digisound

MODULES 80-1:80-1A
±15V POWER SUPPLIES

1. INTRODUCTION

In common with most other analogue synthesisers the DIGISOUND 80 requires a well regulated power supply since several modules, particularly the VCO, make use of the supply to provide current or voltage references. It is also essential, therefore, that the supply be stable both in terms of time and temperature.

The above requirements are met with the 80-1 supplies which are available in two forms.-
a). 80-1 capable of 300mA per rail at +/-15V and is sufficient for about 15 modules plus a monophonic keyboard controller.
b). 80-1A capable of 1A per rail at +/-15V and would be suitable for about forty modules.

The design of the two types is identical and so the notes describe the 80-1 version and then list the component changes required for the larger 80-1A.

2. DESIGN

As will be seen from the circuit diagram, Figure 1, the design is based on the 723 voltage regulator. The technique employed is to generate two identical positive 15V supplies and to tie the output of one to the ground rail of the other to generate the +/-15V required.

The circuit is a typical for the 723 IC. Looking at the first half: R1 improves temperature stability; C3 increases ripple rejection; C4 is for compensation; and C5 reduces noise on the output which originates from the voltage reference diode in the IC. R2 is a current sensing resistor which will limit the output to about 450mA under overload or short circuit conditions. Finally, R3, RV1 and R4 allow precise adjustment of the output voltage and provide about +/-10% adjustment around the nominal 15V output.

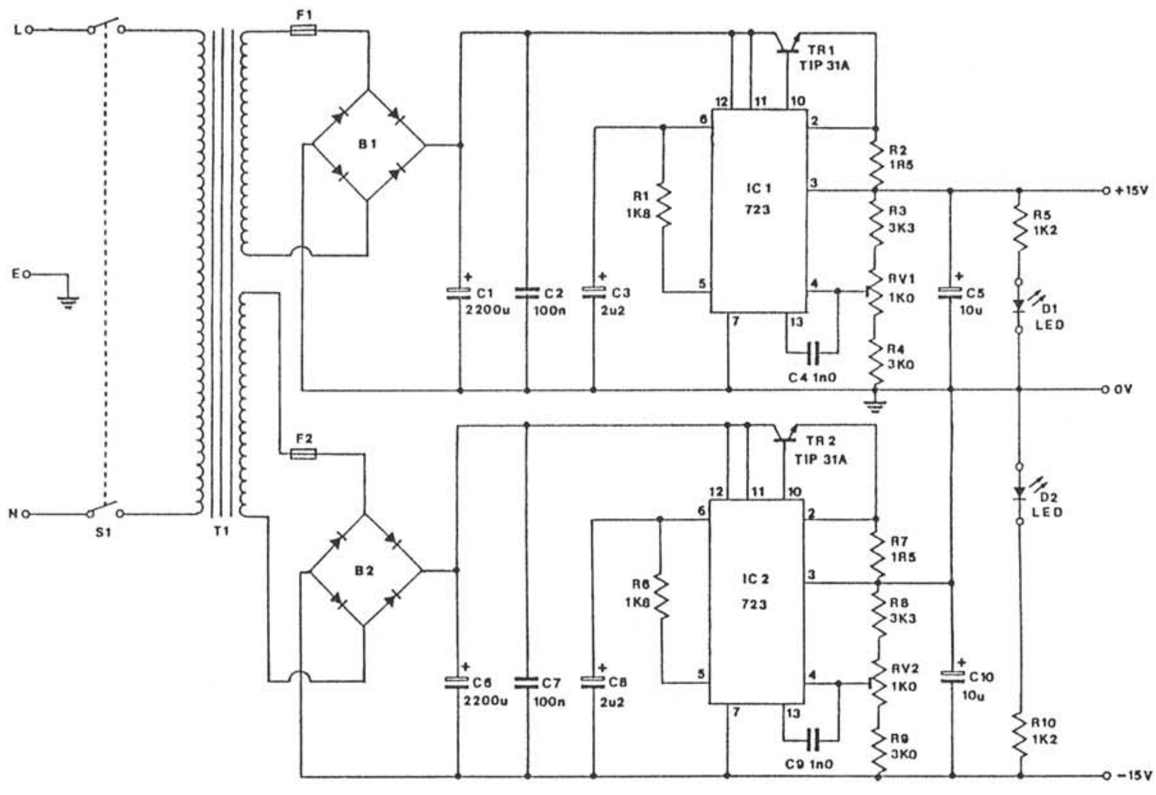


FIGURE 1. CIRCUIT DIAGRAM OF 80-1 POWER SUPPLY

3. CONSTRUCTION

Since the number of components used is low and the PCB is spaciously layed out no component overlay has been printed on the PCB and reference should be made to the overlay shown in Figure 2. Take care on orientation of the IC's; that the bridge is correctly orientated prior to soldering; and also the orientation of the capacitors. As regards the latter, tantalum capacitors will not usually withstand reversal but their body is invariably marked with a '+' and this should face the '+' shown in Figure 2. Large electrolytics, such as C1 and C8 are usually shaped as indicated but additionally they normally have a band at the negative end which may also be identified by the fact that the axial lead is not surrounded by insulation.

Next mount the transistors on the heatsink provided. Space them out on the flat side of the heatsink, as illustrated in Figure 3, and after marking the heatsink drill two 3mm holes - ensuring that these will be clear of the fins. Remove any burrs from around the holes. Place the mica washer on the heatsink (a little heatsink compound should be applied to both sides, if available, but it is not essential in this application since the heatsink is generously sized) and the transistor laid on top such that its mounting hole lines up with that in the washer and also one of the holes made in the heatsink. The round insulating washer is now inserted in the mounting hole of the transistor and the bolt passed through and secured with a nut on the other side of the heatsink. The two washers

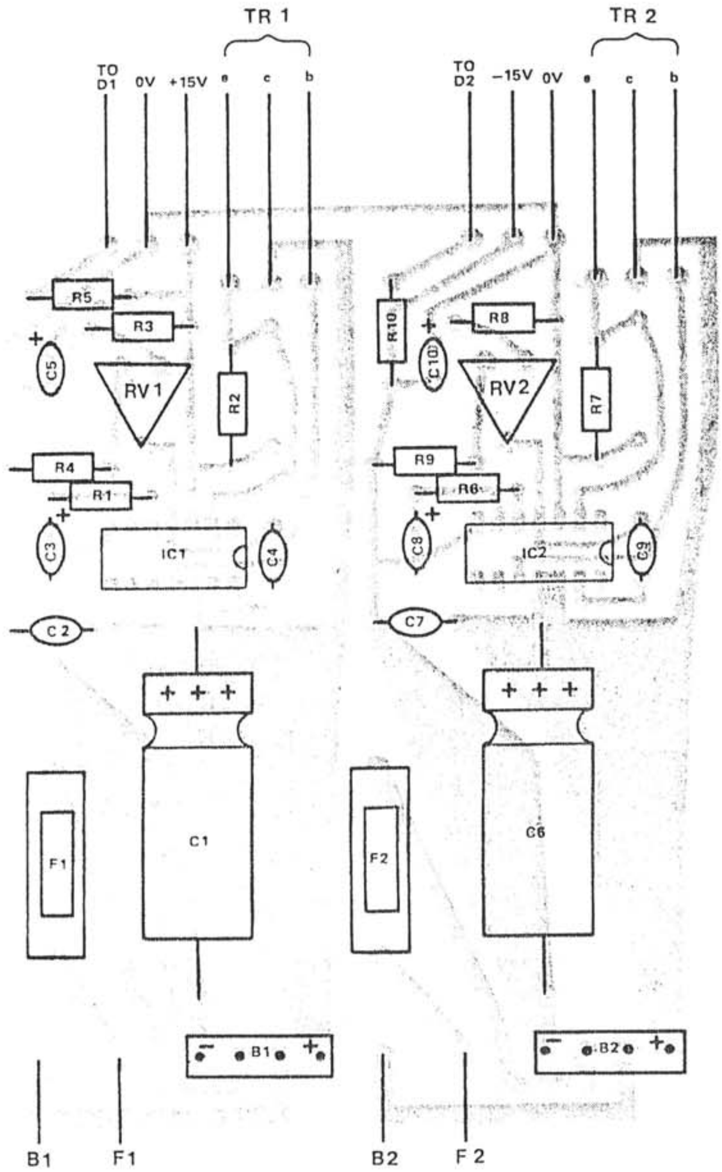


FIGURE 2. PCB COMPONENT OVERLAY

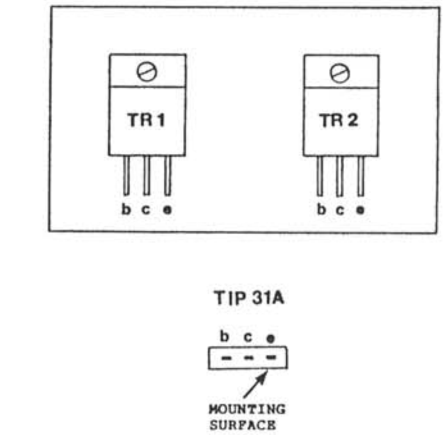


FIGURE 3. MOUNTING TRANSISTORS

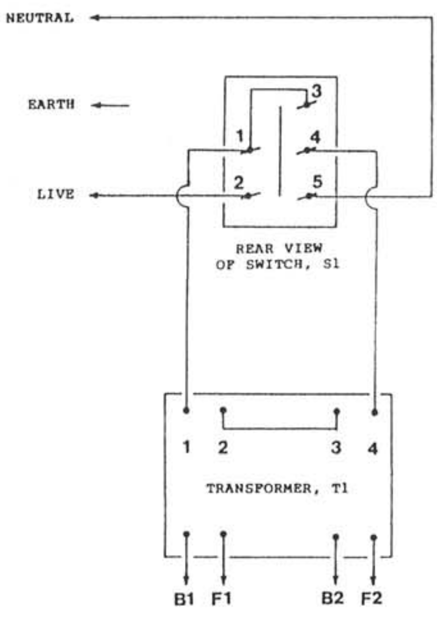


FIGURE 4. CONNECTING MAINS

will ensure that the two transistors are electrically isolated from one another. The latter isolation is essential and if any washer becomes damaged during construction it must be replaced. The heatsink should be mounted vertically and close to the PCB and both parts will accept 'L' brackets to allow vertical mounting. Connect the transistors to the PCB using the information provided in Figures 2 & 3.

LED's tend to vary in respect of whether the short or long lead is the ground connection. It is better to identify the flat on the lens of the LED and the lead on this side will be referred to as the 'flat lead'. The LED's are connected as follows -
+15V LED (D1). Flat lead to ground and other lead to R5.
-15V LED (D2). R10 to flat lead and other lead to ground.

The final step is to wire up the switch and transformer. The wiring is illustrated in Figure 4 and note the interconnections between pins 1 and 3 of the switch and tags 2 and 3 of the transformer (for normal U.K. 200-240V operation). All wiring from the mains to the switch and also the mains connections to the transformer should be made with mains cable.

4. SAFETY

MAINS VOLTAGES CAN BE LETHAL. IF YOU ARE NOT EXPERIENCED WITH CONNECTIONS TO THE MAINS SUPPLY THEN OBTAIN EXPERT ASSISTANCE.

We recommend that an IEL chassis plug is installed in the synthesiser case and the connection made with a separate lead having a mains 3-pin plug at one end and an IEL free socket at the other. The mains plug should be fitted with a fuse of no greater than three amps. Layout will vary a great deal but whatever method is adopted you must ensure that there is no risk of accidentally touching the mains leads and so all of the latter must be insulated. Protective insulating boots may be obtained for the IEL plug, if used, and this same boot will stretch over the switch supplied. These boots must be installed at the wiring stage. The mains tags on the transformer may be covered with a proper insulating tape. If the power supply is housed in a metal chassis then the mains earth lead should be connected to the chassis. Otherwise connect earth to the bolts securing the transformer to the case.

5. CALIBRATION

The only calibration required is to adjust RV1 and RV2 to obtain +15V and -15V as accurately as possible. More important, however, is to set them to a reading close to 15V which may be reproduced. In other words the output of the power supply should be measured from time to time and the trimmers re-adjusted to exactly the same voltage as before. If your oscillators go off tune then always check the power supply first and check it with all the modules normally in use connected up.

6. COMPONENTS

RESISTORS

R1,6	1k8, 5% carbon film
R2,7	1R5, 2.5W wirewound
R3,8	3k3, 1% metal film, 100ppm TC
R4,9	3k0, 1% metal film, 100ppm TC
R5,10	1k2, 5% carbon film

CAPACITORS

C1,6	2200uF, 35/40V electrolytic
C2,7	100nF polyester
C3,8	2u2, 25V tantalum bead
C4,9	1nF polyester
C5,10	10uF, 25V tantalum bead

PRESETS

RV1,2	1k0 cermet

SEMICONDUCTORS

B1,2	2A, 200V bridge
IC1,2	LM 723CN, or equivalent
TR1,2	TIP31A
D1,2	5mm Red LED

MISCELLANEOUS

T1	20VA transformer, 2 X 17V5 secs.
F1,2	1A fuses with PCB holders
S1	DPST rocker switch with neon
-	4.5°C/W heatsink

THE FOLLOWING CHANGES ARE REQUIRED FOR THE 80-1A POWER SUPPLY.-

R2,7	0R47, 2.5W wirewound
TR1,2	TIP 3055
T1	50VA transformer, 2 X 15V secs.
F1,2	2A fuses with PCB holders

NOTE: The orientation of the TIP 3055 is the same as the TIP 31A shown in the construction notes.

