GRM TOOLS CLASSIC VST

User's Guide
Introduction

GRM Tools Classic VST is a bundle of eight plug-ins that provide superb tools for sound enhancement and design. 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 more than 20 years of research and development by composers and sound designers in sound transformation software.

Use these plug-ins with Cubase or any other version of a PC program that is compatible with VST plug-ins.

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.

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Installation

The installer normally puts the eight plug-ins in a GRM folder, which is in the VSTplugins folder in the host application folder in the Program Files folder, as shown in the following line using Steinberg’s Cubase as an example.

```
c:/ Program Files / Steinberg / Vstplugins / GRM
```

In certain circumstances, you may have to search for a VSTplugins folder. You also have the option of creating a folder for the plug-ins as indicated in the following line.

```
c:/GRM Tools VST
```

**Note:** Please read the plug-in installation notes in the operating manual for whatever host software you are using. You may need to manually copy the plug-ins into a folder designated by the host program.

To launch the installer, double-click on the **GRM Tools Classic VST installer icon**.

![GRM Tools Classic VST installer icon](image)

Click **Next** and follow the instructions in the windows. A ReadMe file and Software License will be displayed. The installer also installs the Interlok Extensions necessary for authorization of the software.

If you prefer to install the plug-ins, the documentation, or the Interlok Extensions individually, choose the **Select Components** option. Be sure to select **Interlok Extensions** if you do not already have them installed. See the following window:
For a standard installation, leave all the items checked, click **Next** and continue. At the following option, we recommend that you include an uninstaller. You will then be prompted to choose a destination for the GRM Tools documentation.

When the installation is done, the following window will be displayed:

Click on **Close** to quit the installer.
Authorization

Once GRM Tools is installed, you can use it in demonstration mode for up to 15 days. If you want to continue to use GRM Tools after the demonstration period, you'll need to buy it and authorize it.

Buy GRM Tools

Buy GRM Tools at any time within the 15-day demonstration period. To buy GRM Tools, do any of the following:

- buy the software at a store
- visit the GRM Tools website at www.grmtools.org
- go to www.cdemusic.org and type 'grm tools' into the search engine

When you buy GRM Tools, you'll receive a serial number. If you purchase the software in a store, the serial number will be in the package. If you acquire the software online or from a demo disk, we will email your serial number to you. Keep your serial number in a safe place! It is your proof of purchase, and it will be required when you need to obtain an authorization or receive technical support.

Authorize GRM Tools

Authorizations are done through iLok.com.

To authorize GRM Tools, you'll need an iLok (a USB-compatible dongle), a web browser that is supported by iLok.com, and an iLok.com account. Check our FAQ at www.grmtools.org for suggestions on how to buy an iLok if you don't already have one, how to keep your iLok secure and, if necessary, how to use a USB hub.

- If you do not have an iLok.com account, go to www.iLok.com and create an iLok.com account. Then insert the iLok you plan to use into a free USB port and register that iLok.

- Register your GRM Tools bundle at www.grmtools.org. When you have completed the form, we will deposit an ASSET into your iLok.com account and notify you.

- Go to iLok.com and transfer the ASSET to your iLok. Your iLok will then be authorized for the GRM Tools bundle that you purchased.

IMPORTANT! At this time, there is no way to remove or transfer an ASSET authorization
from an iLok. Please BE ABSOLUTELY SURE that you are purchasing the correct GRM Tools bundle.

Please note also that there are several authorization dialogs that can appear when you do not have an authorized iLok connected to a USB port. Some of these authorization dialogs will request a Response to a Challenge, and some dialogs request an iLok License Card. Neither the Challenge/Response, nor the License Card method are operational with GRM Tools. Please disregard any Challenge/Response or License Card requests.

Check our FAQ at www.grmtools.org for more information about the authorization dialogs.
Universal Controls

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

- Sliders
- Numerical Value Fields
- Elastic String
- Buttons
- Presets
- Tempo
- Super Slider
- Save / Load

**Sliders**

![Slider with value 1.001]

To change the value of a Slider, do any of the following:

- 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 [option] key.

**Numerical Value Fields**

![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.
Elastic String

To achieve smooth movement of Sliders, 2DControllers, or the SuperSlider, click on the object and move the mouse while holding down the [Shift] key. *Note:* In general, the follow-speed of the smoothing function depends upon the length of the Elastic String.

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 contains 16 Presets, identified by location numbers, into which you can save all of your parameter values and from which you can call up a specific previously saved configuration of parameter values.
To save your current configuration of parameter settings into any of the 16 Preset locations, click on a location number while holding down the [Control] key. Note that the Status Field, located under the Preset locations, gives the indication of the last performed operation. In the screen shot above, for example, the Status Field displays 'Save 7', indicating that a configuration of parameter values was most recently saved into Preset 7.

