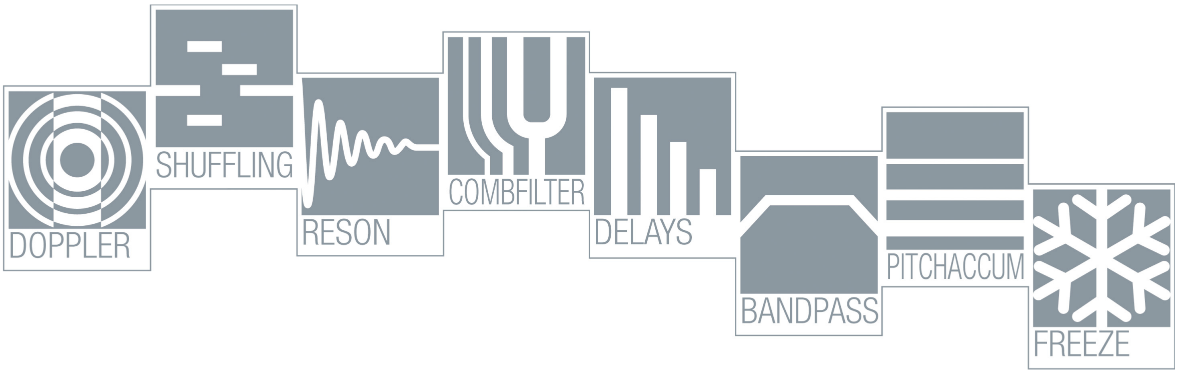
GRM Tools Classic

User's Guide



version 3.7

Introduction

GRM Tools Classic is a bundle of eight 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

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 Classic 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 Classic Installer.exe** icon and follow the instructions.

Authorization

Buy GRM Tools Classic

When you make your purchase, you will receive a serial number and a link to the 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 Classic

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 and 2DControllers

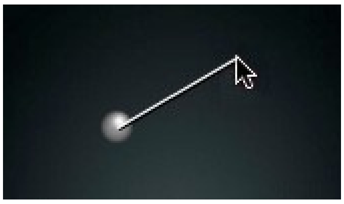
To change the value continuously, drag the handle to the left or right.

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.

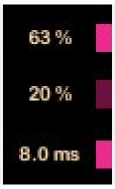
Only for 2DControllers, a click on the handle while pressing the [Shift] key limits movements to vertical or horizontal displacements.

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:

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

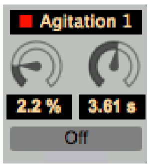
The horizontal SuperSlider and its associated Numerical Value Fields, located at the bottom of each window, provide a powerful control for interpolating between Presets. Use the SuperSlider to interpolate between any sequence of Presets to find new configurations and create new Presets.

- 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

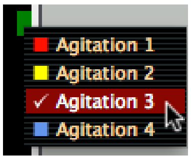
This set of controllers enables the adding of random variations to the processing parameters. The left hand rotating potentiometer gives the amplitude (from 0% to 100%) of the random variation. The right hand rotating potentiometer gives the speed of the variations (from 0 to 60 s). The On/Off button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

Four agitation groups are available:



A left click on the coloured button 

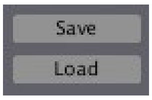
to the right of each alphanumeric value validates agitation for each individual parameter. A right click on the button opens the agitation group selection menu.



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:

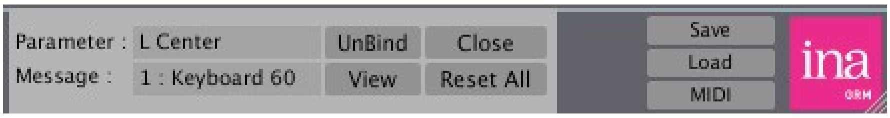
- Control Change
- Note On
- Pitch Wheel
- Program Change
-

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

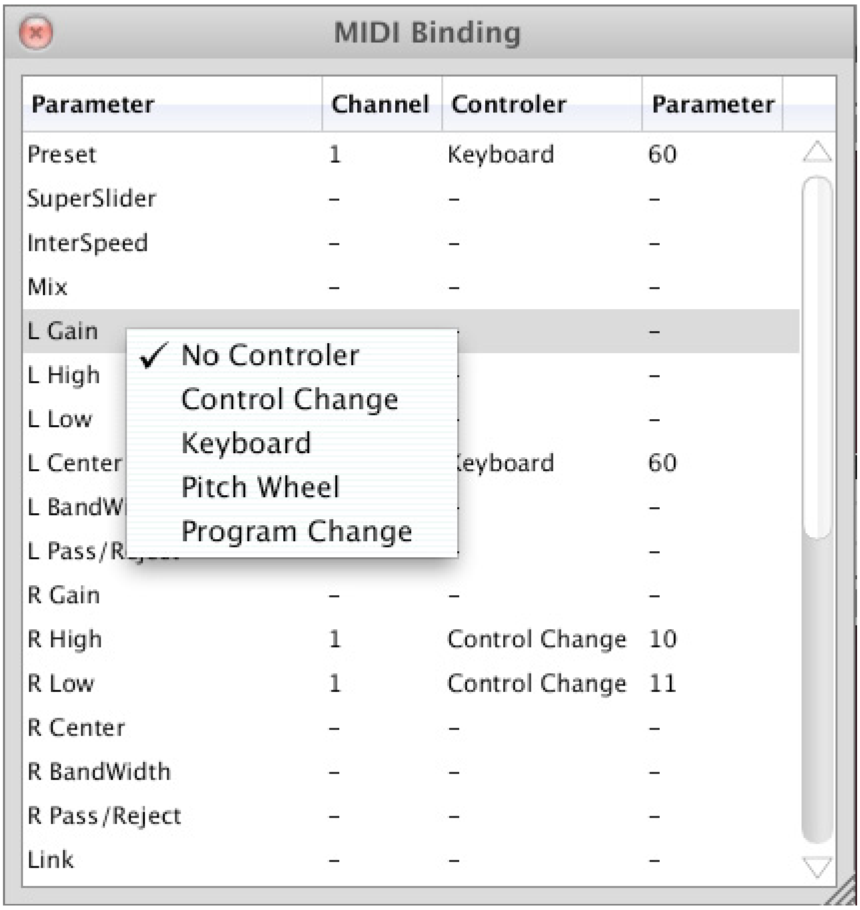


Click on the MIDI button located next to the ina-grm logo, and underneath the **Save** and **Load** buttons. A panel opens to the left of the button. Manipulate the parameter you want to control on the plug-in interface. Its name is displayed on the **Parameter** line. Then send the corresponding Midi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

