Oakley Sound Systems

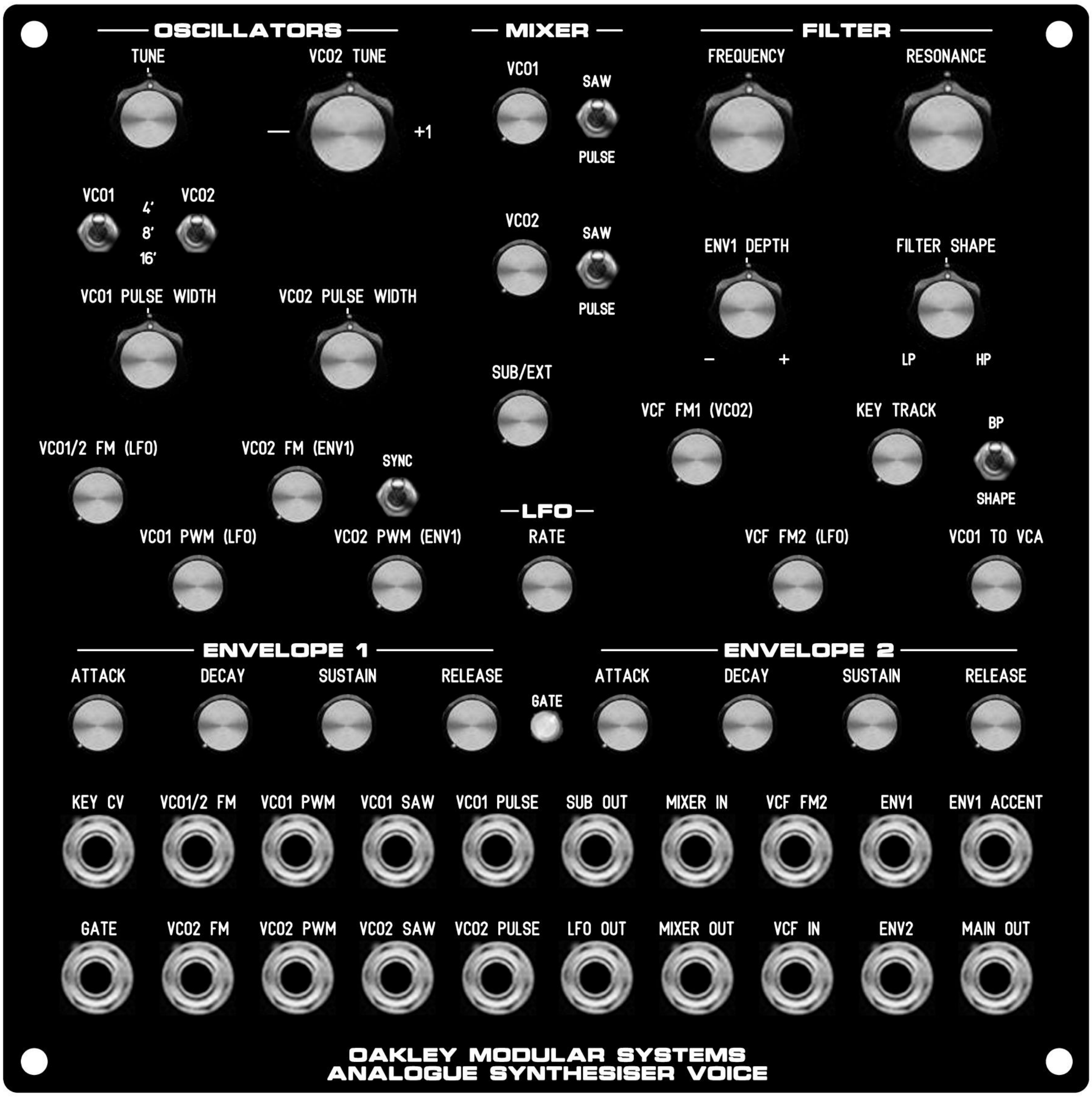
5U Oakley Modular Series

Analogue Synthesiser Voice

User Manual

V1.1

Tony Allgood
Oakley Sound Systems
CARLISLE
United Kingdom



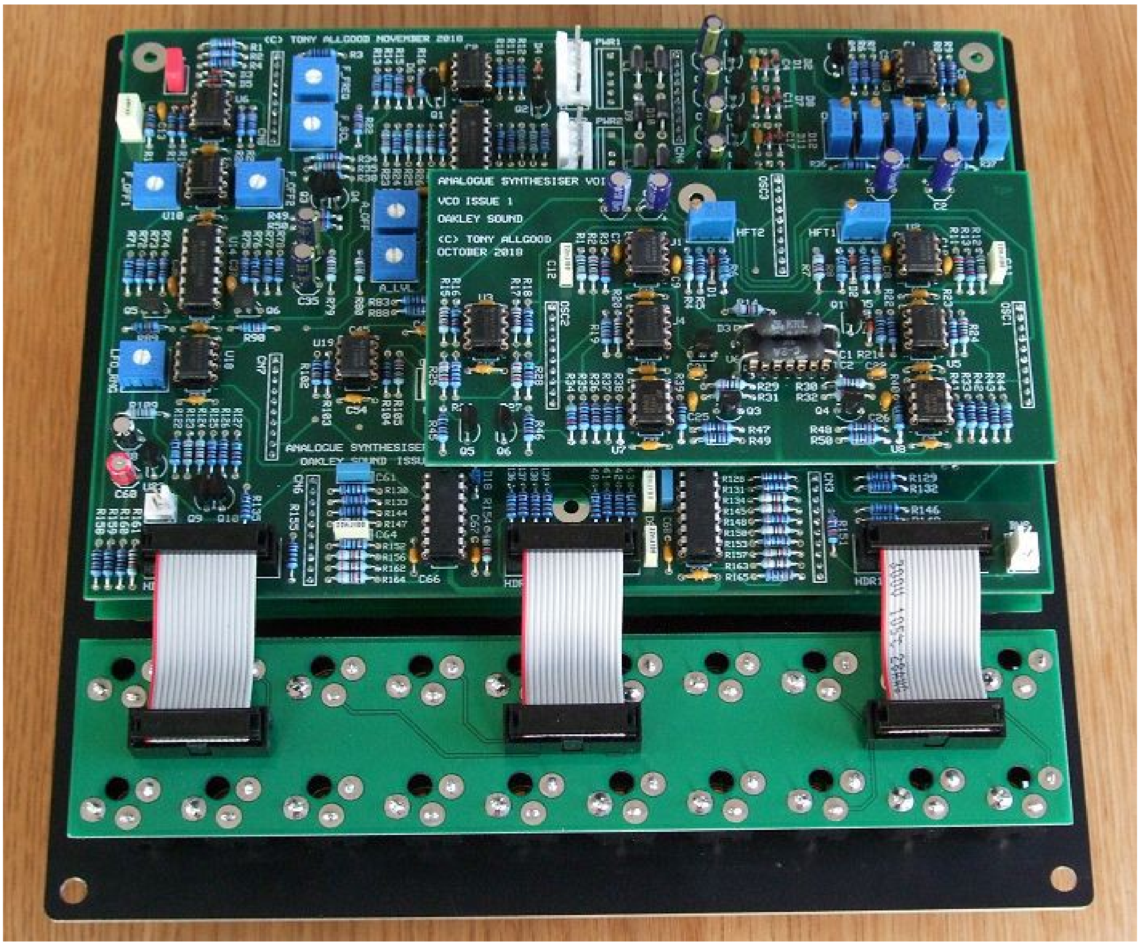
The panel design for the 5U wide MOTM format module.

