GRM Tools Evolution

User’s Guide

Version 3.7

Introduction

GRM Tools Evolution is a new bundle of three plug-in forming part of the GRM Tools collection. It is 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

Evolution
Fusion
Grinder

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 Evolution 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\VSTPlugins 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 Evolution Installer.exe** icon and follow the instructions.

Authorization

Buy GRM Tools Evolution

When you make your purchase, you will receive a serial number and a link to the iLok.com website enabling you to authorise the plug-in. 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 Evolution

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
- Download and install the latest iLok License Manager
- Launch iLok License Manager and sign in with your 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* 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

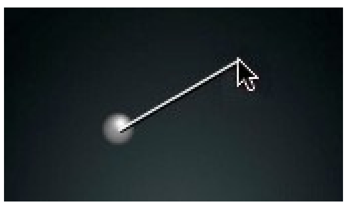
To change the value continuously, drag the handle to the left or right.

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.

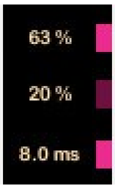
Only for 2DControllers, a click on the handle while pressing the [Shift] key limits movements to vertical or horizontal displacements.

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.

Boutons

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, an Evolution VST preset can be pasted into another Evolution VST or even into an Evolution 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:

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

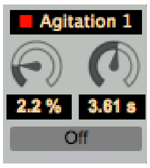
The horizontal SuperSlider and its associated Numerical Value Fields, located at the bottom of each window, provide a powerful control for interpolating between Presets. Use the SuperSlider to interpolate between any sequence of Presets to find new configurations and create new Presets.

- 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

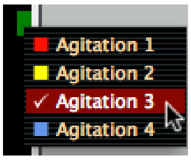
This set of controllers enables the adding of random variations to the processing parameters. The left hand rotating potentiometer gives the amplitude (from 0% to 100%) of the random variation. The right hand rotating potentiometer gives the speed of the variations (from 0 to 60 s). The On/Off button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

Four agitation groups are available:



A left click on the coloured button 

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:

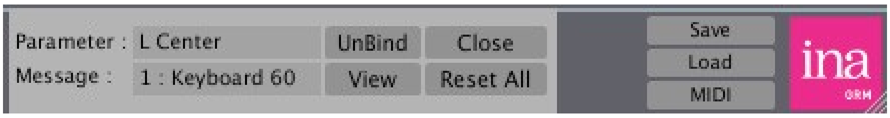
- Control Change
- Note On
- Pitch Wheel
- Program Change
-

