Oakley Sound Systems

Filter Core Series

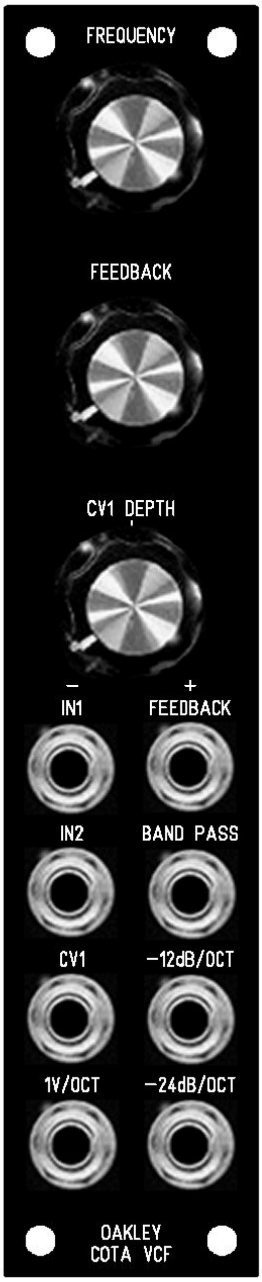
The COTA

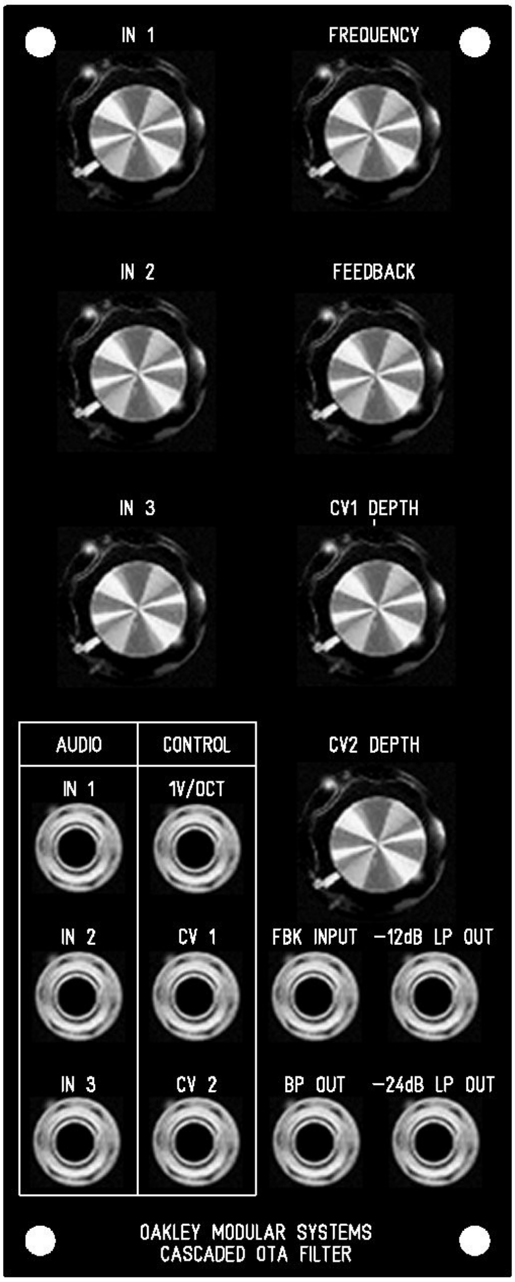
Voltage Controlled Four Stage Cascaded Filter

User's Guide

V1.0.1

Tony Allgood B.Eng PGCE
Oakley Sound Systems
CARLISLE
United Kingdom


FREQUENCY
FEEDBACK
CV1 DEPTH
–
+
IN1
FEEDBACK
IN2
BAND PASS
CV1
–12dB/OCT
1V/OCT
–24dB/OCT
OAKLEY
COTA VCF


IN 1
FREQUENCY
IN 2
FEEDBACK
IN 3
CV1 DEPTH
AUDIO
CONTROL
CV2 DEPTH
IN 1
1V/OCT
IN 2
CV 1
FBK INPUT
-12dB LP OUT
IN 3
CV 2
BP OUT
-24dB LP OUT
OAKLEY MODULAR SYSTEMS
CASCADED OTA FILTER

Introduction

This is a four pole multimode filter capable of -24dB/octave low pass, -12dB/octave low pass and +/-6dB/octave bandpass outputs simultaneously. The filter core is based on four identical cascaded current controlled integrators. This is similar to the topology found on the filters in the SH series of monosynths as well as the CEM3320, IR3109 and SSM2040 filters of the classic polyphonics.

