

**Oakley Sound Systems**

**Filter Core Series**

**The SVF**

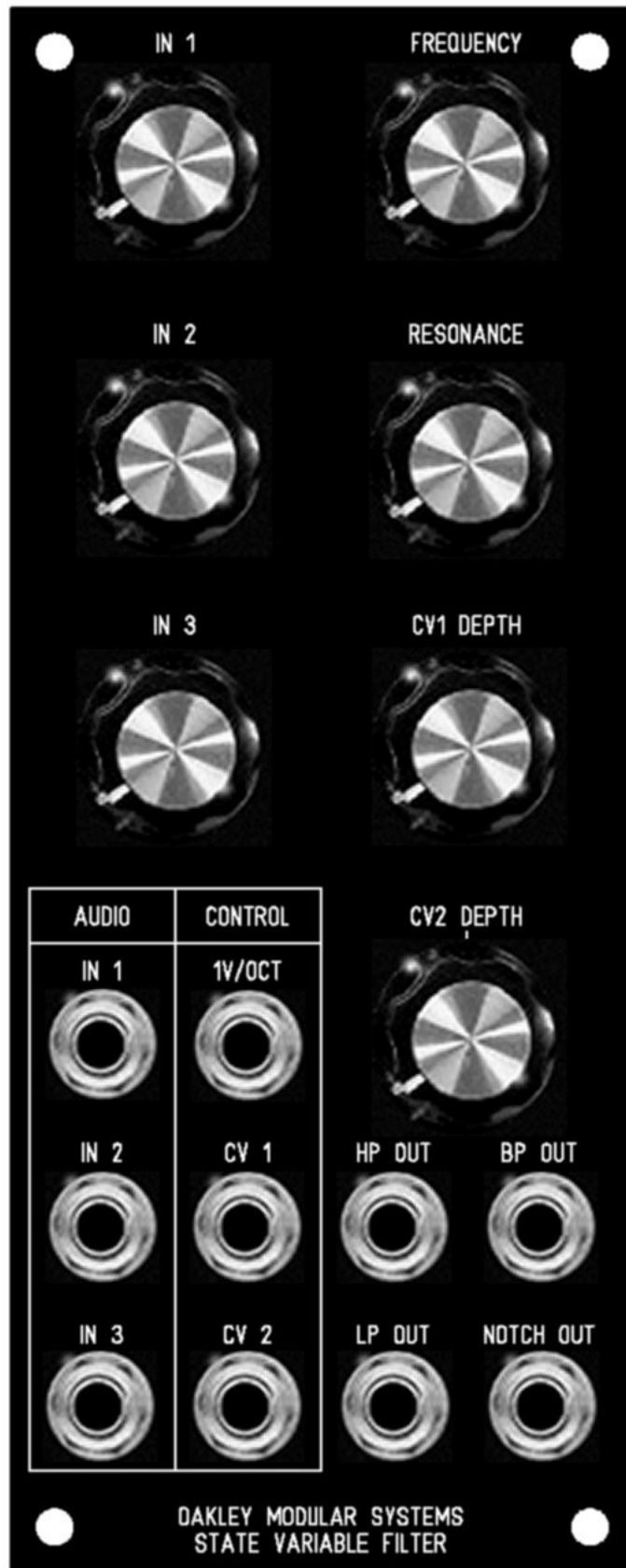
**Voltage Controlled State Variable Filter**

**User's Guide**

**V4.04**

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***Please Note: This User Guide is for issue 4 boards only.***

***This issue PCB is significantly different to previous issues of the SVF , ie. Issues 1, 2 and 3. These earlier issue boards will therefore require a different User Guide and schematic. Older User Guides, as well as schematics, can still be obtained as e-mail attachments from Oakley Sound Systems for £2 each.***

## Introduction

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 sinewave output over the full audio band.

The design is intended to fit into 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 new 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.

Whichever format you chose to build the SVF module, you can also chose from two other suggested alternatives. Firstly, there is the optional Saturation level control. This can either be hardwired into place by a simple wire link, made 'jumperable' with a small 0.1" jumper, or allowed to be selected on the fly with a front panel switch. The saturation option allows for the filter to behave differently with large level inputs and at high resonance. With the contacts closed or linked the filter operates in its default mode. This mode allows for smooth sounding resonance but the oscillation amplitude at full resonance is limited to just over 3V peak. Leaving the contacts left open or the jumper not fitted allows for a harsher resonance and a self oscillation level of around 8V peak.

The second option and new to this issue is the ability to run the active filter electronics in 'soft' or 'hard' mode. The former allows for non linear soft clipping of the two integrators producing a softer overdriven sound when using the module with high level signals or high resonance settings. The 'hard' option is similar to our older issue SVF modules. At low to medium audio input levels this produces a cleaner and quieter output signal, but at the cost of a harder sound

when the module is driven hard. Each of these options must be selected whilst you are building the unit. It is not practical to have these switchable.

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 series of modules is our new way of presenting filter designs and it is the suggested way of making your module. This is especially true if you have not built any of our modules before since the filter core version is a lot easier to make.

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 new 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.

The Filter Core panel design is a lot easier to make of course. This compact panel for the SVF has six sockets and all the wiring is done by using some big solder pads at the bottom of the PCB. For the 2U format, you will need to use these pads and some additional 0.1" headers which are placed near the pots. In a possible future addition to our range, we may provide special pot PCBs to directly attach to these headers to make building the larger modules a lot easier. These pot PCBs have proved very successful in the VCO and other modules where we use them.