Introduction

This is the User Manual for the issue 1 Analogue Synthesiser Voice (ASV) module from Oakley Sound. This document contains an introduction to the module, some information on how to make the best use of your module and the calibration procedure.

For the latest Project Builder's Guide, which also contains the parts list for the components needed to populate the boards and helpful notes on how to build the module, please visit the main project webpage at:

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



The prototype issue 1 ASV fitted to a MOTM format panel.

The Oakley Analogue Synthesiser Voice

The Oakley Analogue Synthesiser Voice, or ASV, is a complete analogue synthesiser in a 5U wide, or four MU width, module for the MOTM and MU formats. It consists of the traditional synthesiser architecture of two oscillators feeding a single filter, via a mixer, and then to an amplifier. Two identical four stage envelope generators (ENV1 & ENV2), and one sine wave low frequency oscillator (LFO) act as modulation sources to control the sound in a dynamically interesting way.

Each voltage controlled oscillator (VCO) is based around a classic sounding sawtooth core. VCO2's frequency can be hard synchronised to VCO1 by a front panel switch. Both sawtooth and variable width pulse waves are generated. These are both available from the socket field but only one from each VCO may be selected in the audio mixer that feeds the filter. An additional triangle wave from VCO1 can mixed with the audio signal from the filter prior to the final amplifier.

The AVS also has a sub-oscillator. This produces a triangle wave output that is half the frequency of VCO 1, that is, one octave below the pitch of VCO1. By default this is sent to the third level control on the mixer although this can be overridden by inserting a jack plug into the 'mixer in' socket. The sub-oscillator's output is also available on the socket field.

The pitch of both VCOs may be controlled by a master tune control on the front panel. Two octave switches independently control the base pitch of each of the VCOs. Each switch covers a range of three octaves. The pitch of VCO2 may also be offset from VCO 1 by the VCO 2 Tune control. The range of this control covers just over one octave from unison at the 9 o'clock position to an octave above at around 3 o'clock.

Pitch may also controlled by external control voltages (CVs) and internal sources such as the LFO and ENV1. Front panel pots control the depth of this modulation.

The pulse wave output from each VCO can be manually controlled by the pulse width controls. At their middle settings both pots will produce a 50% pulse wave, ie. a square wave. The pulse width of both VCOs can be dynamically altered with external CVs, or internal sources. Both VCOs use special circuitry to make the their average voltage output equal to zero. This reduces audible thumping when the pulse width is modulated quickly.

The audio mixer normally sends its output direct to the filter although this can be broken by using the 'filter in' socket. In this way it is possible to process the mixer's output by taking a signal from the 'mixer out' socket, feeding it through another module, and then back into the AVS via the 'filter in' socket for final processing.

The voltage controlled filter (VCF) is a state variable design based on the one in a classic synthesiser first seen in the early 1970s. It has two basic modes, shape and band pass (BP). In shape mode the output response is controlled by the Shape control. Fully counter clockwise the response from the filter is -12dB/octave low pass, and fully clockwise it is -12dB/octave high pass. Between the two extremes are a variety of responses and half way around, ie. the control is pointing upward, you get a notch filter which attenuates a single frequency. The filter has a separate resonance control where increasing amounts will accentuate the frequencies heard around the filter's cut off point.

The filter's cut off frequency is controlled by a front panel pot and several modulation sources including ENV1, LFO, and VCO2. The 'ENV1 accent' socket allows an external CV to additionally control the level of ENV1 reaching the filter.

The voltage controlled amplifier (VCA) shapes the volume of the signal leaving the filter. It is solely controlled by ENV2. Both ENV1 and ENV2 are triggered by a single gate signal present at the gate input socket. If no jack plug is inserted into the gate socket then both envelopes are held in their sustain phase, as if they are permanently gated on. The gate LED will light green when gate is active.

The ASV's LFO is a wide range oscillator that will go from very slow to around 120Hz. It produces a sine wave output which is routed internally to certain functions and is available on its own output socket.

The module is built from four printed circuit boards populated by through hole components. The four boards are: The pot/switch board which holds the front panel potentiometers and switches, and some supporting circuitry. The main board which has the majority of electronics such as the filter, envelopes, LFO, oscillator pitch control and sub-oscillator. The socket board which supports all twenty 1/4” jack sockets and is connected to the main board via three detachable ribbon cables. And finally the VCO board which as its name implies has the core of the two voltage controlled oscillators and pulse generators.

