Freak - Manifold Filter User Manual

Leonardo Laguna Ruiz
leonardo@vult-dsp.com

Introduction

The Freak Manifold Filter is the result of my obsession with analog filters.

When I started with the idea of building my own synthesizer, the part that I found the most intriguing was the Voltage Controlled Filter. Filters for music synthesis are different from the filters that I learned during my Electrical Engineering studies. Filters for synthesis are required to change their parameters on the fly, while all the theoretical filters I studied were fixed.

In order to create filters that change, it is necessary to change the values of some of it's components (like resistors or capacitors), which in principle is not an easy task. To achieve that, the great designers like Moog and Steiner used the characteristics of semiconductors, like transistors or diodes, to emulate variable resistors. In other designs, like Buchla's, light-dependent resistors were used to achieve the trick.

Those methods of changing the parameters of the filters, combined with the nonlinearities of other stages of the circuit that use semiconductors, result in an "imperfect", "distorted", "nonlinear" filter behavior. These imperfections are what give the personality to every filter design. Non linear filters sound rich and warm compared to the plain and cold linear digital filters.

Over the last years I have been obsessively modeling these filters. Thanks to the knowledge that I have acquired in my real job, which involves making mathematical models and simulators, I believe I have managed to capture the soul of these analog filters and convert them into efficient digital models capable or running in a small microcontroller used as Core of the Freak filter.

These models are not perfect emulations; I have had to made some compromises to achieve good performance. However, these models are very good simulations. The Vult filters have attracted the attention of thousands of persons in the VCV Rack community that use them as mainly drivers to create their sounds.

Quick Theory of Filters

Filters are one of the main tools in subtractive synthesis to transform the sounds. The analog filters found in most vintage synthesizers can be classified by two main parameters: Filter type and Slope.

The **Filter type** refers to which kind of frequencies it lets pass e.g. Low Pass (LP) keeps low frequencies and removes high frequencies. High Pass (HP) is exactly the opposite; it keeps high frequencies and removes low frequencies. A Band Pass (BP) is a combination of a LP and HP. It removes low and high frequencies letting pass only a 'band' of frequencies. Using combinations of LP and HP stages is possible to obtain other filter types e.g. the Notch filter.

The **Slope** of a filter defines how good is at removing unwanted frequencies. The analog filters used in synthesizers do not have a very steep slope. The slope is measured in dB of attenuation. The simplest of the filters (one pole) has an attenuation of 6dB per octave, which means that if we input a sine wave of 1V the signal will be reduced to approximately 1/2V if we double the frequency (make it one octave higher). A slope of 12dB of attenuation results in approximately 1/4 of the input voltage. Every 6dB of attenuation that we add will reduce the amplitude of the signal by the half. In principle, it is not possible to have a perfect filter because it will require an infinite attenuation slope.

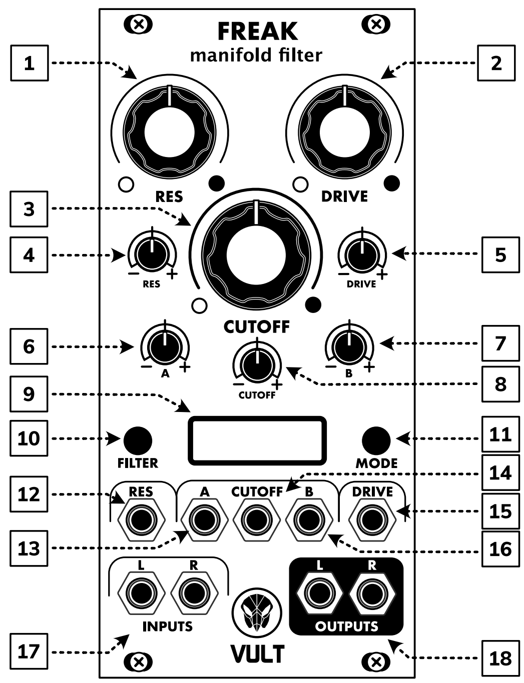
Filters have what is called the **Cutoff** frequency. This frequency is the point at which the signal is attenuated 3dB (approximately 0.7 of gain). Most analog filters used for sound design allow you to control the Cutoff frequency. In modular synthesizers the Cutoff frequency is controlled in Volts per Octave. Similarly to controlling the pitch in oscillators, the Cutoff frequency will double every volt.

