Oakley Sound Systems

5U Oakley Modular Series
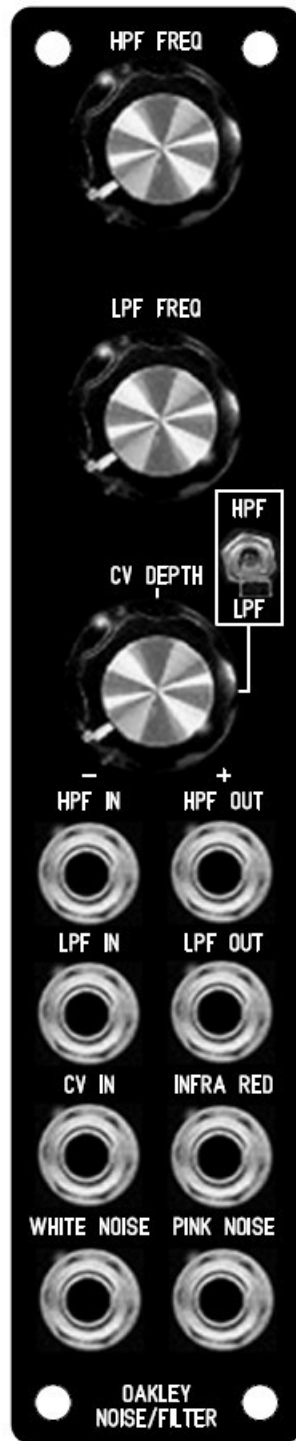
Noise Generator and Dual Filter

PCB Issue 4

Builder's Guide

V4.0.3

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HPF FREQ

LPF FREQ

HPF

LPF

CV DEPTH

−   +

HPF IN   HPF OUT

LPF IN   LPF OUT

CV IN   INFRA RED

WHITE NOISE   PINK NOISE

OAKLEY
NOISE/FILTER

The suggested panel layout for the MOTM format of the Oakley Noise/Filter module.

Introduction

This is the Project Builder's Guide for issue 4 of the Noise/Filter 5U module from Oakley Sound.

This document contains a basic introduction to the board, a description of the schematic, a full parts list for the components needed to populate the boards, and a list of the various interconnections.

For the User Manual, which contains an overview of the operation of the unit, the history of the various board issues, and all the calibration procedures, please visit the main project webpage at:

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

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

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

The Noise/Filter issue 4 PCB
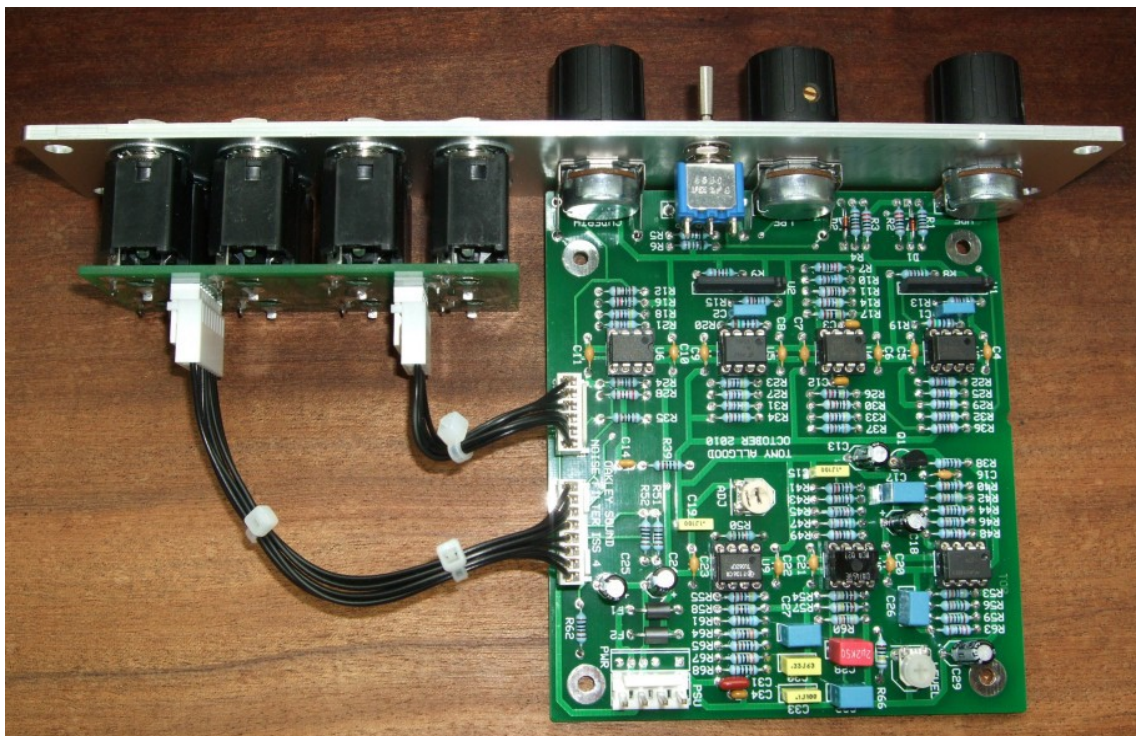
The issue 4 PCB mounted behind a 1U MOTM format Schaeffer panel in natural finish. Note the use of the Sock8 board to make easy the complicated socket wiring.

