



fusion:FILTER User Guide

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OVERVIEW

This manual discusses how to use the Premiere, AudioSuite, and DirectX versions of fusion:FILTER.

Opcode's fusion:FILTER provides powerful, analog-style synthesis filtering and modulation capabilities to the digital audio recordist.

fusion:FILTER contains digital emulations of classic analog filters, ring modulators, and distortion devices. Specifically, it offers:

- **Lowpass** filters. Two different models -- one with variable poles. Filter frequency and resonance can be modulated.
- **Highpass** filter with variable poles. Filter frequency and resonance can be modulated.
- **EQ Boost** filter. Single band EQ. Frequency and depth can be modulated.
- **Bandpass** filter. Filter frequency and band width can be modulated.
- **Notch** (band reject) filter. Filter frequency and notch width can be modulated.
- **Ring modulator** with sawtooth, sine, square, and triangle wave modulators. Frequency and depth can be modulated.
- **OverDrive**, for distortion effects. Frequency and depth can be modulated.

fusion: FILTER lets you use up to three filters in a single patch -- arranged in any serial, parallel, or hybrid combination.

Additionally, fusion:FILTER contains the following fully-programmable modulation sources:

- Up to six separate **low frequency oscillators** (LFO's).
- Up to six separate **pulse sequences**. Use a drum-machine style editor to create pulse sequences and use them to modulate any of the three filters -- turning any pad, drone, or ambient file into a pulsating rhythm loop.
- Up to six **user-drawable envelopes**. Draw any envelope shape and use it to modulate any of the three filters, creating subtle filter sweeps or a chaotic cacophony.
- Up to six **file-extracted envelopes**. Extract an envelope from any digital audio file and use it to modulate any of the three filters -- perfect for imprinting the rhythmic feel of one audio file onto another.
- An **envelope follower**. Use the volume envelope of the file you're processing to modulate any or all of the three filters.

The remainder of this document discusses how to use fusion:FILTER to process your own audio files.



BASIC OPERATION

This section discusses basic techniques for setting and selecting parameters.

Knobs



To increase the value of a knob, position the cursor over a knob, then press and hold the mouse button while either pushing the mouse away from you or dragging it to the right.

To decrease the value of a knob, position the cursor over a knob, then press and hold the mouse button while either pulling the mouse toward you or dragging it to the left.

To instantly set the knob to any value, click anywhere along its colored outer arc.

Faders



Some parameters use faders instead of knobs. Faders can be oriented either vertically or horizontally. To adjust a fader, simply drag the small red "thumb" along the path of the fader. To instantly set the fader to any value, click anywhere along its length. The fader's value is indicated graphically by the cyan trail left by the fader. Some faders display their exact value in a corresponding numerical.

Numericals



Numericals display the exact value of a knob's position. To change the value of a numerical directly, use one of the following techniques:

- Click the small "up arrow" button to the right of the numerical to increment its value. Hold down the button to automatically scroll through increasing values.
- Click the small "down arrow" button to the left of the numerical to decrement its value. Hold down the button to automatically scroll through decreasing values.
- Click anywhere within the numerical to select it (highlighting it), type a new value, then hit the ENTER key.



Some numericals (such as the one shown on the left) contain multiple segments. Multi-segment numericals let you independently select different portions of the display. For example, you can select the left part of the display (the integer values) and enter a new integer value without affecting the decimal values. Similarly, you can select the right part of the display (the decimal values) and enter a new decimal value (without affecting the integer value). Typing the TAB key moves the selection from left to right and, in the case of decimal numbers (like Tempo), typing the period key also moves the selection from the left portion to the right -- this means, for Tempo, you can click in the left field and type "97.86" and your tempo is set to that value.

Buttons



Click a button to activate it. If a button has an LED, then that button has an associated on/off state. If the button is ON, then the LED is lit and the button stays down. If the button is OFF, then the LED is dark and the button is up.

Selectors



Selectors are lighter gray than buttons. Use selectors to select an item from a pop-up or pull-down list of choices. The Patch Selector, which appears in the Control module, is an example of a Selector.

About Box

Open an About box for fusion:FILTER by clicking the fusion logo in the lower left corner of the window.



QUICK START GUIDE

To use Opcode's fusion:FILTER plug-in:

1. Select some audio in your host-application.
2. Open the fusion:FILTER plug-in from your host application's DSP menu (or equivalent).
3. You can either select one of the factory-supplied FILTER patches or create your own.

To open a factory patch:

1. If you want to start with a factory patch and you're using either the Premiere or AudioSuite versions, select the desired factory patch from the Patch Selector menu in the Control module. To select a factory patch in the DirectX version, click the **Import** button in the Control module (or, if the host application supports standard presets, then you can also select a preset from the host application's list).
2. Factory patches are timed at 120 bpm. If the audio file you're processing runs at a different speed, enter a different tempo in one of the Modulator module's Tempo fields, then option-click it to set all other tempos to that value.

3. Preview the effect by clicking whichever button that your host application provides for this purpose. In the case of Adobe Premiere, the **Preview** button is contained within fusion:FILTER's Control module. In AudioSuite and DirectX, the Preview button is provided by the host application.

NOTE: The amount of time required to compute a preview depends on the speed of your computer and the length of the audio file. Use the Preview Status display in the Control module to determine if the preview is currently being calculated.

4. Adjust the Output module's Level control as necessary. Alt-click (or option-click) it to automatically normalize the output level's audio signal.
5. If you're happy with the previewed sound, you can go ahead and process it -- writing it to your hard disk. Process the effect by clicking whichever button that your host application provides for this purpose. In the case of Adobe Premiere, press the **OK** button within fusion:FILTER's Control module.
6. If you're not happy with the previewed sound, it's time to start tweaking. Some basic editing techniques are discussed in the next tutorial.

To create your own filter effects (beginning with the Default patch):

1. From the patch selector in the Control module, select the patch named "Default."

If you've never used fusion:FILTER, this patch makes it easy to experiment with a single filter and modulator without worrying about interactions with other filters. Specifically, it sets the connection order to "Parallel," activates Filter #1 (turning it yellow), and open-circuits filters #2 and #3 (turning them gray).

2. In the upper left corner of the Filter module, click the 1 button to select Filter #1 for editing.
3. From the selector below the Filter Shape display, choose the type of filter you wish to assign to Filter #1.
4. Set the desired **Freq** and **Q** (or **W** or **D**) values using either the numericals or the red handles on the Filter Shape display.
5. Use the **F mod** and **Q mod** (or **W mod** or **D mod**) numericals to specify how much modulation to apply to each parameter (you'll assign and edit modulators in the Modulator module). Modulation can be either positive or negative.
6. In the Modulator module, click the **F** button in the left-most column (under the number, 1)
The three columns correspond to the three filters.
7. From the selector below the six buttons, choose which type of modulator you wish to use to modulate Filter #1's frequency.

You can choose **LFO**, **Envelope**, **Sequence**, or **None**.

8. For this example, choose **LFO**.

The area to the right of the selector changes to display LFO parameters.

9. Use the various LFO parameters to select a waveshape and oscillation rate.

If you've set the Filter module's **F mod** numerical high enough, you should be able to hear the LFO modulating Filter #1's cutoff frequency.

10. In the Filter module, set the output **Level** of the filter.

If you remove a lot of frequencies, you may need to boost the level. If you use a lot of resonance, you may need to cut the level. Option-clicking the **Peak** light automatically normalizes the filter's level to produce a 0dB output.

11. If desired, create a modulator for Filter #1's **Q** (or **W** or **D**) parameter.

12. Experiment with using multiple filters by selecting different connection orders from the Order module and configuring the additional filters as discussed previously.

13. In the Order module, click a filter button to change its state.

Each filter operates in one of three states:

Active - yellow (the audio is processed by the filter)

Bypassed - cyan (sound appearing at the input is passed unfiltered to the output).

Open-Circuited - gray (audio does not pass through the filter).

NOTE: It's possible in some connections to open-circuit filters in such a way that no output is heard. If this happens, the words "Output Disconnected" appear in the Control modules Status area.

14. In the Output module, use the **Level** knob to set an overall output level for the processed file. Option-clicking the **Peak** light automatically normalizes the output level to its maximum, unclipped level.
15. Process the audio by clicking whichever button that your host application provides for this purpose. In the case of Adobe Premiere, click the **OK** button in fusion:FILTER's Control module.

The following chapters discuss all fusion:FILTER parameters and operations in detail.



FILTER MODULE

Use the Filter module to:

- Assign a filter type to one or more of the three filters.
- Set basic parameters for each filter (such as cutoff frequency, resonance, width, or depth)
- Establish the amount of frequency modulation and resonance (width or depth) modulation for each of the three filters.
- Set an output level for each filter.