Most filters have a feedback control that is better know by the name **Resonance**. The feedback (Resonance) has the effect of boosting the frequencies around the Cutoff frequency. This effect is a very important part of the sound of analog filters.

Increasing the Resonance will increase the boost around the Cutoff frequency. But this boost cannot be infinite. An analog filter cannot produce an output voltage larger than its power supply voltage. The output voltage will be clipped. This clipping is a form of distortion or nonlinearity; and thanks to this nonlinearity analog filters could present **self-oscillation**. When self-oscillating, the filter will behave as a Voltage Controller Oscillator (VCO) and it will produce an output signal even when no audio input is present.

Nonlinear effects can occur in many of the stages of analog filters. These nonlinearities are what give the character to the filters. In order to exploit these distortions we need to drive the filter to its limits. Many filters have the **Drive** control which is basically an attenuator that limits the input signal. If the input signal is large, the filter can produce distortion that can spice up the sound.

Some parameters that are not very often controlled in analog filters are the **Slope** and the **Quality**. Slope tends to be fixed in analog filters however it is possible to have variable slope filters. The Quality is basically the bandwidth of the Resonance. You can find filters that vary the Slope and Quality in Serge Tcherepnin desgins.



Main Controls

1 Resonance
2 Drive (input gain)
3 Cutoff Frequency

Attenuverters

4 Resonance
5 Drive
6 Modulation A
7 Modulation B
8 Cutoff

Display

9 Main screen

Buttons

10 Filter selector
11 Filter mode selector

Modulation Inputs

12 Resonance
13 Modulation A
14 Cutoff
15 Drive
16 Modulation B

Audio

17 Left and right inputs
18 Left and right outputs

Figure 1: Summary of all controls of the Freak module

The Freak Module

The Freak module is a digital filter that simulates the sound of a variety of analog filters. It consists of two individual channels that can be operated in Stereo or Dual mode.

Each simulated filter contains one or more operation modes, e.g. low pass, high pass notch etc. All the parameters can be modulated through the provided inputs and attenuverters.

A summary of the controls and inputs is shown in Figure 1.

As of version 2.0 of the firmware, the Freak filter includes distortions and other wave shapers.

Common Filter Parameters

The majority of the filter models provided in Freak have the following parameters to control: Cutoff frequency, Resonance level and Drive level. These parameters are directly controlled by the knobs with the same name in the panel. Other special filters and distortion models do not have these specific parameter names but the same knobs are used to control them. Check the section "List of Available Models" in order to know how these controls behave for an specific filter.

You can see in the panel of the Freak module that each control provides an input jack and an attenuverter with the same name. In general, these pairs are used to control the modulation applied to the parameters. This is the common case in Stereo mode. In Dual mode the attenuverters control the right channel filter. Check the section "Dual Mode" to get all the details on how the module is operated this way.

The attenuverters and control values mix as defined by the formula:

$$parameter = knob + attenuverter \cdot input$$

Where the values from knobs go from $0 \rightarrow 1$ attenuverters and inputs go from $-1 \rightarrow 1$. The equation above is represented in Figure 2.

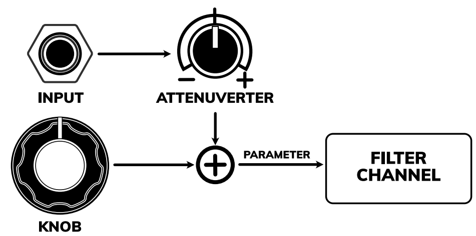


Figure 2: General case of signal mixing involving knobs attenuverters and inputs

Audio Routing

The two channels of the Freak filter are independent and can receive separate audio inputs. However, the right channel input has a hardware switch that when unconnected it will receive the audio input of the left channel. Figure 3 shows the audio routing of the channels.

Navigating the Menus

The FILTER and MODE buttons are used to navigate all menus and options of the module. There is a set of actions that are commonly used in all the menus: Left push (FILTER button), right push (MODE button) and double push (FILTER and MODE buttons at the same time).