I have provided space for the three main control pots on the PCB. If you use the specified 16mm Alpha pots and matching brackets, the PCB can be held firmly to the panel without any additional mounting procedures. The pot spacing is 1.625" and is the same as the vertical spacing on the MOTM modular synthesiser and most of our other modules.

The design requires plus and minus 15V supplies. The power supply should be adequately regulated. The current consumption is about 30mA for each rail. Power is routed onto the PCB by a four way 0.156" MTA156 type connector or the special five way Synthesizers.com MTA100 header. You could, of course, wire up the board by soldering on wires directly. The four pins that are used on the four way header are +15V, ground, earth/panel ground, -15V. The earth/panel connection allows you to connect the metal front panel to the power supply's ground without it sharing the modules' ground line. More about this later.

The PCB has four mounting holes for M3 bolts, one near each corner. These are not required if you are using the two 16mm pot brackets. The board size is 101 mm (high) x 116 mm (deep).

The board has been laid out to accept connection to our Sock8 socket board. This small board speeds up the wiring of the four sockets and reduces the chances of mistakes.

Circuit Description

The noise and the two filter sections are essentially separate circuits that only share the same power supply. The power supply section is shown on page 2 of the schematics. Power is applied to the board through the PSU, a 4-way MTA156 / Molex 0.156" connector, or PWR a special five way MTA header. F1 and F2, small axial ferrite beads, provide some high frequency resistance, and along with C24 and C25 prevent the board from being effected by any noises on the power rails. They also help keep any unwanted noises going the other way too.

Additional decoupling is also provided elsewhere on the board by eight other capacitors shown above right of the power inlet circuitry. 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.

Two grounds are provided, one for the circuit itself, and one for the earthing of the jack sockets on the front panel.

The power supplies to each of the op-amp ICs are shown separately from the main schematics to avoid cluttering the diagram.

Still on page 2 we will consider the two filter sections. These are actually identical circuits even though one is used as a low pass filter and the other as a high pass filter. The only real difference between the two circuits is from where in the circuit we take the output.

The filters are made from a circuit described as a state variable filter or SVF for short. Unlike the Oakley SVF module this filter is only one pole. Only one active part actually controls the cut-off frequency and not two like in the SVF module. The part that gives us control over the cut-off point is an audio voltage controlled amplifier (VCA) made by the THAT Corporation, part number 2180. These parts are normally used in gain control, but it is a simple matter to allow them to work in filters too.

The precise mathematics of the one pole filter is beyond the scope of this User Guide, but a more detailed account of its operation can be found in the THAT Design Note number 130 found on their website. www.thatcorp.com

Those of you who have built our VC-ADSR may recognise the circuitry involved in the one pole filter as I used a similar circuit in the ADSR core. To explain how the circuit works we will look at the low pass filter section in more detail. Two basic components make up the filter circuitry. One is a differential amplifier, this is U5b (pins 5, 6, 7). The other is a more complex circuit called an integrator, this is based around the VCA and U5a (pins 1, 2, 3).

An integrator is a charge storage device. Think of it like a bucket for electricity. It can be filled up and drained at will, and the voltage output of the integrator is analogous to the amount of water in our bucket. The speed at which the bucket can be drained or filled is controlled by the VCA. To be more specific the VCA actually creates a current output that allows C2 to be charged up by the op-amp U5. The VCA's current output is dependant on two things; one, the

current through R15 and the voltage at the VCA's control pin, pin 3. In fact, the current through R15 is multiplied by the mathematical logarithm of the voltage at pin 3.

Now pin 3 is controlled by the front panel pot and any connected external CV. This way the speed of the integrator can be manually and automatically adjusted. The voltage at pin 3 varies from around +290mV (lowest frequency) to -90mV (highest frequency) when controlled by the front pot alone. The gain of the VCA goes up the more negative the voltage. However, if the gain of the VCA is allowed to go too high, then the whole filter will become unstable and an oscillation is seen at the output of U6b (pins 5, 6, 7). The actions of the resistor network R7, R4 and R3 in conjunction with D2 act as a voltage clamp on the control voltage. This prevents excessively negative voltages from reaching pin 3 and thus prevent the VCA from being opened too far and causing the circuit to oscillate.