7. ADDENDA

A. LOCATION. The preferred method is to locate the power supply within the keyboard case. This method provides some shielding between the transformer and the modules as well as preventing heat build-up around the modules. In addition the proposed location will be found convenient when: (i) a power amplifier is fitted at which time the separate transformer for this may also be housed within the case and utilise the same mains plug and switch; (ii) when the keyboard logic controller is housed in the case it is convenient, for a monphonic system, to obtain the small amount of +5V required from the 80-1, or 80-1A, power supply.

B. CONNECTING MODULES. The best method is to have the power wires from each module go directly to a set of connectors which are as close as possible to the power supply. This arrangement is not, however, very convenient and no problems have been encountered as a result of using a bus distribution system. THe latter may take the form of three wires (+15V, -15V and 0V) running along the back of the module housing and connected at one end to the power supply in the keyboard case. Miniature 3-pole connectors are suitable for connecting the two housings. A low cost method consists of using 'TAP-IN' connectors with the appropriate heavy gauge wire for the bus. These connectors make a branch connection, as illustrated in figure 5, by closing and then pressing together with a pair of pliers. Ensure that the connection nearest the hinge is closed first and it is good practice to test that the connection

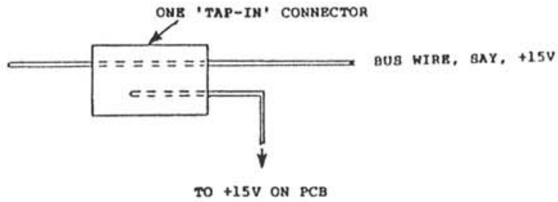


FIGURE 5. USING 'TAP-IN' CONNECTORS

has been properly made by checking the voltage on the branch wire, using the 0V branch as the ground for the voltmeter.

The DIGISOUND 80 system allows you to connect other designs to the 80 series modules. This may cause problems with the bus system if the additions draw a high intermittent current or are noisy in some other respects. Such problems may usually be overcome but in a large synthesiser there are merits in having more than one power supply such that the voltage controlled oscillators, filters and amplifiers are connected to one supply which is free from possible interference on the supply lines.

C. CHIRI CONNECTORS. All 80 series PCB's accept the CHIRI 4123 plug which connects with a non reversible 4113 free socket. The use of these plugs and sockets avoids costly damage that may result from power reversal as well as allowing easy relocation of the modules, as required. When using these connectors do not be tempted to push in the connecting pins of the socket prior to soldering the wires in place. When pressed home, after soldering, they lock into the casing and will then provide a good connection with the pins of the plug.

digisound

MODULE 80-2
VOLTAGE CONTROLLED OSCILLATOR

1. INTRODUCTION

The DIGISOUND 80-2 voltage controlled oscillator (VCO) has a 1V/octave response with a high level of stability and accuracy in the frequency range of 5Hz to 10kHz. The quality of the VCO is such that the modules may be used in polyphonic systems, such as, the ALPHADAC 16 microprocessor controller. The full frequency range is approximately 0.1Hz to 50kHz.

The design is based on the CEM 3340 IC from Curtis Electromusic Specialties. The latter provides triangle, sawtooth and pulse waveforms and in the 80-2 a sine wave is derived from the triangle output. These four waveforms are 0 to +10V in amplitude. Additionally the sine and triangle are available as +/- 5V outputs. The duty cycle of the pulse waveform may be varied from 0 to 100%, either manually or by an external control voltage thus allowing pulse width modulation techniques. In addition to frequency modulation via the exponential control input(s) a linear frequency modulation input is included which produces a 10% change in frequency per volt.

The CEM 3340 has three methods for synchronising the VCO to other oscillators of the same type and these methods have been incorporated in the design to provide an exceptional range of modulation and harmonic locking effects.

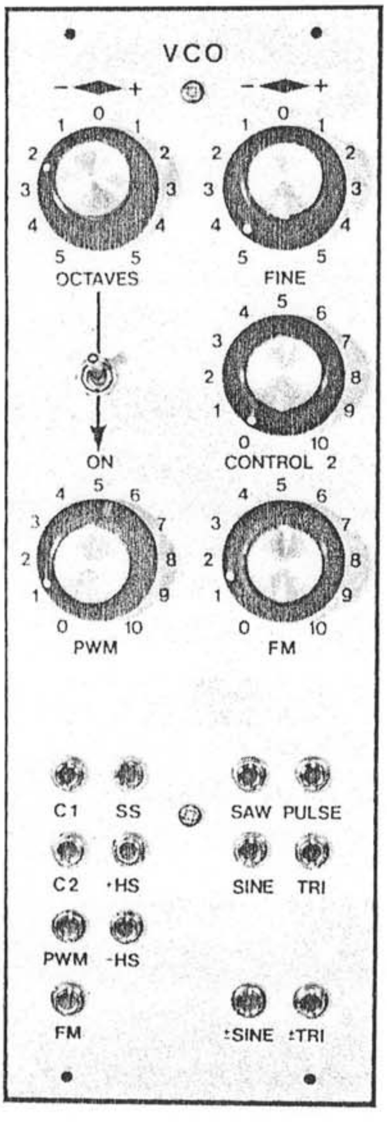


FIGURE 1. 80-2 PANEL

2. DESIGN

The complete circuit diagram for the 80-2 VCO is shown in Figure 2. Most of the circuit is centred around IC1, the CEM 3340, whose pin out and functional block diagram is shown in Figure 3. The frequency control input, pin 15, is a summing stage while pin 14 provides the scaling factor. Since the current gain of the internal multiplier is set near unity 100k input resistors (R5, R6) and a 1k8 scaling resistor (R3) produce the standard 1V/octave response and about 18mV at the base of Q1. Note that R6 has an attenuator, RV9, attached and this input will normally be used for frequency modulation. The input to R5 on the other hand will invariably be connected to the keyboard control voltage and the VCO is calibrated through this input to achieve the accuracy necessary when operated from the keyboard. Components R4 and RV2 set the initial frequency of the oscillator and have been chosen such that with no external voltage applied the frequency may be adjusted to 65.406Hz, i.e., the lowest note of a four octave C-C keyboard. R7 and RV7 allow manual adjustment of the VCO by +/-5octaves. Switch S1 has been fitted between RV7 and R7 so that in normal use, i.e., keyboard; precise octave shift; and external voltage controls, slight variation in RV7 will not cause the oscillator to go out of tune. R8 and RV8 provide a fine adjustment of approximately +/-0.5 octaves and is essentially for tuning purposes. The other components (R9, C5) on the summing input are for compensation and are always required. The sum of the input voltages to pin 15 should always remain positive for proper operation of the oscillator.

For greatest accuracy of the internal multiplier the current flowing out of pin 2 should be close to the current flowing out of pin 1 and this balance is achieved with RV1.

The exponential generator in the CEM 3340 is capable of delivering a current, for charging and discharging the timing capacitor, C7, from greater than 500uA to less than the input bias current of the buffer which gives a typical frequency range of 500,000:1. For synthesiser applications, however, one should use the most accurate portion of this range, which is from 50nA to 100uA. Thus with a 1nF timing capacitor and a positive supply voltage of +15V the most accurate range will be from 5Hz to 10kHz.

As is normal with exponential generators a reference current needs to be established and for the CEM 3340 this is 10uA which is derived from R11 connected to the positive supply. This input, pin 13, may also be used for linear frequency modulation of the VCO. R12 adjusts the FM range to a 10% change in frequency per volt and an attenuating potentiometer, RV10, allows manual adjustment of this range. The FM input has been AC coupled so as to avoid errors from any DC offset on driving inputs. It may, however, be DC coupled so long as the user is aware that any DC offsets will cause the oscillator to go out of tune. Furthermore, a negative current at pin 13 in excess of the reference current will gate the oscillator off. A stable reference current is essential to maintain the accuracy of the oscillator and thus a stable power supply must be used.

One of the biggest drawbacks to exponential VCO's has been their temperature sensitivity which resulted in frequency drift as they warmed up. One of the novel features of the CEM

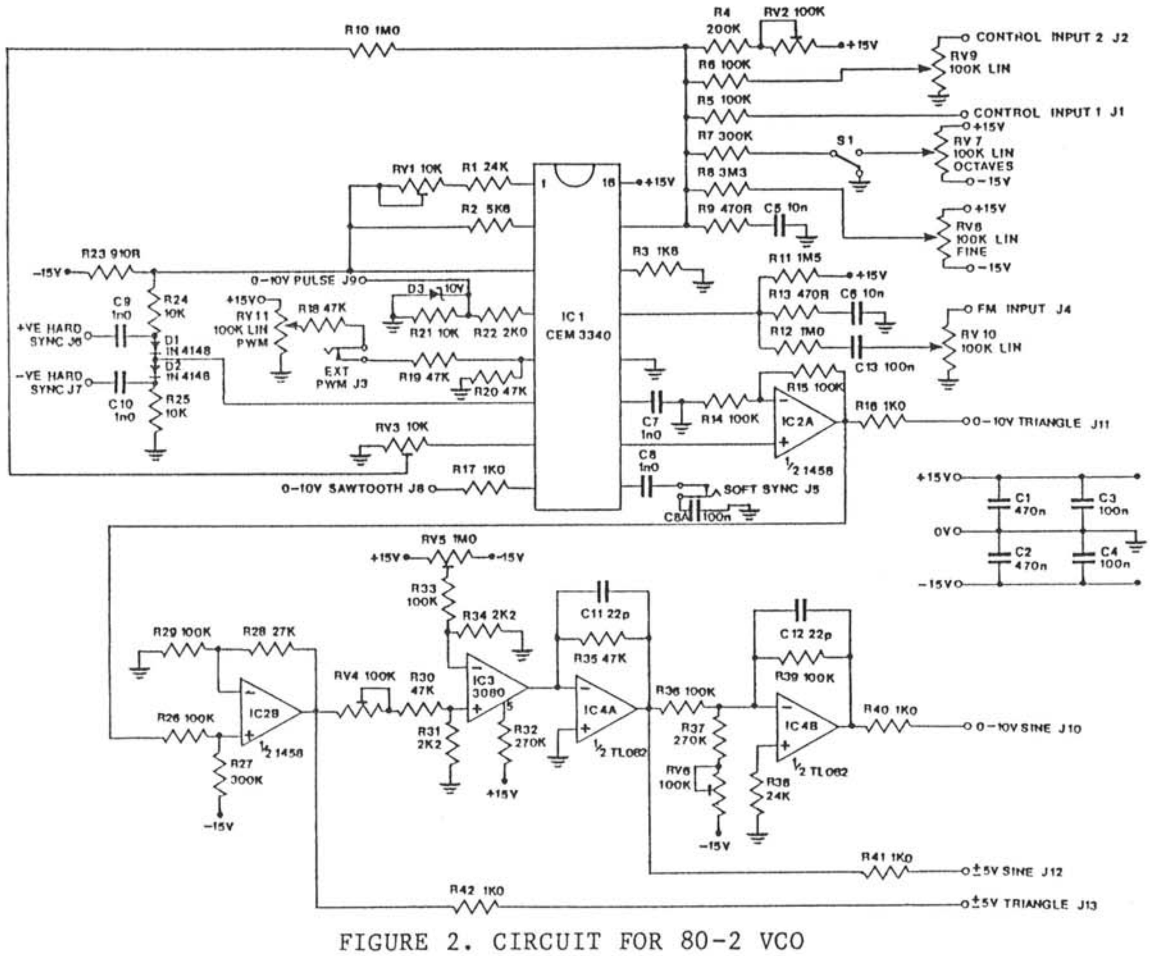


FIGURE 2. CIRCUIT FOR 80-2 VCO

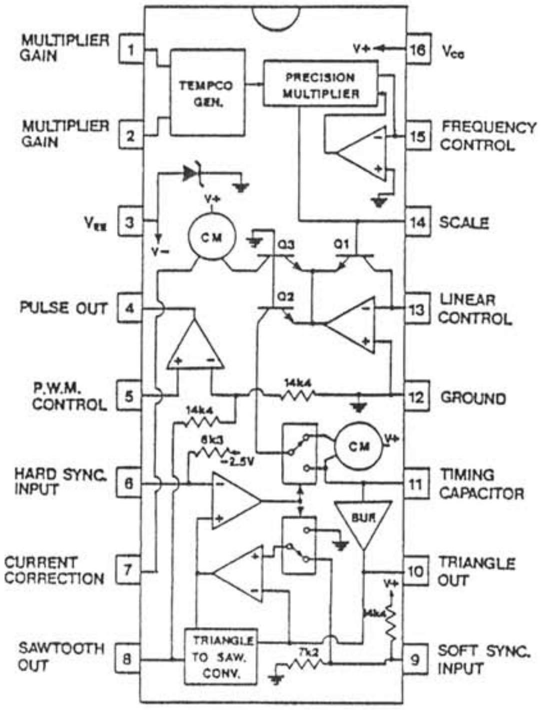


FIGURE 3. CEM 3340 IC

3340 is the incorporation of temperature compensation. This is achieved by multiplying the current sourced into the control pin (pin 15) by a coefficient directly proportional to the absolute temperature. This coefficient is produced by the 'Tempco Generator' using the same mechanism as in the exponential generator and thus cancellation is nearly perfect.