In general, the left button (FILTER) will iterate among the displayed menu options. Right button (MODE) will select the highlighted option. Pressing

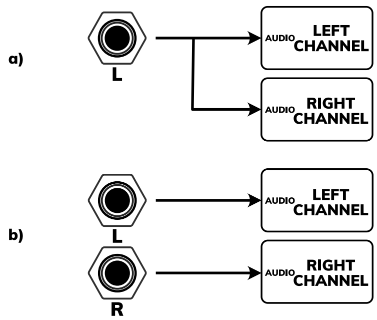


Figure 3: Audio routing a) when the right input is unconnected, the left audio goes to both channels. b) when both inputs are connected, each signal goes into its corresponding channel.

both buttons will take you one level up in the hierarchy of menus.

Figure 4 shows the hierarchy of the menus available in the module.

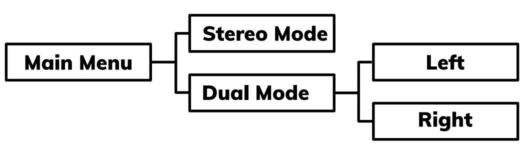


Figure 4: Map of the menus in the module. Pressing both buttons moves one level up. Left button iterates the options, Right button selects the option.

Operation Modes

The module has two operation modes: Stereo and Dual. Dual mode was introduced in v2.0 of the firmware. In order to use it you need to have at least v2.0 of the firmware. The version number is displayed when the module starts. To learn how to update your firmware check the section "Updating the Firmware".

To switch between modes, navigate to the Main menu by pressing at the same time Left and Right buttons as many times as needed. There you should see the option to select either Stereo or Dual mode. You can see the Main menu screen in Figure 5.

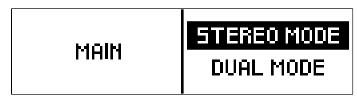


Figure 5: Main menu. Allows selecting either Stereo or Dual mode

Stereo Mode

In Stereo mode the left and right channel run the same filter model and share most of the controls.

The screen will show the current filter, the operating mode and description (as shown in Figure 6). You can use the left button (FILTER) to iterate through the available filter models and the right button (MODE) to iterate through the modes. Check the section "List of Available Models" to get more information about each of the filter models.

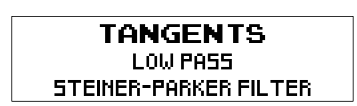


Figure 6: Main screen when working in Stereo mode. It shows the name, mode and description of the filter

In Stereo mode, all controls, attenuverters and inputs (marked with CUTOFF, RES and DRIVE), have effect on both filter channels. The modulations A and B are applied to the left and right channels correspondingly. You can see the signal routing of the controls in Figures 8 to 9.

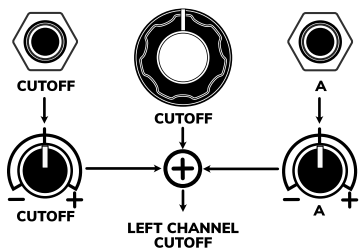


Figure 7: Left channel signal routing for the Cutoff in Stereo mode

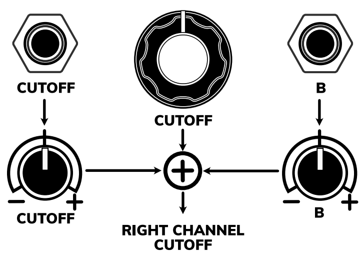


Figure 8: Right channel signal routing for the Cutoff in Stereo mode

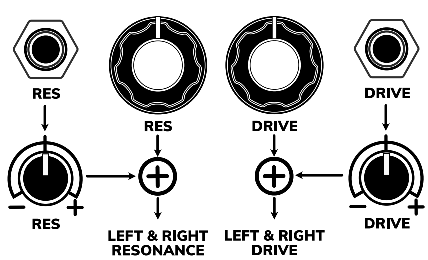


Figure 9: Signal routing for the Resonance and Drive (Left and Right channels) in Stereo mode

When processing stereo signals, it is possible to create interesting effects by modulating the Cutoff of the channels independently through the A and B inputs.

