FilterFreak™

Creative Resonant Filtering

soundtoys™

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THANKS!

First off we'd like to give you a loud THANK YOU for purchasing FilterFreak! We know there are an awful lot of plug-ins for you to choose from and we are truly grateful that you have chosen to spend your hard earned scratch (old school for "cash") on our product. We take a lot of pride in our work and we sincerely hope that you find Filter Freak inspirational and musically useful.

We'd also like to thank all the people that helped in the development of FilterFreak and the whole new SoundToys line of plug-ins:

Nick Caiano for his super-cool demo sessions and guitar work.

Mikail Graham for letting us hijack his wonderful collection of guitar pedals (we'll return them soon – we promise!) and for all the help and feedback along the way.

Wendy Letven for her amazing design work on the SoundToys logo, all the SoundToys packaging, and much more.

Andrew Schlesinger for all of his help from the very start of the SoundToys concept – from product names, design ideas, and presets, to his witty and useful contributions to this user guide and other sage advice.

Christy Gustafson for coming through with the 11th hour layout of this manual.

Our whole crew of beta testers for FilterFreak, especially Arthur Alexander, BT, Jez Colin, Andy Gray, Fab, Lars Fox, Terry Fryer, Peter Freeman, Jon Goldstein, Rich Hilton, Michael James, Pete McCabe, Oliver Momm, Allan Speers, Morgan Page, Michael Costa, Andrew Souter, Willie Wilcox, Doc Wiley, and Richard Zvonar.

-- Ken, Bob, Jamey, and Noah

About Filter Freak

FilterFreak is the first in a series of totally cool new plug-ins to come out of the Wave Mechanics / Sound Toys plug-in development lab. A ridiculous amount of programming time, head scratching, listening and tweaking has gone into the development and design of FilterFreak in order to provide you with a wealth of incredibly realistic, powerful and truly "analog sounding" filters. Filter Freak was designed to take filter related sound mangling to a whole new level.

FilterFreak offers unprecedented plug-in filtering power and provides not only one, but TWO filters, each with a host of easily accessible and highly tweak-able parameters that can be used to create some totally awesome filtering combinations. Special attention has been given to recreating a filter that both sounds and responds like a real analog filter. When you overload the input to FilterFreak it won't "freak out" but instead will "saturate" the way a normal filter would in the analog world. This attention to detail is what you will find in all Sound Toys products and we are really pleased with the way FilterFreak works and sounds ...hopefully you are as well!

What and Who is SoundToys

Ok, so what the heck is "SoundToys" anyway? SoundToys is a concept, a vision, a new line of totally cool, sonically superior audio FX plug-in devices designed to set a new standard in regards to plug in processing and sound.

You see all of us at SoundToys (a "division" of Wave Mechanics) are bunch of audio geeks that thirst for new ways to manipulate and mangle sound in ways that are innovative, unique, and with sound quality that is in a class all its own.

While everyone and their grandmother is modeling older gear these days, we are busy wracking our collective brains to develop NEW devices that are truly inventive and will stand the test of time – the kind of things you'd take with you to a desert island...

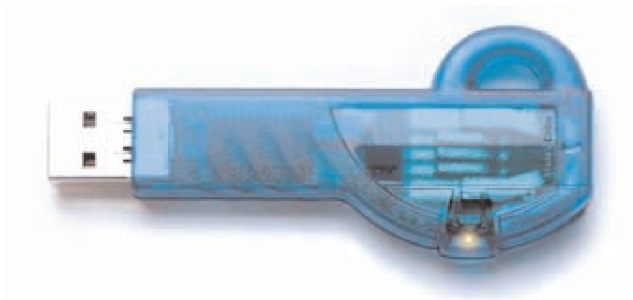
Just so you know, we are the guys that were responsible for designing such groundbreaking and industry standard products as the Eventide® H-3000, DSP-4000, the Wave Mechanics UltraTools line of plug-ins for Pro Tools TDM. Our DSP algorithms can even be found in such high-end studio gear as the TC Electronics Fireworx, G-Force, and G-Major processors, and more.

Our approach in developing the SoundToys series of Pro Tools plug-ins is to create a series of dedicated processors that provide the absolute best possible sound quality, flexibility and creative sound manipulation in the wonderful world of digital, but that also possess that truly analog character and vibe.

So enough of the hype – let's get started, which leads us to...

System Requirements

FilterFreak is a software plug-in for the Digidesign ProTools system. You'll need to have at least one decent Macintosh computer running OS 9.1 or later and of course you'll need a copy of ProTools version 5.0 or later. To use FilterFreak you'll also need one of those little "iLok" hardware keys, – it's a small blue plastic key that looks like this:



If you don't have an iLok we'll be happy to sell you one at a modest price or you can purchase once from Digidesign or your favorite ProTools dealer.

Installing FilterFreak

Installing FilterFreak is really pretty easy. First, make sure that you have your iLok hardware key connected to your computer, and that you have the FilterFreak license card handy. Once you're ready, simply pop the FilterFreak CD into your computer's CD-ROM drive and when the CD icon appears on your screen double-click the FilterFreak installer program to start the process.

Follow any instructions included in the installer and at the end of the installation process you will be prompted to 'authorize' FilterFreak. To do this, carefully detach the license chip from the license card, and insert the chip into the slot in the rear of your iLok when prompted to do so by the installation program.

Once installed, a copy of the FilterFreak™ plug-in will be installed into your Digidesign plug-ins folder along with a cool set of FilterFreak presets. Finally, a SoundToys folder will be created in your Applications folder. The SoundToys folder contains this manual, as well as other useful documentation and tools that you can read or ignore depending on your level of patience or interest in reading babbling things…like this manual.

Registration

Please register your product by going to http://www.soundtoys.com/register

If you chose not to register your product during the FilterFreak installation, please take a moment to register FilterFreak by going to http://www.soundtoys.com/register.

We also strongly recommend that you go to http://www.iLok.com, set up an iLok.com account, and register your FilterFreak authorization.

So, why should you bother with all this tedious registration stuff? When the day comes that your pet boa constrictor swallows your iLok in the middle of a recording session, and you NEED to use FilterFreak, we will be able to get you up and running again MUCH MORE QUICKLY.

If you haven’t taken the time to do this, we will still try to help you, but we’ll have no easy way of knowing whether you are really you, or some bozo who swiped your copy of FilterFreak when you weren’t looking.

What's a Filter? (For those in the know you can skip this section)

Filtering got its start in the world of music with analog synths, and the filter is really the key to what makes those old beasts so cool. Simple put, filters are used to remove parts of the audio signal, and when done well are one of the most useful tools available to shape your sound.

Basics

Note: Since you would probably not be reading this unless you own a Pro Tools system we kind of take it for granted that you have at LEAST a basic working knowledge of your computer and of Pro Tools. If you don't, we strongly suggest you make yourself a strong cup of coffee (or two), get a comfortable chair and familiarize yourself with the ins and outs of your ProTools system before delving too much further into FilterFreak. Digidesign is really nice and provides lots of manuals for your reading pleasure!

Knobs

You can use a mouse to control all of the knobs in FilterFreak and they all work in the same way: To turn a knob up, (to the right, clockwise etc), click on the knob with your mouse and drag the cursor to the right or slide the cursor up (towards the ceiling). To turn a knob down, (to the left, counter-clockwise), click on the knob with the mouse and drag the cursor the left or down (towards the floor). "Mousing" around in a circle doesn't really work; you'll just get frustrated and or dizzy, neither of which is a lot of fun.

Jumping to a Value using Text Markings

Some knobs have text markings showing minimum, maximum, or other values. Clicking on one of these text markings will automatically move the knob directly to that value.

