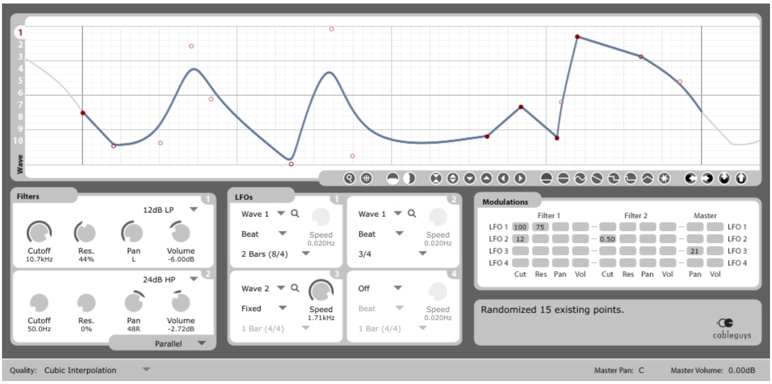
# Cableguys FilterShaper 2

## Create complex volume, pan and filter modulations with ease

## User Manual



February 2011



All magazines are free to include the demo of this version of FilterShaper on their cover CDs/DVDs and to write about this plugin. If you want to know more you can always contact us via contact@cableguys.de. Also it would be nice if you let us know should our effect be featured in your magazine.

The distribution of any kind of the full version of FilterShaper without the written permission of Jakob Rang or Steffen Rose is prohibited.

FilterShaper can be used "as is". In no event shall we be liable for any damages whatsoever.

# Contents

# 1 Introduction

## 1.1 Features

FilterShaper enables you to bring life to your sounds and loops. This unique VST plugin allows you to draw a waveform, and up to 10 of these customized waveforms can be stored. These waveforms can be assigned to any of the 4 LFOs, and the signals produced can be used to modulate cutoff and resonance of FilterShaper's two acclaimed filters. FilterShaper also allows you to make precise volume and pan modulations.

FilterShaper offers precise waveform drawing possibilities that are easily adjustable, and user friendly. You may create waveforms to shape either hard-synced or analog rhythms. You can use this flexible waveform drawing tool creatively to produce interesting rhythmic elements, modulate a bass sound, process vocal recordings, or even make vivid shaker rhythms out of white noise. FilterShaper's flexible options allow you to make either subtle changes, or extreme modulations. You can choose to make a broad range of sounds, from detailed sound corrections or expertly filtered rhythmic loops. Its numerous filter types can be set to a wide assortment of ranges, all the way from zero resonance up to self-oscillation.

One unique aspect of FilterShaper is that you are able to automate every single parameter, providing that your host sequencer is capable of this. Yes, we really mean everything here! Even the shape of the modulating waveform can be easily automated.

And we care about quality. Regardless of the sample rate you choose to work at, FilterShaper's sound will not be altered. We also made sure that FilterShaper's LFOs will always stay in sync, even when you make tempo changes in your host software, or jump ahead to different parts of your track.

## 1.2 Known issues

A sharp waveform is capable of producing clicks in the audio. This is intended, as it allows you produce distorted sounds. You can avoid these clicks by softening the waveform.

There are no known host issues, with the exception of Renoise. In this application, the song needs to play in order to have the waveform display updated.

If you have questions, comments, suggestions, please mail us at:
contact@cableguys.de

Or contact us via our contact page:
http://www.cableguys.de/contact.html

## 1.3 Who created it?

The idea for FilterShaper and this manual was created by Jakob Rang, and was implemented by Jakob Rang and Steffen Rose. Jakob and Steffen both earn their livings in Berlin-based software companies.

The GUI was designed by Kiritan Flux (http://www.kiritanflux.de).



# 2 Installation

## 2.1 Installation

### 2.1.1 Windows XP/Windows Vista/Windows 7

Just copy the plugin's DLL-file to your VST plugins folder. Please restart your host application. Rescan the plugins folder if the plugin is not found automatically.

### 2.1.2 Mac OS X: VST

Just copy the "FilterShaper2.vst"-package / folder to your VST plugins folder. Please restart your host application. Rescan the plugins folder if the plugin is not found automatically.