Dual Mode

Dual mode allows selecting a different filter model on every channel and control them independently. When entering Dual mode you have to select which channel is focused, either the left or right (Figure 10). When a channel is focused you can change the filter and mode (of that channel only) using the buttons. The focused channel is marked in the top of the screen as shown in Figure 11. To switch the focused channel, you have to go one level up by pressing the two buttons and select the other channel. See the section "Navigating the Menus" to find more information.

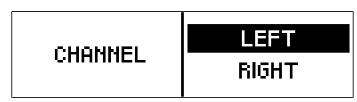


Figure 10: Channel selection menu for Dual mode

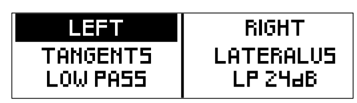


Figure 11: Dual mode. The focused channel is highlighted. Each channel shows the filter and selected mode. Buttons only affect the focused channel

In Dual mode the attenuverters CUT, RES and DRIVE become the main controls for the Right channel. Attenuverters A and B control the Cutoff for the Left and Right channels correspondingly. The inputs CUT, RES and DRIVE have direct control (not attenuated) over the parameters of both channels. In the Figures 12 to 15 you can see the signal routing diagrams used in Dual mode.

Dual mode is great of you want to process two independent signals. But you can also create interesting effects by connecting the two channels either in series or parallel.

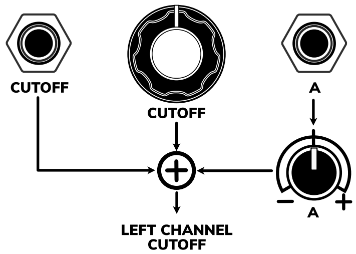


Figure 12: Signal routing for the Left channel Cutoff in Dual mode

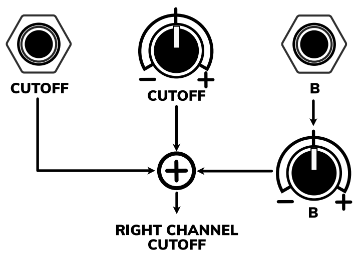


Figure 13: Signal routing for the Right channel Cutoff in Dual mode

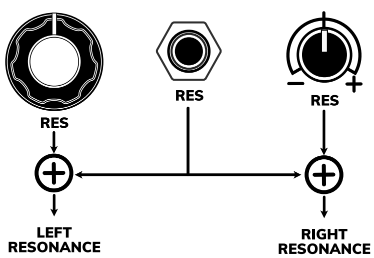


Figure 14: Signal routing for the Resonance in Dual mode (both channels)

Updating the Firmware

We are constantly working on adding new features, filters and effects to the Freak module. For that reason we recommend you to track the updates by subscribing to the Vult mail list or checking the Freak product page.

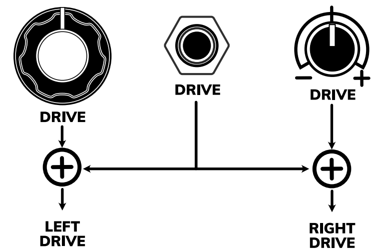


Figure 15: Signal routing for the Drive in Dual mode (both channels)

We recommend you to check the video Vult Freak: Firmware Update Mode where you can see a demo of the whole process.

To update the module you require a way of playing back a Wave file with a sufficiently high level. Playing back the file with a laptop or a phone (through the headphone jack) in most cases is not enough. You can use a specialized audio card (like the Expert Sleepers) or run the headphone signal through a Eurorack module capable of amplifying the level.

IMPORTANT: having low audio signal level can lead the update process to fail. You need to make sure as well that the playback is as raw as possible. Eliminate any effect or equalization from the chain. Music players can have some kind of effect enabled, we recommend using Audacity.

To enter the update mode, press both buttons and hold them while you press the Reset button in the back of the module. Alternatively, you can completely turn off the module, hold the buttons and power the module. When entering the update mode you should see the screen in Figure 16.a.

Once you are in update mode, you will see a screen where you can test the audio level of your signal. Disconnect all the cables from the module except the audio input and play back the file to check if the level of the signal is good. A level around the middle (as shown in figure 16.b) should be good to update.

a)

b)

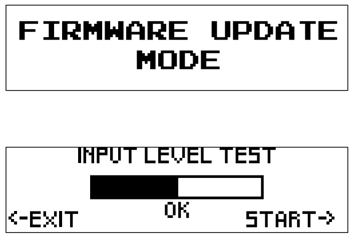


Figure 16: Firmware update screens. a) welcome screen b) audio level test

Once you are ready, press the right button (marked by START in the screen). If you want to abort the mission, press the left one to exit.