Returning a Knob to its Default Value

To return to the knob's "default value", simply hold down the option key and click on the knob. This will automatically move the knob back to its default value.

Viewing a Knob's Exact Value

To view the exact numerical value of a knob simply hold down the control key and click on the knob. To see the knob's title, hold down control and click on the knob a second time.

Adjusting a Knob with Fine Control

To get finer control over knob values, hold down the apple key while dragging the mouse.

Toggle Switches

In an effort to provide the feeling of real analog gear our toggle switches "switch" when clicked. To change the "state" of a toggle switch (like the analog/digital control), simply click on the switch – it will change from where it was to the "other" setting. Click again to change it back. Really simple!

LED Displays

LED displays on FilterFreak work in a couple of ways:

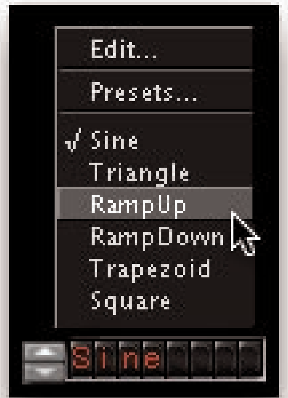
Nudge Buttons

Most text readouts also include a pair of nudge buttons next to the display. Clicking on the upper button will increase the value by one and clicking once on lower button will decrease the value by one. This is useful for tweaking a value by small steps.

Numerical Readouts & Entering Values from the Keyboard

For numerical readouts (like BPM), you can enter exact values right from your computer keyboard. Clicking on the LED display highlights the field and makes it "live". Once highlighted, you can enter a new value via the computer keyboard. Press "Return" to submit the value and deselect the field. You can also click & drag the cursor to change the value. Simply click on the value and drag the cursor up to increase the value or down to decrease the value, similar to using a knob control. Press, "Return" to submit the new value and deselect the field.

Pop-Up Menus & Changing Values:



For most text-based readouts (meaning things that aren't a number), like LFO Wave Shape etc, clicking on the readout with the mouse will display a popup menu. To change a value in the popup, hold down the mouse button and drag to select the desired value and let go. The selected entry will show up in the filed, or a new pop-up window may appear depending on what you have selected.

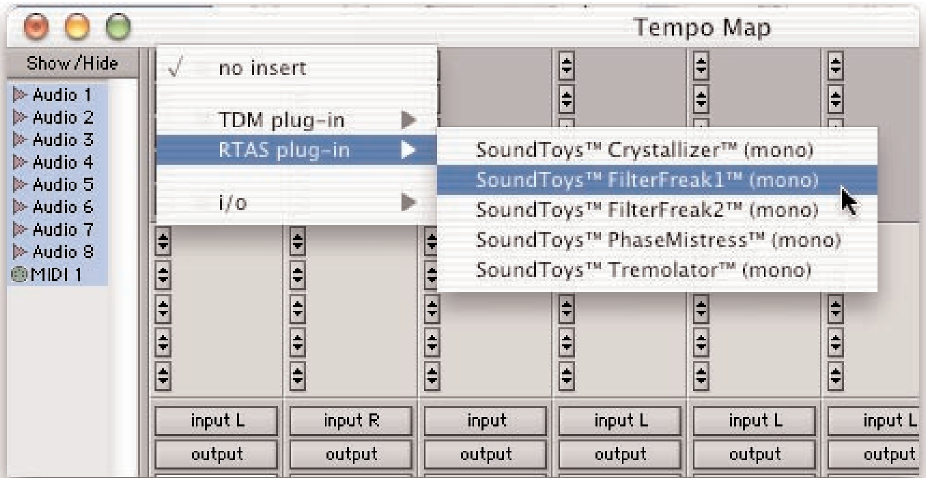
Accessing FilterFreak

Now that you know how to use knobs and pop-up menus, it's time to actually use FilterFreak within Pro Tools. FilterFreak is available to process your audio in two different ways: Real-Time, or Non Real-time.

Real Time Processing (RTAS or HTDM)

As a real-time plug-in, FilterFreak works a lot like a real hardware filter. Whatever sound goes into FilterFreak, comes out filtered, and you can hear it as it is happening (in real time!). Keep in mind that using FilterFreak in this manner always chews up some CPU processing power.

To use FilterFreak in this way, you must first select FilterFreak as one of the insert devices within ProTools, by clicking on the inserts button on a track. You can do this either from the edit or mix window in Pro Tools. From the inserts pop up, select FilterFreak1 or 2: (one and two-band versions of FilterFreak) as shown in the next graphic...



You can insert as many versions of FilterFreak as you'd like, just keep in mind that each one gobbles up certain amount of CPU cycles and can make your computer start to sweat in terms of what's left over for other tasks.

Non-Real Time Processing (AudioSuite)

If you want to save computer power, FilterFreak can also be used to process a specific region of audio destructively. This is done in ProTools using the AudioSuite version of FilterFreak. To do this, first select the region of audio you'd like to process from the Edit window of Pro Tools. Then choose FilterFreak from the AudioSuite menu of ProTools.

Hit the process button to process the selected region with FilterFreak and to write the result back to disk. To hear the effect before committing to disk, click on the preview button.

Using FilterFreak Within ProTools

FilterFreak is designed to be very integrated with Pro Tools, supporting all of the 'standard' Pro Tools plug-in features and controls including parameter automation, MIDI control, etc. Access to these functions can be found in the gray bar that Pro Tools attaches to the top of FilterFreak. The display below shows the real-time version of FilterFreak. When using the AudioSuite version, an additional gray bar will appear on the bottom with buttons to control previewing and processing.

Compare Button

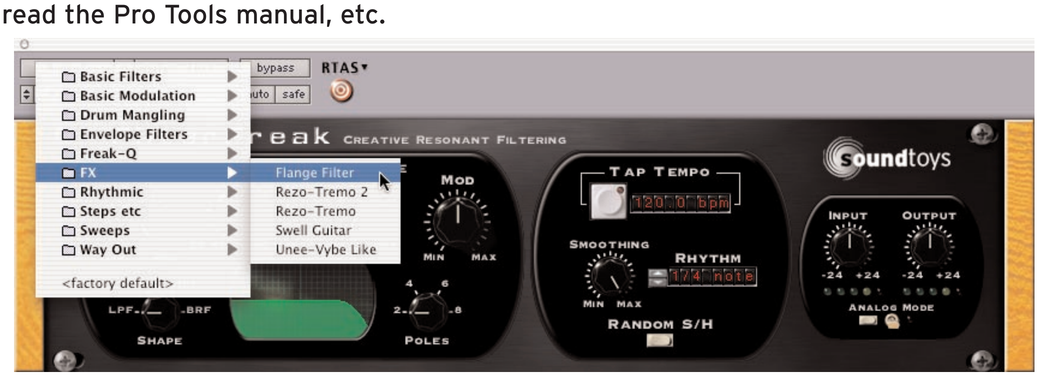
Preset Librarian Bypass Automation Button

The FilterFreak Preset Library

For your immediate listening pleasure and for those who just can't wait, we've included a bunch (hundreds really) of carefully crafted, creative, cool presets with FilterFreak. You might want to check these out as they not only provide a good example of the various types of effects that can be achieved with Filter Freak, but one of them just might be the ticket you need for your latest hit record or production. They also provide a great starting point for you to tweak and mangle to your hearts content. Besides, we spent a whole bunch of time making them so you might as well give them a spin.

Loading a Preset:

To view and load a FilterFreak preset, simply click on the 'Settings Librarian' button located in the standard ProTools plug-in controls area, located at the top of the plug-in window...located in Pro tools, located on your computer, located in your...oops, sorry! And as you would expect, for more info on the Pro Tools 'Settings Librarian', please read the Pro Tools manual, etc.