A further problem with transistor based exponential generators is that their bulk emitter reistance becomes significant as current is increased and this in turn will cause the oscillator to go flat. With the CEM 3340 this situation applies when current from Q2 (see Figure 3) is greater than 50uA. Means of correcting this source of error have been included since pin 7 outputs a current which is a quarter of the exponential generator current. The current is converted into a voltage across RV3 and a proportion fed back into the control input via R10.

All waveform outputs from the IC are short circuit protected and may be shorted continuously without damaging the device. A 0-10V sawtooth waveform is available at pin 8 which can sink at least 0.6mA and source over several milliamps without any effect on oscillator performance and only a negligible effect on waveshape. The pulse output from pin 4 is an open NPN emitter and therefore requires a pull down resistor to ground or a negative voltage. This output has been clamped with a 10V zener diode, D3, to give a 0-10V pulse output. Pin 5 allows pulse width modulation and 0 to +5V applied to this pin will vary the pulse width from 0 to 100%. Attenuating resistors R19 and R20 increase the control range to +10V, which is the standard adopted for the DIGISOUND 80 series of modules. RV11 connected to +15V provides manual control of pulse width and this control voltage is reduced by R18 such that the effective control voltage from RV11 is +10V. RV11 is connected to a jack socket, J3, such that it is disabled when an external control voltage is being applied. Note that 0 to 100% pulse width modulation results in the pulse 'disappearing' at the extremes of the potentiometer or external control voltage. Pin 10 outputs a 0-5V triangle waveform. Although the sink and source capabilities approach those of the sawtooth this triangle output has a finite impedance and also drives the comparator with the result that loading into even a 100k input may lower frequency by 0.15%, in the worst case. This output has therefore been buffered by IC2a and addition of R14 and R15 increases the gain by two to provide a 0-10V output.

The triangle output is converted to +/-5V at IC2b (this output is also made available from the VCO) and attenuated to about +/-100mV by RV4, R30 and R31 prior to IC3 which is a CA 3080E, or equivalent. At high input levels this OTA becomes non linear in response and use is made of this to convert the triangle waveform into a sine wave. RV4 adjusts the third harmonic content while RV5 and associated components at the inverting input of IC3 trim the second harmonic. IC4a converts the current from IC3 to a voltage and provides a +/-5V sine wave. This output is inverted compared to all other outputs but this is not detrimental for normal use. It should be noted, however, that mixing of this waveform with other outputs will have a subtracting effect. Finally the +/- sine output is shifted to a 0-10V output by IC4b with R37 and RV6 used for level shifting.

The CEM 3340 will operate from a wide range of power supplies but +/-15V is the standard adopted for the 80 series. For negative supplies greater than -7V5 a current limiting resistor must be employed at the negative supply input, pin 3. For -15V a 910R resistor, R23, is required.

Synchronisation of oscillators is often used to prevent unpleasant beating effects when two, or more, VCO's are set to ratios to produce a complex waveform. Synchronisation may, however, be used to produce some pleasing timbral effects. The CEM 3340 has a wider range of synchronising effects than found on conventional synchronised oscillators. Soft synchronisation by negative pulses to pin 9 causes the triangle upper peak to reverse direction prematurely with the result that the oscillation period is an integral multiple of the pulse period. If this input is not in use it should be by-passed to ground with a 100nF capacitor, C8A, so as to prevent unwant synchronisation or waveform instability from noise pulses on the positive supply line. Pin 6 is used for hard synchronisation and R24, D1 and C9 allow synchronisation from rising edges while R25, D2 and C10 allow synchronisation from falling edges. A positive sync. pulse will cause the triangle wave to reverse direction only during the rising portion of the triangle, whereas a negative sync. pulse will cause direction reversal only during the falling portion. Figure 4 illustrates the hard synchronisation capabilities of the CEM 3340 and as employed in the 80-2 module.

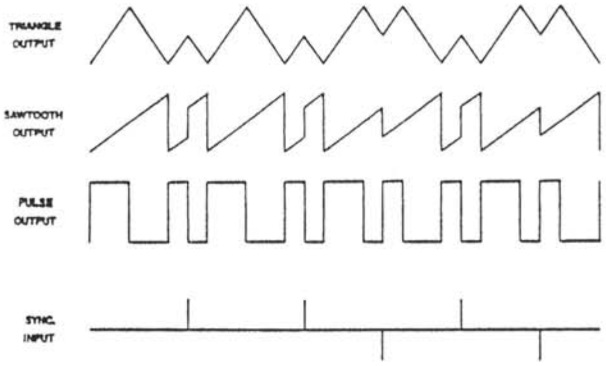


FIGURE 4. HARD SYNCHRONISATION

Reference should be made to the CEM 3340 data sheet for further information regarding component selection for other frequency ranges, alternative power supplies and so on. As regards utilising the synchronisation and other capablities of the VCO then reference should be made to 'USING THE DIGISOUND 80 MODULAR SYNTHESISER'.

3. CONSTRUCTION

The 80-2 PCB is printed with a component overlay which simplifies the construction stage. The overlay is reproduced in Figure 5 to allow checking of component placement after the module has been constructed.

Take special care regarding orientation of the IC's. Even after installing the DIL sockets the number '1', denoting pin 1, will still be visible on the PCB. In any event compare the completed PCB against Figure 5 before applying power. For the diodes a line is printed on the PCB, either above or below the 'D' number and this line corresponds with the band on the diode denoting the cathode.

Wiring of potentiometers and other panel connections to the PCB are shown in Figure 6. This diagram illustrates the components when viewed from the rear of the panel. The arrows and associated letters indicate that a wire connection must be made from the position shown to the PCB. The latter has a connecting point on its front edge with letters corresponding to those shown in Figure 6. DO NOT MAKE THE CONNECTION BETWEEN THE FINE CONTROL POTENTIOMETER, RV8, AND THE PCB AT THIS STAGE. Note the location of C8A on the soft synchronisation jack socket. Unlike most other DIGISOUND 80 modules it will be found most convenient to mount the PCB in such a way that the power connecting point is at the top of the panel. This reduces the length of most wires and also gives ready access to the frequency trimmers by simply tilting the panel forward from its housing.

The jack sockets in the diagram are of the type supplied by Digisound Limited. The top connection, as illustrated, is the connection which is made with the jack when the latter is inserted. The lower connection is disabled by insertion of a jack plug. Finally, the tab under the socket is the ground connection. It is recommended that all of these ground connections are wired to the 0V line since this facilitates connection of the VCO to other equipment which may be using a separate power supply. The ground tabs may be joined together using tinned copper wire but other panel wiring should be made with

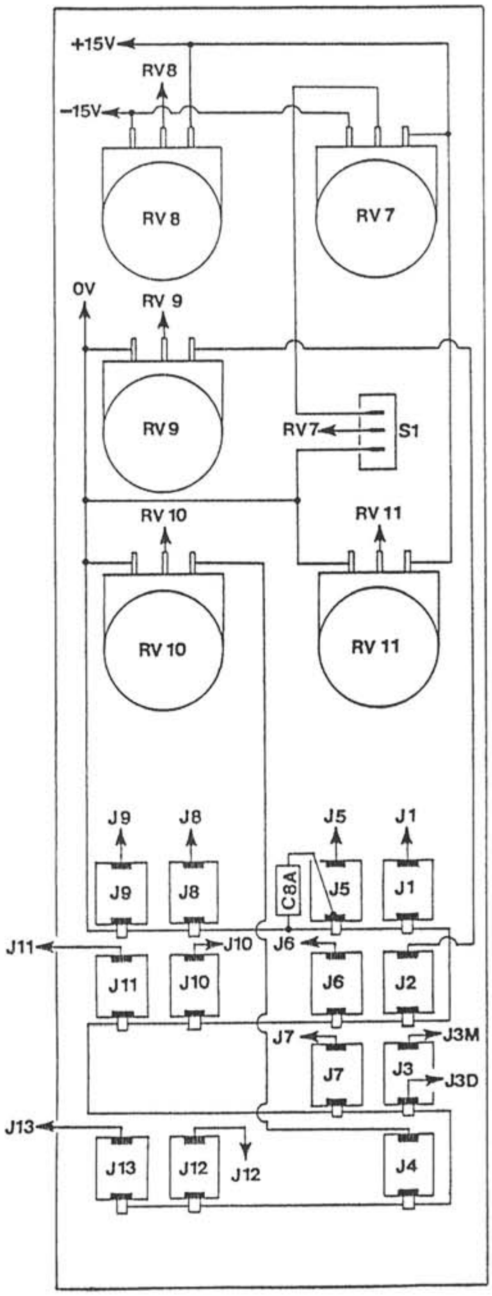


FIGURE 6. PANEL WIRING

insulated wire. 1/0.6mm insulated wire is ideal for panel wiring since it retains any shaping and thus allows a neat appearance to be obtained. Wires should be kept as short as practical.

One special point to note is that cleanliness of the PCB is particularly important in the area around the timing capacitor, C7. At low frequencies the current to this capacitor is only a few nanoamps and so residual solder flux or other dirt around the capacitor may degrade performance. Use either a proprietary PCB solvent cleaner or else carefully scrape clean the foil side of the PCB around the solder points for C7.

On completion of the construction stage carefully examine the underside of the PCB to ensure that all components and connections are properly soldered and that no solder bridges have been formed. Also check polarity of power supplies and that a good ground (0V) connection is present before applying the power to the module. For example, if the recommended CHIRI connector is being used then check voltage at its pins using 0V as the ground for the meter. Do not have the power turned on when the connector is mated with the CHIRI plug on the PCB. A further precaution is to have the VCO connected to an oscilloscope or an amplifier when it is powered for the first time. For the latter connect the +10V triangle to an amplifier, with its 'volume' control nearly off, and apply power to the module. If there is no response

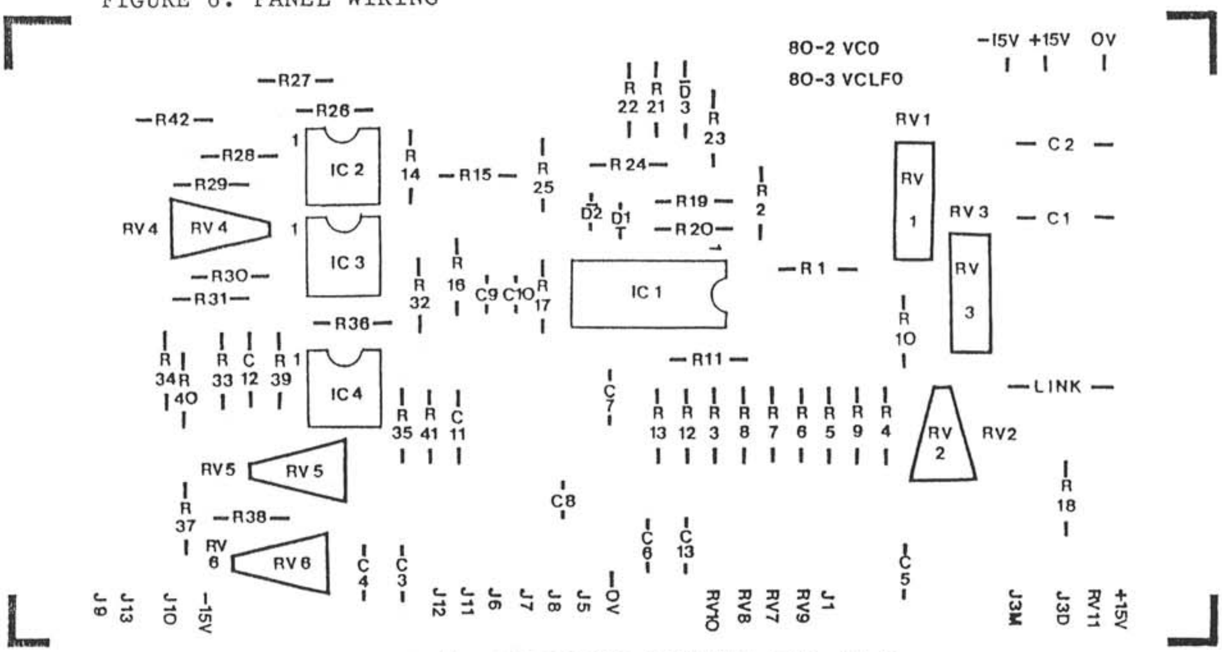


FIGURE 5. COMPONENT OVERLAY FOR 80-2

then switch off immediately and repeat all of the checks mentioned above. If functioning then check the other waveform outputs although remember that the sinewave has not been trimmed at this time.

After constructing the module you will be anxious to try it out. Do not, however, skip the tedious checking procedure since mistakes can be expensive.

