Oakley Sound Systems

Filtrex II

PCB issue 1

Analogue Filter Rack

User Manual

V2.1.01

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Oakley Sound Systems FILTREX 2
Analogue Filter
INPUT GAIN
ON
FREQUENCY
RESONANCE
FIZZ
SMOOTH
MIX OUT
DRY
WET
PEAK
VOLUME
FILTER CONTROL
SIDE
MAIN
UP
TRIG
DOWN
ENV DEPTH
ENV MODE
FLR
AR
AD
THRESHOLD
LFO
LFO RATE
AUTO TRIG
OFF
ON
LFO WAVE
SQR
TRI
LFO DEPTH

Introduction

This is the User Manual for the issue 1 Filtrex II rack module from Oakley Sound. It contains a brief history about the development of the module, a guide to the front panel controls and some notes about the power pack needed to power your module.

For the Builder's Guide, which includes the parts list, schematic description and testing procedures, please visit the project webpage at:

http://www.oakleysound/filtrex.htm

For general information regarding where to get parts and suggested part numbers please see our useful Parts Guide at the project web page or http://www.oakleysound.com/parts.pdf.

For general information on how to build our modules, including circuit board population, mounting front panel components and making up board interconnects please see our Construction Guide at the project web page or http://www.oakleysound.com/construct.pdf.

A bit of history...

In 1998 the Oakley Sound Systems VCF-1 was my first PCB project to be released. It proved to be a moderate success and paved the way for the development of a whole range of PCB projects designed for the audio DIY hobbyist. After the very first stripboard prototype, two printed circuit board issues were made. The first issue used Alps 16mm pots and 30 boards were sold. The second issue corrected a few things and allowed the use of the then popular Omeg P16 plastic pots and brackets. Sixty issue 2 PCBs were sold.

In 2002 I decided to update the design and improve certain aspects of it that I thought could be done better. Three things stood out as being key areas to improve.

1. The VCF-1 created smooth sweeps that warmed up any harsh digital sound. But it couldn't do the opposite. It couldn't be nasty, well, not nasty enough anyway. Enter the Filtrex's one pole or Fizz output. This created a bright bubbly sound that had a whole new character. If you then mixed that in with the ordinary four pole or smooth output you would get a band pass response.

2. The envelope follower (EF) in the VCF-1 was quick responding and worked well with most program material, especially keyboards and drums. But it didn't sound right with guitars. The problem was a decay time that was too fast. Experimenting with the MOTM-820 lag processor and the Oakley EFG on a modular synthesiser allowed me to try out various topologies. In the end I settled for a complex, yet simple to operate, envelope follower with fully variable rise and fall times. This combines the functions of an attack-decay/attack-release envelope generator (EG) with a controllable envelope follower. Rise and fall times can now be set independently which made the Filtrex-1 more useful than its predecessor.

3. The lack of a bypass function on the VCF-1 made the unit difficult to use without a mixing desk. The Filtrex-1's Thru function allowed the 'wet-dry' level to be set. Although in practice finding the control's middle null point was somewhat tricky. It did however allow for complex filter responses. Mixing the filter's output with the unfiltered audio input interacts in such a way as to create new filter and phase responses.

The last Filtrex-1 was sold in 2005. There had been a second issue of the PCB in 2004 but this was to be the last run of boards for a while. However, repeated requests for the board to be reinstated finally had their desired effect. The new Filtrex-II rolled off the production lines in November 2009.

The Filtrex-II is more of a refinement of the design ideas introduced with the Filtrex-1 rather than a totally new module. Filtrex-II boards are different only in the following things:

1. Uses 16mm Alpha/ALPS pots instead of the Omeg P16 types.

2. The 'thru' reversible attenuator has been changed to a dry/wet pot. This gives you the same functionality but makes it easier to find 100% wet.

3. Uses the excellent THAT300P instead of the now obsolete SSM2210 NPN pairs.

4. Resonance can now be turned completely to zero.

5. Ground compensated output driver. This is a bit like an electronically balanced output stage but less fussy.

6. The PCB's are blue instead of green.

Overview of the Filtrex

The complete filter rack module contains:

A four pole true analogue filter. This is based on the famous transistor ladder filter developed by Dr Bob Moog in the 1960s. A front panel pot controls the filter cut-off frequency, another controls the resonance. The resonance control is configured not to cause volume drop when turned up. The filter can be made to oscillate if required.