The current through R15 is dependant on the voltage output of op-amp, U5b (pins 5, 6, 7). This op-amp is a differential amplifier and this particular circuit subtracts the input signal, coming into the circuit via R31, with the output of the integrator, coming in via R23. This now means that the integrator has to respond to both the input signal and its own output. The upshot of this is that the integrator's output will try and follow the input signal, but only as fast as the VCA will allow that capacitor to charge and discharge. This produces the classic 'lag circuit'. This slowness of the integrator's output to react to the input signal gives the desired low-pass effect.

The high pass response from a similar circuit to the low pass filter. Indeed the only difference is that we take the signal from the output of the differential amplifier. You may want to consider this yourself in more detail. However, a simple way of looking at it is to consider that this output is formed by the difference in the input signal and low pass output. If the low frequencies are taken out of the input signal, then you are left with only the high frequencies, and thus, the high pass response. There is one small addition to the high pass filter and that is C14. This low value capacitance, in conjunction with R39, acts to stabilise the filter output at very high frequencies.

It should be noted that a one pole filter creates a roll off of around -6db/octave past its cut off frequency. So the effect of the filter is less pronounced than ones of higher orders like Moog Ladders and so on. It is not possible to make a one pole filter have gain at its cut-off point, so there is no chance of a resonance control or self-oscillation at its cut-off frequency.

Now let us look to the first page of the schematic. This covers the noise generator, the pink noise filter and the unstable chaos of the infra-red output circuity.

The noise generator is Q1 and I will discuss choosing this device in more detail later in this document. All devices produce noise of some sort. This is caused by the random movement of charge carriers within a semiconductor or metal at temperatures above absolute zero. At room temperatures things are really shaking! Various devices can be utilised for their noisy behaviour. Zener diodes are very noisy particularly around their reverse breakdown voltage but, because of their glass enclosures, tend to be affected by ambient light levels. Transistors can also behave as zeners too when the base-emitter junction is connected up the wrong way. The voltage in which the zenering affect is seen is called the reverse breakdown potential and we normally try to avoid getting anywhere near it since it produces poor results in amplifiers and other circuits. However, in this project we lovingly embrace this point of the base-emitter

junction's V/I curve and deliberately force the junction to conduct the wrong way just a little bit.

The emitter is made more positive than the base via R40 and the resistor thus provides the reverse 'emitter' current. The positive end of R40 is a well decoupled (filtered) version of the +15V supply rail. Any extraneous noise on the power supplies is not wanted here and R38, C13 and C16 filter the power supply accordingly.

The base-emitter junction of the transistor will effectively regulate the voltage at the emitter with respect to ground, since it is behaving as a zener diode. However, this voltage is not that stable. Superimposed on this 'stable' DC voltage is a tiny amount of random noise. Noise is a wideband AC signal. It contains many different frequencies. In fact, white noise contains all frequencies (within reason) at the same average power.

C17 allows this small AC signal through while blocking the larger DC voltage. The next stage is a high gain amplifier, with a massive amplification of over a hundred. The noise signal is around 2mV or so, and this first stage of amplification takes it up to around 200mV. It is very possible that DC errors at the input of the op-amp U7a will produce output offsets. Normally these are very small, but any offsets at the input will also be amplified by the op-amp. C26 removes any DC from the output signal before the next gain stage.

The second stage of amplification should bring the noise level up to around +/-5V. However, due to the variable nature of noise generators in general, the gain of this stage is made variable. The LEVEL trimmer allows the user to set the output signal to a correct level and account for any differences between noise making devices. It is important to set this level correctly, otherwise performance of the infra red output will be impaired. The white noise output is available from this second stage of amplification via a 1K resistor, R62.

Now, you maybe thinking, why do we need two stages to amplify? Surely, we could have just used one big amplifier. Although it is possible to make a single op-amp gain stage of 1000 or more, it has one major disadvantage: As an amplifier's gain rises, we find its bandwidth lowers accordingly. The bandwidth of an amplifier is its ability to amplify all input frequencies accordingly. A small bandwidth means a restriction in usable frequency response. For example, it would be not very useful here to have an amplifier with a gain of 1000 but with a maximum frequency of 4kHz. By using two lower gain amplifiers in cascade, each will have a bandwidth exceeding the audio band, so the overall frequency response of both of our amplifiers will not effect the quality of the audible noise. Even so, I do recommend that you use a good quality fast op-amp here. The LF412 is an excellent choice, and also gives less offset error than a TL072.

To make pink noise we need to create a signal that contains all frequencies as before, but this time with equal power per octave. Without going into the mathematics of this, we need to filter our white noise so that frequencies roll off at a rate of 3dB per octave. Now, normal one pole RC filters produce roll off not exceeding -6dB/octave. So to make -3dB/octave we must get very clever. Fortunately for me, other people have done all the difficult work and have published their findings. This one I built many years ago from an original design in Wireless World. It was used in an audio test set and produces a very good quality pink noise. Some synthesisers use simpler methods, but this one sounds better to my ears so why not use it.