At high resonance, we call it 'feedback' in this module, the COTA VCF will oscillate well over the whole of the audio band with a clean sinewave. Its temperature compensated too so you can use it as a reasonably accurate tracking oscillator. The module also features a feedback input. This is similar to the one we introduced in the groundbreaking Multiladder module. In conjunction with the 24dB/octave output this input allows the resonance signal path to be split. This means that additional modules can process the resonance loop independently. This allows a large variety of new timbres to be created. And it also allows for voltage controlled resonance with the use of an extra single VCA module.

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.

As already mentioned voltage controlled resonance can be simply attained with an external VCA module. Set the feedback to maximum, patch the -24dB/octave output to the input of the VCA and then take the output from the VCA to the 'feedback' input on the COTA. Now the VCA is controlling the resonance of the COTA. The higher the gain of the VCA, that is the higher 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 always 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 COTA has eight 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. 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 COTA issue 1 PCB

I have provided space for the three main control pots on the PCB. If you use the specified Spectrol 248 pots and our 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 112mm 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 two basic filter responses of the Oakley COTA:

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. For example a two pole filter will loose output amplitude by 12dB per octave, a four pole filter by 24dB/octave.

The COTA has four identical single pole filters cascaded together. Therefore a maximum attenuation of 24dB/octave is possible with the COTA. In theory, by taking the outputs from each stage you could obtain outputs of -6dB/octave, -12dB/octave and -18dB/octave too. Indeed, the COTA has an additional tap half way along so making the -12dB/octave output available.

By increasing the **resonance** (or **feedback** in the COTA) 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.

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 normally be determined by the number of active elements in the filter. For example, the roll off in a two stage state variable filter like the Oakley SVF, it is -6dB/octave either side of the centre frequency.

The COTA provides a bandpass output by combining the output from the first stage and fourth stage of the cascaded filter elements. At first glance, this may seem an unexpected result getting a bandpass response from two low pass outputs. However, one should remember that a filter also affects the phase of the audio signal. The two outputs, when mixed in the right amounts, cancel themselves out at lower frequencies and reinforce each other at higher frequencies. This produces a classic -6dB/octave band pass response.

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.

COTA issue 1 Parts List

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.

The components are grouped into values, the order of the component names is of no particular consequence.

A quick note on European part descriptions. R is shorthand for ohm. K is shorthand for kilo-ohm. R is shorthand for ohm. So 22R is 22 ohm, 1K5 is 1,500 ohms or 1.5 kilohms. For capacitors: 1uF = one microfarad = 1000nF = one thousand nanofarad.

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, 6n8 is a 6.8 nF capacitor.

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

Resistors

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

1U Filter Core format does not need R11 and R20 so these parts can be omitted.

100R	R14
470R	R35
1K	R7, R26, R46, R45, R36, R8, R27, R39, R52, R51, R40
1K +3000ppm/K PTC	R33
4K7	R15
15K	R9, R2, R1, R28, R19, R18, R41, R48, R53, R47
33K	R32
47K	R10
68K	R24, R37, R49
75K	R42, R29, R55
100K	R6, R4, R3, R16, R20, R25, R38, R50
120K	R31
150K	R12, R11, R13, R44, R34, R30, R54
220K	R5, R21
240K	R43
560K	R17
1M	R22
1M5	R23

Capacitors

330pF 500V silver mica	C1, C6, C8, C16
100nF, 63V axial ceramic	C5, C3, C13, C12, C10, C11, C4
220nF,63V polyester	C2

1uF, 63V polyester	C9, C17, C7
2.2uF, 25V electrolytic	C14, C15

Discrete Semiconductors

1N4148 signal diode	D1, D2
BC550 NPN transistor	Q4
BC560 PNP transistor	Q3
J201 N channel JFET	Q1, Q2, Q5, Q6

Integrated Circuits

LM13700 dual OTA	U2, U4
TL072 dual op-amp	U1
LF412A dual op-amp	U3, U5

Variable Resistors

100K multiturn trimmer	TUNE
20K multiturn trimmer	V/OCT
10K horizontal 6mm trimmer	RES
10KA Spectrol 248 pot	FEEDBACK
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	Two off
DIL8 IC sockets	Three off

