

`/source/directivity/preset <string>`

3.2.6 IV - X-fade setting

The following parameter are specific for the "x_fade" render methods. More information for the "x_fade" render methods can be found in section 2.2.2 on page 8 and in section 3.3 on page 27. These parameter can only be changed if the sound source is not moving.

(IV-1) Fading Threshold

This parameter sets the length in samples, one of the calculated propagation paths has to change, in order to trigger a cross-fade from the current position to a new position.

`/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 which is set in IV-1. The unit of this parameter is samples.

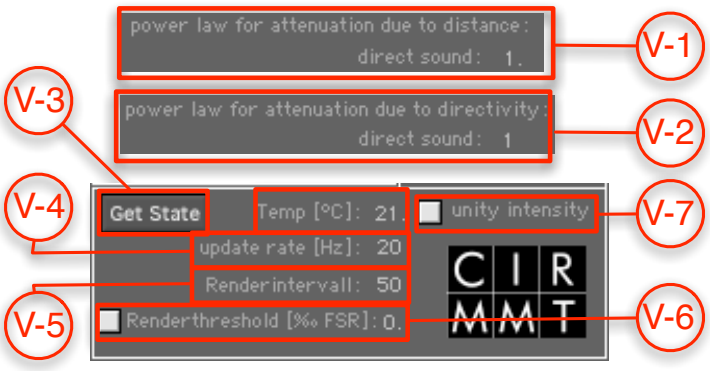
`/rendermode/xfade/fadelength <int> [10 .. 9999]`

(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

`/rendermode/xfade/fadefunction <string>`



3.2.7 V - Miscellaneous parameters

Identifier	Name	Description
V-1	Distance law exponent	p.25
V-2	Microphone Directivity exponent	p.26
V-3	”Get State” button	p.26
V-4	ViMiC update rate	p.26
V-5	ViMiC Renderinterval	p.27
V-6	Renderthreshold	p.27
V-7	Intensity normalization	p.27

(V-1) - Distance law exponent

Two distance functions are implemented, see section 2.1.1 for a comparison of the two distance functions.

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

inverse proportional decrease The parameter sets the exponent q in the attenuation function described in equation 2.1 on page 4. 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.

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

exponential decrease the parameter k from equation 2.2 on page 4 can be modified.

`/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 2.3 on page 5. By increasing the exponent w to a value greater than 1 will produce an artificially sharper directivity pattern. Furthermore, by applying even numbered parameter, anti-phased gain values are avoided. Note, that only integer values are allowed for this exponent.

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

Microphone Polarity

If a microphone directivity ratio smaller than 0.5 is chosen (e.g. a figure-of-eight microphone setting), then certain directivities will be rendered with a negative attenuation factor. 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 an appropriate microphone arrangement, the "Polarity-restricted cosine" panning function described in [Martin et al., 1999] can be achieved.

/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 in order to find bugs ;-).

/getState

(V-4) - ViMiC update rate

To save resources, not every parameter change is causing ViMiC to update (re-render) its virtual microphone signals. This parameter sets the frequency, how often ViMiC is forced to re-render its virtual microphone signals. All messages arrive the ViMiC-external as fast as possible, independently form this update frequency.

/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 to small, audible artifacts appear.

```
/rendermode/intervall <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 set very precisely in one-tenth of a percent of the full scale range. This is an experimental parameter.

```
/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 3.2. This is an experimental option.

$$I = \sum_{i=1}^{n} g_i^2 = 1 \tag{3.2}$$

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

3.3 Rendermodes

ViMiC supports several different Rendermodes which can be chosen from the drop-down menu in the front-end (see (3) on page 13). The following section describes each of them:

3.3.1 ViMiC_XL

If a sound moves, `ViMiC_XL` renders the sound rays with through delay-line interpolation, which causes and audible doppler effect (see section 2.2.1 on page 8).
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 3.2.4 on page 20.

3.3.2 X_fade_XL

`X_fade_XL` renders moving sound sources with the cross-fade method, described in section 2.2.2 on page 8. The special X-Fade settings in the ViMiC inspector (see section 3.2.6 on page 24 come into play. 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.