U9b (pins 5, 6, 7) provides some gain for the losses incurred within the RC network, and the pink noise output is available via R52.

The infra red output is heavily inspired by the Polyfusion noise module. I have made only a few changes to their design since it works so well.

R43 and C15 provide further low pass filtering of the pink noise and prevent any unwanted switching clicks from the next stage from leaking back into our pink noise output. The first half of U8a (pins 1, 2, 3) is configured as a comparator. Pin 1, the output, will either be at around +14V or -14V depending on the voltages present at its two inputs. Without R49, if the '+' or 'non-inverting' input is at a higher voltage than the '-' or 'inverting input', the opamp's output will be positive. If it is lower, then the output is negative. But R49 provides *hysteresis*. This means the voltage at the inverting input must be substantially larger then the non-inverting input to get the op-amp to flip states. This prevents the op-amp's output from juddering when the two inputs are very close to each other. The signal we see at the output of U8a is a rectangular one with random flips from positive to negative. We will talk about R41 and the ADJ pot later.

The next op-amp stage is an interesting low pass filter arrangement. This is not a traditional low pass filter as it has two distinct cut-off frequencies at 0.2Hz and 0.04Hz. Suffice to say, its output will respond very slowly to any changes at its input. The rapidly changing output of U8a is thus turned into slow sluggish random movements by this 'lag' circuit. Furthermore, feedback is provided by R47 which feeds back this slow moving output to the comparator. This increases the randomness of the signal. I found experimenting with the value of R47 produced some unexpected results.

The ADJ trimmer adjusts the offset voltage applied to the comparator circuit. It is provided to keep the overall output centred around 0V. Two things seem to affect the amount of offset required. One; the offset specifications of the op-amp. I have chosen to use a 1458 as in the original design. This does not have the best offset specifications, but does have a limited slew rate (the speed the op-amp's output stages can swing from one voltage to another) which will act as a further low pass filter and prevent any fast edges to contaminate our clean analogue design. And two; the pink noise will contain a certain degree of imbalance which this circuit is sensitive to. This seems to be affected not by the DC offsets of the pink noise output amplifier but by the spectra of the noise itself after filtering.

The output of U8b is typically around 5V to 10V peak. The value of which seems to depend on the quality of the noise source. After further low pass filtering with R50 and C19, the random output is fed to another op-amp to buffer the signal before the final output.

Issue 4 Noise/Filter 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 kilohm. R is shorthand for ohm. So 22R is 22 ohm, 1K5 is 1,500 ohms or 1.5 kilohm. 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.

Resistors

Resistors 1/4W, 5% or better.

22R	R63, R48
100R	R38, R68
220R	R3, R2
330R	R67
620R	R50
1K	R35, R44, R62, R51, R52, R39, R1, R4, R65
1K2	R45
1K5	R59
2K2	R36, R7
3K3	R64
4K7	R58
5K1	R9, R5, R8, R6
6K8	R61
10K	R11, R30, R12, R41, R16
14K	R26, R17
18K	R49, R15, R13
22K	R28
36K	R10, R37
47K	R18, R33, R21, R55, R32, R29, R14, R31, R34, R22, R23, R27, R25
51K	R47
100K	R46, R43, R40, R24
150K	R56
470K	R60, R53, R42
1M	R54, R57, R66
10M	R19, R20

Capacitors

100n axial multilayer ceramic	C4, C5, C6, C7, C8, C9, C10, C11, C16, C20, C21, C22, C23
100p 2.5mm C0G ceramic	C14
470p 2.5mm C0G ceramic	C34, C12, C3
2n2 polyester film box	C1, C2
33nF polyester film box	C31
100n polyester film box	C15, C19, C33
330nF polyester film box	C30
1uF polyester film box	C17, C26, C32, C27
2u2 polyester film box	C28
22uF, 35V electrolytic	C29, C13, C18
2u2, 35V electrolytic	C24, C25

Discrete Semiconductors

1N4148 signal diode	D1, D2
BC547 - see note below	Q1

Q1 is the noise transistor. I tried many different types of transistor in this circuit. I found BC548s and BC547s to be best. I have no idea why they should sound better than the BC549s or any other I tried, but they did. I didn't like the sound from any of the BC550s I tried at all. I suggest buying five BC547s and try each one in turn. See separate section later on how to do this.

Integrated Circuits

TL072CP dual FET op-amp	U3, U4, U5, U6, U9
THAT2180LC voltage controlled amp	U1, U2
LF412CN dual precision FET op-amp	U7
1458 dual op-amp	U8

Please ensure that the THAT2180 VCA chips are seated into the board correctly. The notch at the top of the chip represents pin one (square pad on PCB) and is fitted so that it is towards the top of the board.

