

Fault 1.0 Manual

by unfilteredaudio

The screenshot displays the 'Fault 1.0' software interface, titled 'STEREO SPECTRAL SHIFTER'. The interface is organized into several functional sections:

- INPUT:** A 'Mono' button and a volume knob set to 0 dB.
- FREQUENCY SHIFTER:** A large circular frequency display showing B: 62.500 Hz, L: 62.500 Hz, and R: 62.500 Hz. Below it are knobs for MULT (x1.0), PAN, and SPREAD (x1.0). At the bottom are FM FREQ (x1.0) and FM DEPTH (x0.0) knobs, with a 'Lin. FM' button.
- FEEDBACK MATRIX:** Controls for SHIFTER and DELAY for both L and R channels. It includes 'Link' buttons for SHIFTER and DELAY, and a 'Stabilize' button.
- DELAY:** L and R delay knobs set to 1/8th, with 'Link' and 'Sync' buttons.
- FEEDBACK FILTER:** LP and HP filter knobs, with a 'Filter All' button.
- SPECTRUM:** SAMP RT knob set to 100.0%, and buttons for 'Soft Clip', 'Anfi-Alias', and 'Ring Mode'.
- OUTPUT:** A volume knob set to 0 dB, a 'DRY/WET' knob, and an 'M/S' button.
- WIDTH:** A knob at the bottom right.

At the bottom of the interface, there are 'MODULATION' and 'MUTE ALL' buttons. The top right corner features the 'unfilteredaudio' logo and a 'Default' button with settings icons.



Introduction

Fault is a Stereo Spectral Shifter, a new kind of effect that combines pitch shifting, frequency shifting, and more into an exciting plug-in designed to warp your audio in incredible ways.

Features

- Six completely customizable frequency shift feedback paths allow for unique combinations available in no other plugin.
- Channel-independent frequency and pitch shifting engines. Use pitch shifting for smooth, harmonic shifts, or frequency shifting for aggressive, metallic timbres.
- Internal Complex Linear and Exponential Thru-zero FM synthesis takes the frequency shifter into unheard territories.
- Multiple spread controls create deep possibilities for stereo processing.
- Two delay lines with independent time controls.
- UA's new flagship modulation system featuring a direct cable-patching interface, six modulation types (LFO's, input follower, sample and hold, etc.), fully compatible with host automation and tempo sync, and featuring a real-time visualization of each modulated parameter value. Modulation parameters can themselves be modulated, creating emergent systems with limitless complexity.
- "Anti-Alias" mode to remove foldover frequencies in the frequency shift engine.
- "Ring Mode" turns Fault into a complex, stereo ring modulator.
- Variable Sampling Rate control over the entire DSP. Don't just crush your audio, crush your entire plug-in.
- Wet/Dry mix for parallel processing.
- Spatial width control, featuring the ability to use Fault as an M/S encoder.

Getting Started

Installing Fault

Simply run the provided installer. On Windows, you will need to select your VST directory if the installer doesn't pick the right one by default.

Browsing Presets

Fault comes with a wide array of presets to get you started. The preset manager on the top-right of Fault is used to load, save, or browse presets. If you want to get a good feel for the power of Fault, spend some time with these presets before browsing the manual.

Standard Unfiltered Audio Features

CONTROLS

- All knobs and vertical sliders are controlled in a smooth up-and-down motion.
- All horizontal sliders are controlled in a smooth left-to-right motion.
- Use the Command key (on OS X) or Control key (on Windows or OS X) to fine-tune controls.
- Double-click or Alt-click on a control to return it to a default setting.

PRESETS

- All presets are saved with a .uap file extension. These presets are compatible across all platforms and plug-in formats. They use a simple, non-encrypted XML format for easy editing.
- Use the "Lock" menu to prevent specific parameters from changing while browsing presets.

Input/Output Configurations

1 IN/1 OUT (MONO)

In this configuration, Fault takes in a mono input and creates a mono output. Many of Fault's controls exhibit interesting stereo behavior, so this mode isn't recommended for most cases. In this mode, Fault processes the mono input internally as a stereo sample. The left and right stereo outputs are then averaged together to create a mono output.

1 IN/2 OUT (STEREO UPMIXER)

This mode is available in hosts like Logic Pro X that support it. In this mode, Fault takes in a mono sample and produces a stereo output. The mono sample is used as a dual mono input for the stereo processing callback.

2 IN/2 OUT (TRUE STEREO)

Fault's native mode. In this mode, the left input runs through the left processing controls (pitch shift, frequency shift left, delay left, etc.), while the right input runs through the right controls. If you want both inputs to run through both channels, enable the "Mono" button above the input gain control.

Controls

Input



In Gain: Controls the amount of gain applied to the incoming audio signal. This gain only affects the Wet signal. Fault has separate input and output gain controls due to the number of Feedback controls present.