3.3.3 ViMiC_lite & X_fade_lite

The render methods `ViMiC_lite` and `X_fade_lite` differs from their bigger brothers `ViMiC_XL` and `X_fade_XL` in the quality of the room model: Instead of having the choice of a individual wall absorption parameter for each wall, the entire virtual room has the same kind of wall absorption. To edit the wall absorption, the parameters of the "left" wall in the ViMiC inspector (the first wall in the ViMiC Room settings see 3.2.4 on page 20) have to be manipulated.

3.3.4 Static

This rendermode is similar to `ViMiC_lite`, but no delay-line interpolation is performed. This setting is meant to be a cheap and dirty solution – sometimes it doesn't need more than this.

3.3.5 Panning

The Panning 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. Regarding of what order of reflections was selected in (5), no early reflections are calculated, because the reflections would appear undelayed. If the gain normalization option is active (V-7), and microphone characteristic is set to `omni`, DBAP-like (distance based amplitude panning) [Lossius, 2007] panning is created.

3.3.6 Comparison

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

+++ indicates processor overload

Max/MSP Version 4.6.3

numbers are taken from the internal Max/MSP "CPU Utilization" info

Chapter 4

The Jamoma module jmod.sur.vimic8~

This module contains basically 8 jmod.sur.vimic8~ 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 8 times. Because the module renders 8 sources, it has a lot more inputs than the regular jmod.sur.vimic8~. The first inlet is (as usual in Jamoma) reserved for communication via OSC. The next 8 inlets are conventional signal inlet to connect audio sources directly to the rendering algorithm. The rightmost inlet is a multicable inlet for convenient patching. The first 8 signals inside this multicable are internally separated and routed to the rendering algorithm. Because

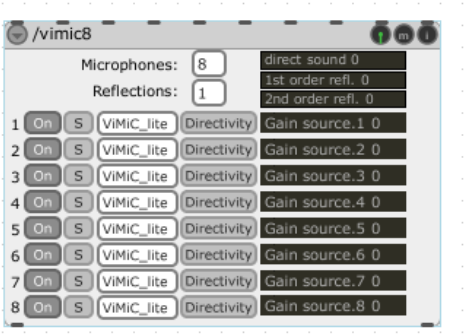


Figure 4.1: jmod.sur.vimic8~

there are now 8 sources to be controlled, the OSC namespace is slightly different 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:

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

4.0.7 Description message

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

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

4.1 The Jamoma module jmod.sur.vimic8poly~

The module jmod.sur.vimic8poly~ is basically the same as jmod.sur.vimic8~, but uses the multi threading functionalities of Max5. If you are using a multi-core computer, the rendering of the eight sound sources can be distributed amongst different cores. There are two new parameter to control the multi-threading.

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

if `/parallelization/numThreads 0`, then Max will automatically use as many cores as available. Another number than 0 specifies explicitly how many cores are going to be used. These two parameter can also be manipulated over the panel:

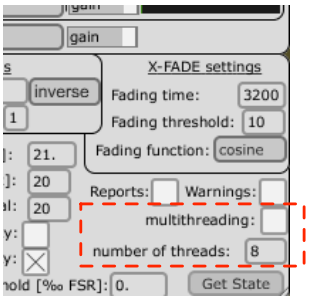


Figure 4.2: multithreading options in jmod.sur.vimic8poly~

Chapter 5

The Jamoma module `jmod.sur.reverb~`

5.1 Introduction

The late reverb is implemented as a feedback delay network (FDN) with 16 delay lines, displayed in Figure 5.1. The use of FDN for artificial reverberation can be justified fundamentally by a stochastic model of late reverberation decays assuming that sufficient density of acoustic model in the frequency domain and of reflections in the time domain are achieved. Under these assumptions, later reflections can be modeled as a Gaussian exponentially decaying random process, characterized by a spectral envelope. This approach cannot guarantee the same degree of accuracy a convolution with a measured impulse response, but provide a more efficient parametrization for dynamic control of the synthetic reverberation effect like reverberation time, modal density or reverb coloration.
Any FDN can be represented as a set of digital delay lines whose inputs and outputs are connected by a feedback/mixing matrix which defines the FDN structure. 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 (see Figure 5.2). The advantage of this matrix is that it can be efficiently implemented with a butterfly algorithm using $N \log 2N = 64$ additions. As proposed in [Dattorro, 1997] the delay-lines in the FDN can be modulated by low frequency oscillators in terms of delay-length and attenuation coefficient.