Trimmers

20K 6mm horizontal	ADJ
50K 6mm horizontal	LEVEL

Pots

47K linear 16mm Alpha pot	LPF, HPF, CVDEPTH

Alpha pot brackets   2 off for LPF and CV DEPTH

Switch

SPDT, on-off-on   LP/HP - to be connected with single core wire

Miscellaneous

Leaded Ferrite beads   F1, F2
Knobs to fit 6mm shafts   Three off

Molex or MTA 4 way header   PSU – Oakley/MOTM power supply
MTA100 6-way header   PWR – Synthesizers.com power supply

Molex/MTA 0.1" header 6-way   UPR – for connecting to sockets
Molex/MTA 0.1" header 8-way   LWR – for connecting to sockets
Molex/MTA 0.1" housing 6-way   UPR – for connecting to sockets
Molex/MTA 0.1" housing 8-way   LWR – for connecting to sockets

1/4" sockets   Eight off mounted either on the Sock8 board or on panel

Hook up wire (26awg).

You may well want to use sockets for the ICs. I would recommend low profile turned pin types as these are the most reliable. You need seven 8-pin DIL and two 8-pin SIL sockets.

Additional components required if using optional Sock8 board

Molex/MTA 0.1" header 6-way   UPR
Molex/MTA 0.1" header 8-way   LWR
Molex/MTA 0.1" housing 6-way   UPR
Molex/MTA 0.1" housing 8-way   LWR

112APCX Switchcraft 1/4" socket   SK1, SK2, SK3, SK4, SK5, SK6, SK7, SK8

If using Molex KK you'll also need at least 28 crimp terminals.

Suitable lengths of wire to make up the two interconnects and three cable ties.

Wire Link

For this module do not fit the wire link, L1, on the Sock8 PCB.

Mounting the LPF/HPF switch

Simply connect the switch to the board so that the switch's solder tangs connect to the PCB pad immediately below it. This can be done with either insulated multistrand wire or solid core wire.
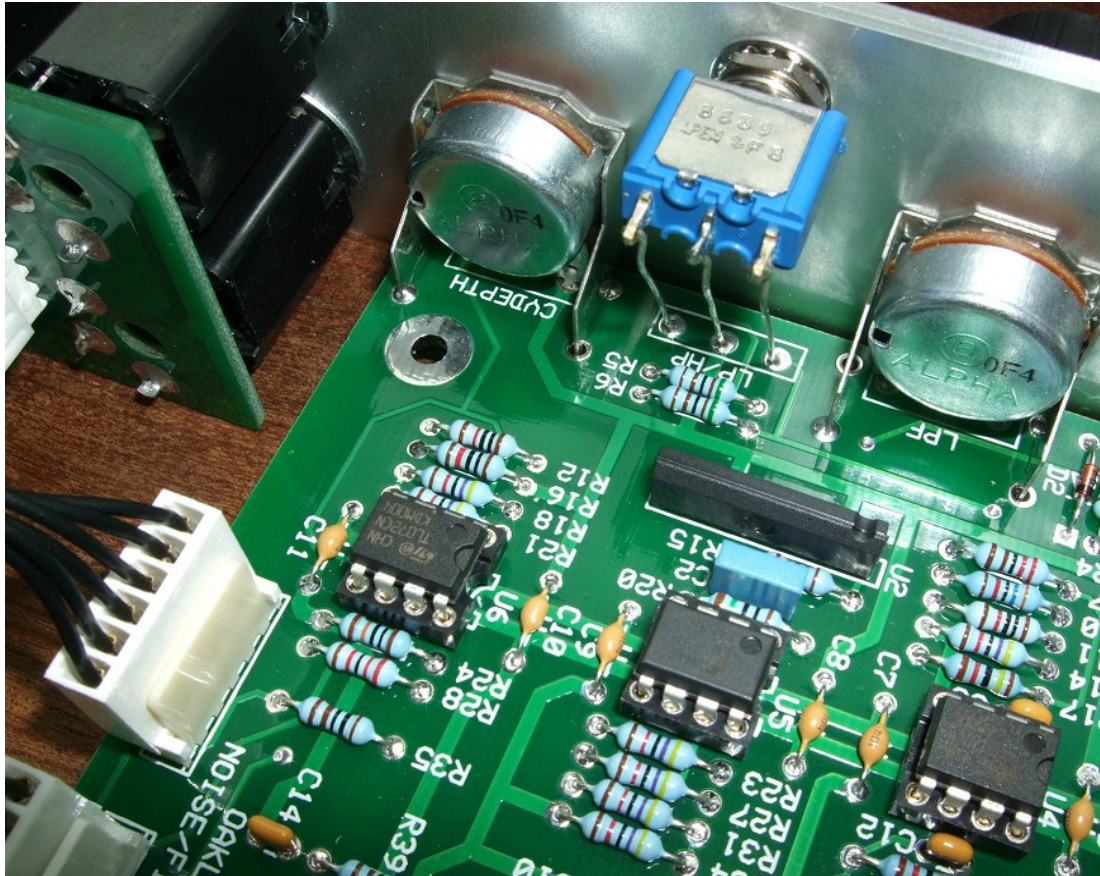
This is an APEM on-off-on toggle switch. Note the spring washer of the switch is on the inside of the panel. The switch can then be tightened from the rear reducing the chances of scratching the front panel.