Two 0.1” (2.54mm) Molex KK headers are fitted to the main board. The two way OUT header carries a copy of the ASV's main audio output. The three way BUS header connects to the Oakley Bus system which allows normalisation of Key CV and Gate signals behind the panel. See later for more details on the Oakley Bus.

The module requires +/-15V at around +/-180mA. MOTM/Oakley or MU power headers can be fitted to the board. MOTM/Oakley power leads are to be preferred as they are more rugged and can carry more current without problems. Whichever you choose two power leads are required for normal operation.

Failure to connect one or the other connector will not result in damage to the ASV or your power supply. However, the ASV will not work correctly. The lower headers, PWR2 and PSU2, power the LFO and VCO2. The upper headers, PWR1 and PSU1, power the rest of the ASV. Sharing the power over two leads reduces the chances of unwanted interaction between the two VCOs and helps provide a good solid 0V connection.

Frequently Asked Questions

Will this fit into a Eurorack? The ASV is a 5U format module and as such is too big to fit into a Eurorack case.

Can it be powered from +/-12V? No. Although the ASV has internal +/-10V references, it also uses +/-12V to power the two VCOs. It must therefore have more than +/-14V to work correctly.

Is it a SEM clone? No. The only thing that is similar to the SEM, other than the fact it is a two oscillator subtractive synthesiser, is the filter topology. All the rest of the circuitry is very different.

If I have four of them can I use the ASV as a polyphonic synthesiser? Yes. But be aware that to change the sound you will need to change the sound similarly on each of the ASVs. There is no way to change all ASVs with one set of controls. You will also need a midi-CV convertor that features polyphonic note assignment. Mutable Instrument's Yarns is one such device.

The Front Panel Controls

TUNE

The control pot adjusts the pitch of both voltage controlled oscillators (VCOs). The range of this control is +/- 1.2 semitones. The ASV is normally calibrated to be in tune when this control is in its middle position.

The ASV is fully analogue and its VCOs will drift a little in pitch over time and with ambient temperature. The unit takes ten minutes or so to fully stabilise after powering up.

VCO2 TUNE

Controls only the pitch of VCO2. The range of this control covers just over one octave from being matched with VCO1 at the 9 o'clock position to an octave above at around 3 o'clock.

VCO1 and VCO2 Octave Switches

Each VCO has its own octave switch which controls the pitch of that VCO. 16' produces the lowest pitch, 8' is one octave higher, and 4' is one octave higher than that. The actual number of the 'footage' may not necessarily be directly equivalent to other VCOs you may have. In the ASV they simply represent relative pitch in octaves.

VCO1 PULSE WIDTH

This control varies the pulse width of VCO1's pulse output. In the middle position the pulse output is a square wave, that is, the output spends 50% of its time 'up' at +5V and the other 50% 'down' at -5V. A square wave has a hollow sound with no even harmonics. Changing the pulse width varies the ratio between the time spent in the up position compared with the down and varies the harmonic structure of the sound. There is little appreciable difference in sound between pulse waves at equivalent sides of the middle position. That is, a 60:40 (60%) pulse wave sounds much the same as 40:60 (40%). However, when used as a modulating source, in other words, when the pulse output is used to alter another module's parameters, the actual pulse width is very important.

At either of the extremes of the pulse width control there is no sound heard. This is because the ratio has exceeded either 0% or 100%. At these extremes the pulse has become so thin, or so fat, the output is now constant and not oscillating.

Between 0% and 100% pulse wave output is always 10V 'peak to peak', but the actual peak levels vary with pulse width. The ASV's VCOs use an interesting technique to maintain the average voltage over one complete waveform cycle to zero volts. The circuit essentially adds a voltage offset to the pulse output to compensate for the non zero average voltage for any pulse wave that isn't a square wave. Ordinarily, one would see average voltage varying from +5V to -5V as the pulse width is swept from one end to the other. In the ASV, this average voltage is kept at zero. This means that for narrow pulses, say at 5%, you now have a wave that goes from just below 0V (down) for 95% of the time, to just below +10V (up) for 5% of the time. For square waves, you have the usual +5V up and -5V down. For wide pulses, you have a wave that is just below 0V (up) and just below -10V (down). The benefits of doing this is that fast changes in pulse width no longer adds unpleasant artefacts to the audio output. ie. fast EG sweeps of PW will not cause thumping sounds.

VCO2 PULSE WIDTH

The control works in the same way as above but on VCO2. It should be noted that in conjunction with the synchronisation feature changing the pulse width can create many new textures.

VCO1/2 FM (LFO)

This control alters the depth of frequency modulation (FM) of both oscillators. That is, it determines how much the applied control voltage at the VCO1/2 FM input socket has an effect on the pitch of both VCOs. Turning the control to full with make the VCOs very sensitive to the CV connected to the VCO1/2 FM socket. The maximum sensitivity is approximately 1V/octave.

When no plug is inserted into the VCO1/2 FM socket then the ASV's own sine wave LFO will be used as the modulation source. The control may then be used to introduce vibrato type effects.

VCO1 PWM (LFO)

This control alters the depth of pulse width modulation (PWM) that is applied to VCO1's pulse output. That is, it determines how much the applied control voltage at the VCO1 PWM input socket has an effect on the pulse width of VCO1. It acts in conjunction with the setting of the main pulse width control.

When no plug is inserted into the VCO1 PWM socket then the ASV's own sine wave LFO will be used as the modulation source. The control may then be used to introduce chorus type effects.

With the pulse width control set to its middle position, and the VCO1 PWM control set to maximum, a voltage greater than +5V, or -5V, at the VCO1 PWM socket will mean that no sound is heard from the pulse output as the pulses have become too thin or too thick. Allowing the pulse width ratio to exceed 0% or 100% in this way will not damage the unit and can be used to create interesting gating effects.