Most likely the path to copy to is one of these:

- /Library/Audio/Plug-Ins/VST
- <your home folder>/Library/Audio/Plug-Ins/VST

### 2.1.3 Mac OS X: AU

Just copy the "FilterShaper2.component"-package / folder to your AU plugins folder. Please restart your host application. Rescan the plugins folder if the plugin is not found automatically.

Most likely the path to copy to is one of these:

- /Library/Audio/Plug-Ins/Components
- <your home folder>/Library/Audio/Plug-Ins/Components

## 2.2 Demo version

The demo version of FilterShaper gives you full functionality, but you may only run one instance of the demo at a time; also please note that it is not possible to run two Cableguys plugins in demo mode simultaneously. Furthermore, you can open the presets, test-drive the plugin and make adjustments to the parameters. You are able to save in the demo version, but you can only recall your saved material when you purchase a license.

To avoid these restrictions, you will need to purchase the full version. The full version allows for unlimited instances of FilterShaper to be used at the same time, and also for recall of customized adjustments to the presets.

You can buy a license at http://www.cableguys.de

## 2.3 Unlocking

Once you have bought FilterShaper, you will be emailed your unlock key and a link to download the full version of the software. Please save this key file to a local directory. When you start FilterShaper, please enter your email (the one that we sent the unlock key to), locate the key file and then click on "Unlock".

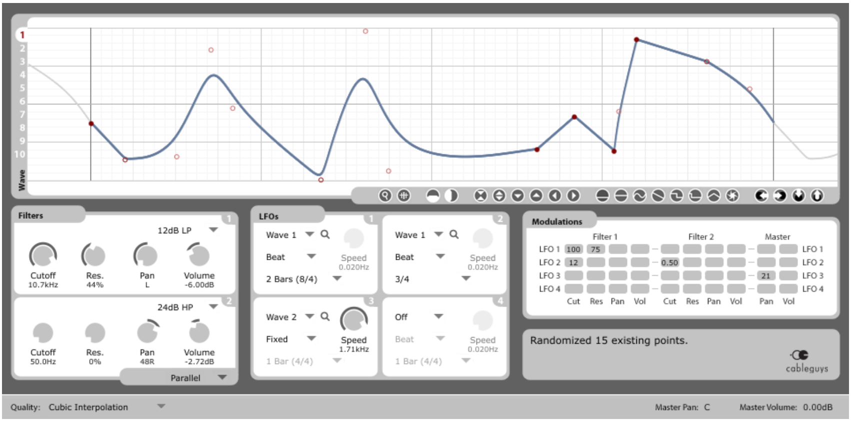
If you have any problems you can email us at:
contact@cableguys.de

Or contact us via our contact page:
http://www.cableguys.de/contact.html

## 2.4 Updating from FilterShaper 1

FilterShaper 2 cannot open songs or presets made with FilterShaper 1. It is no problem to run FilterShaper 1 and FilterShaper 2 in parallel.

# 3 Controls

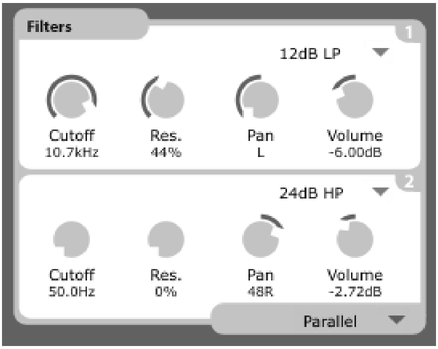



## 3.1 Setting values

To change the values of FilterShaper's knobs and cells in its modulation matrix, simply click on them and move the mouse up or down.

For precise changes, SHIFT-click on a knob or on a cell in the modulation matrix while moving the mouse, or left-click and right-click at the same time. CTRL-clicking sets the controls back to their default value.

## 3.2 Filters




FilterShaper has two filters, and a variety of filter types to choose from, with a range of responses from 6dB to 24dB per octave, with high-pass, low-pass, band-pass, notch and peaking shapes.

All filter types enable you to set the cutoff frequency, and most of them also allow you to set the filter's resonance. The latter can be set to a wide range of values, from zero resonance up to self-oscillation.

You can choose to route the filters in either serial or parallel:

- **Serial:** The output signal of Filter 1 is processed by Filter 2.
- **Parallel:** The incoming audio is processed by both filters in parallel. Afterwards the output of both of the filters are summed together. If using parallel mode, each filter has individual controls for both panning and volume.