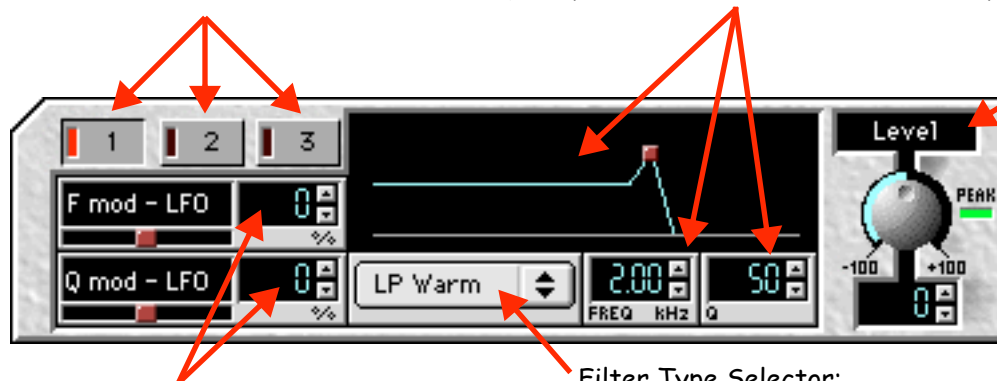
The following pages discuss all the Filter modulator parameters in detail.

Filter Overview

The following figure illustrates the layout of a typical Filter module:

Filter Select Buttons:
Click to select which filter to view and/or edit.

Filter Shape Display and Parameters:
Graphical representation of selected filter type. Drag handle(s) to change filter frequency and resonance (or width if bandpass/notch is displayed).



Filter Level:
Use to set output level of each filter.

Modulation Amounts:
Set a positive or negative amount by which to modulate each parameter (modulation sources are selected and edited in the Modulator module).

Filter Type Selector:
Use to select type of filter.

Figure 1: Typical Filter Module Layout

Filter Select Buttons



fusion:FILTER contains three independent filters, numbered 1, 2, and 3. Click a numbered button to display all the parameters for the corresponding numbered filter.

When you click one of the Filter Select buttons, the Modulator module changes to display the most recently edited modulator for that numbered filter. If you want to view a different filter without causing the modulator display to change, alt-click (or option-click) the Filter Select button.

- ! NOTE: If a filter is either bypassed or open-circuited (as described in ["Filter States" on page 28](#)), then most (or all) of its parameters are disabled and hidden from view. In the Order module, click a filter number to change its state to "active" (yellow).

Filter Type Selector



From the drop-down list of choices, select the type of filter you wish to assign to the currently displayed filter number. You have numerous choices, which included:

- **LP 2 pole** -- this lowpass filter features variable resonance and rolls off at 12dB/octave.
- **LP 4 pole** -- this lowpass filter features variable resonance and rolls off at 24dB/octave.
- **LP 6 pole** -- this lowpass filter features variable resonance and rolls off at 36dB/octave.

- **LP Warm** -- this is the second of two lowpass filter models contained in fusion:FILTER. This version features variable resonance and rolls off at 24dB/octave. Although it's similar to the "LP 4 pole" filter, its sonic characteristics are somewhat different.
- **HP 2 pole** -- this highpass filter features variable resonance and rolls off at 12dB/octave.
- **HP 4 pole** -- this highpass filter features variable resonance and rolls off at 24dB/octave.
- **HP 6 pole** -- this highpass filter features variable resonance and rolls off at 36dB/octave.
- **EQ Boost** -- this fixed-width (1-octave), single band of equalization features variable frequency and depth controls, rolling off at 24dB/octave.
- **Bandpass** -- this bandpass filter features variable width and rolls off at 24dB/octave.
- **Notch** -- this notch (band reject) filter features variable width and rolls off at 12dB/octave.
- **RM Saw** -- this ring modulator multiplies a sawtooth wave with your original audio file, creating the most dissonant overtones.
- **RM Sin** -- this ring modulator multiplies a sine wave with your original audio file, creating a less dissonant sound than either sawtooth or square wave modulators.
- **RM Squ** -- this ring modulator multiplies a square wave with your original audio file, creating fairly dissonant overtones.

- **RM Tri** -- this ring modulator multiplies a triangle wave with your original audio file, creating a little more dissonance than the sine wave modulator.
- **OverDrive** -- this special type of lowpass filter first distorts the input waveform, then wave-shapes it through a lowpass filter.

Filter Shape Display and Parameters



The Filter Shape display changes to graphically illustrate the type of filter chosen from the Filter Type selector. For example, the screenshot to the left shows a 4-pole lowpass filter.

Each filter has two parameters, which you can edit using either the handle(s) on the Filter Shape display, or the numerals below it. The various filter types are discussed on the next few pages.

- **Lowpass filters:**

Lowpass filters remove high frequencies from your audio (allowing low frequencies to pass through the filter). When you use a lowpass filter, the Filter Shape display looks similar to the following illustration.



Handle:

Drag left/right to set the cutoff frequency.
Drag up/down to set the Q (resonance).

Figure 2: Typical Lowpass Filter Display

You can set a lowpass filter's cutoff frequency using either the **Freq** numerical or the handle on the Filter Shape display. To change the cutoff frequency using the handle, drag it left (to lower the cutoff frequency), or right (to raise the cutoff frequency).

In addition, you can set the resonance (or "Q") for the filter. Resonance emphasizes the frequencies around the cutoff frequency. Set the resonance using either the **Q** numerical, or by dragging the Filter Display handle up and down.

- ! **NOTE:** fusion:FILTER allows resonance levels that are capable of self-oscillating. Instead of "playing it safe" and scaling back the filters so that they never self-oscillate, fusion:FILTER lets you decide how much resonance is "too much." The up side of this is you can create some very aggressive resonant peaks. The down side is, with certain input

signals, you can drive a filter into an oscillating frenzy. Be careful of very high Q values or very high amounts of Q modulation -- they may cause the filter to self-oscillate (especially with the 4 and 6 pole models). We know some of you will love the self-oscillating filters -- those that don't should keep the total Q level (static plus modulated amount) to below 85-90. Another good tact is to set the default Q between 85-90, then apply *negative* modulation to *remove* resonance from the filter -- this produces an effect similar to setting a low Q and adding positive modulation, except that the results are more predictable. If a filter still self-oscillates with a low Q value, try shifting the filter's cutoff frequency up or down slightly, so as not to accentuate a resonant peak that probably exists within the audio file you're processing.

- **Highpass filters:**

Highpass filters remove low frequencies from your audio (allowing high frequencies to pass through the filter). When you use a highpass filter, the Filter Shape display looks similar to the following illustration.



Handle:
Drag left/right to set the cutoff frequency.
Drag up/down to set the Q (resonance).

Figure 3: Typical Highpass Filter Display

You can set a highpass filter's cutoff frequency using either the **Freq** numerical or the handle on the Filter Shape display. To change the cutoff frequency using the handle, drag it left (to lower the cutoff frequency), or right (to raise the cutoff frequency).

In addition, you can set the resonance (or "Q") for the filter. Resonance emphasizes the frequencies around the cutoff frequency. Set the resonance using either the **Q** numerical, or by dragging the Filter Display handle up and down.

- ! NOTE: fusion:FILTER allows resonance levels that are capable of self-oscillating. Instead of "playing it safe" and scaling back the filters so that they never self-oscillate, fusion:FILTER lets you decide how much resonance is "too much." The up side of this is you can create some very aggressive resonant peaks. The down side is, with certain input signals, you can drive a filter into an oscillating frenzy. Be careful of very high Q values or very high amounts of Q modulation -- they may cause the filter to self-oscillate (especially with the 4 and 6 pole models). We know some of you will love the self-oscillating filters -- those that don't should keep the total Q level (static plus modulated amount) to below 85-90. Another good tact is to set the default Q between 85-90, then apply *negative* modulation to *remove* resonance from the filter -- this produces an effect similar to setting a low Q and adding positive modulation, except that the results are more predictable. If a filter still self-oscillates with a low Q value, try shifting the filter's cutoff frequency up or down slightly, so as not to accentuate a resonant peak that probably exists within the audio file you're processing.

- **EQ Boost:**

This fixed-width (1-octave), frequency-variable single band of equalization lets you boost any desired frequency. It's particularly useful for increasing the effects of resonance when placed immediately after a resonant filter in a serial connection. When you use the EQ Boost filter, the Filter Shape display looks similar to the following illustration.



Handle:
Drag left/right to set the center frequency.
Drag up/down to set the depth.

Figure 4: Typical EQ Boost Display

You can set the EQ's frequency using either the **Freq** numerical or the handle on the Filter Shape display. To change the EQ's frequency using the handle, drag it left (to lower the frequency), or right (to raise the frequency).

In addition, you can set the depth of boost. The greater the depth, the more the specified frequency is emphasized. Set the depth using either the **Depth** numerical, or by dragging the Filter Display handle up and down.

- ! **NOTE:** Since this filter always increases the audio signal level, you may have to turn down the filter's **Level** control (or option-click its Peak light to auto-normalize the level).

- **Bandpass filter:**

Bandpass filters remove both low and high frequencies from your audio (allowing only frequencies centered around a specific frequency band to pass through the filter). When you use a bandpass filter, the Filter Shape display looks similar to the following illustration.

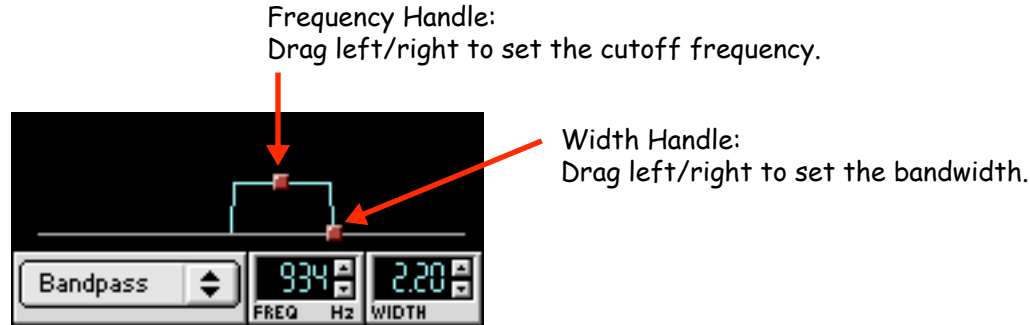


Figure 5: Typical Bandpass Filter Display

You can set a bandpass filter's cutoff frequency using either the **Freq** numerical or the Frequency handle left (to lower the cutoff frequency), or right (to raise the cutoff frequency).

