Oakley Sound Systems

Filtrex

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Analogue Filter Rack

User's Guide

V2.1

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A bit of history...

The popular Oakley Sound VCF-1 design has now been totally updated to create a new powerful filter processing unit, the Filtrex.

The VCF-1 was my first PCB project to be released and it proved to be a moderate success. Two issues were made, the second one allowing the use of Omeg Plastic pots and brackets. In the end some eighty or so PCBs were made and bought. Everyone who built one and wrote back to me felt that it was a very useful unit with a great sound.

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

1. The filter itself created smooth sweeps that warmed up any harsh digital sound. But it couldn’t do the opposite. It couldn’t be nasty, well, not nasty enough anyway. Enter the Filtrex’s one pole or Fizz output. Plenty of cutting edge. Mix that in with the ordinary four pole or smooth output and you get band pass to cut through any mix. It still doesn’t mangle the sound like the Sherman Filter Bank can, but that's a different beast altogether.

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

3. The lack of a bypass function on the VCF-1 made the unit difficult to use without a mixing desk. The Filtrex’s Thru function allows the ‘wet-dry’ level to be set simply. It also allows for complex filter responses as the filter’s output interacts with the audio input.

Some things present in the VCF-1 were left out in the Filtrex. The biggest loss to me is the syncable mode on the LFO. The LFO is just free running now. I just didn’t have enough room on the PCB to fit it in. Also gone is the ability to mix the envelope generator with the envelope follower. You can only now have either one or the other, not both. I don’t think this one will be missed by anyone, and the new combined EG/EF more than makes up for it.

Overview of the Filtrex

The complete filter module contains:

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

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

- A wide range preamplifier is included. You can use the Filtrex-1 with virtually any input from guitars to synthesisers. A peak indicator will allow you to monitor overloads, although the Filtrex-1 is designed to overdrive beautifully.

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

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

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

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

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

- The final output volume can be controlled with a level control. The output circuitry is configured to drive long length cables with ease.

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

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

The board is designed to be powered by an external ac supply of 15 to 22V rms. You can use an externally mounted AC wallwart. Provision is also provided for connection to an internal power transformer. This will be discussed later in the document. However, this should only be done by experienced builders.

What all those front panel controls actually do...

The PCB has thirteen rotary pots and four toggle switches to control the filter. Please refer to the first appendix sheet at the end of this document that shows the complete front panel layout.

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

Let us first look at the main audio control pots:

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

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

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

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

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

Note that this version of the ladder filter will not oscillate at very low frequencies.

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

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

Thru: Output level of the filter bypass signal. This signal has been through the pre-amp, so the GAIN pot will affect the level too. Just like the other two previous controls, this is a reversible attenuator. For no bypass signal to be present, this pot must be set straight up.

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

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

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

That completes the description of the audio controls at this stage. A more detailed look at the circuitry behind the dials will be found in the next section. The rest of the pots and switches control the envelope and low frequency oscillators. These form part of the control circuitry and these will be described now.

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

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

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

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

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

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

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

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

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

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

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

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

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

Threshold: The audio-trigger circuit. The peak level of the audio signal, either the main input or the 'side chain', is analysed in this section. If the audio input is higher than the level set by the threshold pot, then a gate output is produced that can trigger the EG if turned on. With the pot set less than fully clockwise, a loud signal is required to trigger the EG. With it set fully anti-clockwise very quiet sounds will trigger it. However, setting it too low can cause false triggering. This pot's operation can get some getting used to. Stick with this, the results are worth it... especially if you haven't got an external midi-CV converter.

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

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

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

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

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

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

That completes the overview of the front panel operation.

Circuit Description

Like many complex analogue circuits the Filtrex circuit can be split up in to little bits. The first bit we will look at is the preamplifier stage on page one of the Schematic.

The pre-amp is built around U1. I have specified the low noise audio op-amp, the OP-275. The preamplifier is a two stage design. The first stage is a non-inverting amplifier whose voltage gain can be varied from 1 to 12. C1 keeps the gain for DC and very low frequency signals at near to one. This prevents any offsets within U1 from being amplified unnecessarily C3 provides a little bit of high frequency roll-off to keep the amplifier stable.

The second stage of the pre-amp is an inverting amplifier. The GAIN pot is used in a slightly offbeat way. It is in both the feedback and the input resistors. This way we can control the gain over a wide range from -0.4 to -10.1. The minus in these numbers shows the inverting properties of the amplifier.

The voltage gain of the two preamplifier stages in tandem can be varied from -0.4 to -122. A gain of 0.4 means that the output of the pre-amp is only 40% of the input level. While a gain of 122 means that the output level is 122 times bigger than the input. In audio circles this would normally defined in dB. This preamplifier will give you a gain from -8dB to +42dB. Because the preamplifier is made from a inverting and non-inverting stage, the overall behaviour is inverting. This means the output is completely out of phase with the input. This is not a problem since the inversion is corrected later on the Filtrex.

Q7 and associating circuitry drive the peak LED. This is designed to light up just as the ladder filter starts to show heavy distortion.

The main audio path continues on to the filter ladder itself via one half of the resonance pot. However, the pre-amp also provides the signal for the THRU circuit and the envelope processor. We will deal with these two later.

The filter is based around the traditional ladder as designed originally by Dr. Moog. I have used SSM2210 matched pair for the top and bottom pairs in the ladder. This minimises control current breakthrough to almost zero. Current breakthrough manifests itself as a copy of the modulating signals on the output. Generally, this is not a good thing. ‘BAL’ biases the base of U3b, via R20, by a small amount to even out any differences within the ladder. This minimises breakthrough still further.

Two of the rungs of the ladder are ‘sniffed’ by a differential amplifier. Each of these is identical, based around the classic three-op-amp implementation. They are all DC coupled, and rely on ‘close’ matching to remove any DC offset. A differential amplifier is a device that makes larger the voltage difference between two points. In our case, the voltage across the top and bottom filter capacitors. The gain of the differential amplifiers is set higher than normal ladder filters to improve signal to noise ratio in the following mixer stage.