Here are two examples of what parallel mode could be useful for:

- To provide some subtle changes to the stereo image, you may choose to hard pan one filter to the left and the other to the right. Use the same filter type on both of them, but try using slightly different cutoff frequencies.

- Or say perhaps you have a mix that includes a bass, as well as shakers. You might choose to use a low pass filter, panned to the center on the low end, and use a high pass filter and modulate its panning controls to provide some movement in the high frequencies.

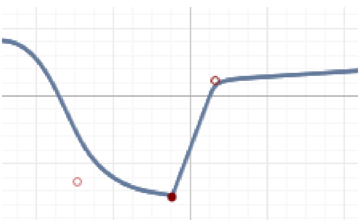
## 3.3 Waveforms

### 3.3.1 Drawing a waveform

The waveform drawing area is what makes FilterShaper a unique, ultra-flexible effect for filter modulations. A lot of effort was put into this feature to make it work smoothly for you!

The waveforms which are drawn in this area can be used by the LFOs to modulate the cutoff and resonance of the two filters. They can also be used to modulate pan and volume.

The waveforms are interpolated according to the points drawn into the window. Each point in the waveform can be one of three following types.



Have a look at the points in the screenshot:

The point on the left is Type 1, it does not have that much influence on the waveform.

The point in the middle is Type 3, a hard breakpoint that produces a sharp bend in the waveform.

The point in on the right is Type 2, and is a bit softer than a point of Type 3. Type 2 points can be used to avoid sharp modulations which may otherwise cause clicks in the resulting audio.

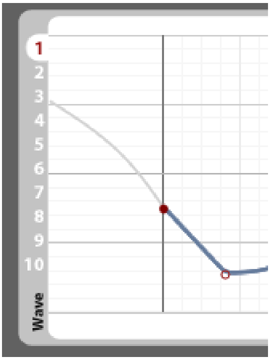
You can create, delete and move points, and change the type of a point:

- Left-click on a free area to create a Type 1 point.
- Left-click on an existing point to move this point.
- Right-click on a free area to create a Type 3 point.
- Right-click on an existing point to change its type from Type 1 to Type 2, or from Type 2 to Type 3. If you click on a Type 3 point, the point will be deleted. So, you can delete a Type 1 or 2 point by clicking on it two (or three) times.

When moving a point, its exact x- and y-values are shown in the bottom right hand corner of the wave area.

Please note that when the plugin is bypassed, the waveform that is shown will not be updated until the plugin is activated again.

### 3.3.2 Visible waveform




Next to the waveform area there are buttons which allow you to choose which waveform is displayed.

### 3.3.3 Waveform buttons

There are several buttons below the waveform:

 Scales the displayed width of the waveform.

 Inverts the x- or y-values of the points.

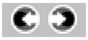
  Decreases or increases the waveform's amplitude by dividing or multiplying their current values. SHIFT-click on these buttons for precise adjustments.

 Moves the waveform to the left or to the right. SHIFT-click on these buttons for precise adjustments.

 Selects one of two flat waveforms.

 Selects a classic analogue waveform: sine, saw, square, pulse or triangle.

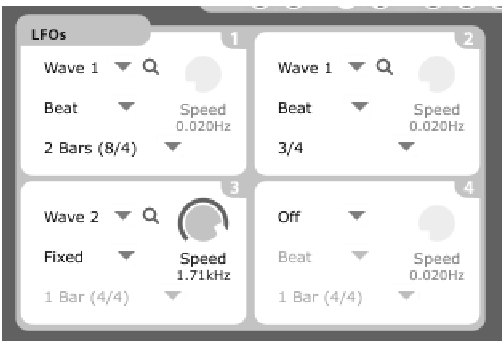
 Generates a random waveform. The randomized waveform will contain the same number of points as the current waveform.

 Undo/redo your last waveform edits.

 Takes a snapshot of the current waveform.

 Recalls the last snapshot. If no snapshot has been taken, clicking on this button has no effect. Note that you can copy and paste snapshots between FilterShaper's 10 waveforms. This gives you the ability to back up a waveform. You can also begin editing a waveform using a copy of another waveform. This is especially useful if you automate LFOs to switch between them.