4. CALIBRATION

The first step is to adjust the sinewave outputs and trimmers RV4 to RV6 should be set to their mid position. With an oscilloscope, or by ear via an amplifier, adjust RV4 then RV5 for purest sinewave output. These adjustments may have to be repeated a few times to obtain the best result. Next adjust RV6 to get the +10V sine output referenced to ground. The simplest method is with a DC coupled oscilloscope and RV6 is adjusted such that the bottom of the sinewave is at 0V. An alternative method is to use a voltage controlled amplifier (VCA), such as the DIGISOUND 80-9. Connect a signal to the VCA and insert a jack plug into the exponential control input and while listening to the output, using an audio amplifier, adjust signal level or amplifier volume such that no sound is heard. Now connect the +10V sinewave to the VCA and adjust RV6 until no sound is heard. For the latter step the VCO should be set to its lowest frequency, that is, S1 to the 'ON' position and RV7 fully anti-clockwise. If a VCA is not available at this stage then this step may be left until such times as suitable equipment is available. One should remember, however, that it is not trimmed if it is used in a patch.

The last and most important step is to calibrate the oscillator to the 1V/octave relationship. Before starting first check the voltage from the power supplies and trim to +/-15V, if necessary. Calibration is greatly simplified if one has available: a variable voltage source (from a potentiometer or from a calibrated keyboard); an accurate voltmeter; and a digital frequency meter. There are, however, a number of other methods. One is to use a previously calibrated oscillator or a musical instrument and make the calibration using the beat frequency technique. Another approach is to build two simple, but stable, fixed frquency oscillators and use them in conjunction with an oscilloscope to calibrate the VCO by generating Lissajous figures.

Whichever method is used the calibration procedure is as follows. Set the wiper of RV3 at the grounded end by turning the adjusting screw clockwise until clicks are heard (or seen if a multiturn with a transparent cover is supplied) and the wiper of RV1 to about mid position. Put switch, S1, to the 'OFF' position so as to disable the 'octaves' control, RV7. Next apply a positive voltage to Control Input 1 (R5) until the frequency is about 200Hz. Increase voltage by exactly one volt (as accurately as the measuring equipment allows) and adjust RV1 until the frequency is double that of the first frequency. These steps may have to be repeated several times in order to achieve an exact doubling of frequency per volt applied. If using a digital frequency meter then the best approach is to note the first measurement, increase voltage by one volt and divide the second reading by the first. If the ratio is greater than 2 then turn RV1 anti-clockwise and if less than 2 then turn it clockwise. Now decrease the voltage by exactly one volt and calculate the new ratio and adjust RV1 as before. Continue the procedure until a ratio of exactly 2 is obtained and during the calibration it may be necessary to alter the voltage level such that the calibration is carried out in the general range of 150 to 500Hz.

Repeat the above procedure except for starting at an initial frequency of about 5kHz and adjusting RV3 until a doubling of this frequency is obtained when the applied voltage is increased by exactly one volt. Note that this is a fine adjustment and if the low frequency calibration has not been carried out accurately then it may be found impossible to carry out this step effectively. If accurate calibration is not possible at this stage then set the wiper of RV3 to its mid position. If the high frequency trim has been carried out then re-check the low frequency calibration and the performance of the VCO over its specified range.

Two important things to note are: (a) That the VC0 has been calibrated for Control Input 1 which will normally be connected to the keyboard. It may not be in exact calibration for Control Input 2 due to small variations in components; (b) Resistors and other components will age and their change may be quite significant for a period following soldering. If convenient the best approach is to make a quick initial calibration and then a more accurate one after using, or keeping the module powered up, for several hours.

The oscillator may now be adjusted so that with no input voltages it will be tuned to the lowest frequency of a four octave C-C keyboard. Adjust RV2 to 65.4Hz if connected to a keyboard with zero volts at lowest key. This step is not essential until the VCO is connected to a keyboard. The value of RV2 or R4 may be altered to suit other keyboard requirements.

Finally, the wire to the wiper of the fine control potentiometer, RV8, may be connected up and this may also be used to tune to particular keyboard requirements or to other instruments.

5. COMPONENTS

RESISTORS, 5%, 1/4w carbon film

R8	3M3
R9,13	470R
R12	1M0
R14,15,26,29,33,36,39	100k
R16,17,40,41,42	1k0
R18,19,20,30,35	47k
R21,24,25	10k
R22	2k0
R23	910R
R27	300k
R28	27k
R31,34	2k2
R32,37	270k
R38	24k

RESISTORS, 1%, 1/4w metal film, 100ppm

R1	24k
R2	5k6
R3	1k8
R4	200k
R5,6	100k
R7	300k
R10	1M0
R11*	1M5

*may be replaced by low TC carbon film resistor.

CAPACITORS

C1,2	470n polyester
C3,4,13	100n polyester
C5,6	10n polyester
C7	1n0 polystyrene
C8,9,10	1n0 polyester
C8A (see text)	100n polyester
C11,12	22p polystyrene

POTENTIOMETERS, PRESETS

RV1,3	10k cermet multiturn
RV2	100k cermet
RV4,6	100k carbon
RV5	1M0 carbon
RV7,8,9,10,11	100k lin.

SEMICONDUCTORS

IC1	CEM 3340
IC2	LM 1458*
IC3	CA 3080E*
IC4	TL 082
D1,2	1N4148
D3	BZY88 10V

*or equivalent

MISCELLANEOUS

S1	SPDT sub. min. switch

MODULE 80-3
V.C. LOW FREQUENCY OSCILLATOR

1. INTRODUCTION

The DIGISOUND 80-3 voltage controlled low frequency oscillator (VCLFO) is designed to the same standard as the 80-2 VCO so as to provide some unique synthesis capabilities. An external +10V control voltage will sweep the oscillator over the range of 0.2 to 205 Hz but by using the manual controls plus external control voltages the range of the oscillator is approximately 0.01Hz to 5kHz. The range may be altered, as described in the construction notes.

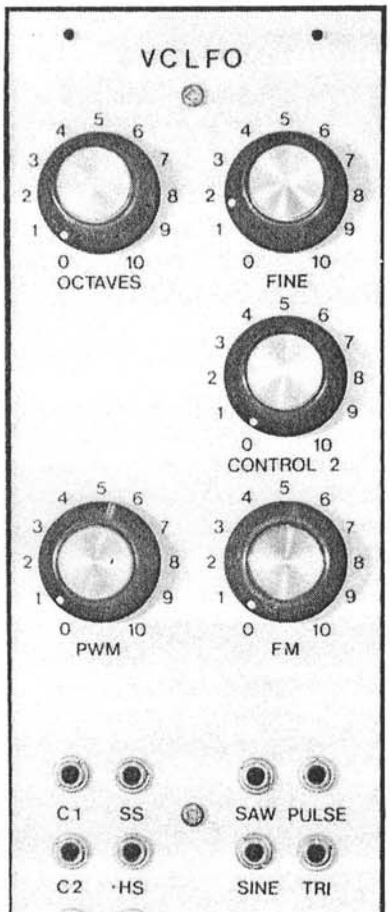


FIGURE 1. 80-3 PANEL

The design is based on the CEM 3340 IC from Curtis Electromusic Specialties. The latter provides triangle, sawtooth and pulse waveforms and in the 80-3 a sine wave is derived from the triangle output. These four waveforms are 0 to +10V in amplitude. Additionally the sine and triangle are available as +/-5V outputs. The duty cycle of the pulse waveform may be varied from 0 to 100%, either manually or by an external control voltage thus allowing pulse width modulation techniques. In addition to frequency modulation via the exponential control inputs(s) a linear frequency modulation input is included which produces a 10% change in frequency per volt. The scaling of the VCLFO is the standard 1V/octave.

The VCLFO also includes the three synchronising techniques employed in the 80-2 VCO and thus may be used for synchronising techniques with the latter modules.

2. DESIGN

The complete circuit diagram for the 80-3 VCLFO is shown in Figure 2. Most of the circuit is centred around IC1, the CEM 3340, whose pin out and functional block diagram is shown in Figure 3. The frequency control input, pin 15, is a summing stage while pin 14 provides the scaling factor. Since the current gain of the internal multiplier is set near unity 100k input resistors (R5, R6) and a 1k8 scaling resistor (R3) produce the standard 1V/octave response and about 18mV at the base of Q1. Components R4 and RV2 allow setting to some initial frequency, typically 0.2Hz. R7 and RV7 allow manual adjustment of the oscillator by 10 octaves up from the initial setting while R8 and RV8 allow a fine adjustment of a further octave higher. The other components (R9, C5) on the summing input are for compensation and are always required. The sum of the input voltages to pin 15 should always remain positive for proper operation of the oscillator.

For greatest accuracy of the internal multiplier the current flowing out of pin 2 should be close to the current flowing out of pin 1 and this balance is achieved with RV1.

The exponential generator in the CEM 3340 is capable of delivering a current, for charging and discharging the timing capacitor, C7, from greater than 500uA to less than the input bias current of the buffer which gives a typical frequency range of 500,000:1. For synthesiser applications, however, one should use the most accurate portion of this range, which is from 50nA to 100uA. Thus with the 10nF timing capacitor and a positive supply voltage of +15V the most accurate range will be from 0.5Hz to 1kHz.

As is normal with exponential generators a reference current needs to be established and for the CEM 3340 this is 10uA which is derived from R11 connected to the positive supply. This input, pin 13, may also be used for linear frequency modulation of the VCO. R12 adjusts the FM range to a 10% change in frequency per volt and an attenuating potentiometer, RV10, allows manual adjustment of this range. The FM input has been AC coupled so as to avoid errors from any DC offset on driving inputs. It may, however, be DC coupled so long as the user is aware that any DC offsets will cause the oscillator to go out of tune. Furthermore, a negative current at pin 13 in excess of the reference current will gate the oscillator off. A stable reference current is essential to maintain the accuracy of the oscillator and thus a stable power supply must be used.

One of the biggest drawbacks to exponential VCO's has been their temperature sensitivity which resulted in frequency drift as they warmed up. One of the novel features of the CEM 3340 is the incorporation of temperature compensation. This is achieved by multiplying the current sourced into the control pin (pin 15) by a coefficient directly proportional to the absolute temperature. This coefficient is produced by the 'Tempco Generator' using the same mechanism as in the exponential generator and thus cancellation is nearly perfect.

A further problem with transistor based exponential generators is that their bulk emitter reistance becomes significant as current is increased and this in turn will cause the oscillator to go flat. With the CEM