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

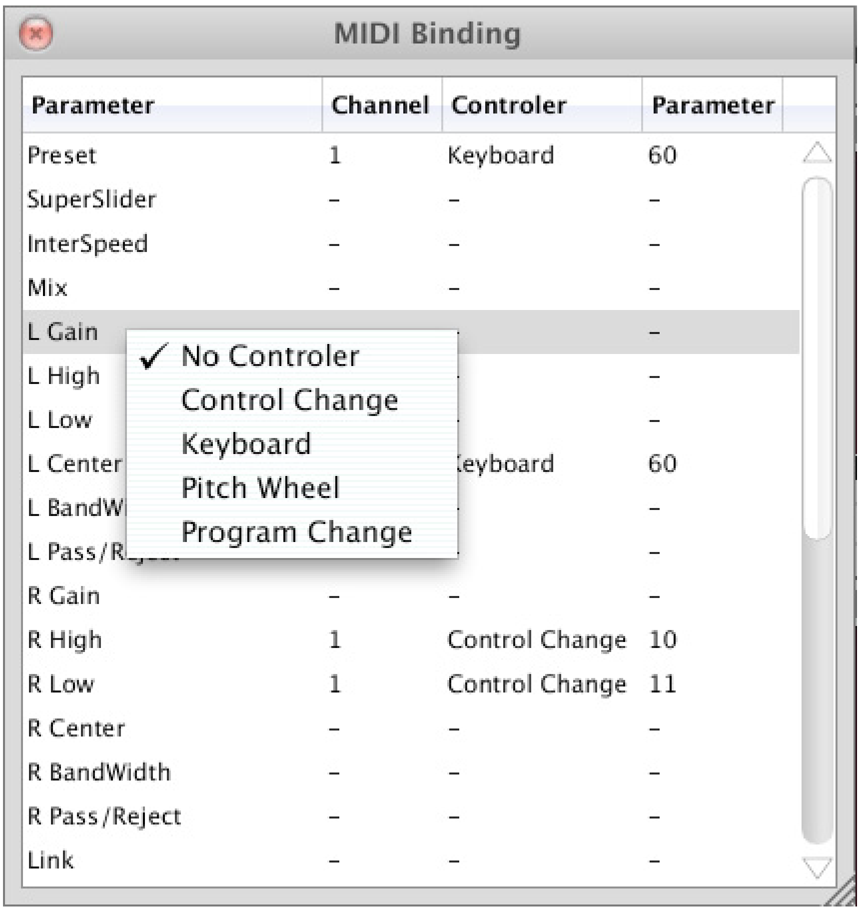


Click on the MIDI button located next to the ina-grm logo, and underneath the **Save** and **Load** buttons. A panel opens to the left of the button. Manipulate the parameter you want to control on the plug-in interface. Its name is displayed on the **Parameter** line. Then send the corresponding Midi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

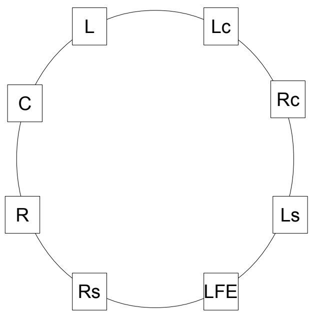
Fusion proposes outputs on several channels in AAX, RTAS, VST and Audio Units.

The channel allocations follow the standard configurations of each system:

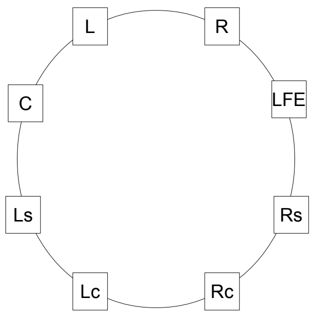
Configuration	RTAS/AAX	Audio Units	VST
Quad	L R Ls Rs	L R Ls Rs	L R Ls Rs
5.0	L C R Ls Rs	L R Ls Rs C	L R C LS Rs
5.1	L C R Ls Rs LFE	L R C LFE Ls Rs	L R C LFE Ls Rs
7.1 (8.0)	L Lc C Rc R Ls Rs LFE	L R C LFE Ls Rs Lc Rc	L R C Ls Rs Cs Lc Rc

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

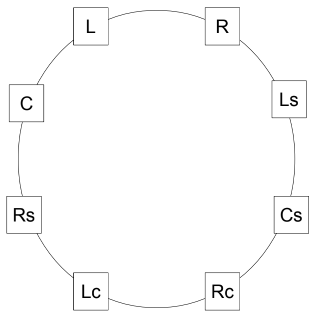
RTAS/AAX (SDDS)



