Ohm Force

QUAD FROHMAGE

Virtual Filter

User manual version 1.20 – 2004.06.21

TABLE OF CONTENT

1. Getting started

Thank you for purchasing the Quad Frohmage effect. This manual is divided into three sections :

Getting Started, which explains how to install the Quad Frohmage plug-ins and get it working;

Common features, rounds up the common features you can see on every Ohm Force effects;

Effect usage, which shows you how to operate the plug-in.

1.1 Features and requirements

Quad Frohmage is a high quality audio filter, available in two interfaces, the "Classic Skin" and the "Funky Skin".

It can work as a stand-alone program, or as plug-in for various platforms. The requirements given below can vary depending on the latest program versions, see the latest release notes in the file `readme.txt`.

You will need at least 64 MB of RAM, 25 MB on your hard-drive, a Pentium II-compatible CPU on PC machines and a G4-compatible CPU on Apple Macintosh.

- MacOS Classic

Requires OS 9.x and CarbonLib 1.5 or higher. See how to update on the Apple website: http://www.apple.com/support/. The plug-in is available for the VST, MAS and RTAS platforms.

- MacOS X

Requires OS 10.1 or higher, Jaguar is recommended. The plug-in is available for VST and AudioUnits platforms.

- Windows

Requires Windows 98, 98 SE, ME, 2000 or XP. The plug-in is available for the VST, DirectX and Winamp 2 platforms.

1.2 Installing

Run the installer and follow the instructions, which may vary depending on the platform. The plug-in has a unique serial number needed for installation. Check your purchase receipt, or the label on the manual cover, where there should be a User Name / Serial Number. Copy these fields exactly as they appear on the label when prompted by the installer.

☞ **Warning**: Serial numbers are checked only when the plug-in is actually used. If you mistyped the codes, a dialog box will warn you only when trying to run the plug-in. To enter the correct serial number again, you have to quit every application trying to use the plug-in and to reinstall it at the same location as previously. If you decide to change the installation location, *uninstall first the plug-in*. It would be a prudent to systematically uninstall it in the case of wrong code.

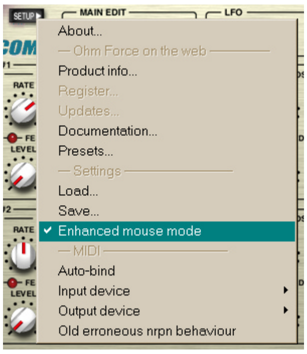
☞ **Note**: The stand-alone versions do not require any code for installing.

1.3 First contact

Open Quad Frohmage and feed it with some sound. The MacOS Classic users should ensure that there is enough memory reserved for the application.

A good way to start the tour is to try the *factory presets*. On the plug-in you will find a frame with eight numbered buttons in it. Click on these buttons to activate each of the eight factory presets.

Turn the knobs by clicking on them and dragging the mouse vertically.



☞ **Note**: If your mouse suddenly goes mad, stay calm and locate the **SETUP** button. Click on it to open the menu and unselect **ENHANCED MOUSE MODE**. This may happen with some mouse, graphic tablets or trackball devices.

1.4 Register

In order to get the latest updates and support directly from the Ohm Force website, you can register your plug-ins on-line. To do so, click on the **SETUP** menu and select **REGISTER**. Your computer should be connected to the Internet. The Register option will open a browser on the Ohm Force web site (http://www.ohmforce.com). You also need to have an account on this web site; if you still do not have one, you will be given the opportunity to create one. Follow the instructions on the web page.

2. Common features

Ohm Force's plug-ins share a lot of important features. They might not look alike because of graphic design differences, but they have the same basic functionalities. Let’s review them.

2.1 Preset support

A bank of eight slots enables you to memorize your sound settings, and can be saved on your hard disk. Those banks are multi-platform; therefore, you can use your presets on another computer or with another sequencer.

You can add a transition time when applying a preset, during which the buttons are going to turn slowly to go from the current setting to the new one you have chosen.

2.1.1 Presets / Memorize

To memorize the effect current setting in a preset, click once on the **STORE** button; it will light up. Then click on the button of the preset in which you wish to memorize the effect.

To apply a preset, make sure the **STORE** button is off. To turn it off, click it once. Then click on the preset you want to activate.

2.1.2 Rename

Once you have stored your preset, you can rename it by clicking on the display screen. Press **[RETURN]** to validate your change or **[ESCAPE]** to cancel it. Note that the name applies only to the last selected or memorized preset.

You can check the names in the preset menu by clicking on the small down arrow. This menu acts exactly like the preset buttons, you can apply or record a preset by selecting an entry in the menu.

2.1.3 Transition time

This potentiometer enables you to vary the time the plug-in will take to go from a sound to another when a preset is activate. The time measured in seconds is displayed beside. By default, this duration is to one second. Set it to 0 if you want the immediate preset application.

2.1.4 Load / Save Bank

These two buttons will help you save and load your preset banks on the hard disk for a later use. The 8 presets are memorized or loaded at once. During disk loading, the current setting is not modified. There are a lot of presets bundled with your Quad Frohmage. If you're trying to achieve a particular sound, you can start with a preset close to the idea you have in mind, and tweak the settings to get the wanted result.

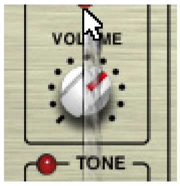
2.2 Using knobs and faders

Not all the knobs and the faders work the same way. There are two modes: direct action or slide-clicks.

2.2.1 Direct action

You can catch the button by giving a long click on it (on the mobile part for the fader) and moving the mouse up or down, keeping the button pressed.

Actually, each button has a *preferred* direction for the mouse movements : it is vertical for the knobs and according to their orientation for the faders. If you move the mouse in the preferred direction, the move will be quick. But, if you move your mouse in the *perpendicular* direction – horizontally for knobs, then the move will be slow and therefore very accurate.



Some buttons have notches to constraint certain values. However it is possible to set the button position between two notches. To achieve this, move the mouse along the perpendicular direction mentioned above.

2.2.2 Side-clicks

The button is divided into two zones on which you can click to turn the button on the right or on the left. For the faders, those two zones are on both sides of the moving part. For the knobs, they are at 4:30 and 7:30 on the button. The button will move slowly if you give a long click on these zones without moving the mouse. This can help slightly and quickly adjust a parameter value.

If you click on this zone then move your mouse without releasing it, the button will move automatically, and keep moving even after having released it. When you move the mouse away from where you clicked, the movement of the button will get faster. To stop that move, just click again on the button.

2.2.3 Linked buttons

On a few Ohm Force plug-ins, knobs can be linked, because they are associated to similar parameters. For instance the parameters of the four Quad Frohmage filtering bands. Indeed you can affect one parameter of Band 1, 2, 3 and 4 at the same time, with a single click.

To do so, you have to click on the button with right button of the mouse (click while holding the **[Control]** key on Macintosh systems). The parameters of both lines take the same value.

If you hold the **[SHIFT]** key pressed and click on the right button of the mouse, both knobs move at the same time but keep their own original gap. For instance, when the original value of the first knob is 10 % and the original value of the second knob is 50 %, if you increase the value of the first button up to 30 %, you increase the value of the second button up to 70 % at the same time.

You can also combine the right click to the **[CONTROL]** key (the **[COMMAND]** key on Macintosh). It will reverse the move of the "slave" knob(s).

2.3 Parameter information and modulation

2.3.1 Parameter

This contextual zone depends on which parameter you have chosen – it has a colored outline. Indeed, it would be horrifically complex if all the numeric parameter values were displayed on the interface at once.

- **NAME**

Name of the selected parameter.

- **VALUE**

It is the parameter value expressed with the selected unit. You can edit this value by clicking on it. Press **[RETURN]** to validate your change or **[ESCAPE]** to cancel it.

2.3.2 Tempo control

Because many of plug-in's applications are related to music, and therefore to rhythm, it was necessary to take the song tempo into account. Indeed various settings are oscillating or beating, and synchronization with the piece is really convenient.