C19 and C17 provide AC coupling of the outputs to remove the slight DC offset. The feedback to the ladder traditionally comes from the output of the differential amplifier at top rung of the ladder. It does this via R25, the TRIM preset, and the Resonance pot itself. R25 and the TRIM control set the point at which oscillation will occur in the travel of the Resonance pot. R23 doesn’t actually do anything and can be omitted if wished. I added it to the circuit to do some experimentation with the value of the resonance pot. By setting it to a value of 1M, it has no effect on the circuit.

It is traditional to use a 50K reverse log pot for the resonance control in the classic moog ladder. A value of 50K pretty much allows the feedback path to be ignored, and no resonance can be heard. The ‘reverse log law’ is needed so that as you turn the pot the resonance increases smoothly. An ordinary linear pot would do nothing for most of its travel, and then all the resonance would be introduced in the last quater of a turn. Not very ‘smooth’ or musical. Unfortunately, 50K reverse log pots are difficult to find and quite expensive. In the Filtrex, we use an 10K linear pot. We can’t turn the resonance effect completely off, but this is perfectly all right in a post processing filter like the Filtrex. By limiting ourselves to 10K maximum, we still have a nice ‘feel’ as we turn the pot even with a traditional linear pot law.

The resonance pot is a dual gang type too. One gang controls the feedback loop as we have seen, and the other sets the gain of the input. Normally the passband gain of a moog ladder decreases as you turn the resonance up. In other words the volume drops as you increase the resonance. This can be quite a problem for a post processor like the Filtrex. So in the Filtrex, the input level is automatically turned up as resonance increases. Thus the overall effect is of a constant volume at all values of resonance. I thought this was quite clever of me to invent a way of doing this without increasing noise levels. However, I found out later that the very same principle was used in the Roland SH-2000 as long ago as 1974.

C11 provides the appropriate decoupling and is sufficiently large so as to allow the filter to oscillate above 200Hz or so.

Four inputs control the filter cut-off frequency via an exponential convertor based around U9. The filter frequency can be directly controlled with the FREQ pot via R63. R65 sets the sensitivity of the envelope processor’s output. While R67 does the same job for the LFO. The CV input provides a nominal 1V/octave response, and would normally be accessed via a jack socket on the rear panel.

The exponential convertor is temperature compensated. R61 is the positive temperature coefficient resistor providing an approximate cancellation of the exponential convertor’s inherent temperature coefficient.

The audio path continues from the filter’s differential amplifiers by going on to the mixer stage. This is shown on page two of the schematics at the top of the page. The mixer stage is designed to combine the three audio signals created by the Filtrex so far: The first and fourth rung ladder outputs, which will make up the ‘fizz’ and ‘smooth’ outputs respectively. And the main pre-amped signal which will be the ‘thru’ output.

The three signals go to identical ‘reversible attenuator’ circuits. This is 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, although I have to say I haven’t seen it in any other text book. Let us look at the first attenuator connected to the fourth rung output.

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 clockwise, the wiper connects to ground, pin 3 of the pot. 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 R34, 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 R36. 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 IN-1 was 3V, the output of the op-amp would be -3V.

Now, let us move the wiper to the opposite side, that is, fully counter 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 R34. Why? Because you need a difference in voltage across a device to create a current. So if there is no current through R34; there is none flowing through the feedback resistor R36. To do this, the op-amp must produce the same voltage at its output as on its two inputs. Et viola... 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.

Pins 1, 2 and 3 of U7 form the main summing amplifier. Currents flowing through all the input resistors, R37, 38 and 41 come together at the inverting input. The op-amp's output will alter to create the same current through R43. If all the resistors are the same value, the voltage at the output of the op-amp would be proportional to the sum of all the voltages presented to the input resistors. However, note that R41 is actually ten times higher then the others. This is because the signal level from the filter is about a tenth of the straight through signal.

The output of the mixer stage is fed via a DC blocking capacitor to the 'Volume' pot. The wiper of this is connected to the final output amplifier stage. This is a low noise inverting amplifier circuit with a gain of -4.7. The op-amp chosen is the 5534 which is capable of driving larger currents than most ordinary devices. C24 and R55 form a compensation circuit to allow the op-amp to drive fairly high capacitive loads. It has no effect on the audio performance, but reduces any chance of high frequency ringing in long cable runs. A full explanation of this is beyond the scope of this user guide, but more information can be found on the Analog Devices website in their Applications Note AN-257. I was first introduced to this method when I worked at Soundcraft, and have since used it on other Oakley projects notably the original VCF-1 filter rack's main output with great success.

That completes the description of the audio path. Now let us take a look at the processing circuitry of the Filtrex. Staying on page two of the schematics, the second set of circuits down the page belongs to the full wave rectifier and threshold detector. This takes its input from wither the pre-amp output or the external side chain input. The choice is determined by the 'cont' switch.

The switch's wiper leads straight into some amplification based around U14a. This is a non-inverting amplifier of around 2. The main pre-amp is expected to produce a maximum output of around 5Vp-p in normal use, so U14a boosts this signal up to 10Vp-p. The amplified signal is now full wave rectified by the circuitry based around U14b and associating circuitry. Full wave rectification can be described by the mathematical 'absolute' function. In other words, the output of the full wave rectifier (FWR) is always positive. If you present +10V to the input, you will get +10V. But if you present it with -10V you will also get +10V. Likewise, -5V turns into +5V, -3V into +3V. Now if you put an audio signal into this circuit, you will get a series of positive bumps that correspond to the up and downs of the audio signal.

U14d forms a special buffer circuit. This configuration, allows the op-amp to drive medium to high capacitive loads without instability. We also saw it being used in the output amplifier of the Filtrex. The output of the full wave rectifier is therefore protected by the odd load presented by the next set of circuits.