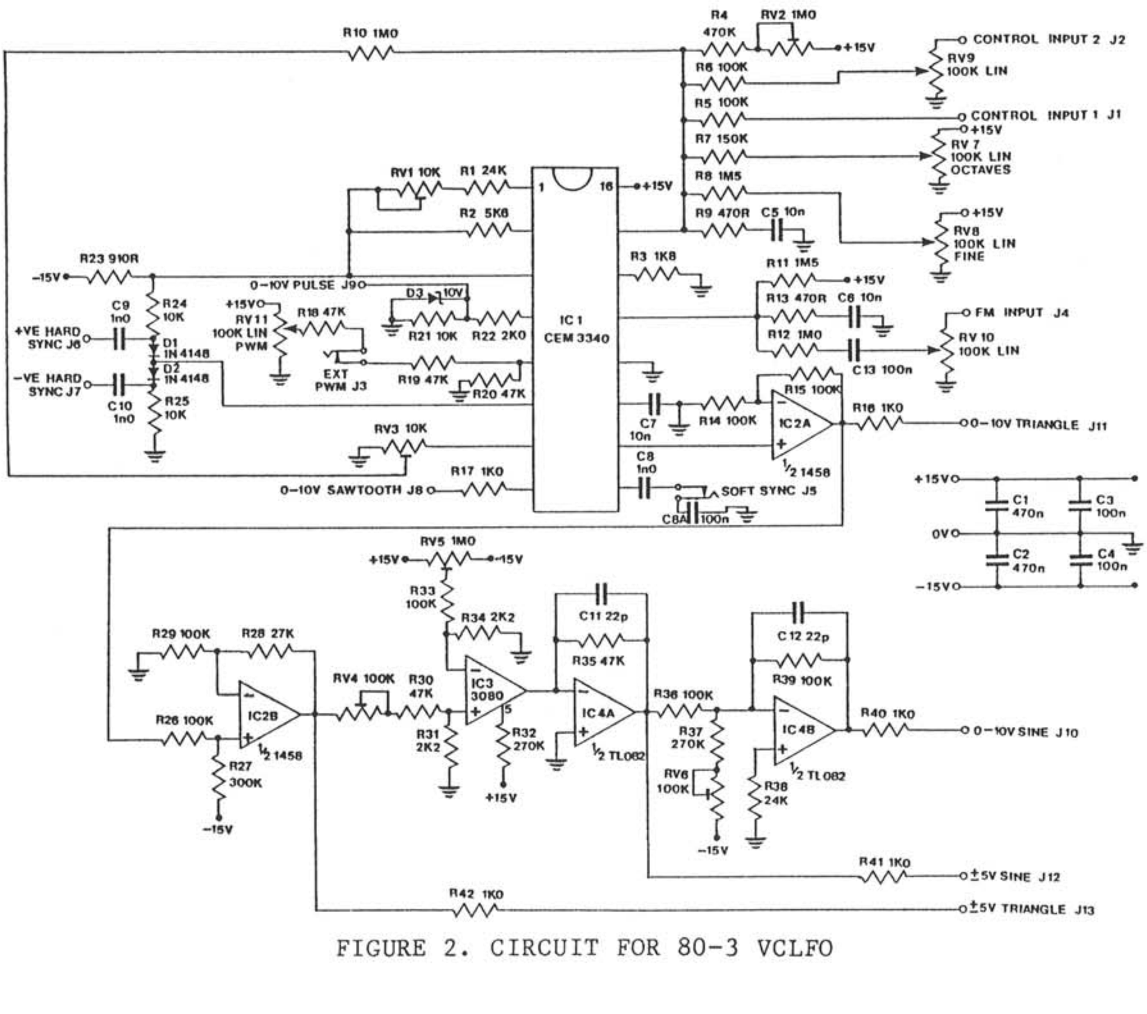


FIGURE 2. CIRCUIT FOR 80-3 VCLFO

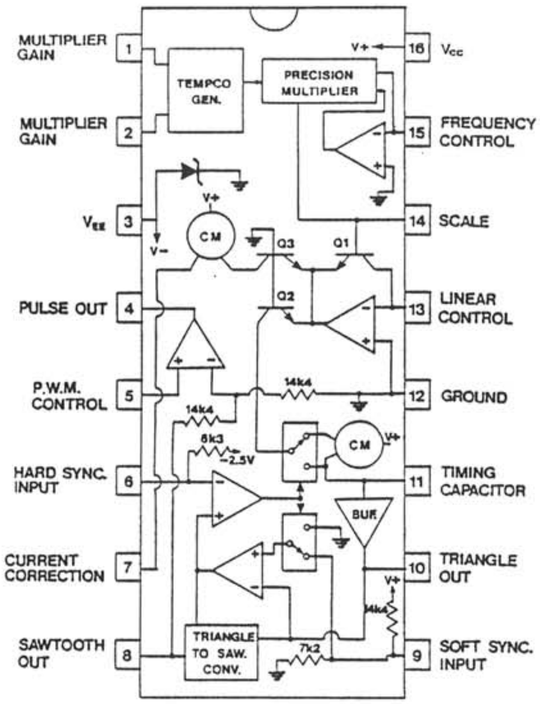


FIGURE 3. CEM 3340 IC

3340 this situation applies when current from Q2 (see Figure 3) is greater than 50uA. Means of correcting this source of error have been included since pin 7 outputs a current which is a quarter of the exponential generator current. The current is converted into a voltage across RV3 and a proportion fed back into the control input via R10.

All waveform outputs from the IC are short circuit protected and may be shorted continuously without damaging the device. A 0-10V sawtooth waveform is available at pin 8 which can sink at least 0.6mA and source over several milliamps without any effect on oscillator performance and only a negligible effect on waveshape. The pulse output from pin 4 is an open NPN emitter and therefore requires a pull down resistor to ground or a negative voltage. This output has been clamped with a 10V zener diode, D3, to give a 0-10V pulse output. Pin 5 allows pulse width modulation and 0 to +5V applied to this pin will vary the pulse width from 0 to 100%. Attenuating resistors R19 and R20 increase the control range to +10V, which is the standard adopted for the DIGISOUND 80 series of modules. RV11 connected to +15V provides manual control of pulse width and this control voltage is reduced by R18 such that the effective control voltage from RV11 is +10V. RV11 is connected to a jack socket, J3, such that it is disabled when an external control voltage is being applied. Note that 0 to 100% pulse width modulation results in the pulse 'disappearing' at the extremes of the potentiometer or external control voltage. Pin 10 outputs a 0-5V triangle waveform. Although the sink and source capabilities approach those of the sawtooth this triangle output has a finite impedance and also drives the comparator with the result that loading into even a 100k input may lower frequency by 0.15%, in the worst case. This output has therefore been buffered by IC2a and addition of R14 and R15 increases the gain by two to provide a 0-10V output.

The triangle output is converted to +/-5V at IC2b (this output is also made available from the VCO) and attenuated to about +/-100mV by RV4, R30 and R31 prior to IC3 which is a CA 3080E, or equivalent. At high input levels this OTA becomes non linear in response and use is made of this to convert the triangle waveform into a sine wave. RV4 adjusts the third harmonic content while RV5 and associated components at the inverting input of IC3 trim the second harmonic. IC4a converts the current from IC3 to a voltage and provides a +/-5V sine wave. This output is inverted compared to all other outputs but this is not detrimental for normal use. It should be noted, however, that mixing of this waveform with other outputs will have a subtracting effect. Finally the +/- sine output is shifted to a 0-10V output by IC4b with R37 and RV6 used for level shifting.

The CEM 3340 will operate from a wide range of power supplies but +/-15V is the standard adopted for the 80 series. For negative supplies greater than -7V5 a current limiting resistor must be employed at the negative supply input, pin 3. For -15V a 910R resistor, R23, is required.

Synchronisation of the VCLFO may be used to provide some harmonic locking effects when used with 80-2 VCO's and produce some useful effects. The CEM 3340 has a wider range of synchronising effects than found on conventional synchronised oscillators. Soft synchronisation by negative pulses to pin 9 causes the triangle upper peak to reverse direction prematurely with the

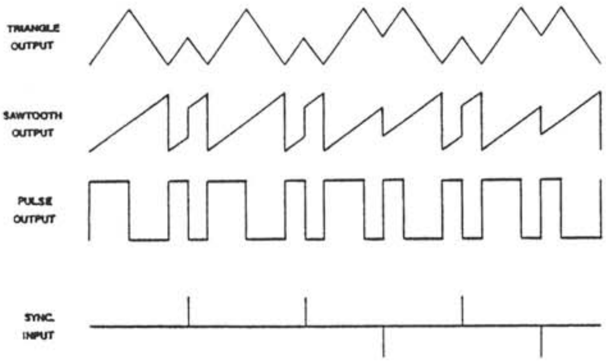


FIGURE 4. HARD SYNCHRONISATION

result that the oscillation period is an integral multiple of the pulse period. If this input is not in use it should be by-passed to ground with a 100nF capacitor, C8A, so as to prevent unwant synchronisation or waveform instability from noise pulses in the positive supply line. Pin 6 is used for hard synchronisation and R24, D1 and C9 allow synchronisation from rising edges while R25, D2 and C10 allow synchronisation from falling edges. A positive sync. pulse will cause the triangle wave to reverse direction only during the rising portion of the triangle, whereas a negative sync. pulse will cause direction reversal only during the falling portion. Figure 4 illustrates the hard synchronisation capabilities of the CEM 3340 and as employed in the 80-3 module.

One, or more, VCLFO's in a synthesiser allow some unusual effects and reference should be made to 'USING THE DIGISOUND 80 MODULAR SYNTHESISER'.

3. CONSTRUCTION

The 80-3 PCB is the same as that used for the 80-2 oscillator and is printed with a component overlay which simplifies the construction stage. The overlay is reproduced in Figure 5 to allow checking of component placement after the module has been constructed.

Take special care regarding orientation of the IC's. Even after installing the DIL sockets the number '1', denoting pin 1, will still be visible on the PCB. In any event compare the completed PCB against Figure 5 before applying power. For the diodes a line is printed on the PCB, either above or below the 'D' number and this line corresponds with the band on the diode denoting the cathode.

Wiring of potentiometers and other panel connections to the PCB are shown in Figure 6. This diagram illustrates the components when viewed from the rear of the panel. The arrows and associated letters indicate that a wire connection must be made from the position shown to the PCB. The latter has a connecting point on its front edge with letters corresponding to those shown in Figure 6. Note the location of C8A on the soft synchroni-sation socket. Unlike most other DIGISOUND 80 modules it will be found most convenient to mount the PCB in such a way that the power connecting point is at the top of the panel. This reduces the length of most wires and also gives ready access to the frequency trimmers by simply tilting the panel forward from its housing.

The jack sockets in the diagram are of the type supplied by Digisound Limited. The top connection, as illustrated, is the connection which is made with the jack when the latter is inserted. The lower connection is disabled by insertion of a jack plug. Finally, the tab under the socket is the ground connection. It is recommended that all of these ground connections are wired to the 0V line since this facilitates connection of the VCO to other equipment which may be using a separate power supply. The ground tabs may be joined together using tinned copper wire but other panel wiring should be made with insulated wire. 1/0.6mm insulated wire is ideal for panel wiring since it retains any shaping and thus allows a neat appearance to be obtained. Wires should be kept as short as practical.

One special point to note is that cleanliness of the PCB is particularly important in the area around the timing capacitor, C7. At low frequencies the current to this capacitor is only a few nanoamps and so residual solder flux or other dirt around the capacitor may degrade performance. Use either a proprietary PCB solvent cleaner or else carefully scrape clean the foil side of the PCB around the solder points for C7.

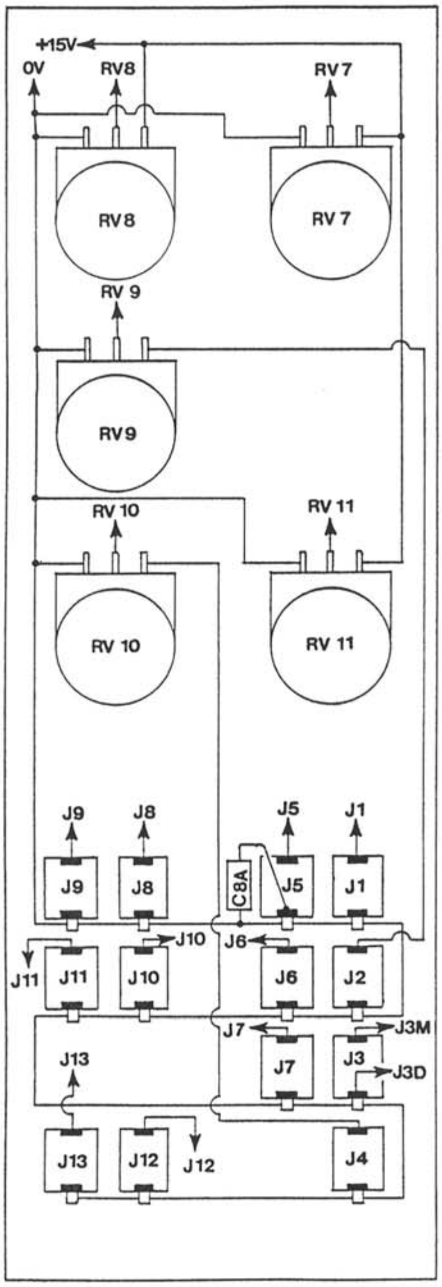


FIGURE 6. PANEL WIRING

On completion of the construction stage carefully examine the underside of the PCB to ensure that all components and connections are properly soldered and that no solder bridges have been formed. Also check polarity of power supplies and that a good ground (0V) connection is present before applying the power to the module. For example, if the recommended CHIRI connector is being used then check voltage at its pins using 0V as the ground for the meter. Do not have the power turned on when the connector is mated with the CHIRI plug on the PCB. A further precaution is to have the VCO connected to an oscilloscope or an amplifier when it is powered for the first time. For the latter connect the +10V triangle to an amplifier, with its 'volume' control nearly off, and apply power to the module. If there is no response then switch off immediately and repeat all of the checks mentioned above. If functioning then check the other waveform outputs although remember that the sinewave has not been trimmed at this time.

After constructing the module you will be anxious to try it out. Do not, however, skip the tedious checking procedure since mistakes can be expensive.

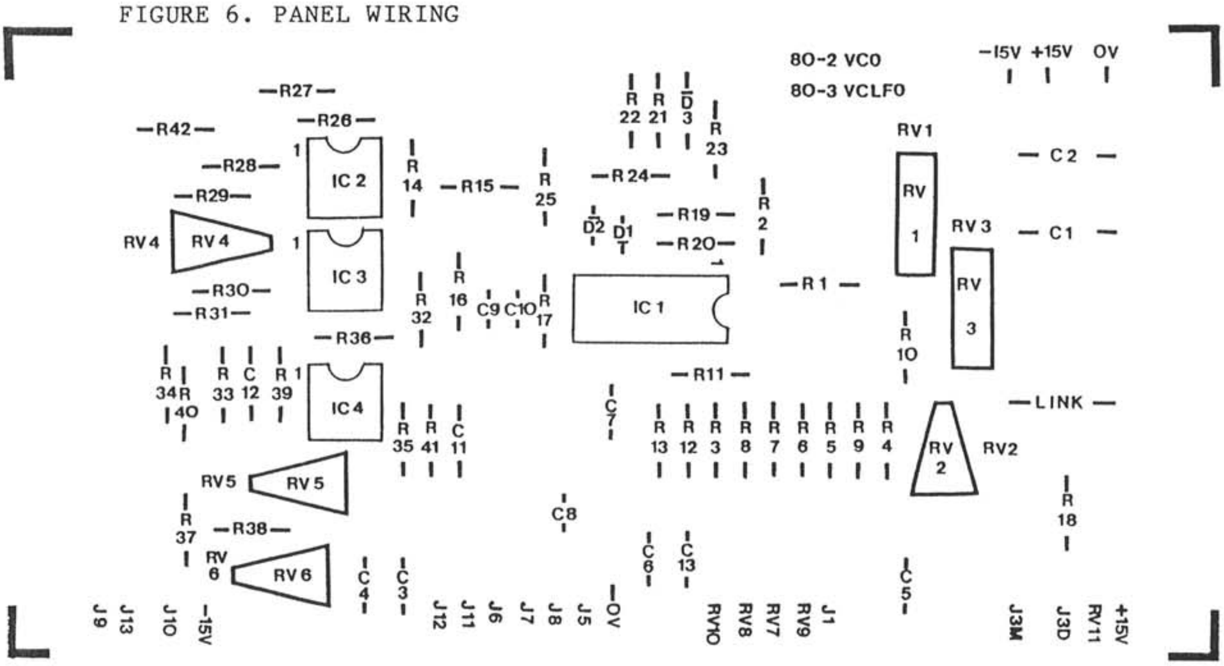


FIGURE 5. COMPONENT OVERLAY FOR 80-3

4. CALIBRATION

The first step is to adjust the sinewave outputs and trimmers RV4 to RV6 should be set to their mid position. With an oscilloscope, or by ear via an amplifier, adjust RV4 then RV5 for purest sinewave output. These adjustments may have to be repeated a few times to obtain the best result. Next adjust RV6 to get the +10V sine output referenced to ground. The simplest method is with a DC coupled oscilloscope and RV6 is adjusted such that the bottom of the sinewave is at 0V. An alternative method is to use a voltage controlled amplifier (VCA), such as the DIGISOUND 80-9. Connect a signal to the VCA and insert a jack plug into the exponential control input and while listening to the output, using an audio amplifier, adjust signal level or amplifier volume such that no sound is heard. Now connect the +10V sinewave to the VCA and then adjust RV6 until no sound is heard. For the latter step the VCLFO should be at a fairly low frequency, say, 1 to 5Hz. If a VCA not available at this stage then this step may be left until such times as suitable equipment is available. One should remember, however, that it is not trimmed if it is used in a patch.

