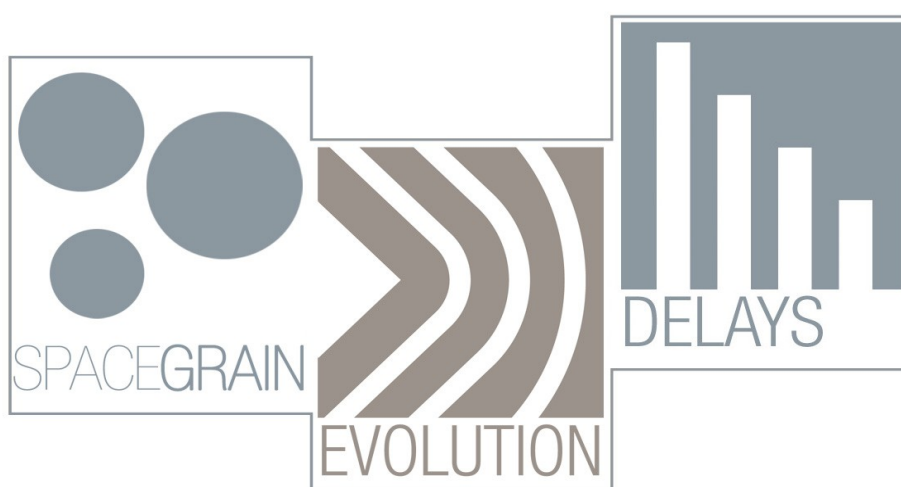


# GRM Tools Creative

## User's Guide



version 3.7

# Introduction

GRM Tools Creative is a bundle of three plug-ins that provide superb tools for sound enhancement and design. They are available in AAX, Audio Unit and VST (32 & 64 bits) and RTAS (32 bits) formats, and also as Stand-Alone application. Conceived and realized by the Groupe de Recherches Musicales (Musical Research Group) of the National Audiovisual Institute, Paris, France, GRM Tools is the result of numerous years of research and development by composers and sound designers in sound transformation software.

The following pages will take you through the installation and authorization process, describe the innovative interface devices created to make these plug-ins intuitive and musical, and explain the operations of the individual plug-ins.

**Installation**  
**Authorization**  
**Universal Controls**  
**Midi Management**  
**Multi-Channel**  
**Stand-Alone**

**Delays**  
**Evolution**  
**SpaceGrain**

# Installation

## Mac OS X

The installer puts the different files in the following folders :

- RTAS : Library\Application Support\Digidesign\Plug-Ins
- AAX : Library\Application Support\Avid\Audio\Plug-Ins
- VST : Library\Audio\Plug-Ins\VST\GRM
- Audio Units : Library\Audio\Plug-Ins\Components
- Stand Alone : Applications\GRM
- Documentation : GRM Tools Documentation

It installs also the Interlok Drivers necessary for authorization of the software.

To launch the installer, double-click on the **GRM Tools Creative Installer.pkg** icon and follow the instructions. If you do not want to install all available architectures, select "Customize" in the "Installation Type" dialog and select the desired components.

## PC Windows

The installer puts the different files in the following folders :

- RTAS : Program Files\Common Files\Digidesign\DAE\Plug-Ins
- AAX : Program Files\Common Files\Avid\Audio\Plug-Ins
- VST : GRM folder in the default VST folder (usually : Program Files\Steinberg\Plug-Ins\VST) or Program Files\GRMVstPlugins if the default folder is not defined.
- Stand Alone : Program Files\Ina-GRM\GRM Tools
- Documentation : My Documents\GRM Tools Documentation

It installs also the Interlok Drivers necessary for authorization of the software.

On 64 bits systems, 32 bits plugins and applications are installed in « Program Files (x86) » .

To launch the installer, double-click on the **GRM Tools Creative Installer.exe** icon and follow the instructions.

# Authorization

When you make your purchase, you will receive a serial number and a link to the [iLok.com](https://iLok.com) website enabling you to manage your licenses. Take great care of the serial number, as it constitutes proof of your purchase. You will need it to authorize the plug-in and to contact the technical support unit.

## Authorize GRM Tools Creative

Authorization is carried out through the iLok License Manager application. iLok License Manager is a new application for Mac and PC that allows you to more easily manage your licenses and iLok dongles.

The iLok dongle is no longer required. GRM Tools licenses can be either moved on the computer or on an iLok dongle.

- Create an account on [iLok.com](https://iLok.com)
- Download and install the latest iLok License Manager
- Launch iLok License Manager and sign in with your [iLok.com](https://iLok.com) ID
- Select « Redeem Activation Code » in « Licenses » menu
- Copy the activation code
- Drag the generated license on a location displayed on the left column (iLok dongle or computer).

Documentation and videos showing details of dongle and license management are available on the [iLok.com](https://iLok.com) website.

# Universal Controls

Every plug-in contains a variety of interactive controls to vary, display, store, recall, and otherwise manipulate parameters. The controls are:

- Sliders
- Elastic String
- Numerical Value Fields
- Buttons
- Presets
- Tempo
- SuperSlider
- Agitation
- Save/Load
- Window resizing

## Sliders and 2DControllers



To change the value continuously, drag the handle to the left or right.

To jump to a new value, click along the path of the handle.

To reset the default value, click on the handle while pressing the [Alt] key.

Only for 2DControllers, a click on the handle while pressing the [Shift] key limits movements to vertical or horizontal displacements.

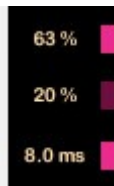
## Elastic String



To achieve smooth movement of Sliders, 2DControllers, or the SuperSlider, click on the object and move the mouse while holding down the [Command] key on Mac or [Ctrl] key on PC.

Note: In general, the followspeed of the smoothing function depends upon the length of the Elastic String.

## Numerical Value Fields



Note that a slider is often associated with a Numerical Value Field that shows the parameter value as a number or other alphanumeric character. You can change the value in a Numerical Value Field directly.

To change the value in a Numerical Value Field, click within the field and drag vertically upwards to increment a value or downwards to decrement a value. To modify the increment of change, drag while pressing the [Command] key on Mac or [Ctrl] key on PC.

Double clicking in a Numerical Value Field opens an editor enabling the direct modification of a parameter value. Clicking outside the field or the [Return] key ends value editing.

**Caution:** with some host applications, the [Return] key is interpreted as a host command and does not therefore end the editing of the value. In this case, click outside the editable field to end editing.

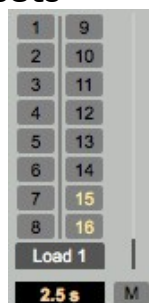
## Buttons



Buttons are used to change a state or start a process.

To change the state of a Button, or to start a process, click on the Button.

## Presets



Each plug-in has memorization capacities for all settings, and recall capacities for the memorizations.

Gradual transition from one preset to another is carried out by linear interpolation of

parameter memories. The memorisation zone is at the right of each window. It includes sixteen memorization boxes, an interpolation time control slider and a status field.

- To save your current configuration of parameter settings into any of the 16 Preset locations, click on a location number while holding down the [Command] key on Mac or [Ctrl] key on PC. Note that the Status Field, located under the Preset locations, gives the indication of the last performed operation.
- To call up a configuration of parameter settings from any of the 16 Presets, click on that Preset number. Note that the Status Field indicates 'Load'.
- To reload the factory configuration of parameter values, click while pressing down the [Alt] key. Note that the Status Field indicates 'Reset'.
- Note that the factory default settings for presets 15 and 16 are random values. Preset 15 applies a random variation of about 10% deviation from the current value as set by the user. Preset 16 generates a completely random set of parameter values.

The timing of the change from current values to the recalled Preset values is determined in one of two ways:

- It can be determined by the current position of the vertical Slider located to the right of the Preset numbers. To change the time of interpolation between current parameter values and Preset values, move the vertical Slider up or down to reflect your preferred timing.
- It can be recalled as one of the parameters saved in the Preset. To recall the time of interpolation from a Preset configuration along with other parameters, click on the Interpolation Button (M), which is just underneath the vertical Slider, to activate it before you click on the Preset.

