Oakley Sound Systems

5U Oakley Modular Series

The SVF issue 5

Voltage Controlled State Variable Filter

User Manual

V5.0.1

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The 1U Filter Core Module in standard MOTM compatible format.

Introduction

This is the User Manual for the issue 5 State Variable Filter module from Oakley Sound.

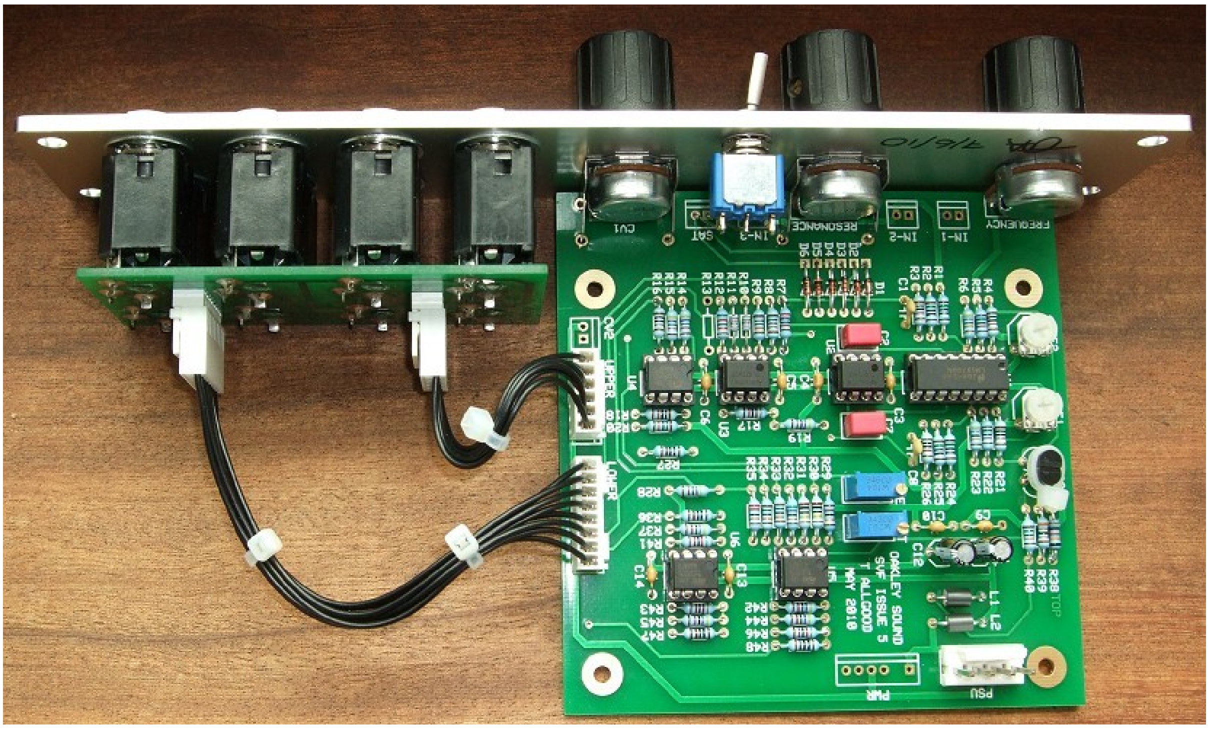
This document contains an overview of the operation of the unit and all the calibration procedures.

For the Builder's Guide which contains information on how to construct the module from our PCB and parts kits please visit the main project webpage at:

http://www.oakleysound.com/svf.htm

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



The prototype 1U SVF in Filter Core MOTM format. This one has been built with a natural finish Schaeffer panel.

The Oakley State Variable Filter

The Oakley State Variable Filter is a two pole voltage controlled multimode filter module. It generates high pass, low pass, band pass and notch responses simultaneously. The cut-off frequency of the filter is voltage controlled, while the resonance is manually controlled with a front panel pot. The filter can be stimulated to oscillate with a sine wave output over the full audio band.