Audio Unit

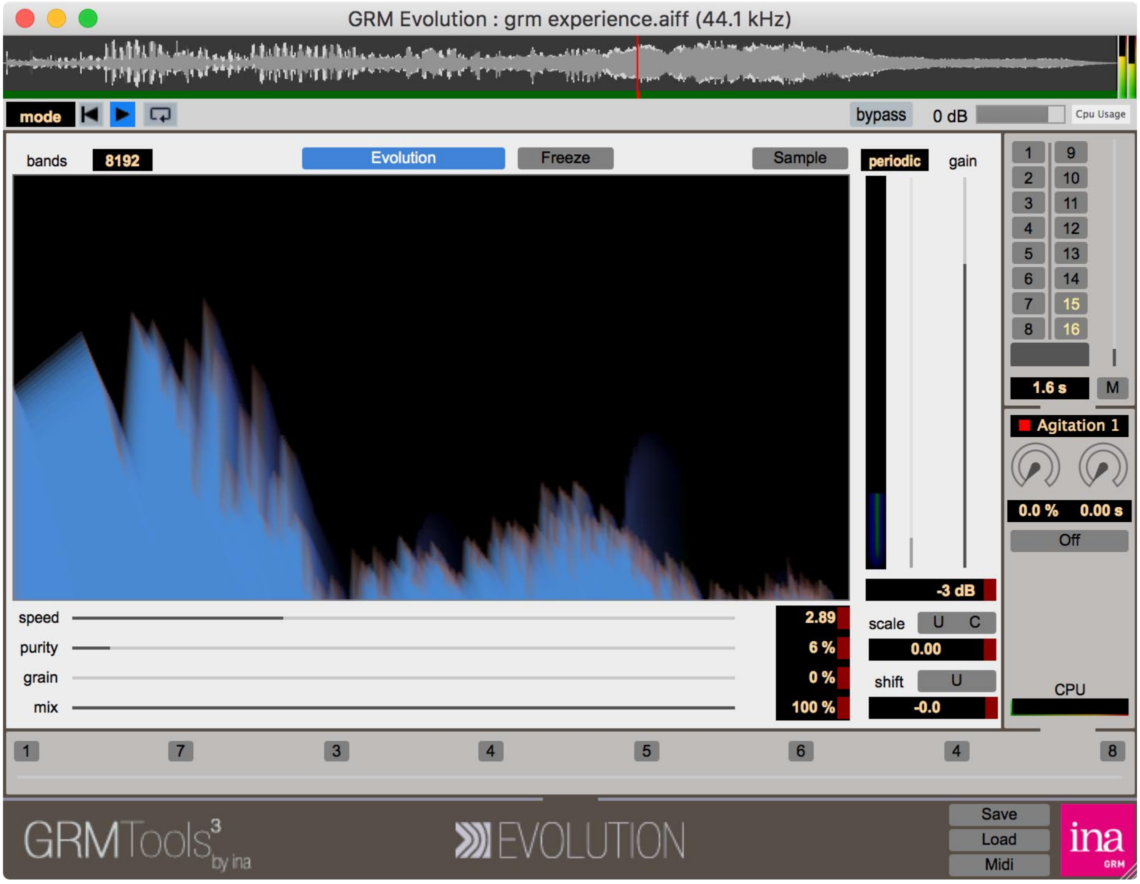


VST



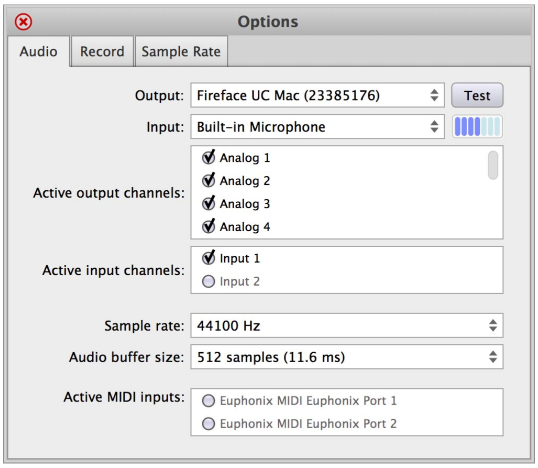
Stand Alone

The following descriptions only concern the stand-alone versions.



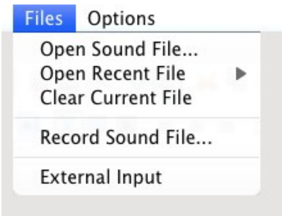
In this version, the processing window includes at the top a strip to control the reading and writing of sound files.

Multichannel processing is available for Fusion. 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

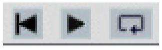
Select **Open Sound File...** or **Open Recent Files** in the **Files** menu.

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:

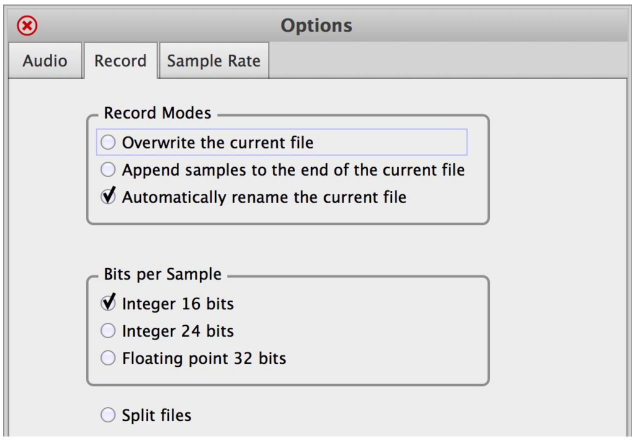
select **New Output Sound File...** in the **Files** menu.