In addition, you can set the bandwidth using either the **W** numerical, or by dragging the Width handle left (to narrow the band) or right (to widen the band).

- **Notch filter:**

Notch filters remove frequencies centered around a specific frequency (allowing only low and high frequencies outside the notch to pass through the filter). When you use a notch filter, the Filter Shape display looks similar to the following illustration.

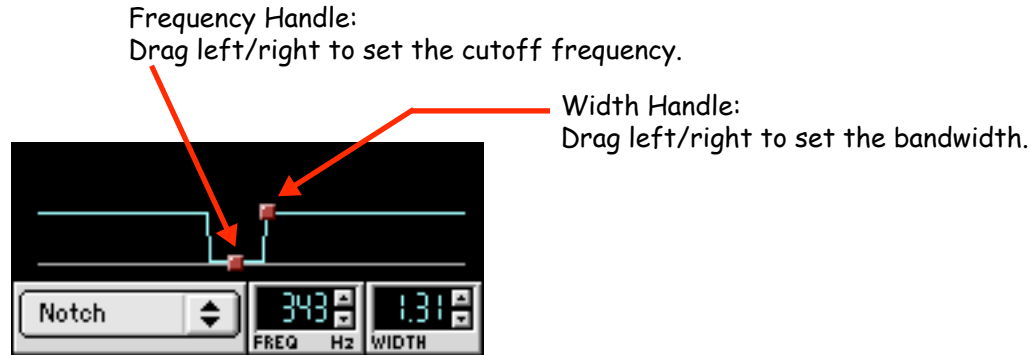


Figure 6: Typical Notch Filter Display

You can set a notch filter's cutoff frequency using either the **Freq** numerical, or by dragging the frequency handle left (to lower the cutoff frequency), or right (to raise the cutoff frequency).

In addition, you can set the bandwidth for the notch using either the **W** numerical, or by dragging the notch width handle left (to narrow the notch) or right (to widen the notch).

- **Ring Modulators:**

Ring modulation multiplies a pair of audio signals and outputs their sum and difference frequencies. Ring modulation can produce very complex, clangorous sounds with many dissonant overtones. In fusion:FILTER, one of the audio signals is the audio you wish to process -- the other is a constant waveform (sawtooth, sine, square, triangle). When you select a ring modulator, the Filter Shape display looks similar to the following illustration.



Handle:

Drag left/right to set the modulator's frequency.

Drag up/down to set the depth of modulation.

Figure 7: Typical Ring Modulator Display

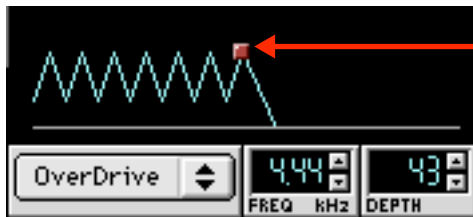
The Filter Shape display illustrates the shape of the ring modulator waveform.

You can set the modulator's frequency using either the **Freq** numerical or the handle on the Filter Shape display. To change the modulator's frequency using the handle, drag it left (to lower the frequency), or right (to raise the frequency).

In addition, you can set the depth of ring modulation. The greater the depth, the greater the ring modulation effect. Set the depth using either the **Depth** numerical, or by dragging the Filter Display handle up and down.

- **OverDrive:**

OverDrive distorts the input waveform, then wave-shapes it through a lowpass filter. The higher you set the cutoff frequency, the "grungier" the sound. When you select OverDrive, the Filter Shape display looks similar to the following illustration.



Handle:
Drag left/right to set the cutoff frequency.
Drag up/down to set the depth of distortion.

Figure 8: Typical Overdrive Display

You can set the overdrive waveshaping frequency using either the **Freq** numerical or the handle on the Filter Shape display. To change the cutoff frequency using the handle, drag it left (to lower the frequency), or right (to raise the frequency).

In addition, you can set the depth of the overdrive effect. The greater the depth, the greater the distortion. Set the depth using either the **Depth** numerical, or by dragging the Filter Display handle up and down.

Modulation Amounts



Each of the two parameters in fusion:FILTER's three independent filters can be modulated. Frequency (F) is always one of the parameters that you can modulate; the other depends on the type of filter used: Lowpass and Highpass filters can have their resonance (Q) modulated; Bandpass and Notch filters can have their widths (W) modulated; Ring Mods, Overdrive, and EQ Boost can have their depths (D) modulated.

The modulators themselves are assigned and configured in the Modulator module. The Modulator Amount controls (shown here) determine how *much* to modulate each of the two filter parameters. Modulation can add to or subtract from the static values set in the Filter Shape display.

The Modulation Amount labels change depending on:

- The type of filter selected -- Bandpass/notch filters will have a **W mod** parameter; lowpass/highpass filters will have a **Q mod** parameter; RM, OverDrive, and EQ Boost will have a **D mod** parameter. All four filter types will have an **F mod** parameter.
- The type of modulator selected -- If the parameter is being modulated by an LFO, the label will say **LFO**; if it's being modulated by an envelope, the label will say **ENV**; if it's being modulated by a pulse sequence, the label will say **SEQ**; if the modulator is disabled, the label will say **NONE**.

Filter Level



Each of the three filters has its own output **Level** control. These can be particularly useful when you chain filters together in some form of parallel order (as discussed in ["Connection Order" on page 31](#)). You may boost or cut individual filter levels to "mix" the various filter effects together.

In addition, if you bypass a filter (as discussed in ["Filter States" on page 28](#)), the **Level** control is still active. In certain parallel connections, this lets you mix some of your unprocessed audio signal into the final output of fusion:FILTER.

To the right of the **Level** parameter is the **Peak** indicator, which behaves as follows:

- The peak indicator turns green if any portion of the previewed filter output exceeds -24dB.
- The peak indicator turns yellow if any portion of the previewed filter output exceeds -6dB, but remains at or below 0dB.
- The peak indicator turns orange when any portion of the previewed audio exceeds 0dB (see ["Filter Levels in Excess of 0dB" on page 27](#) for more information).

If you Alt-click (or Option-click) the **Peak** indicator, fusion:FILTER automatically adjusts the **Level** parameter to produce a filter output of 0dB.

Filter Levels in Excess of 0dB

The peak indicator turns *orange* rather than *red* when 0dB is exceeded because fusion:FILTER's internal floating-point processing affords enough internal headroom to prevent clipping. Clipping occurs only when the final signal is converted to integer values in the Output module.

Therefore, it's OK if a filter level exceeds 0dB (turning orange), as long as the signal eventually gets reduced before the final output (usually by lowering the Output Module's **Level** knob, or alt-clicking (option-clicking) the Output module's **Peak** indicator).

Filter peak indicators turn orange when their output exceeds 0dB to warn you that clipping *might* occur if you don't lower the signal somewhere further down the processing chain.



ORDER MODULE

The Order module has two functions:

- To specify the state of each filter (active, bypassed, open)
- To specify the connection order of the filters

You can achieve many radically different sounds simply by combining different filter states with different connection orders.

The following sections discuss filter states and connection orders in detail.

Filter States

Each filter can operate in one of three states, as shown below:

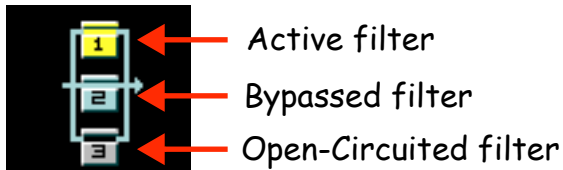
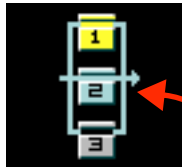


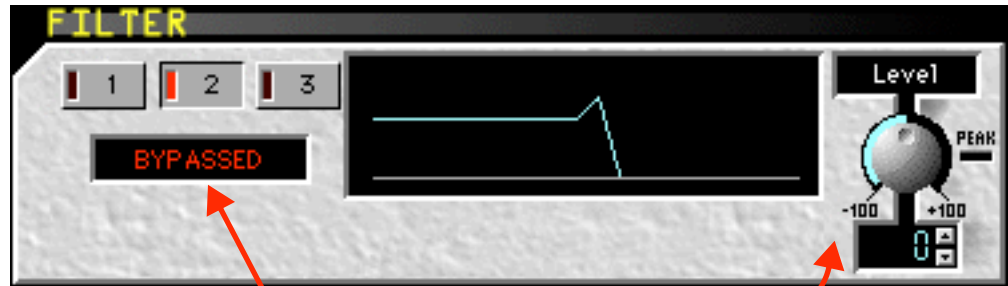
Figure 9: Three filter states

The three states operate as follows:

- **Active** - The filter is turned "on" (as indicated by its yellow color). It processes the audio received at its input in accordance with the parameters set in the Filter module.
- **Bypassed** - The filter is "short circuited" (as indicated by its pale cyan color, which is the same color as the "wires" that connect the filters). Any audio arriving at a bypassed filter's input is passed through to its output. The only parameter that you can control for a bypassed filter is its **Level**, which allows you to increase or decrease the level of the audio passing through the filter, creating an effective way to "mix" unprocessed audio with processed audio in certain parallel connection orders. When a filter is bypassed, its non-editable parameters are hidden in the correspondingly numbered Filter module.