The last and most important step is to calibrate the oscillator to the 1V/octave relationship. Before starting first check the voltage from the power supplies and trim to +/-15V, if necessary. Calibration is greatly simplified if one has available: a variable voltage source (from a potentiometer or from a calibrated keyboard); an accurate voltmeter; and a digital frequency meter. There are, however, a number of other methods. One is to use a previously calibrated oscillator or a musical instrument and make the calibration using the beat frequency technique. Another approach is to build two simple, but stable, fixed frquency oscillators and use them in conjunction with an oscilloscope to calibrate the VCO by generating Lissajous figures.

Whichever method is used the calibration procedure is as follows. Set the wiper of RV3 at the grounded end by turning the adjusting screw clockwise until clicks are heard (or the wiper seen if a multiturn with a transparent cover is supplied) and the wiper of RV1 to about mid position. Turn both RV7 and RV8 fully anti-clockwise and ensure that they remain in this position during the calibration stage. Next apply a positive voltage to Control Input 1 (R5) until the frequency is about 20Hz. Increase voltage by exactly one volt (as accurately as can be measured) and adjust RV1 until the frequency is double that of the first frequency. These steps may have to be repeated several times in order to achieve an exact doubling of frequency per volt applied. If using a digital frequency meter then the best approach is to note the first measurement, increase voltage by one volt and divide the second reading by the first. If the ratio is greater than 2 then turn RV1 anti-clockwise and if less than 2 then turn it clockwise. Now decrease the voltage by exactly one volt and calculate the new ratio and adjust RV1 as before. Continue the procedure until a ratio of exactly 2 is obtained and during the calibration it may be necessary to alter the voltage level such that the calibration is carried out in the general range 15 to 50Hz. If using the beat frequency technique these low frequencies are not very convenient and may be increased slightly with little loss in calibration accuracy.

Repeat the above procedure except for starting at an initial frequency of about 500Hz and adjusting RV3 until a doubling of this frequency is obtained when the applied voltage is increased by exactly one volt. Note that this is a fine adjustment and if the low frequency calibration has not been carried out accurately then it may be found impossible to carry out this step effectively. If accurate calibration is not possible at this stage then set the wiper of RV3 to its mid position. If the high frequency trim has been carried out then re-check the low frequency calibration and the performance of the VCO over its specified range.

Two important things to note are: (a) that the VCLFO has been calibrated via Control Input 1 and this will remain the most accurate input; (b) Resistors and other components will age and their change may be quite significant for a period following soldering. If convenient the best approach is to make a quick initial calibration and then a more accurate one after using, or keeping the module powered up, for several hours.

The oscillator may now be adjusted to some initial frequency and with RV7 and RV8 fully anti-clockwise a suitable value would be 0.2Hz. The actual value is not very important but it is useful to know the 'no voltage' condition.

5. MODIFICATIONS

Some users may prefer to have the VCLFO set to some other scale, for example, a much wider manual control over lower frequencies. The easiest method is to replace the 10nF timing capacitor, C7, to 100nF. This step will reduce the scale by a factor of ten and allows manual adjustment (with RV7 and RV8) over a range of 0.02Hz, which is one cycle every 50 seconds, to 41Hz. Application of positive control voltages will allow the range to be extended to about 500Hz. The timing capacitor must be a low leakage type (silver mica, polystyrene or polycarbonate) for reliable results.

Another alternative is to wire RV7 between +/- 15V supplies, as in the 80-2, which theoretically would provide a range from this potentiometer of between 0.0002Hz and 205Hz. The lower value, however, exceeds the guaranteed range of the CEM 3340 with a 10nF capacitor and there is a chance that the oscillator will cut off below about 0.01Hz. Furthermore the scale of RV7 is no longer in octaves. To avoid this situation R8 should be replaced by a 300k resistor which will then give RV7 a range of 0.006Hz (close to specified lower limit) up to 64Hz. If RV7 is wired between the +/-15V supplies then the wire to its wiper (middle pin of potentiometer) should be disconnected during the calibration stage.

6. COMPONENTS

RESISTORS, 5%, 1/4w carbon film	
R8	1M5
R9,13	470R
R12	1M0
R14,15,26,29,33,36,39	100k
R16,17,40,41,42	1k0
R18,19,20,30,35	47k
R21,24,25	10k
R22	2k0
R23	910R
R27	300k
R28	27k
R31,34	2k2
R32,37	270k
R38	24k

RESISTORS, 1%, 1/4w metal film, 100ppm	
R1	24k
R2	5k6
R3	1k8
R4	470k
R5,6	100k
R7	150k
R10	1M0
R11*	1M5

*may be replaced by low TC carbon film resistor.

CAPACITORS	
C1,2	470n polyester
C3,4,13	100n polyester
C5,6	10n polyester
C7	10n polycarbonate
C8,9,10	1n0 polyester
C8A (see text)	100n polyester
C11,12	22p polystyrene

POTENTIOMETERS, PRESETS	
RV1,3	10k cermet multiturn
RV2	1M0 cermet
RV4,6	100k carbon
RV5	1M0 carbon
RV7,8,9,10,11	100k lin.

SEMICONDUCTORS	
IC1	CEM 3340
IC2	LM 1458*
IC3	CA 3080E*
IC4	TL 082
D1,2	1N4148
D3	BZY88 10V

*or equivalent

MODULE 80-4
VOLTAGE CONTROLLED MIXER

1. INTRODUCTION

The DIGISOUND 80-4 Voltage Controlled Mixer (VCM) provides proportional (linear) mixing of up to four input signals and has a master gain control so as to be able to maintain its output at a level compatible with other modules in the series. The peak output is visually indicated by a LED. The combined output may be panned between two outputs termed 'left' and 'right'.

The design is based on three CEM 3330 Dual Voltage Controlled Amplifiers in order to facilitate voltage control of the functions of the VCM as well as ensuring exceptionally low noise.

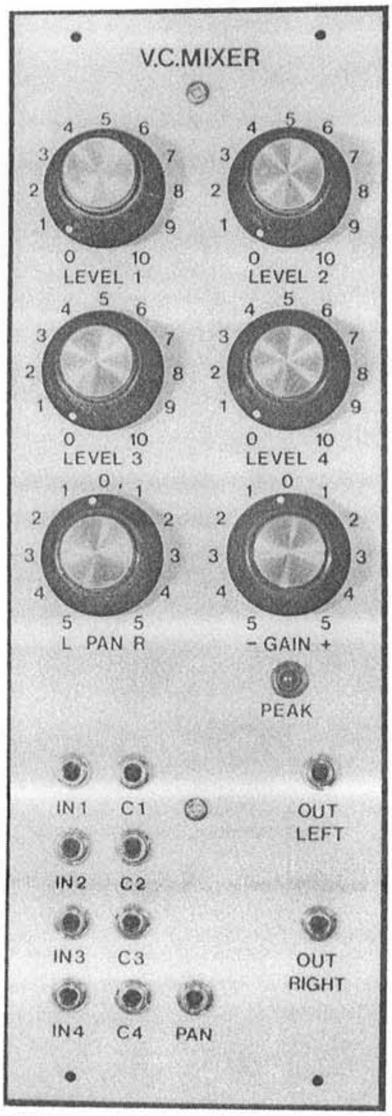


FIGURE 1. 80-4 PANEL

Figure 1 illustrates the panel lay-out of the VCM. The four signal inputs are marked 'IN 1' to 'IN 4' and the input signals may be up to +/-10V. The module is DC coupled. The signals may be mixed manually using the potentiometers marked 'LEVEL 1' to 'LEVEL 4' and their action is linear, that is, when set mid-way the signal level to the mixing stage is 50%. Additionally, however, each input may be varied by an external control voltage applied to the inputs marked 'C1' to 'C4' and a voltage of 0 to +10V will vary the level between 0 and 100%. In effect, therefore, the four channels respond in the same manner as four independent VCA's (voltage controlled amplifiers) and the control source may be low frequency oscillators for modulating any, or all, of the inputs or one may use envelope generators to switch channels on and off in a controlled manner. This makes the 80-4 VCM an extremely versatile module - see Section 4. After mixing, the 'GAIN' control allows the gain of the combined signals to be increased by a factor 5 or attentuated by a factor of 6.25. The 'PEAK' LED will iilluminate when the combined signal going to the final stage reaches about +10V. The combined signal may be manually panned between the left and right outputs. The signal may also be panned automatically by applying a +10V control signal to the 'PAN' socket. If only one output channel is used then the output of the mixer may be faded manually or automatically by using these pan controls.

These construction notes relate to PCB's marked DIGISOUND 80-4A which has some slight modifications compared to the original 80-4. The main change is an improved method of compensating the linear control inputs of the CEM 3330 although this does not affect performance of the original version. The other changes may be applied to the original version and are passive component changes to: (1) improve voltage control feedthrough when modulating the inputs; and (2) an increase in the gain factor to allow for attenuation of filtered signals.

2. DESIGN

The CEM 3330 contains two voltage controlled amplifiers each of which consists of a variable gain cell and a log converter. The gain cell is the current-in, current-out type and has simultaneous linear and exponential controls. The log converter generates the logarithm of the linear control input current while the exponential control input is transmitted unchanged to its output.

Reference to the circuit diagram shown in Figure 3 and pins 1 to 9 of IC1 (refer also to the block diagram of the CEM 3330 in Figure 2) illustrate the basic principle of the design as well as some of the features of the CEM 3330. The signal input (pin 4) is a summing node and can, therefore, accept multiple inputs. In the VCM application where we require independent control over each input only one input has been provided and with R2 = 100k the signal level should be kept to within +/-10V. R3 and C7 are compensation components and the diode, D1, is to avoid latch-up. R1 connected to +15V provides a reference current to the gain cell and this current should be limited to 100uA for best linearity. The design is based on proportional mixing of up to four signals and thus the linear control input is used to independently control the gain of each signal input. Again this is a summing node input at pin 7 which allows manual control of gain via RV1 and R6 or external control via R5 without using additional summing stages. By using a 150k resistor for R6 the control pot can be wired to the +15V supply and provide the same gain as a +10V external control signal applied to the 100k resistor, R5. C8 is provided to stabilise the log converter. This latter method of compensation should be used in place of the 1k0 and 10nF network (at pin 7) shown in the data sheet for the CEM 3330 although in this application the previous method is satisfactory. C9 is for compensation of the gain cell. A master gain control is obtained by injecting a small voltage into the exponential control input (pin 6). This voltage is derived from RV5, R31 and R32 and is common to the four input stages.

For very critical applications the CEM 3330 may be trimmed for lowest

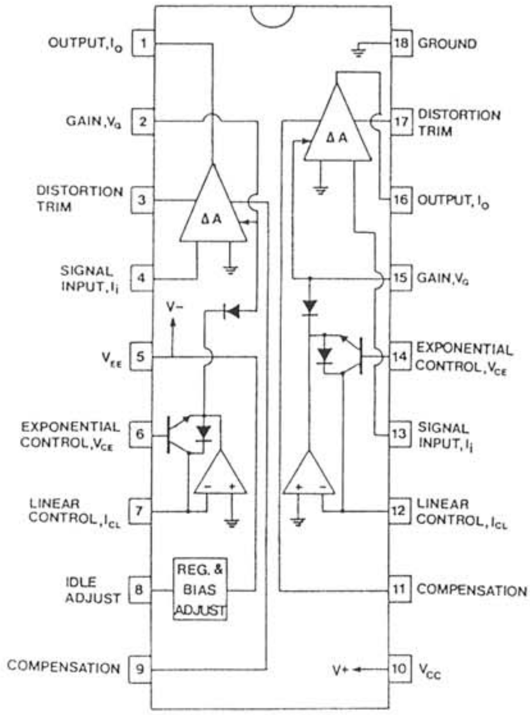


FIGURE 2. CEM 3330 IC

distortion and control voltage feedthrough. Without trimming these two factors are opposing one another but for a mixer application a good balance may be achieved by varying the operating point of the amplifiers. This is a unique feature of this VCA and is achieved by varying the quiescent standby current of the signal carrying transistors by placing a resistor between the I_{EE} pin (pin 5) and the idle current adjust pin (pin 8). In this application the amplifiers are run Class B with a 15k resistor (R4) providing a standby current of about 3uA. The CEM 3330 requires a current limiting resistor when operated from negative supplies greater than -7V5 and this is provided by R7 which is the correct value for a -15V supply with the idle current employed.