Some host programs can synchronize the plug-in internal tempo with their own tempo. In this case, the BPM display is only a display. Otherwise, you can change the tempo by clicking on the handle right to the numeric display. You can also edit the numeric display itself.

2.3.3 Automation

2.3.3.1 Support

Every parameter including the LFOs and other modulation devices are potentially automatable on the RTAS, MAS, VST and AudioUnit platforms. However depending on your host's capabilities, you might be restricted to only 16 fixed parameters, or even none. Check your host reference manual to find out how to automate a parameter.

Digital Performer and ProTools show directly on the plug-in interface which parameters are currently automated. A green triangle on a knob indicates that the automation is playing, and a red disc shows an automation recording.

The DirectX version does not support DXi automation yet, please use the MIDI automation instead.

2.3.3.2 The VST case

Some host programs like Steinberg Cubase or Emagic Logic Audio have several limitations regarding VST plug-in automation. They can handle only a few parameters whereas Quad Frohmage has hundreds of them. In consequence, some important parameters cannot be automated. This can often be done by using MIDI commands, but this solution is not always the most convenient.

To fix this problem, we give you the possibility to change the order in which VST parameters are presented to host. Proceed by selecting **SETTINGS** / **LOAD** in the menu opened by a click on the **SETUP** button. Locate the file `easy_vst_automation.cfg.txt` in your Quad Frohmage installation folder and open it. Now the important parameters are located on the top of the VST list so they can be automated.

Still not happy with the provided configuration file ? You can make your own. First, save a configuration using **SETTINGS** / **SAVE**. Load it in a text editor, along with `easy_vst_automation.cfg.txt` so you can take a look for reference. You can see that a configuration file is made of "keys". They have a name and a content, which can be made of other keys, in a recursive structure that scientists are calling a tree. Key name and content are separated by an equal sign (=), and complex key contents are enclosed by braces.

The provided configuration file is way lighter than yours. This is because it is a *partial* configuration, whereas yours is a *complete* one. Suppress some subkeys, like the whole `midi` section, in order to make the two files look alike. Anyway, yours will remain longer.

Let's focus on the `parameter_reorder_map` key. You'll see several parameter names. In the file you have saved, there are all the plug-in parameters. Now, just move the parameters you want to automate at the top of the list. You can leave the other parameters if you want to specify a particular order, or you can simply suppress them. This does not means that they will not appear any more or not being automated at all. When loading the configuration file, the plug-in will find automatically the best mapping for the missing parameters. Once you have finished to sort the parameters, save your work and reload it into the plug-in.

☞ **Important**: If you already did settings before applying the mapping file, you should better save them into an *internal* preset – the one described in the Preset section of this manual. You should not use the VST host presets anymore because they will be completely randomized after the change. Instead, apply your saved preset to get your sound again. Fortunately, the new presets you'll make after the change can be stored as VST ones and reloaded.

2.4 MIDI support

The Quad Frohmage plug-in can receive MIDI commands to control parameters. MIDI can even replace automation, because the plug-in is not only able to receive these commands, it can also emit them when the user twists knobs or pushes buttons.

Effects are in "Omni" mode, meaning they can receive MIDI commands from any channel. However all the commands are sent via Channel 1. Commands can be regular CC (Continuous Controllers), but also RPN and NRPN (Non-Registered Parameter Numbers). Depending on what your MIDI device is supporting, it can be better to use CC or NRPN. CC are the most commonly used by hardware devices, but NRPN have a higher resolution.

The factory MIDI settings are using NRPN, but it is possible to change the mapping at any time. The default mapping is listed in the last section of this manual.

☞ **Note**: MIDI support is disabled in the demo version of the plug-in.

2.4.1 Selecting MIDI ports

Depending on your host, your MIDI devices and your system settings, you may have one or more available MIDI port, for input and for output. It is possible to select the virtual ports for receiving and sending MIDI events.

To chose the input port – the one on which MIDI data is received by the plug-in, click on the **SETUP** button, go to the **MIDI** / **INPUT DEVICE** menu and select the one you want. Do the same thing to select the output port. The selected MIDI port will appear checked in the menu. It is possible to chose only one MIDI port for input, and one for output.

A reason for a failed connection is the use of the port by another piece of software or plug-in, likely your host program. In this case, check your host operating manual to know how to free up the port.

2.4.2 Binding parameters to MIDI controls

The easiest way to associate a parameter with a specific MIDI controller knob, or any MIDI Control Change, is to use the auto-bind feature. First, activate the auto-bind mode by checking **MIDI** / **AUTO-BIND** in the **SETUP** menu.

If you already selected a parameter before, its name will display in the menu, between brackets, like this: `Autobind [target : Volume]`. If not, select the one you want to bind on a MIDI control change. After auto-bind activation, you can change the selected parameter. Only the last selected one will be taken into account for binding.

Once you have chosen the parameter, send a MIDI event to the plug-in (for example, turn a knob on you external MIDI controller). It can be a simple CC, a RPN or a NRPN. As soon as the event is received, the connection is done, the MIDI command will remain associated with the parameter. Only one parameter can be bound to a MIDI command, and vice et versa.

If you want to bind more parameters, repeat the procedure: select another parameter, and send another MIDI event. Do not forget to exit the auto-bind mode by unchecking the corresponding entry in the **SETUP** menu.

2.4.3 Saving and restoring the MIDI configuration

If you have numerous parameters to bind each time you want to use the plug-in, you can save the configuration for later use. The currently selected ports will also be saved.

To do so, select **SETTINGS** / **SAVE** in the **SETUP** menu. You can restore the settings at any time by selecting **SETTINGS** / **LOAD**.

☞ **Important**: the MIDI configuration is not part of the presets therefore it is not saved with the host song. You have to load the settings manually after having loaded a song on your host application.

The true tech freaks among you will notice one can open the saved file in a text editor and tweak the configuration from here. It is possible to build "partial" configurations by only keeping a couple of the "keys". The content syntax is rather simple but will not be covered much in this manual.

2.4.4 Other MIDI features

- **OLD ERRONEOUS NRPN BEHAVIOUR**

This option is checked by default, and is here in some plug-ins for historical reasons. Our plugins used to interpret RPN and NRPN controls erroneously. As a consequence, automation recorded using old versions cannot be interpreted by the recent versions, unless this option is checked. You are advised to uncheck this option if you are a new Ohm Force user.

2.5 Stand-alone versions

Each plug-in exists also as a stand-alone version, capable of playing and looping a sound file, or processing the soundcard input.

2.5.1 Setup

The first time you run a stand-alone version of an effect, you will be asked to setup your sound card. You can change it later by clicking on **MENU** / **AUDIO SETUP…** at any time. The dialog has several fields.

- **AUDIO ENGINE**

Here you can select the driver category used to produce and capture audio. Use ASIO only if you have specific ASIO drivers installed on your computer. Windows user may prefer the DirectX drivers over the MME. Changing the audio engine will automatically stop the sound input and output, even if you have not pressed the **OK** button yet.

- **INPUT DEVICE**, **OUTPUT DEVICE**

Here you can select the devices for sound ouput and input. Normally, you'll have to specify at least the output device to make the program work.

- **SAMPLE RATE**

Choose the sample rate. Default is 44.1 kHz, but you can set it higher if your soundcard supports more.

- **BUFFER SIZE**

This is the buffer size, in samples. Buffer size affects the latency, which is the unavoidable delay between sound and user input, and what you actually hear. You will need small buffer size to achieve a low latency. However, small buffers tend to increase CPU load. 256 samples seems to be a fair choice in most cases.

2.5.2 Loading sound files

When no sound file is loaded, the effect processes input of the sound card. To load a sample, click on **MENU** / **LOAD SAMPLE…** and select a sample from your disk. The sample will be integrally loaded in RAM before any processing, so be careful not to pick up a multi-gigabyte file !

