Oakley Sound Systems

5U Oakley Modular Series

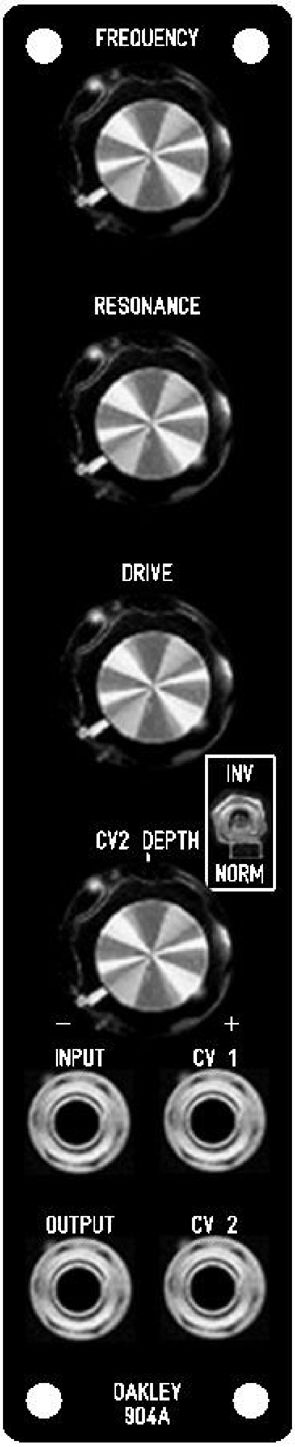
DFL – Discrete Ladder Filter

PCB Issue 1

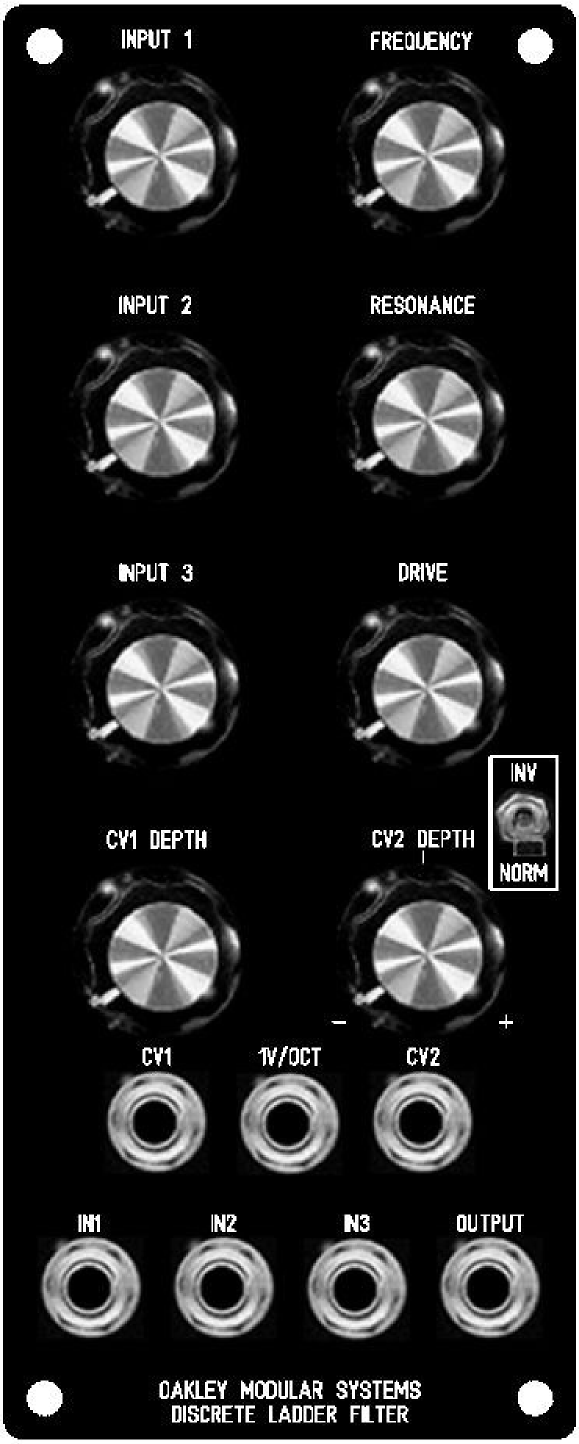
User Manual

V1.0

Tony Allgood
Oakley Sound Systems
CARLISLE
United Kingdom



The Discrete Ladder Filter in the 1U wide 'filter core' format.