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, press down the [Control] and [Shift] keys and click. 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 values called up from a Preset, 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, which is just underneath the vertical Slider, to activate it before you click on the Preset.

**Tempo**

The functionality described here is available only in certain applications, such as Cubase SX, etc. Consult the manual of the program you're using or simply try this feature.

Certain parameters, such as the time of interpolation between presets, can be synchronized with variations of tempo. To initiate synchronization, hold down the [Control] and [Shift] keys 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**

![SuperSlider](image)

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 disable a Numerical Value Field, click in the field and drag vertically downwards to an 'Off' position.

To change continuously from one Preset to another, drag the handle of the SuperSlider to the left or right.
Save / Load Buttons

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.

To save a user file that contains the complete configuration of all of the current values of the parameters and the 16 Presets, click on the Save Button.

To recall a user file that contains a complete configuration of all of the current values of the parameters and 16 Presets, click on the Load Button.
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 frequency. 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 the basic operations of your host software, that you know how to record or import a sound into a mono or stereo track, and how to insert VST plug-ins. For this tutorial, use a song or a wide bandwidth sound.

Set the controls for the plug-in as you see them above. Start the playback.

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 [control] key 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 button. 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.

**2DController.** 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 [control] + [shift] keys and click on the Preset slider or on the Numerical Value Field associated with the Presets.

**Additional controls for the stereo version.** The following controls are found only in 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 high-frequency 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 the basic operations of your host software, that you know how to record or import a sound into a mono or stereo track, and how to insert VST plug-ins. For this tutorial, use a hand-drum or other rhythmic sound.

Set the controls for the plug-in as you see them above. Note that the filter frequencies are set very close together to create a dense resonant sound. Start the playback.

Move the **mix** slider slowly towards the right. 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** control 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 **Universal 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 of the comb filters by a number between 0.8 and 1.2.
**LP.** Multiplies the cutoff frequencies of all of the 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.

**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.

**Tempo.** The interpolation time between presets can be synchronized to tempo. To initiate synchronization, hold down the [control] + [shift] keys and click on the Preset slider or on the Numerical Value Field associated with 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?

Up to 128 delays, controlled in their amplitudes and timing, are distributed within a range of up to 6 seconds. In the stereo version of this plug-in, the delays are assigned alternatively to the left and right channels.

The 2DController lets you set the start time of the first delay and the range within which all the delays will follow. Note the nr of delays slider. If you select 24 delays, for example, with the first delay set at 50 milliseconds and the range set at 3000 milliseconds, the 24 delays will be distributed within the 3000 milliseconds range following the first delay at 50 milliseconds. Within that range, you can control the pattern of delays by moving the sliders. You can set the curve of the amplitude distribution to start loud and get softer, or to start soft and get louder, or to remain equal. You can set the delay distribution to start fast and get slower, or to start slow and get faster, or to remain equal. If you want to introduce a random element in the delay distribution, you can vary the extent of the random deviation and the timing of the random changes. You can also add feedback.

The controls are further explained below in the section called Reference.
A quick tutorial

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

Set the controls for the plug-in as you see them above. Start the playback.

When you hear the sound, move the mix slider to the right 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 1.000.

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

Now, change the amp distrib slider to approximately 1, and drag to control ball to the bottom center of the control window. You should be hearing repetitions of the sound, not quite in sync.

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 [control] + [shift] keys and click on the slider or on the 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?

In general, a Doppler effect gives you the impression that a sound moving towards you rises in pitch while a sound moving away from you falls in pitch. This plug-in, shown above in the stereo version, simulates the Doppler effects of two sounds moving around an axis.

The position of the 2DController, which is the silver ball shown above in the center of the control window, represents an axis around which the sounds move. The little red and green balls on either side of the 2DController represent the sounds from left and right channels as if they were coming from two loudspeakers swinging on ropes suspended from the ceiling. By manipulating the sliders to the right of the control window, you can determine the variation in amplitude and pitch of each sound as it swings, the timing of each of the sounds as it follows the other, the size of the circle in which each sound swings, the speed with which it swings, how far apart they are as they swing through space, and their relative positions in their swing cycles.

The controls are further explained below in the section called Reference.
A quick tutorial

We assume that you are familiar with the basic operations of your host software, that you know how to record or import a sound into a mono or stereo track, and how to insert VST plug-ins. For this tutorial, use a song or a melodic sound.