You can change the sound file by repeating the operation described above, or return back to input processing by choosing **CLOSE SAMPLE** in the **MENU**.

2.5.3 Playback toolbar

This is where you control the playback settings.

- **PLAY – STOP**

Push this button to start the sample playback, when one is loaded. The button icon turns into a square. Press again on it stop the playback.

- **REWIND**

Restart the sample without interrupting playback.

- **LOOP**

Activate or deactivate the loop mode of the sample playback.

- **PANIC**

Cut out the sound. Useful when the beast is becoming enraged and gets out of control.

- **MENU**

Click here to access the configuration menu.

- **TEMPO**

You can slave the plug-in tempo with the BPM Beats Per Minute value indicated in the box. Use the handle and the arrow buttons to change it, or enter the tempo manually by clicking on the box.

- **STRETCH**

Stretch the sample playback to match the duration given by the tempo and the number of beats.

- **Beats**

This field is taken into account only when the **Stretch** function is activated. It indicates to the program how many beats are in the sample loop, because it cannot guess it itself. Of course it depends on the sample you loaded, so you will have to adjust it after each loading, if required.

2.5.4 Volume control

- **Input**

Controls the level applied to sample playback or audio input. The vu-meter indicates the volume after amplification.

- **Output**

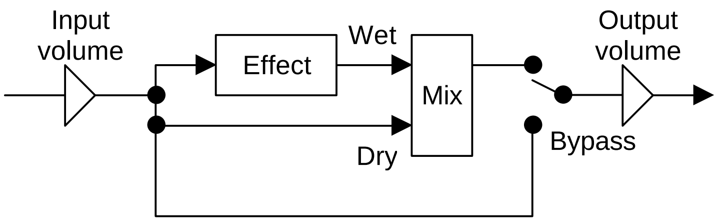
Controls the final volume, plug-in output eventually mixed with the input. The vu-meter indicates the final volume of the signal sent right to the output.

- **Mix**

Set the amount of effect input ("Dry") and output ("Wet") sent to the sound card. Turn it to the left, you will get only the input. To the right, it gives pure output.

- **Bypass**

When activated, the effect is bypassed, meaning output is directly fed with input or sample playback. However input and output volumes are still effective.

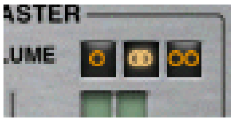


3. Effect usage

3.1 Functioning overview

Quad Frohmage is a complex filter effect. Its core is made up of four identical modules, known as bands. Each one has a one-tap delay, synchable to the song tempo, a multi-mode filter and an optional distortion. Every parameter of the module is automatable – assuming that your host application supports it, and most of them can be modulated, via MIDI or their own modulators (LFO, ADSR).

3.1.1 Stereo mode

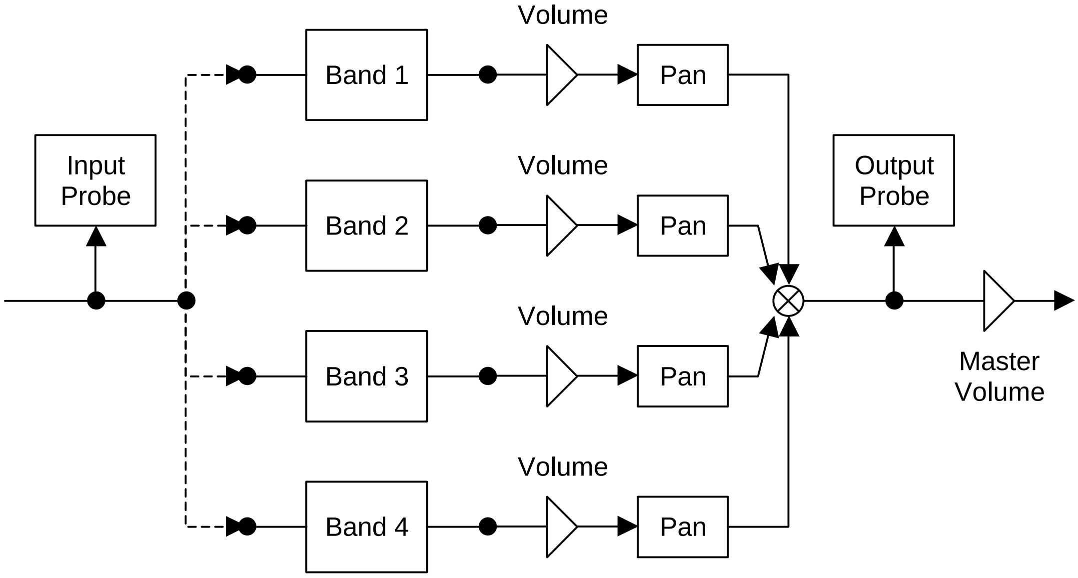


The Quad Frohmage has various way to deal with stereo signals. It can work in mono, dual mono and stereo mode.

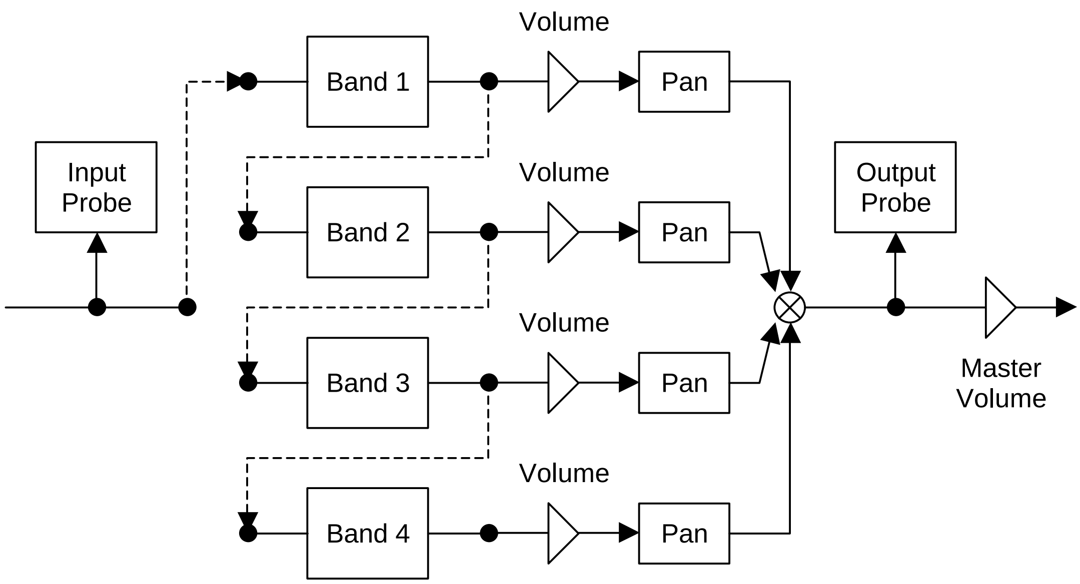
In mono mode, each band is monophonic, and the input signal is summed in a mono signal.
In dual mono mode, each band is still monophonic, but the input signal is split in two mono signals. Depending on the bands routing, a band will be able to process the left or the right channel.
In stereo mode, the way things work is pretty close to the mono mode, excepted the signal path is in stereo. Each band processes a stereo signal.

3.1.2 Routing

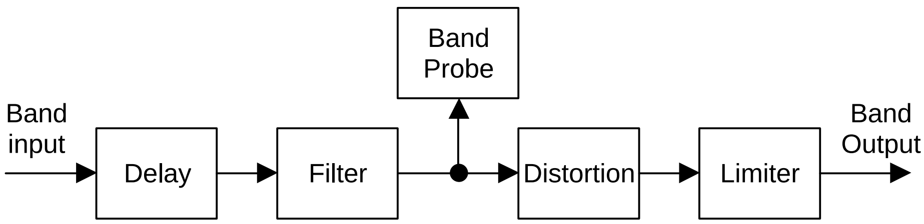
First, let's review the signal path and routing, probably the most complex part of the filter. The four bands can be arranged in parallel, in serial, or in one of the 8 possible combinations. Whatever the configuration, the output of every band can be mixed into the final output. The bands themselves and the probes will be described later.