Mono: When this button is active (highlighted red), the incoming stereo audio signal will be converted to mono by summing the two channels together and averaging them. This can be useful if you have a signal with uneven balance and want it to be evenly affected by Fault's controls.

Pitch Shifter



L/R Shift: Controls the amount of pitch shifting applied to each channel. The value is listed in semitones. The amount of possible pitch shifting is +/- 1 octave.

Link: When enabled, the Left Pitch Shift amount is copied to the Right channel. You can disable this to create unusual stereo pitch shift effects.

Quantize: When enabled, the pitch shifter will only lock onto whole semitone values. This is very useful for harmonic, tuned material. Turn this off for deeper processing of percussive signals.

Frequency Shifter



Shift Amount: The large knob in the top middle controls the overall frequency shift. This shift amount is affected by the Mult, Spread, and Sampling Rate parameters. This knob will display the Base (“B”) shift amount, which is the frequency shift selected by this knob (-/+ 500 Hz). It will also display the Left (“L”) and Right (“R”) shift amounts, which are the Base shift amount modified by the other parameters.

Mult: This control multiplies the Shift Amount 0-10x, extending the maximum shift range of Fault to +/- 5,000 Hz. Instead of going for a toggled Low/High Range mode like other shifting plug-ins, this control allows you to set your own shifting range for the large knob. This is especially useful for Fault due to the complexity of the modulation system.

Pan: This control determines how much of each Shifter + Delay channel is heard at the final output. You can use this control to smoothly crossfade between two completely different channel settings (especially useful in tandem with the spatial Width control).

Spread: This control determines the frequency of the Right Shifter channel as a multiple of the Left Shifter’s frequency. At small Spread amounts (0.99), this will create a super-wide stereo phasing effect. At more dramatic settings, this will create extremely unusual timbre and stereo effects, especially when combined with the X-Feedback controls.

FM Freq: Fault contains an internal complex oscillator that is used for creating the frequency shifting effect. There is an additional modulation oscillator that can modulate the frequency of

the shift. This control determines the frequency of the modulation oscillator. This frequency is determined as a multiple of the base frequency.

FM Depth: This control determines the amount of FM that is applied to the complex shifting oscillator. This depth is determined as a multiple of the base frequency.

Lin. FM: When active, a linear FM algorithm will be used to determine the shifter's frequency. When inactive, an exponential FM algorithm will be used. Without diving too much into technical detail, this switch is used to change the timbre of the FM in an interesting way.

*Note regarding FM: An easy way to think of Fault's internal FM synthesis is to use one of Fault's sine wave modulators to modulate the large shift knob. The internal FM oscillator does the exact same thing, but with greater resolution, a larger modulation range, and an algorithm that is better tuned to a great sounding FM response.

Feedback Matrix



This section is perhaps the signature feature of Fault, as it gives you control over six separate feedback paths!

The six controls provide three paths each for the Shifter and Delay sections. Shifter feedback is single-sample, meaning that the shifter's output is immediately added to its next input. The delays' feedback time is determined by the delays' lengths. As a result, you may notice that the shifter feedback can sound different if you change your DAW's sampling rate.

The three controls are identical for each section.

L: Controls the amount of feedback on the left channel.

R: Controls the amount of feedback on the right channel.

X: Controls the amount of cross-feedback. This is the amount of feedback sent from the right channel to the left, and the left channel to the right. This can get out of hand very quickly, so smaller amounts are recommended.

*Great technique: Use cross-feedback between the two delay lines, and set the delays to different lengths. This will give you huge stereo echoes.

Stabilize: This algorithm attempts to prevent runaway feedback oscillations. This will calculate the total feedback between all controls. If the amount of feedback is greater than 99%, it will reduce the level of all feedback controls (but keep them set to the same amplitude ratios). If you

are using a lot of additional input/output gain and heavy modulation, the algorithm may fail to prevent all oscillations.

*Great technique: When Stabilize is turned off, Fault can be used as a standalone texture generator. Modulate the shifter frequency and delay times to create unusual synth tones and bird sounds.

Link: When active, the amount of feedback on the left channel will be copied to the right channel.

Delay



Delay L/R: There are two delays in Fault, one for each channel. These knobs will change the length of each delay.

Link: When active, the length of the left delay will be copied to the right delay.

Sync: When active, both delays will be synced to the host tempo. Instead of being set in seconds, the delay lengths will be in musical intervals (eighth notes, quarter notes, etc.).

Feedback Filter



This section provides two filters that are run serially. These filters are placed inside the feedback paths of the Shifters and Delays. These can be used to tame feedback, prevent harshness, or modify timbres.

LP: Controls the cutoff of a low-pass filter.

HP: Controls the cutoff of a high-pass filter.