## 3.4 LFOs




FilterShaper has 4 LFOs which can be used to modulate cutoff and resonance of the two filters, and to modulate pan and volume.

### 3.4.1 Choosing the waveform

Each LFO has a menu for choosing one of the 10 waveforms. If it is set to "Off", the LFO is not used, which saves you a little CPU consumption.

There is a magnifier button next to that menu. Click on it, and the chosen waveform will be displayed in the waveform window.

Changing waveforms allows you to easily compare different waveforms to each other. Also, it can come in handy with automation. You may choose to use one waveform for the intro of your song, and change to another waveform later.

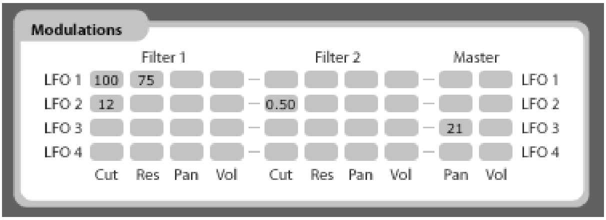
### 3.4.2 LFO modes

You can set the LFOs to different modes:

- **Beat mode:** In beat mode a menu is displayed which allows you to set the loop length in rhythmic units, which are always in sync with your host software.

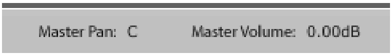
- **Fixed mode:** If you choose fixed mode, a knob appears for setting the speed of the loop in hertz. This mode operates independently of the host sequencer's tempo.

## 3.5 Modulation matrix




You can modulate cutoff, resonance, pan and volume using the LFOs. The modulation intensity is set within the cells of the modulation matrix. Higher values will result in more intense modulations, while a value of zero means that a parameter is not modulated at all. A negative value behaves as if the waveform, which is used by the LFO, is mirrored on the x-axis.

## 3.6 Master section




There is one control for setting the master pan and another one for setting the master volume. CTRL-click on them to set them to their default value.

## 3.7 Interpolation


The interpolation type determines how precise the modulations will be.

- **No Interpolation:** We thought it might be a nice idea to give you the opportunity to use no interpolation at all, although this result is not as exact as linear or cubic interpolation. Choose "no interpolation" if you are looking for distorted digital sounds with modulation artefacts.

- **Linear Interpolation:** This results in fairly precise modulations.

- **Cubic Interpolation:** The difference between linear and cubic interpolation is minimal, and to be honest, we are able to measure it, but the effect is almost not audible. However, if you would like to achieve the highest sound quality possible, then use this setting. Cubic Interpolation is slightly more CPU intensive than linear interpolation.

## 3.8 Automation

You can change all parameters via automation. Everything. The cutoff frequency and resonance, the LFOs, all point movements and even the visible waveforms and interpolation type. That means that you can record and play back all of your edits, and you can use any hardware controller to control any parameter in FilterShaper.

You may choose to use Waveform 1 for the intro of your song and change to Waveform 2 later. Or you can change the speed of your modulation by automating the loop length.

FilterShaper has a lot of parameters, as every possible point in a waveform adds three parameters (type, time position, and amplitude). Please note that some hosts do not allow to access all parameters. E.g. if your host is limited to 128 parameters you may only be able to automate some of the points of the first waveform. However, all parameters will still be able to be saved and loaded correctly.

# 4 Extended Knowledge

## 4.1 Parameter changes

Due to limitations of the VST specifications, parameter changes (made via automation, drawing in the wave area or changing the control knobs) do not happen on a per-sample basis. Instead, parameters are updated only once before an audio buffer is processed. A parameter update happens before the next audio buffer is processed. In short, parameter changes are quantized to the size of the audio buffer, which is set in the preferences of your VST host or audio interface settings.

Setting small buffer sizes gives you smoother control, but may also drastically increase the CPU consumption for FilterShaper, especially when moving points in the waveform area.

The most CPU intensive calculation is updating the waveform that results from changing the waveform's points. To prevent audio dropouts due to high CPU consumption, we split this calculation across several audio buffers. As a result, it may take a moment to update the waveform when moving points in the waveform area.

Decreasing the audio buffer allows for a faster response of changes made to the parameters, but will also increase the CPU load.

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