During interpolation, clicking on a slider or a value field, stops the interpolation of this parameter. The other parameters continue to be interpolated. To completely stop the interpolation, click the Status Field.

A preset content can be copied and pasted into another preset of the same kind. For instance, a BandPass VST preset can be pasted into another BandPass VST or even into a BandPass StandAlone.



A right-click on a preset opens a pop-up menu allowing to copy the preset into the clipboard. When a compatible preset is available, its number is shown and it can be pasted into the chosen preset. This new preset is now loaded.

## Tempo

This feature is only available with certain applications such as Cubase SX, ProTools, etc..

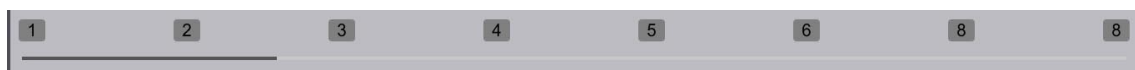
Certain parameters, such as the time of interpolation between presets, can be synchronized with variations of tempo in a Pro Tools session. To initiate synchronization, hold down the [Shift] key and click on the Slider or Numerical Value Field associated with the parameter you want to synchronize. The display will indicate in bold characters the figures that represent the possible tempos. The figures and the tempos they represent are:

<b>T/64</b>	64th-note triplets
<b>1/64</b>	64th-notes
<b>T/32</b>	32nd-note triplets
<b>./64</b>	dotted 64th-notes
<b>1/32</b>	32nd-notes
<b>T/16</b>	16th-note triplets
<b>./32</b>	dotted 32nd-notes
<b>1/16</b>	16th-notes
<b>T/8</b>	8th-note triplets
<b>./16</b>	dotted 16th-notes
<b>1/8</b>	8th-notes
<b>T/4</b>	quarter-note triplets
<b>./8</b>	dotted 8th-notes
<b>1/4</b>	quarter-notes
<b>T/2</b>	half-note triplets
<b>./4</b>	dotted quarter-notes
<b>1/2</b>	half-notes
<b>T/1</b>	whole-note triplets
<b>./2</b>	dotted half-notes
<b>1/1</b>	whole-notes
<b>./1</b>	dotted whole-notes
<b>1 bar</b>	measure

Larger numbers are indicated in numbers of measures, as in 3 bar for three measures.

The parameters of each plug-in that can be synchronized are itemized in the sections dealing with the individual plug-ins.

## SuperSlider



The horizontal SuperSlider and its associated Numerical Value Fields, located at the bottom of each window, provide a powerful control for interpolating between Presets. Use the SuperSlider to interpolate between any sequence of Presets to find new configurations and create new Presets.

- To select a Preset number in a Numerical Value Field, click in the field and drag vertically upwards or downwards.
- To change continuously from one Preset to another, drag the handle of the SuperSlider to the left or right.
- To disable a Numerical Value Field, click in the field and drag vertically downwards to an 'Off' position.



## Agitation

This set of controllers enables the adding of random variations to the processing parameters. The left hand rotating potentiometer gives the amplitude (from 0% to 100%) of the random variation. The right hand rotating potentiometer gives the speed of the variations (from 0 to 60 s). The On/Off button under the two rotating potentiometers activates the variations.

Four agitation groups are available:



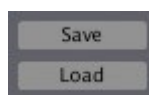
A left click on the coloured button **63 %** to the right of each alphanumeric value validates agitation for each individual parameter. A right click on the button opens the agitation group selection menu.



When the button is bright, the parameter is subjected to the variation. When it is dark, the variation has no effect.

**Caution :** In the default configuration, agitation is de-activated for all parameters.

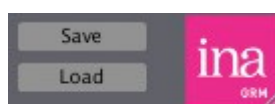
## Save/Load



As an alternative to the save / load action in the host application, each plug-in contains Save / Load Buttons located in the bottom of the windows. These buttons allow you to save configurations of your plug-ins in a folder that you choose. They also allow you to exchange configurations of GRM Tools plug-ins in other environments in which GRM Tools is used.

- **Save** opens the file selector to save the complete configuration (current values of the parameters and the 16 presets).
- **Load** opens the file selector to recall a complete configuration (current values of the parameters and the 16 presets).

## Window resizing



The plug-in window can be resized by clicking and dragging using the small triangle in the bottom right hand corner of the ina-grm logo.

**Caution :** Excessively large dimensions may slow down the display of data in the plug-in interface.

## Midi management

All the processing parameters can be controlled by MIDI messages.

Important note: Some applications (for example Logic) do not send MIDI messages directly to the processings, but propose other solutions to bind the messages to the parameters.

### MIDI messages

The MIDI messages recognized by the processings are the following channel messages:

- Control Change
- Note On
- Pitch Wheel
- Program Change
- 

The discrete controls (buttons, menus, Preset) behave differently depending on the messages:

- **Control Change** : the message values (from 0 to 127) are mapped on the parameter variation range. For example, for a button, the values 0 to 63 trigger the “released” state, and the values 64 to 127 trigger the “pressed” state.
- **Pitch Wheel** : similar to the previous control, but with a 14 bit message value range, that is from 0 to 16383.
- **Program Change** : The program numbers correspond to the state of the parameter. For example, **Program Change** 1 and 2 correspond to the “released” and “pressed” state of a button. Alternatively, **Program Changes** from 1 to 16 correspond to the 16 processing presets.
- **Note On** : A basic note is bound to the parameter.
  - For parameters with two states (buttons, or two-choice menus) each sending of the **Note On** message with the same basic note switches the state of the parameter. The other **Note On** messages have no effect, and can therefore be used to control other parameters.
  - For parameters with more than 2 states (Preset, menus, etc.), the basic note corresponds to the first state, and the following note to the second state, etc. For example, if the LA 440 (midi 69) is bound to the Preset parameter, the LA will load preset 1, LA# preset 2, SI preset 3, etc. **Note On** messages outside the parameter variation range (for example, notes below LA440 in the previous example) are not taken into account.

The **Note On** messages can be processed in a special way by certain types of processing (for example, for **Evolution** transposition parameters). Refer to the description of each type of processing for more information on these special cases.