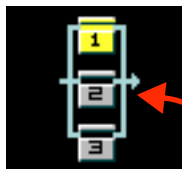


When, for example,
Filter #2 is bypassed...

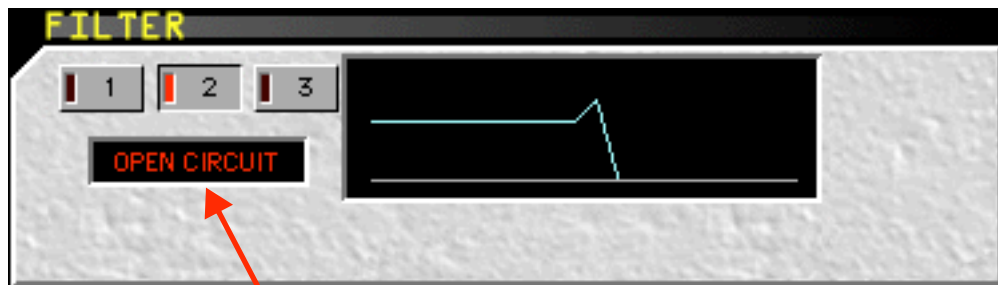


... then the word **BYPASSED** appears in the edit area
for Filter #2 and only the Level knob is active.

- **Open-Circuited** - The filter is "open circuited" (as indicated by its dark gray color and broken "wire"). Open circuiting a filter is just like cutting the wire -- any audio arriving at the filter's input does not pass through to its output. When a filter is open-circuited, all parameters are disabled in the correspondingly numbered Filter module.



When, for example,
Filter #2 is open
circuited...



... then the words OPEN CIRCUIT appear in the edit area
for Filter #2 and all parameters are disabled.

- If open-circuiting a filter prevents all audio from reaching the final output stage (as would be the case in a serial connection order), "Output Disconnected" is displayed in the Control module's Status area.



When open-circuiting a filter prevents audio
from reaching the final output stage...



... the Status area reads "Output Disconnected"

To change states:

1. Click the filter number in the connection order diagram.

The filter number acts like a three way switch, changing from "Active" to "Bypassed" to "Open", then back to "Active."

NOTE: You can step through the states in *reverse* order by alt-clicking (option-clicking) the filter number.

Connection Order

fusion:FILTER contains three independent filters, which you can connect in any of five different orders. These are:

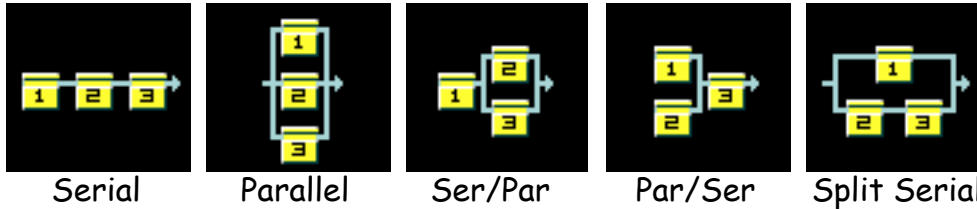


Figure 10: Various Filter Orders

Different orders provide different benefits. Specifically:

- **Serial:** The output of the first filter is further processed by the second filter, whose output is then processed by a third filter. This results in a heavily processed, uniform sound with none of the original signal mixed in. Open-circuiting any filter breaks the chain and results in no audio output.
- **Parallel:** Each of the three filters acts directly on the selected audio, allowing you to create three different filter effects, which you can mix together using each filter's Level control. If you bypass one or two of the filters, some of the unprocessed audio will be mixed into the output (using the bypassed filter's Level control to set the amount of unprocessed audio that appears at the output).
- **Serial/Parallel:** This lets you first filter the entire signal, then split its output such that it's further processed by two independent filters, whose outputs are mixed together (using their respective Level controls).
- **Parallel/Serial:** The first two filters act directly on the selected audio, allowing you to create two completely different filter effects, which you can mix together using each filter's Level control. The output of these mixed filters is then sent through a single filter, giving the sound more uniformity.

- **Split Serial:** Provides two separate processing paths. The bottom path contains a pair of serially connected filters whose output is mixed with the single filter in the top path (using each filter's respective Level control). If you bypass the top filter, some of the unprocessed audio will be mixed into the output (using the bypassed filter's Level control to set the amount of unprocessed audio that appears at the output).

To change connection order:

1. Choose the desired connection order from the selector menu immediately above the connection order diagram.



MODULATOR MODULE

Use this module to select, assign, and configure modulators to control up to two parameters in each of the three filters (six modulators in all).

Modulator Overview

The following figure illustrates the layout of a typical Modulator module:

Modulator Select Buttons:

Each button represents a parameter that can be modulated in each of the three filters. Click any button to set the parameters for that modulator.

Modulator Specific Parameters:

This area changes depending on the type of modulator selected. This example shows a Pulse Sequencer.

Type Selector:

Use this to select which type of modulator to use.

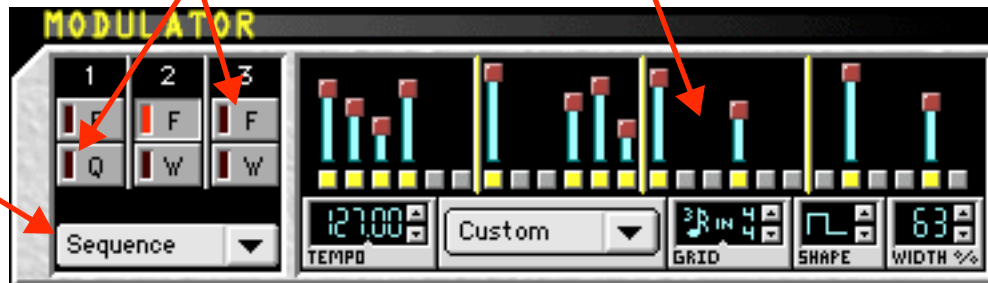


Figure 11: Typical Modulator Module Layout

Modulator Select Buttons



Opcode's fusion:FILTER contains three independent filters, numbered 1, 2, and 3. Each filter has two parameters that can be modulated. A filter's frequency (F) is always one of these parameters. The other depends on the type of filter used -- Lowpass and Highpass filters can have their resonance (Q) modulated. Bandpass and Notch filters can have their widths (W) modulated. OverDrive, EQ Boost, and Ring

Modulation can have the depths (D) modulated.

Use the Modulator Select buttons to determine which modulator you wish to view and/or edit. There are six buttons, arranged in three columns of two buttons each. Each column corresponds to a filter number (1, 2, or 3). Each row represents one of each filter's two modulated parameters.

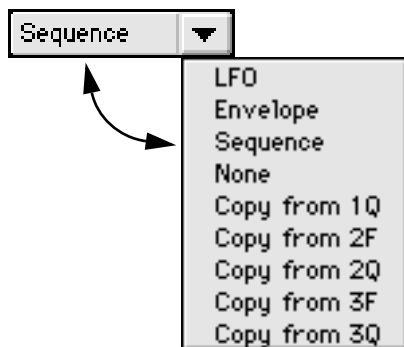
The top button is always labeled "F" (frequency), since frequency is always one of the parameters that can be modulate, regardless of the selected filter type.

The label on the bottom button changes between "Q" (resonance), "W" (width), or "D" (depth) depending on the type of filter you've selected in the Filter module.

Click any button to view the modulator assigned to the indicated filter parameter. In the above illustration, the modulator assigned to filter #2's cutoff frequency would be displayed.

! NOTE: When you click one of the Modulator Select buttons, the Filter module changes to display the filter whose modulator you're editing. If you want to view a different modulator without causing the filter display to change, alt-click (or option-click) the Modulator Select button.

Type Selector

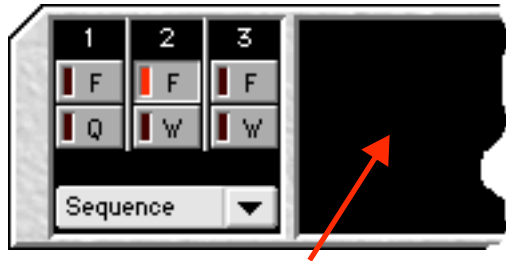


Use the Type Selector to assign a modulation type to the currently displayed modulator. There are three main types of modulation sources: **LFO**, **Envelope**, and **Sequence**. Each of these modulation types will be discussed later in this manual.

In addition, you can select **None** to turn off a modulator.

Finally, you can use one of the **Copy from** choices to copy all the parameters from some other modulator to the currently displayed one.

Modulator Specific Parameters



This area changes to show parameters for selected modulator type.

The area to the right of the Modulator Select buttons changes depending on which modulator is selected and what type of modulator it is.

The following sections discuss each type of modulator in detail.

LFO Modulation

When you select an LFO modulator, the edit area looks as follows.