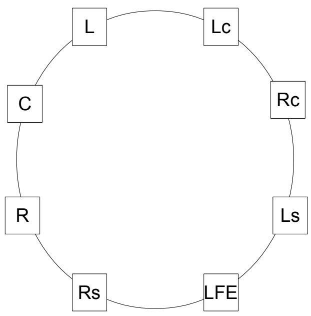
Delays, Doppler, Reson and Shuffling propose outputs on several channels in AAX, RTAS, VST and Audio Units.

The channel allocations follow the standard configurations of each system:

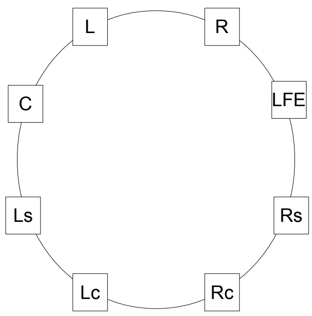
Configuration	RTAS/AAX	Audio Units	VST
Quad	L R Ls Rs	L R Ls Rs	L R Ls Rs
5.0	L C R Ls Rs	L R Ls Rs C	L R C LS Rs
5.1	L C R Ls Rs LFE	L R C LFE Ls Rs	L R C LFE Ls Rs
7.1 (8.0)	L Lc C Rc R Ls Rs LFE	L R C LFE Ls Rs Lc Rc	L R C Ls Rs Cs Lc Rc

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

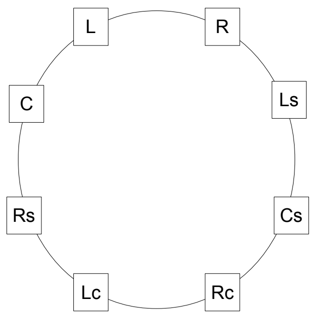
RTAS/AAX (SDDS)



Audio Unit



VST

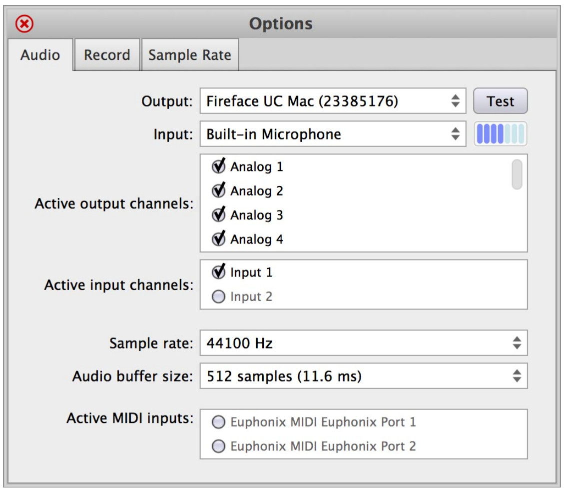


Stand Alone

The following descriptions only concern the stand-alone versions.

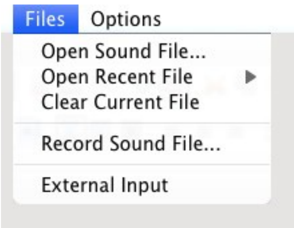
In this version, the processing window includes at the top a strip to control the reading and writing of sound files.

Multichannel processing is available for 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

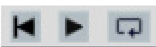
Select **Open Sound File...** or **Open Recent Files** in the **Files** menu.

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:

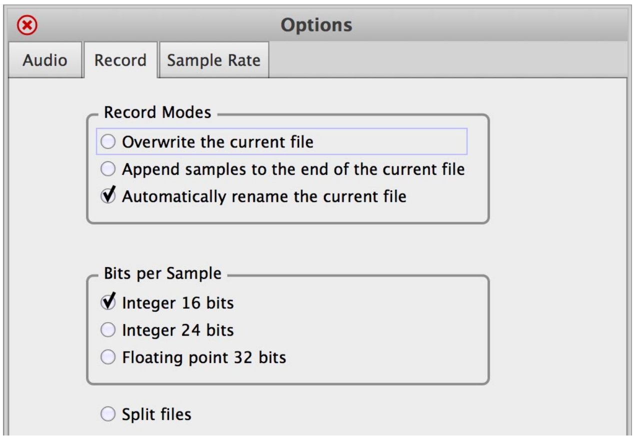
select **New Output Sound File...** in the **Files** menu.

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

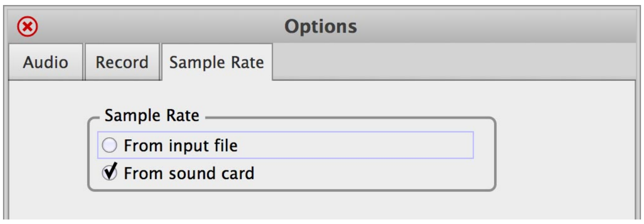
When an output file is open, the appearance of the advance button bar changes:

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.