Note also the position of the THAT2180LC chip. The notch on the chip's case is towards the top of the board.

Connections

Power connections – MOTM and Oakley

The PSU 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

Pin 1 on the LWR header has been provided to allow the ground tags of the jack sockets to be connected to the powers supply ground without using the module's 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.

Power connections – Synthesizers.com

The PWR power socket is to be fitted if you are using the module with a Synthesizers.com system. In this case you should not fit the PSU header. The PWR header is a six way 0.1" MTA, but with the pin that is in location 2 removed. In this way location 3 is actually pin 2 on my schematic, location 4 is actually pin 5 and so on.

| Power	Location number	Schematic Pin number
+15V	1	1
Missing Pin	2	
+5V	3	2
Module GND	4	3
-15V	5	4
Not connected	6	5

+5V is not used on this module, so location 3 (pin 2) is not actually connected to anything on the PCB.

If fitting the PWR header, you will also need to link out pins 2 and 3 of the PSU header. This connects the panel ground with the module ground. Simply solder a solid wire hoop to join the two middle pads of PSU together.

Building the Noise/Filter module using the Sock8 board

This is the simplest way of connecting all the sockets to the main board. The Sock8 board should be populated in the way described in our construction guide found on the project webpage. There are only two headers, UPR (for upper) which is six way, and LWR (for lower) which is eight way. Both headers are fitted to the bottom side of the board.

You need to make up two interconnects. The six way one should be made so that it is 80mm long. The eight way should be made to be 130mm.
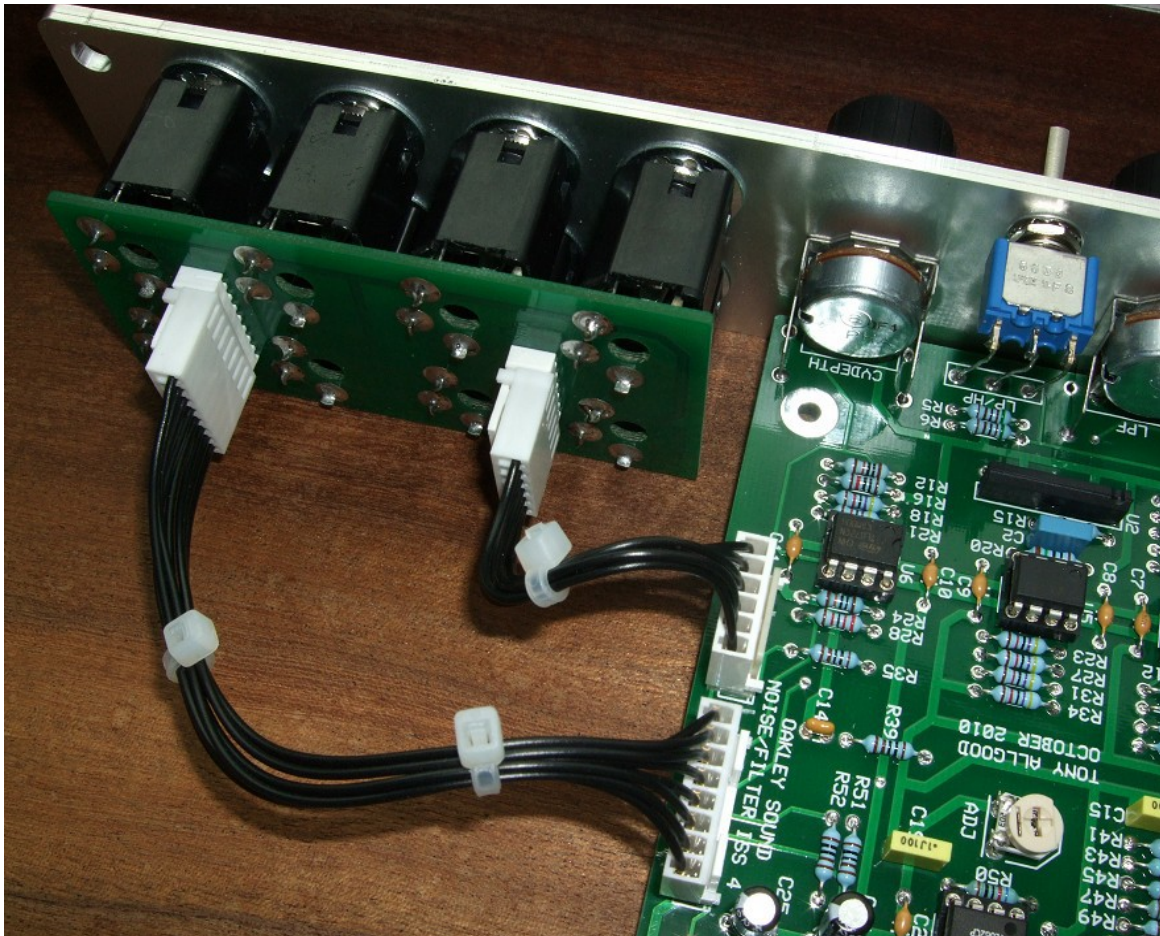
The prototype Noise/Filter module showing the detail of the board to board interconnects. Here I have used the Molex KK 0.1'' system to connect the Sock8 to the main PCB.