The design is available in either a 1U wide 'filter-core' module or a more fully featured 2U wide panel with seven control pots.

The 1U 'Filter Core' format is our usual way of handling filter modules. Although the 1U module can be used as a filter module on its own, it is expected that users will make use of external mixers to control CV and audio levels going into the filter. In this way, you will be able to have a collection of space saving 1U filter cores that can be used with any generic mixer module. The Oakley Multimix is an ideal choice for a handy mixer module.

The 1U model features two CV inputs, one is fixed at a sensitivity of 1V/octave, while the other is controlled by a reversible attenuator. Two audio inputs are provided which are summed equally into the filter.

For the 2U design three audio inputs are provided each with its own attenuator. Three CVs can control the cut-off of the filter. One is fixed at approximately 1V/octave, the other two have input attenuators. CV1 features a reversible type attenuator with inverting/non-inverting properties.

Power (+/-15V) is provided to the board either by our standard Oakley 4-way header or Synthesizers.com header. Current draw is around 30mA

The resonance mode option allows for the filter to behave differently with large level inputs when resonance is turned up. The NORM mode allows for smooth sounding resonance but the oscillation amplitude at full resonance is limited to just over 3V peak and it does not self oscillate at frequencies below 1kHz. Switching the unit into HIGH mode allows for a much stronger resonant sound. In this mode the unit will self oscillate across the whole audio band with an output amplitude of around 7V peak.

Voltage controlled resonance can be simply attained with an external VCA module. Set the resonance to maximum, patch the BP output to the input of the VCA and then take the output from the VCA to the spare input on the SVF. Now the VCA is controlling the resonance of the SVF. The smaller the gain of the VCA, that is the lower the VCA control voltage or VCA's gain setting pot, the greater the resonance.

The Filter Core Idea

As you have read this module can be made into either a standard 2U wide module, or as a compact 1U filter core module.

The Filter Core idea has come from the fact that many of our customers were buying different filter types, eg. they may have an MS-20 clone, a Moog ladder filter and an SVF. Each filter type gives a different sound so its worthwhile having a few in your modular set up. However, each filter module also has its own input mixer for audio and an input mixer for CVs. This adds to panel real estate and soon your modular is filling up very quickly. While this does look very impressive, it does mean that, in many patches, you have a lot of redundant electronics in your modular.

Step forward the 'filter core'. This is quite simply a 1U module that contains only the filter and a few important front panel pots. All the audio and CV mixing is done externally with a dedicated mixer module, like the Multimix. The good thing about this is that any unused filter module is only 'wasting' 1U of panel space. So you can afford to have many different flavours of filter without the additional cost and panel space of mixers.

However, as with all things, there are disadvantages too. The lack of inbuilt mixers mean that you will need to get more dedicated mixer modules. But remember that these relatively cheap mixer modules can be used for **any** mixing or level controlling within your modular. Thus, you have more flexibility, at the expense of a little more patching.

The great thing about the Oakley Filter Core modules is that they will all be designed so that they can still be used in the full format design. All the Filter Core modules will have input summing amplifiers built onto the PCB. You won't be using these circuits in the 1U format, but they are there if you want to go for the larger 2U or 3U designs.

What is a MultiMode filter?

An electronic filter is a device that lets you separate out some of the elements that make up the whole input. With a filter we can remove parts of the audio signal to give us something that sounds different to the original. Let us take a look at the four basic filter responses:

Low pass: This type of filter will pass all frequencies below the cut-off frequency, *fc*. Above this frequency, the output amplitude or level of the filter will drop as the frequency is increased. The rate at which this output drops is normally determined by the number of active poles in the design. A two pole filter will loose output amplitude by 12dB per octave. A four pole filter by 24dB/octave.

By increasing the **resonance** (or **emphasis** in Moog filters) a band of frequencies around *fc* will be emphasised. This creates a more artificial or electronic sound.