Set the controls for the plug-in as you see them above. 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 setting of circle amplitude.

Now experiment with the doppler variation slider but note that the actual pitch variation will depend upon the setting of '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.

**channel separation.** In the stereo version, this control determines the distance between the two channels.

**channel phase.** In the stereo version, 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.

**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 [control] + [shift] keys and click on the slider or on the Numerical Value Field associated with the parameter.
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?

A sound passes through the waveform window. When you click on the freeze bar, a 3-second segment of the sound is frozen in the waveform window as a number of simultaneous loops. The loops can be varied in duration and pitch to encompass the whole or any part of the 3-second segment.

As a sound passes through the waveform window, click on the freeze bar to freeze the 3-second segment of the sound that was in the window at the moment you clicked. The sound is frozen as a number of loops between 2 and 32, as set by the nr of loops slider. The loops can be varied in pitch together by the pitch slider, individually by the pitch offset slider, and randomly by the random pitch slider. The durations of the loops will depend upon their pitch and the setting of the random dur slider. The starting times of the loops can be coordinated by the reset Ph or randomized by the random Ph button. By dragging the control ball up and down, you control the duration of the looped part of the segment that you actually hear. By dragging the ball left and right, you scrub the sound through the segment that is showing in the window.

The controls are further explained below in the section called Reference.
A quick tutorial

We assume that you are familiar with the basic operations of your host software, that you know how to record or import a sound into a mono or stereo track, and how to insert VST plug-ins. For this tutorial, use a rhythmic drum track.

Set the controls for the plug-in as you see them above. 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 duration 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 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.

begin. Specifies the start time of the loops within the freeze bar.

end. Specifies the end time of the loops within the freeze bar.

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.
random phase. Unsynchronizes the playback of the loops.

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 32. 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 3-second field. Moving the 2DController horizontally lets you scrub through the 3-second field.

Tempo. The parameters that can be synchronized to tempo are the interpolation time between presets and end. To initiate synchronization, hold down the [control] + [shift] keys and click on the slider or on the 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 the basic operations of your host software, that you know how to record or import a sound into a mono or stereo track, and how to insert VST plug-ins. For this tutorial, use a melody.

Set the controls for the plug-in as you see them above. 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. Modulates the pitch set in 'transpose' in the shape of a sine wave.

frequency. 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'.

frequency. 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.

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

**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.

**pitch detector.** Ensures that the durations of the sound fragments remain consistent with the frequency of the input signal. It is advisable to use this with periodic signals.

**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), **frequency** (periodic modulation), and **frequency** (random modulation). To initiate synchronization, hold down the [control] + [shift] keys and click on the slider or on the 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?

Up to 128 resonating high-pass, low-pass or band-pass filters, each of which resonates a single frequency, are distributed within a frequency range to transform a sound's timbre.

Specify the number of filters with the **nr of filters** slider. Move the **2DController** up or down to widen or narrow the frequency range within which the filters are distributed, and move it left or right to place the range higher or lower. Use the **distribution** slider to distribute the filters within the range. Note that the window below the control window displays the distribution curve.

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

We assume that you are familiar with the basic operations of your host software, that you know how to record or import a sound into a mono or stereo track, and how to insert VST plug-ins. For this tutorial, use a series of staccato sounds or footsteps.

Set the controls for the plug-in as you see them above. 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.

Move the random factor slider and S&H rate sliders to the right. The sounds will seem 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 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.

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 in the mono version from 2 to 128 and in the stereo version from 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.

filter type. Selects lowpass, bandpass, or highpass filter modes. The effect will be heard most clearly with weak resonance values.

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 [control] + [shift] key and click on the Slider or Numerical Value Field associated with 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?

A sound is fragmented and the fragments are shuffled in time. You control the durations and attacks of the fragments, their pitches, and their densities.

The fragment and envelope sliders let you control the size of the fragments and the sharpness of their attacks, the delay slider lets you control how far apart the fragments will be, the initial pitch, final pitch, and random pitch sliders let you vary the pitches of the fragments, the feedback slider adds reverb, the density slider determines the number of fragments that will be generated ...

The controls are further explained below in the section called Reference.
A quick tutorial

We assume that you are familiar with the basic operations of your host software, that you know how to record or import a sound into a mono or stereo track, and how to insert VST plug-ins. For this tutorial, use a solo voice or melody.

Set the controls for the plug-in as you see them above. Start the playback.

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

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 the [shift] key 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 the [shift] key 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.

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.

2DController. Moving the 2DController vertically lets you control the fragment slider. 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 [control] + [shift] key and click on the Slider or Numerical Value Field associated with the Presets.