VCO2 FM (ENV1)

This control alters the depth of frequency modulation (FM) of just VCO2. That is, it determines how much the applied control voltage at the VCO2 FM input socket has an effect on the pitch of both VCOs. Turning the control to full with make VCO2 very sensitive to the CV connected to the VCO2 FM socket. The maximum sensitivity is approximately 1V/octave.

When no plug is inserted into the VCO2 FM socket then the ASV's Envelope 1 (ENV1) will be used as the modulation source. The control may then be used to introduce pitch sweeps, and if synchronisation is engaged, then dynamically varying sounds can be created.

SYNC

The Sync switch turns on the synchronisation feature where VCO2's output waveforms are reset to a specific value each time VCO1 starts a new cycle. In other words VCO2's fundamental frequency is locked to the free running VCO1. VCO2 has become a slave to VCO1. VCO2 will still oscillate as before, and is still controlled by the voltage on the KEY CV input, but each time VCO1 begins a new cycle VCO2 will be reset. When the two oscillators are at a similar frequency engaging sync will simply lock VCO2 hard to the same frequency as VCO1. Any rich rolling beating effects will no longer be heard. However, as VCO2's pitch is increased significantly above that of VCO1 the output waveform of VCO2 becomes more complex and interesting timbres can be created.

Using the VCO2 FM control to allow ENV1 to sweep VCO2's frequency can produce the classic 'Laser Harp' sync sweep sound.

VCO2 PWM (ENV1)

This control alters the depth of pulse width modulation (PWM) that is applied to VCO2's pulse output. That is, it determines how much the applied control voltage at the VCO2 PWM input socket has an effect on the pulse width of VCO2. It acts in conjunction with the setting of the main pulse width control.

When no plug is inserted into the VCO2 PWM socket then the ASV's Envelope 1 will be used as the modulation source to produce dynamically varying timbres.

With the pulse width control set to its middle position, and the VCO2 PWM control set to maximum, a voltage greater than +5V, or -5V, at the VCO2 PWM socket will mean that no sound is heard from the pulse output as the pulses have become too thin or too thick. Allowing the pulse width ratio to exceed 0% or 100% in this way will not damage the unit and can be used to create interesting gating effects.

ENVELOPE 1 and ENVELOPE 2

Both envelope generators (EGs) function in the same way and differ only in their destinations. Envelope 1 natively controls the pitch of VCO2, the pulse width of VCO2 and the cut off frequency of the filter. Envelope 2 controls only the final VCA which controls the final output volume of the ASV. Both envelope generators have their outputs available on the socket field as ENV1 and ENV2. These output levels vary from 0V to approximately +5V.

Both envelope generators are triggered from the GATE socket. Any signal greater than +2V will be enough to activate the envelopes. The GATE LED will light when an active gate signal is present. With no plug inserted into the Gate socket the ASV will be in drone mode and the gate is permanently active, so both envelope generators will be in their sustain phase.

The EGs have four distinct phases; attack, decay, sustain and release. When the attack phase is initiated by an active gate signal, the output will rise to +5V at a time determined by the Attack control. The decay phase then starts and the output voltage will then fall, at the rate set by the Decay control, to the level set by the Sustain pot. This sustain level can be any value from 0V to +5V. The output will remain at this level for as long as the gate is high. But as soon as the gate is removed the release stage is initiated. This causes the output to fall at a rate determined by the Release control.

Attack, Decay and Release are controlling the times taken to rise and fall. Sustain controls a level.

If the gate input falls below +2V then the envelopes will go into the release phase irrespective of whether the attack phase has been completed. EGs that always complete their attack phase are called 'one shot' envelopes. They have their place but the ASV's EGs behave in the traditional manner to allow for more expressive playing. Furthermore, the attack phase always starts at a voltage determined by the envelope's current output voltage. For example, if the EG has finished completely and its output is resting at 0V then the attack phase will start from 0V. If however the EG is still in its release phase and the voltage is, say, at +2V, then the attack will start from +2V. This is the natural way of things and has been so since EGs were first invented. An EG that always starts from 0V irrespective of what it was doing just before the new gate signal has arrived is called a 'return to zero' (RTZ) EG. They do have their uses but RTZ behaviour should never be the default behaviour of any EG.

The ASV's envelopes have the following specifications:

Minimum attack time is around 1.8ms.
Minimum decay and release time is around 3.5ms.

Maximum attack time is around 25s.
Maximum decay and release time is around 40s.

MIXER

The mixer has three level controls, VCO1, VCO2 and SUB/EXT. Each one controls the signal levels that will appear on the output of the mixer. The output of the mixer is present at the MIXER OUT socket and will be automatically sent to the ASV's filter if no plug is inserted into the VCF IN socket.

The two VCO level controls have next to them a two position switch which selects whether the sawtooth or the pulse output is to be used.

The SUB/EXT level control adjusts the signal level of the sub-oscillator circuitry. The sub-oscillator is actually a complex waveshaper which derives its output from VCO1. It is locked at exactly one half of the frequency of the VCO1, that is, one octave lower. The output is a reasonable approximation of a triangle wave and as such has only a few obvious overtones. This makes it ideal for adding sonic power to the main VCO outputs without excessive complexity. The sub-oscillator's output is available from its own socket, SUB OUT. The output signal level at the socket is 12V peak to peak.

Alternatively if a jack plug is inserted into the MIXER IN socket, the SUB/EXT control now adjusts the level of the external signal that is present at MIXER IN.

The gain of the mixer is around -10dB. That is a single 10V peak to peak signal will be reduced to the around 3.3V peak to peak at the mixer's output even with its level control at full. This gain drop is significant but it allows the mixer to sum together all three of its inputs at full without the output clipping and distorting.