Once you start the update process the module will be waiting for the audio signal. Playback the file to start receiving data. You should see the screen shown in Figure 17.a.

IMPORTANT: Once the update process starts receiving data, the old firmware will be overwritten and will not be usable anymore.

a)

b)

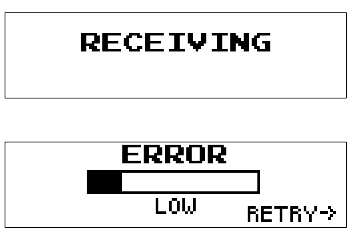


Figure 17: Firmware update screens. a) receiving data b) error in the update

If the update fails you will the the error screen show in figure 17.b. Here you will be able to check the signal level one more time. In order to retry the update press the right button (marked in the screen as RETRY).

If the update process fails, you should be able to enter the update mode again to retry the process as many times as necessary.

List of Available Models

The Freak module provides a variety of filters that share the same control parameters: Cutoff, Resonance and Drive. However some of the models use other parameters mapped to the same controls, for example, the distortions or the Comb filter.

In this section you will find a short description of each model, starting with the common filters and continuing with the special models.

Models with Common Controls

Tangents - Steiner-Parker Filter

The Tangents filter models the Steiner-Parker filter circuit. This model comes in three variations:

- TANGENTS (original)
- TANGENTS-MS
- TANGENTS-XX.

The original version is based on the first Eurorack filter I built. It is derived from the Yusynth design published by Ives Usson. The main changes in my model are that the feedback signal is more controllable in order to avoid the sudden jump to self oscillation in this filter. The MS and XX versions are variations on this principle. The XX goes on the other direction and makes it more aggressive than the others.

All the Tangents versions feature LP, HP and BP modes of the filter.

Lateralus - Ladder Filter

Lateralus is a model of the diode/transistor ladder filter. It's a hybrid model because I could not pick which version I liked the most. The initial version was based on a breadboard circuit I made. This filter features outputs at every stage of the ladder providing 6dB, 12dB, 18dB and 24dB slope options.

This kind of sound can be found in filters from Moog and Roland.

Nurage - Low Pass Gate/Borg Filter

Nurage is a special version of the filter found in the low pass gate designed by Buchla. Some of the variations in my model are that the VCA mode is not enabled and also that the circuit features a high resonance mode. Nurage also simulates the Vactrol behavior of the low pass gate.

This filter has been dubbed (not by me) Borg which is a combination of Buchla + Korg names.

Ferox - CMOS Filter

Ferox is a model of a CMOS filter. This kind of filter uses a "not" logic gate which is typically used in digital electronics. I this case, the logic gate is used as an inverting amplifier. One of the variations found in my model is that the filter is more stable than the original circuit, which makes it more usable. The Wasp synthesizer used a CMOS filter.

Ferox is a filter that presents some distortion. But to spice it a bit more, the Drive control after 50% will blend-in extra CMOS distortion.

Ferox provides LP, HP, BP and Notch modes.

Vortex - Russian Filter

Vortex, is based on the circuit found in the Polivoks synthesizer. This circuit uses a special kind of OPAMPs that allow you to control their bandwidth. The original circuit is very minimal, but modeling it proved to be a challenge.

Vortex provides LP and BP modes.

Stabile - State Variable Filter

Stabile is a well-behaved State-Variable filter. Stabile presents a very clean sound and a nice soft resonance. Sometimes you want the sound to be as clean as possible, for those occasions Stabile is the best.

Stabile provides LP, BP and HP modes.

If you like Stabile, but want a bit more crunch, you can try Unstabile.

Unstabile - Bent State Variable Filter

Unstabile is the circuit-bent version of the Stabile. This model is based on the idea of starving the circuit until the electronic components start behaving strange. In this case, the OPAMPs are running with low voltage providing an interesting distoreted sound.

Unstabile has LP, BP and HP modes.