The four signal input (and control stages) are identical and their output currents are summed at IC3 and converted to a voltage across R30. This voltage is applied to TR1 which is turned on when the peak output voltage is about 9V5, which is set by the voltage divider R25 and R26. TR2 is also turned on when this peak voltage is reached and the LED (D7) will then light up. At constant amplitude high frequency the LED will tend to glow dimly while intermittent peak voltages are clearly indicated.

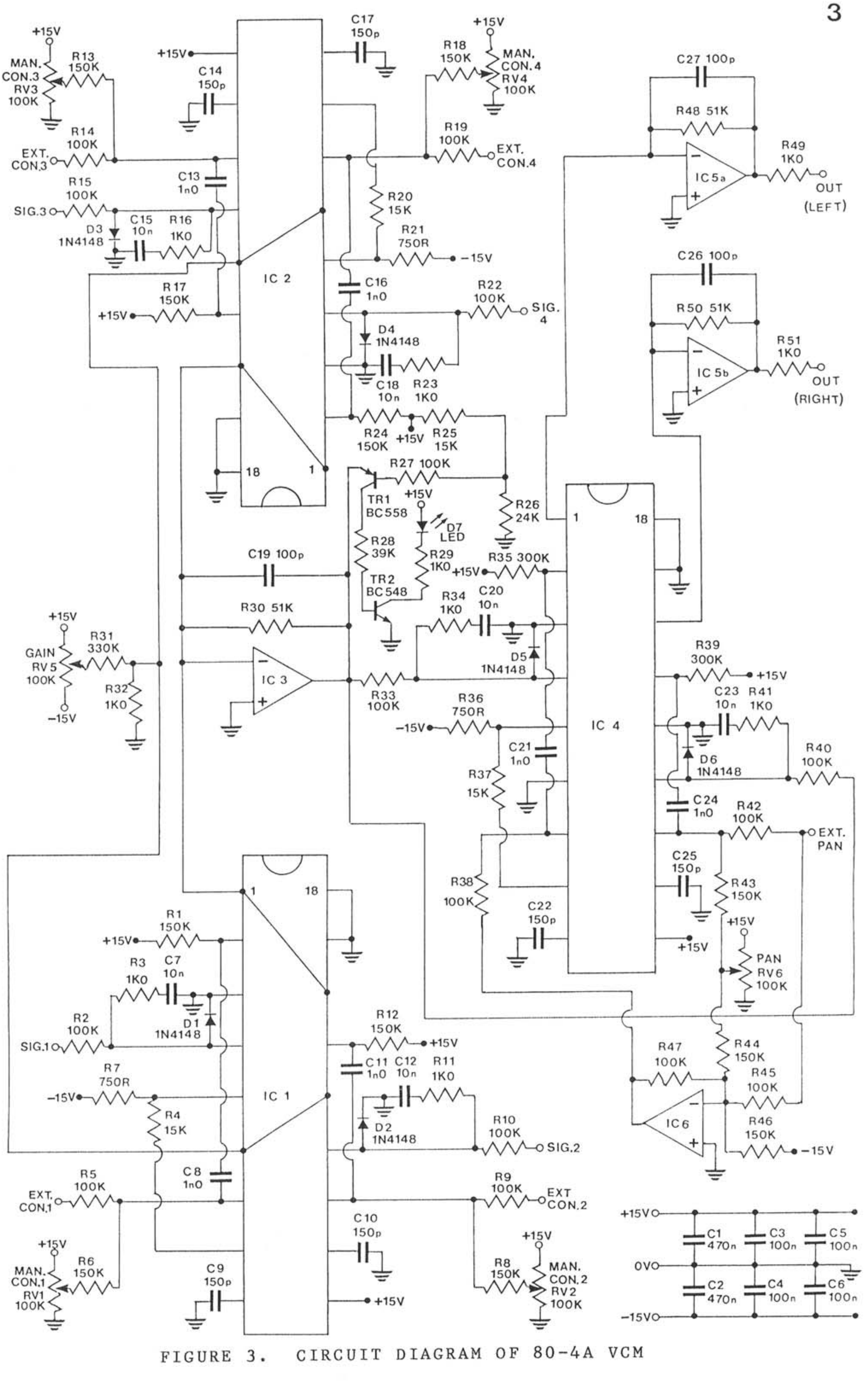


FIGURE 3. CIRCUIT DIAGRAM OF 80-4A VCM

The output voltage from IC3 also goes to both VCA's in IC4 which is configured in a similar manner to IC's 1 and 2 except that their exponential inputs (pins 6 and 14) are grounded. The amplifiers and associated op amps (IC5a and IC5b) are set to unity gain when a +10V control voltage is applied to R38 or R42. The panning effect is obtained through IC6 and associated components which provide a +10V output with zero volts at R42, or when RV6 is fully clockwise. Thus when using a +10V control voltage for panning then RV6 should be fully anti-clockwise. The left and right outputs are obtained by converting the current to a voltage across resistors R48 and R50 respectively. The use of IC5 provides low impedance outputs.

3. CONSTRUCTION

The 80-4A PCB is printed with a component overlay which aids the construction stage. The overlay is reproduced in Figure 4 to allow checking of component placement after the module has been completed.

The usual approach to soldering components onto a PCB is selecting them by increasing height. For example, start with wire links, followed by resistors then DIL sockets and so on. This approach is recommended with the 80-4A and after the DIL sockets have been installed it is worth inserting the IC's since the close proximity of capacitors to the sockets makes their insertion more difficult unless an IC insertion tool is used. The difficulty is aggravated by the fact that most IC's are supplied with their pins splayed outwards. Once the pins are vertical then there is little difficulty in inserting and removing the IC's and so if required the IC's may be removed again while the rest of the PCB is being constructed.

Take particular care with the orientation of the IC's, the diodes and the transistors. For the IC's note that IC2 is orientated opposite to that of all other IC's. The reason for this is evident from the circuit diagram which illustrates that this orientation makes the current paths from IC1 and IC2 to IC3 shortest so avoiding noise pick-up. Even after the DIL sockets have been installed the number '1', denoting pin 1, will still be visible on the PCB. For the diodes a band (line) is shown on the PCB overlay either before or after the diode number and this band indicates that the band on the diode, denoting its cathode, should be towards the hole nearest the band on the PCB. The transistors should be orientated in accordance with the shape printed on the overlay. In any event compare the completed PCB against Figure 4 before applying power to the module. Also before powering up check that the five

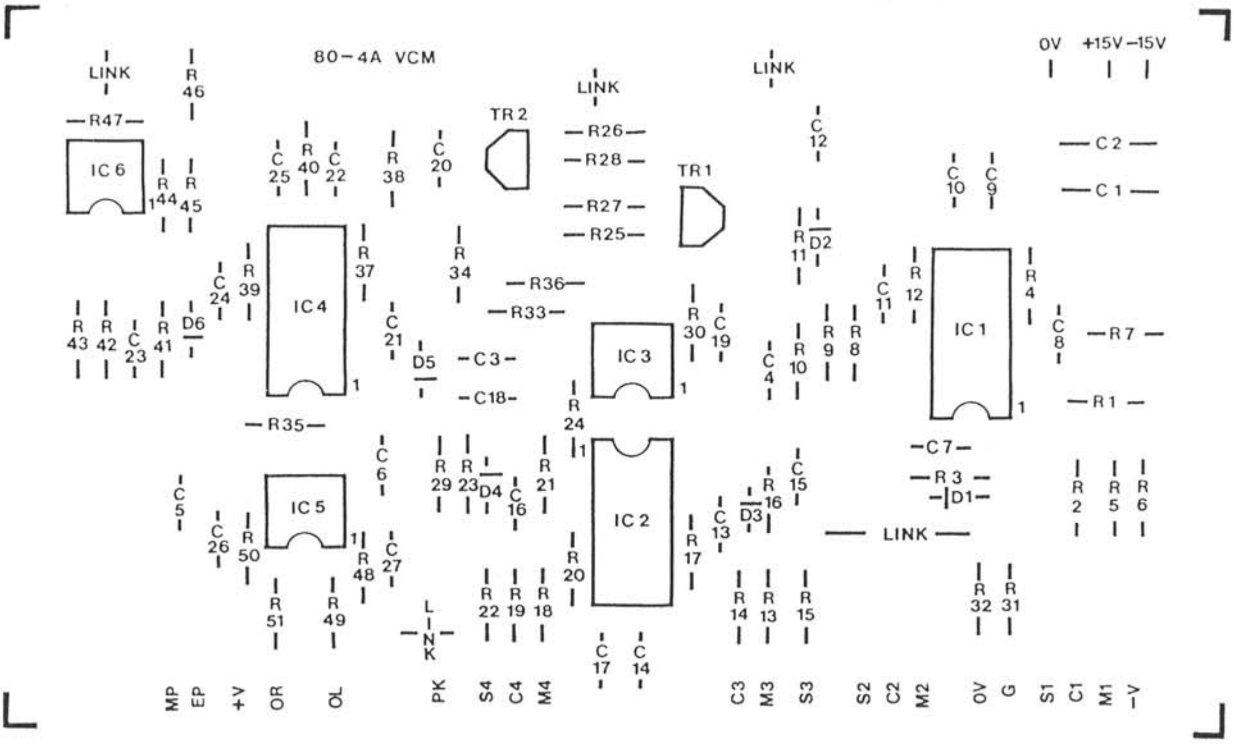


FIGURE 4. 80-4A COMPONENT OVERLAY

wire links have been made (uninsulated wire may be used for these) and inspect the foil side of the PCB for solder bridges and suspect joints.

The panel wiring is illustrated in Figure 5 and this diagram illustrates the components when viewed from the rear of the panel. The arrows and associated letters indicate that a wire connection must be made from the position shown to the front edge of the PCB which has corresponding letters. The long lead of the LED is connected to the +15V line.

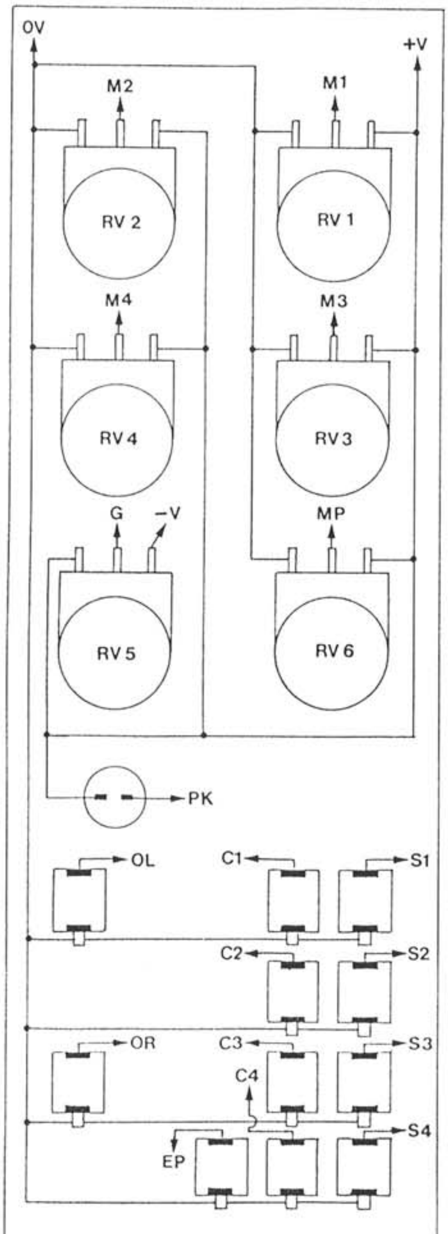


FIGURE 5. PANEL WIRING

The jack sockets illustrated in Figure 5 are of the type supplied by Digisound Limited. The top connector, as shown, is the connection which is made with the jack plug when the latter is inserted. The lower connection is disabled by insertion of a jack plug. Finally, the tab under the socket is the ground connection. It is recommended that all of these ground connections are wired together and taken to the 0V connection on the PCB since this facilitates connection of the module to other equipment which may be operating from a separate power supply. The ground tabs may be soldered together using tinned copper wire but other panel wiring should be made with insulated wire. 1/0.6mm insulated wire is ideal for panel wiring since it retains any shaping and so allows a neat appearance to be obtained. Wires between panel components and PCB should be kept as short as possible.

No adjustments are needed for the 80-4 VCM and after carefully checking construction the unit simply requires to be connected to a +/-15V supply.

4. USING

The DIGISOUND 80-4 Voltage Controlled Mixer is extremely versatile as will be evident from study of the 'Using the Digisound 80 Modular Synthesiser' manual. To give a taste of its applications some of the uses listed in the original constuction article are given below.

A. A simplistic view would be to consider the mixer as four voltage controlled amplifiers with a common output. One technique often applied to a VCA is amplitude modulation (tremolo). Usually, however, the VCA is one of the last stages and if a number of signals have been combined in a conventional mixer prior to the VCA then the total signal has to be amplitude modulated. Using the VCM one may choose which signals are to be modulated and this selected modulation can be far more pleasing.