Saving a Preset:

Saving a preset is just as simple. Click on the small button to the left of the preset name and choose ‘Save’ to update the current preset, or ‘Save As’ to create a new preset.

Compare Button

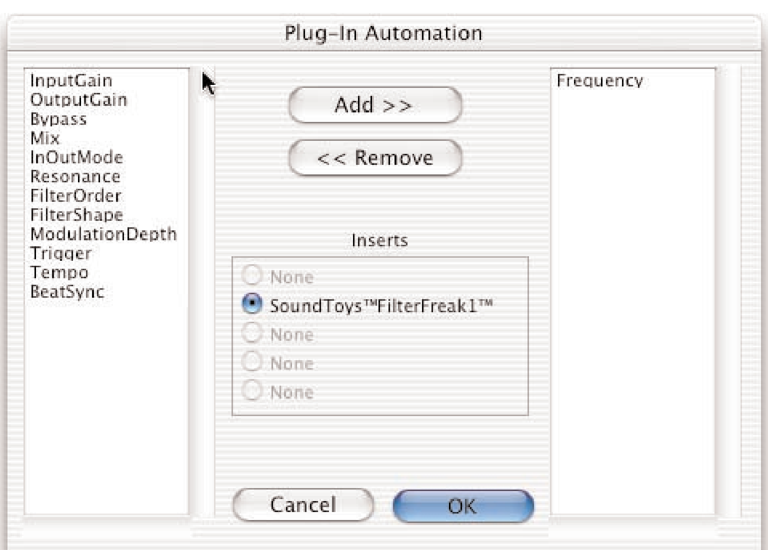
The compare button (often called the compare ‘light’ by Pro Tools pros) is a great way to audition the effect of any changes you’ve made to one of the FilterFreak presets. As soon as you change any parameter in FilterFreak, the compare light will come on. Click on the compare light to toggle between the original preset and your current changes.

Bypass

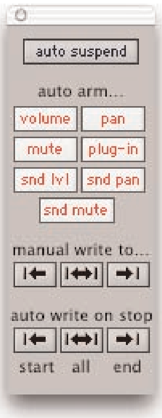
One of the more useful features on any plug-in is the Bypass button. Click on this to bypass the effect of FilterFreak. This also recovers any CPU cycles being used by FilterFreak.

Using Parameter Automation

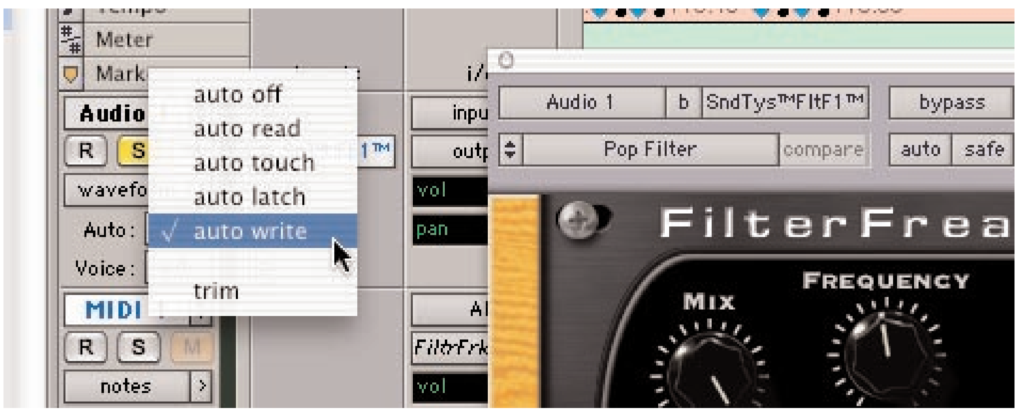
Many of the controls in FilterFreak can be automated. Click on the ‘auto’ button to bring up the Plug-In Automation window. At the left is a list of parameter available for automation. On the right are any of the parameters that are currently automated. Select whichever parameter you’d like to automate then click on the ‘Add’ button. Those parameters will appear on the right side of the window, and can now be automated from within Pro Tools.



To record automation data, you'll first need to do a few things within ProTools. First, bring up the Pro Tools automation window (by pressing - 4) and make sure 'plug-in' is enabled.



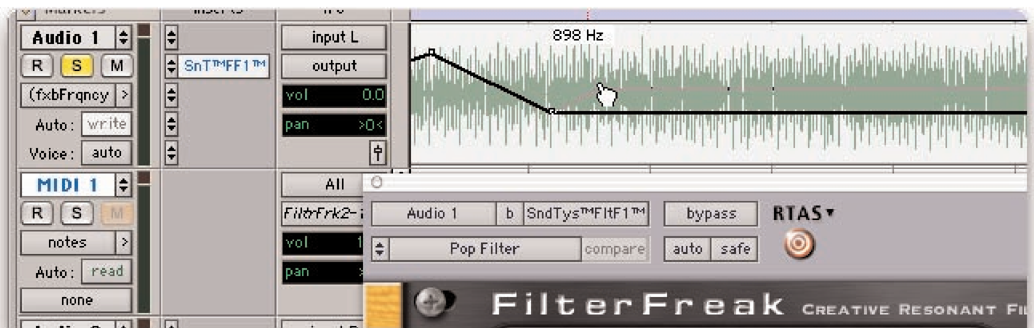
Next, in the track you're working on, select 'auto-write' to enable the real-time recording of automation data.



Now, press play and fiddle with the FilterFreak controls. Pro Tools should now record any control changes you make. When the transport is stopped, the automation will automatically switch to 'auto-touch'. *Please Note: In auto-touch mode, the automation data you've just recorded will be played back, and any new control changes you make will be written over the older automation data.* To prevent these new control changes from being recorded, change the automation mode to 'read'. To turn off all automation on this track, set the mode to 'off'.

It would be really cool to edit these control changes, wouldn't it? Pro Tools has got you covered. From the edit window, click on the waveform button and slide the mouse over until you see FilterFreak and its automated parameters. Choose whichever parameter you'd like to edit (it would be nice to see all parameters at once, but for now you've got to work on one at a time.)

Now you'll see the automation control superimposed on the audio waveform. From this you can click and drag to your hearts content, creating all sorts of wild and crazy filter sweeps!



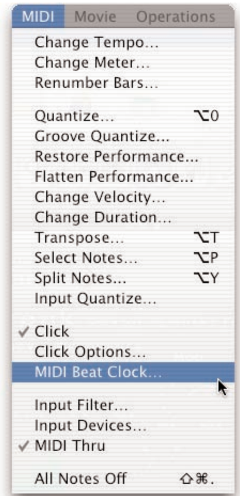
Locking FilterFreak to Tempo and Down Beat (or How to Give FilterFreak a Real Beating)

One of the coolest features of FilterFreak is its ability to synchronize its sweep to the tempo (BPM) and downbeat (the '1' of 1, 2, 3, and 4) in your music. This is a totally awesome way to make FilterFreak hip-hop and groove along with your tunes. Mastering this feature will open up a new world of creative filtering options, especially when you get two different types of filters dancing and sweeping in different directions with different patterns and with different sound.

How to Get Filter Freak to Sync

FilterFreak uses MIDI clock to synchronize its LFO with your music. To sync FilterFreak to your tune...

1. Select ‘MIDI Beat Clock’ from the ProTools MIDI menu.
2. From the dialog box, make sure ‘enable MIDI Beat Clock for…’ is checked and that FilterFreak (and any other SoundToys plug-in) are also checked. You will need to repeat this step every time you insert a new FilterFreak plug-in.