## Power Supplies

This module is designed to run from plus and minus 15V supplies. These should be adequately regulated. Although quite large perturbations in the supply will not cause any depreciable effect, DC offset and hence CV breakthrough will be affected by an unregulated supply. The current consumption is about 25 mA per rail. Power is routed onto the PCB by a four way 0.156" MTA or Molex type connector. You could, of course, wire up the board by soldering on wires directly. The four pins are +15V, ground, panel, -15V. The panel connection allows you to connect the metal front panel to the power supply's ground without it sharing the modules' ground line.

This unit will also run from a +/-12V supply with a slight reduction in dynamic range.

## The SVF issue 4 PCB

I have provided space for the three main control pots on the PCB. If you use the specified Spectrol pots and matching brackets, the PCB can be held firmly to the panel without any additional mounting procedures. The pot spacing is 1.625" and is the same as the vertical spacing on the MOTM modular synthesiser.

The PCB has four mounting holes for M3 bolts, one near each corner. These are not required if you are using our specially made pot brackets. The size of the board is 103mm high by 94mm deep.

## 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,  $f_c$ . 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 lose 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  $f_c$  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,  $f_c$ . 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 lose 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  $f_c$ . 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 affects 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 Equinox.

A true multimode filter will offer all types of responses simultaneously. Many however give you only a choice just one at any given time. The Oakley SVF will give you access to all four outputs at the same time.

## Circuit description

The design of the Oakley state variable filter (SVF) is very traditional and I am making no claims for originality. The classic voltage controlled SVF design harks back to the time when manufacturers were looking for an alternative to the moog ladder. Dr. Moog had patented his very clever idea of using the transresistance of the bipolar transistor to create low pass and high pass filtering, so other means of filtering needed to be used.

The state variable circuit had been around for a long time and had been used in variety of filters and oscillator circuits for quite some time. The problem for synthesiser builders was how to make the filter cut-off frequency controllable by an external voltage. Without this type of control, dynamic filtering, the key to a musical instrument filter, could not be realised.

Now it is possible to make a state variable circuit with the variable parameters controlled by variable resistors or pots. This is often used in commercial analogue parametric equalisers. One could even make these variable resistors 'vactrols' and have some degree of voltage control. However, with the invention of the integrated OTA, or operational transconductance amplifier, the implementation of current controlled filters and oscillators became economically viable.

State variable filters are found in various commercial synthesisers. Some examples are the Oberheim SEM and OB range, OSC Oscar, Roland JP-6 and the Transcendant 2000 and Polysynth.

A full description of the state variable circuit is beyond the scope of this User Guide since it requires the reader to be comfortable with either second order differential equations or complex transfer functions. Suffice to say, the SVF core topology can be considered as a pre-prepared circuit block in its own right with the complex mathematics already done for us.

The Oakley design is a second order state variable filter that uses the well respected LM13700 as the OTA element. U2a and U3a form one current controlled integrator. U2b and U3b forming the other integrator. Both halves of U2 are driven by the same current to make the each integrator behave identically as possible.

All OTAs are non linear. This means that they affect the input more than they should and produce audible distortion at their outputs. However, the LM13700 features a simple waveshaping circuit at its two inputs. These waveshaping circuits produce their own non-linearities, but the shape of these non linearities are designed to cancel out the other ones present in the rest of the chip. The waveshapers are 'turned on' by the presence of currents at pin 2 for U2a and pin 15 for U2b. R9 and R16 set the current so that the waveshaper should be able to deal with most signals we will have in our modulars.

The issue 4 SVF module allows the builder to decide whether to use the waveshaping circuitry or not. Leaving it in gives you the 'hard' mode because the although the inputs are linearised over their operating range, once the input levels exceed the ability of the waveshapers' beneficial action, the distortion comes in very quickly. This is more akin to the hard clipping exhibited by op-amp output stages.

By omitting R9 and R16 and changing the values of R7 and R21 you can run the two halves of the LM13700 without additional waveshaping. This means that the inputs will start to overdrive much easier, but in a much smoother and predictable way. In a synthesiser VCA this is probably a bad thing, but in the SVF the resultant audio output is actually quite musical. I will leave it to the builder to decide which one they like the best.

I should add that both modes will distort in the end. The op-amps used in the other audio parts of the circuitry will clip if the combined input signals to the module are too high and you have the resonance settings up full. Op-amp's outputs can only swing between around +/- 12V. Trying to drive them higher than this will just result in hard clipping. However, this may be what you are looking for and you certainly won't do them any harm.

The LM13700 is not perfect in other ways too. Its inputs, in common with most linear ICs, do suffer from a small input asymmetry. This causes a small voltage offset between the two input



pins. It is relatively small in the case of the LM13700, but we can minimise it further by introducing a deliberate offset voltage of the opposite polarity. This is the job of OFF-1 and OFF-2 trimmers. Any untrimmed offset will end up as an unwanted DC voltage on the outputs.

All the filter outputs are available direct from the state variable circuit except for the notch output. This is easily created by adding, or summing, the low pass and high pass outputs together. You would think that this simply reproduces the input signal, but because of phase differences at the cut-off frequency, the two outputs cancel each other out at and around the cut off frequency. This produces the required notch in the frequency spectrum. The addition of the LP and HP outputs falls to U4b, which is configured as a simple inverting summing amplifier.