Now, no real time system can recover envelope information without some disadvantage of some sort. Some systems employ the *peak and droop* method. These are fast to respond to sudden changes in loudness or envelope. They work by simply charging a capacitor from the

FWR through a diode. The capacitor is then discharged through a resistor, sometimes variable, causing the stored voltage to droop at a determined rate. However, they are often plagued by ripple. Ripple is the bumps from the FWR creeping through to affect the required output. This tends to manifest itself in a ‘buzz’ to the output CV. If you increase the discharge resistor, you can reduce the bumps but this tends to not allow the CV to drop quick enough when the sound ends.

Another method involves low pass filtering of the FWR output. This leads to less bumps if the correct filter cut-off frequency is chosen, but does lead to longer attack times. There are more complex ways too, involving sample and holds and other clever methods. Some three years ago when I designed my VCF-1 Filter rack, I sat down and compared the many different circuits. I didn’t want a complex circuit, I didn’t have the board space to do that. So I stuck to either filtering or the *peak and droop* topologies. The one that most excited me was the low pass filter, but only if you got the frequency right. I decided to use a four pole filter with the cut-off frequency at around 33Hz. I chose the best sounding and the most natural with all sound sources. This method is not as quick at responding to fast attack signals as the *peak and droop* but it did react equally to rises and falls in signal level. However, the *peak and droop* wouldn’t be forgotten, I used that for the threshold detector.

In the Filtrex, things are different. The output of the full wave rectifier, labelled FWR on the circuit, is passed to a special circuit called the ‘lag processor’. This is cleverly combined with the function of the envelope generator and is described in detail later. At this point I will just say that it functions as a simple low pass filter with controllable rise and fall times.

But let us stay on page two for now. To create a gate signal we need a very fast response. In an ideal world this signal must go high the moment the signal arrives and goes low the moment the signal dies away. In this case I have used the *peak and droop* method. This does give us a fast as response as possible, but what about the ripple. Well, ripple is not *that* important here. Remember the gate output only goes high or low. What we have to do is make sure our gate doesn’t ‘rattle’ when it picks up the ripple. In other words, we need our gate to come cleanly on and off with no spurious states as the signal rises and falls.

U14b is a comparator. This is a device based around an op-amp in this case, that determines whether a signal is higher than a pre-selected threshold voltage. The threshold voltage is controlled by the user, and is set by the ‘Threshold’ pot. The threshold voltage can be set between 12V and 0.7V. C43 is charged via D15 from the FWR output. D15 allows the capacitor to be charged up, but not discharged, by the FWR’s output. R102 allows the capacitor’s stored peak voltage to droop at a controlled rate.

Most gate extractors provide a gate signal when the voltage on the capacitor is above a certain value. The Filtrex is different. Once the gate does go high, a certain proportion of the opamp's high level output is fed back to keep the input higher. This forces the comparator to stay high longer than it would normally do. This allows more ripple to be present before ‘rattling’ occurs, giving us a cleaner edge to our gates. You don’t have any control over this amount of positive feedback, its set by the value of R103. Now many good comparator designs have a little positive feedback anyway, its called hysteresis, and in our case its provided by R104. But the additional path offers a type of one way hysteresis that gives us better high to low gate transitions.

The comparator’s output is fed via D17 to a transistor Q13. This transistor is turned on when the comparator’s output goes positive. D17 protects the transistor form damaging negative output voltages. Q13’s collector will be pulled down to ground when the transistor is turned on. This in turn controls the envelope generator’s logic circuitry described later.

Lets have a look at the third page and the envelope processor itself. This is quite a hard bit to understand... so go and get some coffee now.

The heart of this unit is the circuitry based around U10b. This, along with the ‘up’ and ‘down’ pots, make up the lag generator. What is a lag generator? Basically it is a capacitor, C36 in this circuit, that can discharged and charged at a controlled rate. The level to which the capacitor charges to, or discharges to, is determined by the input voltage applied to pin 5 of U10. The voltage across the capacitor will directly control the output of the envelope processor.

U12 is an analogue switch. Its a good old 4016, and this IC is found in hundreds of synth circuits. In the Filtrex, it doesn’t do a great deal other than select which mode the envelope processor is going to be in. The 4016 is controlled by the ‘mode’ switch. For the envelope processor to be in envelope follower, or EF, mode, the FWR output needs to be patched into the lag generator. U12 (6,8,9) switches on, and U12 (10,11,12) is off. The positive voltage that is being produced by the FWR will now start to charge or discharge C36 up and down. The speed of the charging will be controlled by the ‘up’ pot, and the speed of the discharge wil be controlled by the ‘down’ pot. U10c buffers the voltage across C36 to create the positive going EF output signal. U10d inverts this to produce negative going voltages. The ‘envelope’ pot controls the depth of the effect. The position of the pot’s wiper will determine the polarity and the level of the final output signal. D3 and 4, along with R59, create a dead band around zero volts so the pot doesn’t have to be exactly in the middle for no modulation.

In EG mode, U12 is switched over to allow the output of the EG logic circuity to control the lag generator. The output of this logic circuitry is either high, +7.5V or low, 0V. The logic circuitry can operate in two modes, attack-decay (AD) or attack-release (AR).

Several sources can initiate the attack phase. One is the external ‘gate’ signal. This is a switch type signal that is either at around 0 volts when off, or any positive voltage greater than 3V when on. The Filtrex can easily handle greater voltages, within reason, without damage. D5 protects Q11 from any negative inputs and huge positive ones.

Other sources of triggering the attack phase come from the LFO and the threshold detector already discussed in this document. Both of these trigger the unit by pulling the collector of Q11 down to zero.

When a positive gate signal arrives, Q11 turns on and pulls its collector down to ground or 0V. This inverse version of the applied gate signal is sent to two destinations. One is another transistor, Q15. This is configured as another inverter. Thus the output of Q15 produces a copy of the gate signal that swings from 0 when off to +15V when on. R87 passes some current back to the first transistor. This creates a type of Schmitt trigger action which makes the transistors change state faster. It also allows slowly varying signals to trigger the Filtrex. For example you can use a slow sine wave or aftertouch CV to fire the EG.