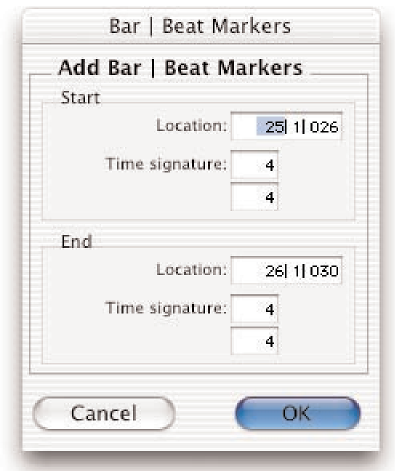


3. Select ‘OK’ to lock to the ProTools tempo and beat clock.
4. But WAIT! There are still a few things to note before this will all work. New Pro Tools sessions always set the default tempo to 120 BPM, and the downbeat will start at the beginning of the session. In order for tempo locking to be useful, the downbeats and tempo of your music needs to line up with the downbeats and tempo in your Pro Tools session. If you've recorded everything to a click track within Pro Tools, everything should be cool. If not, you'll need to enable the conductor track within ProTools and create a tempo map for your song.

Creating a Tempo Map

1. Enable the conductor track by clicking on the conductor button in the transport window of Pro Tools. If you don't see the conductor button on the transport window, Select Display->Transport Window Shows from the Pro Tools menu. Make sure MIDI and “Expanded” are both checked.
2. Create a tempo map by defining bar | beat markers in your song. A simple way to do this is to select the first measure of your song (very precisely) in the Edit window, and then use the Identify Beat command in ProTools (- I) to define the starting and ending beat numbers for the selection. ProTools will then calculate

the correct tempo based on this. If the tempo of your music varies (which it probably does if it wasn't recorded to a click), then you may need to repeat this procedure at various points in your song to keep the beat locations lined up correctly.



Hint: It is very, very useful to use a drum track along with the tab to transient feature within Pro Tools to help with this. If you're a Pro Tools TDM user, you may also want to learn about Beat Detective, which can help to automate this potentially tedious process.

More Info: All of this and more are covered in great detail in the Pro Tools manual. For info about tempo maps and conductor tracks look at Chapter 21 of the ProTools 6 reference guide. For info about Beat Detective see Chapter 22, and for info about the extremely useful tab to transients feature read Chapter 16, Playing and Selecting Audio.

Using the HTDM version of FilterFreak

If you are a Pro Tools TDM user, FilterFreak is also available as an HTDM plug-in, and will appear in the TDM inserts menu of ProTools. The HTDM version is available because RTAS plug-ins cannot be used on aux tracks and cannot have real-time monitoring when used within Pro Tools TDM systems.

The HTDM format allows native plug-ins like FilterFreak to be used just like TDM plug-ins (except that the host CPU is used for processing instead of the DSP chips on the DSP farm cards). One very big caveat here is that the HTDM format causes significant latency (or delay) in the audio stream.

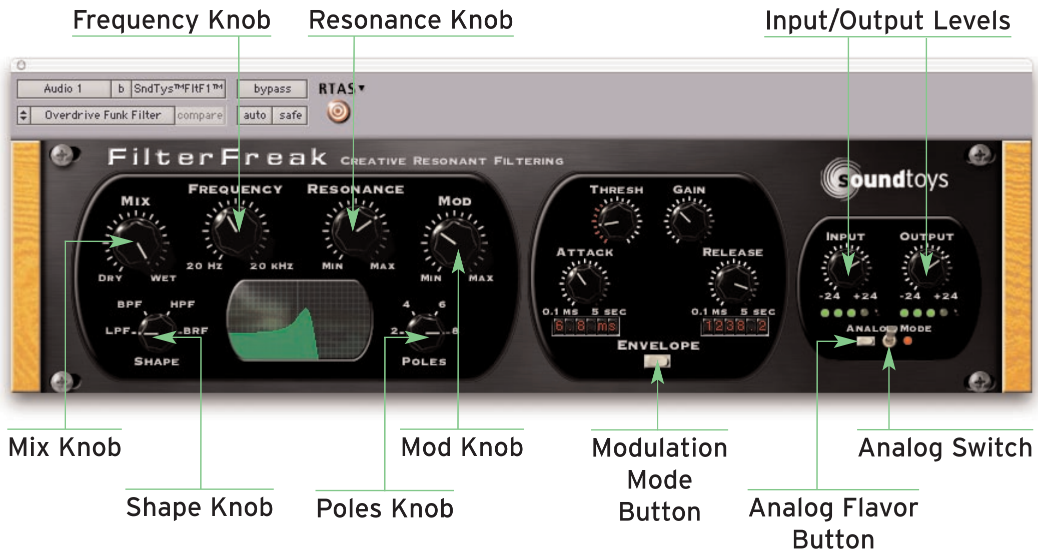
IMPORTANT: Digidesign has a special low-latency version of the "StreamManager" driver for ProTools. This can reduce the latency from 1024 audio samples to 256 audio samples – (from 21 milliseconds to 5 milliseconds at 48 kHz sample rate!) For the latest and greatest info on optimizing HTDM for your system and to download the low latency StreamManager, please visit the HTDM Q&A page on the Digidesign website. The link to it is:

http://www.digidesign.com/support/faq/htdm.html

Optimizing CPU Usage

FilterFreak is a *native* plug-in, which means that it uses your computer's processor to do its audio processing. Because of its advanced DSP algorithms, FilterFreak can put a pretty heavy load on your processor. Here are some tips to reduce the processing:

- Use Digital mode instead of Analog. This uses about half the CPU power.
- Set the poles switch to the lowest setting needed for the sound you'd like to get.
- Use FilterFreak1 instead of FilterFreak2 if you only need one band of filtering.
- If you're running multiple tracks through the same effect, use FilterFreak on an aux track and use sends to route the audio through FilterFreak.
- Commit your effect to disk by using AudioSuite or by bouncing to disk.



Analog Mode Switch (Distortion / Overload Characteristics)

The analog mode toggle switch and flavor button allow you to choose how FilterFreak will distort or "saturate" as the signal input increases, which is more noticeable at high signal levels.

Analog Mode

In 'analog' mode, FilterFreak will saturate in the nice, warm and friendly manner similar to the way real analog gear responds. The analog setting adds a certain amount of distortion at all signal levels. The Analog setting sounds really good but keep in mind the "Analog" setting eats up significantly more DSP resources.
Sorry, no free DSP lunch here!

Digital Mode

When the analog mode switch is off, FilterFreak is in 'digital' mode. In this mode, higher signal levels will clip in the typically nasty, crunchy digital way. This too can be desirable depending on what effect you're trying to achieve. Lower level sounds are left pretty much unchanged and sound spic-n-span clean. Also note that the digital setting uses significantly less DSP horsepower than the analog setting. (What are horses doing in my computer anyway?)

Needless to say you should experiment with both settings using various types of source material and cranking the input levels up and down to hear what it all sounds like. A fun and educational exercise!

Analog Flavor

The analog flavor control is used to adjust how the input and output behaves (or doesn't behave) when it is overdriven. To access this control, click on the small button to the left of the Analog switch to see the analog flavor pop-up menu. For best results, experiment with different sources when tweaking this control:

Clean: Maximum non-distorted range, with fairly hard clipping.
Fat: Smooth low-frequency distortion.
Squash: Similar to above but more compressed.
Dirt: Smooth broadband saturation.
Crunch: Exaggerated high-end clipping.
Shred: Lots of asymmetrical clipping.
Pump: Extreme pumping compression.