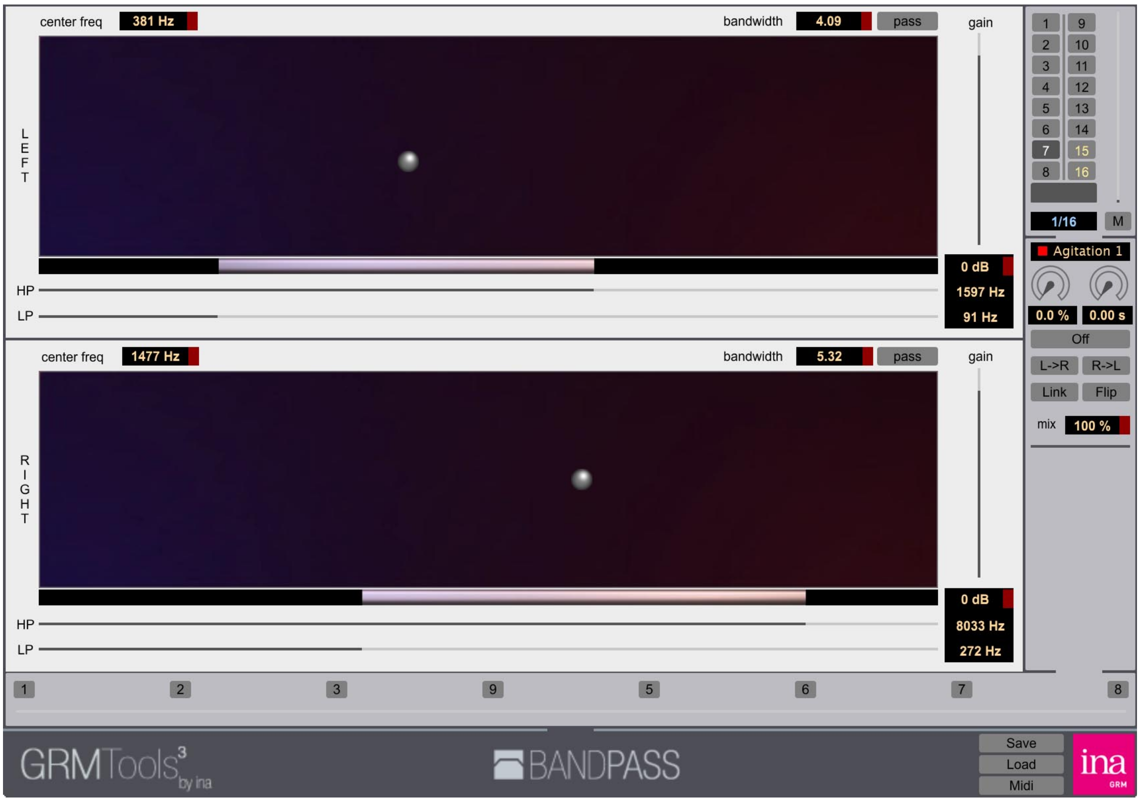
Sample Rate option allows to choose the sample rate when the file sample rate and the output sample rate are different:

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.

Band Pass

This plug-in gives you a dynamically controllable electronic "chisel" to carve your way through the highs and lows of a sound. Create dramatic effects as you focus on the high or low frequencies in a sound, simulate analog subtractive synthesis techniques ...



How does it work?

The 2DController, which is the ball in the center of the control window, functions like a handle to let you control lowpass and highpass filters together as a variable bandpass or bandreject filter. Left to right movement changes the center fequency. Top to bottom movement changes bandwidth. The bandwidth control is particularly effective because the filters have exceptionally steep slopes, creating sharp cutoffs and dramatic effects.

This plug-in will appear in a mono or stereo configuration depending upon whether the track is a mono or stereo track. In the stereo version, the two channels can be controlled independently or together. The screen shot shown above is the mono version.

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 song or a wide bandwidth sound.

Grab the 2DController with your mouse and drag it through the space. The effect will be dramatic and clear.

Use the Elastic String: Hold down the [Command] key on Mac or [Ctrl] key on PC while you grab the 2DController with your mouse and drag it through the space. Note: You're pulling the 2DController with the Elastic String. The longer the string, the slower the motion. The Elastic String smooths out your actions.

Create a stereo track and use the stereo version of Band Pass. Note the **link** button (only in the stereo version). Unlink the left and right channels, then switch one of them to **bandreject** mode, then link them again. One channel will be bandpass and the other will be bandreject.

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

center freq

Sets the center frequency of the filtered band to any frequency from 23 Hz to 22 kHz. This is both a display of the horizontal position of the **2DController** and a numerical value field that you can change by dragging the number up or down with the mouse.

bandwidth

Sets the width of the filtered band. The value is given as a percentage of deviation from the center frequency. For example, for a center frequency of 1000 Hz and a bandwidth of 2, the values for lowpass and highpass will be respectively at the nearest to 500 Hz (1000 / 2) and 2000 Hz (1000 x 2).

pass/reject

Selects the filtering mode. Clicking on the button sets it to the opposite state.

- In bandpass mode, the frequencies between the lowpass cutoff and the highpass cutoff are passed through the filter.
- In bandreject mode, the frequencies between the lowpass cutoff and the highpass cutoff are rejected.

gain

Sets the level of the input signal from -96 dB to + 12 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.

HP

The slider changes the highpass cutoff frequency in 512 steps between 23 Hz and 22050 Hz.

LP

The slider controls the lowpass cutoff frequency in 512 steps between 23 Hz and 22050 Hz.

2D Controller

Moving the 2DController vertically lets you control the filter's **bandwidth**. Move the 2DController horizontally lets you control the filter's **center frequency**.

tempo

The interpolation time between **presets** can be synchronized to tempo. To initiate synchronization, hold down the [Shift] key and click on the Numerical Value Field under the Presets.

Additional controls for the stereo version :

L -> R

This button copies the settings from the left track to the right track.