The output of Q15 is passed on to a CR network that acts as a differentiator. This circuit produces a positive voltage spike when the gate goes high. The duration of the spike is determined principally by the values of C42 and R95. D12 prevents a negative spike being produced when the gate goes low. The positive spike triggers an RS flip-flop circuit based around two NOR gates, U13.

A flip-flop is a sort of a one bit memory, or latch. Once triggered by a positive going pulse at pin 12, it stays latched. You can only reset it by removing the power or a reset pulse at its other input, pin 9. When the flip-flop is latched, pin 10 goes high and pin 11 goes low. The output at pin 10 is passed via R90 and U12 to the lag generator's input, thus causing C36 to start to charge upwards. R90 is chosen to interact with R73 to give an input signal of 7.5V in the high state.

In any mode, removing the gate will reset the flip-flop. The inverted gate signal from Q11 goes to a second differentiator, C38, R93 and D11. When a gate signal is removed, the positive going edge runs through U13 (pins 1,2,3 & 4,5,6) to reset the flip-flop. Thus removing a gate signal will cause the lag generator's output to fall.

Another way to reset the flip-flop is in the AD mode. This utilises the actual output of the lag generator to control the discharging process. When the output of the lag generator exceeds a certain value, approximately +3.8V, the flip-flop is reset and the output voltage will drop.

This job is performed by a comparator based around U10a and Q10. The output of the lag generator is passed onto the comparator by another analogue switch U12 (1,2,13). In AR mode, this is switched off and the input to the comparator is held low by R84. In AD mode, the switch opens to allow the comparator to sniff the output of the lag generator.

When the voltage exceeds +3.8V or so, the comparator's output goes from 0V to +15V. This tells the flip-flop that the attack phase is over and the decay phase is about to start. Pin 10 therefore goes low and C36 is discharged via the 'down' pot.

R69 and R70 set the +3.8V threshold level. R78 with R79 provides a thin slice of positive feedback to force the comparator to switch cleanly... its another Schmitt trigger again.

The LFO circuit is quite simple. Its on page two of the schematics.

The first TL072 op-amp, U16a (1,2,3) forms part of the integrator. Any positive voltage applied to the right of R121 will cause the voltage to fall at the output of the op-amp. The speed at which the voltage falls is controlled by C51 and the size of the voltage applied to R121. If the applied voltage is negative the op-amp's output will rise. It is the integrator's output that will be used as the source for the triangle wave output.

The second half of the TL072 op-amp is used as a Schmitt trigger. It's output is either high at +13V, or low at -13V. If the output of the Schmitt is initially low, it requires +6V at the output of the integrator to make it go high. The integrator will need to produce an output of -6V to make the Schmitt go low again.

To make any oscillator you normally require an output to be fed back into the input. In a standard LFO like this one, the integrator is fed by the output of the schmitt trigger. Thus, a low at the output of the schmitt causes the integrator to rise. When the integrator’s output reaches a certain point, the schmitt switches state and the integrator’s output falls. The schmitt trigger changes state once again, and the process repeats itself....

The ‘LFO-rate’ pot allows a only a controlled proportional of the schmitt’s output voltage to reach the integrator. If the proportion is large, the voltage on R121 is large, and the integrator sweeps fast. If the proportion is small, the integrator sweeps slowly. R117 sets the minimum speed. Don’t be tempted to lower this value any more to get really slow sweeps. Input errors within the integrator op-amp will take over and your LFO won’t oscillate any more.

The square wave output is derived from the schmitt trigger’s output. D25 allows only positive excursions through. R124 and C54 act as a simple low pass filter to round of the waveforms edges a little bit. Very fast edges end up as CV breakthrough on the main audio output and are pretty unpleasant.

The trigger output is simply generated by a transistor, Q12, that turns on when the output of the schmitt trigger goes high. The ‘auto’ switch switches the function off by shorting the base to the emitter when not required.

The last thing to describe is the power supply. This is a standard ‘three terminal regulator' design straight out of the data book. R98 and C41 provide a decoupled version of +15V for the logic circuitry. The logic circuitry can generate little spikes on the power supply, that could get back into the audio if not decoupled properly. R8 and the ‘on’ LED provide power supply indication. I have put it on the negative supply only to even up the power drains on both rails.

There are three power inputs. This is different to the VCF-1 design. The Filtrex’s power supply can function either with a half wave rectifier for wall warts, or with full wave for internal transformers. This will covered in more detail later on the document.

Components

Most of the parts are easily available form your local parts stockist. I use Rapid Electronics, RS Components, Maplin and Farnell, here in the UK. The Filtrex was designed to be built solely from parts obtainable from Rapid Electronics and myself only. Rapid’s telephone number is 01206 751166. They offer a traditional ‘paper’ catalogue and take VISA card orders over the telephone.

In North America, companies called Mouser, Newark and Digikey are very popular. In Germany, try Reichelt, and in Sweden you can use Elfa. All companies have websites with their name in the URL.

The pots are Omeg Eco types with matching brackets. You could use any type you want, but not all pots have the same pin spacing. Not a problem, of course, if you are not fitting them to the board. In the UK, CPC, Maplin and Rapid sell the Omeg pots at a very good price. But note that none of these sell the pot brackets. The pot kit that I supply contains all thirteen pots and the pot brackets.

Alternatively, one can use the Piher P16 series of pots. These can be obtained from various sources including Farnell in the UK. However, I have found that getting the 1MB pots is problematic.

The resistors are generally ordinary types, but I would go for 1% 0.25W metal film resistors throughout, since these are very cheap nowadays. For the UK builders, then Rapid offer 100 1% metal film resistors for less than 2p each! For R104 and R178, which are 3M3 resistors you better go with a 5% type since high values of resistance are harder to obtain in metal film.