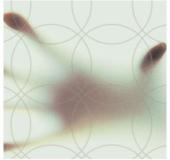
Input and Output Level

The Input and Output level controls are used to (you guessed it) boost or attenuate either the input or output of FilterFreak. The default setting of the controls approximates "unity gain" (what goes in comes out the same level) and should provide the best overall "normal" sound quality when set to these levels. The LEDs beneath the Input and Output knobs provide visual display of the input and output signal levels. The yellow LED indicates that the signal is 6dB below clipping. The red LED indicates maximum signal level, and possible clipping which may or may not sound so good depending on the switch setting and what your ears like to hear.

These controls are particularly useful in the Analog mode, allowing you to control the amount of saturation and distortion in FilterFreak. You can crank either the Input or the Output to create distortion and cranking both to extreme levels can really futz with your sound.

Distortion combined with filtering can be lots of fun. If you saturate the input stage (by turning up the Input level), the distortion and harmonics added by the saturation will be filtered by the FilterFreak filter. If you saturate the output stage by turning up the Output level, the signal will be filtered first and the distortion and harmonics will be added post filtering. These can sound quite different and there are a lot of variations available using both input and output saturation at different levels.

It's important to note that the input and output level *only affects the filtered signal* and leaves the dry signal unchanged. This approach may be slightly different than that of some other plug-ins and we found this implementation to work best for FilterFreak.



Mix

The Mix control is used to set the balance between the filtered sound created by FilterFreak's filters and Input / Output control settings, and the dry signal. Because FilterFreak can mangle your sound in radical ways, it's sometimes useful to be able to mix in some of the original signal. A setting of 100 percent will give you pure filtered sound, and a setting of 0 percent will give you only unfiltered sound. The Mix control provides a convenient means of setting just the right balance between the effect and dry signals right in FilterFreak.

If you're using FilterFreak on an Aux Send / Return configuration, you may want to leave the mix at 100 % and use the fader on the return to control the amount of the effected sound. Please note that in this configuration you may get some phase cancellation due to the slight processing delay introduced by FilterFreak.

Tip: Because of the above it might be preferable to use FilterFreak as a channel insert and use the Mix knob to determine the mix between the dry and filtered sound.

Frequency

The frequency knob is probably *THE* most important control in FilterFreak. This knob determines what area or region of the overall sound spectrum FilterFreak will affect. What effect the Frequency knob has on the sound is greatly dependent on the setting of the **Shape** knob.

When the Shape knob is set to "Lowpass" the filter becomes a "Lowpass Filter" (Duh!) and the Frequency knob is used to set the "cutoff frequency" of the filter. (Again, where along the audio spectrum the filtering affect will occur).

With Shape set to "Lowpass" any harmonics in the sound that are lower than the Frequency knob setting are pretty much "passed" through the filter unaffected. Any harmonics in the sound that are higher than the Frequency knob will be attenuated, cutoff, lowered, turned down, shut out…well you get the picture. So setting the Frequency knob to its maximum value (20 kHz) will pass the entire audio signal mostly unaffected. Setting Frequency to its minimum value (20 Hz) will pretty much make the signal go bye-bye.

Of course most of the really cool effects available with FilterFreak happen when the Frequency is yanked around and moved across the sound spectrum. You can do this manually (as manually as using a mouse can be) simply by clicking on the Frequency knob and adjusting the knob and then recording the changes with Pro Tools automation. For even cooler effects you can also use the FilterFreak's extensive built-in modulation section (Discussed in the Modulation Section) to automatically change the filter frequency in a multitude of different and bizarre ways.

Resonance

The **Resonance** control works in conjunction with the Frequency knob and greatly enhances the effect of the filter by "boosting" the harmonics that are located close to the Frequency setting. So as you change the frequency of the filter those harmonics right around the cutoff frequency are exaggerated, "picked out" and are more pronounced. This gives the filter that squeaky, wheezy sound. Cranking the Resonance all the way up will drive FilterFreak into oscillation (it will actually create a signal), just like those real analog filters do. FilterFreak performs this function with aplomb making the filter squeal, screech and wiggle…which sounds just great!

A Word of CAUTION! – Extreme settings of the Resonance controls can create VERY high signal levels…enough to fry your tweeters and blow out your woofers if the volume is way up. You may want to turn down the output level of FilterFreak when using high values of resonance. We cannot be responsible for any damage done due to being koo-koo enough to listen REALLY loud while cranking the Resonance!
So please…be cool and careful.

Shape

Ok, let's talk shape. The "Shape" knob allows you to select the type of filter that will be will be used and offer four distinct flavors: "LPF" or Lowpass, "BPF" or Bandpass, "HPF or Highpass and finally "BRF" or Band Reject / Notch Filter. We kind of hope that you already have some understanding about filters in general as describing the differences can take lots of words and you probably won't want to read about it now anyway. When it comes right down to it, the best way to understand how the different filter shapes sound is to dive right in and pump some audio though FilterFreak, flip through each shape and twiddle the knobs. You will for sure hear a difference and as they say, a sound is worth a thousand words! But just in case, here's a short description of each of the filter shapes available in FilterFreak.

LPF – Lowpass Filter

The **Low Pass** filter shape will remove or reduce any harmonics *above* the Frequency setting. This type of filter shape is really the most common and is used in most synths and effects. The Minimoog® had a Lowpass filter and the sound is unmistakable once you hear it.

BPF – Bandpass Filter

The **Band Pass** filter shape is like a "cone" and will pass or "pick out" the harmonics in the sound both above and below the frequency setting, passing a "band" of frequencies. As you get further away from the center frequency the harmonics are reduced depending on how " wide" or "thin" the filter's "band" is. The Resonance knob controls how wide or narrow this band is; the lower the resonance the wider the band, the higher the resonance the thinner the band.

HPF – Highpass Filter

The **High Pass** shape is the exact opposite of the Lowpass shape and works in reverse; it passes the harmonics in the sound above the frequency setting and attenuates and removes the harmonics that are below the frequency setting. A highpass filter is useful in removing the bass from a sound etc.

BRF – Band Reject or Notch Filter

The **Band Reject** or **Notch Filter** is also the opposite of a band pass filter and passes all the harmonics in the sound but attenuates the band of harmonics near the filters frequency setting. The "width" of the notch determines how many of the harmonics near the frequency are removed. The Resonance knob controls how wide or narrow this band is; the lower the resonance the wider the notch, the higher the resonance the thinner the notch. In addition to the Resonance knob, the Poles knob also affects how the filter responds as described below.

Poles

The **Poles** knob and associated setting determines how strongly the filter will "filter out" and affect the harmonics either above or below the frequency set by the Frequency knob. This is also referred to as how "steep" the filter is. This function is measured and displayed in "dB" with each Pole equal to a change of 6 dB. The gain reduction of 6 dB is referring to the gain of harmonics that are one octave away from the cutoff frequency. So in a 1 Pole LP filter the harmonics one octave above the cutoff frequency will be lowered by 6dB. Additionally, filter Poles usually come in pairs, i.e. 2, 4, 6, and 8 Pole etc. The more poles there are, the greater the attenuation of the harmonics that are one octave above or below the setting of the Frequency knob (depending on the type of filter).

The lower the number of Poles the more gentle (less-steep) the filter's slope will be with the effect being more subtle and "smooth". The higher the Pole setting the steeper the filters slope will be with the effect being more pronounced. The most common filters are 2 or 4 poles with have 12 and 24dB slopes respectively. However, having a filter with 6 or 8 poles is really cool as the effect can be mucho pronounced. Of course FilterFreak offers you this option for more extreme filtering affects. Just to clarify all this babble above, here is a list of the Pole settings and the associated slope of the filters...