Phase:

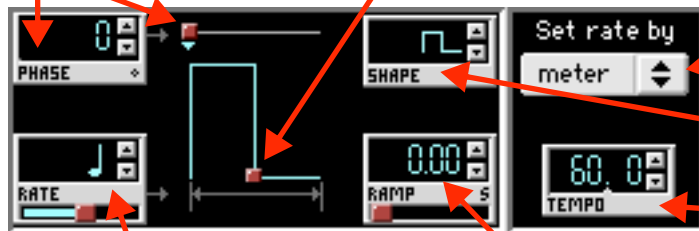
Sets start (trigger) position within LFO. Use the Fader to set the trigger position visually, or the numerical to set it exactly.

Width/Symmetry:

Drag horizontally to create asymmetric LFO shapes. For square and sawtooth waves, it controls width of wave. For triangle wave, it controls asymmetry of wave. Does not exist for random waveshape.

Rate Mode Selector:

Determines whether Rate parameter is set using note values (meter) or Hz (freq). If using meter, a Tempo numerical appears.



Shape:

Sets the basic LFO waveshape (triangle, sawtooth, square, random).

Rate:

Sets the rate at which the LFO oscillates. If Rate Mode Selector is set to "meter," then Rate is expressed as a note value. If Rate Mode Selector is set to "freq," then Rate is expressed in Hz.

Ramp:

Sets how long (in seconds) it takes for LFO to reach its maximum amplitude. This lets you create LFO effects in which the modulation gradually increases over time.

Tempo:

Set sequence tempo (appears only in meter mode).

Figure 12: Typical LFO Modulation (showing Rate expressed in metrical terms)

The following sections discuss the various LFO parameters in detail.

▶ Rate Mode Selector

You can select whether you want to express the LFO's oscillation rate in terms of *Meter* or *Frequency*.

- **meter** -- The **Rate** numerical is expressed in note values and a **Tempo** numerical appears below the Rate Mode selector. Use the **Tempo** numerical to enter the tempo of your audio file, then use the **Rate** numerical to set the periodic rate of the LFO.
- **freq** -- The **Rate** numerical is expressed in Hz (cycles per second) and no **Tempo** numerical appears below the Rate Mode selector. Use the **Rate** numerical to set the periodic rate of the LFO.

Select **meter** if you want to achieve synchro-sonic filtering effects that modulate in time with your sequence. Select **freq** if you want free-form modulations, or LFO rates that are slower or faster than those obtainable in meter mode.

- ! NOTE: LFO parameters set in **meter** mode are more precise than those set in **freq** mode. In general, it's best to keep each LFO in one mode or the other and not switch between them.

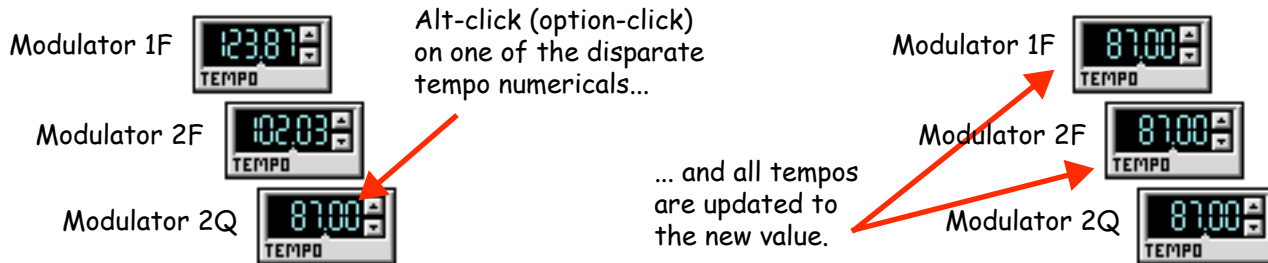
Tempo



This numerical appears ONLY if the Rate Mode Selector is set to meter.

Generally, you'll want to set this to the tempo of your host-application's sequence. For example, if you'll use the filtered audio file in a 120 bpm sequence, you should set the **Tempo** numerical to 120. The **Tempo** numerical uses a multi-segment numerical as described in ["Numericals" on page 5](#).

- ! NOTE: fusion:FILTER lets each modulator have its own tempo, which can create some very interesting polyrhythmic effects. If, however, you want to set ALL tempos in ALL modulators to the same value, Alt-click (or option-click) the **Tempo** numerical and all other tempos in all other modulators will be set to that same value. This new tempo also becomes the master tempo used by the **Trigger** button, discussed in ["Trigger Button \(DirectX\)" on page 72](#).



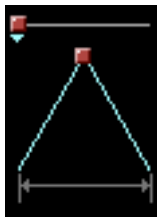
Rate

Use this numerical/fader to set the frequency at which the LFO oscillates. Rate values are expressed either in Hz or in note values, depending on whether you selected **freq** or **meter** in the Rate Mode selector.

- ! NOTE: LFO rates set in **meter** mode are more precise than those set in **freq** mode. In general, it's best to keep each LFO in one mode or the other and not switch between them.

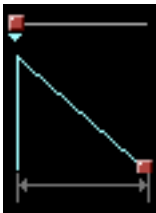
Shape

Use this numerical to set the basic LFO waveshape. You can choose between triangle, sawtooth, square, and random.



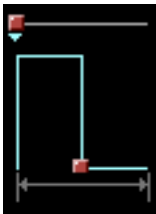
•Triangle

The LFO oscillates using a triangular waveshape, which it displays graphically in the large edit area. You can change the waveshape's symmetry by dragging the handle at its peak. You can change the phase of the wave by dragging its phase handle (or using the **Phase** numerical).



•Sawtooth

The LFO oscillates using a sawtooth waveshape, which it displays graphically in the large edit area. You can change the waveshape's pulse width by dragging the handle at its peak. You can change the phase of the wave by dragging its phase handle (or using the **Phase** numerical).



•Square

The LFO oscillates using a square waveshape, which it displays graphically in the large edit area. You can change the waveform's pulse width by dragging the handle at its peak. You can change the phase of the wave by dragging its phase handle (or using the **Phase** numerical).



•Random

The LFO oscillates randomly, which it displays generically in the large edit area. Random oscillation uses square waves of random height, resulting in a "sample-and-hold" effect. You cannot edit a random LFO's shape, nor can you change the phase of the wave (the phase numerical and handle disappear).

Phase

Use this parameter to set the exact position in the oscillation at which the LFO begins its cycle. You can set this value using either the **Phase** numerical or the graphical phase pointer over the LFO waveshape display.

By modulating different parameters with the same LFO rate/shape, but different phases, you can get harmonically interesting "movement" and "texture."

- At 0 degrees, the LFO begins oscillating at the beginning of its waveform.
- At 180 degrees, the LFO begins oscillating in the middle of its waveform.

! NOTE: This parameter is not available for LFO's of random shape.

Width

The graphic representation of triangle, sawtooth, and square waves all contain an edit handle. For sawtooth and square waves, drag this handle (located at the bottom of the waveshape) left/right to change the pulse width of the waveshape. For triangle waves, drag this handle (located at the waveshape's apex) left/right to change the symmetry of the waveshape.

Ramp

Sets how long (in seconds) it takes for the LFO to reach its maximum amplitude. This lets you create LFO effects in which the modulation gradually increases over time.

DirectX users can retrigger a ramp by clicking the **Trigger** button in the Control module. The **Trigger** button is not available for the Premiere or AudioSuite versions. For more information, see ["Trigger Button \(DirectX\)" on page 72](#).

- ! **PREMIERE USERS:** Since, in general, you can only preview the first few seconds of audio, a Ramp time of more than a few seconds will make it impossible for you to hear the LFO effect (since the preview time will have ended before the LFO ramped up to full amplitude). Therefore, you should set all other LFO parameters first -- setting the Ramp speed only after you're satisfied with the sound.

Pulse Sequence Modulation

When you select a Sequence modulator, the edit area looks as follows:

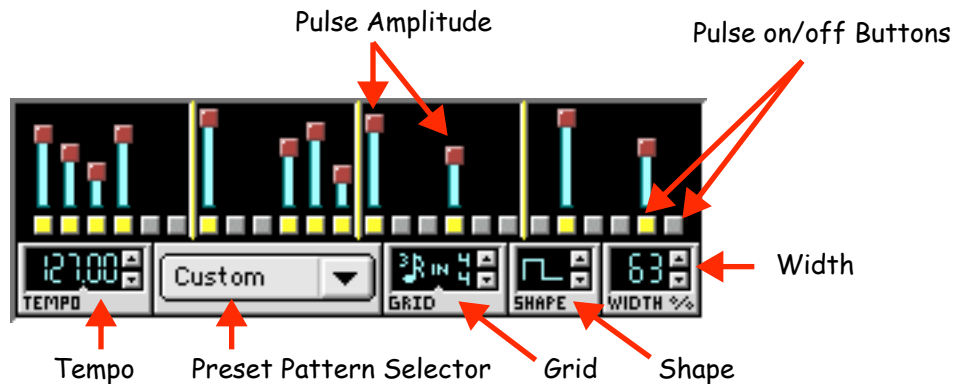


Figure 13: Typical Pulse Sequence Modulator

The following sections discuss the various pulse sequence parameters in detail.

Tempo



Use this numerical to enter the tempo at which you want the pulse sequence to play. Generally, you'll want to set this to the tempo of your host-application's sequence. For example, if you'll use the filtered audio file in a 120 bpm sequence, you should set the pulse sequencer's **Tempo** numerical to 120. The Tempo numerical uses a multi-segment numerical as described in ["Numericals" on page 5](#).