The Oakley Discrete Ladder Filter in the 2U wide full version format.

Introduction

This is the User Manual for the Discrete Ladder Filter 5U module from Oakley Sound. This document contains a brief introduction to the module and the calibration procedure.

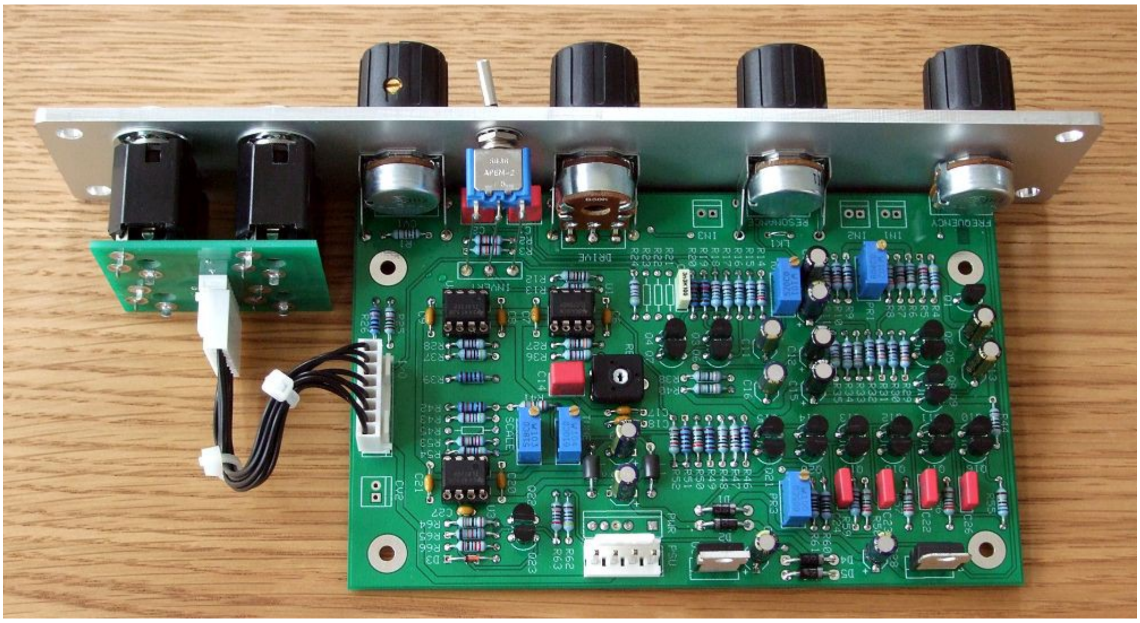
For the latest Project Builder's Guide, which contains a basic introduction to the board, a full parts list for the components needed to populate the boards, and a list of the various interconnections, please visit the main project webpage at:

http://www.oakleysound.com/dlf.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 Oakley Discrete Ladder Filter Module



The prototype Oakley Discrete Ladder Filter as a single width MOTM format module in a natural finish Schaeffer panel.

The Oakley discrete ladder filter (DLF) is my reworking of the classic 1960's low pass filter module using the circuits from the 904A filter and CP3 mixer and combining them into one excellent sounding module. The module also features a drive control which allows you to heavily overdrive the filter circuit without producing a big change in output volume.

The audio signal pathway is almost all discrete components with a dual IC op-amp only providing the input and output signal processing for the drive control. The frequency control circuitry is based around IC op-amps much like my other filter designs and is temperature stabilised.

Prior to entering the filter circuit, the audio signal passes through a clone of the CP3 mixer. This is true even in the 1U wide version of the module which only has the one input socket. The original CP3 module had two outputs, an inverted output and a non inverted output. Apart from the change in phase the two outputs sounded different when the input signal was particularly loud. The Oakley DLF has a front panel switch to select which of the input stage's outputs go on to the filter circuit. The DLF's input stage will start to clip at around +/-4V, but keeping the signal below this will ensure that the filter receives a clean signal.