Building the Noise/Filter module by wiring the sockets manually

If you have bought Switchcraft 112A sockets you will see that they have three connections. One is the earth or ground tag. One 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) tag. The NC tag is internally connected to the signal tag when a jack is not connected. This connection is automatically broken when you insert a jack.

Once fitted to the front panel the ground tags of each socket can be all connected together with solid wire. I use 0.91mm diameter tinned copper wire for this job. It is nice and stiff, so retains its shape. A single piece of insulated wire can then be used to connect those connected earth tags to pin 1 of LWR. Pin 1 is the square solder pad.

All the other connections are shown in the tables below:

UPPER

Pin	Pad name	Socket Connection	Lug Type
Pin 1	LP-OUT	Connect to LPF OUT	Signal lug
Pin 2		Not connected	
Pin 3		Not connected	
Pin 4	HP-OUT	Connect to HPF OUT	Signal lug
Pin 5	WHITE-OUT	Connect to HPF IN	NC lug
Pin 6	HP-IN	Connect to HPF IN	Signal lug

LOWER

Pin	Pad name	Socket Connection	Lug Type
Pin 1	Panel ground	Connects to all sockets	Ground lugs via wire frame
Pin 2	PINK-OUT	Connect to PINK NOISE	Signal lug
Pin 3	WHITE-OUT	Connect to WHITE NOISE	Signal lug
Pin 4	IR-OUT	Connect to INFRA RED	Signal lug
Pin 5	PINK-OUT	Connect to LPF IN	NC lug
Pin 6	LP-IN	Connect to LPF IN	Signal lug
Pin 7	module ground	Connect to CV IN	NC lug
Pin 8	CV-IN	Connect to CV IN	Signal lug

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

Apply power to the unit making sure you are applying the power correctly. Reversal of the power's polarity will usually destroy all the op-amps in an instant.

If you can monitor current, check that it doesn't exceed 30mA or so from each rail. If you can't check current directly, check with your finger that none of the op-amps or VCAs are getting warm.

You will now need a audio VCO and an LFO of some sort. Some form of audio monitoring will be needed too so you can hear the output of the Noise/Filter module.

Connect the audio output of the VCO to the IN socket of the low pass filter. A sawtooth wave in the bass registers is best. Listen to the audio output from the OUT socket. Adjusting the LPF FREQ pot you should notice the sound getting duller as you turn the pot anti-clockwise. This should sound like simple filtering, which is exactly what it is. At the maximum point the output of the filter should sound more or less the same as the input signal. The minimum point should just let through a dull rumble.

Set up an LFO so that its frequency is at around 1Hz. Now attach the LFO's sawtooth output to the CV IN socket. Set the LPF FREQ pot to its middle position, click the switch to the middle position and rotate the CV DEPTH pot to beyond the central position. The cut-off frequency should now be being modulated with the LFO. Check that the minimum modulation occurs in the middle position of the CV DEPTH pot. You should have a 'dow, dow, dow' sound with the pot in the '+' position, and 'yit, yit, yit' in the '-' position. Click the switch into the HPF position and the modulation should stop.

Now we need to test the high pass filter. Repeat the above section to check whether the high pass filter is working. This time as you increase the HPF FREQ pot the output should get 'fizzier' and lose all of its bass end. Check also the CV input for the high pass filter. Again, check that the modulation is at its minimum when the CV DEPTH pot is in the middle position.

If this is all correct you almost certainly have a working module. However, we can't check the noise section yet because we haven't inserted the noisy transistor.

If you have found a fault, then the usual course of action is to try and find the wrong component. It'll usually be a wrong value resistor, like swapping a 100K resistor for a 100R.

Other things to check are the connections between the socket and main board. Make sure too that your headers have been fitted the right way.

Selecting your transistor and final setting up

As soon as you know the filters are working the next thing you need to do once you have built up your module is to fit the noise transistor. You should have a few BC547, BC548 or BC549 NPN transistors ready. You can try each one in turn into the Q1 position on the PCB and see which one sounds best.

Connect your power supply up to the module, and attach your monitoring amplifier to the white noise socket. Keep the volume levels fairly low, because as you remove and add the transistors it will thump a bit.