- ! NOTE: fusion:FILTER lets each modulator have its own tempo, which can create some very interesting polyrhythmic effects. If, however, you want to set ALL tempos in ALL modulators to the same value, Alt-click (or option-click) the **Tempo** numerical and all other tempos in all other modulators will be set to that same value. This new tempo also becomes the master tempo used by the **Trigger** button, discussed in ["Trigger Button \(DirectX\)" on page 72](#).

Modulator 1F



Modulator 2F



Modulator 2Q



Alt-click (option-click)
on one of the disparate
tempo numericals...

Modulator 1F



Modulator 2F



Modulator 2Q



... and all tempos
are updated to
the new value.

Grid



Use this multi-segment numerical to set the time signature and pulse spacing of the desired sequence (multi-segment numericals are discussed in ["Numericals" on page 5](#)).

Shape



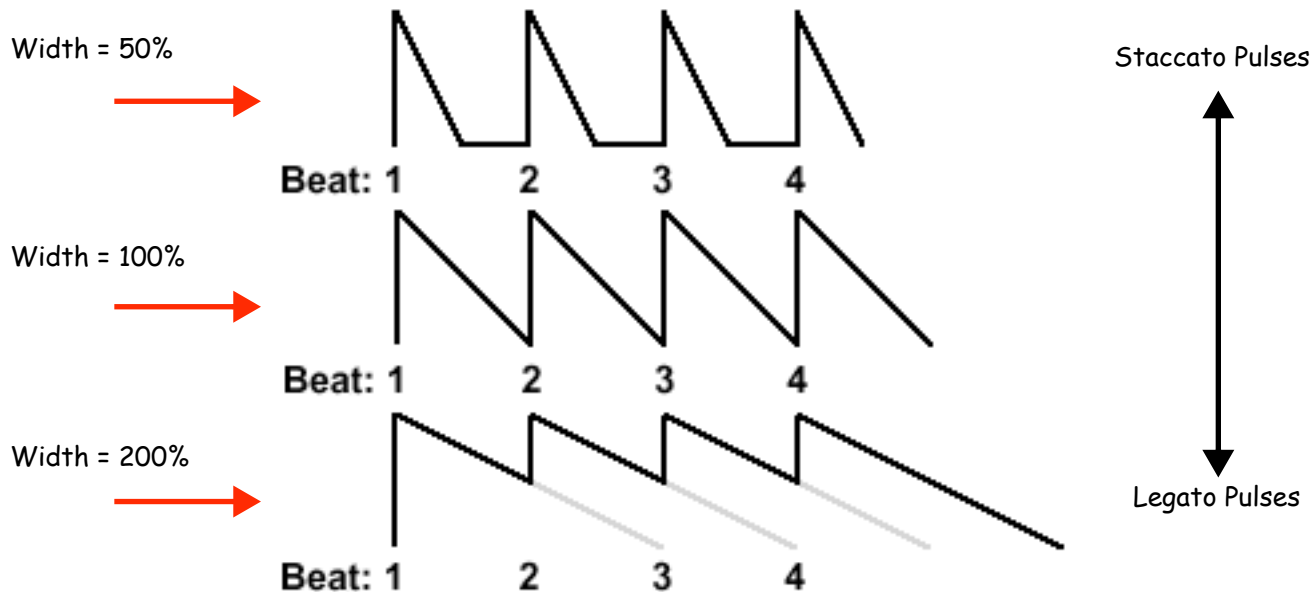
This sets the shape of each "pulse" created by the pulse sequencer. the shape of the pulse can dramatically change the sound of the sequence. You may select either square, triangle, sawtooth, or reverse sawtooth.

Width



This sets the width of each pulse, expressed as a percentage of the current **Grid** size.

For example, assume the **Grid** size is set to quarter notes and the **Shape** is set to a sawtooth wave. The illustration on the following page shows the effect that different widths have on the modulator.



Pulse On/Off Buttons



Click a button to create a pulse at the indicated position (causing the button to turn yellow). Click it again to turn the pulse off.

Pulse Amplitudes



When a pulse is turned on, fusion:FILTER creates an amplitude fader immediately above the Pulse On/Off button. Use this fader to set the amplitude of each pulse.

Preset Pattern Selector



This selector contains a number of preset pulse sequence patterns. Preset patterns store all sequence parameters except Tempo.

Envelope Modulation - Drawable

fusion:FILTER contains three types of envelopes. This section discusses **Draw** envelopes, which let you draw your own envelope shapes -- great for creating filter sweeps or subtly shifting textures.

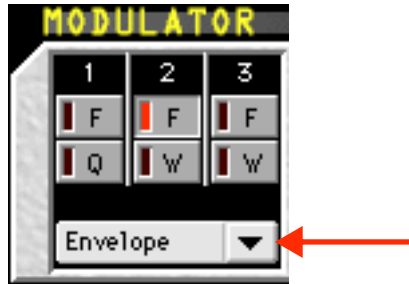
The other two types of envelopes are:

- **Extract** -- extracts an envelope from any audio file, applying the rhythmic effects of a different file onto the audio you're filtering. See ["Envelope Modulation - File Extraction" on page 58.](#)
- **Follow** -- uses the processed file's own volume envelope to modulate the filters. See ["Envelope Modulation - Following" on page 63.](#)

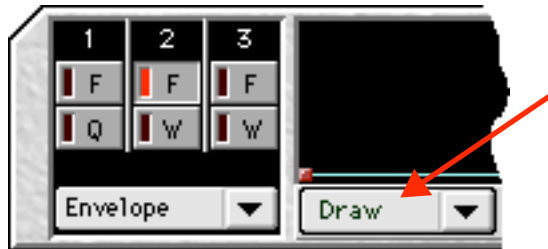
Entering Draw Mode

To enter envelope drawing mode:

1. Select **Envelope** from the Modulator Overview's Type selector.



2. In the lower left corner of the envelope edit area, select **Draw** from the Envelope Type selector.



You must tell fusion:FILTER how long the envelope is (in time). There are two ways to do this: 1) use the **Duration** numerical or 2) use the **File** button.

- Using the **Duration** numerical lets you set the exact time over which you want the envelope to complete its cycle. The envelope is processed exactly once at the start of audio processing and does not loop. See ["Drawing in Duration Mode" on page 52.](#)
- Using the **File** button lets you display the waveform for any audio file. It sets the envelope duration to the length of the selected audio file, and lets you draw an envelope over top of the waveform for precise positioning of the handles. See ["Drawing in File Mode" on page 54.](#)

Drawing in Duration Mode

To use the **Duration** numerical:

1. Set the length of the desired envelope (in seconds) using the **Duration** numerical.

The width of the edit area always equals the amount of time specified in the **Duration** numerical. For example, if Duration is 10 seconds, the left handle would be at 0 seconds and the right handle at 10.

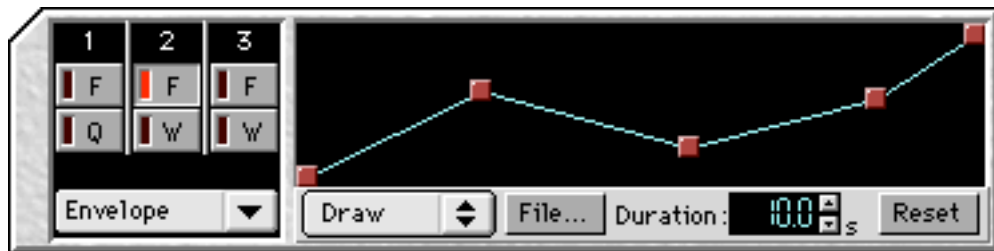


2. Drag either of the two end handles to reposition them.



3. Click anywhere along the envelope to create a new handle, and drag it to the desired position.

4. Continue creating new handles and dragging them until you achieve the desired envelope shape.



5. To erase a handle, option-click it.
 6. To reset the entire envelope to its default value, click the **Reset** button.
- ! DIRECTX USERS: You can retrigger an envelope by clicking the **Trigger** button in the Control module. The **Trigger** button is not available for the Premiere or AudioSuite versions. For more information, see ["Trigger Button \(DirectX\)" on page 72](#).

▶ Drawing in File Mode

To use the **File** button:

1. Click the **File** button to open a standard file dialog box.
2. Use the File dialog box to locate and select the audio file whose waveform you wish to view.

Audio files can be either AIFF, WAV, AU, or SDII (Mac only). If you select a stereo file, the displayed waveform will be the sum of the left and right sides.

The waveform begins drawing from left-to-right.



NOTE: In **Draw** mode, the **File** button does not extract an envelope -- it merely sets a duration and provides visual feedback over which to draw an envelope.

3. During this time, the **File** button changes to a **Clear** button, which you can click to stop the waveform drawing at any time.

The amount of time required to draw an waveform depends on both the speed of your computer and the length of the audio file. Drawing is complete when the entire waveform is displayed and a **Duration** value is shown (which equals the duration of the selected audio file).

4. Drag either of the two end handles to reposition them.



5. Click anywhere along the envelope to create a new handle, and drag it to the desired position.
6. Continue creating new handles and dragging them until you achieve the desired envelope shape.