2 poles = 12 dB / octave (harmonics one octave from the Frequency are lowered by 12dB)
4 poles = 24 dB / octave (harmonics one octave from the Frequency are lowered by 24dB)
6 poles = 36 dB / octave (harmonics one octave from the Frequency are lowered by 36dB)
8 poles = 48 dB / octave (harmonics one octave from the Frequency are lowered by 48dB)

Note: At lower pole settings FilterFreak doesn't have to work quite as hard and will not use as much CPU power!

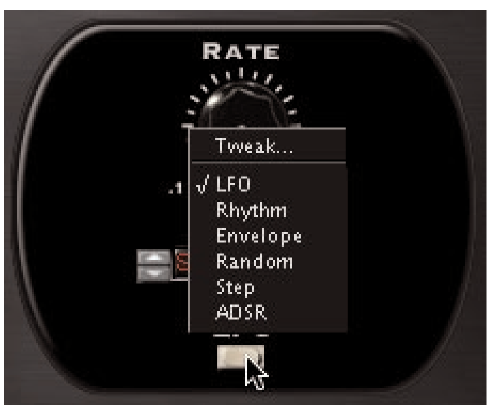
Mod

As we mentioned earlier the real action starts when you pump audio through FilterFreak and tweak the knobs. But what is even more cool is when you "modulate" the filter automatically with various control sources so you don't have to sit there and turn the knobs up, down, up, down...which is kind of silly and gets really tedious really fast. So it should come as no surprise that FilterFreak provides a "modulation" section that allows you to dynamically adjust the filter frequency rhythmically, in response to the audio input, or in a whole bunch of other interesting ways.

The **Mod** knob is used to control the "depth" or amount of modulation applied to the filter's Frequency. The higher the Mod knob setting the greater the modulation depth and the greater the filter sweep, the lower the Mod setting the lower the amount of sweep or modulation. Keep in mind that how the filter is swept is based on the type of modulation signal used.

MODULATION CONTROLS

Choosing a Modulation Mode



When is comes to modulation the more sources you have the greater the sonic possibilities. FilterFreak includes a number of different modulation "Modes", called funny enough "Modes" that allow you to modulate FilterFreak in a variety of cool ways. To select one of the Modes, click on the "**Mode**" pop-up button, drag the mouse to highlight the desired Mode and let go.

LFO Mode

The "**LFO**" mode stands for Low Frequency Oscillator. An LFO creates a repeating waveform (usually with a selection of various wave shapes) that oscillates at a rate between 0 and 20Hz. This is why it is referred to as a "Low" frequency oscillator as 20Hz is still considered pretty slow in comparison to an audio signal. However, the FilterFreak LFO has a slightly wider range and can go as fast as 100 Hz. Modulating the filter with a repeating LFO can provide the ubiquitous "Auto-Wah" type of effect as well as many other effects depending on the type of modulation, wave shape, filter, Poles, Mode, Resonance, etc.

Rate

The "**Rate**" knob controls the speed or rate of the LFO's sweep and is displayed in Hertz. A sweep rate of "1 Hertz" (1Hz) means that the LFO modulation will repeat once per second. As we mentioned, the maximum setting of the LFO in FilterFreak is 100 Hertz (100Hz) meaning it will repeat 100 times per second. Modulating the filter at this speed is pretty darn fast and can create weird "side bands" that are inharmonic making the sound kind of "clangorous". This is "kind of" like the sound created by a Ring Modulator and can be a useful effect. Note that at this speed the sound of the original signal will often be totally warped out and unrecognizable.

Shape

The "**Shape**" nudge buttons are used to select from the list of available built-in LFO wave shapes. FilterFreak includes all the "standard" LFO shapes like sine, triangle and square etc, but also includes a method for you to create you own custom wave shapes, which greatly increases the sonic modulation abilities of FilterFreak. The number of possible shapes you can create is virtually limitless.

Select the Custom LFO Shape Screen:

To create a custom LFO shape, you first need to get to the LFO Edit screen. Click on the LFO shape LED display to display the shape pop-up menu. Select "**Edit**" at the top of the popup menu. This will launch a screen where you can create custom LFO shapes.

Creating a New LFO Shape / Adding "Points" to the Waveform:

Creating a new shape is really pretty easy. To create a new shape you first need to add a new "point" in the waveform (you'll notice that the starting shape is a sine wave with three points: one at each end and one in the middle at the top). To add a new "point", simply click anywhere on the green waveform; this will insert a new point. To remove a point, hold down the option key while clicking on it with the mouse. To change the shape of the wave, click on the new point and drag it up, down sideways etc. You will see that the shape of the green waveform line will change based on where you position the point Letting go of the mouse will position the point where you left it. You can create as many points on the waveform you like and use the "grab/move" operation to reposition any of the points. This process allows you to create some REALLY complex wave shapes.

Smoothing:

The "Smoothing" control allows you to "round out" the edges in the wave shape between the points. When smoothing is set to zero (no smoothing), the waveform will have a stair-step appearance and will jump abruptly from point to point. By increasing the “Smoothing” control you can decrease the abrupt changes between the points and smooth out the transitions between points by varying amounts. When set to its maximum value the waveform will be completely smooth.

Smoothing Mode:

The Smoothing Mode determines the “shape” of the smoothing that will be used to connect the points. This further increases the variety of waveforms you can create within FilterFreak’s Mod section. The Smoothing Mode choices are as follows:

<u>Linear:</u> Points are connected using straight lines
<u>Sine:</u> Produces a sinusoidal-like waveform, which is very smooth.
<u>Exp:</u> Produces a “scooped”, curved waveform where the curve is not even but kind of “rises quickly”, similar in shape to those used in an exponential analog ADSR envelope
<u>Sym:</u> Produces a curved shape that is even and symmetrical.

Naming and Saving Custom Waveforms:

Once you have edited a shape and created a new LFO waveform, the Shape LED display will show the word “custom” indicating that a new shape has been created. To keep your wave shape handiwork from going to data-heaven, you need to save it and give it a name. Pressing the **Save** button allows you to name and save the waveform for future use. Once saved, the new shape will appear in the “Shape” popup menu under the **Preset** menu entry and can be selected as previously described.

Rhythm Mode

The “Rhythm” mode is a much more sophisticated version of the LFO and provides the means to sync the LFO (regardless of the shape) to the tempo of a song in Protools. Using the Tempo mode allows you to produce complex filter modulations that can be programmed in musical and rhythmic ways.

Tempo

You can use Tempo in one of two ways; "Lock to Midi Beat Clock" or "Tap Tempo". To lock Filter Freak to Midi Beat Clock, follow the procedure on page 15. Locking the LFO to Midi Beat Clock insures that the sweeps will lock in march-step to your songs tempo and not waiver.

If you're *not* using MIDI beat clock to lock FilterFreak's tempo, you can simply tap in a tempo by clicking on the "Tap Tempo" button. You can also enter an exact tempo (in BPM) by clicking on the Tempo LED display and entering the desired value with the computer keyboard.

Rhythm

You can also choose specific rhythmic divisions using the "Rhythm" menu. You can select from one of the included simple rhythms by clicking on the Rhythm LED display, which will launch a pop-up showing the available preset options. Simply highlight the desired rhythmic interval with your mouse and let go. You can also use the two nudge buttons to move through the various rhythm choices.

Custom Rhythm Editor:

In addition to the preset rhythms provided, FilterFreak includes a very powerful and unique **Custom Rhythm Editor** feature that allows you to create your own rhythms. To access the Custom Rhythm Editor, select the **Edit** option from the Rhythm pop-up menu.

The Rhythm Editor works a bit like a simple drum machine. By default, the basic rhythm pattern is one bar long and is shown in the rhythm display. For each selected event in the rhythm pattern, one entire cycle of the LFO Shape will be triggered and played.