For the each transistor 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. For now there is no need to solder it in place. If you don't like the sound of just pushing the transistor into the unsoldered holes in the board you can use a turned pin socket. I buy these as 20 or 32 way SIL sockets and then simply cut them down to size.
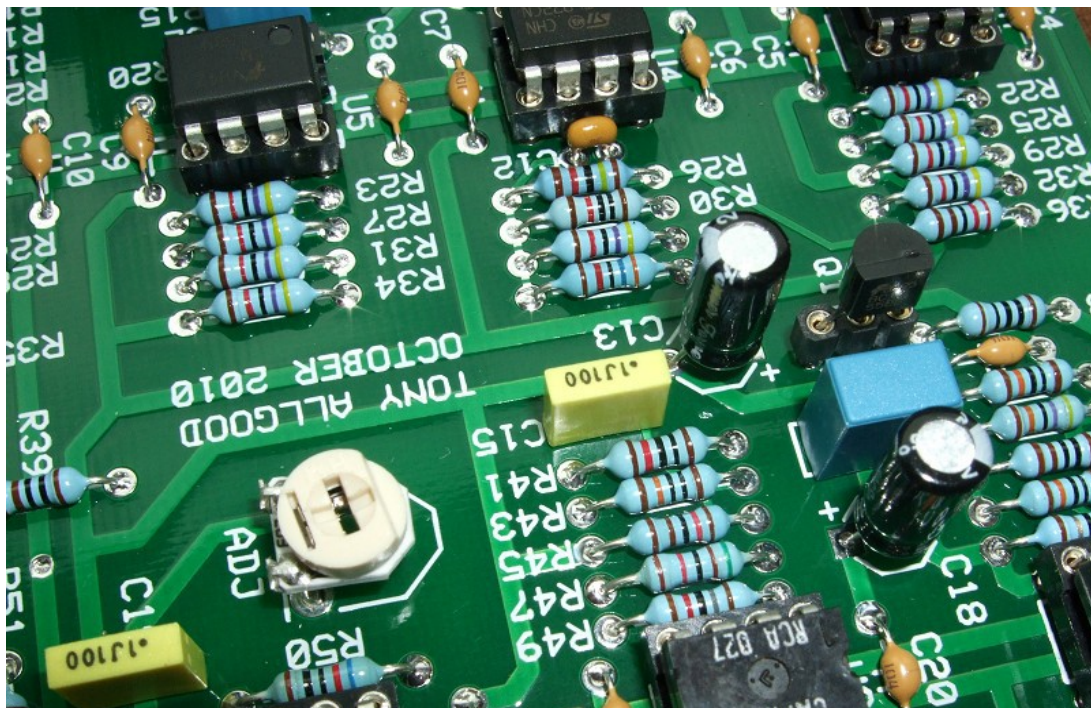
Q1 fitted into a three way section of a SIL turned pin socket.

Sometimes transistors come with the middle leg preformed away from the other two. This is all right, the part will still fit into the board. However, if I get these parts, I tend to 'straighten' the legs out by squashing gently all the three of them flat with a pair of pliers. The flat surface of the pliers' jaws is parallel to the flat side of the transistor.

Once the transistor is inserted the circuit will take time to settle. After a few seconds you should hear the tell tale hiss of a reverse biased transistor junction.

You are listening out for the best quality noise sound. Don't get too concerned with overall loudness as some of the quieter ones can actually sound better. You may need to alter the

LEVEL trimmer so that you get an even tone. If the LEVEL trimmer is set to high then the noise pre-amps may clip and that won't sound very nice either. There's no hard and fast rules here, just enjoy the process. If you have a 'scope then you can monitor what's going on by looking at the white noise output. You'll find that the best noise sources are the ones with an even voltage swing about 0V. Bad sounding devices tend to have more positive voltage spikes. Once you have found a nice sounding transistor, switch off and solder the little chap in.

There are just two trimmers to be set before you are finished. As we have seen the LEVEL trimmer allows you to trim the output of the white noise and also the pink noise. Monitor the white noise with an oscilloscope. Adjust LEVEL until the peak signal is around +/-6V. If you don't have a 'scope, then just adjust it until you get the roughly the same volume as one as your modular's VCOs.

The ADJ control can be set in a variety of ways. What we need to do is to set the average value of the IR output to zero. One way is to use a 'scope and try and get equal swings either side of zero volts. Another way, is to use your Multimix output LEDs. Adjust until you get equal 'green' and 'red' times. Beware, the infra red output is very random and very slow. Be prepared to spend around a minute in waiting for the output to settle to the new ADJ position.

If you find you cannot get the IR output to get away from either supply rail, then please re-adjust the LEVEL trimmer to give a higher level of noise to the IR circuitry. Too little pink noise will mean you haven't got enough signal to flip the comparator U8a.

Final Comments

If you have any problems, 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

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