## Binding a Midi message to a parameter

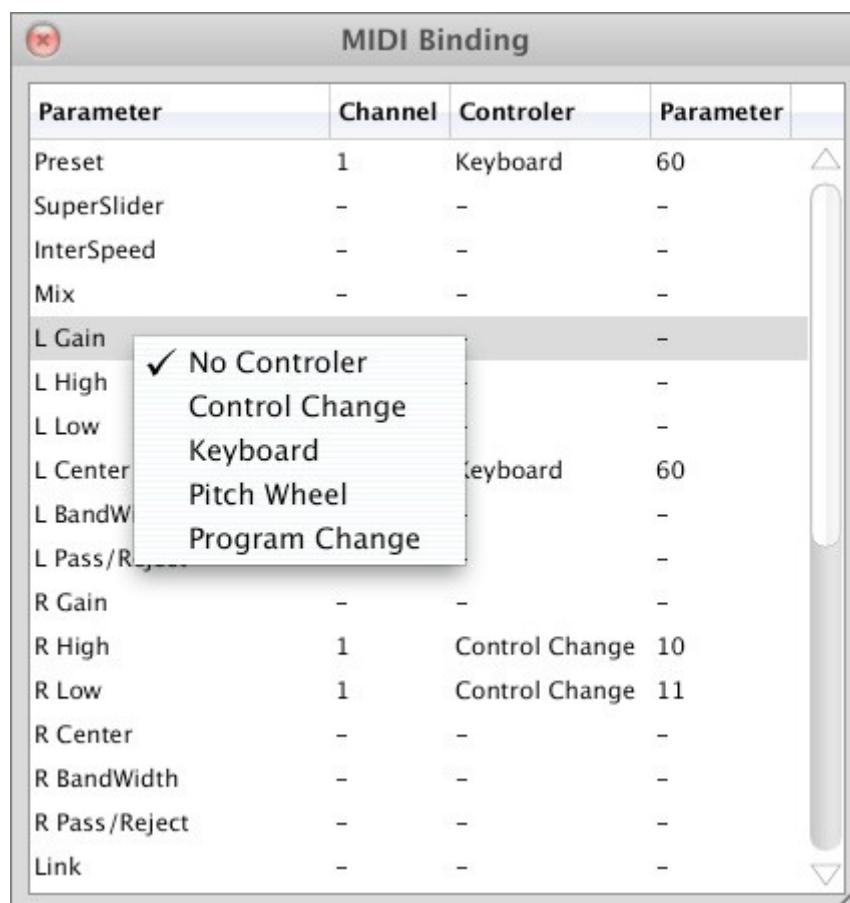


Click on the MIDI button located next to the ina-grm logo, and underneath the **Save** and **Load** buttons. A panel opens to the left of the button. Manipulate the parameter you want to control on the plug-in interface. Its name is displayed on the **Parameter** line. Then send the corresponding Midi message, which is displayed on the **Message** line in the following format:

[channel number][controller name][optional parameter]

The binding between the Midi message and the parameter is carried out and memorized.

- **Unbind** : cancels the Midi binding of the parameter displayed
- **Close** : closes the panel
- **Reset All** : cancels the Midi binding of all parameters.
- **View** : opens the window shown below, which enables the viewing of all bindings, their modification and the addition of new ones.



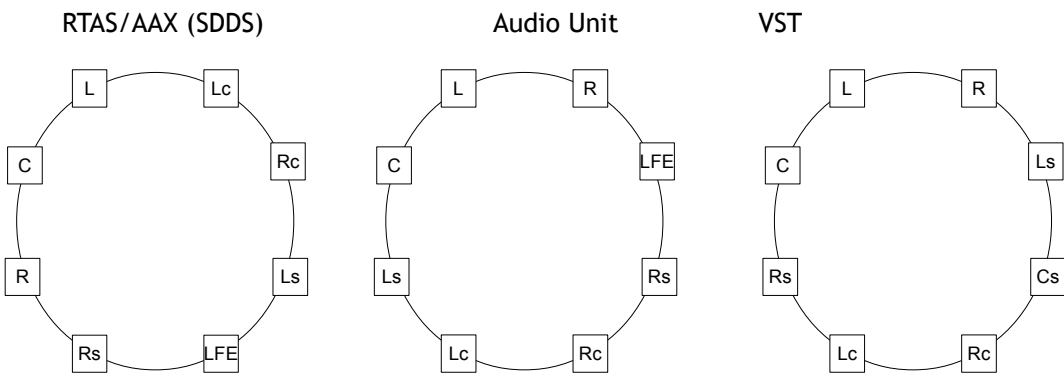
# Multi-channel

Some GRM Tools plug-ins propose outputs on several channels in AAX, RTAS, VST and Audio Units.

The channel allocations follow the standard configurations of each system:

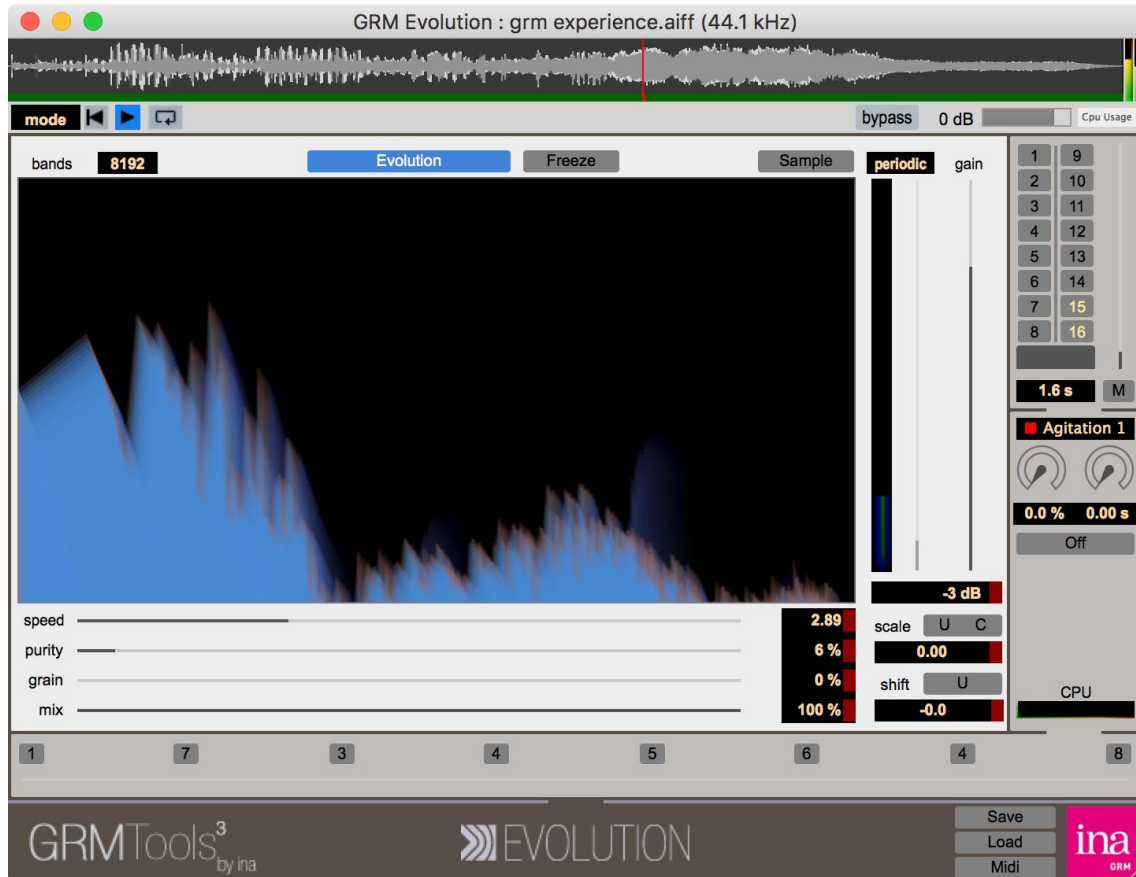
Configuration	RTAS/AAX	Audio Units	VST
Quad	L R Ls Rs	L R Ls Rs	L R Ls Rs
5.0	L C R Ls Rs	L R Ls Rs C	L R C Ls Rs
5.1	L C R Ls Rs LFE	L R C LFE Ls Rs	L R C LFE Ls Rs
7.1 (8.0)	L Lc C Rc R Ls Rs LFE	L R C LFE Ls Rs Lc Rc	L R C Ls Rs Cs Lc Rc

Configurations 7.1 & 8.0 are non-standard. It is a regular octophonic configuration without a central channel staging successive stereo couples.



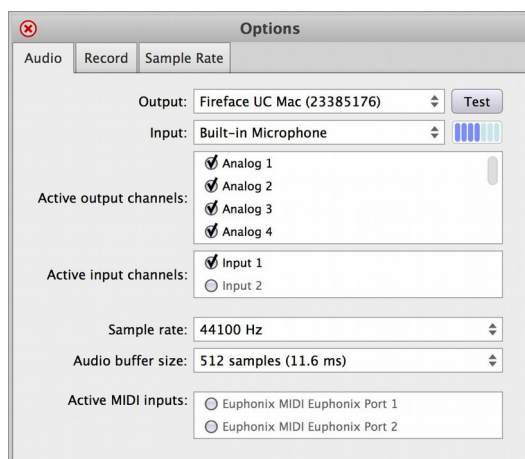
# Stand Alone

The following descriptions only concern the stand-alone versions.