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

Additional parts required for the 2U version

Miscellaneous

1/4” sockets	IN3, 1V/OCT

Offboard Pots (2U format only)

47K or 50K Log	IN1, IN2, IN3
47K or 50K Linear	CV2

Circuit description

The design of the Oakley **C**ascaded **O**perational **T**ransconductance **A**mplifier filter is very traditional and I am making no claims for originality. The design topology is found in many synthesizers. It is simply four voltage controlled low pass elements put in cascade to give us a great sounding four pole low pass filter. Each low pass element is made from an integrator, built from an OTA and a timing capacitor, a buffer of some sort, and some resistative feedback. The active parts of the design were put into chip form as the SSM2040 (eg. Prophet 5 rev 2, Octave CAT SRM) , CEM3320 (eg. Prophet 5 rev 3 and Oberheim OB-Xa) and the IR3109 (eg. Jupiter 8 and SH-101). Each one of these chips behaves differently in terms of signal linearity, hence their different sounds, but the internal topology was much the same.

The Moog ladder can be thought of as a cascaded design too although this is constructed so differently to warrant its own classification.

Roland initially used transistor Moog ladders and then diode ladders in their first synths like the SH-3, SH-5 and SH-2000. However, later synths such as the SH-7, SH-1 and System 700 used a cascaded design built from CA3080s. The 3080 is an operational transconductance amplifier, OTA, which is probably more commonly used as a VCA, but here it was used as the control element in the integrator. Later Roland synths such as the SH-09 used their own version of the CA3080, the BA662, as the OTA. Later still, the JP-8 and SH-101 used four matched OTAs in one package complete with its own onboard exponential convertor, this was the IR3109 filter chip. The fundamental sound was much the same as the early more discrete design – although there are some sonic differences between all the versions.

The Oakley design is a four stage filter that uses the well respected LM13700 as the OTA element. There are two OTAs in one LM13700, and the two internal devices are quite similarly matched. The OTA part of the LM13700 can be considered to be very similar to the CA3080 and probably the BA662.

Before we look at the core in detail, we need first to have a look at how the signals are treated before they reach the filter circuitry. There are three main audio inputs to the COTA, each one has a two pin header with the second pin being used for a ground, or 0V, connection. These are shown on the schematic as IN1, IN2 and IN3. Two of these inputs have an alternative input solder pad – these are to be used in the 1U Filter Core module. These have no individual ground since they will going straight to the input jack socket's signal lug.

U3 and its associating components form a inverting summing circuit. This common circuit block is used to add the various input signals together. The gain of the summing is set to be roughly a third. That is, if all three inputs are at 5V peak, the output at pin 1 of U3 will be approximately 5V peak. Note though that it is an inverting summer, which means that the phase of the summed output is in opposition to the input signals. This means a negatively going signal will be converted to a positive going one.

C2 provides AC coupling of the audio output of the summer. This acts to block any DC or very low audio frequencies. As we have seen the whole filter is built from four identical single pole low pass filters. Each filter element is basically a current controlled integrator with some negative feedback to provide the required low pass function.

Let us look at the first element in the signal chain, that is the one made based around U2a, Q1 and C1. The first element is identical to the others but does have some additional bits added to allow for the resonance feedback input. We'll look at this particular bit in more detail later.

The OTA is very useful device. It produces an output current that is proportional to the difference in voltage between its two inputs, pins 4 and 3. Furthermore, that output current is also proportional to current drawn from a control pin, which is pin 1 for this integrator. This control current is called Iabc - abc standing for 'amplifier bias current'.

In this application we can see that one of the inputs to the OTA is tied only to ground through a low value resistor, R7. Thus our only voltage input is at pin 4 which is the inverting input. This means that our output current is proportional, but of opposite polarity, to the voltage at pin 4. That current is used to charge and discharge the timing capacitor, C1. The more negative the input voltage, the faster the capacitor charges. The more positive the input voltage the faster the capacitor discharges. This is the action of an integrator.

Remember too the affect of Iabc. This controls the output current too, and a bigger Iabc will mean a faster rate of charging and discharging. Iabc is provided by the control voltage (CV) input circuitry and we will look at this later, but at the moment we can simply think of Iabc being larger for higher cut-off frequencies.