R -> L

This button copies the settings from the right track to the left track.

link

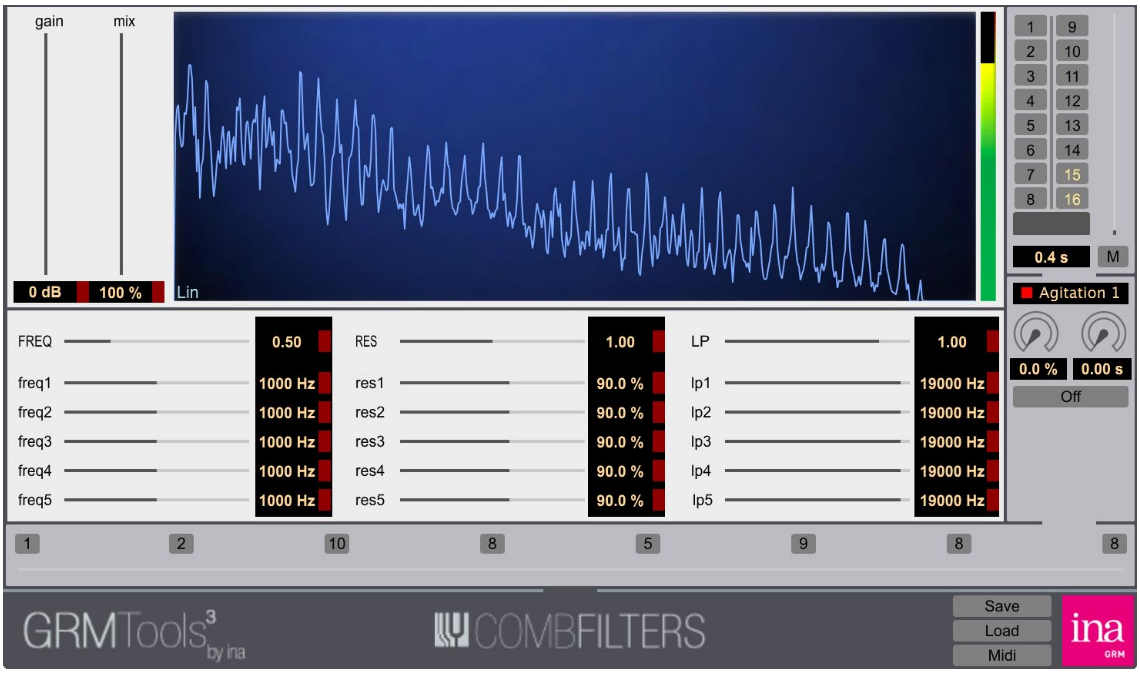
This button links the settings of both tracks. As a setting is changed in one track, the same change will occur in the other.

flip

This button exchanges the settings from one track to the other.

Comb Filters

This plug-in lets you add power and resonance to a sound, transform a sound's timbre, smooth rhythmic sounds into long continuous sounds ...



How does it work?

A comb filter, in general, is a filter that resonates a selected frequency and all of the harmonics of that frequency. This plug-in gives you five high-Q in-parallel comb filters, controllable independently or in ensemble, plus five low-pass filters to control the highfrequency content of the output.

The **freq** sliders let you control the frequencies of each of the filters. The **res** sliders let you control the amount of resonance of each of the filters. The **lp** sliders let you control the cutoff frequencies of the lowpass filters, allowing you to control high-frequency content. The **FREQ**, **RES**, and **LP** sliders are the master sliders for the filters.

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 diferent windows and insert plug-ins. For this tutorial, use a hand-drum or other rhythmic sound.

Click on preset 6. Note that the filter frequencies are set close together to create a dense resonant sound. Start the playback.

Move the **mix** slider slowly downward and upward. You'll hear the sound change until you're hearing only the processed sound.

Set the **RES** slider at about .90 and the **LP** slider at 1.00. Then play with the **freq** controls and note how they affect the sound. Click on different **Presets**. Move the **FREQ** slider to the left and right. For a special effect, move it all the way to the left, then slowly to the right.

Move the **RES** slider all the way to the left and then all the way to the right.

Move the **LP** slider to the left and right. Note: Changes in resonance. Transformations in the sound.

Here's a potentially dangerous effect: Create a self-sustaining resonance that will continue even after you stop the input sound by moving the **RES** and **LP** sliders all the way to the right. To stop the resonance, move the **LP** slider to the left.

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 U**niversal Controls**.

Reference

gain

Sets the level of the input signal from -96 dB to 0 dB.

mix

Determines the relative strengths of the processed and original signals. 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.

FREQ

Multiplies the resonant frequencies of all of the comb filters by a number between 0 and 2. If you multiply by 2, for example, the resonant frequencies of the five filters are transposed upwards one octave. If you multiply by .5, the resonant frequencies of the five filters are transposed downwards one octave.

RES

Multiplies the resonance duration of all comb filters by a number between 0.8 and 1.2.

LP

Multiplies the cutoff frequencies of all lowpass filters by a number between 0 and 1.2.

freq1 - freq5

Each control sets the frequency of one of the five comb filters from 45 Hz to 20 kHz. Values very close to each other (for example with a gap of 1 Hz) will create beat or phasing effects. Some values enable the obtaining of chords which resonate with the sound processed.

res1 - res5

Each control sets the resonance of one of the five filters from 0 (no resonance) to 100 (the longest possible resonance). Note that the actual duration depends also upon other parameters, such as the lowpass filter.