7. To erase a handle, option-click it.
8. To reset the envelope to its default value, click the **Reset** button.

9. To clear the waveform display and bring back the **Duration** numerical, click the **Clear** button.
- ! **DIRECTX USERS:** You can retrigger an envelope by clicking the **Trigger** button in the Control module. The **Trigger** button is not available for the Premiere or AudioSuite versions. For more information, see ["Trigger Button \(DirectX\)" on page 72](#).

Envelope Modulation - File Extraction

fusion:FILTER contains three types of envelopes. This section discusses **Extract** envelopes, which let you draw your own envelope shapes, allowing you to apply the dynamic or rhythmic effects of a different audio file onto the audio you're filtering.

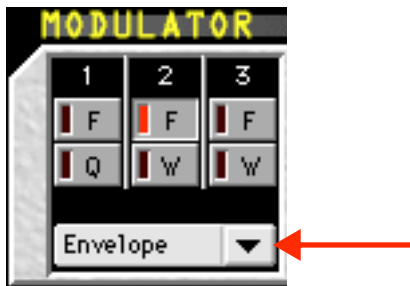
The other two types of envelopes are:

- **Draw** -- allows you to draw your own envelope shape, which is great for creating filter sweeps or subtly shifting textures. See ["Envelope Modulation - Drawable" on page 50.](#)
- **Follow** -- uses the processed file's own volume envelope to modulate the filters. See ["Envelope Modulation - Following" on page 63.](#)

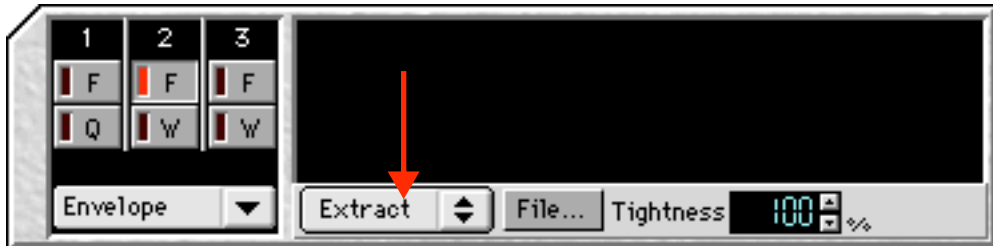
Extracting an Envelope

To extract an envelope from an audio file:

1. Select **Envelope** from the Modulator Overview's Type selector.



2. In the lower left corner of the envelope edit area, select **Extract** from the Envelope Type selector.



3. Click the **File** button to open a standard file dialog box.

4. Use the File dialog box to locate and select the audio file from which you wish to extract an envelope.

Audio files can be either AIFF, WAV, AU, or SDII (Mac only). If you select a stereo file, the envelope will be extracted from the sum of the left and right sides.

5. The waveform begins drawing from left-to-right.



6. During this time, the **File** button changes to a **Clear** button, which you can click to stop the file extraction at any point.

The amount of time required to extract an envelope depends on both the speed of your computer and the length of the audio file from which you extract the waveform. The file extraction is complete when the entire waveform is displayed and the red modulation envelope appears.

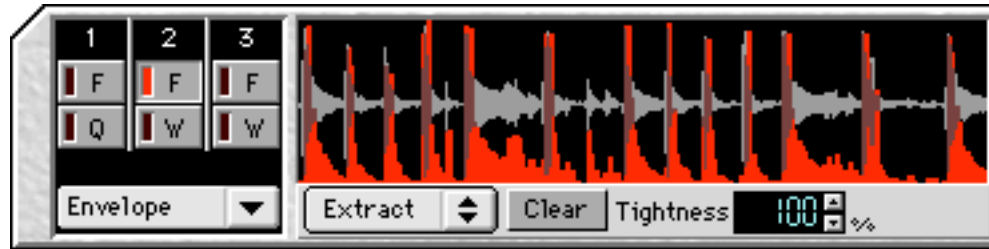


Figure 14: High Tightness Value

7. Use the **Tightness** numerical to determine how tightly the extracted envelope conforms to the audio file.

For example, if you extract an envelope from an audio drum loop and you set a low **Tightness** value, the extracted envelope will modulate just slightly in rhythm with the highest peaks in the audio file. If you set a high **Tightness** value, the extracted envelope will very closely follow all the peaks, valleys, and rhythmic nuances of the audio file.

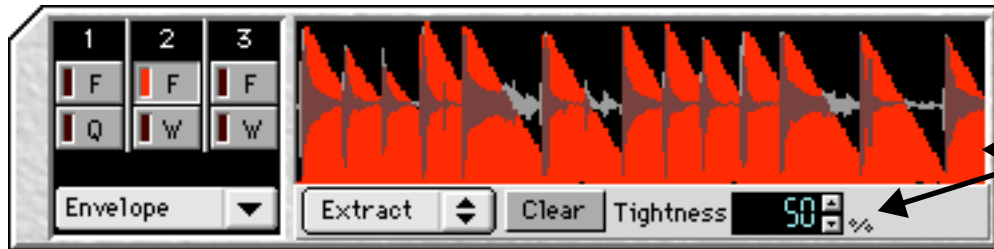


Figure 15: Medium Tightness Value

8. Click the **Clear** button to clear the waveform and extracted waveform.

When the waveform is cleared, the **Clear** button becomes a **File** button and you can select a different audio file from which to extract an envelope.

- ! NOTE 1: Extracted envelopes are always normalized, so that the highest peak always produces full modulation.
- ! NOTE 2: Extracted envelopes "loop." For example, if you're processing an 8 bar file and you extract an envelope from a 2 bar drum loop, the extracted envelope will repeat itself every 2-bars, modulating the entire 8-bar file.
- ! NOTE 3: Extraction occurs much faster if you turn off Previewing.

Envelope Modulation - Following

fusion:FILTER contains three types of envelopes. This section discusses **Follow** envelopes, which cause the audio file you wish to process to modulate the filters using its own dynamics.

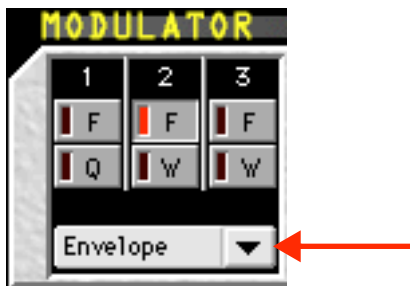
The other two types of envelopes are:

- **Draw** -- allows you to draw your own envelope shape, which is great for creating filter sweeps or subtly shifting textures. See ["Envelope Modulation - Drawable" on page 50.](#)
- **Extract** -- extracts an envelope from any audio file, applying the rhythmic effects of a different file onto the audio you're filtering. See ["Envelope Modulation - File Extraction" on page 58.](#)

Following the Dynamics of the File You're Processing

To create an envelope that follows the dynamics of the audio file you're currently processing:

1. Select **Envelope** from the Modulator Overview's Type selector.



2. In the lower left corner of the envelope edit area, select **Follow** from the Envelope Type selector.



The modulator will use the volume envelope of the file you're currently processing to create the envelope.

3. Use the **Tightness** numerical to determine how tightly the envelope follows the dynamics of your audio.

For example, if you're processing a drum loop and you set a low **Tightness** value, the envelope will modulate just slightly in rhythm with the highest peaks in the drum loop. If you set a high **Tightness** value, the envelope will very closely follow all the peaks, valleys, and rhythmic nuances of the drum loop.

Tightness affects an envelope follower exactly the same as an extracted envelope (as depicted on [page 61](#) and [page 62](#)).

- ! **IMPORTANT:** If an audio file doesn't contain a lot of dynamic range, the audible effects of envelope following will be very subtle. The same is true if the audio file has not been normalized. To achieve maximum effect, envelope following should be applied to files that, themselves, have a lot of dynamic range and that have first been normalized.



OUTPUT MODULE

Level



Use the Output module's **Level** parameter to set the overall output level of the processed audio file.

To the right of the **Level** parameter is the **Peak** indicator, which behaves as follows:

- The peak indicator turns green if any portion of the previewed output exceeds -24dB.
- The peak indicator turns yellow if any portion of the previewed output exceeds -6dB, but remains at or below 0dB (clipping).
- The peak indicator turns red when any portion of the previewed audio "clips" the output signal.

fusion:FILTER can automatically adjust the **Level** parameter to the loudest possible unclipped output, as described in the next section, "Auto-Normalizing."

Once the peak indicator lights, you can reset it by either:

- clicking the **Peak** indicator.
- setting a new **Level** (using either the knob or the numerical).

Auto-Normalizing

Opcode's fusion:FILTER provides a convenient way to automatically set **Level** parameters to create the loudest possible unclipped audio file. To do this:

1. Tweak some fusion:FILTER parameters.
2. Wait for the preview to finish processing.
3. Alt-click (or Option-click) the **Peak** indicator.

This resets the peak indicator and automatically sets the **Level** parameter to the value that will produce the loudest possible unclipped, output file.

Caveats

There are a few caveats to keep in mind:

- The section you're previewing may be only a small portion of the entire audio file. If this is true, it's possible that some portion of the audio file outside the preview range might still cause the processed output to clip.

- If you Alt-click (or Option-click) while the preview is still calculating, fusion:FILTER may set the wrong output level -- it's best to wait for the preview calculation to finish.