Adding and Deleting Events:

To add an event in a specific location simply click in the rhythm grid at the desired location and a new event will be added to the pattern. To remove an event, just click on the event you wish to remove and it will go bye-bye.

Changing Event Level and Duration:

To change event level or duration, option-click and drag on any existing event. Dragging up/down will change the level, and dragging left/right will change the duration.

Adjusting Grid Length and Spacing:

You can further adjust the grid spacing and length of any new events using the Grid control.

Pattern Length / Number of Beats per Bar:

Increasing the length of a pattern is easy. To create patterns longer than 1 bar simply adjust the Pattern Length control to the desired number. You can also adjust the number of Beats per Bar for the pattern up or down.

Saving A Custom Pattern:

Once you have created a custom pattern it can be stored and named for future recall. After creating your rhythm, the Rhythm LED display will show the word custom. Pressing the **Save** button will allow you to name and save the pattern. Once saved, the new rhythm will appear in the Shape popup menu under the **Preset** menu entry.

Selecting a Rhythm Preset:

To select a Rhythm Preset, choose the "Preset Menu" option to view the list of available rhythms.

Groove

The **Groove** control is used to adjust the feel and define the amount of "swing" feel to be added to the selected rhythm pattern. Turning the knob will add a '*swing*' feel to the rhythm. Turning the knob down will add a '*shuffle*' feel. This allows you to swing and shuffle all night long! Have mercy!

Shape

The shape control works just like the Shape control for the LFO mode, except that one cycle of the shape is inserted for each rhythm event.

Envelope Mode

The Envelope Mode employs an "envelope follower" that "follows" the volume level of the input signal and dynamically controls the amount of filter modulation based on volume changes in the input signal. The Envelope Mode works great on dynamic signals that dynamically change in volume in rhythmic ways like guitar or drums. Classic effects like the Mutron® used an envelope controlled filter to create its effect. FilterFreak allows you to emulate this kind of effect and a whole lot more using the envelope mode.

Threshold

The **Threshold** knob works similar to the threshold control on a compressor. It defines at what volume the envelope follower will start "tracking" the volume changes in the input signal. As the input level rises above the threshold, the envelope follower 'follows' the signal and will modulate the filter frequency accordingly. The farther above the threshold level and the louder the input signal, the more the filter will be modulated. As the signal falls below the threshold level no modulation occurs.

It is important to adjust the Threshold based on the type of input audio and the amount of modulation you wish to achieve. Setting the Threshold very high will only modulate the filter at loudest peaks of the incoming signal. Setting it too low can cause the filter to be modulated like crazy and jump around following all the changes in the signal. It all depends on the amount of modulation you wish to hear and there is really no right or wrong setting.

Gain

The **Gain** knob works together with the threshold control, and is similar to the ratio control on a compressor. It determines the overall "sensitivity" of the envelope follower, and is used to boost any signal that exceeds the setting of the threshold control. This can be very useful when you are using either a very high threshold, or slow attack settings. At the highest gain settings the envelope follower will begin to function more like a gate, turning 'on' when the input goes above the threshold, and turning 'off' when the signal goes below. At lower gain settings the envelope follower will be more touch sensitive and dynamic.

Attack

The **Attack** knob controls the how fast the filter will react to an increase in signal level. A fast setting will cause the envelope follower react very quickly to transients, and will produce a very dynamic, staccato-like filtering effect. Setting the Attack knob to a slower setting will smooth out and lengthen the attack response the envelope follower. The filter will behave like it has had a few too many beers!

Release

The **Release** knob controls how fast or slow the filter will react to the input signal as it decreases (the opposite of the Attack knob). Again, a faster release setting will produce a more dynamic effect, and slower release times producing a slower, smoother effect as the sound decays.

Random S/H Mode

The Random S/H (Sample and Hold) Mode produces a waveform that jumps from one value to another at each cycle. This effect was used (and abused by ELP) in the beginning of Brain Salad Surgery, you know..."Welcome back my friends to the dinosaur that never ends" as well as other cheesy sci-fi movies to mimic the sound of bleeping and burping computers. However, it is still a cool effect and FilterFreak offers a few twists so that its sample and hold can be synced to the tempo of your music.

Tempo

As before, you can either use Tap Tempo to define the BPM of the random signal or sync it to Midi Beat Clock. When using MIDI beat clock the display will show the current tempo. Use the Tap Tempo as previously discussed.

Rhythm

The **Rhythm** pop-up allows you to set the speed or rate at which the sample and hold will jump to a new value.

Smoothing

The **Smoothing** knob works much the same as it does in the Rhythm Editor and can be used to smooth out transitions between the random values. A setting of zero equals no smoothing and the values will jump from one to the next. The maximum setting will give you a smooth, constantly changing random filter variation sometimes referred to as the "drunken walk" for obvious reasons.

Random Step Mode

The Random Step mode combines the sample and hold effect with an envelope follower. Instead of changing to a new value at a specific set rate, a new random value is triggered when you press the **Trigger** button, in response to a **MIDI note** trigger, or when the input signal exceeds the **Threshold** setting.

This works really great on drums and other highly percussive signals, and can be used to create a creative, dynamic effect that varies with each audio event.

Trigger

Press this button to trigger a new random value. The trigger button will also respond to any MIDI note event.

Threshold

As previously discussed, the **Threshold** knob controls the level at which the input will trigger a new random value. To turn off audio based triggering (if you're using MIDI or manual triggering), turn the threshold knob all the way up.

Smoothing

The **Smoothing** knob controls how fast the filter will move from the previous value to the new random value once the input has passed the Threshold. The minimum setting will cause abrupt transitions to the new random value and at higher settings the filter will smoothly "ramp" to the new value.

ADSR Mode

ADSR Mode is a recreation of the standard envelope generator found on most synthesizers. If you're new to synthesis technology, ADSR stands for Attack, Decay, Sustain, and Release, which also happen to corresponds to the four knobs of this mode. Funny how it works that way!

With the ADSR you can define a specific envelope shape that will be used to modulate the filter each time it receives a trigger *based on the level* of input signal. This is quite a bit different than the Envelope mode whose shape changes and responds dynamically to the input signal

On a keyboard synth, the ADSR envelope is triggered each time you press a key. In FilterFreak, the "ADSR" is triggered either by pressing the **Trigger** button, by receiving a **MIDI note** event, or when the input signal exceeds the **Threshold** setting.

Trigger

Press and hold the **Trigger** button to trigger the ADSR envelope. As long as you hold the button the envelope will move through the Attack, Decay and Sustain portions of the envelope. The modulation envelope will increase the filter frequency and attack to its maximum value, then decay to the sustain value based on the settings of the individual Attack, Decay and Sustain knobs. When the trigger button is released, the envelope will decrease to zero (release) based on the setting of the Release knob.

Threshold

You can also trigger the ADSR based on the level of the input signal. Just set the **Threshold** knob to a value *other than maximum* value. Each time the input signal moves *above* the Threshold level the envelope will be triggered. If you *don't want* the audio to trigger the ADSR make sure the Threshold is set to its maximum value.

A - (Attack)

The **Attack** knob determines how fast the envelope will increase to its maximum value once it has been triggered. The lower the setting of the Attack knob the faster the attack time! As you increase the setting of the Attack knob the attack time will get longer / slower (like increasing the Smoothing parameter) Keep in mind that the possible modulation level goes from a hypothetical 0 to 100%, so the Attack time determines how fast the envelope modulation signal will move from a level of "0" to a level of "100".