lp1 - lp5

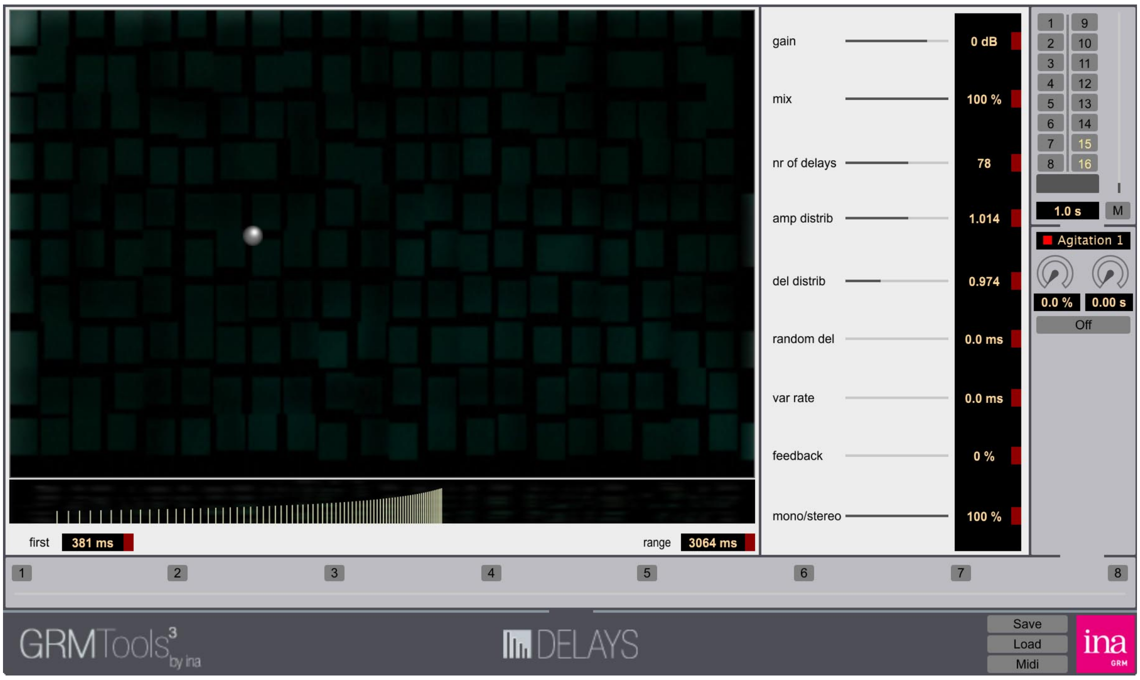
Each control sets the cutoff frequency of one of the lowpass, enabling the adjustment of the quantity of harmonics present in the resonances, and thus the richness of the resulting sound. High values will cause slight filtering of the signal, giving a rich and metallic sound quality, while low values will smooth the sound by cutting off the highest resonances.

tempo

The interpolation time between presets can be synchronized to tempo. To initiate synchronization, hold down the [Shift] key and click on the Numerical Value Field under the Presets.

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 elemnents 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

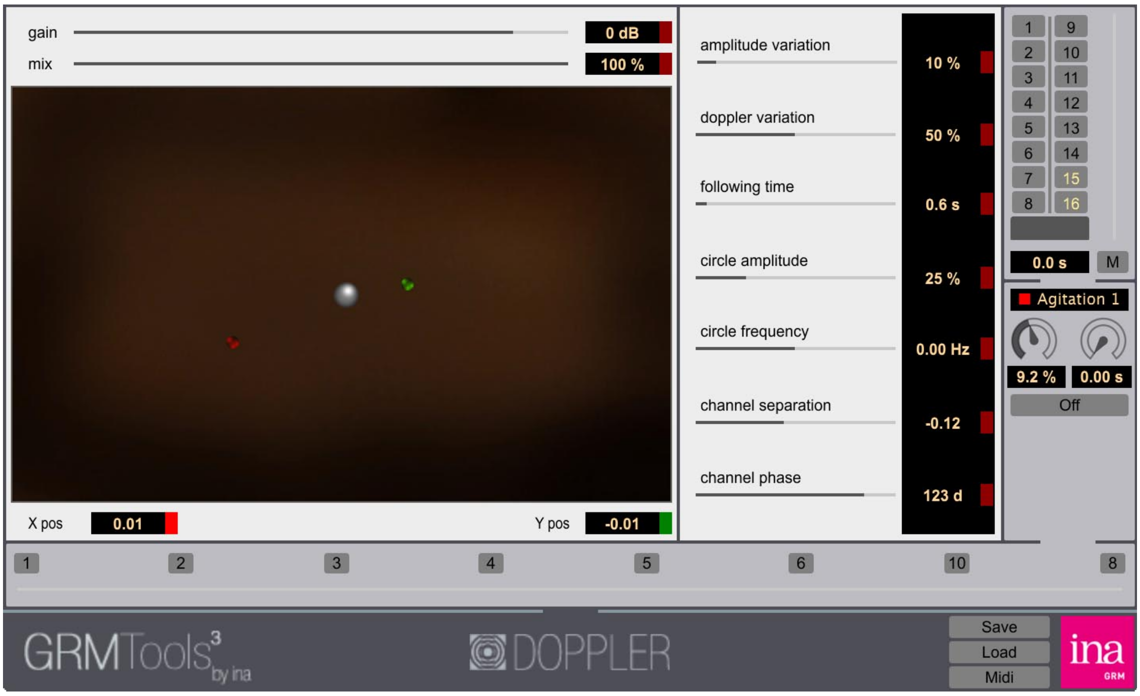
The vertical dimension controls the range of the delays in time. The horizontal direction controls the center of the range. Note that the window under the control window contains a graphic display of the values for **first**, **range**, **nr of delays**, **amp distrib**, **del distrib**, **random del**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

Doppler

This plug-in lets you move sounds through an audio space with changes in pitch that correspond to their movement. You can also transform sounds through loudspeaker modulation, create unusual vibrato effects ...



How does it work?

Doppler simulates the effect obtained by moving sound through a two-dimensional space. A sound moving closer to the hearer (placed in the centre of the two-dimensional slider) will be transposed upwards, while a sound moving away from the hearer will be transposed downwards. Movements follow the displacement of the two-dimensional cursor, to which may be added an automatic circular movement. Based on a physical propagation model, this processing also enables the obtaining of non-realistic results by the decoupling of variations in amplitude and in the Doppler effect.

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 song or a melodic sound.

Click on preset 1. Start the playback.

Move the **circle amplitude** slider slightly to the right. Notice the balls start to move. You'll hear changes in pitch.

Move the **circle frequency** slider to the right. Notice that the balls start to move faster.

Move the **channel separation** slider to the right. Notice that the balls are swinging further out from the center. Now move the slider towards the left to about -0.85, so that the balls are both swinging around the center. Now move the **circle amplitude** slider to the right. The balls are now swinging around the center in a larger circle.

Move the **channel phase** slider to the right and the left. Notice that when the slider is positioned in the center, the balls swing together, but when it is to the left or right, the balls follow one another.