Unlike the moog ladder, feedback is required around the SVF to prevent it from oscillating. The moog filter is inherently stable and you have to tickle it into oscillation. The SVF topology will oscillate if not enough feedback is provided. The resonant feedback path comes in two bits. Firstly there is the controlled path. This goes via the resonance pot. Applying more feedback keeps the filter stable, so the pot is wired in 'backwards'. The second feedback path is via the little network of diodes. This also acts as a sort of automatic variable resistor. The resistance of which is determined by the voltage across it. When the voltage across it becomes more than a certain amount, the resistance of the network drops and the feedback is applied. This acts a pressure relief valve. If too much resonance is selected by the resonance pot and the unit goes into oscillation, the diode network keeps a check on the output level and keeps it down to a respectable level. With the SAT pads shorted the Oakley SVF will oscillate with quite a nice sine wave at a fairly constant 6.5 volt (peak to peak) across most of the audio band. At frequencies above 10kHz the level of the sinewave becomes larger and the waveform itself gets more distorted.

There is scope for experimentation in the diode network. I have added the SAT pads to allow you to switch between two different zener values. With the SAT pads closed D6 is the predominant zener in the network, and we can effectively ignore D4. This means we have a 5.6V zener diode controlling the feedback level. With this value you get a smooth sounding filter at high resonance. One could make D6 even smaller; early prototypes of the SVF used a 5.1V zener diode. This produces a type of resonant sound similar to the Moog ladder in some respects.

Opening the SAT pads disables D6 from the circuit and D4 takes over. This is an 8.2V zener diode and thus the level of oscillation is higher, around 16V peak to peak, at full resonance. This gives rise to some quite tearing resonant sounds, and is quite dramatic if a little overpowering.

The SAT pads can be connected to a simple SPST switch so you can change modes on the fly. Or you can simply link across it with a piece of wire to permanently set the resonance value to be the lower of the two options. There is some scope too for a variable resistor to be used, but there is no space on the standard front panel layout.

It is crucial if you want your filter to oscillate to make sure you use good components for C1 and C5. 2.5% tolerance, or better, polystyrene capacitors are recommended. If you use 10% polyester types your filter will probably only oscillate at higher end of the audio range. Even

so, the Oakley SVF is more prone to oscillation at higher frequencies due to the various phase shifts caused by the small stray intrinsic capacitances within the op-amps and OTAs. Now it is possible to cludge some additional small value capacitors across R21 and R7. These act as phase compensators and will even out, to some extent, the resonant response over the audio range. Feel free to experiment here, but I actually preferred the sound of the prototype filter without the capacitors in place. Because the degree of compensation is very small the value of each additional capacitor is likely to be below 10pF.

The upshot of the Oakley SVF having no phase compensation means that if you want to make it oscillate at frequencies below 800Hz you have to either, have some sort of audio input connected, or that you briefly take the cut-off frequency to a higher setting and then lower it to your chosen value. Indeed, the Oakley SVF will oscillate down to sub hertz frequencies once oscillating.

Having a filter that can oscillate allows you to generate 'ringing' effects. Ringing can be produced by 'hitting' the audio input with a short pulse from an envelope generator or a fast edge on an LFO. At just below self oscillation, the filter will ring at the cut-off frequency producing a drum or bongo like sound. Fine tweaking the resonance just a bit will alter the decay time of the ringing. Remember again that the uncompensated Oakley SVF rings easier at higher frequencies so adjusting the cut-off frequency will also affect the decay time.

Control of the OTAs is carried out by a current called  $I_{abc}$ . The 'abc' bit standing for 'amplifier bias current'. This is provided by Q2 which acts as an exponential convertor and voltage to current driver in one. Its driven by an emitter follower, Q1. Both transistors act together to eliminate the effects of  $V_{be}$  drift.  $V_{be}$  drift, if uncompensated, would cause the filter's cut off frequency to change with temperature. However, any drift in  $V_{be}$  in Q2, is effectively cancelled out by an opposite drift in Q1. In theory, the transistors must be matched for  $V_{be}$ . However, this is not really necessary in my opinion.

$V_{be}$  drift is not the only source of inaccuracy in the exponential convertors. Drift in the scaling factor caused by a changing ambient temperature can be cancelled by the PTC resistor R30. The PTC resistor's resistance goes up by 0.3% every degree C, and this almost compensates for the -0.35% drift in scaling. Its not essential to get this exactly cancelled in a VCF, since the imperfections create only a slight change in the tonal characteristics as the temperature changes.

U1b acts a summing amplifier taking all the CV inputs together to create one CV line to control the exponential convertor.  $V/OCT$  sets the relationship between the input voltage and the affect on the cut-off frequency. It is normally trimmed so that an increase in 1V on the KBD input will produce a one octave shift in cut-off frequency.

U1a and associating components act as a **reversible attenuator**. This is based on a simple but very effective circuit. I first saw it in the classic text 'The Art of Electronics' by Horowitz and Hill. They probably aren't the originators of the circuit.

The best way to figure out how it works is to think what happens when the pot wiper (pin 2) is moved to each of the ends. When the pot is turned fully counter clockwise, the wiper connects to ground, pin 1. Thus, the non-inverting input to the op-amp is now connected to ground. The pot has no effect on the circuit other than loading the input, which we can ignore.

One of the golden rules of op-amps is that both inputs must be at the same voltage. (This is not actually true, but it is a useful starting point.) If the non-inverting input is at ground, then so must the inverting input. A voltage at the input is turned into a proportional **current** through R4, because one end is at 'ground'. The op-amp acts in such a way as to produce the **same current** through its feedback loop, namely R3. It does this by changing the voltage on its output. Because the resistors are the same value, you need to have the same voltage but of opposite sign appearing at the output of the op-amp to produce the same current. Thus the op-amp is working as an inverting amplifier with a gain of -1. So if the voltage on CV1 was 3V, the output of U1a would be -3V.