D - (Decay)

Once the Attack has reached its maximum value, the envelope moves to the Decay stage and the modulation signal "decays" then until the Sustain value is reached. The **Decay** knob controls the speed or length of the decay time. The *lower* the setting of the Decay knob, the faster the decay time will be. Conversely, the higher the setting of the Decay knob the longer the decay time will be and the longer it will take to reach the Sustain level.

S - (Sustain)

The **Sustain** knob controls at what level (between 0 –100%) the envelope will "hold" or sustain at as long as the Trigger button is held down, or, as long as the input signal is above the Threshold setting.

R - (Release)

The **Release** knob adjusts the time it takes for the envelope signal to move from the Sustain level back to a "0" level once the Trigger button is released, or after the input goes below the Threshold.

Tweak Page

Also included in the Modulation pop-up menu is a unique feature called "**Tweak**". Selecting the Tweak option will open, you guessed it, the Tweak page. This page is where you can tweak and individually adjust the *depth and direction* of modulation applied to the filter's Frequency, Resonance and Output Level. It's important to note that with these controls you can define either *positive or negative* modulation for Frequency, Resonance and Output.

Frequency Depth

The Frequency Depth controls to what degree the selected modulation signal will modulate the filter's frequency, and is set in "Octaves". Setting the Depth control to "+1 Octave" means that the maximum modulation will *increase* and sweep the filters frequency up to one octave above Frequency setting. A setting of "–1 Octave" will sweep the filter frequency up to one octave *lower* than the Frequency knob setting.

Resonance and Level

The Tweak menu allows you to modulate the filter's Resonance and Level along with the filter's Frequency. Normally the Resonance and Level modulation depth controls are set to zero. By turning up the Resonance and Level knobs you can determine how much the Resonance and Level will be modulated. Positive knob values increase the amount of modulation while negative settings will decrease the amount of modulation. By combining differing amounts of Frequency, Resonance and Level modulation an incredible range of filtering effects can be created.

FILTERFREAK 2:

We should probably mention that there is a second plug-in included with FilterFreak called interestingly enough: "FilterFreak2". FilterFreak 2 is identical to FilterFreak 1 except that it provides *TWO* separate filters that can be used together to create an even greater set of filtering effects! All of the previously discussed parameters also apply to FilterFreak2. The filters themselves are identical but can be set totally independently of each other. You can combine a Lowpass with a Band Pass, a Notch with a Highpass, a Lowpass with Highpass, etc to create all sorts of different filter sounds.

Serial / Parallel Mode:

The Serial / Parallel switch determines how the incoming audio signal is fed through FilterFreak's two filters.

Parallel Mode:

In Parallel mode the input signal is sent through each filter individually and the outputs of the two filters are then mixed together. In this mode each filter affects the sound individually and there is no real interaction between the two filters.

Serial Mode:

In Serial mode the input signal is first sent through Filter 1, and the output of Filter 1 is then fed into Filter 2. In this mode, the output of Filter1 is re-processed by Filter 2 and the filters will interact with each other to create "combined" filtering. Depending on the Types filter and Poles selected and their settings it is actually possible to make the sound virtually disappear.

Serial Mode can create some really whacked-out sounds but can also get pretty nasty when the Resonance is set to higher values. This can create some *VERY* loud peaks so please use caution as you could potentially damage your speakers at high volume settings. We have chosen not to limit the creative possibilities with FilterFreak so please keep in mind that a little common sense goes a long way.

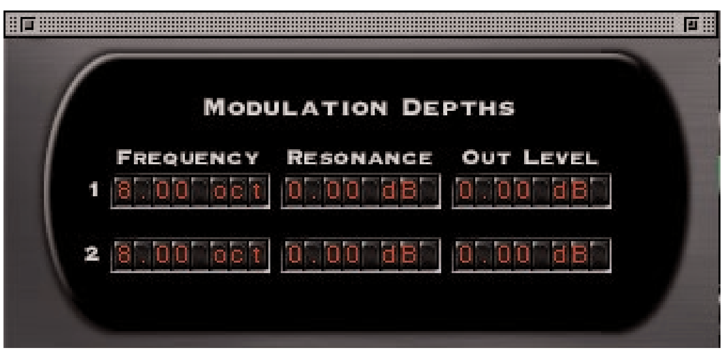
Link

Switching on the Link switch "links" the Frequency, Resonance and Gain controls of the two filters together. Once the "Link" is turned on, moving any of the knobs on *either* filter will move the corresponding knob on the other, in the same direction and by the same amount.

Also note that depending on where a particular knob is set it is possible that *nothing* will change on one of the filters.

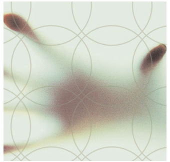
For instance, if a particular knob on Filter 1 is set about half way and the same knob on Filter 2 filter is set to maximum, turning UP the knob on Filter 1, will have no effect on Filter 2 as it has nowhere to go; it's value is already as high as possible. However, using the same example, if you were to turn DOWN the value of the same knob on Filter 2, the corresponding knob on Filter 1 will move as it is not set to it's extreme and has "room to move". This is just one example but hopefully you get the basic idea.

Tweak Page



Like FilterFreak1, FilterFreak2 has a single modulation source, but can get more varied and interesting effects by using different modulation amounts for the two filter sections. You can even set the direction of the modulation for Filter 1 to move in one direction while having the modulation for Filter 2 going in the opposite direction. This can be used to create really interesting vocal-like "Wah" sounds.

The above Tweak page for FilterFreak2 is accessible from the Modulation Mode popup, and is the place to go to dial in modulation depths for both filters.



Summary

Hopefully this manual has provided enough of the information you need to have a basic understanding of how FilterFreak works. We could go on and on providing LOTS of picayune details but you probably would get bored, throw the manual down and start tweaking anyway. Which in all honesty is what you should do. FilterFreak and the whole series of SoundToys plug-ins were MADE for tweaking. There's nothing to break, no parts to rust or replace and you really can't screw it up. What you CAN do is screw up your sounds and that is what FilterFreak is for: creative sound mangling! We sincerely hope you find FilterFreak inspiring and useful in your musical endeavors.

Enjoy!

Getting Help

We offer free technical support for all registered users. We love to hear from you, but if you are having problems, first try to look in the manual or on the support page of our web site for an answer. If you are still stumped, please e-mail us with the following info:

- The product version and serial number.
- The version number of your Pro Tools system, and type of hardware (e.g. Mix, HD, Digi-001, etc.)
- Your computer type and operating system version number (e.g. System 9.1, OS X, etc.)
- A detailed description of the problem

The e-mail address for support is: support@wavemechanics.com

If you don't have e-mail, you can call us at 1-802-951-9700.

Wave Mechanics, Inc.
PO Box 528
Burlington, VT 05402

Phone: 802-951-9700
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A classic analog tape flanger with ultra smooth sound. Reproduce analog flanging effects with the same kind of creative options offered with FilterFreak.

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Decapitator™

Instant Lo-Fi. Add grunge, aliasing and distortion with ease – great on drums!

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Tremolator™

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Wave Mechanics continues to make its UltraTools line of plug-ins for ProTools TDM systems. UltraTools is the most complete package of advanced audio manipulation tools for ProTools, giving you complete power over time and pitch.

PurePitch™

PurePitch is the tool of choice for vocal manipulation on Pro Tools. The first formant-preserving TDM pitch processor, PurePitch is a great tool for creating harmony vocals or altering vocal character.

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Speed™

Jaw-dropping time-compression, expansion and pitch transposition that works incredibly well even on full mixes. Wish we had this when we were trying to learn all those Hendrix licks.

SoundBlender™

An amazing multi-effects processor in a single plug-in. Includes a complete library of stunning effects, from lush chorus and swirling delays to truly mind-bending sonic destruction. What else would you expect from the creators of the Eventide's original H3000 effects processor?

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