Now experiment with the **amplitude variation** slider but note that the actual level of loudness variation will depend upon the settings of **circle amplitude** and **circle frequency.**

Now experiment with the **doppler variation** slider but note that the actual pitch variation will depend upon the settings of **circle amplitude** and **circle frequency.**

Note that when the value of **circle frequency** is a positive number, the balls swing in one direction and when it is a negative number, the balls swing in the opposite direction.

Move it in either direction to a higher number, for example to 27 hz, to achieve a novel type of spatial modulation.

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 +12 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.

X pos

Horizontal position. -1 indicates that the sound is at the left. 0 indicates that the sound is in the center. +1 indicates that the sound is at the right.

Y pos

Vertical position. -1 indicates that the sound is at the bottom. 0 indicates that the sound is in the center. +1 indicates that the sound is at the top.

amplitude variation

Determines the amount of intensity variation relative to the distance of the position of the sound relative to the center of the 2-dimensional potentiometer. At 0%, no variation occurs. At 100%, maximum variation occurs.

doppler variation

Determines the amount of pitch variation relative to the speed with which the sound is moving. At 0%, no variation occurs. At 100%, maximum variation occurs.

following time

Determines the time from 0 to 10 seconds that the sound, represented in mono as the little red ball or in stereo as the little red and green balls, will take to reach a position as represented by the grey ball.

circle amplitude

The amplitude of the circular movement of the sound around the grey ball. In the stereo version, two circular movements are represented for each channel.

circle frequency

Frequency of rotation of a sound around the grey ball. Positive values from 0 to 100 Hz represent motion in a clockwise direction. Negative values from 0 to -100 Hz represent motion in a counterclockwise direction.

2DController

Lets you control simultaneously the parameters X pos and Y pos. The effective position of the sound is displayed by a little red ball. In the stereo version, a little red ball and a little green ball represent the two channels.

tempo

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

Additional controls for the stereo version :

channel separation

This control determines the distance between the two channels.

channel phase

This control determines the phasing of the rotations of the sounds. At 0, the rotations are synchronous. At 180 or -180, the rotations are out of sync by half a circle.

Freeze

This plug-in lets you freeze a fragment of a sound and scrub through the fragment with loops of different sizes at different pitches ...

How does it work?

Freeze is an algorithm for time freezing by looping into a 12-second zone. The read point, the time, the number of simultaneous loops (up to 128) and their synchronization are variable, as are read speeds. Freeze can be used to create rhythmic motifs, phasing effects, temporary time lags, and many other effects.

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 rhythmic drum track.

Start the playback.

Position the **2DController** in the center close to the bottom. Wait until you hear a particularly interesting segment of your sound, then click on the **freeze** bar. Then drag the control ball slowly through the control window. As you move it left and right, you'll notice a scrubbing effect. As you move it up and down, you'll notice that you're sampling a larger or shorter du ration of the frozen sound.

Click on the **reset Ph** and **random Ph** buttons and note the coordination of the loops. Move the **pitch** slider to the left to -15. Note that sounds are lower and longer. Move the **pitch offset** slider to the right to 100% to note how the pitch changes. Then move the **random pitch** and **random dur** sliders. Then move the **pitch** slider to the right and note the differences in the sound.

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 U**niversal Controls**.

Référence

gain

Sets the level of the input signal from -96 dB to 0 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.

begin

Specifies the start time of the loops within the 12-second field.

end

Specifies the end time of the loops within the 12-second field.

center

Specifies the center of the loops within the 12-second field. Changes **begin** and **end.**

width

Specifies the width of the loops within the 12-second field. **Changes begin and end.**

freeze

Triggers the capture of the three last seconds of the input sound and begins the loop replay. The display shows the portion of sound that is saved and the start and end points of the loops. Click again on this button to resume listening to the input signal.

Reset phase

Synchronizes all loops to start together. Enables interesting percussion, phasing and harmonic beat effects with very low pitch offset parameter values.

random phase

Unsynchronizes the playback of the loops.Gives an ensemble or nappe effect if the nr of loops parameter has a high value.

pitch

Controls the pitch of all loops from -48 semitones to +24 semitones. Note that as the pitch changes, the duration of the loops will change accordingly.

pitch offset

Controls the pitch variation from one loop to the next. Note that as the pitch changes, the duration of the loop will change accordingly. To achieve phasing effects, set the interval of variation to very small values and all random controls to 0.

random pitch

Lets you specify a random pitch variation for each loop. A new value is chosen randomly at the beginning of each loop replay and then remains constant during the time of the loop.

random dur

Lets you vary the duration of each loop randomly from 0 to 400 ms.

nr of loops

Specifies the number of loops from 2 to 128. Note that the maximum effective number of loops possible depends on the power of the processor and on the number of plug-ins in use.

mono / stereo

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

2DController

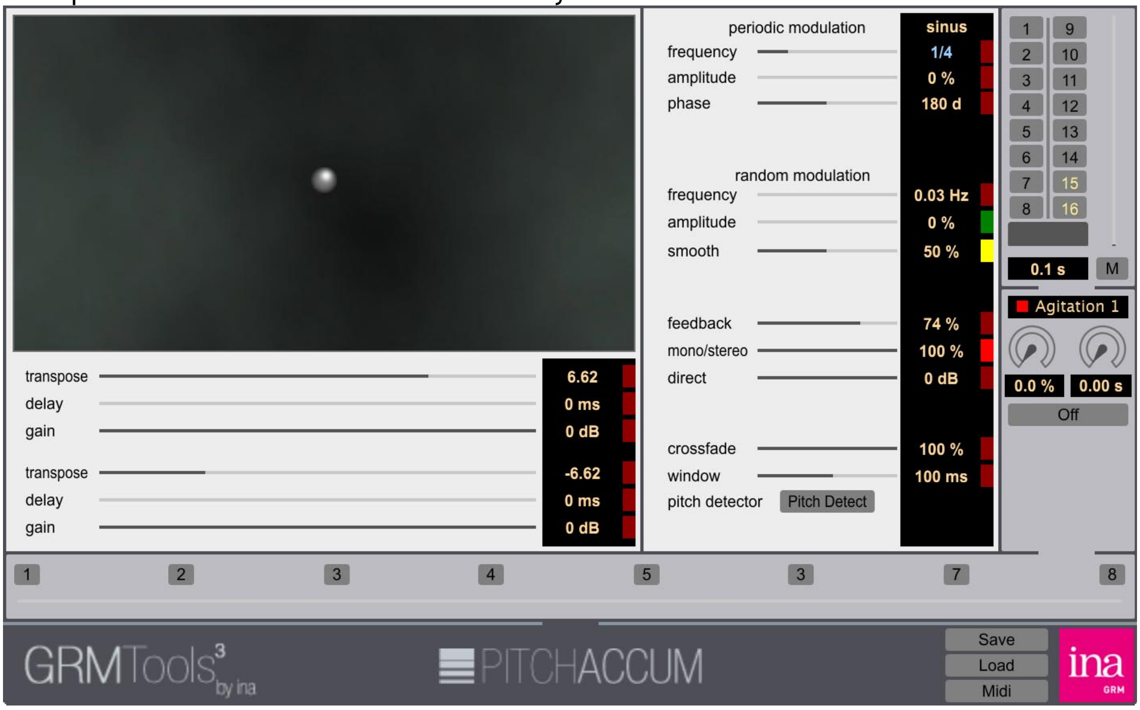
Moving the 2DController vertically lets you simultaneously control the start and end times of the loops within the 12-second field. Moving the 2DController horizontally lets you scrub through the 12-second field.