Now, let us move the wiper to the opposite side, that is, fully clockwise. This one is harder to figure out. This means that the non-inverting input is connected to the input signal. The golden rule says that the inverting input is also at the same voltage. This means that no current flows through R4. Why? Because you need a difference in voltage across a device to create a current. So if there is no current through R4; there is none flowing through the feedback resistor R3. To do this, the op-amp must produce the same voltage at its output as on its two inputs. Thus the gain of the op-amp is +1. This means a +3V signal going in, ends up as a +3V signal going out. I will leave it to you to prove that the central position of the pot gives no output.

Power is supply via the usual four way MTA or Molex connector. As is the custom for Oakley modules, I have used ferrite beads to act as high frequency filters on the power lines. Decoupling at the point of entry is provided by C6 and C8 for the positive rail, and C7 and C9 for the negative rail. Additional decoupling is also provided elsewhere on the board by the other capacitors shown at the bottom of the schematic. All these capacitors keep the power supply clean of noise, and provide a reservoir for the little bursts of current that the circuit takes in normal operation.

## Buying the components

Most of the parts are easily available from your local parts stockist. I use Rapid Electronics, RS Components and Farnell, here in the UK. The SVF module was designed to be built mostly from parts obtainable from Rapid Electronics.

The resistors can be 5% carbon 0.25W types except where stated. However, I would go for 1% 0.25W metal film resistors throughout, since these are very cheap nowadays. If you do want to use 5% types, the please note that three of the resistors **have to** be 1% or better types. Failure to use good quality parts in these locations will affect the quality of the notch output. These critical parts are indicated in the parts list.

R14 can be a 5% type since getting hold of 3M3 in 1% can be quite difficult.

The pots are Spectrol 248 conductive plastic types and are held onto the boards with specially made Oakley pot brackets. Three pot brackets are required. These are provided with the 'pot mounting kit' which also contains the extra nuts required to correctly fit the panel to the boards.

The PCB is another Oakley board to feature axial ceramics for the power supply decoupling. These are good components with an excellent performance. The PCB legend for these devices features a lead spacing of 0.3". Various types of axial ceramics exist. There are the more expensive C0G types from Farnell, but the other types like Y5V and X7R are perfectly good in this application too. I use Rapid part number: 08-0240.

For the 390pF capacitors in positions C1 and C5, you must use good quality capacitors. I have suggested using 2.5% tolerance 63V polystyrene film types. These are generally axial types, but radial types do exist. These are little silver coloured cylinders, and offer exceptional quality. You may also chose a 1% or 2% polypropylene type. Most of the smaller PCB mounted ones normally have a lead spacing of 5mm so should fit nicely into the PCB.

Another good quality capacitor one could use is the silvered mica type. These are quite expensive but offer exceptional performance in a small package. Failure to use low tolerance parts in these locations will mean your filter will be unlikely to oscillate over the full audio range.

The two offset trimmers are a single turn cermet type. This type of trimmer has a different PCB footprint than other types you may have used. Lead spacing is 0.2" for the track ends, and the wiper is 0.2" away above a line joining the other two pads.

The two multiturn trimmers are the ones that have the adjustment on the top of the box. Spectrol and Bourns make these. Some types are 20 turns, while others are 25 turns. Either will do. They should have three pins that are in a line at 0.1" pitch. Don't chose the older style 10-turn ones with the adjustment on the end, they won't fit on the PCB.

The BC550 and BC560 devices are discrete low noise transistors. The former is NPN, while the latter is PNP. You can replace the NPN with BC549, and the PNP with BC559. Quite often you see an A, B or C suffix used, eg. BC550C. This letter depicts the gain or grade of the transistor (actually hfe of the device). The SVF is designed to work with any grade device although I have used BC550C and BC560C in my prototype. In theory the PNP should be matched for Vbe against the NPN. But I don't think that this is really necessary.

All ICs are dual in line (DIL or DIP) packages. These are generally, but not always, suffixed with a CP or a CN in their part numbers. For example; TL072CP. Do not use SMD, SM or surface mount packages.

U2 is specified as a LM13700, but you can also use the older LM13600 or the now extinct Phillips' part, the NE5517. All should behave identically in this application. However, I have come across one older Raytheon LM13600 device that had very poor matching between the two OTAs within. It didn't oscillate well nor could I make it track at 1V/octave. Its best to use a IC socket for U2 and swap out any duff devices.

I have recently been using the LF412 as a replacement for the TL072 in more critical positions. Although there is very little audible difference I recommend trying an LF412 in positions U3 and U4. I also tried the OPA2134. This is an excellent audio part, but it didn't work so well as U3 probably because of its larger input bias current.

R30 is a PTC or positive temperature coefficient resistor. This means its resistance goes up with temperature. Its there to keep the filter's cut-off frequency relatively stable as the ambient temperature changes. Its a bit of a luxury because its not essential to the working of the filter. If you don't want the expense of fitting it, you can use an ordinary 1K resistor for R30. The PTC I use for this job is Farnell part number 117-4306. Its a 1K +3000ppm/K 900mW device.

L1 and L2 are leaded ferrite beads. These are little axial components that look like little blackened resistors. They are available from most of the mail order suppliers. Find them in the EMC or Inductor section of the catalogues. Farnell sell them as part number: 108-267. Rapid sell them as part number: 24-4860.