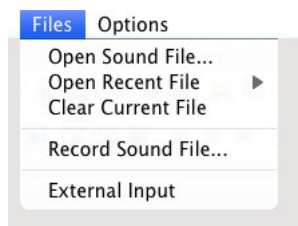
In this version, the processing window includes at the top a strip to control the reading and writing of sound files.

Multichannel processing is available for Delays, Doppler, Freeze, Reson & Suffling. To select the number of output channels open the **Audio** tab in **Option->Audio & Midi Setting** menu :



The channel allocation is the Audio Units channel allocation (see above)

To select a sound input:



Select **External Input** in the **Files** menu to process an external sound.

Slide an audio file from a file browser into the horizontal grey zone at the top of the window

Select **Open Sound File...** or **Open Recent Files** in the **Files** menu.

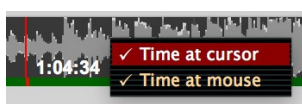
The file is loaded with a selection equal to its total time. To modify the start of the selection, click close to the start and slide the mouse. To modify the end, click close to the end and slide the mouse. Click&Drag inside the selection lets you simultaneously control the start and the end of the selection. A single click moves the cursor under the mouse.


The green bar under the waveform controls the zoom of the display.



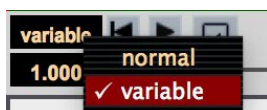
The green zone corresponds to the displayed part, the red to the non-displayed. A click in the green zone followed by a vertical drag expands or narrows the zone. A horizontal drag moves the zone in time. A clic in the red zone moves the nearest bound.

A right clic on the waveform shows a contextual menu allowing to display the current time under the mouse and/or under the cursor.



The  buttons control the read cursor. The first button sends the cursor back to the start of the selection, the second starts the reading, or pauses it, and the third one loops reading on to the selection.

Two playing modes are available :



normal and variable. In variable mode, a slider and an alphanumeric field allow to change the playing speed from -2 (double speed backward) to +2 (double speed forward)

To record a sound file:

select **New Output Sound File...** in the **Files** menu.

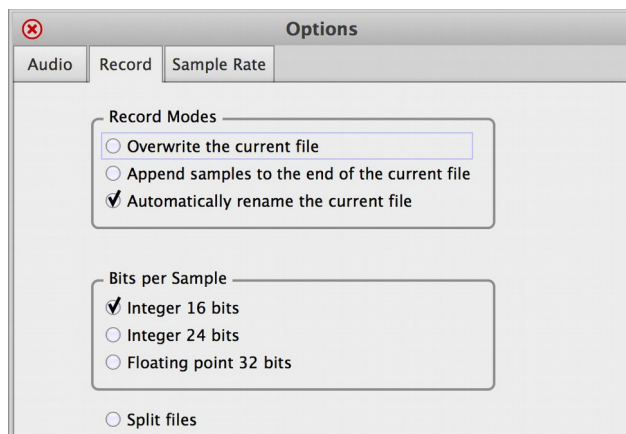
The file will be created in the WAV format. We recommend adding the extension .wav to avoid any confusion at a later stage.

When an output file is open, the appearance of the advance button bar changes:

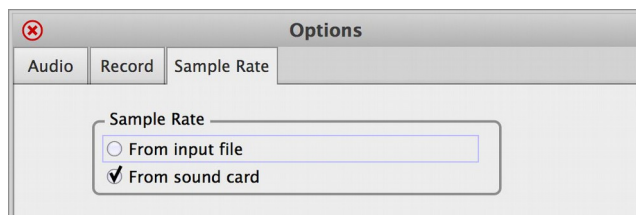


A new button can be used to start and stop recording. The name and the path of the file are indicated, and its time.

**Option->Audio & Midi Setting, Record tab** allows to choose the file resolution (16, 24 bits integer or 32 bits floating point) and the record mode:



**Split File** option allows to split the output file in multiple mono files.



**Sample Rate** option allows to choose the sample rate when the file sample rate and the output sample rate are different:

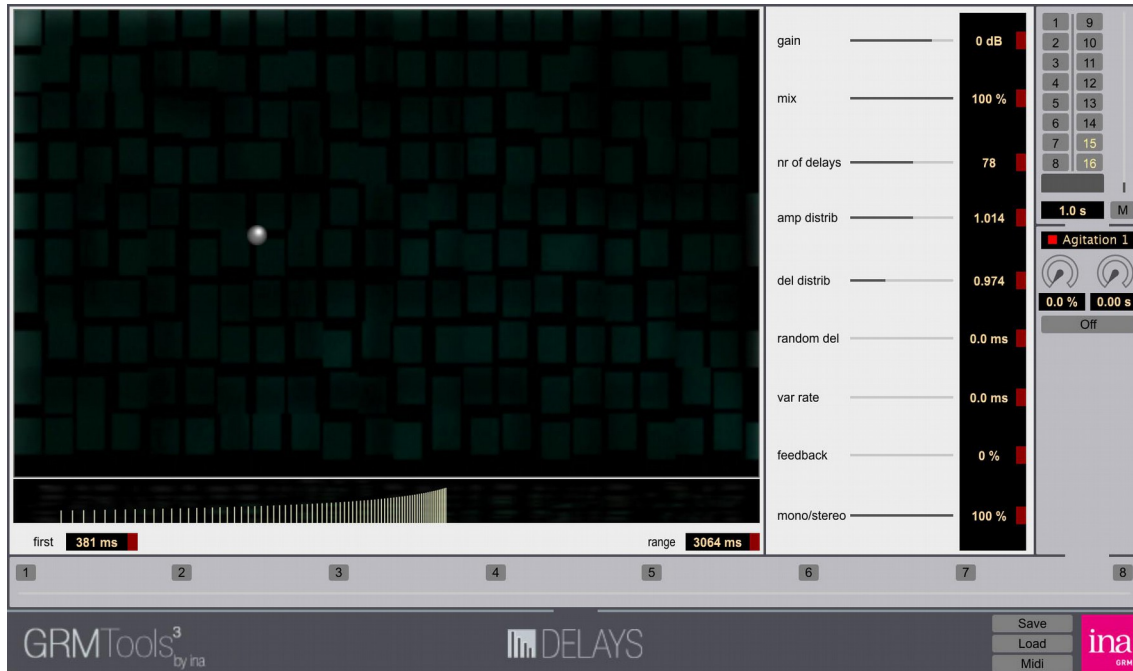
**From input file** : the output sample rate is set to the file sample rate, if possible.

**From sound card** : the output sample rate is not modified, and the file is resample to this rate.



# Delays

This plug-in lets you create and control any type of delay line from echo to reverberation to phase shift, soften instrumental attacks, create a wide variety of special effects ...



## How does it work?

Delays consist of a set of delays which are variable from 0 to 6 seconds. Depending on the power of the processor, it is possible to have up to 128 simultaneous delays, whose amplitude and position are controlled. In the stereo version, the delays are assigned alternately to the left hand and right hand channels.

The controls are further explained below in the section called **Reference**.

## A quick tutorial

We assume that you are familiar with basic operations, that you know how to record or import a sound into a mono or stereo track and how to access the different windows and insert plug-ins. For this tutorial, use a series of staccato sounds or a voice counting.

Click on preset 2. Start the playback.

When you hear the sound, move the **mix** slider to the left to about 70%. You'll hear the original plus the delays at equal intervals, fading out.





Move the **amp distrib** slider to the right. Note that the amplitude pattern of the delays changes to equal and then to soft-to-loud. Set the slider wherever you like.

Move the **del distrib** slider to the left. Note that the delays get faster towards the end of the range. Now move it to the right and notice that the delays get slower towards the end of the range. Set the slider to approximately 1.

Now, to introduce some random elements in the delay pattern, move the **random del** slider to the right to about 400 ms and set the var rate slider at about 76 ms. Note the irregularities in the delay timing.

Store any configuration of the variables in a **Preset** and perform any sequence of Presets with the **SuperSlider**. If you're not familiar with these controls, have a look at the page called **Universal Controls**.

## Reference

### gain

Sets the level of the input signal from -96 dB to + 24 dB.