When you charge a capacitor up the voltage across it changes. It is this voltage that we are interested in since this is what we want to hear. However, we can't just take this voltage from the capacitor because by doing so we will steal current from it and affect the output level. So we use a buffer made from a JFET, Q1, and a resistor, R9. This simple circuit allows us to make a copy of the capacitor voltage without affecting that voltage itself. That copy is presented at the source pin of the JFET. There's actually a small error in this copy, in that it is slightly higher in value than that across the capacitor. However, this is no issue because the error is constant and we're only interested in the audio part of the signal, the wiggly bit, and not any constant DC term. C9, a largish value capacitor, will block the DC and allow the audio through.

Note R6, this is critical to the operation of the low pass element. This passes back some of the output signal back to the input. Actually, its passing current back to the input and because it is of opposite polarity to the input signal it acts against it and regulates the output. It is this negative feedback that gives us our overall low pass filter response. Without R6, we just have an integrator and the voltage across the capacitor would simply rise and rise with any positive input voltage, or fall and fall with any negative signal. Of course, it can't rise or fall beyond the limits of the power supply, but even so, it wouldn't give us the desired low pass effect.
I will point out one more thing about this feedback path. The value of the feedback resistor, R6, in conjunction with the value of R5, sets the gain of the filter stage. Unlike many cascaded OTA designs, this is not set to one. In other words, the Oakley COTA is unusual in that the passband signal level changes as it runs through the filter. The first stage there is an gain of approximately one half. This is to reduce the large signal levels coming out of the audio summing circuit. Then the other three following stages have a gain of approximately 1.47. This means that when the filter is put into self-oscillation the magnitude of the resultant sinewave output is the same from each filter stage.

However, it also means that the audio level gets hotter as it works its way through the filter. This means that any non-linearities in the circuits are going to be exacerbated as the signal progressively gets closer to overdriving the sensitive OTA input circuitry. So why have I done this because surely this is a problem? Over the years I have heard many different synthesisers and some I like a lot, and some not so much. I analysed the circuits of one of my favourite synths and wondered why this 1978 synth sounded better than those that followed it. One of the differences was the filter gain structure, this one had been made for constant oscillation levels, but the later synth designs had unity gain in each filter stage. So what I have done for the COTA is copied the unusual gain structure from that wonderful 1978 synthesiser. So which synth is it? You'll have to guess.

You can of course change the values of R24, R37 and R49 to 100K to give each stage unity gain. I found the basic sound to be much the same, but there was something smoother sounding about the earlier circuit value. Why not experiment yourself?

As I have hinted earlier all OTAs are non linear. This means that they affect the input more than they should and produce audible distortion at their outputs. Now the LM13700 features a simple waveshaping circuit, based around some internal diodes, at each of 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 and pin 15. However, in this design I have chosen **not** to utilise the diodes. This will decrease the overall signal to noise ratio, ie. make the module more noisy, but it will allow the signal to gracefully overdrive. It also makes it closer to the behaviour of the CA3080 and hence give the module a more classic vintage tone.

The COTA has three audio outputs. The -12dB/octave output is taken from the output of the second stage of filtering. The -24dB/octave output is taken from the final fourth stage. Both of these are inverted and amplified by identical op-amp circuits, based around U3b and U5b respectively. C7 and C17 provide AC coupling – allowing the audio to go through but blocking the DC voltage output of the filter stages. The inversion of these stages is not a problem since it simply reverses that was done by the audio input summing circuit, U3a. The gain of each output stage is set to -2, so as to bring the output signal level, when the filter is oscillating, to roughly 5V peak, or 10V peak to peak.

It should be noted that due to the non unity gain of each filter stage, the passband audio signal does get larger as it works its way down the filter stages. Thus the -24dB/octave output will be greater than the -12dB/octave output. It may not necessarily be louder though, since the -12dB/octave output will be brighter in tone.

The bandpass output is a derived output and not generated in a true bandpass filtering process. The fourth stage output is summed with the first stage output in a specific proportion. This summing allows for the frequencies below the cut-off frequencies to be attenuated whilst reinforcing those above it. This gives an eventual overall roll off of -6dB/octave above and below the cut-off frequency.