tempo

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

Pitch Accum

This plug-in lets you create two distinct 'shadows' of a sound at different transposition levels and at different delay intervals.



How does it work?

Two independent and simultaneous transposers are used to transpose the pitch of an input sound. A pitch follower detects the pitch of the input sound and optimizes the transposition. The transposed sounds are then transformed with delays and modulations.

In the mono version, the same input signal is sent to both transposers. In the stereo version, the left channel of the input signal is sent to one transposer and the right channel is sent to the other.

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 melody.

Click on preset 1. Start the playback.

Adjust the first transposer to a positive number. Note: The transposition is higher than the original.

Adjust the second transposer to a negative number. Note: The transposition is lower than the original.

Set the delays at different lengths.

Change the settings for **periodic modulation**, **random modulation**, and **feedback**.

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

transpose

Sets the interval for each transposition from -24 semitones to +24 semitones.

delay

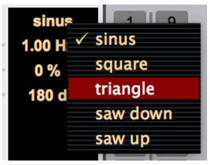
Sets the delay of each transposition from 0 to 2972 ms.

gain

Sets the level of the input signal from -40 dB to 0 dB.

periodic modulation

A click on the modulation name allows to select the modulation waveform.



frequence. Sets the modulation frequency from 0 to 20 Hz.
amplitude. Sets the modulation amplitude from 0 to 100 %.
phase. Synchronizes the modulations from 0 to 360 degrees.

random modulation

Randomly modulates the pitch set in **transpose** :
frequence. Sets the modulation frequency from 0 to 20 Hz.
amplitude. Sets the modulation amplitude from 0 to 100 %.
smooth. Smoothes the random variations from 0 (abrupt changes) to 100 (slow variations).

feedback

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

mono/stereo

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

direct

Determines the level of the non-processed input signal from -40 dB to 0 dB.

window

Sets the durations of the sound fragments used by the transposition algorithm. For periodic sounds or sounds with quick changes, a value between 10 - 20 ms is recommended. For complex sounds with slow variation, a longer value may prove to be preferable.

crossfade

Controls the way that fragments overlap from small values and abrupt transitions to longer values and softer transitions.

pitch detector

Ensures that the du rations of the sound fragments remain consistent with the frequency of the input signal. It is advisable to use this with periodic signals with small values of the **window** parameter (10-20 ms).

2DController

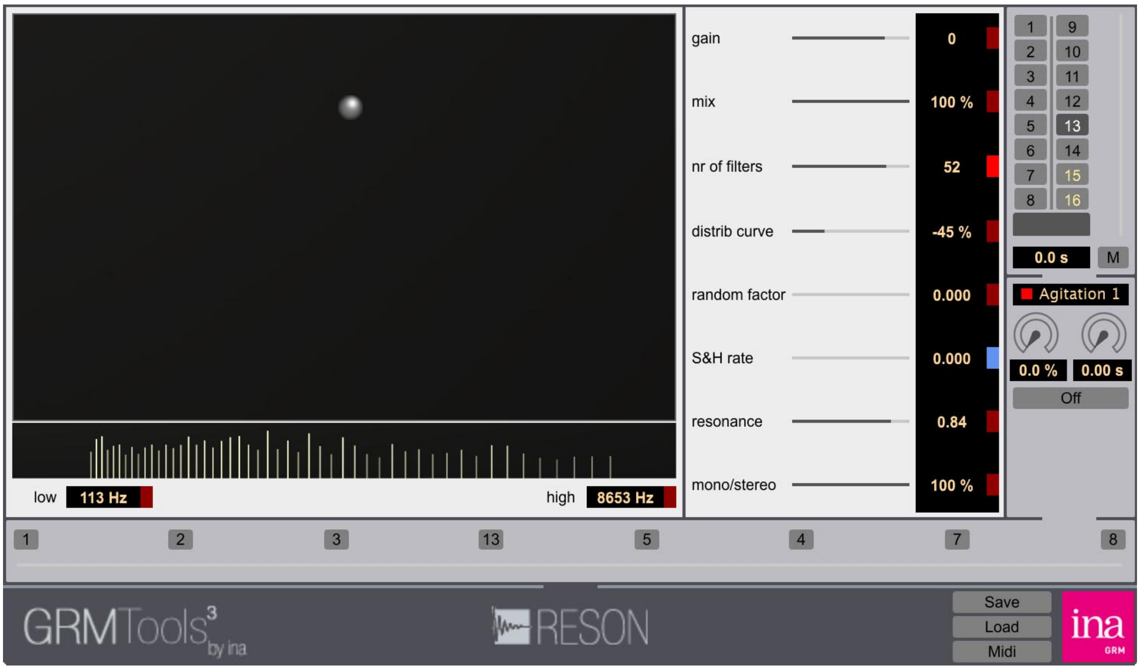
Moving the 2DController vertically lets you control the range of transposition levels between the two fragments. Moving the 2DController horizontally lets you control the transposition levels of both fragments in parallel.

tempo

The parameters that can be synchronized to tempo are the interpolation time between presets, **delay** (both voices), **frequence** (periodic modulation), and **frequence** (random modulation). To initiate synchronization, hold down the [Shift] key and click on the Slider or Numerical Value Field associated with the parameter

Reson

This plug-in lets you create new sounds from current sounds by rebalancing and redistributing the sound's resonant frequencies.