### mix

Determines the proportion of the processed signal that is mixed with the original signal. At 0%, only the original signal is heard. At 100%, only the processed signal is heard. In the stereo version, the same settings are applied simultaneously to both channels.

### nr of delays

Number of delays. Selects a number of delays between 2 and 128. Note that the effective maximum number depends upon the power of your computer and the number of plug-ins you are using. Note also that when this parameter is varied, the amplitudes and the timing of all of the delays are recalculated.

### amp distrib

Amplitude distribution. Specifies the amplitude of each delay relative to the previous delay. For example, an amp distrib value of 2 signifies that each delay will have an amplitude that is double the amplitude of the preceding delay. Values of less than 1 mean that successive delays will have less amplitude than preceding delays. A value of 1 means that delays will have equal amplitude.

### del distrib

Delay distribution. Specifies the timing of each delay relative to the timing of the previous delay. Values of more than 1, for example, mean that each delay will be longer than the previous delay and that, consequently, there will be a continually longer time between successive repetitions of a sound. A value of 2, for example, means that each delay will be twice as long as the previous delay and that the time between repetitions of a sound will be twice as long as the time between the previous repetitions of the sound. Values of less than 1 mean that the time



between repetitions of a sound will be shorter and that sounds will occur more and more quickly. A value of 1 means that sounds will occur at regular intervals.

**random del**

Randomized delays. Specifies a random number from 0 to 1000 ms which is used to vary the timing of the delays. The actual timing will result from the combined settings for **first**, **range** and **del distrib** plus the random number between 0 and the number specified by **random del**.

**var rate**

Rate of variation, used in conjunction with **random del**. Specifies the rate at which random numbers are generated.

**feedback**

Specifies the percentage of the delayed signal which will be fed back into the input.

**mono/stereo**

Sets the distribution of delays to the outputs, from 0 (monophonic, same signal to both tracks) to 100 (2-track output).

**first**

Sets the timing of the first delay to a time between 0 and 5914 ms.

**range**

Sets the difference in time between **first** and the last delay. Note that **del distrib** sets the distribution of delays within this range.

**2DController**

The vertical dimension controls the range of the delays in time. The horizontal direction controls the center of the range. Note that the window under the control window contains a graphic display of the values for **first**, **range**, **nr of delays**, **amp distrib**, **del distrib**, **random del** and **var rate**.

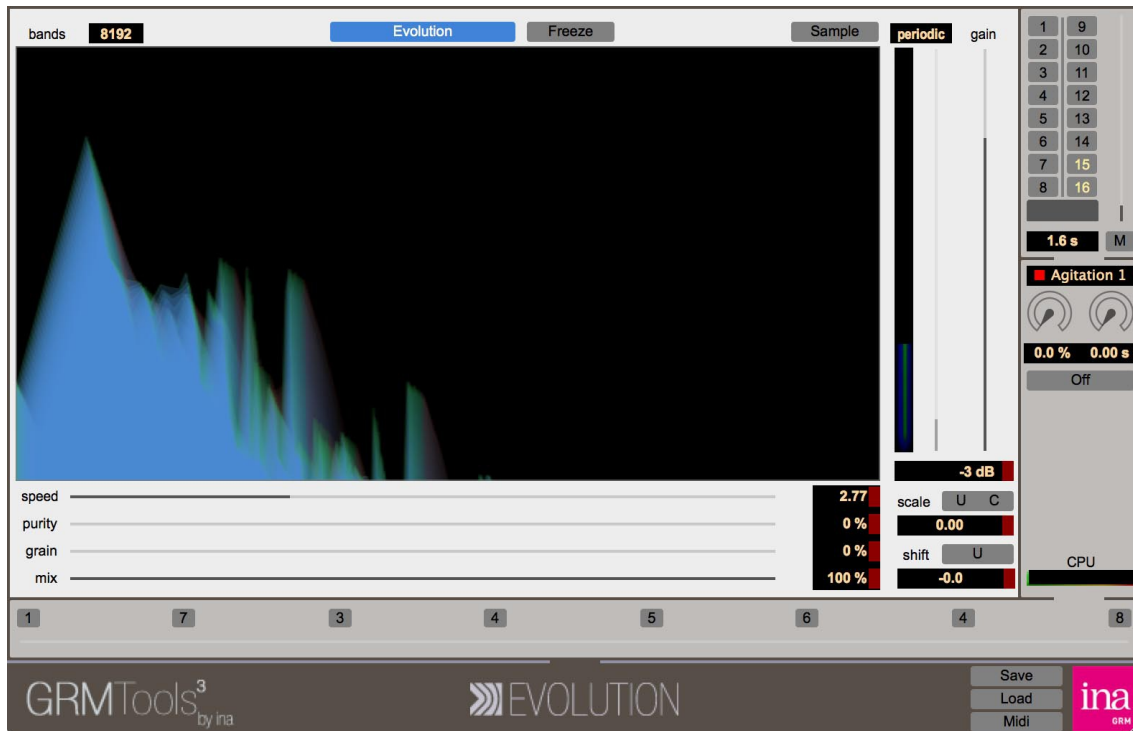
**tempo**

The parameters that can be synchronized to tempo are the interpolation time between presets, **first** and **range**. To initiate synchronization, hold down the [Shift] key and click on the Slider or Numerical Value Field associated with the parameter.



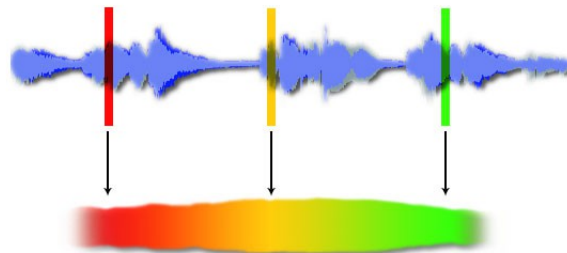
# Evolution

Evolution is used to obtain continuous evolution of timbre by frequential sampling of the input signal.



## How does it work?

The timbre of the input signal is sampled at more or less regular intervals. The output signal is obtained by interpolation between the sampled timbres. In the following example, the three sampling instants are represented by the colours red, yellow and green. The resulting signal is a frame obtained by interpolation between the red timbre and the yellow timbre, and then between the yellow timbre and the green timbre. This signal is represented by the red → yellow → green shading.





## A quick tutorial

We assume that you are familiar with host application, that you know how to import or record a sound into a mono or stereo track, and how to access the different windows and insert plugins. Carry out the following examples using orchestral sounds or wide bandwidth sounds.

Start reading while maintaining the default values. Click the **Evolution** button.

The sound evolves at a rate of 3 seconds. Observe the vertical gauge on the right which indicates the progression of the interpolation.

Modify the value of the **Speed** slider and observe its influence on the interpolation speed.

Click the **Evolution** button again to hear the original sound. Select **One Shot** in the menu above the interpolation gauge. Set the **Speed** value to 0. Click the **Evolution** button.

The sound is frozen. Clicking the **Sample** button samples the original sound and freezes it again. The **Speed** slider sets the interpolation time of the current status to the new frozen timbre.

Return to **Periodic** mode in the menu above the interpolation gauge. Set the **Speed** value to around 3 seconds. Use the **Purity** slider.

The more this value increases, the more the sound is simplified and becomes 'electronic'. **Purity** filters the components of the sound according to their amplitudes. A high **Purity** value will only allow the strongest components to pass.

Set **Purity** to 0 or to a very low value, and select 256 in the **Bands** menu (to the left of the **Evolution** button).

The sound has become 'electronic' again because there are now only 256 sine waves to describe its evolution. Use the **Grain** slider which will add random variations to the amplitudes and frequencies of each component.

Record the settings in **Presets** and play the sequence with the **Superslider**. Refer to the page on **Universal Controls** for more information about controls.



## Reference

### Bands

Number of sine wave components used to represent the sound (from 128 to 65536). The higher this number, the more faithfully the timbre will be reproduced, but the less the attacks and rapid variation will be conserved. The value of 8192 bands is a good compromise.