Two filter outputs are available. One is the Fizz output. This is a type of -6dB/octave filter and it gives a unique filtering effect with a real bite. The other is the Smooth output. This is a classic -24dB/octave filter, and gives a smooth warm analogue filter sound. Each output goes via its own reversible attenuator. This means not only can you adjust the output level, but also the phase of the output signal. This allows complex filter types to be made when the two outputs are mixed together in various degrees.

A wide range preamplifier is included. You can use the Filtrex-II with virtually any input from guitars to synthesisers. A peak indicator will allow you to monitor overloads, although the Filtrex-II is designed to overdrive beautifully.

An inbuilt low frequency oscillator with triangle and square waveforms. The frequency can be changed from 0.2Hz to over 40Hz. A LED gives visual indication of the speed of the LFO. Use this to create auto wah, vibrato and phaser type effects.

An ingenious dynamic envelope section that can move the filter cut off point up or down automatically. It can be put into one of three modes at a flick of a switch:

FLR It can function as a useful envelope follower, with controllable attack and decay times. Perfect for simulating many classic dynamic follower filter boxes. Play louder and hear the filter open up.

AR It can be a standard attack-release envelope generator. This is triggered either by the LFO, external gate signal or by an automatic audio trigger circuit with a variable threshold level. Powerful for creating rising crescendos.

AD It can be a standard attack-decay envelope generator. This is a type of one shot envelope, that rises and falls at the presence of an initial trigger signal. Useful for percussive type sounds.

The output level can be controlled with a master volume control. The output circuitry is configured to drive long length cables with ease and offers a ground compensating facility to reduce the likelihood of ground loops.

A side chain input is available. This is a second audio input that can control the main input, but is not heard directly. Use this to trigger the envelope generator, or drive the envelope follower. A common use of this is to make the main audio program respond to a drum machine's output for syncopated effects. Use a deep rich string patch as your main input, and listen to the drum machine automatically pulse the output to the beat.

CV and gate inputs are available for connection to a modular synth or midi-CV convertor. The CV input will control the filter frequency, and the gate control will trigger the envelope generator if selected.

The board is designed to be powered by an external ac supply of 15 to 22V rms.

What all those front panel controls actually do...

The PCB has thirteen rotary pots and four toggle switches to control the filter.

The first set of pots seen on the left up to the first switch are in the main audio pathway. These pots will directly control what you hear from the main output. The pots and switches to the right of these are the controllers. They do not control the audio direct but control voltages or currents in the processor to affect the audio by controlling the filter's cut-off point.

Let us first look at the main audio control pots:

Gain: Input level. This controls the level of the input audio signal to the filter. The design features a fully active pre-amp with variable gain. Use this in conjunction with the peak LED to obtain the best sound quality.

The voltage gain of the pre-amplifier stage can be varied from 0.4 to 122. A gain of 0.4 means that the output of the pre-amp is only 40% of the input level. While a gain of 122 means that the output level is 122 times bigger than the input. In audio circles this would normally defined in dB. This pre-amplifier will give you a gain from -8dB to +42dB.

On: This is the power on light. It will be on if the power supply is switched on.

Frequency: Manual control of filter cut-off frequency. Covers the whole audio range. Fully clockwise opens the filter and allows all signals to go through. Fully counter clockwise will filter virtually everything and very little audio will get through the filter.

Resonance: Resonance or emphasis. Controls the Q of the filter. Fully clockwise the filter will oscillate at the cut off frequency. Not always desired... but can be useful sometimes. There is a trimmer on the circuit board inside, called TRIM, that will control the point at which oscillation occurs. It can be set so that oscillation will not occur at all.

This control allows emphasis of the cut-off frequency. Although the filter is generally thought of as cutting frequencies above a certain point, this pot allows you to accentuate a narrow band of frequencies. The leads to very distinctive 'electronic' sounds.

Note that this version of the ladder filter will not oscillate at very low frequencies even if the TRIM and Resonance controls are set to their maximum positions.