Although it is unlikely to make any difference it should be noted that the mixer is also inverting. That is, positively going input voltages are turned into negatively going output voltages – a sawtooth waveform is thus converted to a ramp waveform. However, the ASV's output stage is also inverting so by the time the signal comes out from the ASV's main output the waveforms are the right way around again.

FILTER

The ASV features a voltage controlled state variable filter. With a filter we can cut and boost certain parts of the audio spectrum. The **FREQUENCY** control manually adjusts the cut-off frequency, Fc, and therefore the actual part of the audio spectrum which is to be boosted or cut. Fc can also be controlled via control voltages (CVs) both internal to the ASV and external via the socket VCF FM2.

The behaviour of the filter is also affected by the **RESONANCE** and **FILTER SHAPE** controls. The FILTER SHAPE control allows the filter's response to be swept from low pass (LP) to high pass (HP) through 'notch' in the middle.

Low pass (LP): In this mode the filter will pass all frequencies below Fc. Above this frequency, the output amplitude or level of the filter will drop as the audio input frequency is increased. The ASV's filter has a roll off of -12dB/octave.

By increasing the RESONANCE control the narrow band of frequencies around Fc will be emphasised. This creates a more artificial or electronic sound.

Low pass filtering is the most common form of active filtering in most analogue synthesisers, and generally the most useful.

High pass (HP): In this mode the filter will pass all frequencies above the cut-off frequency, Fc. Below this frequency, the output amplitude or level of the filter will drop as the audio input frequency is decreased. Again, the rate at which this output rolls off is -12dB/octave.

The high pass filter is very useful in creating thin sounds. This is because most input waveforms are rich in lower frequencies, and by removing the low partials of the sound, you tend to end up with just the fizzy part of the sound.

Notch: This is not often found in music synthesisers but is a useful addition to the ASV's musical armoury. The notch filter is in essence the opposite of a band pass filter. It allows all frequencies to pass, except for a very narrow band around the centre frequency. Sometimes this filter shape is called 'band stop'.

Sweeping Fc in notch mode produces a type of effect similar, but weaker, to that of a phaser like the Oakley Equinoxe.

Just below the FILTER SHAPE control is a switch that selects between the audio output from the FILTER SHAPE control and a fourth type of filter response, band pass.

Band pass (BP): This will pass a band of audio frequencies centred around Fc. All other frequencies are attenuated. The roll-off on either side of the centre frequency is -6dB/octave.

Turning up the resonance control will effectively narrow the band of frequencies passed, making the filter more selective. This is a very useful response and results in powerful filtering effects. It is more drastic than the low pass filter since it effects the audio on either side of the cut-off frequency.

The output of the filter, irrespective of the mode used, is passed directly onto the voltage controlled amplifier (VCA) circuit. The output of the VCA connects directly to the MAIN OUT socket. The VCA is solely controlled by ENV2 which shapes the volume of the audio output.

ENV1 DEPTH

This control is a bipolar attenuator, sometimes called a reversible attenuator. It allows the user to adjust the amount of control ENV1 has over the cut-off frequency, Fc, of the filter. In its middle and off position, ENV1 has no control over Fc. Fully clockwise, ENV1 can sweep the filter over most of its operating range with the output of ENV1 adding to the value set by the FREQUENCY control. Fully anti-clockwise ENV1 can again sweep the filter over a wide range but this time subtracts from the value set by the FREQUENCY control.

The depth of ENV1 sweep can also be controlled by an external control voltage (CV) inserted into the ENV1 ACCENT socket. A +5V CV here will make the ASV behave as normal but anything else will either increase the sweep depth (above +5V) or decrease the sweep (below +5V). If the velocity CV from a midi-CV convertor is connected to ENV1 ACCENT simple velocity sensitivity may be obtained.

VCF FM1 (VCO2)

This control adjust the levels of frequency modulation applied to the cut-off frequency of the filter from the output of VCO2. The waveform selected in the mixer stage is the opposite of what is applied to modulate the filter. If sawtooth is selected from the mixer then the filter will be modulated by the pulse wave, and vice versa. This produces more sonically interesting results than modulating with the same waveform that is running through the filter. Extreme modulation depths with high resonance settings can produce some very harsh results. Unlike VCF FM2 there is no related input socket for VCF FM1.

KEY TRACK

This control adjusts the amount of effect that the control voltage applied to the KEY CV socket has on the cut-off frequency (Fc) of the filter. At KEY TRACK's maximum setting Fc will match the 1V/octave sensitivity of the VCOs, and in this way, maintain the harmonic relationship across all notes of the keyboard. In practice this tends to lead to overly bright high notes and for more natural sounds a lower setting is to be preferred.

VCF FM2 (LFO)

This control alters the depth of frequency modulation (FM) of the filter's cut-off frequency (Fc). That is, it determines how much the applied control voltage at the VCF FM2 input socket has an effect on Fc. Turning the control to full with make the VCF very sensitive to the CV connected to the VCF FM2 socket. The maximum sensitivity is approximately 0.5V/octave.

When no plug is inserted into the VCF FM2 socket then the ASV's own sine wave LFO will be used as the modulation source. The control may then be used to introduce tremolo or wah-wah type effects.

VCO1 TO VCA

This control adjusts the level of VCO1's triangle wave output that is sent direct to the VCA's audio input. This signal bypasses both the mixer and the filter sections of the ASV and as such is not affected by the settings of either. The output of this control is added to the output of the filter. It therefore offers a way of reinforcing the fundamental frequency of VCO1 even if the filter has filtered out that part of the signal. It is a very powerful feature and not often seen on fixed architecture synthesisers.

More about the Oakley Bus and Module Normalisation

The Oakley Bus, previously known as the Oakley Buss, is a three way connector found on various Oakley modules. Pin 1 carries the keyboard control voltage (KCV) input for note pitch control, and pin 3 carries gate input for note on and note off. The ASV module can connect to the Oakley bus and receive both gate, for its two envelope generators, and KCV for its two audio oscillators. Pin 2 is connected to 0V on the module and acts only as a 'fire break' between the gate and CV lines preventing any crosstalk between the two.