### Evolution

This button switches processing from the bypass mode to the interpolation mode. When interpolation is activated, the button is blue.

### Freeze

This button stops the interpolation. It is active only in interpolation mode.

### Sample

This button forces a new sampling of the sound input.

### Mode

Defines the interpolation mode.

**Periodic.** A new sampling is carried out at the end of each interpolation, and a new interpolation is initiated automatically.

**One-Shot.** The sound remains frozen at the end of each interpolation. The Sample button must be pressed to carry out a new sampling, and activate an interpolation.

**Auto.** Another sampling is carried out each time the input timbre is modified. The vertical gauge displays the variation rate (calculated by the spectral flux), and the vertical slider just to the right of the gauge is used to adjust the trigger threshold.

### gain

This parameter controls the output signal level from -96 to +24 dB.

### Speed

Interpolation speed. Varies from 0 (no interpolation) to 60 seconds.

### Purity

Selects the components used for interpolation according to their amplitudes. At 0, all the components are used, and the timbre obtained is very close to the original timbre. When the value increases, only the largest components will be used, and the timbre is simplified and becomes 'electronic'.



### Grain

Adds random variations to the frequencies and amplitudes of each component. These variations are only carried out inside each band. The effect will be all the more noticeable if the number of bands is low. Almost inaudible at 8192 bands, it adds slight roughness at 1024 bands, and a very noticeable noise at 128 or 256 bands.

### Mix

This parameter controls the balance between the original sound and the processed sound. At 100% only the processed sound can be heard. At 0% only the original sound can be heard.

### Scale

Used to select a value between -24 and 24 which is used to multiply the frequency of each spectral component and to obtain a transposition.

A value above 1 will transpose each spectral component upwards. A value of 24 will transpose each component two octaves higher.

A value below 1 will transpose each spectral component downwards. A value of -24 will transpose each component two octaves lower.

To lock the chosen value, click the **U/L** button so as to display the initial **L**. To change the values, display the **U**.

To limit the transposition to integer-valued half-tones, click the **C/D** button to display the initial **D** (discrete). To obtain continuous variation, display **C** (continuous).

### Shift

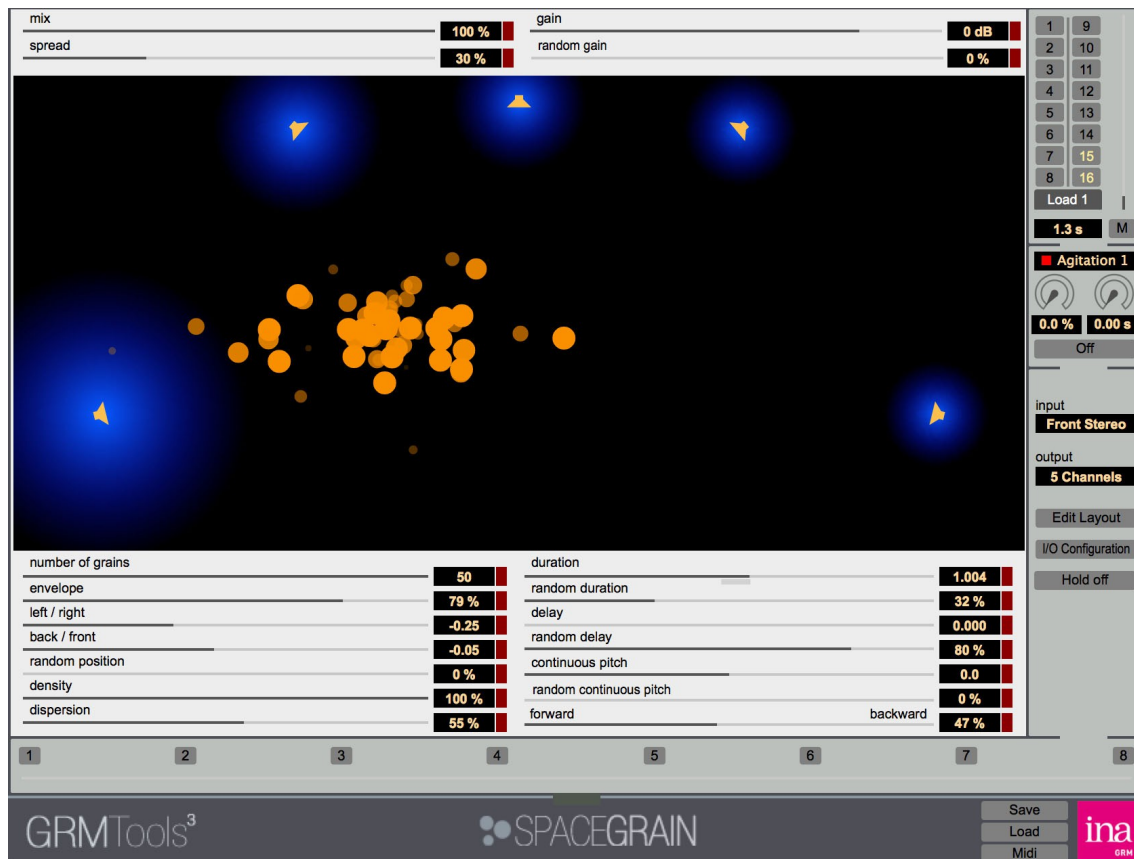
Used to select a value between -4000 and +4000 Hz so as to remove it from or add it to each of the spectral components of the original sound.

To lock the chosen value, click the **U/L** button so as to display the initial **L** (lock). To change the values, display the **U** (unlock).



# SpaceGrain

**SpaceGrain** enables the generation of up to 100 grains and their placing in a multi-channel sound space.



## How does it work ?

Variable duration, delay, amplitude and transposition grains are sampled in the incoming signal and distributed in a multi-channel space (up to 32 channels, depending on the environment). Each grains performs a mono read. The input channels (up to 8) are randomly allocated to the grains.

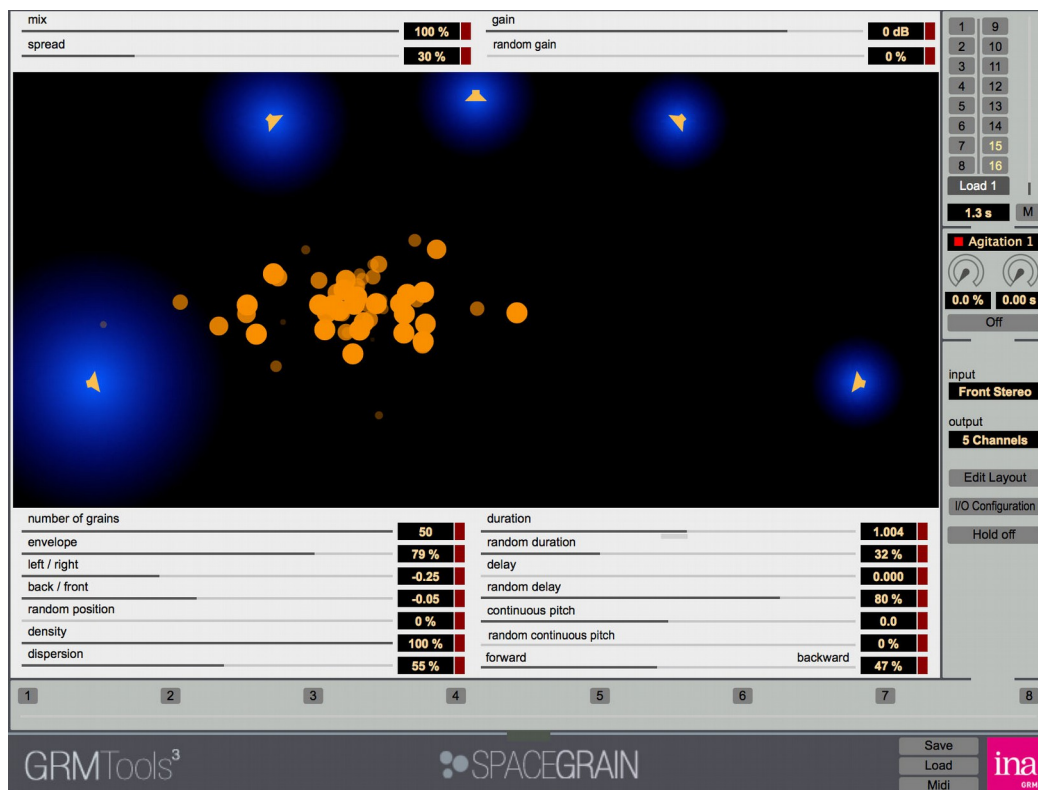
The controls are explained below in the paragraph entitled **Reference** (page 25).