For the capacitors, there are no set rules. All the electrolytics (abbreviated to 'elect') should be over 25V, except where stated, and radially mounted. However, don't chose too higher voltage either. The higher the working voltage the larger in size the capacitor. A 220V capacitor will be too big to fit on the board. 25V or 35V is a good value to go for.

The pitch spacing of all the non-polar capacitors is now 5mm (0.2"). This may differ from some of the older Oakley boards you have built. For values between 1nF and 680nF, I use metalised polyester film types. These come in little plastic boxes with legs that stick out of the bottom. Try to get ones with operating voltages of 63V or 100V.

The ceramic capacitors should be 'low-K' ceramic plates. The lead spacing is 0.2" or 5mm. Do not chose cheap and nasty ceramic types, usually 'high-K', obtainable from some surplus places. These can lead to a noisy audio output.

The two horizontal preset or trimmer resistors are just ordinary carbon types. No need to buy the expensive cermet types. Carbon sealed units have more resistance to dust than the open frame types. Piher make a suitable type to use here. Pin spacing is 0.2" at the base, with the wiper 0.4" away from the base line.

The BC549 transistors can be pretty much any NPN transistor that corresponds to the same pin out. For example: BC550, BC548, BC547 etc. However, for the ladder transistors (Q1-6) I recommend using BC549 or BC550 only. These are low noise devices. Quite often you see an A, B or C suffix used, eg. BC549B. This letter depicts the gain or grade of the transistor (actually hfe of the device). The Filtrex is designed to work with any grade device although I have used BC549B throughout in my prototypes.

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. The two logic ICs are standard 4000 series CMOS. Typical part numbers are CD4001BE or HCF4016BE. You can use a 4066 in place of the 4016 if you wish.

The matched pairs required by the ladder are SSM2210P made by Analog Devices. You could use National's part LM394. Both are excellent devices but rather expensive. However, to reduce CV breakthrough you must use these parts. Each SSM2210 contains two ready matched NPN transistors. You could use individual transistors and match them yourself. But life is too short to spend an hour searching for the ideal pair. And they won't be thermally matched either.

The LM394 is not made in a DIL8 package any more. The only version that I could find is a metal can. Now, it is possible to get these to fit in the board. However, the pins must be bent to match the two rows that the DIL spacing allows. This job must be done very carefully so that the pins don’t get pulled from the package. The SSM2210P is supplied in an 8 pin DIL package and the Filtrex board was designed for this part.

R61 is a PTC or positive temperature coefficient resistor. This means it 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 R61. The PTC I use for this job is Farnell part number 732-278. Its a 1K +3000ppm/K 900mW device.

Both the two SSM2210P and the PTC are provided in the odd parts kit.

The 7815 and 7915 are three terminal voltage regulators. These are standard parts and have various prefixes and suffixes, eg. LM7815CT. The 7815 is a 1A +15V regulator, and the 7915 is 1A -15V regulator.

Input and output sockets are not board mounted. You can choose what types of sockets to use. I used plastic 1/4" jacks mounted on the rear of the rack. If you use metal sockets, like the excellent Switchcraft 112, you ought to insulate the sockets’ earth lugs from the metal casing. It is not essential but may help to alleviate hum pick up due to earth loops.

Each jack socket has two connections on the PCB, one signal and a ground. There is no need to use screened cable, so long as the connections are fairly short and you twist each pair of wires together.

The LEDs can be any type, although I recommend the use of standard round 3mm types. You will need to bend their legs if you want them to stick through the panel. More detail about mounting the LEDs is given on this later on this document. Many manufacturers do ready made preformed LEDs in little plastic boxes. These may be perfect for the job, but be careful that your LEDs have the cathode on the right hand side as you look at the front of the device. The Scheaffer front panel database was designed so that ordinary 3mm LEDs are to be used. The prepackaged preformed types will probably not fit. The LFO LED must be a bi-colour type, which means the one package contains both red and green LEDs wired in anti parallel. These are made in a water clear package.

The switches can be any style if you are not fitting them directly to the PCB. However, the PCB is designed to use the miniature toggle range from C&K. These are ‘type 2 horizontal’ non-sealed units. Only SPDT types are used. Three of the switches are ordinary ON-ON switches, whilst the MODE switch is an ON-OFF-ON. The latter is the same as an ordinary ‘up-down’ toggle switch, but it has an additional position in the middle. This third position neither connects the switch's wiper to either the bottom or top connection.

The manufacturer’s part numbers for the chosen switches are:

On-On	7101MD9AV2BE
On-Off-On	7103MD9AV2BE

Note that C&K are now part of the mighty ITT-Cannon organisation.

The mode switch may be obtained from Farnell, part number: 917-813. The other three switches are part number: 917-801. I believe these switches also to be stocked by Mouser and Digikey. I hope to be able to provide these switches as a switch kit at some point in the near future.

Heatsinks are not normally required. BUT this does depend on the power supply, local temperature etc. If you cannot hold the regulators for more than 5 seconds after it is has been powered up for at least 5 minutes, then you need heatsinks. I use the clip on type of heatsinks available from most suppliers. These require no nuts and bolts, or heatsink compound. They simply push on, and friction holds them in place. Try not to bend the legs of the power devices when you push the heatsinks into place.

The topic of power supplies will be covered in its own section later on in this document.

Finally, if you make a component change that makes the circuit better, do tell me so I can pass it on to others.

Parts List

Resistors

All resistors 5% or better. 0.25W types. Those items marked with * need to be 1%, or better, 0.25W metal film resistors.

R is shorthand for ohm. So 22R is 22 ohm. 1K5 is 1,500 ohms or 1.5 kilohms.