Filter All: When active, the filters will be used to filter the entire output signal, and not just the feedback portion.

*Note: The HP filter's lowest cutoff is 20 Hz. This means that DC offset will be filtered out of the feedback path and final signal. This was implemented to prevent build-up of sub-bass frequencies in the feedback path as well.

Spectrum



This section provides timbral manipulation over the entire plug-in.

SAMP RT: A signature feature of Unfiltered Audio's plug-ins. If you've used Sandman, then you are familiar with this. This control determines the sampling rate of the *entire* plug-in, not just the input. Changing the sampling rate will affect the shifter frequency, the length of the delay units (when not set to Tempo Sync), the behavior of the filters, and more. Needless to say, entire worlds of unusual sounds and artifacts are contained within this knob!

Soft Clip: When active, extremely loud sounds will be saturated instead of clipped. This provides a pleasant, analog-sounding grit instead of harsh, digital distortion.

Anti-Alias: When active, an interesting filtering method will be applied to the frequency shifter. This filtering will remove fold-over aliasing from the shifted signal. This doesn't have too drastic of an effect on signals shifted positively, but it dramatically changes the spectrum of signals shifted downwards.

If you've used other frequency shifters, many of them have the tendency to sound like they are shifting upwards, despite the frequency shift control being set to negative frequencies. This is because of fold-over aliasing. What happens is that when a harmonic component of a signal is shifted below 0 Hz, it "bounces" off of 0 Hz and starts shifting upwards. With this mode enabled, that harmonic component is simply suppressed once it reaches 0 Hz. The overall effect is a downward frequency shift with elements of high-pass filtering, as the lower harmonic components of the signal start to disappear.

Ring Mode: In this mode, Fault becomes a dual ring modulator. To do true ring modulation, you will need to turn off the Anti-Alias mode. At low frequencies, this will turn Fault into a stereo

tremolo device (try it with the Spread control and light FM!). At high frequencies, it creates metallic sounds and textures.

Output



Out Gain: Controls the amplitude of the Wet output.

Dry/Wet: Controls the balance between the dry input signal and the wet output signal. Unlike many effect plug-ins, many of Fault's most interesting sounds can be found with a 50/50 balance (due to phase cancellation effects).

M/S: When active, Fault will output in Mid/Side mode. This can be used for advanced mixing and processing techniques. If you are unfamiliar with Mid/Side processing, the simplest explanation is that "Mid" can be thought of as the mono or "shared" component of the two channels, while "Side" can be thought of as the stereo or unique components of the two channels. In M/S mode, the first channel output will be Mid, and the second channel output will be Side. You can output M/S signals onto a L/R configuration without negative side effects. It will definitely sound unusual!

Width: Controls the perceived width of the output signal. In L/R mode, 0% is mono, 50% (12 o'clock) is stereo, and 100% is super-wide (Note that components of the left and right signals will bleed over into each other). In M/S mode, 0% is mid-only, and 100% is side-only.

Modulation System

Introduction

Fault utilizes Unfiltered Audio's flagship modulation system, with which you can quickly patch anything from simple parameter changes to complex and even self-evolving systems. Starting off is very easy due to the direct cable-routing user interface.

The first step is to show the modulation manager by clicking the "Modulation" button in the bottom corner of the interface. You'll see the plugin interface get taller, revealing modulation sources, outputs, and inputs on each modulatable plugin parameter.

Patching

Input and Output Ports

Outputs and inputs are both denoted by a simple circle. To avoid confusion, just remember that outputs are always located on the top left of each modulator. Everything else is an input.

Output ports have a small circular scaling knob below them which can be used to alter its calculated output value. This knob is set to 1.0 by default (fully clockwise), but can be turned down to 0.0 (center) to zero out its values entirely, or -1.0 (fully counter clockwise) to invert them.

Creating Connections

To create your first modulation "patch" simply click and drag on one of the outputs. You'll see a cable appear next to your mouse, the end of which you can drag to any input on the interface. When you get close enough you'll see the cable "lock" into place, showing you that the path is valid. Once you release the mouse, the connection completes and you'll see the modulation start to animate in real-time.

You can create a second connection from the same output source by clicking on it again. Likewise, try clicking and dragging from an input to an output, which works in the same fashion.

Moving or Removing Connections

In order to move a connection, click on one of the thicker ends of the cable, near the input or output port. You'll see the cable lock onto your mouse, allowing you to move it to a different input/output source. If you release the mouse while the cable is not "locked" onto any port, it will be removed entirely, which is how connections are deleted.

Inter-Modulation

As you might have noticed, all of the knobs on each modulator have input ports of their own, capable of receiving modulation just like parameters on the main section of the interface. Any output port can be connected to these inputs, meaning that a modulator can even modulate its

own parameters! Experimenting with complex inter-modulation chaining can lead to dynamically evolving behavior, and even emergent systems.