The filtering bands are connected in parallel.

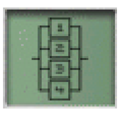


The filtering bands are connected in serial.



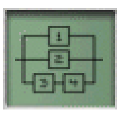
Detail of a band.

The following table sums up all the possible routing, and clarifies the exact behavior of the dual mono mode:



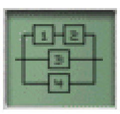
All bands are in parallel. This is the default routing, and the most obvious, since each band uses the plugin input as input.

☞ In dual mono mode, left channel feeds bands 1 and 3, right channel feeds bands 2 and 4.



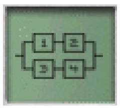
In this case, band 3 feeds band 4. If you set volume of band 3 to –∞ dB, you can hear on band 4 the signal passing through the two filters. If you mute band 3, it's like having bands 1, 2 and 4 in parallel.

☞ In dual mono mode, band 1 process the left channel, band 2 the right channel, and band 3 the sum L+R.



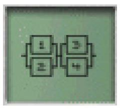
This is very close to previous routing.

☞ In dual mono mode, band 3 uses the left input, band 4 use the right input and band 1 uses the mono mix.



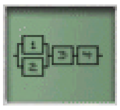
Here two blocks are in parallel. First block is made of band 1 and 2 in serial, second block of bands 3 and 4 in serial.

☞ In dual mono mode, band 1 uses left input and band 3 uses right input.

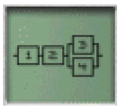


With this configuration, you have two blocks in serial: first block is made of band 1 and 2 in parallel. This block feeds another block, made of bands 3 and 4 in parallel.

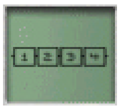
☞ In dual mono mode, band 1 uses left channel input and band 2 uses right channel. Bands 3 and 4, in any case, use a mix of band 1 and band 2.



This routing is close to previous one, excepted that the second block is made of band 3 and 4 in parallel.



With this routing, being in mono or dual mono mode is the same in the end: band 1 uses a sum of the left and right inputs. Bands 3 and 4 are fed by the band 2.



This is the “all serial” mode, each band feeding the next one. As previous mode, its behavior is unaffected by the stereo mode. You can get a 144 dB/octave low-pass filter using a 36 dB/octave LPF on every bands ! Set the bands 1, 2 and 3 volumes to 0 so that only band 4 goes to the output, link the bands cutoff frequencies, with a ratio of 0 %, and you are done. You can also get a 16 beat delay using the same configuration and setting the delay of each band to maximum (4 beats).
As with previous routing, dual mono and mono modes behave the same way.

3.1.3 Modulation

Traditionally in Ohm Force plug-ins, parameters can be modulated by LFOs. But in Quad Frohmage things have changed and the LFO is joined by envelope followers and ADSR generators. These modules produce signals, which are added together to define the final modulation, which will be applied to the parameter.

However the new boys are not just modulation generators like the LFO. Instead, they convert the audio signal into a modulation signal. To do this, they use “probes”, plugged into the audio path to analyze it. Envelope follower detects the sound energy to transform it into modulation. ADSR is less trivial: it detects sudden variations of energy and detects any “On” and “Off” events, driving the ADSR generator accordingly.

3.2 Band

In this section we’ll review the core of the Quad Frohmage – the filtering bands. They are made of four chained blocks: a delay, the filter itself, a distortion and a limiter. Finally the band is mixed to the output.

3.2.1 Delay

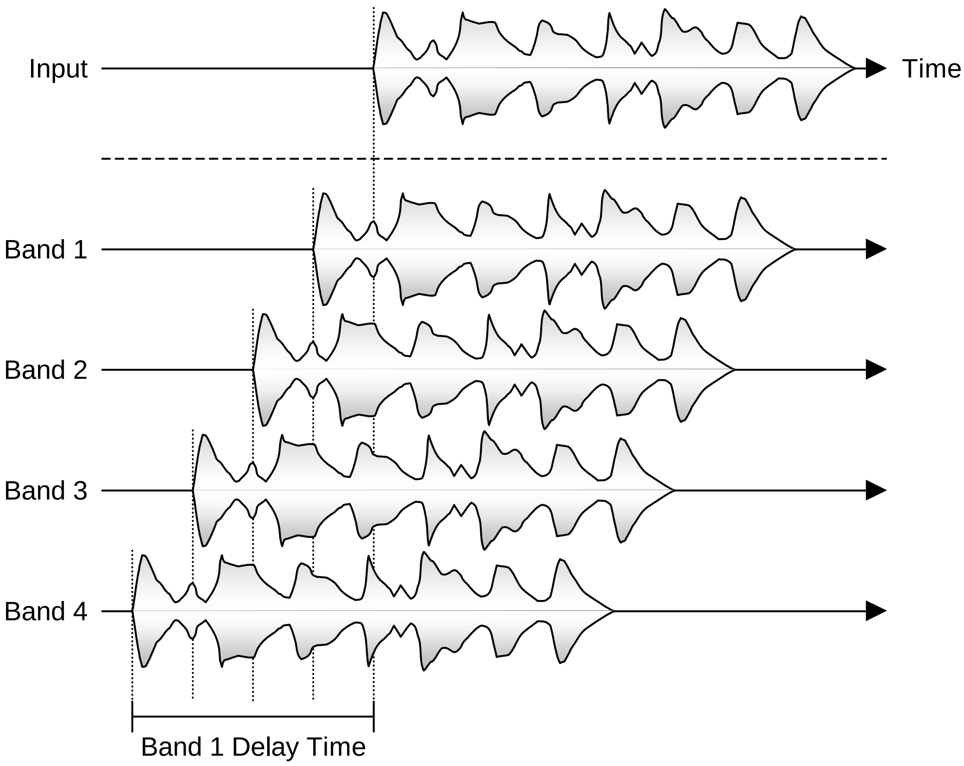
Before entering the filter, sound can be optionally delayed. Delaying parallel bands differently allows you to achieve spectacular rhythmic effects, especially when complex modulation enters the arena. Please note that it's a simple delay; there is no feedback involved.

- **DELAY TIME**

This fader controls the amount of delay. Although it can be specified in beats or in seconds, depending on the state of the Beats/ms button, the effective delay time will always be tempo-dependent. The maximum delay time is 4 beats, meaning 4 seconds at 60 BPM.

- **DELAY LINK**

It is possible to automatically link the delay times of the four bands. When linked, only one knob is controllable, the others becoming transparent. The delays are then equally spaced on every band; for example if the knob is set to the time T, band 1 has a $\frac{T}{4}$ delay, band 2 has $\frac{2T}{4}$, band 3 has $\frac{3T}{4}$ and band 4 is delayed by T. This is a bit weird, as the Band 1 Delay Time parameter is not the actual band 1 delay but the band 4 delay. When Delay Link is activated, the Link LED is lit.



The linked time is post-modulation. This means that band 1 modulation also applies to other bands, and the band 2–4 modulations are simply ignored.

☞ **Tip**: An amazing use of the delay is the reverse playback. First, set a 1 beat delay. Then activate the delay LFO: a 1-beat period, a Ramp-up waveform and an amplitude of 100%. Deactivate the S&H and the smoothing effect by setting them to 0 %. And voilà ! You can decrease the playback latency by lowering both delay and LFO period. However it may reveal small pops and a ring modulation effects if set to extreme values.

3.2.2 Filters

The Filters are the main sound-sculpting element in Quad Frohmage, the kernel of the bands. A filter attenuates and removes some frequencies from the spectrum. For example, it gives muffled sounds when high frequencies are attenuated, or aerial sounds when low frequencies are missing. When removing both low and high frequencies, you can achieve phone-like sounds. Of course there are many combinations possible, but you'll get a better idea by experimenting yourself with the filters included in the effect.