100K	R66, R67, R64, R41, R115, R107, R121, R68, R89, R119, R120, R126, R93, R95, R85, R90, R73, R96, R97, R72, R77, R84
100R	R117
10K	R3, R37, R51, R52, R38, R114, R111, R101, R94, R102, R124, R59, R79
10K*	R45, R46, R32, R31, R33, R35, R42, R44
12K	R108
15K	R69, R40
1K	R11
1K +3000ppm/K	R61
1K5	R25, R118
1M	R23, R87
220K	R20, R63, R91
22K	R7, R12, R43, R100, R113, R106, R75
22K*	R21, R22, R29, R28
22R	R123, R9, R122, R5, R4, R53, R54, R116, R110, R76, R81, R82, R98
2K2	R57, R58, R92, R8, R47
2K2*	R24, R26
330R	R10, R18, R17, R15, R13

39K	R105
3K	R109
3K9	R1
3M3	R104, R78
470K	R2, R62, R50
470R	R19, R14
47K	R36, R34, R27, R30, R56, R49, R48, R125, R71, R80, R83, R86, R39
47R	R55, R112
4K7	R99, R88, R74
56K	R60
680K	R103
68K	R65
6K8	R6, R16, R70

Capacitors

1000uF, 35V elect	C39, C48
100uF, 25V elect	C41
100nF, 63V poly	C51, C20, C18, C54, C30, C37, C49, C40, C53, C34
10uF, 35V elect	C36, C21
1nF, 100V poly	C28, C38, C42
220pF ceramic	C31
220uF, 10V elect	C9, C11
22pF ceramic low-K	C3, C7
22uF, 25V elect	C2, C52, C55, C1, C8, C16, C10, C29, C4, C6, C27, C45, C47, C23, C25, C44, C35, C33, C32, C50
33pF ceramic low-K	C26, C24, C46
470nF, 63V poly	C5, C17, C19, C22
680n, 63V poly	C43
68n, 63V poly	C15, C14, C13, C12

Discrete Semiconductors

BC549	Q1-15
1N4002 or 1N4004	D8, D23, D9, D24, D14, D13, D19, D18
1N4148 or 1N914	D1, 2, 3, 4, 6, 7, 10, 11, 12, 15, 16, 17, 20, 21, 22, 25
10V 400mW zener	D5
LED 3mm green	TRIG
LED 3mm red/green*	LFO
LED 3mm red	PEAK
LED 3mm yellow	ON

* This is a bi-colour LED in a waterclear package. Do not fit an ordinary single colour LED in this position.

Integrated Circuits

OP275	U1
NE5534	U8

TL072	U5, U6, U7, U9, U16
TL074	U14, U4, U10
7815	U15
7915	U11
SSM2210P	U2, U3
4001	U13
4016 or 4066	U12

Others

10KA X 2	RESONANCE
10KB	VOLUME
1MB	UP, DOWN
47KA	SMOOTH, FREQ, FIZZ, THRU, THRESHOLD, ENVELOPE
47KA X2	GAIN
47KB	LFO-RATE, LFO-DEPTH
2K2 trimmer	TRIM
470K trimmer	BAL
SPST on-on	CONT, AUTO, WAVE
SPST on-off-on	MODE
1/4" Sockets	input, output, side chain input, CV, Gate

You will also need a power source of some kind. The recommended supply is a wall-wart supplying 15V AC at 300mA minimum. See later for more information.

A small amount of insulated multistrand wire is needed. This will be used to connect the sockets and power supply to the board.

IC sockets are to be recommended, especially if this is your first electronics project. You need nine 8 pin DIL sockets, and five 14 pin DIL sockets. Choose 'turned pin' or 'dual wipe' types.

Building the Filtrex

The printed circuit board is flashed with solder around the pads. This helps the soldering process and keeps the board solderable for many years. This flashing is with solder that contains lead. You should therefore wash your hands after handling the boards and do not place the boards in your mouth. It is also recommended that for best results this board is soldered with lead-tin solder.

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. To date this has happened only a few times across the whole range of Oakley PCBs. The most common error 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!

I always use water washable flux in solder these days for my board manufacture. In Europe, Farnell and Rapid sell Multicore's Hydro-X, a very good value water based product. You must wash the PCB at least once every two or so hours while building. Wash the board in warm water on both sides, and use a soft nail brush or washing up brush to make sure all of the flux is removed. Make sure the board is dry before you continue to work on it or power it up. I usually put the board above a radiator for a few hours. It sounds like a bit of a hassle, but the end result is worth it. You will end up with bright sparkling PCBs with no mess, and no fear of moisture build up which afflicts rosin based flux. Most components can be washed in water, but **do not** wash a board with any trimmers, switches or pots on it. These can be soldered in after the final wash with conventional solder or the new type of 'no-clean' solder.

I have found that if you are using a very hot soldering iron it is possible to run your iron so hot as to boil the flux in the 'water washable flux' or some types of 'no-clean' solder. This is not a good idea as it can create bubbles in the solder. If you prefer to have a fixed temperature iron, then it is best to get a 18W one for this purpose. I use an ordinary Antex 240V 25W iron with a Variac power supply running at 200V. This seems to work well for 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 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. There are three types of diodes used in this project. Most are ordinary signal diodes, the 1N4148. You have eight bigger black 1N4004 types. And one zener diode. When all the diodes are in place, double check all are pointing the right way.

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.

The transistors are all the same type on this board. 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. Turn the board over and cinch the two outer leads on the flip side, you can leave the middle one alone. Now solder the middle pin first, then the other two once the middle one has cooled solid.

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 parallel to the flat side of the transistor.

The polyester capacitors are like little blue 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.

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.

The bigger 1000uF capacitors should be soldered flush onto the board using no-clean or conventional solder. On no account must these two parts be put in the wrong way.

For U11 and U15, the two power devices, it may be necessary to preform the leads before putting them into the board. I use a pair of fine nosed pliers to bend the middle leg outwards. I bend the lead firstly near the body of the device at an angle of 45 degrees. Then where the metal leg thins, I bend it again so that it becomes parallel to the other two. The device should then fit snugly into place on the board.

Whenever you need to bend legs of any semiconductor device be gentle. Its a good idea to earth yourself too. Touching a nearby radiator or oscilloscope earthing tag is usually enough.