Fizz: Output level of the one pole filter output. This is a reversible attenuator. The pot is at its minimum position when pointing straight up. Fully clockwise allows the signal to be at full strength with no inversion of the signal. Fully counter clockwise allows the signal to be at full strength but with inverted properties. The two types actually sound identical on their own, but when mixed with the other two outputs strange new filter sounds can be made.

Smooth: Output level of the four pole filter output. As with FIZZ, this is a reversible attenuator.

Mix Out: Controls the wet/dry mix. That is it adjusts the ratio between the pre-amplifier output and the output of the combined smooth and fizz signals. Set the Mix Out pot to 'wet' to hear just the output of the filter. Set the pot to 'dry' to hear just the output of the pre-amplified input signal with no filtering.

The Fizz, Smooth and Mix Out pots control the audio signal, and it will take a little bit of practice to understand how these controls affect the final output signal. When each is used on its own, the results are predictable enough. They act a simple level controls. However, by mixing the signals together new and interesting effects can be heard. The simplest one to try is setting the Mix Out to ‘wet’, the Smooth to full ‘+’ and the Fizz to full ‘-’. This creates a simple band pass effect. What you will have here is not the traditional low pass filter sound, but just one band of enhanced frequencies. You can sweep these up and down manually with the filter frequency pot, or let the Filtrex do it all automatically with its many modulation routes. At increased resonance, the output can become nothing like the original input signal.

Peak: This led glows fiery red when the filter starts to clip. However, the ladder filter actually sounds very good when overdriven, so this may be what you want. Generally the GAIN pot should be set so that this LED just flickers occasionally when in normal operation. For overdriven sounds it is usually best to stick with simple monophonic input signals. Playing complex chords through an overloaded filter can be too harsh for most people.

Output: The final output level is controlled by this pot. The output amplifier of the Filtrex is set quite high, so in normal use this pot would normally be at the 1 o’clock position.

That completes the description of the audio controls at this stage. The rest of the pots and switches control the envelope and low frequency oscillators. These form part of the control circuitry and these will be described now.

Control: This is a simple two way switch that allows the filter’s processing circuitry to be controlled by the main input or a second input, sometimes called the ‘side chain’. The side chain input is never heard directly from the main output, but when selected it will be able to control the envelope generator or follower.

Up: Controls the speed at which the envelope generator or follower ramps upwards to its maximum output value.

Down: Controls the speed at which the envelope level falls from its maximum value back to zero.

Env Depth: This controls the amount that the envelope output controls the cut-off frequency of the filter. It is a reversible attenuator, so straight up produces no modulation, and therefore no movement of the filter frequency. Fully clockwise will produce rising cut-off frequencies. Fully anti-clockwise will produce falling cut-off frequencies.

Mode: This is a three position switch and its simple function hides some quite clever circuitry behind it. It controls the mode of the envelope section and can be set to FLR (follower), AR (attack-release) and AD (attack-decay).

The follower mode is similar to the envelope follower filters like the Mutron and Doctor-Q guitar stomp boxes of the past. Simply put, the input is analysed for volume, the louder the volume of the incoming signal, the bigger the envelope follower's output. This essentially means that loud sounds will move the filter frequency more than quiet sounds.

The key to a good follower is the speed at which it reacts to the input signal. In the Filtrex, the 'up' control determines the speed at which the follower's output voltage rises. Set this to its maximum value and the envelope output will slowly rise when the signal is present. Set to the minimum value, the output will rise very quickly indeed. The 'down' control affects the speed of which the output falls once the input signal is no longer there. Getting the 'down' time right will be crucial in getting the sound that you need. Set this too fast, and you will hear an odd stuttering from the filter. Set it too long and the envelope output won't shut down fast enough to respond to the changes in the music. The actual correct setting of these two pots is determined by the music material you are putting into the Filtrex and the sound you require. Experimentation will lead you to learning how this important section works.

Both the AR and AD modes refer to the operation of the envelope generator or EG for short. This is similar in some ways to the operation of the follower, but different in one major thing. It is not the audio input that controls the EG circuitry but a gate signal. A gate signal is either on or off. Remember an audio signal is a true analogue signal and can be many values, not just the two extremes of on and off.