Remember if you would like to utilise the saturation function then you will need either a single pole switch, or a two way 0.1" header to which you can fit a jumper to select which mode you like best.

Input and output sockets are not board mounted. You can choose whichever type of sockets you wish. I use the excellent Switchcraft 112 as used on the Moog and MOTM modulars.

Finally, if you make a change that makes the circuit better, do tell the 'Oakley-synths' mailing list or myself directly. Any updates are added to the current user guide as quick as possible.

UK builders should know that there is now a 'Oakley Preferred Parts List' online. This can be found at [www.oakleysound.com/parts.pdf](http://www.oakleysound.com/parts.pdf).

## Parts List

The components are grouped into values, the order of the component names is of no particular consequence. Please read the above section for more details about the parts used in this module.

A quick note on European part descriptions. R is shorthand for ohm. K is shorthand for kilo-ohm. For capacitors: 1uF = 1000nF. To prevent loss of the small '.' as the decimal point, a convention of inserting the unit in its place is used. eg. 4R7 is a 4.7 ohm, 4K7 is a 4700 ohm resistor, 4n7 is a 4.7 nF capacitor.

For pots: A = Linear, B = Logarithmic/audio

### Resistors

All 5% carbon 1/4W or better, except where stated.

**For 'soft' mode: Do not fit R9 and R16 and leave their spaces empty. Make R7 and R21 both 150K resistors.**

**1U Filter Core format does not need R11 so this part can be omitted.**

330R

R5, R6, R19, R20

1K	R32, R33, R35, R37
1K, +3300ppm/K	R30
10K	R15, R8
15K	R9, R16
22K	R7, R21
27K	R25
33K	R29
47K, 1% Metal film	R34, R36, 38
62K	R27
100K	R24, R26, R2, R3, R4, R11
120K	R18
150K	R22, R23, R28, R31
220K	R12
560K	R1
1M	R10, R13, R17
3M3	R14

### Capacitors

100nF, 63V axial ceramic	C2, C3, C10, C4, C11, C6, C7
390pF 63V 2.5% polystyrene	C1, C5
2.2uF, 25V electrolytic	C8, C9

### Discrete Semiconductors

1N4148 signal diode	D1, D2, D3, D5
5V6 500mW zener diode	D6
8V2 500mW zener diode	D4
BC550 NPN transistor	Q1
BC560 PNP transistor	Q2

### Integrated Circuits

LM13700 dual OTA	U2
TL072 dual op-amp	U1
LF412A dual op-amp	U3, U4

### Variable Resistors

100K trimmer	OFF1, OFF2
100K multiturn trimmer	TUNE
20K multiturn trimmer	V/OCT
10KA Spectrol 248 pot	RESONANCE
50KA Spectrol 248 pot	CV1-DEPTH, FREQUENCY
Oakley pot brackets	Three off

## Miscellaneous

0.156" MTA 4-way header	PSU
Leaded ferrite beads	L1, L2
Knobs to fit 6mm shafts	Three off
1/4" sockets	Eight off
DIL16 IC sockets	One off
DIL8 IC sockets	Three off

Hook up wire (26awg) in several different colours.  
Cable tie for holding Q1 and Q2 together.

## Optional Parts

SPDT or SPST toggle switch or 0.1" header/jumper for SAT mode. If you do not wish to have this option it is recommended that you link out the two pads in the SAT position. You link out simply by inserting a small loop of wire, like a resistor lead clipping, between the two pads.

## Additional parts required for the 2U version

### Miscellaneous

1/4" sockets	IN3, 1V/OCT
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### Offboard Pots (2U format only)

50K Log	IN1, IN2, IN3
50K Linear	CV2

## Populating the State Variable Filter Circuit Board

### Warning:

Oakley Modular PCBs are now supplied with the RoHS compliant Ni/Au finish. This is a high quality finish but does possess slightly different soldering characteristics to the traditional lead based HASL finish. Handle the boards with care, and avoid touching the Ni/Au that can cause premature tarnishing of the finish. Shelf life is hard to predict but we recommend soldering in all the components less than one year from when you receive your board.

**We are not responsible for any accidents caused whilst working on these boards. It is up to you to use your board responsibly and sensibly.**

Occasionally people have not been able to get their Oakley projects to work first time. Some times the boards will end up back with me so that I can get them to work. The most common error with most of these was parts inserted into the wrong holes. Please double check every

part before you solder any part into place. Desoldering parts on a double sided board is a skill that takes a while to master properly.

If you have put a component in the wrong place, then the best thing to do is to snip the component's lead off at the board surface. Then using the soldering iron and a small screwdriver prize the remaining bit of the leg out of the hole. Use wick or a good solder pump to remove the solder from the hole. Filling the hole with fresh solder will actually make the hole easier to suck clean!

Sometimes people like to substitute parts in place of my own recommendations. Feel free to do this, but remember that there is normally a good reason why I have selected that particular part. If you do find that, say changing an op-amp with another one, makes an improvement, please do let me know either via the Oakley-Synths list or directly to me.

All resistors should be flat against the board surface before soldering. It is a good idea to use a 'lead bender' to preform the leads before putting them into their places. I use my fingers to do this job, but there are special tools available too. Once the part is in its holes, bend the leads that stick out the bottom outwards to hold the part in place. This is called 'cinching'. Solder from the bottom of the board, applying the solder so that the hole is filled with enough to spare to make a small cone around the wire lead. Don't put too much solder on, and don't put too little on either. Clip the leads off with a pair of side cutters, trim level with the top of the little cone of solder.