The discrete low pass filter circuitry is also liable to distort when driven hard but does so in a less aggressive manner than the mixer. The DLF's drive control alters the input signal level so the filter will operate from clean to heavily overdriven. With variations in input level and drive level it is possible to utilise the overdrive characteristics of only the input stage, only the filter, or both for a really heavy sound.

The filter will self oscillate at high resonance. However, like the original module, the filter won't self oscillate below 100Hz. When the filter's cut-off frequency is swept at high resonance this limited resonance at low frequencies gives the filter a powerful sound.

The Filter Core Idea

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

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

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

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

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

Calibration

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

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

The first three trimmers probably should be adjusted on the bench rather than in your modular because getting the voltmeter's probes into the right places is tricky if you can't get full access to the PCB. Allow the board to stabilise by powering it up for at least five minutes before making any of these adjustments.

PR2: This adjusts the offset output voltage from the module's input stage amplifier. Measure the voltage between the left hand end of R16 and metal tab of U4. Adjust PR2 so that the voltage is as close to 0.000V as you can get it.

PR3: This adjusts the offset balance voltage of the ladder's input stage. Measure the voltage between the base pins of Q14 and Q20. The easiest way is probably to put your probes on the left hand ends of R60 and R61. Adjust PR3 to give 0.000V or as close as you can get it.

PR1: This adjusts the offset between the inverting and non inverting outputs of the ladder's differential amplifier. Measure the voltage between the collectors of Q2 and Q5. The easiest way is to put your probes between the positive pin of C5 and the left hand end of R30. Adjust PR1 to give 0.000V or as close as you can get it.

The remaining trimmers can be adjusted in your modular.

RES: The sets the point at which the resonance pot needs to be turned to make the filter self-oscillate. There is no correct place to put this but I tend to like the self-oscillation to start when the resonance pot is at 3 o'clock. Set the frequency pot to its middle position and put the resonance pot to the 3 o'clock position and adjust RES so that the filter just starts to oscillate.

TUNE: This adjusts the filter’s cut-off frequency. Set this so that the filter’s frequency pot covers your chosen range. I would normally place this in the middle position for now, that is 10 turns or so, from one of the end points.

SCALE: This adjusts the scaling (or sensitivity) of the exponential inputs. In a perfect world SCALE should be adjusted so that there is an exactly an octave jump in cut-off frequency when the CV1 input for the 1U module, or the 1V/OCT input for the 2U module, is raised by one volt. However, this is not going to be the case over anything more than a few octaves.

Turn the resonance pot fully up and let the filter oscillate. Turn the drive pot down to maximise the oscillating output. Then connect the CV1 (1U) or 1V/OCT (2U) socket to the 'keyboard CV' out of your midi-CV convertor or analogue keyboard.

Play an A on your keyboard and adjust the front panel's frequency pot so that the filter gives out a sine wave like signal of around 440Hz. Now play the A note one octave higher than you were pressing. By adjusting TUNE and SCALE you should aim to get the higher A to be 880Hz, ie. double 880Hz. However, any change in SCALE will also change the lower note as well as the higher note. So you will have to move back and forth between altering TUNE and SCALE until you get the octave spread you require.

Remember though that the Discrete Ladder Filter will not be able to be made to track perfectly over a very wide range. As such, even if you have achieved a perfect octave spread with your two A notes, you won't be able, for example, to hear a perfect octave spread from two similarly spaced A notes a couple of octaves above that. However, since the action of the Discrete Ladder Filter is, for the most part, to filter and not to act as an oscillator this lack of perfect scaling should not be a problem.

Final Comments

I hope you enjoy using the Oakley Discrete Ladder Filter module.

If you have any problems with the module, an excellent source of support is the Oakley Sound Forum at Muffwiggler.com. Paul Darlow and I are on this group, as well as many other users and builders of Oakley modules.

If you have a comment about this user manual, or have a found a mistake in it, then please do let me know.

Last but not least, can I say a big thank you to all of you who helped and inspired me. Thanks especially to all those nice people on the Synth-diy and Analogue Heaven mailing lists and those at Muffwiggler.com.

Tony Allgood at Oakley Sound

Cumbria, UK