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

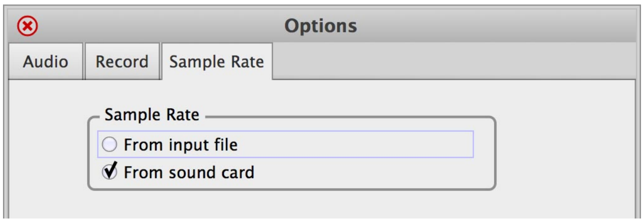
When an output file is open, the appearance of the advance button bar changes:

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.



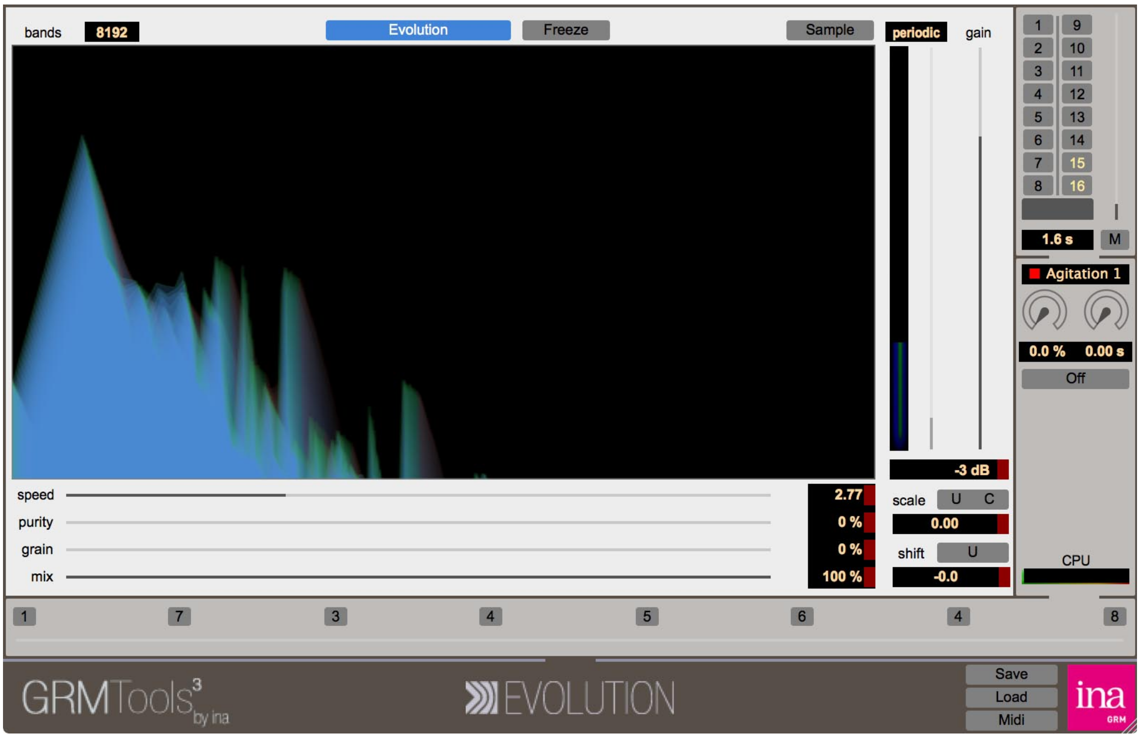
Sample Rate option allows to choose the sample rate when the file sample rate and the output sample rate are different:

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.

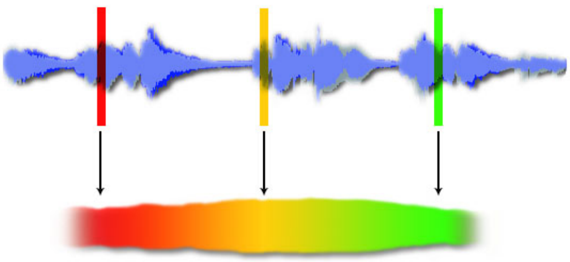
Evolution

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 ecord a sound into a mono or stereo track, and how to access the different windows and insert plug-ins. 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 eacht 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).