The Dizzy PCB, our main form distributing power around our modular systems, includes up to eight three way Oakley Bus headers alongside the much larger power headers.

The word bus is perhaps a little grand for something that has just two control lines and a single ground. However, it still adheres to the principle of a common set of conductors that is available to all modules.

Normalising is the process by which some signal paths are already made for you. In other words no patch leads are needed to make those connections; they are connected internally either within the module or between different modules but behind the faceplates. However, normalising can always be overridden by the user. The name itself comes from the use of normalised connections on sockets. When a socket does not have a jack inserted it is in its normal position. There is often a connection between the signal lug of the socket and an extra contact called the NC (normally closed) lug. It is this third lug on the socket that is used for the normalisation. Inserting a jack plug will break the connection between the NC and the signal lug.

To help us understand where normalisation is useful consider a VCO with a 1V/octave, or Key CV, socket on its front panel. To connect KCV to this socket one would ordinarily need a patch lead. But imagine a system where you have four VCOs and two VCFs that all need the same KCV signal. It can take many patch leads to do this; seven if you have a large multiple panel. Now suppose that the NC lug of every 1V/octave socket is connected to a common KCV bus. All six modules can now be driven without the need for those seven patch leads. This saves you leads, time, and also gives you a better working environment because you don't have to fight your way through a tangle of leads to get to the module's knobs. Inserting a jack into any one of those sockets would disconnect it from the KCV bus, so you still have complete modularity.

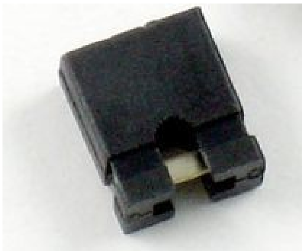
The Oakley VCO, VCO Controller, midiDAC, ADSR, VRG and ASV have the three pin headers on the main or socket boards ready for easy direct connection to the Dizzy.

The Oakley ASV has its Oakley Bus input header near the bottom right of the main board. Pin 1 of this header is connected to the NC lug of the Key CV input socket. So any voltage on pin 1 of the Oakley Bus is automatically connected to the module's Key CV input unless as jack is inserted into that socket. Likewise pin 3 is connected to the NC lug of the Gate socket. So any gate signal present on pin 3 of the Oakley Bus will trigger both envelopes unless there is a plug inserted into the gate socket.

To connect to the Bus you will need to make, or purchase, an assembly that has a three way 0.1" Molex KK or MTA100 socket at each end. There should be two wires of suitable length connecting the two ends. One wire for pin 1, and one wire for pin 3. It is very important that the middle pins are not connected together. The length of wire used should be long enough to allow the module to taken out of its case and disconnected. Because the cable is not screened the wires should not be longer than necessary and anything over 1m is probably too long.

It should be noted that the ASV features a 'drone mode' so if neither the Oakley Bus is being used, nor the front panel's gate socket, then both envelope generators will be in sustain mode.

If the ASV's Bus header is not being used to connect to an Oakley Bus then it can either be left as it is, or a 0.1” (2.54mm) jumper be put across pins 1 and 2. Pin 1 is the upper pin, and pin 2 the middle one. The jumper then connects pin 1 to 0V and prevents any stray interference from reaching the Key CV socket when not the socket is not in use. Pin 3 is left unconnected in this instance.

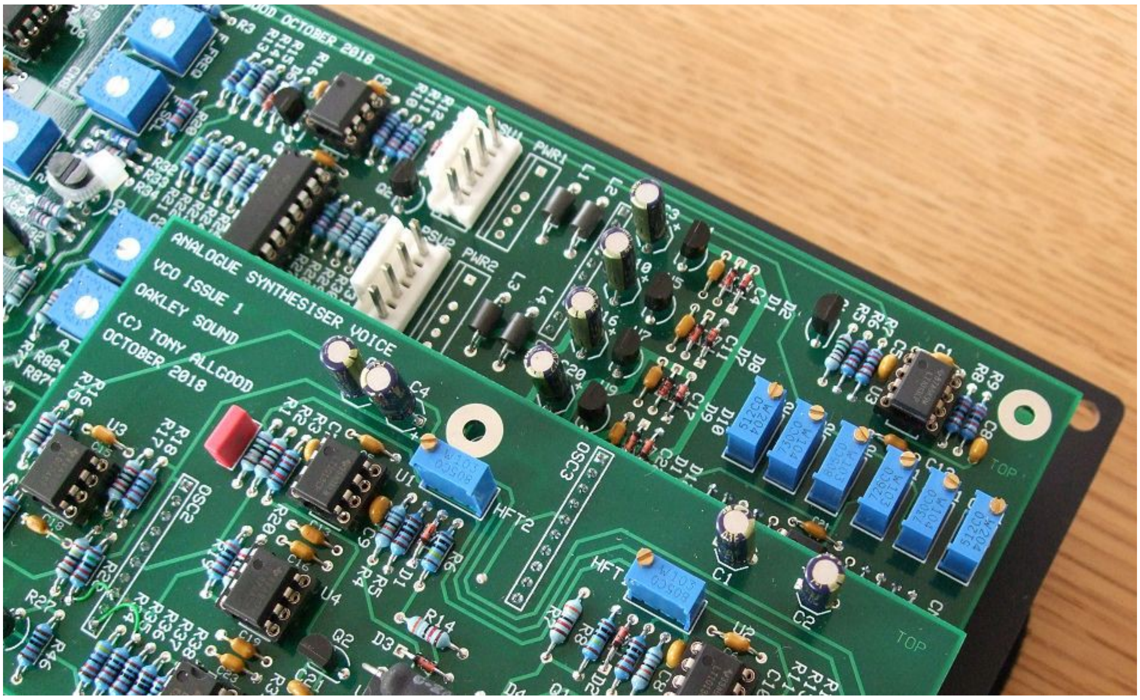


A 0.1” jumper that can be used to short pins 1 and 2 together on the ASV's unused Oakley Bus header.