The gate signal is derived from several sources; the LFO, the threshold detector and the external gate input. Each of these will discussed in detail later on, but for now, we will consider just the operation of the EG.

When a gate signal is received from any of the three sources, the attack mode is started. This means the output of the EG will rise from zero to a fixed peak value. The time taken to reach this peak value is determined by the 'up' control. If the gate is removed at any point during this phase the release phase begins and the output falls. The speed at which the output falls is controlled by the 'down' pot. Assuming the gate signal does stay high, and the peak signal is reached, what happens next depends on the mode the EG is in.

In the AR mode, the output will stay high as long as the gate is active high. In other words the output is 'sustained'. Once the gate is turned off and goes low, the output of the EG will fall.

In the AD mode, the output of the EG will drop as soon as the peak is reached. There is no sustain and the release mode is started prematurely even if the gate is still high. This premature release mode is correctly called decay.

AR mode produces sustained effects like an organ. The AD mode produces percussive sounds like a guitar or marimba.

Threshold: The audio-trigger circuit. The peak level of the audio signal, either the main input or the 'side chain', is analysed in this section. If the audio input is higher than the level set by the threshold pot, then a gate output is produced that can trigger the EG if turned on. With the pot set less than fully clockwise, a loud signal is required to trigger the EG. With it set fully anti-clockwise very quiet sounds will trigger it. However, setting it too low can cause false triggering. This pot's operation can get some getting used to. Stick with this, the results are worth it.

The effects of the audio-trigger circuit can be turned off by simply setting the threshold pot to its most clockwise position. No matter how loud your incoming signal the audio-trigger circuit will not detect any audio with the pot in this position.

LFO: This little bi-colour LED responds to the output of the LFO. It will gently pulse in brightness and colour according to the speed of the LFO.

LFO rate: Low frequency oscillator (LFO) frequency. Controls the frequency of the LFO. From about 0.2Hz (slow) to 30Hz (fast). Great for producing 'wah-wah' and trancey swishes when the filter is set to self oscillate.

Auto Trig: LFO trigger. The LFO will trigger the EG automatically. The Trig LED will flash at the speed determined by the LFO frequency when this is turned on.

LFO wave: This controls the waveform that will modulate the LFO via the LFO depth control. Triangle or square wave outputs. Triangle will move the cut-off frequency up and down smoothly, like wah-wah. Square wave will move the cut-off point rapidly between two points, creating 'bip-bip-bip' sounds.

LFO depth: Controls the amount that the LFO can affect the filter.

That completes the overview of the front panel operation.

Power Supply

The recommended option is to use an insulated wallwart or AC adapter. These can be bought from most places and are used external to the Filtrex housing. They are very safe since all the nasty dangerous stuff is kept inside the wall-wart. You won't hurt yourself with the output from one of these unless you stick it in your mouth!!

You need a 15V or 18V alternating current (AC) output at 250mA or higher rating. Do not use a DC output type. Although the latter are the most common type of wallwart for guitar effects pedals, they will not work with the Filtrex. To reiterate, because this is really important, it must say 15VAC or 18VAC on it somewhere.

In the UK they can be bought from Maplin Electronics. The one we recommend is their part number N57AT. This is a variable supply which means it can supply a variety of different voltages. The adjustment is on the underside of the unit. Simply set this to 15V with a small blade screwdriver.

Some 12V AC output types may also work but this is only because some AC output wallwarts tend to be poorly regulated and have a lot of overhead. Do be aware that if they do work some of the time they may not work all of the time. If the voltage does fall below the required operating voltage of the Filtrex-II the most obvious sign is an audible hum from the outputs. You will not damage the Filtrex-II by doing this although any connected amplifier or speaker system may object to the humming.

Final Comments

I hope you enjoy using the Oakley Filtrex-II.

If you have any problems with the module, an excellent source of support is the Oakley Sound Forum at Muffwiggler.com. Paul Darlow and I are on this group, as well as many other users and builders of Oakley modules.

If you have a comment about this user guide, or have a found a mistake in it, then please do let me know.

Last but not least, can I say a big thank you to all of you who helped and inspired me. Thanks especially to all those nice people on the Synth-diy, Oakley-Synths and Analogue Heaven mailing lists.

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