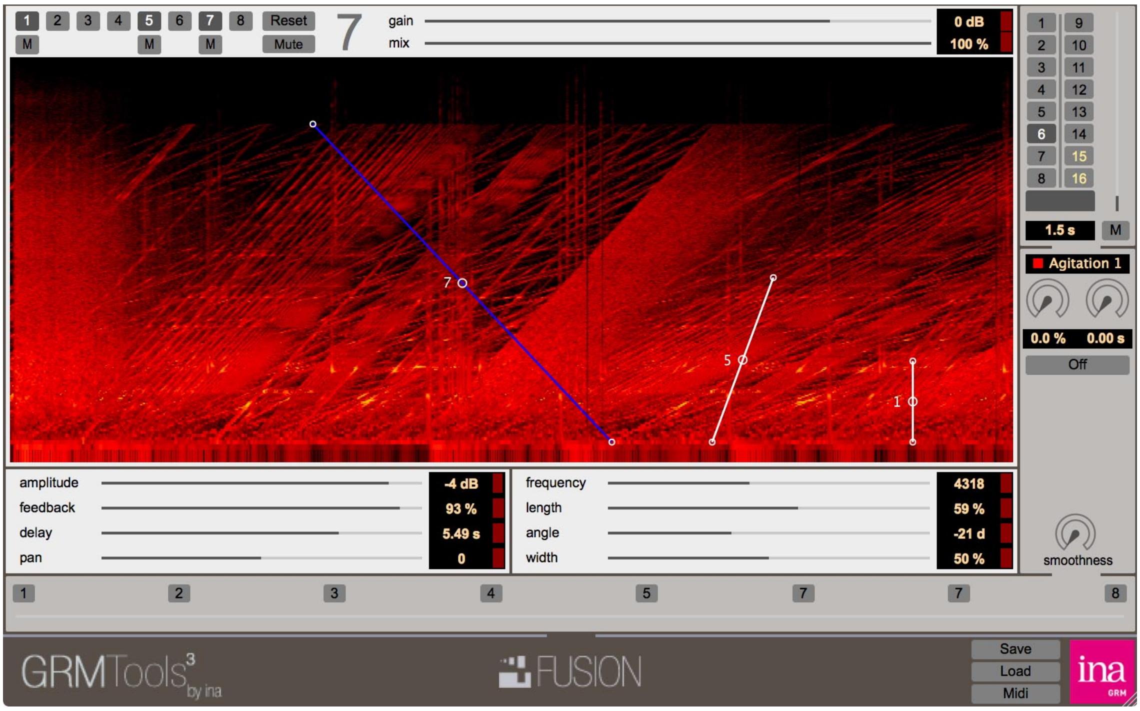
Shift

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).

Fusion

Fusion modifies a sound by playing on delays, filters offset and frequential sliding.



How does it work?

The input signal is represented by a sonogram which moves from right to left. 8 player heads of variable size and orientation can be placed on the sonogram. The players capture the signal running below them, like player heads placed along a magnetic tape or placed at different locations on a vinyl disc. They behave like band-pass filters and combine delay, filtering, reinjection and glissando effects.

Getting started

It is assumed that the user is familiar with the manipulation of the host applications, the import and the recording of a sound on a mono or stereo track, access to the different windows and the insertion of the plug-ins. Try the following examples on low density sounds, such as small percussion sounds.

Launch playing and click on the **Reset** button to have the default configuration. Enlarge the window to have a comfortable view of the sonogram.

> The sonogram starts running. To the right of the screen a small vertical blue bar symbolizes the first player. Its parameters are displayed under the sonogram.
>
> Manipulate **frequency** and **length** and note the filtering variations.
>
> Now manipulate **angle** and note the glissando effects.
>
> You can also click directly in the centre of the blue bar to modify **frequency** and **delay**, and on the ends to modify **length** and **angle**.
>
> Click on button **4** at the top, towards the left of the window. A new player appears.
>
> Click on the **M** button just below the **4** button. The blue bar symbolizing the fourth player becomes shaded and this player is no longer active. Click again on the **M** button, and the fourth player is again active.
>
> Adjust **agitation depth** to 40% and a**gitation speed** to 4 seconds (parameters to the right of the interface, underneath the presets). Validate agitation for **delay**. Note the delay variations.
>
> Validate agitations for the **frequency** and **angle** parameters.
>
> Play with **length** and **feedback**.

Reference

1 - 2 - 3 - 4 - 5 - 6 - 7 - 8

Activation and selection of players. If the player is not active, a click makes it appear and selects it: it appears in blue on the sonogram and its parameters are displayed. If it is already active, it is selected. Selection can also be carried out by clicking directly on one of the small circles on the bar. To deactivate a player, drag it outside the sonogram.