Once all the resistors have been soldered, check them ALL again. Make sure they are all soldered and make sure the right values are in the right place.

The diodes can be treated much like the resistors. However, they must go in the right way. The cathode is marked with a band on the body of the device. This must align with the vertical band on the board. In other words the point of the triangular bit points *towards* the cathode of the diode.

IC sockets are to be recommended, especially if this is your first electronics project. Make sure, if you need to wash your board, that you get water in and around these sockets. Also, make sure that any water drops left between the pins of the sockets are fully dried up before switching the board on.

For the two transistors match the flat side of the device with that shown on the PCB legend. Push the transistor into place but don't push too far. Leave about 0.2" (5mm) of the leads visible underneath the body of transistor. Note how the flat sides face each other. The idea is that the two devices should be in contact with one another. So before soldering you should wrap a small cable-tie around the bodies of the devices. Tighten the cable-tie carefully so that the flat face of each transistor is touching the other. Snip of any excess plastic from the cable tie. This will hold them together to ensure good thermal contact. There is no need for any thermal compound between them, but if you have some, you can add a small bit to ensure good thermal transfer. Now solder all six leads as you would normally.

Sometimes transistors come with the middle leg preformed away from the other two. This is all right, the part will still fit into the board. However, if I get these parts, I tend to 'straighten'



the legs out by squashing gently all the three of them flat with a pair of pliers. The flat surface of the pliers' jaws is parallel to the flat side of the transistor.

The polypropylene capacitors are like little blue, green or red boxes. Push the part into place up to the board's surface. Little lugs on the underside of the capacitor will leave enough of an air gap for the water wash to work. Cinch and solder the leads as you would resistors.

If using polystyrene capacitors then be very careful not to melt the cases when soldering. They are easily destroyed by overheating their leads.

The smaller electrolytic capacitors are very often supplied with 0.1" lead spacing. My hole spacing is 0.2". This means that the underside of these radial capacitors will not go flat onto the board. This is deliberate, so don't force the part in too hard. The capacitors will be happy at around 0.2" above the board, with the legs slightly splayed. Sometimes you will get electrolytic capacitors supplied with their legs preformed for 0.2" (5mm) insertion. This is fine, just push them in until they stop. Cinch and solder as before. Make sure you get them in the right way. Electrolytic capacitors are polarised, and may explode if put in the wrong way. No joke. Oddly, the PCB legend marks the positive side with a '+', although most capacitors have the '-' marked with a stripe. Obviously, the side marked with a '-' must go in the opposite hole to the one marked with the '+' sign. Most capacitors usually have a long lead to depict the positive end as well.

Remember that if you are not using the saturation option to link out the SAT header. A simple loop of uninsulated wire, like a resistor wire clipping, will suffice.

I would make the board in the following order: resistors, IC sockets, small non-polar capacitors, transistors, electrolytic capacitors. Then the final water wash if you are using water washable solder. You can then solder the trimmers in place, but do not mount the pots just yet. The mounting of the pots requires special attention. See the next section for more details.

## Mounting the Pots

**NOTE:** This procedure is rather different to that of the Omeg/Piher pots you may have used on the older Oakley boards.

The first thing to do is to check your pot values. Spectrol do not make it that easy to spot pot values. You only need three pots. These are mounted onto the PCB directly, and are held in place by our specially made pot brackets.

<b>Value</b>	<b>Marked as</b>	<b>Quantity</b>	<b>Location</b>
10K linear	M248 10K M	1 off	RES
50K linear	M248 50K M	2 off	FREQ, CV1

Fit a pot bracket to each of the three pots by the nuts supplied with the pot bracket kit. You should have two nuts and one washer per pot, including the one you got with the pot itself. Fit

only one nut at this stage to hold the pot to the pot bracket. Make sure the pot sits more or less centrally in the pot bracket with legs pointing downwards. Tighten the nut up carefully being careful not to dislodge the pot position. I use a small pair of pliers to tighten the nut. Do not over tighten because if you do the pot shaft will not rotate smoothly.

Now, doing one pot at a time, fit each pot and bracket into the appropriate holes in the PCB. Solder two of the pins attached to the pot bracket. Leave the other two pins and the three pins of the pot itself. Now check if the pot and bracket is lying true. That is, all four pins are through the board, and the bracket should be flat against the board's surface. If it is not, simply reheat one of the bracket's soldered pads to allow you to move the pot into the correct position. Don't leave your iron in contact with the pad for too long, this will lift the pad and the bracket will get hot. When you are happy with the location, you can solder the other two pins of the bracket and then the pot's pins. Do this for both pots and snip off any excess wire from the pot's pins at this point.

You can now present the front panel up to the completed board to check that it fits. However, I usually fit the sockets before I do this, and wire up the ground tags first. Then I mount the board up proper. You need to add the washer between the panel and the nut. Again, do not over tighten and be careful not to scratch your panel.

The pots shafts of the four pots will not need cutting to size. They are already at the correct length.

The Spectrol pots are lubricated with a light clear grease. This sometimes is visible along the top of the mounting bush of the pot body. Try not to touch the grease as it consequently gets onto your panel and PCB. It can be difficult to get off, although it can be removed with a little isopropyl alcohol on cotton wool bud.

## Connections

The power socket is 0.156" 4-way header in common with rest of the Oakley and MOTM modules. Friction lock types are recommended.