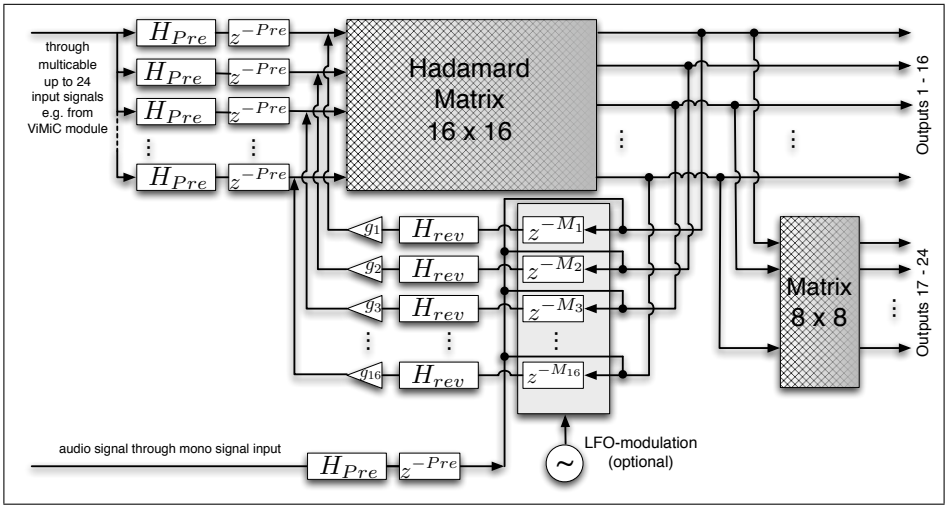


Figure 5.1: FDN-reverb flowchart

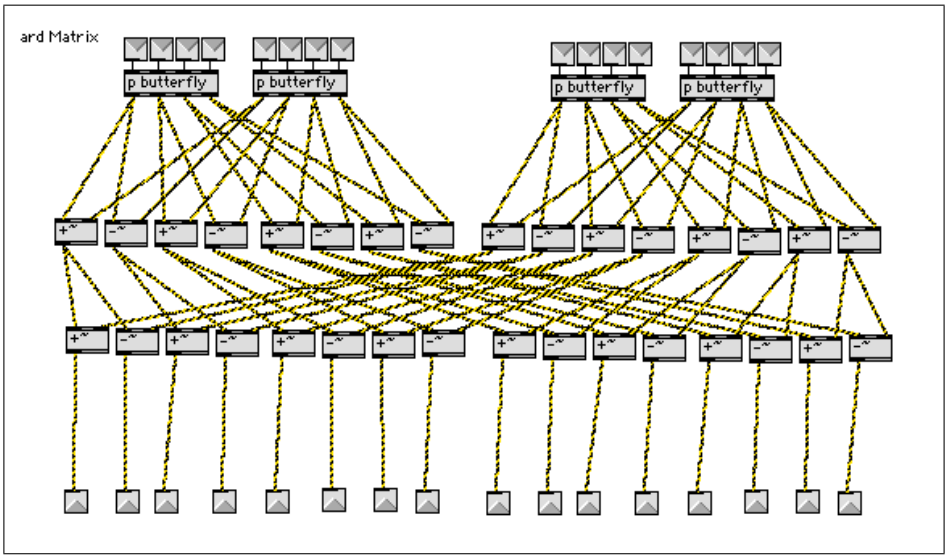
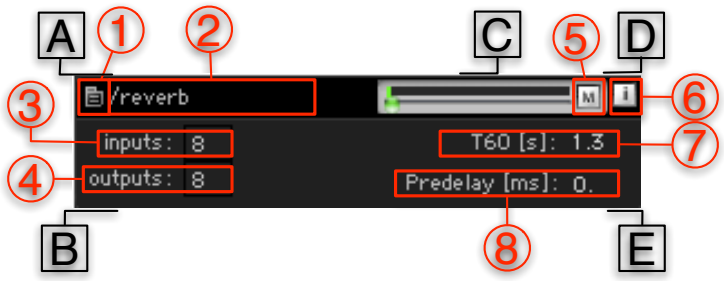