## Getting started

It is assumed that the user is familiar with the manipulation of the host applications, the importing and recording of a sound on a mono or stereo track, access to the various windows and the insertion of the plugins. The following examples have been carried out using the Stand Alone version of SpaceGrain (see the Stand Alone chapter, page 13, to find out more about this version). These examples are transposed without difficulty to the other versions (VST, AAX, RTAS and Audio Unit). It is assumed that the user has an audio interface with at least 5 outputs.

Start SpaceGrain StandAlone. Open **Audio & Midi settings** in the **Options** menu. Select 5 outputs in the **active output channels** list. Close the **Options** panel. The application has now been configured for an output on 5 channels.



Check that the menu above **Edit Layout** does indeed indicate **5 Channels**. Open this menu: only compatible layouts with fewer than 5 channels can be selected (Stereo and Quadraphonic).

Load a stereo sound file and start the reading. Observe the orange discs appearing in the main window. They symbolize the grains. You can modify their quantity with **number of grains**, their duration with **duration** and their spatial distribution with **random position**.

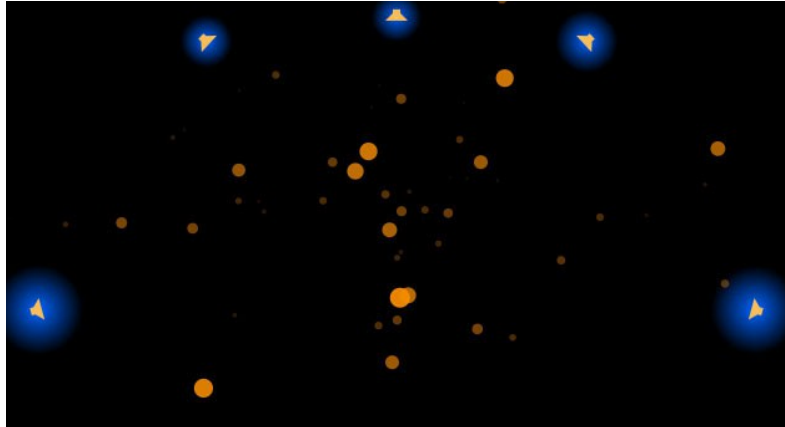
Set **number of grains** to 10, **duration** to 2 and **random position** to 30%. Click in the window and move the mouse. The grains will follow the mouse with a delay that depends on their duration.





## Reference

### Spatial window



This window displays a multi-channel space symbolized by N speakers that correspond to N output channels. The grains are displayed as orange circles whose intensities depend on the envelope and amplitude of each grain.

#### clic + move

moves the center of the grains distribution in the multi-channel space

#### right clic

displays a context menu



**Display Gains.** The blue haloes represent the amplitude of each speaker.

**Display Levels.** The blue haloes represent the actual level of the signals sent to each speaker, just as with a VUmeter.

**Display Infos.** Displays the numbers, names, output channels and amplitudes of dB levels of each speaker.

**Rectangular Grid.** Displays a rectangular grid as the window background.

**Polar Grid.** Displays a polar grid as the window background.

**Open Layout Editor.** Opens the layout editor (see page 28)

#### mix

This parameter controls the mix between the original sound and the processed sound. At 100% only the processed sound is heard, and at 0% only the original sound is heard.

#### spread

controls the spread of the influence of the output channels. At 0, there is very little overlap between the channels, and at 1 the influence is almost constant over the whole space and the overlap is maximum.

**gain**

This parameter controls the level of the output signal from -96 to +24 dB.

**random gain**

Applies a random variation on the amplitude of each grain. At 0%, no attenuation is applied, and at 100% attenuation varies randomly between 0 dB and -40 dB.

**number of grains**

The number of grains varies from 1 to 50. Depending on the **envelope** parameter, grains may overlap more or less. In the event of overlapping, the quantity of effective grains at a given moment can reach 100. Each grains performs a mono read. The input channels are randomly allocated to the grains.

**envelope**

Amplitude envelope of each grain. At 0% the envelope is rectangular, and the transition from one grain to the next is sudden. At 100%, the envelope is triangular, and the grains are superimposed for half of their duration.

**left / right**

Left/right coordinate of the position of the centre of the source. From -1 (left) to 1 (right).

**back / front**

Back/front coordinate of the position of the centre of the source. From -1 (back) to 1 (front).

**random position**

Random variation of the position of the grains. At 0%, all the grains are positioned at the last known position of the mouse. At 100%, the grains are distributed over the whole work space.

**density**

Density of grains that are in fact active. At 0% no grain is active, and 100% all grains are active.

**dispersion**

The grains move linearly outward from the cursor position in a random direction. 0 % no movement, 100 % maximum dispersion, about half of the main window.

**duration**

Duration of grains from 1 ms to 10 s. This duration may be automatically modified as a function of the pitch parameters to avoid certain reading incoherencies.

**random duration**

Random variation of duration of grains. At 0% duration is determined by the **duration** parameter. At 100%, duration varies randomly between 1 ms and the duration determined by the **duration** parameter.



### delay

Delay of each grain relative to direct sound. The delay varies between 0 and 10 s. The delay may be automatically modified as a function of the pitch parameters to avoid certain reading incoherencies.

### random delay

Random variation of delay. At 0% the duration is that determined by the **delay** parameter. At 100% the duration varies randomly between 0 s and that determined by the **delay** parameter.

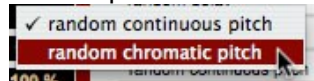
### continuous/chromatic pitch

Transposition of + or - one octave. The transposition is performed by reading at variable speed. Depending on the duration of the grains, the acceleration, or the slowdown, of the content of the grains will be perceptible or not. A click on the label opens a menu for selection of continuous or chromatic transposition.



### random continuous/chromatic pitch

Random transposition of + or - one octave. The random variation is added to the previous **pitch** parameter. A click on the label opens a menu for selection of continuous or chromatic transposition.



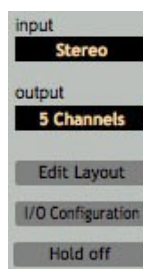
### forward/backward

Proportion of grains that are played backward. At 0% all grains are read normally, 100% all grains are played in reverse.

### input

Menu enabling the selection of the input layout. Only layouts with a number of channels lower than or equal to the number of channels of the track on which the processing is inserted can be selected. Only the number of channels is taken into account, and not the detailed layout of the channels. **Stereo** and **Front Stereo** are therefore equivalent.

With **I/O Configuration**, this menu enables the selection of the input channels that are actually used by the grains.





## output

Menu enabling the selection of the output layout. Only layouts with a number of channels lower than or equal to the number of channels of the track on which the processing is inserted can be selected.

## Edit Layout

Button that opens the layout editor. See the description of this editor on page 28.