<i>Power</i>	<i>Pin number</i>
+15V	1
Module 0V	2
Earth/Screen	3
-15V	4

The PN1 and PN2 pads on the PCB has been provided to allow the ground tags of the jack sockets to be connected to the power supply ground without using the modules 0V supply. Earth loops cannot occur through patch leads this way, although screening is maintained. Of course, this can only work if all your modules follow this principle.

Whether you have chosen to make your module in a 1U or 2U format will determine what you do next. The 1U format is considerably easier to build and test. The 2U is more daunting, but

if you take your time, you should not find it excessively difficult. I shall deal primarily with the 1U format first since this is the recommended approach.

## **1U Filter Core module**

You have eight sockets to wire up. If you have used Switchcraft 112 sockets you will see that they have three connections. One is the earth lug or ground tag, this is indicated by a bevel in the socket's housing. The second is the signal tag which will be connected to the tip of the jack plug when it is inserted. The third tag is the normalised tag, or NC (normally closed) lug. The NC lug is internally connected to the signal tag when a jack plug is not inserted. This connection is automatically broken when you insert a jack.

Firstly we are going to 'common' all the sockets' ground lugs, and half of the sockets' NC lugs. This means that some of the sockets' lugs are going to be joined together. I normally do this part of the wiring without the PCB or pots in place.

Fit the eight sockets onto the panel so that the bevel on the side of the socket is facing top right as you look at the rear of the panel.

The first lugs we are connecting together will be the ground or earth tags on the four left hand sockets. I use 0.91mm diameter tinned copper wire for this job. Its nice and stiff, so retains its shape. Solder a length of this solid core wire right across the four earth tags in the left hand column. Trim off any excess that sticks out on either end. Then do the same with the right hand four. What you have now done is common each column's earth tags together, but each column is separate for now.

The next set of lugs to common will be the NC lugs of the input sockets. These will be on the right hand side as you look at the module from the rear. Again, this can be made with some 0.91mm diameter stiff wire. However, be careful that the wire is not too close to the body of the socket so as to foul any inserted jack plug. This can happen with some types of socket. Check this by inserting a jack plug and making sure the wire doesn't accidentally touch the tip of the plug. I always use Switchcraft 112APC sockets and this always have enough height to avoid this from happening.

Do not common the NC lugs of the four output sockets. These lugs are left unconnected.

Fit the SVF PCB against the front panel if you haven't done so already. Now solder a piece of ordinary insulated multistrand wire to the earth lug on the top socket on the left. The other end of this wire needs to go to the pad on the PCB marked PN1. Now solder another piece of wire to the earth lug of the right hand socket. This wire will be going to the pad PN2. Your earth tags are now commoned together since PN1 and PN2 are electrically connected together on the circuit board.

Solder a similar length wire to the NC lug on the top right socket. Solder the other end to panel named GND.

Now its time to wire up the eight signal lugs to the board. Use multistrand hook up wire to connect each socket's signal lug to the relevant pad on the PCB. Keep your wires short but

not too short and you can use as many different colour wires as you can. There is absolutely no need to use screened cable for such short runs.

The connections of the signal lugs of the CV and audio sockets that go directly to the PCB are summarised below:

<i>Socket name</i>	<i>PCB pad name</i>
IN1	IN1
IN2	IN2
CV1	CV1
1V/OCT	KEY
NOTCH	BS
BP	BP
HP	HP
LP	LP

That completes the connections to the sockets. Your Filter Core module is now ready to test and calibrate.

## **2U SVF full format**

I am not going into great detail with this format as the PCB has been primarily designed with the 1U filter core module in mind. However, I will mention a few things that may be useful to you if you do decide to build the larger format design.

The 2U format contains ten sockets and four additional pots. You can use any pots you like, but I am rather partial to the 16mm Alpha pots sold by Banzai. These have nice solderable lugs on them and are therefore easier to wire up than the Spectrol 248 types used elsewhere in the module. One thing to bear in mind though, is that the Schaeffer database I have provided on the site is set up for using Spectrol 248 pots. As such, it has a 10mm diameter hole for the bush. You'll need to make this smaller, to around 8mm, if you are using the Alpha pots. You may be tempted to use the larger 24mm Alpha pots, indeed, these are great pots, but the width of them may mean that the top pot will clash with your choice of mounting rail.

As with the 1U module, you need to ground the sockets' earth lugs. Do this by joining the earth lugs of each vertical row of sockets together first with stiff single core wire. Then connect the four solid wires back to the PN1 or PN2 pads on the PCB with three pieces of thin insulated multistrand wire. You'll need to put two of the wires into each pad, but there is plenty of room in to accommodate this. The PN1 and PN2 pads are connected to panel ground on the power socket, pin 3 on the MTA/Molex connector.

It is also advisable to ground the NC lugs of the six input sockets too. Do this in the same way you have commoned the earth lugs. Simply connect all three sockets' NC lugs in each of the right hand columns together with two pieces of stiff wire. Then connect these two wires to the GND pad on the PCB with two more pieces of insulated multistrand wire.

The connections of the signal lugs of the CV and audio output sockets that go directly to the PCB are summarised below:

<i>Socket name</i>	<i>PCB pad name</i>
CV1	CV1
1V/OCT	KEY
HP OUT	HP
LP OUT	LP
BP OUT	BP
NOTCH OUT	BS

There is no connection to the pads labelled IN1 and IN2.

All your other connections will be made via the four two way 0.1" headers that are situated on the board near the pots. These are labelled appropriately to help you connect up your module correctly. They are IN-1, IN-2, IN-3 and CV2.

Pots have three pins. Two of these pins will be connected to PCB, whilst the remaining one will be connected to the appropriate socket's signal lug.