I would make the board in the following order: resistors, diodes, IC sockets, small non-polar capacitors, transistors, electrolytic capacitors. Then the final water wash. Do not fit the pots, trimmers, LEDs or switches at this stage. The mounting of the pots, LEDs and switches requires special attention. See the next section for more details.

Mounting the Pots, LEDs and Switches

If you are using the recommended Eco pots, then they can support the PCB with specially manufactured pot brackets. You will not normally need any further support for the board. When constructing the board, fit the pot brackets to the pots by the nuts and washers supplied with the pots. Now fit them into the appropriate holes in the PCB. But only solder the three pins that connect to the pot. **Do not** solder the pot bracket at this stage. When you have soldered all the pots you can fit the board to your front panel. Position the PCB at right angles to the panel, the pot's own pins will hold it fairly rigid for now. Then you can solder each of the brackets. This will give you a very strong support and not stress the pot connections.

The Omeg pots are labelled A, B or C. For example: 47KA or 100KB. Omeg uses the European convention of A = Linear, B = logarithmic and C = Reverse logarithmic. So a 1MB is a 1 megohm log pot.

The LEDs must be fitted carefully if you are using the directly mounted technique in conjunction with the Schaeffer panel design. Although this sounds fiddly, its actually quite easy and it reduces wiring, interference and possible errors.

Remove the front panel so that you just have the board again. Get the four LEDs and find the cathode for each one. Make sure the cathode of the LEDs will go into the square pad, pin 1, on the board. Carefully bend the LED's legs at a point 6mm away from the plastic body of the LED. The legs should be bent by 90 degrees so that the legs are pointing straight down. Check to see if they fit into the board. The bottom of the LED's body should fit just flush to the board edge. Fit all four LEDs to the board but do not solder them in at this stage. Let their legs poke through, there's no need to cut them down yet. Now fit the front panel again to the board and tighten the pot nuts. You should find that the board now fits snugly into position and each LED should be just poking out of its hole neatly, albeit loosely. Align the LEDs if they aren't quite straight and solder each one in turn, trimming its leads nice and short afterwards.

With panel removed once again, you can now fit the switches. The C&K PCB mountable switches should fit tightly into their respective holes on the board. Make sure the 'on-off-on' switch goes into the MODE location. You may need to use a pair of fine nosed pliers to help the flexible gold pins into the board holes. Make sure the switch body is flat against the board. The switch should stay in position without solder for the moment. Refit the front panel and make sure the round switch barrel fits into its hole in the front panel. Now solder all the pins on the switch including the securing pins to the front.

That completes the soldering of the front panel components.

Connections

You are going to need five 1/4" jack sockets to connect your PCB to the outside world. Each jack is connected to the rectangular boxes on the PCB marked with its function. The ground is always on the left hand side as you look at the board with the pots facing forwards. This goes to the jack socket's ground pin, ie. the one that will connect to the barrel of the plug when connected. For each socket, it is a good idea to twist the wires together in pairs. Use two different colours to tell them apart, and try to keep the wiring as short as possible to prevent picking up hum and other stray fields. You can use screened cable if you wish, and this must be the case if you are using a wooden case to house the Filtrex. The screen must go to the left hand side pad of each connection, ie. it carries the ground. If you are using a metal case, then it will be a good idea to use plastic jack sockets. This way you can be sure that you will not get any secondary paths to ground via the casing.

Power Supplies and things that can kill if you don’t do them properly...

The recommended option is to use an insulated wallwart or AC adapter. These can be bought from most places and are used external to the Filtrex housing. They are very safe since all the nasty dangerous stuff is kept inside the wall-wart. You won’t hurt yourself with the output from one of these unless you stick it in your mouth!! You need a 15V or 18V AC output at 250mA or higher rating. Do not use a DC output type. Although the latter are the most common type of wallwart for guitar effects pedals, they will not work with the Filtrex. To reiterate, because this is really important, it must say 15VAC or 18VAC on it somewhere.

In the UK they can be bought from Maplin Electronics. In North America, US Robotics make various types.

To connect your wall wart to the Filtrex, you need a suitable connector. The standard type is the barrel type as found on most effects pedals. Make sure you get the right socket for the plug you have on the wall wart. Some wall warts give you a little bundle of different types to choose from. Either way, make sure the socket you get allows the plug to slip in easily yet not break connection when wiggled gently. If you are making up your own plug for it, since it is AC, it does not matter which wire goes to what. There is no + or -.

The socket must be connected to AC1 and AC2 on the PCB. AC0 and AC3 are both left unconnected. If you have a metal socket and metal case, make sure the that AC1 is connected to the outer shroud of the socket, ie. the one that goes to the barrel of the plug. Failure to do so may result in burnt out wall warts or at the worst excessive hum. This is not to do with the polarity of the AC. This is to do with the fact that the AC1 pad on the PCB is connected to analogue ground. You don’t want the case to be connected to AC2, which will have a voltage on it that is bouncing up and down 50 or 60 times a second.

The following advice is only for those who know how to wire mains rated equipment safely. If you do not know how to do this then make no attempt to do so. I do not endorse this method of powering any Oakley equipment. It is up to you to use your PCB wisely. I take absolutely no responsibility for your actions with this board. I will offer no further advice than what you see below in italics:

Transformer rating: Secondaries: 18-0-18 @ 250mA or 18-0, 18-0 @ 15VA total
Connect common, or centre tap, to AC1 and/or AC0. Others to AC2 and AC3.
Line fuse: T250mA

The Filtrex may also be powered from the MOTM or Oakley power busses. The power socket is 0.156” Molex/MTA 4-way header is marked as MOTM on the PCB. Friction lock types are recommended. The pin out is as follows:

Power	*Pin number*
+15V	1
Module GND	2
Module GND	3
15V	4

Pin one is depicted by the diagonal on the legending. If you are using the MOTM system to power your Filtrex, be sure not to fit the following components:

U15, U11, D24, D9, D13, D14, D18, D19, C48, C39, C37, C49

You can also use the Filtrex to power other MOTM/Oakley modules using the MOTM header. However, be sure not to exceed the power rating of the wallwart, transformer, heatsinks and smoothing capacitors.