It is also possible to boost a specific frequency band to make it prominent. This boost is called resonance, and is useful for adding color and character to sound. In fact, these filters are so extreme that, when you push the resonance into the red zone, they can produce a plethora of tones and whistles, even without any signal input.

Most of these filters are carefully modeled from their analogue counterparts. We also added two essential tools which can be assimilated with filters: a Comb and a Ring Modulator.

3.2.2.1 Selecting the filter type

The first group of buttons let you choose the primary type of filter. The five leftmost are classical filters.

	LOW-PASS	It passes the frequency components below the cutoff frequency and rejects other frequency components.
	HIGH-PASS	It is the exact inverse of the low-pass filter, removing low-frequency content.
	BAND-PASS	It passes only a band of frequency components centered on the cutoff frequency. The resonance knob determines the bandwidth.
	PEAK Resonator	This filter increases the loudness of a particular frequency range leaving the rest as it is. It is most commonly used for wha-wha effect.
	NOTCH Band-Cut	This removes the frequency content around the cutoff frequency.

These five filters are declined into two flavours:

- **STANDARD**

Offering high resonance capabilities, these are the preferred choice to start with.

- **SVF**

These filters can also self-oscillate when the resonance is set high, over about 80 %. In this case they produce nice whistles even with silent input. SVF also sound a bit different from the classic filters and have their own character.

☞ **Warning**: for stability reasons, the SVF filters have their input hard-clipped to 0 dB. Be careful to feed them with properly leveled signal to ensure maximum quality.

For LP, HP, BP and Peak filters, you can choose the steepness of the slope. The steepness determines how fast the frequencies are attenuated when moving away from the cutoff frequency. The steeper the slope, the closer to ideal the filter is. For the LPF and HPF, you can choose a slope ranging from 6 dB/octave to 36 dB/octave. BPF can achieve from 6 dB/octave to 18 dB/octave on each side, while Peaking filter has no such measurement. Instead, you can choose if it is a 2-pole or 4-pole filter.

Then the last three types of filter:

MOOG-LIKE This is a replica of the famous analogue Moog filter, a self-oscillating low-pass filter with a 24-dB/octave slope.

COMB This rich filter produces several peaks regularly spaced in the spectrum. At low resonance, your audio will sound as if it is being packed down a steel tube with a plunger. Turn the dial, and your drums transform themselves into plucked strings. With an LFO on the cutoff you get a flanger. Again, two modes can be selected on the screen below: positive or negative feedback. this changes the peak distribution across the spectrum.

RING MODULATOR Essentially, this inter-modulates the input with a sine wave whose frequency is tuned on the cutoff. It shifts and mirrors the spectrum, detuning the sound to produce bell-like tones.

Summary of the possible filter combinations:

Type	Sub-type	Order
LP	Classic	6, 12, 24, 36
	SVF	12, 24, 36
HP	Classic	6, 12, 24, 36
	SVF	12, 24, 36
BP	Classic	6, 12, 18
	SVF	6, 12, 18
Peak	Classic	2, 4 poles
	SVF	2, 4 poles
Notch	Classic	
	SVF	
Moog-like		
Comb	Positive feedback	
	Negative feedback	
Ring Modulator		

3.2.2.2 Primary filter parameters

Having selected the filter type, you can play with the parameters to modify the filter effect.

- **CUTOFF FREQUENCY**

This is the main filter control. The knob lets you set the frequency in the 20 Hz – 20 kHz range. However it is limited to 11 kHz for comb filters if the host sample rate is set to 44.1 kHz.

- **RESONANCE**

When available, it controls the amount of resonance in the system. At low values, the filter is not colored, whereas at high values a small band of frequencies is highly amplified, giving ringing tones.

On the BPF, the resonance controls the bandwidth of the filter. Low values means low selectivity, therefore wide bandwidth - almost the whole spectrum. High values make it very selective, especially if a steep slope is selected.

3.2.2.3 Fatness, Color and damping

Depending which filter is selected, the same button can be either the Fatness, Color or Damping.

- **FATNESS**

On filters, the resonance dramatically increases the volume of a given band. This can increase the overall sound energy, sometimes more than desired. You can always adjust the final volume manually, but it can be painful if resonance changes often, or is modulated by a LFO.

To deal with this issue, it would be useful to have an automatic gain control, keeping constant the volume loudness. This is exactly what the Fatness knob is intended to do. Low values make the volume quite constant while resonance is changing. High values do not affect the sound, thus the "pass-band" signal has a constant volume. The pass-band signal is the part of the spectrum, which is not filtered, for example the band below the cutoff frequency for a low-pass filter.

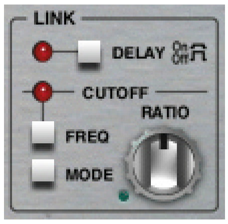
- **COLOR**

This parameter is active only for the Ring Modulator. It sets the amount of ring modulated signal, so at 100%, the filter output is a pure Ring Mod, and at 0%, the filter is bypassed.

- **DAMPING**

6 dB/octave LPF and HPF are actually shelves: you can adjust the minimum stop-band level with the Damping button. When Damping is set to 0 %, the filter has a real 6 dB/octave slope. At 100%, the filter is bypassed.

3.2.2.4 Linking bands



It is possible to automatically link the frequencies of the four filters. Thus you can turn one knob to modify every cutoff frequency. This is useful when the ratio between frequencies has to remain fixed to get a particular timbre.

- **FREQUENCIES**

Push this button to link or unlink the frequencies. In Link mode, only first filter frequency modulation is taken into account for the group, other filter frequency modulations are ignored.

- **MODE**

This switch determines how the frequencies are linked:

Linear: the frequencies are regularly spaced, in a linear way.
Harmonic : the frequencies are exponentially spaced, meaning there are always the same number of semi-tones between two adjacent filters.

- **RATIO**

This indicates the amount of space between two bands. At maximum ratio in Harmonic mode, the spacing is one octave between bands. A ratio of 0 indicates that all the bands have the same value, whatever the mode is.

3.2.3 Distortion

To really get things buzzing, an optional distortion can be added after the filter stage. This comes in two distinct flavours, Classic Overdrive and Puncher, with the option of morphing between the two. The latter adds a lot of low harmonics, excellent for fattening up the puniest of sounds into a ravening audio monster.

The distortion stage is always followed by a dynamic limiter, reducing the louder peaks caused by extremely high resonance. The threshold is set to about +6 dB RMS, so you have a lot of headroom before limiting.

- **GAIN**

This sets the amount of distortion you want to apply. It works like a volume control, amplifying the signal. Then it is clipped. The clipping threshold depends on the gain. It is high at low gain to let high dynamic signals pass without distorting them too much. As the gain increases, the threshold decreases to keep the sound at a reasonable volume.

- **SHAPE**

This knob morphs the clipping curve between the Overdrive and the Puncher. Overdrive is a classic soft-clipping algorithm, producing rich harmonics and high frequency content. The sound remains clear and recognizable. Puncher is a “fold-back” shaper, giving fewer harmonic at low gain, and probably a warmer sound, especially for bass. High gain generates a lot of harmonics, turning into noise if pushed to extreme limits.

- **FATNESS**

Usually distortion makes the sound a lot louder, reducing the dynamics and the punch of the attacks. This is not always desirable, so we included this fatness control. Turned to the right, it gives full-distorted sound. To the left, the volume is adapted from the non-distorted sound.

3.2.4 Mix and activity

In this part of the panel you can control and monitor the band activity, and choose how it is mixed with the final output.

- **VOLUME**

This controls the amount of band signal injected into the final effect output. It is important to note that this volume does not affect the band chaining. For example, if bands 1 and 2 are chained in serial, band 1 volume will not have effect on the amount of signal feeding band 2.

- **PAN**

Select the panoramic stereo location of the band output in the final mix. Like the **VOLUME** setting, this button has no influence on the signal transiting between bands.
☞ **Note:** the pan parameter behaves as a balance in stereo mode.