How does it work?

Reson consists of a set of second order resonating filters. This type of filter amplifies a single frequency, giving very gentle resonances. Depending on processor power, it is possible to have up to 64 simultaneous filters, whose frequency, amplitude and resonance are controlled.

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 footsteps.

Click on preset 1. Start the playback.

When you hear the sound, gradually move the **resonance** slider to the right. The sounds will first take on a metallic quality, then change into bells.

Increase the value of the **nr of filters** and observe the **distrib curve** effect.

Move the **random facto**r slider and **S&H rate** sliders to the right. The sounds will seems like loose metal.

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 U**niversal Controls**.

Référence

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.

low

Sets the resonance of the lowest filter to a frequency between 59 Hz and 15 kHz. The group of filters defined by **nr of filters** will be distributed between **low** and **high** according to the **distrib curve**.

high

Sets the resonance of the highest filter to a frequency between 59 Hz and 15 kHz. The group of filters defined by **nr of filters** will be distributed between **low** and **high** according to the **distrib curve**.

nr of filters

Determines the number of filters ifrom 2 to 64. Note that the maximum number of filters depends on the power of the processor and the number of plug-ins in use.

distrib curve

Determines the distribution of filter frequencies between **low** and **high**. For a value of 0, the filters are distributed equally according to musical intervals. For example, if the interval between **low** and **high** is one octave, and the **nr of filters** is 12, the filters will be tuned in semitones. For a value of 100, the distribution is by equal frequency intervals, and you obtain a sound close to that of the comb filter.

random factor

Sets the extent of random variation in the distribution of filter frequencies between **low** and **high**. At '0', there is no variation. At '1', there is a variation between one octave below and one fifth above the unvaried filter frequency.

S&H rate

Sets the rate at which random variations occur in individual filter frequencies. At '0', there is no variation. At '1', there is a variation in a filter every millisecond.

resonance

Determines the duration of the resonance in the filters from '0' (no resonance) to '1' (infinite resonance). At maximum resonance, the filters behave like sine wave oscillators.

mono/stereo

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

2DController

Moving the 2DController vertically lets you control the frequency bandwidth within which the filters are distributed. Moving the 2DController horizontally lets you control the center frequency of this band within the spectrum. Note that the positions of the filter frequencies are displayed in the display window under the 2DController.

tempo

The interpolation time between presets can be synchronized to tempo. To initiate synchronization, hold down the [Shift] key and click on the Numerical Value Field under the Presets.

Shuffling

This plug-in lets you create an unusual resonance or reverberation, fill an audio space with overlapping fragments of a sound, turn a single voice into a crowd ...

How does it work?

The processing proposed here is based on a traditional practice of musique concrète, which consisted of cutting up fragments of magnetic tape and sticking them back together in a different order, so as to introduce local disruptions into the signal, while preserving its global continuity.

Shuffling randomly samples fragments of variable dimensions in the last three seconds of the sound processed. It is possible to modulate the density of restitution of the fragments and their pitch. Depending on the size and density chosen, Shuffling enables a wide range of sound transformations : from the most conventional (chorus, flanger, harmonizer) to the most unexpected.

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 solo voice or melody.

Click on preset 1. Start the playback.

When you hear the sound, gradually move the **delay** slider towards the right. You'll hear the fragmentation begin.

Modify the size of the fragments with the **fragment** slider.

Move the **feedback** slider to the right. You'll hear a reverb effect.

Play with the **pitch** controls and **density** slider.

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 0 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.

fragment

Sets the duration of the fragments from one millisecond to about 3 seconds.

envelope

Controls the attacks and releases of the fragments from sharp to gradual.

delay

Sets the longest possible delay time. In other words, each fragment will randomly occur at a delay that is less than this number of milliseconds. If you want the fragments to occur further apart, move this slider towards the right. If you want the fragments to be more mixed together, move this slider towards the left.

initial pitch

Determines the transposition at the beginning of each fragment. To get the classical harmonizer effect, hold [Shift] down and move the initial pitch potentiometer. The **final pitch** potentiometer will move simultaneously.

final pitch

Determines the transposition at the end of each fragment. To get the classical harmonizer effect, hold [Shift] down and move the final pitch potentiometer. The **initial pitch** potentiometer will move simultaneously.

random pitch

Lets you randomize the transposition of each fragment.

feedback

Specifies the percentage of the output signal which will be fed back into the input. This produces a reverb effect.

density

Lets you determine the number of fragments that will be heard. At 0%, no fragment will be heard. At 100%, the maximum number of fragments will be heard. If sidechain input is used, silence is replaced by this input.

mono/stereo

Sets the distribution of the fragments to the outputs, from monophonic at 0%, which means the same signal is directed to both tracks, to stereo at 100%, which means that different fragments are directed to different tracks.

2D Controller

Moving the 2DController vertically lets you control the size of the fragments. Moving the 2DController horizontally lets you control the **delay** slider, which determines the extent to which the fragments occur separately or overlapped.

tempo

The interpolation time between presets can be synchronized to tempo. To initiate synchronization, hold down the [Shift] key and click on the Numerical Value Field under the Presets

Credits

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Interface design
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Documentation
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Translation
Peter Thomas, Joël Chadabe

Tests
Diego Losa
Jean-François Minjard
Emmanuel Richier

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dans le domaine du son
et des musiques électroacoustiques