Vorg - MS-20 Style Filter

Vorg is modelled after the OTA (Operational Transconductance Amplifier) version of the MS-20 filter. More specifically, Vorg was modelled based on the circuit used my Vorg Eurorack module.

One of the differences between the real analog Vorg and the modelled Vorg (included in Freak) is that the modelled version provides a better Drive control that makes it possible to go futher into the distortion mode.

Vorg provides LP and HP modes.

Boomstick - Sallen-Key Filter

Boomstick is modelled after the basic Sallen-Key filter architecture. The original circuit I built was very minimalist; just a couple of capacitors, vactrols and operational amplifiers. However, one thing I removed from this model was the vactrol behavior since I was aiming to get the simplest and best sounding filter model.

Boomstick provides (for now) only LP mode.

Special Models

Rescomb - Resonant Comb Filter

Rescomb is a resonant Comb filter. Comb filters are a special kind of filters that can be obtained by adding to the original signal a delayed copy of the same. The resulting effect is the cancellation some of the frequencies according to the length of the delay. In addition to the basic Comb filter, Rescomb allows you to control the level of feedback inserted in the filter.

Rescomb provides two modes, Comb++ and Comb- -. The Comb mode defines the direction of the Comb. The result can resemble either a group of Notch filters or a group of Peak filters.

The following table shows the mapping of the filter parameters to the Rescomb parameters:

Filter Parameter	Rescomb Parameter
Cutoff	Comb Frequency
Resonance	Feedback Level
Drive	Input Gain

Debriatus - Wave Destructor

Debriatus is a chain of wave-shapers that will help you destroy your input signal. Debriatus features:

- Wave Folder
- Bit Crusher
- Asymmetric Distortion
- Saturation

The four effects can be operated in two modes: DIST+FOLD+SAT (Distortion → Wave Folding → Saturation) of CRUSH+FOLD+SAT (Bit Crush → Wave Folding → Saturation). Each effect is controlled with a single parameter.

The following table shows the mapping in DIST+FOLD+SAT mode:

Filter Parameter	Debriatus Parameter
Cutoff	Wave Folder
Resonance	Asymmetric Distortion
Drive	Saturation

In the CRUSH+FOLD+SAT mode the parameters are the following:

Filter Parameter	Debriatus Parameter
Cutoff	Wave Folder
Resonance	Bit Crushing
Drive	Saturation

It is important to mention that these distortions are not analog models and can introduce a bit aliasing, specially the Bit Crusher.

Summary of Models

The following table summarizes all the available filter models and their corresponding operation modes as displayed in the module:

Filter Model	Operation Modes
TANGENTS	LOW PASS
	HIGH PASS
	BAND PASS
TANGENTS-MS	LOW PASS
	HIGH PASS
	BAND PASS
TANGENTS-XX	LOW PASS
	HIGH PASS
	BAND PASS
LATERALUS	LOW PASS 24dB
	LOW PASS 18dB
	LOW PASS 12dB
	LOW PASS 6dB
NURAGE	LOW PASS
FEROX	LOW PASS
	HIGH PASS
	BAND PASS
	NOTCH
VORTEX	LOW PASS
	BAND PASS
UNSTABILE	LOW PASS
	HIGH PASS
	BAND PASS
STABILE	LOW PASS
	HIGH PASS
	BAND PASS
RESCOMB	COMB++
	COMB- -
VORG	LOW PASS
	HIGH PASS
BOOMSTICK	LOW PASS
DEBRIATUS	DIST+FOLD+SAT
	CRUSH+FOLD+SAT

Licenses

This project uses the following third-party sources:

- The Freak Bootloader is a modified version the bootloader code, Copyright 2014 Emilie Gillet released under MIT license.
- Lua VM is included, Copyright 1994–2019 Lua.org, PUC-Rio released under MIT license.

Technical Specifications

- Audio sampling rate: 96 kHz
- Modulation sampling rate: 6 KHz
- Power consumption:
  - +12V: 110 mA
  - -12V: 13 mA
  - 5V: 0 mA
- Depth: 50 mm
- Width: 12 HP

Final Notes

I am proud to say than all the filter models were developed from scratch and that all the DSP code was written in Vult Language.