One final point, the ASV's drone mode works by gently pulling Oakley Bus pin 3 line to +15V. It does this via a 220K resistor connected to the +15V power supply. 220K is a very high resistance and will only supply a very small and very safe current into the Oakley Bus's gate signal. However, if you haven't connected the gate bus to a gate output, for example, from a midi to CV convertor or keyboard, then that 220K resistor will try and pull the whole gate bus high. This may mean any other connected ADSRs or VRGs will appear to be gated on if the 220K resistor can supply enough current to them. This will not damage anything, and indeed may well be useful, but it may surprise you at first.

Note: I have previously used the term 'buss' instead of 'bus'. A bus to me was a form of public transport, and buss was an distributed electrical connection. It seems I was not alone, you'll find various audio sites talking about the 'mix buss'. But from where the extra 's' came nobody seems to know. Perhaps it was a confusion between Bussman the fuse manufacturers. Whatever the history, the use of the word 'buss' is probably wrong. So for now I'll try to stick with bus. Although the plural, buses, looks wrong to me...

Calibration



The ASV has two types of trimmer, the multiturn ones seen here on the right and centre, and the single turn ones seen in the top left.

Although you can adjust the trimmers with a small blade screwdriver, Vishay, Bourns and others make special trimmer adjusters, which are easier to use and less likely to damage the trimmers.

Power up the module and make sure it has been powered up for at least twenty minutes prior to calibration. It is a good idea to have the room temperature close to what it would normally be when playing your modular.

Voltages should always be measured with respect to 0V or module ground. The easiest reference point for the black lead of your voltmeter is any one of the ground lugs of the sockets, ie. the leftmost pin of any socket when viewed from the back.

Set the filter Frequency and ENV1 Depth pot to their middle positions. Set all three mixer level pots, Resonance, VCF FM1, VCF FM2 and Key Track to their minimum positions. These three trimmers should be adjusted in the following order:

F_FREQ: This sets the median operating frequency of the filter. Set this to its middle position for now. We will adjust this properly once the VCOs have been tuned.

F_OFF1: Adjust so that pin 7 of U10 is as close to 0.00V as you can make it.

F_OFF2: Adjust so that pin 1 of U10 is as close to 0.00V as you can make it.

LFO_RNG: Set the LFO Rate control to its maximum. Connect the LFO OUT socket to a digital tuner or a frequency counter. Adjust the LFO_RNG trimmer so that the output frequency is 120Hz +/-5Hz. If you are using a tuner this will equate to the note B2 (where A4 = 440Hz). The LFO frequency will waver around a little so this does not need to be done accurately.

A_OFF: This controls the offset error in the VCA input stage. It should be set so that the control voltage from ENV2 breaks through to the audio output as little as possible.

Set the ENV2 pots as follows, attack, decay and release to their minimums, and the sustain pot to its maximum. Connect a patch lead from the LFO OUT socket to the GATE socket. Set the LFO Rate to its maximum. Make sure the ENV Depth is set to its middle position and all other filter controls are set to their minimum. Ensure also that VCO1 to VCA is set to minimum. Connect the ASV's audio output to your monitoring system. You should hear the LFO breaking through to the audio output as a low hum or buzz.

Adjust A_OFF so that the buzz or hum is minimised. You will not be able to silence it but the optimum position of A_OFF is when the sound heard is as unobtrusive as it can be.

VCO Tuning

All the VCO adjustments are done with multiturn trimmers. VCO1 and VCO2 are to be treated similarly but VCO1 should be trimmed first if the unit has not been calibrated before.

In the mixer section on the front panel turn VCO1 to its maximum and select 'saw'. The other two mixer pots should be at their minimums. Turn the filter Frequency and ENV2's Sustain to their maximums, the ENV1 Depth to its middle position, and all other filter pots to their minimums. You will need a digital frequency counter, or my favourite, a guitar/chromatic tuner or tuner plug-in. It is essential to be able to hear what it happening as well. Connect your audio system and tuner to the main output of the ASV. The ASV will be in drone mode with nothing inserted into the Gate socket so you should be able to now hear VCO1.

SCL1: This is the scale trimmer for VCO1. Use this to generate a perfect 1V/octave scaling.

Set VCO1 to 4' using the switch on the front panel. Set the Tune pot to its central position.

Plug your midi-CV convertor or 1V/octave keyboard into the KEY CV input of the ASV. Play a very high note. To adjust SCL1 properly we will need to negate the actions of the high frequency tracking compensation circuitry. Adjust HFT1 on the VCO board so that the pitch drops as far as it will go.

Now set the VCO to 8' and play a lowish note on the keyboard, then play two octaves higher. Adjust SCL1 until the interval is **exactly** two octaves. I normally try to work between the two As of 110Hz and 440Hz.

Note we are only setting the interval and not the actual frequency. It does not have to be a perfect A when A is being pressed on the keyboard. It could be an F or whatever. The important thing is that we are setting the musical gap between the notes. If you do need to alter the actual pitch of the VCO to help you tune then use the TUN1 trimmer.

For any interval, if you find the higher note is flat, then turn the SCL1 trimmer to make it flatter still. This actually reduces the range between the two notes. Conversely, if you find your interval is greater than an octave, turn the trimmer to make the top note even higher. I always adjust SCL on the high note of any interval, and only adjust the TUN1 trimmer on the lower.

This will probably require some patience and plenty of twiddling. But you will get there. Once you get the hang of it, its easy. I can do it in about one minute but I've had a lot of practice.

You should be able to get it as accurate as +/-1 cent.

Now leave it on for a further 20 minutes and then check the scaling again. Re-adjust if necessary. You can of course move onto to the rest of the calibration while VCO1 is bedding in.