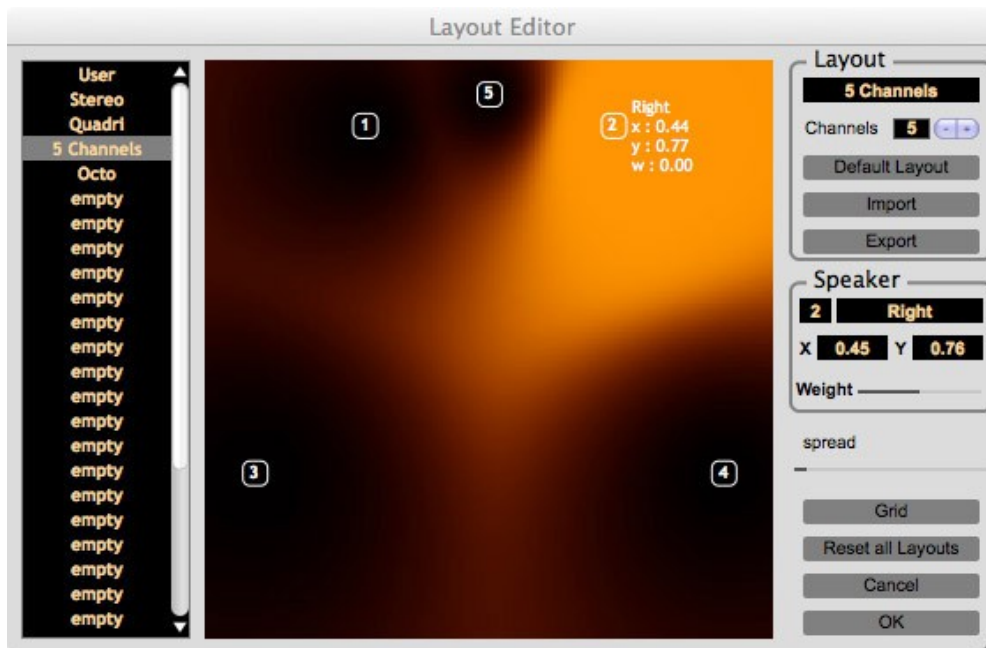
## I/O Configuration

Button that opens the input/output editor. See the description of this editor on page 30.

## hold

Button enabling the suspension of the writing of the incoming signal (Hold on). The grains are then sampled in the last 10 seconds recorded.

## Layout editor



The left hand column lists the 32 possible layouts.

- Clicking on a name displays the layout in the central window, and its name and characteristics in the Layout frame in the top right hand corner.
- Click+drag is used to move an item in the list
- Shift+Click+drag copies the start item into the end item.

The central window displays the selected layout. The orange zone represents the influence zone of the selected output channel. A source located in this zone will be transmitted in the corresponding channel with an amplitude that is proportional to the intensity of the color. This zone has no significance when the layout is used for a source. The **spread** potentiometer controls the spread of the



influence. At 0, there is very little overlap between the channels, at 1 the influence is almost constant throughout the space, and the overlap is maximum. The channels can be freely moved in the space displayed. When resizing the window, press the [Shift] key to maintain the proportions of the central window.

The **Layout** frame on the right is used to:

- **edit** the name of the layout
- **change** the number of channels. The direct edition of the numerical field initialises the position of the speakers in the default layout. Pressing the + or - button either adds a speaker in the centre or deletes the speaker without modifying the position of the other speakers.
- **apply** the default layout, depending on the number of channels
- **import** and **export** layouts in XML files. The file format is as follows:

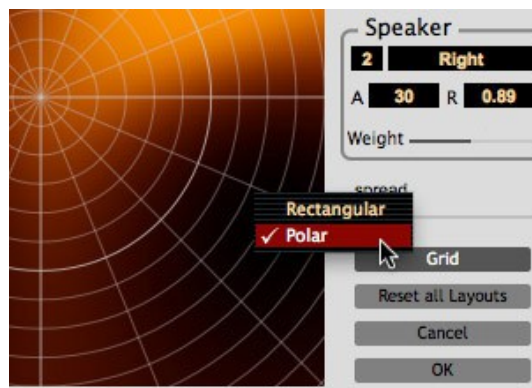
```
<?xml version="1.0" encoding="UTF-8"?>
<Speakers name="Quadri">
  <S x="-0.625" y="-0.625"/>
  <S x="0.625" y="-0.625"/>
  <S x="-0.625" y="0.625"/>
  <S x="0.625" y="0.625"/>
</Speakers>
```

The **name** attribute indicates the name of the layout as it will appear in the list, and the attributes **x** and **y** are the coordinates of the speakers between -1 and 1. The number of channels is equal to the number of **S** items.

The **Speaker** frame on the right is used to:

- **select** a speaker by its number
- **edit** its name (Right in the example shown)
- **modify** its position
  - using Cartesian coordinates (X and Y) by default
  - using polar coordinates (Angle and Radius) if Grid polar is selected. (0° at top, 90° on right, 180° at bottom and 270° on left)
- **modify** its influence using the Weight parameter (in dB)

**Grid**



displays a magnetic grid which can be rectangular or polar.

**Reset all Layouts**



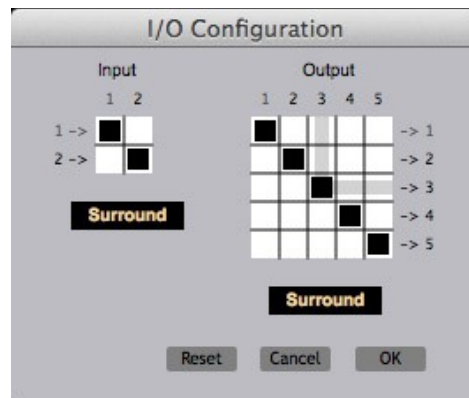
Resets the layout list:

1. Mono
2. Frontal stereo
3. Quadriphony
4. 5 channels
5. Symmetrical octophony
6. Centered octophony
7. Stereo

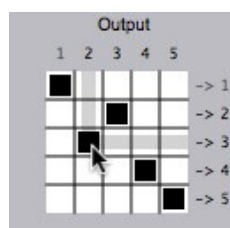
The following layouts (empty) are reset in frontal stereo

### I/O configuration

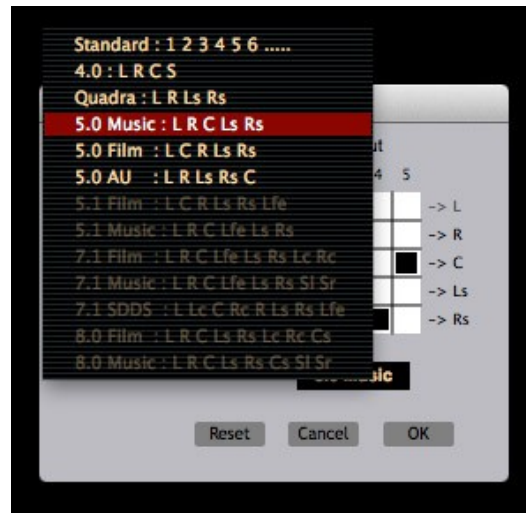
In most cases, the distribution of the input-output channels is carried out automatically (see the multi-channel chapter, page 12). To modify this distribution, click on the **I/O configuration** button:



In this example, **Spaces** is configured with a stereo input and a 5 channels output. The sequencer track channels (n-> or ->n) correspond to the matrix lines. The input and output channels of **Spaces** correspond to the matrix columns. To direct an output channel of **Spaces** towards an output track channel (or an input track channel towards an input channel of **Spaces**), click in the corresponding box of the matrix.



In this example, speaker 2 of **Spaces** is routed towards channel 3 of the output track. To help identify the channels, the surround menus placed under the two matrices propose the most standard configurations:



Only configurations compatible with the number of channels can be selected. The track channel numbers are then replaced by their standardized names (L, R, Ls, etc.) and the configuration of the matrix is performed automatically. Caution: the names of the channels and of the configuration are not memorized. This is only a matrix definition aid.

It is thus possible to manage up to 32 inputs and 32 outputs:

# Credits

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Emmanuel Favreau, Suzy Vincens, NO PUB

Documentation  
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Translation  
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La création et la recherche  
dans le domaine du son  
et des musiques électroacoustiques