Trimmers

TRIM: Adjust this trimmer so that the filter bursts into oscillation when the 'resonance' pot is moved close to its maximum setting. The filter should oscillate from about 100Hz to over the range of your hearing. Or you can adjust it so that it never goes into oscillation at all. This will prevent you from accidentally damaging your ears, your tweeters and upsetting your neighbour's dogs. But hey, that would be boring. Live life in the fast lane. Trim it up so that it just oscillates when the resonance it up full.

BAL: Listen to the 'Smooth' output. Set the Resonance and Frequency pots to their mid positions. Turn the LFO modulation depth pot to its maximum value and select the triangle wave. Adjust BAL until the clicking or buzzing becomes minimised.

Housing your unit

The PCB has been designed to fit into a standard 1u high 19" rack unit. Your local parts distributor will have these. Good rack units are quite expensive, and will contribute heavily to the final cost of your completed Filtrex. Expect to pay around £35 or so.

If you find a good supplier of low depth 1U metal racks in the UK, I would be pleased to hear from you. Maplin sell excellent range racks made by Sherman, but all of them are very deep. Bryant Broadcast and RS Electronics Ltd do have a range of rack units that may be suitable.

The Bryant Broadcast ones are superbly made, but they do not allow you to use a 3mm thick Scheaffer front panel. Their cases actually utilise the front panel as part of the enclosure. Simply swapping the Bryant panel with one obtained from Schaeffer will not work. Of course, if you are drilling out the Bryant panel to the Scheaffer plan, then this would indeed work wonderfully. Bryant do custom metal work, so it may be possible to try their services. This is one area I would like to try in the near future. Scheaffer are also able to engrave panels that are sent to them. In theory, one could send them the blank Bryant panel and they could engrave this with the Frontplatten database found on the Filtrex website. I have heard of one person you has tried this and he was very pleased with the result. Remember if you do decide to do this, you must remove the four mounting holes from the database as these are already present on the Bryant panel.

If you buy the cases made by Vero, you will find that the height of the unit internally is quite restricting. The bottom and lower panels have 6mm folds in them at the front. This effects the amount of space available for the pots and circuit board at the front panel. It is possible to use these cases as I have done, but I needed to cut back the three pins on each pot to prevent them shorting with the case. The pot bracket pins actually prevent the case from then touching the pot’s pins. This is all right, but you need to allow a minimum of 0.5mm slack when you fit the front panel to the case.

The other thing to beware is the heatsinks. Don’t let either one of them touch the top panel, since this would cause major problems. This shouldn’t happen if you make sure the regulators ICs are fitted tight against the PCB.

For those of you fitting an internal toroidal transformer. Please, please make sure there is no way the top metal disc of the transformer's mounting can touch the top of the casing. If the metal support of transformer together with the case makes a complete loop around the core, then you have a shorted one turn secondary. (‘well, there was a large hum, more of a buzzy rattle really, then a smell of burning rubber and then a lot of smoke... ’)

It is possible to fit the unit in a 2U unit. This was successfully done by Klas Anderson with his VCF-1. He simply mounted half of the pots on the PCB. The other half were mounted beneath the board and wired in by hand. This format allows you to use bigger knobs for easier tweaking.

The PCB will be supported well by the pots and pot brackets. However, this does give some people nightmares so in this case it may be a good idea to provide additional support. Small holes, to fit M3 bolts, have been provided on the PCB to do this. Feel free to enlarge these holes if you wish. My prototypes have been very happy just supported by the pots. However, my rack is bolted to the wall, so it doesn’t get moved around much! If you intend to take it out on the road, extra support may be a good idea.

Final Comments

I hope you enjoy building and using the Oakley Filtrex.

Please feel free to ask any further questions about construction or setting up. If you cannot get your project to work, do get in touch with me or the Oakley Synths list on Yahoo groups. Sometimes, it can be the simplest things that can lay out a project. If we still can't get the completed and undamaged module going together, you can send it back to me to fix, but you will have to pay for postage both ways, any parts required and my time at £20 per hour. This service is taken up only very rarely, so it just goes to show how easy it is to get an Oakley project to work first time. If you are sending the item from outside the EU, then be sure to say on the customs label ‘item being sent for repair only’.

Occasionally, there may be an error in the parts list. I have checked the documentation again and again, but experience has taught me to expect some little error to creep past. The schematic is always the correct version, since the parts list is taken from the schematic. So if there is any problem, use the schematic as the guide. If you do notice any error, please get in touch.

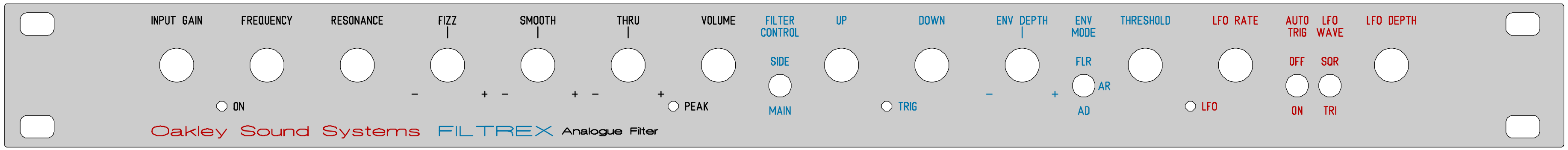
Please further any comments and questions back to me, your suggestions really do count. If you have any suggestions for new projects, feel free to contact me. You can e-mail, write or telephone me. If you telephone then it is best to do this on Monday to Friday, between 9 am and 6 pm, British time.

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 Oakley-synths, Synth-diy and MOTM mailing lists.

Tony Allgood

Cumbria, January 2005

Errors will always occur in the preparation of a document. Please forward any errors found to me so I can correct them.


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