M

Situated under each activation button, these buttons control the **Mute** of each player. These controls are also available on the keyboard with keys 1 to 8.

Reset

Resetting of processing. Only the first player is active, with a delay of 1 second and a central frequency of 500 Hz.

Mute

Global mute for all players.

Amplitude

Output amplitude of selected player. This control acts after **feedback**.

Feedback

Percentage of output of selected player to be reinjected at input of processing.

Delay

Average delay of selected player. The displayed delay corresponds to the central position of the player bar on the sonogram. It is zero when the bar is completely to the right. It is maximum (10 seconds) when the bar is completely to the left.

Frequency

Central frequency of the selected player. The frequency displayed corresponds to the centre of the player bar on the sonogram.

Length

Length of bar representing the player. The maximum length corresponds to the diagonal of the sonogram display. In this case a glissando is obtained on all the frequencies of a duration equal to the maximum delay, that is 10 seconds.

Angle

Angle of the bar representing the player. At 0° the bar is vertical and the player behaves like a delay followed by a band-pass filter. For the other values, the various frequencies are no longer play simultaneously and a glissando effect appears.

Pan

Output panning. The panning effect depends on the number of output channels. For example, in octophony, the pan goes from the rear left hand side to the rear right hand side, describing a circle passing through the 8 outputs. See the **Multichannel** chapter at the start of the manual for more information.

Width

Width of stereophonic image. Active only with stereophonic inputs.

Smoothness

Shape of the band-pass filter. At 0%, the filter is rectangular, with very steep transitions. At 100% the filter is bell-shaped, with much smoother transitions. The same value is applied to all players.

Mix

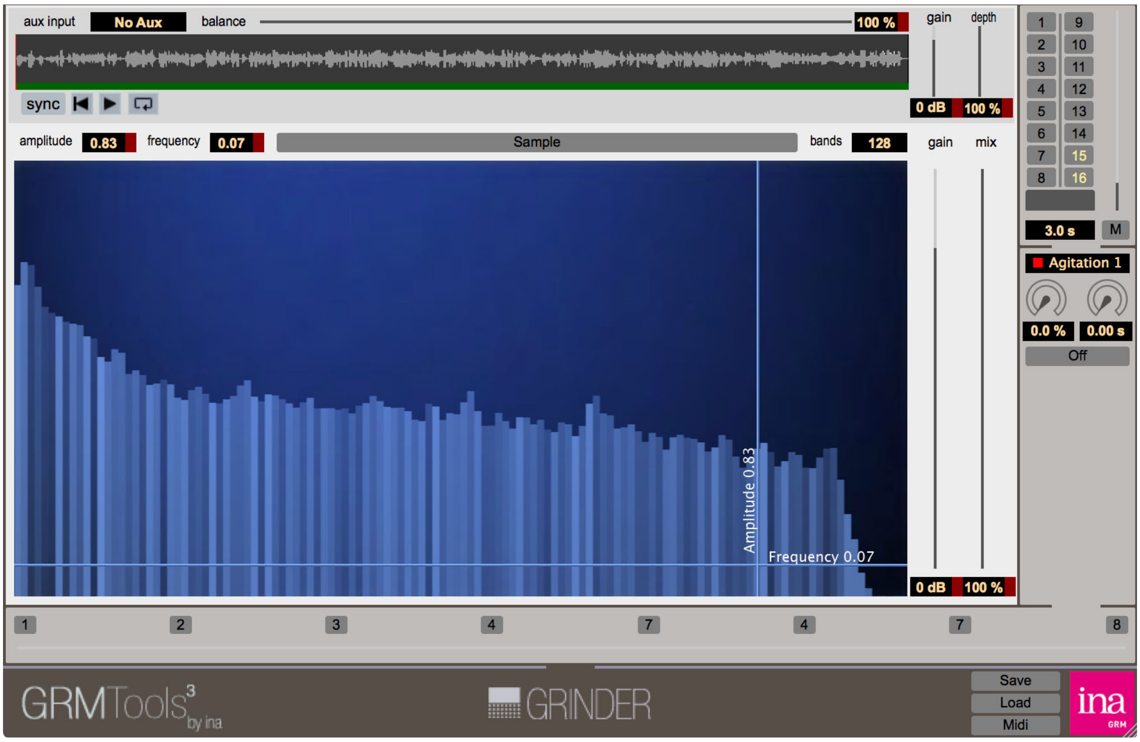
This parameter controls balance between the original sound and the processed sound. At 100% only the processed sound is hear. At 0% only the original sound is heard.