The middle pin of the pot, the wiper, will carry the signal to the appropriate header on the PCB. The pots' wires will attach to the underside of the board at each header, and thus be soldered from the topside of the board. For each header, pin 1 is connected to the wiper of the pot. Pin 1 is the square pin so its easily spotted even from the underside of the board.

The pot has two other pins, one will be connected to ground, the other to the signal lug on the socket it controls.

With pins facing down and looking at the back of the pot, the right hand pin should go to the ground connection of the header, that is pad 2 on each of the headers. Take a wire from the right hand pin to the round pad on the PCB next to the one that the associating wiper connects.

Now each pot will have one unsoldered pin left. Connect these to the appropriate socket. The wire should go to the signal lug of the socket. IN 1 goes to the signal lug on the socket labelled IN 1, and so on.

There are a quite lot of wires here, but it should be quite neat once it is all done.

## Power Connections

The power socket is 0.156" Molex/MTA 4-way header. Friction lock types are recommended. This system is compatible with MOTM systems.

<i>Power</i>	<i>Pin number</i>
+15V	1
Module GND	2
Earth/PAN	3
-15V	4

As stated before the PN pads on the PCB has been provided to allow the ground tags of the jack sockets to be connected to the powers supply ground without using the modules 0V supply. Earth loops cannot occur through patch leads this way, although screening is maintained. Of course, this can only work if all your modules follow this principle.

## The Front Panel

On the website I have included a 1:1 FPD database of the suggested 1U and 2U front panel layouts. Actual panels can be obtained from Schaeffer-Apparatebau of Berlin, Germany. The cost is about £17 for the 1U panel, and about £30 for the 2U panel. VAT and the postage is extra, so it usually helps to order a few panels at the same time.

All you need to do is e-mail the fpd file to Schaeffer in Germany, or Frontpanel Express in the US, and they do the rest. You can also use the Frontplatten Designer program's own online ordering procedure which also works very well.

The panel is black with white **engraved** legending. The panel itself is made from 3mm thick black anodised aluminium. The fpd panel can be edited, including changing the colour, with the Frontplatten Designer. The program available on the Schaeffer web site but it should be noted that the program is for Windows only.

## Testing, testing, 1, 2, 3...

Apply power to the unit making sure you are applying the power correctly. Check that no device is running hot. Any sign of smoke or strange smells turn off the power immediately and recheck the polarity of the power supply, and the direction of the ICs in their sockets.

Assuming everything is OK so far, it is time to apply an audio input. Use a bright signal like a sawtooth output from a VCO. Middle A, 440Hz is a good note to use.

Moving the FREQUENCY control should produce the usual and distinctive filter effect from the low pass output. From the high pass output, you should hear the sawtooth get brighter and less bassy as the cut-off frequency is increased. The band pass output should produce a sort of

wah-wah sound as the Frequency pot is moved back and forth. The notch output will be more subtle, and will appear to sound like a mild phasing effect. Turning the Resonance up will accentuate the ‘electronic’ nature of the sound on all four outputs.

Remove the audio input and check that at maximum resonance the filter output will oscillate across the whole audio band. You'll probably need to turn up the cut-off frequency pot to above the mid-point first to actually get the SVF to start to oscillate – it does not tend to oscillate naturally at frequencies below 1kHz or so. Beware, it is quite possible to get this filter to oscillate above the range of hearing. So be careful so as not to damage your studio monitor’s tweeters.

Listening to the band pass output with the sawtooth input still connected, patch a LFO or EG output to the CV inputs. The 1V/octave input should produce large sweeps of cut-off. Check also that with the LFO or EG connected to the CV1 input, the CV1 pot allows you to control the depth of the sweep. Fully clockwise the CV1 input should produce very deep sweeps.

Notice that the minimum sweep depth should occur with the CV1 pot at its mid point. Use a sawtooth waveform on your LFO, and see if the CV1 depth pot allows you to invert the modulation input. You should get a ‘dow-dow-dow...’ from one side and a ‘yit-yit-yit...’ from the other.

Check that both the audio inputs should behave identically.

If all this happens, the chances are that you have a working module.

## Trimmers

There are four trimmers on the PCB.

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 to get the filter oscillating. Now listen to the output coming from the low pass output. I found it is best to use the Frequency pot on the front panel to set the filter oscillating at quite a high frequency tone. Somewhere around 880Hz (two As above middle C) will do. Now play a note on your keyboard and then the same note an octave above. 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.

The two OFF trimmers control the amount of DC on the output. Use a scope or a voltmeter to measure the voltage at pin 7 of U1. 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 +/-5mV 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 +/-5mV or less.

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

## Final Comments

If you have any problems with the module, an excellent source of support is the Oakley-Synths Group that can be found at <http://launch.groups.yahoo.com/group/oakley-synths/>

If you can't get your project to work, then Oakley Sound Systems are able to offer a 'get you working' service. If you wish to take up this service please e-mail me, Tony Allgood, at my contact e-mail address found on the website. I can service either fully populated PCBs or whole modules. You will be charged for all postage costs, any parts used and my time at 20GBP per hour. Most faults can be found and fixed within one hour, and I normally return modules within a week. The minimum charge is 20GBP plus return postage costs.

Your comments and questions are important to both Oakley Sound and Oakley Modular. In the first instance, please use the Oakley Synths Group where a wealth of experience resides! Please do not contact me or Oakley Modular directly with questions about sourcing components or general fault finding.

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.

Tony Allgood at Oakley Sound

Cumbria, UK  
May 2008

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