Figure 5.2: FDN Hadamard matrix

5.2 Parameter & Controlling

5.2.1 Front-end



Identifier	Name	Description
A	Message inlet for OSC-commands	p.34
B	Message outlet for OSC-commands	p.34
C	Multichannel signal inlet	p.34
D	Signal inlet	p.34
E	Multichannel signal outlet	p.34
1	Module pop-up menu	p.34
2	Modules OSC name	p.34
3	Number of input channels	p.35
4	Number of output channels	p.35
5	Mute button	p.35
6	Inspector button	p.35
7	Reverb time T_{60}	p.35
8	Pre-Delay	p.35

Reverb Connections

NOT WRITTEN YET

Module pop-up menu

NOT WRITTEN YET

Modules OSC name

NOT WRITTEN YET

Number of input channels

NOT WRITTEN YET
realted OSC namespace:

```
/inputs <int> [1, 2, 3, 4, 5, 8, 12, 16]
```

Number of output channels

```
/outputs <int> [1, 2, 3, 4, 5, 8, 12, 16, 24]
```

Mute button

```
/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 parameter described in 5.2.2.
equivalent OSC-message:

```
/view/panel
```

Reverb time T_{60}

Reverberation time in ms.

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

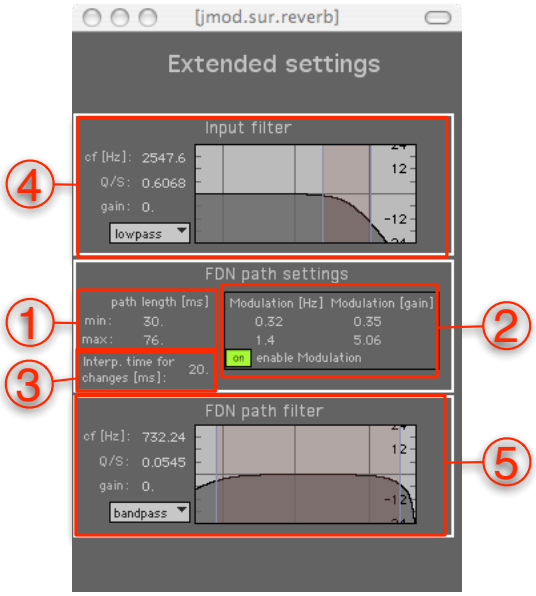
Pre-Delay

This parameter delays the incoming signals of the reverb to a specified time in ms. (described as z^{-pre} in Figure 5.1)

```
/t60 <float> [0...1000.0]
```

5.2.2 Inspector window

Identifier	Name	Description
1	Range of the FDN delay lengths	p.36
2	FDN Modulation options	p.36
3	Interpolation time	p.36
4	Input filter settings	p.36
5	FDN path filter settings	p.37



Range of the FDN delay lengths

Minimal and maximal delaylength of the feedback paths in [ms].

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

FDN Modulation options

```
/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 for changes of feedback path length and path modulation.

```
/interpolationtime <float>
```

Pre-filter settings

Parameter for the filter described as H_{pre} in Figure 5.1.

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

FDN Filter settings

Parameter for the filter described as H_{rev} in Figure 5.1.

```
/fdn/frequency  <float> [Hz]
/fdn/filtergain <float> [dB]
/fdn/filtertype <string>
        [lowpass, highpass, bandpass, bandstop, peaknotch, lowshelf, highshelf]
/fdn/q  <float>
```

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 5.2.2 on page 36).