Gain

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

Grinder

Grinder is used to degrade a sound by freezing its frequential or time changes, and by modifying its spectral resolution. It also enables interesting vocoder effects.



How does it work?

The input signal is transformed into a sum of sine components whose quantity, and change in frequency and amplitude can be controlled. By adjusting the amplitudes, effects of signal distortion, saturation and degradation can be obtained, particularly for small numbers of components. By adjusting the changes of frequencies, it is possible to obtain vocoder effects. An integrated file player enables the use of another signal as a frequency or amplitude signal.

Getting started

It is assumed that that the user is familiar with the manipulation of the host applications, the import and recording of a sound on a mono or stereo track, access to the different windows and the insertion of the plug-ins. Apply the following examples on orchestral sounds with a vocal part, or sounds with large bandwidth.

Start playing, keeping the default values. Check that the **bands** menu is positioned at 64 and that the **amplitude** and **frequency** parameters are at 0.

Move the central cursor horizontally to increase **amplitude**. The change in amplitudes slows until it freezes completely at the value **1** and produces a saturation effect. The spectral envelope remains constant while the frequencies continue to change.

Click on **Sample**. A new amplitude profile is sampled.

Modify the band parameter and **note** the different types of sound degradation obtained.

Position **band** on 256 and **amplitude** on 0.5. Increase **frequency** up to the value 1. The frequencies remain constant while the amplitudes change, giving a sound similar to the vocoder.

Click on **Sample** several times. Modify the band parameter and **note** the different types of sound degradation obtained.

Position **band** on 128, **frequency** on 0.0 and **amplitude** on 0.7.

Click on **load file** (on the **aux input** line) and choose an audio file containing a speaking voice. The wave form is displayed. Click on the **start** button situated under the wave form display. Note the vocoder effect thus obtained.

Manipulate the **frequency**, **amplitude** and **band** parameters and note the different degradations and saturations.

Reference

Amplitude

Controls the change in signal amplitudes. At **0**, the amplitude variations are those of the original signal. At **1**, the amplitudes are frozen, and there is no time change in the amplitudes. The S**ample** button is used to capture the instantaneous amplitudes of the original sound. The color of the sound obtained is heavily dependent on the value of the **bands** parameter.

Frequency

Controls the change in signal frequencies. At **0**, the frequency variations are those of the original signal. At **1**, the frequencies are frozen, and there are no more frequency changes. The S**ample** button is used to capture instantaneous frequencies of the original sound. The color of the sound obtained is heavily dependent on the value of the **bands** parameter.

Sample

This button samples the instantaneous values of the frequencies and amplitudes of the signal input in processing. The effect is only perceptible for large values (very close to 1) for the **amplitude** or **frequency** parameters.

Bands

Number of sine components used to represent sound (from 2 to 4096). The smaller this number, the more the sound will be degraded. This parameter has an enormous impact on the color and quality of the sound for large values for the **amplitude** and **frequency** parameters.

Gain

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

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.

Aux input

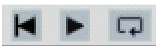
No Aux. No auxiliary input.

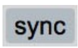
File. Using the file as auxiliary input. The file must have been previously loaded by Load File.

Side Chain. Using the sidechain input if available in the environment.

Load File. Opens the file selector.

The file is loaded with a selection equal to its total duration. 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 buttons  control the progress of the play cursor. The first one sends the cursor to the start of the selection, the second starts play or pauses it, and the last one loops playing on the selection.

The 

button synchronize playback on the host application playback.

Balance

Controls the frequency and amplitude proportions coming from the audio input and the auxiliary input.

0%: frequency=auxiliary input, amplitude=audio input

100%: frequency=audio input, amplitude=auxiliary input

Gain

Amplitude of auxiliary input.

Depth

Depth of modulation by auxiliary input. At 100% the modulation is maximum. At 0%, there is no further modulation (same as **aux input Off**)

Credits

Design, development
Emmanuel Favreau

Interface design
Emmanuel Favreau, Suzy Vincens, NO PUB

Documentation
Emmanuel Favreau

Translation
Peter Thomas

Tests
Diego Losa
Jean-François Minjard
Emmanuel Richier

Powered by JUCE, www.juce.com

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