- **POWER**

Switches the band on or off. When the band is deactivated, it feeds nothing to the final mix. It acts like a bypass on serially connected bands. For example, if band 1, 2 and 3 are serially linked, and band 2 is switched off, the processing becomes: band 1 followed by band 3. The connected probes are also disabled. For information about probes, see the Modulation section.

- **VU-METER**

This monitors the band activity. The vu-meter analyses the pre-mix signal, meaning that the **VOLUME** setting does not influence it.

3.3 Modulations

The Quad Frohmage come packed with modulation effects. Nearly every parameter has it's own modulation set, and without them, the sound would remain somewhat static.

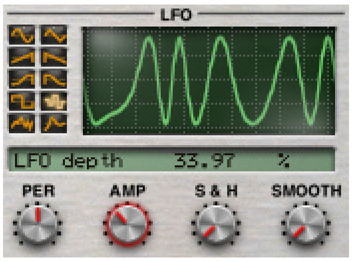
The Quad Frohmage modulation set include a LFO, an envelope follower and a triggered ADSR. The modulation controls are located next to the parameter value display zone, and, in a similar way, are displayed in a contextual way: when a parameter has been selected, the corresponding modulation controls are activated. If the parameter cannot be modulated, for example the filter type, the knobs are hidden from the panel.

The three modulation generators are streamed together to generate the final modulation value, which modifies the parameter. You can monitor the different modulations and the final value on the oscilloscope screens. You can also quickly check if a parameter is modulated by looking at the small green LED at the bottom-left of each knob or fader. If it is lit, the parameter is currently being modulated.

☞ **Tip**: If the oscilloscope display looks jagged, try to decrease the buffer size in the audio settings of your host application. Generally a buffer size of 1024 samples at 44.1 kHz sample rate is enough to get a smooth display.

3.3.1 LFO

Most of the Ohm Force plugins come with one kind of modulation: the LFO, which stands for Low Frequency Oscillator. It is an oscillator producing a signal usually below the audio frequency range. This signal additively modulates the parameter, which it is associated with. It makes the parameter oscillate around its central value.



- **Period**

This is the time taken by an LFO oscillation. LFOs are synchronized to the tempo value to allow them to stay in sync with the music.

- **Amplitude**

This is the amplitude of the oscillations. 0 % means that the LFO does not affect the sound.

- **Waveform**

This parameter affects the shape of the oscillations. Seven of the shapes are classic, the three others are random oscillations.

	SINE	It is the default waveform.
	TRIANGLE	Oscillation goes and returns linearly between the two extreme points.
	SQUARE	LFO stays a half-period on the maximum point, then the other half-period on the minimum point.
	RAMP UP	Oscillation goes linearly from the minimum point minimum to the maximum one.
	RAMP DOWN	Like Ramp up, but in the other direction.
	COS UP	A bit like Ramp up, but LFO go and arrive more gently at the extreme points (a kind of shelf).
	COS DOWN	Like Cos up, but in the other direction.
	RANDOM	This waveform is a chaotic oscillator. It is based on a sine wave, sometimes slowing down, sometimes accelerating, or even going back and forth. Anyway, whatever these random variations, it always keeps the same average phase. This waveform is ideal for organic, live and human beating.
	BROWN NOISE	LFO value changes randomly, combining wide but slow moves with small and fast oscillations. With a very long period, this kind of LFO is perfect to give a parameter a natural and nervous random variation.
	RED NOISE	This is a bit like Brown Noise, but fast variations are more damped, giving even smoother random walks.

- **SAMPLE AND HOLD**

This function slices the waveform periodically and holds the LFO value on each slice, making steps. The knob controls the ratio of the S&H period over the main LFO period. Thus 0% disables it, and 25% samples the LFO each quarter period.

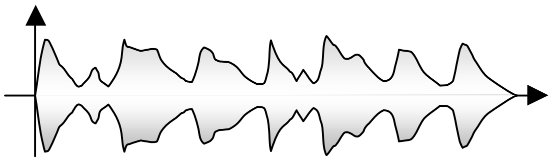
- **SMOOTH**

This knob allows you to smooth fast transitions of an LFO signal; it occurs after the Sample And Hold. The smoothing factor depends on the main LFO period; therefore the final shape does not change when the period is changed. Again, 0% disables the smoothing and leaves the waveform untouched.

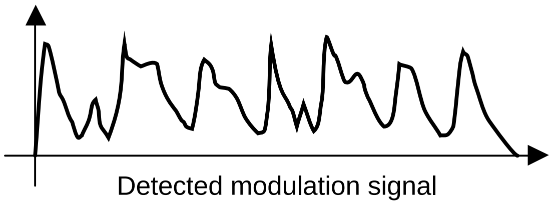
☞ **Tip**: The phase of each LFO can be reset and synchronized with the help of MIDI NRPN commands. See the MIDI Factory Settings section for more information. It is also automatically reset to 0° when the LFO amplitude is null.

3.3.2 Envelope follower

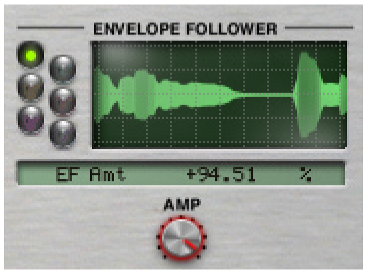
This modulator uses the audio signal energy to produce a slowly varying modulation signal. It is different from the LFO because it does not work on his own. It needs to analyse the sound energy varying over the time – the envelope – to generate the modulation signal. To be clear, the audio signal is made of very fast oscillations whose amplitude varies with the time. When you zoom in sound waveform view, you can see these oscillations. When you zoom out, these oscillations become closer and closer, to finally form a vague shape outlined by the oscillation amplitude. This is what the envelope follower outputs.



Incoming audio signal



Detected modulation signal

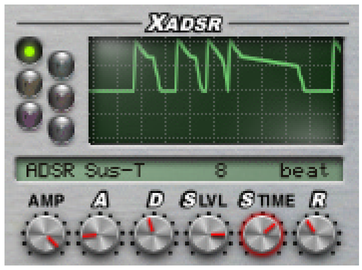


Use one of the six colored buttons to select the probe analyzing the signal, i.e. which point of the sound path you wish to follow. See the Probe section for more information.

- **AMP**

This is the amount of envelope follower output, which is added to the final modulation.

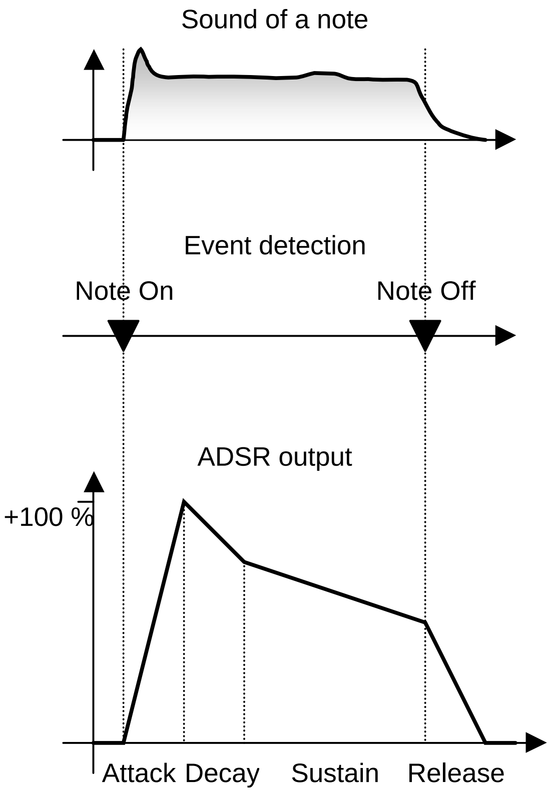
3.3.3 ADSR envelope generator

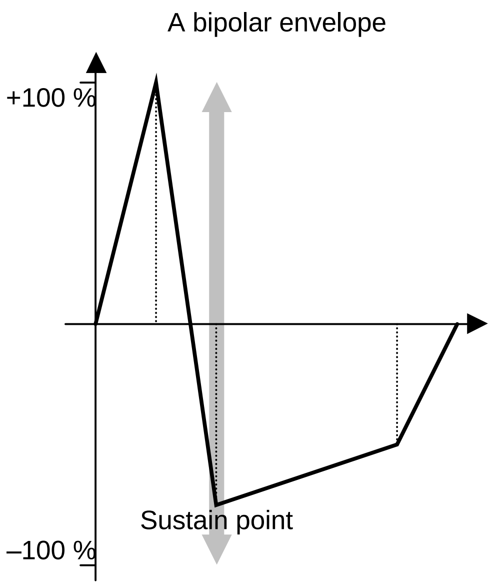


Like the envelope follower, ADSR uses an audio signal as input. Here, the sound energy is analysed and large variations produce "Note On" and "Note Off" events. The purpose of this modulation is to detect the note beginning and the note end from a musical source.