Low pass filtering is the most common form of active filtering in most analogue synthesisers, and generally the most useful.

High pass: This type of filter will pass all frequencies above the cut-off frequency, *fc*. Below this frequency, the output amplitude or level of the filter will drop as the frequency is decreased. Again, the rate at which this output drops is determined by the number of active poles in the design. A two pole filter will loose output amplitude by 12dB per octave. A four pole filter by 24dB/octave.

The high pass filter is very useful in creating thin sounds. This is because most input waveforms are rich in lower frequencies, and by removing the low partials of the sound, you tend to end up with just the *fizziness* of the sound.

A high pass filter will often be used to create string type sounds.

Band pass: This will pass a band of frequencies centred around *fc*. All other frequencies are attenuated. The roll-off on either side of the centre frequency will be determined by the number of poles within the filter. In the two pole case, the roll off is -6dB/octave.

Turning up the resonance control will effectively narrow the band of frequencies passed, making the filter more selective.

This is a very useful response and results in powerful filtering effects. Its more drastic than the low pass filter since it effects the audio on either side of the cut-off frequency. Several band pass filters may be used in parallel to achieve natural resonant or vocal effects.

Notch or Band Stop: This is not often found in music synthesisers but is a useful addition to your synth's musical armoury. The notch filter is in essence the opposite of the band pass filter. It allows all frequencies to pass, except for a very narrow band around the centre frequency. Sometimes it is called 'band stop'.

Sweeping the notch frequency produces a type of effect similar, but weaker, to that of a phaser like the Oakley Equinoxe.

Trimmers

There are four trimmers, or presets as we used to call them in the UK, on the PCB. You do not need any special equipment, other than a decent voltmeter, to set these correctly. However, a digital tuner, or a VST tuner plug in, is very useful for setting the V/octave trimmer.

You should use a proper trimmer tool or a fine blade jeweller's screwdriver for adjusting the two multiturn trimmers. Vishay and others make trimmer adjusters for less than a pound. The two 6mm round trimmers need a small electrician's screwdriver.

OFF1 & OFF2: These trimmers control the amount of DC on the output. Use a scope or a voltmeter to measure the voltage at pin 1 of U5. Rotate the Frequency pot until the voltage is approximately zero volts, ie. +/-10mV. Without any audio input, measure the voltage at the BP output. Set OFF-2 so that the voltage at the BP output is as close to zero as you can get it. You should be able to get it down to +/-10mV or less. Now measure the voltage at the LP output. Set OFF-1 so that this voltage is as close to zero as you can get it. Again aim for +/-10mV or less.

V/OCT: This adjusts the scaling of the exponential inputs. Adjust this so that there is an octave jump in cut-off frequency when the 1V/OCT input is raised by one volt.

Plug a 1V/octave source into the 1V/OCT socket. This may be your keyboard's pitch CV output, or from the CV output of a midi-CV convertor. Set the RESONANCE pot fully clockwise and set the mode switch to HIGH to get the filter oscillating. Now listen to the output coming from the low pass output. Play a note on your keyboard and adjust the FREQUENCY pot so that the note heard is approximately 880Hz – which is two As above middle C. Then play the same note an octave below. Repeat this again and again and adjust the V/OCT trimmer to get the filter's oscillations to jump an octave too. Don't worry about the actual pitch the VCF is producing. Just concentrate on getting roughly one octave difference between the low note and the high note.

It is a fiddly adjustment and it takes a while to get it right. But remember that this is filter and not a VCO, so you don't have to be too accurate.

TUNE: This adjusts the filter's cut-off frequency. Set this so that the filter's FREQ pot covers your chosen range. In a polyphonic modular, this is needed to make each voice's VCF behave identically.

Once that is completed the unit is ready to be used to make music.

Final Comments

I hope you enjoy using the Oakley SVF module.

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 manual, 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 and Analogue Heaven mailing lists and those at Muffwiggler.com.

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