HFT1: This is the high frequency tracking trimmer and it compensates for the slight flattening of pitch at when running the VCO at high frequencies. If you don't go above 3kHz that often there is a good chance you won't even have to touch this one. Like the SCL trimmer it will have a small knock on effect on the absolute pitch of the SVCO lower down too.

If you only have a small keyboard use the keyboard's octave transpose setting to get the VCO playing a really high note. I normally work between the two As of 3520Hz and 7040Hz, although you can work an octave below that if you wish. Once again, you can ignore the actual pitch, it's the interval we are wanting to get right. Once you have set up the perfect octave at these frequencies, then check down at the lower end that everything is still responding to 1V/octave.

Remember, if you have skimped on the SCL1 trimming, no amount of tweaking of HFT1 will get it to play in tune.

TUN1: This is the tune trimmer and it sets the range over which VCO1 acts. You are trying to set this so that VCO1 in the ASV will behave in the same way as your other VCOs or other electronic musical instruments.

As far as I am aware there is no standard amongst modular systems that defines what pitch corresponds to what CV input. However, I choose to make my VCOs produce middle C (C4) at their typical settings when the 1V/octave input is at 5.00V. Thus I would expect the AVS to be producing 261.6Hz when its KeyCV input is 5V and the VCO's octave switch is in the 16' position. Since you may not have calibrated the octave switch yet, for an all Oakley system you need to adjust TUN1 so that VCO1 produces 523.25Hz (C5) when the octave switch is in its middle, 8', position.

MU and other makes of systems will be different but the range of the TUN1 trimmer is around two octaves so it should be wide enough to cope with a variety of requirements.

OCT1: This sets the interval of the octave switch for VCO1. It has no effect on the pitch in the 8' position but effects equally the 4' and 16' settings. Set the switch to 8'. Play an A below middle C and using the front panel tune pot set the frequency to be exactly 440Hz. Now flip the switch up to its 4' position. Adjust OCT1 so that the note heard is 880Hz +/- 1 cent. Flipping the switch down to 16' should give you 220Hz +/- 1 cent.

SCL2, HFT2, TUN2, and OCT2: These are treated similarly to VCO1's trimmers. Note that VCO2 has an additional pitch control pot, VCO2 Tune, on the front panel. Prior to calibrating VCO2 this pot should be set so that the pointer lines up with the line on the panel at the 9 o'clock position. Care must be taken that this pot doesn't move during the trimming process. Once trimming has been done it's worth checking that the turning this pot will shift the pitch of VCO2 by one octave when it is moved from the 9 o'clock position to the 3 o'clock position. Make sure that the Sync switch is off, ie. in the up position, while trimming VCO2.

F_SCL: The filter within the ASV is not self oscillating and as such it is not that important to get filter tracking perfect. Indeed, you can simply leave this trimmer in its mid position and save yourself some bother. To set it correctly requires either a spectrum analyser (these are often included with your DAW software) or a keen pair of ears.

Turn the resonance and Key Track controls to their maximum value. The level of VCO1 should be turned up full, and the level of VCO2 and the Sub Osc turned off. Set VCO1 to sawtooth.

Play an A below middle C (220Hz) and use the filter Frequency control on the front panel to tune the filter so it accentuates the third harmonic (second overtone) of the sawtooth. Now play an octave higher. An ideally scaled filter will also be accentuating the third harmonic of this new note. If not, move the F_SCL trimmer so that it does. Now play the original note again and retune the filter from the front panel to pick out the third harmonic if it isn't already doing so. Then go up again one octave and re-adjust F-SCL to once again emphasise the third harmonic. Repeat until you are happy that the filter is tracking the oscillators frequency over a reasonable range. The four octaves between 110Hz and 880Hz is usually more than sufficient.

F_FREQ: This trimmer sets the median operating frequency of the filter. Set this so the front panel Frequency control is able to operate over a useful range. The actual setting needed will be determined by the relationship of Key CV with VCO frequency, that is, what note is heard with a certain keyboard control voltage. In an Oakley Modular we use C4 = 5V. Other systems may be different.

To set F_FREQ turn the Filter FM1, Filter FM2, and Key Track controls to their minimums, set Frequency and ENV2 Depth to their middle positions and turn Resonance to its maximum. Turn up VCO1, set the octave switch to 16' and set the waveform to sawtooth. Play the A two below middle C (110Hz). Adjust F_FREQ until the volume of the third harmonic (second overtone) is accentuated.

A_LVL: This sets the final signal amplitude. Ideally it should be adjusted so that the signal level at the output of the ASV is the same as the signal going into the filter, via the VCF In socket.

Turn up the level of VCO1 only, set the octave switch to 16' and set the waveform to sawtooth. Set the Resonance, Filter FM1, Filter FM2, and Key Track controls to their minimums, set ENV2 Depth to its middle position, and turn Frequency and ENV2's Sustain to their maximums. With a patch lead connect the VCO1 Saw socket to the VCF Input socket. Play the A below middle C (220Hz). Adjust A_LVL so that the output sawtooth signal is no more than 10.0V peak to peak. The output signal is also available on the lower pin, pin 1, of the two pin header OUT.

If you can't measure audio AC signals, and remember that most handheld voltmeters can't, then don't patch the sawtooth output as described above, but simply listen to the audio signals coming from Mixer Out, and Main Out, one at a time. Adjust A_LVL so that the volume of the sound heard from the Main Out is roughly the same as, or slightly below, that from Mixer Out.

It's worth pointing out that the filter in the ASV is capable of producing large increases in volume of specific frequencies at high resonance settings. Although I didn't manage to find any setting that caused the main output to clip and become distorted, it got very close when all three mixer levels, and the VCO1 to VCA level, were set to their maximum values. I would therefore urge a little caution and set A_LVL to below the 10.0V peak to peak level, perhaps around 8V peak to peak.

Final Comments

I hope you enjoy using the Oakley ASV module.

If you have any problems with the module, an excellent source of support is the Oakley Sound Forum at Muffwiggler.com. I am 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