	Outputs	8	16	24
Inputs	FDN-modulation			
1	off	10 %	10 %	10.5 %
1	on	11.5%	11.5 %	12 %
8	off	10.6%	10.6 %	11 %
8	on	12 %	12 %	12.5 %
16	off	11 %	11 %	12.3 %
16	on	12.5 %	12.5 %	13 %

Table 5.1: Max/MSP Processor load for one sound source at different late reverb settings, Max/MSP overdrive option disabled , $f_s = 44.1kHz@16Bit$, measured at mac mini 1.66 GHz intel core duo. Numbers are taken from the internal Max/MSP "CPU Utilization" info.

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Appendix A

Coordinate System

ViMiC uses a right-handed cartesian coordinate system, which limits are defined by the virtual room size (see. 3.2.4).

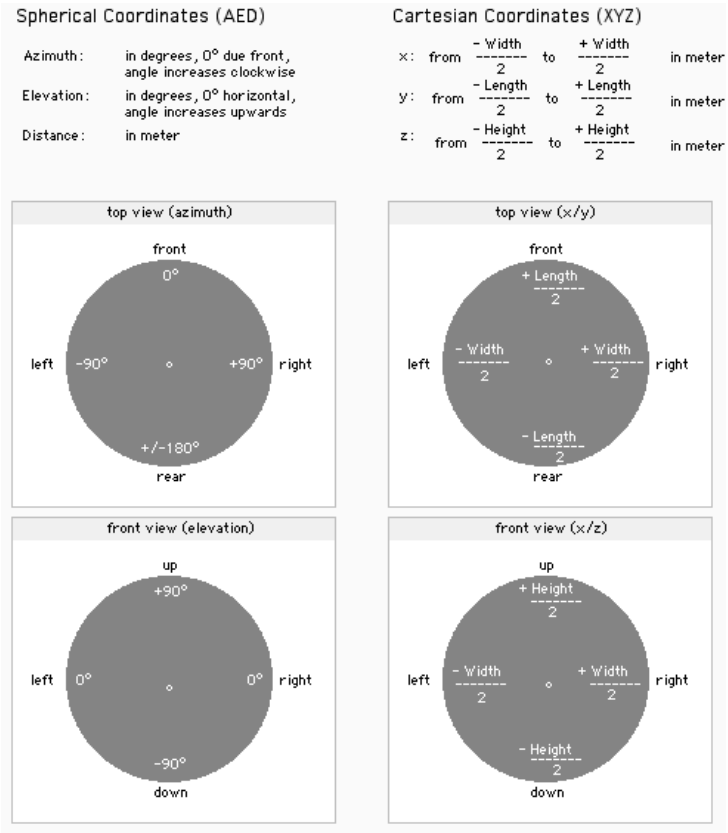

Figure A.1: Coordinate Systems: Spherical (left) and Cartesian (right)

Appendix B

Jamoma's MIDI to gain conversion

All messages controlling gain properties are expecting values in the MIDI range (0 - 127). The following table B.1 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 aplification, whereas MIDI values below 100 will cause an attenuation.

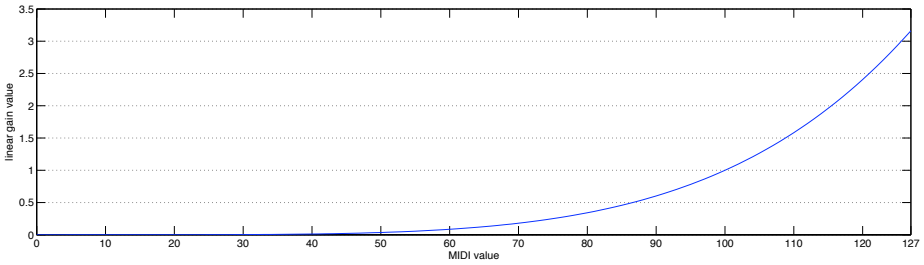

Figure B.1: Jamoma's MIDI to gain conversion

Appendix C

OSC namespace

On this page I want to list the entire OSC namespace for ViMiC. As long this is not written, please refer to the file `jmod.sur.vimic~.html` and `jmod.sur.vimic8~.html` in the ViMiC directory. If anybody know a working html2tex converter, please let me know.

Appendix D

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_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 beeing ramped.
- Quick pick configuration (see p.19) 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.~~