The Note On event trigs the ADSR envelope, starting to generate the modulation signal. The ADSR envelope is a piecewise segmented curve, made of four parts: A – Attack, D – Decay, S – Sustain and R – Release.

Select the probe analyzing the signal with one of the six colored buttons. It determines the part of the sound path trigging the ADSR. See the Probe section for more information.





- **AMP**

The amount of ADSR output which is added to the final modulation.

- **A**

Duration of the Attack stage. This is the time it takes to reach the maximum envelope amplitude. Attack begins at the detection of a Note On event.

- **D**

Duration of the Decay stage, during which the envelope goes to the sustain point. Decay occurs right after the Attack.

- **S-LEVEL**

Level of the sustain point. Sustain level can be negative.

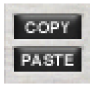
- **S-TIME**

Time taken for the envelope to reach 0, after the Decay stage, starting from the sustain point.

- **R**

This is the duration of the release stage, during which the envelope fades out. The Release stage occurs as soon as the Note Off event is detected.

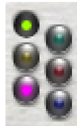
3.3.4 Copy and paste



Use the **COPY** button to copy the whole modulation settings of the currently selected parameter into the internal clipboard. Use **PASTE** to apply them on another parameter. This is very useful to quickly duplicate the modulation settings on a large range of parameters and mutate them afterwards.

3.3.5 Probes

The envelope follower and ADSR triggers use the audio signal energy to do their job. They collect this energy via the “probes”, sensors plugged into the circuit. There are 6 different probes. Each probe can provide information for both envelope follower and ADSR trigger.



The probe configuration is located in the Probe And Routing panel, at the top of the plug-in. After having opened it with the **EXPERT** pushbutton, you'll find six colored buttons, similar to the six close to the Envelope Follower and ADSR oscilloscopes. Push one of them to configure the associated probe. The buttons are also lit up if the corresponding probe is used by at least one EF or ADSR in the circuit.

- **SOURCE**

This parameter specifies where each probe picks up the signal in the circuit.

INPUT The probe is connected at the effect input. Signal is converted to mono before analyzing, even in Stereo Input mode.

OUTPUT This is the effect output. As with Input, sound is converted to mono before the analysis. The signal is taken before the Master Volume.

BAND 1
BAND 2
BAND 3
BAND 4
The signal is taken after the band N filter.

☞ **Warning**: the band deactivation also switches the probe off. So if the band is Off, the connected probes will not output anything.

MIDI PITCH It is possible to use a MIDI keyboard instead of the audio signal to trig the ADSR envelope. Each time you press a key, it generates a Note On event for the ADSR. The Note Off event is sent when the last key is depressed. The envelope follower is replaced by the value of the highest key pressed. The modulation is bipolar here, meaning that notes above Middle C give positive modulation, whereas notes bellow are give negative control signal.

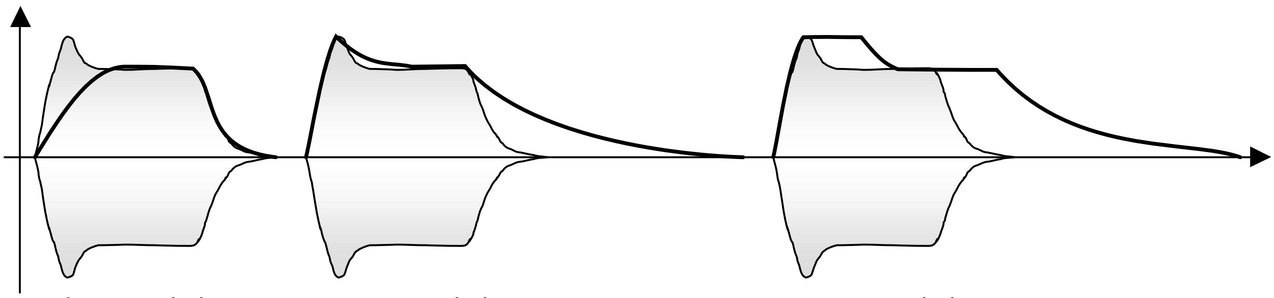
MIDI VELOCITY This works like MIDI Pitch mode, but the MIDI velocity of the last key pressed replaces the envelope follower. The ADSR amount is also controlled by the velocity value.

It is a matter of convention, here at Ohm Force, that we often assign the two bottom probes to MIDI sources. The violet one uses MIDI Pitch, whereas the blue one is assigned to MIDI Velocity. Most of the presets bundled with Quad Frohmage use at least one of these probes, so playing with a MIDI keyboard connected to the effect brings instant gratification. If you intends to publish your presets on our website, we advice you to use the same convention. You can use the `neutral.pbk` preset bank as starting point.

3.3.5.1 Envelope follower configuration

The sound energy detection is not as trivial as it could seem. It needs some parameters to control the evolution of the envelope. The two main parameters are the Attack time and the Release time. They indicate how quickly the envelope follows the energy curve changes. When the envelope is below the current energy level – because of an energy peak for example, it is an Attack section. Release sections are the opposite.

There can be many small attack and release sections during a single sound. To avoid their proliferation, one can block the envelope at the top of an attack peak to prevent or limit unwanted release sections. This is the function of the Hold section. If set long enough, it can also be used to make a plateau on the curve, which will change rhythmically.



- **ATTACK**

Sets the Attack Time. Short times are better suited for processing percussive sounds, because the envelope does not have the time to rise before the Release/Hold section. This is the only parameter used when the source is MIDI. It just smoothes the transitions between the different states.

- **HOLD**

Sets the Hold time. This parameter has no effect when the event source is MIDI.

- **RELEASE**

Sets the Release time. This parameter has no effect when the event source is MIDI.

3.3.5.2 ADSR triggering configuration

When working with audio signals, ADSR triggering is quite difficult to set up. There is no universal solution for detecting note beginnings and ends, because they are extremely dependent on the nature of the instrument. Our detector has two parameters which have to be set carefully to get the desired results.

- **SENSITIVITY**

This knob helps to adjust the detector sensitivity to the average level of the sound. Lower the threshold if you don't get enough On or Off events, and raise it if you have way too many of them.

- **DENSITY**

This parameter limits the numbers of detected events. Below to 25%, Note Off events aren't triggered anymore. Beware, the lowest values tends to delay the event generation.

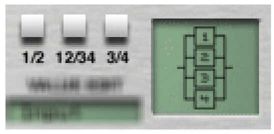
3.4 Miscellaneous controls

3.4.1 Routing

- **EXPERT MODE**

Press this button to reveal or hide the Probe And Routing panel. Here you can control the probe settings, and how the signal is routed in the Quad Frohmage.

- **ROUTING**



Push these buttons to change the band chaining: serial or parallel. **1/2** and **3/4** select the chaining between band 1 and 2 and band 3 and 4, whereas **12/34** selects how the band 1 and 2 block is chained with the band 3 and 4 block.