AudioSuite Technique

Due to architectural constraints, AudioSuite previews may be inaccurate by 1 or 2dB. Therefore, for AudioSuite, you should use the following auto-normalizing technique:

1. Set up the parameters, preview the audio, and option-click the **Peak** indicator to set an approximate output level.
2. Click the AudioSuite-provided **Process** button.
3. After processing finishes, select the **Undo** command.
4. Option-click the **Peak** indicator, then click the **Process** button again.

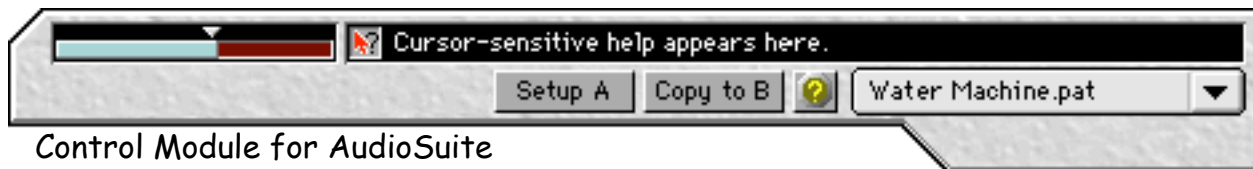
The resulting file will be accurately set to its maximum unclipped level.



CONTROL MODULE



Control Module for Premiere



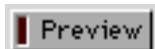
Control Module for AudioSuite



Control Module for DirectX

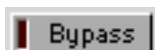
As depicted by the previous illustrations, the appearance of the Control module changes depending on the plug-in architecture you're using. The following sections discuss the various Control module elements.

▶ Preview Button (Premiere)



Press this button to audition a small, looped segment of audio. The length of the preview is determined by your host application. Preview allows you to hear the effect of your edits before you actually process the audio.

▶ Bypass Button (Premiere)



If you're previewing audio, press this button to listen only to the selected input audio, while muting the processed signal.

▶ Preview Status (All)

The function of the Preview Status area changes depending on the plug-in architecture used by your host application.

▶ For Premiere & AudioSuite



It takes a little time for fusion:FILTER to internally process your edits and apply them to the previewed audio. The faster your computer, the faster this calculation occurs.

Use the Preview Status display to determine whether you're listening to processed or unprocessed audio.

It works like this: The width of the display represents the entire preview length of your audio file. In the top-half of the display, a pointer moves left to right to indicate the current play location within the preview time. In the bottom-half of the display is the processed/unprocessed indicator. When you change a parameter, the computer recalculates the output starting from the point indicated by the pointer. As your computer recalculates the data, a light blue bar indicates that the region has been reprocessed. A dark red bar indicates that the region has not yet been reprocessed.

Therefore, if the pointer is over a light blue bar, it means you're listening to an audio preview that accurately reflects the parameters shown in the plug-in window. If the pointer is over a red area, it means you're listening to an audio preview that does not yet reflect the settings shown in the window.

For DirectX



PROCESSING

Use the Preview Status display to view the current processing status of the plug-in. If fusion:FILTER is currently processing your edits, this area says "PROCESSING," otherwise the area is blank.

▶ Trigger Button (DirectX)

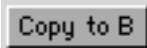
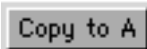
A rectangular button with a light gray background and a thin black border. The word "Trigger" is centered on the button in a black, sans-serif font.

Click this button to retrigger all the modulators. This is particularly useful for retriggering envelopes or LFO ramps in the middle of an audio file.

The **Trigger** button always triggers the modulators at the NEXT quarter note, assuring that your envelopes and music stay in sync.

Since the idea of a quarter-note is dependent on tempo, and different modulators can have different tempos, the latency will always use fusion:FILTER's master tempo, which defaults to 120 bpm, but can be changed in any patch by setting a new tempo in any tempo numerical, then alt-clicking (option-clicking) that tempo numerical. This sets both fusion:FILTER's master tempo and all other modulator tempos to this value. For more information, see either ["Tempo" on page 40](#) or ["Tempo" on page 46](#).

▶ Setup/Copy to Buttons (All)

A rectangular button with a light gray background and a thin black border. The text "Setup A" is centered on the button in a black, sans-serif font.A rectangular button with a light gray background and a thin black border. The text "Copy to B" is centered on the button in a black, sans-serif font.A rectangular button with a light gray background and a thin black border. The text "Setup B" is centered on the button in a black, sans-serif font.A rectangular button with a light gray background and a thin black border. The text "Copy to A" is centered on the button in a black, sans-serif font.

These buttons work as a "compare" feature, allowing you to compare one filter setting with another. Basically, the filter gives you two memory buffers, labeled **Setup A** and **Setup B**.

Whenever the button says **Setup A**, you are editing the parameters stored in Setup A -- you can copy them to Setup B by clicking the **Copy to B** button. Whenever the button says **Setup B**, you are editing the parameters stored in Setup B -- you can copy them to Setup A by clicking the **Copy to A** button.

Press the **Setup** button to switch back and forth between the two different sets of filter parameters.

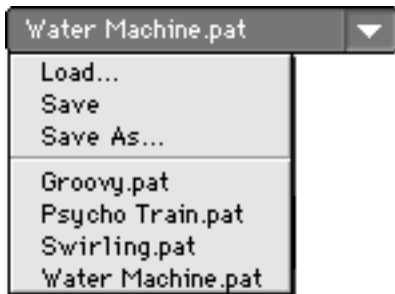
Use the **Copy to** and **Setup** buttons together as follows:

When you first open fusion:FILTER, the buttons read **Setup A** and **Copy to B**. The parameters you first begin editing belong to Setup A. Any changes that you make to these parameters are automatically remembered by Setup A. If you create a filter patch that you like, click the **Copy to B** button, which copies your parameter setup into Setup B. You can continue to make parameter adjustments in an attempt to fine-tune your sound, and these edits continue to be stored in Setup A. At any point, you can click the **Setup A** button to recall the parameter set that you saved when you clicked the **Copy to B** button. You can then switch back to your most recent edits by clicking the **Setup B** button. This lets you compare two different filter patches.

Patch Selector (Premiere & AudioSuite)



Use this area to load or save fusion:FILTER patches. fusion:FILTER ships with a number of factory patches (templates), which you can use as starting-points to build your own sounds. Also, you can save any of your own parameter sets as patches.



To open a patch, simply select it from the Patch Selector menu. Factory patches are stored in "System Folder/Extensions/Opcode Folder/Audio Plug-ins/Filter," and must remain here in order to appear in the Patch Selector list.


Aside from containing a list of fusion:FILTER patches, the Patch Selector menu also has facilities for saving and loading your own custom patches:

- Choose **Load** to open a fusion:FILTER patch that isn't stored in the "System Folder/Extensions/Opcode Folder/Audio Plug-ins/Filter" directory.
- Choose **Save** to overwrite an existing patch with the parameters currently displayed in the fusion:FILTER window.
- Choose **Save As** to name and create a new patch using the parameters currently displayed in the fusion:FILTER window. For example, to create your own patch:


1. Set all filter parameters to their desired values.
2. From the Patch Selector menu, choose the **Save As** command.
3. In the resulting dialog box, type the desired patch name, then click the **OK** button.

If you wish to share patches with Windows users, be sure to give the patch name a ".PAT" file name extension.

Import/Export Buttons (DirectX)

A rectangular button with a light gray background and a thin black border. The text "Import..." is centered in a black, sans-serif font.

Press the **Import** button to import the settings contained in any fusion:FILTER patch into the open plug-in window. Use it to import Opcode's factory-supplied patches, or to import patches created by any version of fusion:FILTER -- regardless of the plug-in architecture or its host application. Factory supplied patches contain a ".PAT" extension.

A rectangular button with a light gray background and a thin black border. The text "Export..." is centered in a black, sans-serif font.

Press the **Export** button to export the current fusion:FILTER settings to a standard patch format, which can be shared with fusion:FILTER customers that use other plug-in architectures (such as Premiere or AudioSuite), or other host applications (Sound Forge, Cakewalk, etc.) Exported patches should be named with a ".PAT" extension.

Cursor Help (All)



Cursor-sensitive help appears here.

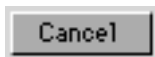
Cursor help appears in the Control module whenever you move your cursor over any element in the filter. You can disable cursor help by clicking the small on/off button at the far left of the text area.

Online Help (All)



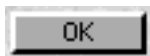
Press this button to open a detailed online Help window for fusion:FILTER.

Cancel Button (Premiere)



Press this button to close the plug-in window and return to your host application, leaving the selected audio file unprocessed.

OK Button (Premiere)



Press this button to close the fusion:FILTER window and pass its current settings to the host application.

At this point, some host applications immediately process the audio and create a new audio file on your hard disk. Other host applications simply apply the settings to the internal audio preview and don't write new audio files until later. See your host application manual to see how it handles Premiere plug-ins.



CREDITS, COLOPHON, & NOTICES

Credits

The following people were responsible for the creation of fusion:FILTER:

Engineering (alphabetically): John S. Cooper; Daniel Steinberg; Dan Timis; Doug Wyatt

Special Engineering Thanks to: David Zicarelli, Mike Berry, Muscle Fish

Product Design, Architecture, and Documentation: Gregory A. Simpson

Additional Art and fusion Logo: Dean Suko

Management: Bruce Nolen; Tim Self

Colophon

This manual was written and produced in Adobe FrameMaker. Graphics were created using a combination of Photoshop, ClarisWorks, and Macromedia Freehand. Adobe Acrobat was used to create this PDF file.

Notices

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