Automation

Our modulation system provides 16 automation “slots” that can be assigned to one or more modulation parameters. Every knob and button on a modulator can be assigned to one of these automation slots by right clicking on it and following the “Add X to Automation Slot...” menu. Once assigned to a slot, this element will mirror the automated value sent in through your DAW. You’ll also see a small number appear next to the element, denoting the automation slot number to which it is assigned.

Adding Modulators

You can add a new modulator by clicking the “+” button in the modulation header. At this time a maximum of 6 modulators can be used at once, although that is a somewhat arbitrary maximum that we plan to increase in the future.

Removing and Duplicating Modulators

Modulators can be removed or duplicated by right clicking on them and then selecting the appropriate option.

Mute Modulation

The “Mute All” button in the modulation header allows you to quickly mute and un-mute all modulation sources.

Saving and Loading

All modulation routing, values, automation, and general state is saved with the preset and/or with the DAW’s session. Presets are a great way to quickly share interesting modulation patches with other users.

Modulator Types

You can change a modulator’s type by clicking the dropdown menu next to its symbol. There are currently six types to choose from, each with its own characteristics and use case. Common parameters such as frequency will remain at their selected value, making the process of auditioning different modulator types both quick and easy.

LFOs: Sine/Saw/Tri/Square

The LFO family of modulators all use common primitive wave types which are useful for classic “envelope” style parameter modulation.

The Amplitude knob is common to all of them, corresponding to the depth of modulation. This can be either Bipolar or Unipolar which is determined by the state of the +/- button in the modulator’s bottom left corner. Bipolar modulation always fluctuates between negative and positive, whereas unipolar modulation is always positive when the amplitude is above 0% and always negative when it is below 0%.

The Frequency knob corresponds to the speed of modulation. It is set in Hz (cycles per second) when in non-tempo-synced mode. Tempo Sync can be activated by clicking the “Hz” button on the left side of the modulator, which will turn into a musical note symbol indicating that it has been initiated. In tempo sync mode, modulation speed is synchronized to divisions of the DAWs global tempo setting- for instance setting the frequency to $\frac{1}{4}$ will correspond to quarter-note modulation cycles.

The Saw/Tri modulator has an additional Shape knob which determines the slope of the triangular wave. At the default value of 0.5, you get a normal triangle wave. At 0.0, you get a downward sloping triangle wave and at 1.0 you get an upward sloping triangle.

The Square modulator has a Width knob which changes the pulse width of the square wave. At the default value of 0.5, you get a normal half-up, half-down square wave. As you approach 0.0, the wave will be almost entirely down, and conversely as you approach 1.0, the wave will be almost entirely up.

Input Follower

The input follower modulator allows you to easily “sidechain” parameters to the incoming audio stream. The Smooth knob determines the speed of the rms meter used to track the incoming audio. Lower values correspond to a more instantaneous reading, while higher values smooth everything out.

Macro Control

The macro modulator can act as a manual control knob capable of outputting values to multiple parameters. In this way you can control many knobs with a single gesture. Alternatively, you can patch many modulation sources into the amplitude knob’s input port, turning the macro control into a summing modulation bus.

Sample and Hold Noise

The sample and hold noise modulator chooses random values at the chosen frequency rate. The Slew knob controls the amount of interpolation time whenever a new value is set. At a slew of 0.0 values change instantly, whereas at 1.0 they interpolate over the entire cycle period. Tempo Sync and Uni/Bipolar can be set on this modulator just like LFOs (see above).

Recipes and Ideas

Wide Stereo Tremolo

- Turn on “Ring Mode”.
- Make sure “Anti-Alias” is disabled.
- Set the main shifter knob to a low frequency (.5-10 Hz or so).
- For extra fun, use the Spread control and turn up the FM depth.

Ping-Pong Echo Chamber

- Turn off the “Link” control between the two delays.
- Set the delays to slightly different values. Tempo Sync helps here. Try 1/4 and 1/8.
- Turn off regular delay feedback. Start to increase the X Feedback parameter.

Stepped Pitch Shifter

- Add multiple, tempo-synced Square LFOs.
- Set the LFOs to metric multiplications and divisions of each other. For example, try 1/4, 1/8, and 1/16.
- Turn on Pitch Shifter quantization and link.
- Connect all three LFOs to the Pitch Shifter L modulation input.
- Adjust the levels of the LFOs until a pleasant sequence is found.

Credits and Thanks

Fault is written by Joshua Dickinson and Michael Hetrick.

We would like to thank Lance Putnam for his excellent Gamma DSP library, Andres Cabrera, Curtis Roads, and Matthew Wright for being excellent teachers, and Yon Visell for his assistance with anti-aliasing filtering.

<http://www.unfilteredaudio.com>