The actual summing proportions were determined firstly mathematically and then verified empirically. To create a bandpass output the mathematics tells us that the output of the first and fourth filter stages shall be summed equally. However, the mathematics assumes that each filter stage has a gain of unity - ours do not. Each of our filter stages is designed to produce an equal output when oscillating and not for a unity gain in the passband. In practice this means that each of our filter stages has the effective gain of around 1.47. Over the three stages, from first to fourth, this means we have an overall gain of 3.18 or so. Thus R43 should be 3.18 times bigger than R42. If we choose R42 to be 75K, R43 should then be 238K. Using 240K is the nearest real resistor value.

The feedback path is normally wired from the fourth output stage back to the input – although the feedback input socket does allow you to change this. For the moment we will consider the normalised route from the -24dB/octave output via the RES trimmer and R35 to the FBO solder pad. This is then wired to the normalised lug of the feedback socket, which when no jack is inserted, simply carries the signal back to the FBI solder pad. The amount of feedback routed back to the input stage of the first filter element is controlled by the 'FEEDBACK' pot, D1 and D2 act as voltage limiters preventing feedback runaway and ever increasing self-oscillation. R1 limits the input current and prevents these diodes from having a detrimental effect on any input plugged into the feedback socket. Even so it should be noted that the feedback input is quite low in impedance compared with the minimum value of 47K of ordinary Oakley audio inputs. However, this should not cause any problems in practice.

Control of the OTAs is carried out by a current called Iabc. The 'abc' bit standing for 'amplifier bias current'. This is provided by Q3 which acts as an exponential convertor and voltage to current convertor in one. Its driven by an emitter follower, Q4. Both transistors act together to eliminate the effects of Vbe drift. Vbe drift, if uncompensated, would cause the filter's cut off frequency to change with temperature. However, any drift in Vbe in Q3, is effectively cancelled out by an opposite drift in Q4. In theory, the transistors must be matched for Vbe. However, this is not really necessary in my opinion.

Vbe 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 R33. The PTC resistor's resistance goes up by 0.30% 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 R3, 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 R4. 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 R3. Why? Because you need a difference in voltage across a device to create a current. So if there is no current through R3; there is none flowing through the feedback resistor R4. 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 supplied 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 C12 and C14 for the positive rail, and C13 and C15 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.

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.

Value	**Marked as**	**Quantity**	**Location**
10K linear	M248 10K M	1 off	FEEDBACK
50K linear	M248 50K M	2 off	FREQ, CV1-DEPTH

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 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.

Power	***Pin number***
+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, ignore the 'feedback' socket for now. 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 sockets on the right hand side. The three output socket NC lugs are left unconnected, and we will be using the NC lug on the 'feedback' socket for another purpose.

Fit the COTA 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:

Socket name	***PCB pad name***
IN1	IN1
IN2	IN2
CV1	CV1
1V/OCT	KEY
FEEDBACK	FBI
-12dB/OCT	2LP
-24dB/OCT	4LP
BAND PASS	BP

There is one connection left to make. This is the connection to the NC lug on the 'feedback' socket. This comes from the pad labelled FBO, which should be the last remaining big solder pad left unsoldered.

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

2U COTA 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 case diameter 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:

Socket name	***PCB pad name***
CV1	CV1
1V/OCT	KEY
FEEDBACK	FBI
-12dB/OCT	2LP
-24dB/OCT	4LP
BAND PASS	BP

There is no connection to the big solder 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.

Power	***Pin number***
+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 £20 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 -24dB/octave low pass output. From the -12dB/octave pass output, you should hear the same sort of sound, but slightly brighter and more electronic in tone. It will also be slightly quieter in volume than the -24dB/octave output. The band pass output should produce a sort of wah-wah sound as the Frequency pot is moved back and forth. Turning the Resonance up will accentuate the 'electronic' nature of the sound on all three outputs.

Remove the audio input and check that at just under maximum resonance the filter output will oscillate across the whole audio band. 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 -24dB/octave low pass output with the sawtooth input connected again, 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 three 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 last trim to do is to set the RES trimmer. This allows you to set where you want the resonance to occur as you turn up the Feedback pot. I normally make resonance start at around three quarters of the way around.

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 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 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 25GBP per hour. Most faults can be found and fixed within one hour, and I normally return modules within a week. The minimum charge is 25GBP plus return postage costs.

If you have a comment about this builder's guide, or have a found a mistake in it, then please do let me know. But please do not contact me or Paul Darlow directly with questions about sourcing components or general fault finding. Honestly, we would love to help but we do not have the time to help everyone individually by e-mail.

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.

Tony Allgood at Oakley Sound

Cumbria, UK