- **STEREO IN**

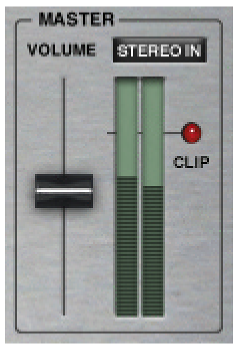
This button, located in the Master panel, activates the stereo input processing. When a stereo signal enters the effect, it is possible to mix both channels and to process the sound. This is the default. If Stereo In is activated, the channels are sent to different bands if possible.

- **BAND INPUT**

Look at the display on the top of each band. They indicate what signal is feeding it. It’s useful to know if a given band processes a mixed signal (“L+R Input”) or each channel individually.

3.4.2 Other settings

- **MASTER VOLUME**



Controls the master volume, result of the band mix. It does not influence output probes. The vu-meter monitors the output peak level. The clip LED lights up when the signal jumps over 0 dB. It does not mean it actually clips – most plug-in standards support floating point data format and volume can still be lowered on the host’s mixing console.

- **NOTE / HZ MODE**

This mode controls the display of the frequencies. When set to Note mode, the corresponding note is displayed instead of a frequency in Hertz. Also the knobs are notched to match notes accurately.

3.5 MIDI factory settings

3.5.1 NRPN, what the hell is this?

It’s a way to send 14 bit data to a MIDI device. There are 16384 possible NRPN, each one corresponding to a 14 bit value (i.e. a value between 0 and 16383). This system uses 4 pre-defined CC. You have to send to the MIDI device the NRPN number, and then the NRPN value.

CC 99 and 98 are used to send the NRPN number. The value conveyed by CC 99 represent the 7 first bits of the value, the one being conveyed by CC 98 the 7 last bits of the NRPN number. Then, you use CC 6 and 38 to send the required value. If you want to only send a 7 bits value, simply send the CC 6.

So, for example, to send NRPN #831 value 257, you have to send :

```
CC #99, value 6
CC #98, value 63
CC #06, value 2
CC #38, value 1
```

☞ **Note**: You don’t have to send the NRPN number each time, as the Symptohm remembers the current NRPN number selected.
If we go on with the example above, if you want to set NRPN #831 to value 0 and NRPN #832 to value 127, you would have to send the following :

```
CC #06, value 0
CC #38, value 0
```

Now, NRPN #831 is 0.

```
CC #98, value 64
```

Now, the NRPN number is 832. You don’t have to send the MSB number if it did not change.

```
CC #38, value 127
```

As we already sent [CC #06, value 0], we don’t need to send it again

3.5.2 Modulation sub-parameters

Most of the Quad Frohmage parameters come with modulation capabilities. These capabilities are manifested in sub-parameters accompanying the main parameter. The NRPNs used to control these sub-parameters are always layered in the same order, following the main parameter NRPN.

It would be needlessly dull to list all parameters with their full modulations, so, instead, the names of modulated parameters will just be followed by an asterisk (*). Use the table below to find the right modulation NRPN from the relevant main parameter one.

NRPN	Sub-parameter
N	Main modulated parameter
N + 1	LFO Period
N + 2	LFO Depth
N + 3	LFO Waveform
N + 4	LFO Sample And Hold
N + 5	LFO Smoothing
N + 6	ADSR Source Probe
N + 7	ADSR Amount
N + 8	ADSR Attack Time
N + 9	ADSR Decay Time
N + 10	ADSR Sustain Time
N + 11	ADSR Sustain Level
N + 12	ADSR Release Time
N + 13	Envelope Follower Source Probe
N + 14	Envelope Follower Amount

3.5.3 Generic parameters

NRPN	MSB-CC99	LSB-CC98	Parameter
0	0	0	Tempo
1	0	1	Master Volume
2	0	2	Band Routing 1–2 / 3–4
3	0	3	Band Routing 1 / 2
4	0	4	Band Routing 3 / 4
5	0	5	Stereo Input Routing Mode
166	1	38	Delay Link
227	1	99	Frequency Link
228	1	100	Frequency Link Mode
229	1	101	Frequency Link Ratio *

3.5.4 Bands

The numbers inside the parenthesis represent the MSB and LSB values corresponding to the NRPN number. The MSB value is sent via CC #99, and the LSB value using CC #98

NRPN for band...				Parameter
1	2	3	4	
6 (0+6)	37 (0+37)	68 (0+68)	99 (0+99)	Power
7 (0+7)	38 (0+38)	69 (0+69)	100 (0+100)	Gain *
22 (0+22)	53 (0+53)	84 (0+84)	115 (0+115)	Pan *
167 (1+39)	182 (1+54)	197 (1+69)	212 (1+84)	Delay Time *
244 (1+116)	290 (2+34)	336 (2+80)	382 (2+126)	Filter Type
245 (1+117)	291 (2+35)	337 (2+81)	383 (2+127)	Filter Cutoff Frequency *
260 (2+4)	306 (2+50)	352 (2+96)	398 (3+18)	Filter Fatness *
275 (2+19)	321 (2+65)	367 (2+111)	413 (3+29)	Filter Resonance *
428 (3+44)	473 (3+89)	518 (4+4)	563 (4+51)	Distortion Type *
443 (3+59)	488 (3+104)	533 (4+21)	578 (4+66)	Distortion Level *
458 (3+74)	503 (3+119)	548 (4+36)	593 (4+81)	Distortion Fatness *

3.5.5 Probes

NRPN for probe...						Parameter
1	**2**	**3**	**4**	**5**	**6**	
130	136	142	148	154	160	Signal Source
131	137	143	149	155	161	EF Attack Time
132	138	144	150	156	162	EF Hold Time
133	139	145	151	157	163	EF Release Time
134	140	146	152	158	164	ADSR Sensitivity
135	141	147	153	159	165	ADSR Density

Here, the MSB is always 1. NRPN #130 corresponds to LSB value 2, #131 to 3… #165 to 34.

3.5.6 LFO phases

With the following NRPN, you can set the current phase for each LFO. 0 sets a phase of 0°, and 127 (or 16383 if you use 14-bit NRPN values) is about 360°.

NRPN	MSB-CC99	LSB-CC98	Parameter
608	4	96	Band 1 Gain
609	4	97	Band 1 Pan
610	4	98	Band 2 Gain
611	4	99	Band 2 Pan
612	4	100	Band 3 Gain
613	4	101	Band 3 Pan
614	4	102	Band 4 Gain
615	4	103	Band 4 Pan
616	4	104	Band 1 Delay Time
617	4	105	Band 2 Delay Time
618	4	106	Band 3 Delay Time
619	4	107	Band 4 Delay Time
620	4	108	Frequency Link Ratio
621	4	109	Band 1 Filter Cutoff Frequency
622	4	110	Band 1 Filter Fatness
623	4	111	Band 1 Filter Resonance
624	4	112	Band 2 Filter Cutoff Frequency
625	4	113	Band 2 Filter Fatness
626	4	114	Band 2 Filter Resonance
627	4	115	Band 3 Filter Cutoff Frequency
628	4	116	Band 3 Filter Fatness
629	4	117	Band 3 Filter Resonance
630	4	118	Band 4 Filter Cutoff Frequency
631	4	119	Band 4 Filter Fatness
632	4	120	Band 4 Filter Resonance
633	4	121	Band 1 Distortion Type
634	4	122	Band 1 Distortion Level
635	4	123	Band 1 Distortion Fatness
636	4	124	Band 2 Distortion Type
637	4	125	Band 2 Distortion Level
638	4	126	Band 2 Distortion Fatness
639	4	127	Band 3 Distortion Type
640	5	0	Band 3 Distortion Level
641	5	1	Band 3 Distortion Fatness
642	5	2	Band 4 Distortion Type
643	5	3	Band 4 Distortion Level
644	5	4	Band 4 Distortion Fatness