B. One of the early works with a synthesiser was Morton Subotnick's "The Wild Bull", recorded in 1968. In this work extensive use is made of a sawtooth waveform which is separated into four octave bands (two 80-16 modules would allow this) to provide signals for a four-channel voltage controlled mixer. Each channel was controlled by an ADSR envelope generator gated from a sequencer. This arrangement allows the separate timbral characteristics of any sound to be independently treated. Furthermore by varying the speed of the sequencer the characteristics of the sound can be made to vary widely, for example, as the rate is increased the four bands begin to sound simultaneously. Only a simple digital sequencer is required for the above and the VCM becomes the heart of a useful music making instrument within the body of the synthesiser.

C. The 80-4 is very suitable for exploring both additive and subtractive synthesis. For example, in the simplest of cases, the addition or subtraction of the sinewave output from an 80-2 VCO to another waveform in order to boost or remove (reduce) the fundamental.

D. Another useful application of the VCM is to alter loudness and harmonic content in relation to pitch. One of the criticisms of 'live' electronic music is the precise nature of its sounds and the initial excitement of a 'new' sound turns to boredom as the brain reacts adversely to its repetitive nature. By applying the keyboard control voltage (or its inverse, or a proportion of either) to one or more of the mixer control inputs then the amplitude or harmonic content (often both) will vary with pitch and so provide a useful means of dynamically altering the timbral characteristics of the sound.

E. Applying a low frequency waveform to the pan control input can produce some interesting effects but for greatest impact this technique should be used sparingly.

As inferred earlier more information on most of these techniques will be found in the User's Manual.

5. COMPONENTS

RESISTORS, 5%, 1/4w, carbon film

R2,5,9,10,14,15,19,22,27,33,40	100k
R3,11,16,23,29,32,34,41,49,51	1k0
R4,20,25,37	15k
R6,8,13,18	150k
R7,21,36	750R
R26	24k
R28	39k
R30,48,50	51k
R31	330k

RESISTORS, 1%, 1/4w, metal film

R1,12,17,24,43,44,46	150k
R35,39	300k
R38,42,45,47	100k

POTENTIOMETERS

RV1,2,3,4,5,6	100k lin.

CAPACITORS

C1,2	470n polyester
C3,4,5,6	100n polyester
C7,12,15,18,20,23	10n polyester
C8,11,13,16,21,24	1n0 polycarbonate*
C9,10,14,17,22,25	150p polycarbonate*
C19,26,27	100p polycarbonate*

* may be replaced by axial polystyrene

SEMICONDUCTORS

IC1,2,4	CEM 3330
IC3	TL 081
IC5	TL 082
IC6	LM 741
TR1	BC 558
TR2	BC 548
D1,2,3,4,5,6	1N4148
D7	5mm Red LED

PROCESSOR 80-5

1.SPECIFICATION

LAG PROCESSOR. EXPONENTIAL DELAY TYPE.
INVERTERS. 2 OFF WITH ATTENUATION.
ATTENUATORS. 4 OFF, WHEN INVERTERS NOT IN USE OR 2 WITH INVERTERS.
POWER REQUIREMENTS: ±15V @ 9mA per rail.

2.APPLICATION

.1 DIGISOUND synthesiser modules have a high input impedance (normally 100k) and a low output impedance (normally 1k). This combin--ation allows one output to be used as a control for, or signal into, several other modules without loading problems. To conserve both panel space and reduce cost this multiple distribution is best accomplished by using the 'Processor' module. In its simplest application one output may be distributed to four inputs, or when two outputs are provided (common practice) to eight other modules. There are four channels on the 'Processor' with this 1 to 4 capability.

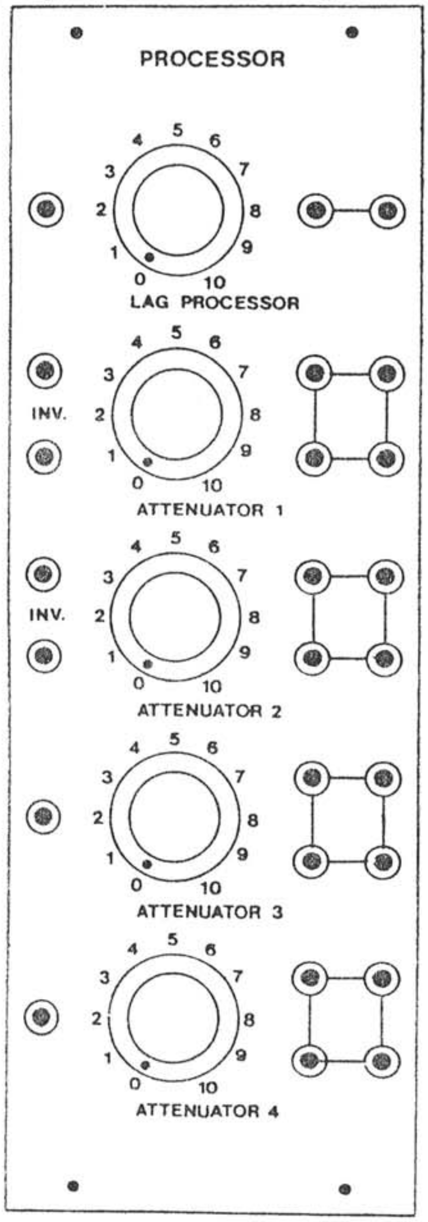


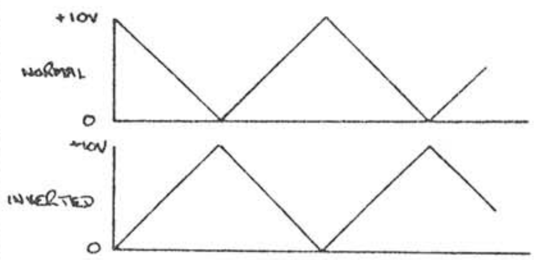
FIGURE 1. PROCESSOR FRONT PANEL

.2 Each of the four distribution channels has an attenuating potentiometer for adjust--ing the level of the output voltage.

.3 Very often, however, one does not wish to have the same level of control voltage to other modules - which would happen if an attenuator was placed on the outputs of the modules. As a simple illustration, assume that an 80-3 VCLFO is being used to modulate a number of 80-2 VCO's. One may wish to vary the depth of modulation to each VCO and the 'Processor' allows this.

.4 There are many useful effects in synthesis which may be obtained by having a control voltage that is simultaneously increasing and decreasing, for example, panning effects. Attenuators 1 and 2 thus have an 'inverting' input which converts a 0 to +10V signal into a +10V to 0V signal. These outputs may be attenuated if required.

.5 The inverters also result in phase invers--ion as illustrated below -



This effect may be used for split phase trem--elo in combination with the 80-9 Dual VCA module.

.6 A so-called 'lag processor' is also incl--uded in 80-5. Essentially this is a crude low pass filter and similar to a portamento circuit. Its main purpose is to slow down control signals, or slew them so that the time taken to reach peak voltage is increased. One application, again with the Dual VCA, would be to take the VCLFO output direct to one channel of the VCA for tremelo and then take the out-put into the second channel of the VCA and modulate it again but this time using the VCLFO signal after it has been delayed by the lag processor. This unit may also be used for smoothing of signals.

3.CONSTRUCTION

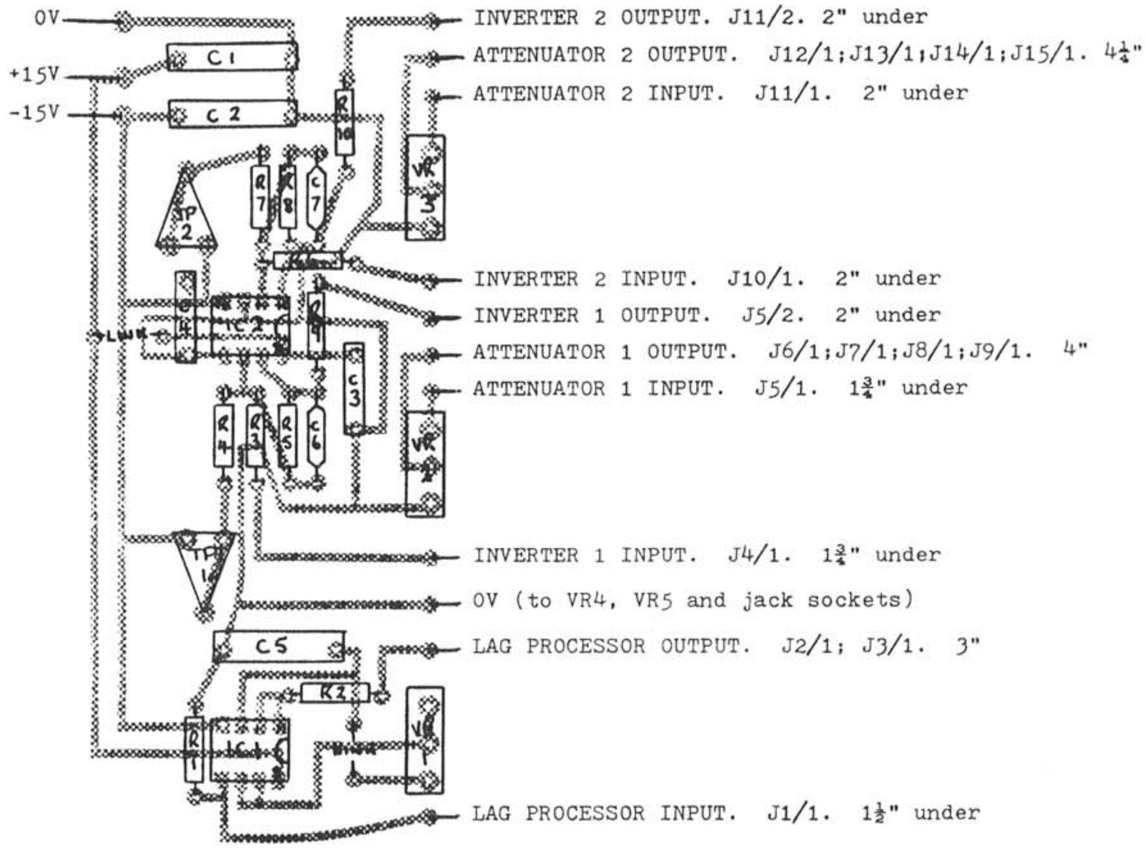
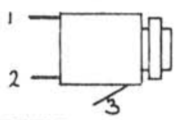


FIGURE 2. COMPONENT OVERLAY AND WIRING CONNECTIONS.

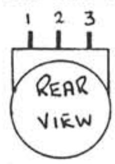
.1 GENERAL

a) WIRE: For wiring between the PCB and panel hardware we recommend a solid wire, such as 1/0.6mm. This is sufficiently rigid to allow neat placement of wires and also to allow bare wire to be used for connecting up a number of commoned jack sockets. Colour coding of wires (3 X power lines; control inputs; signal inputs; and outputs) will aid any fault finding later (if necessary!).

b) JACK SOCKETS: Connecting points will be numbered as shown below. (1) connects with jack plug; (2) connects with (1) with jack plug removed; (3) is grounded - see general construction notes provided with 80 series modules. Thus 'to J2/2' means connect to Jack Socket 2, connecting point (2).



c) POTENTIOMETERS: Potentiometer connections are numbered 1,2 and 3 as shown and when viewed from the rear (normal situation during construction). Thus 'to VR2/1' means to potentiometer VR2 and connecting pin 1.



.2 80-5 MODULE

a) The following components are supplied.-

RESISTORS, ¼w, 5% (gold band) carbon film

R 1, 3, 5, 6, 8	100k (br,b,y)
R 2, 9, 10	1k0 (br,b,r)
R 4, 7	130k (br,o,y)

(br=brown;b=black;y=yellow;r=red;o=orange)

POTENTIOMETERS

VR 1	2M2 log.
VR 2, 3, 4, 5	100k lin.

TRIMMERS

TP 1, 2	47k carbon

CAPACITORS

C1, 2	470nF(0.47mfd) polyester
C3, 4	100nF(0.1mfd) polyester
C5	220nF(0.22mfd) polyester
C6, 7	22pF polystyrene

SEMICONDUCTORS

IC 1 1458 - 8pin

IC 2 TL 082CP, or equivalent

MISCELLANEOUS

2 X 8 pin DIL sockets; 3 extra nuts for VR1, 2 and 3; PCB.

b) Refer to the component overlay shown in Figure 2. Solder in the two wire links and then the remaining components except VR 1,2 and 3. Since there is little working space behind the panel after insertion of the PCB the wire connections to the PCB should be soldered in place next. In Figure 2 a guide to the wire length required is given based on an allowance of ¼" for connecting to PCB; ¾" for connecting to two jack sockets; and 2" when four sockets are to be connected. The word 'under' also appears after some wire lengths which indicates that these wires should be bent under the PCB (towards the foil side) prior to fixing to the panel. Finally place the extra nut provided (for spacing) to VR 1, 2 and 3 and solder these pots to the PCB. Check component placement and inspect the underside of the PCB to ensure that all conn--ections have been properly made and that no solder bridges have formed between the tracks.

c) Mount all panel components and wire up the Attenuators VR 4 and VR 5 (refer to the general construction notes for the 80 series if in doubt). The ground (OV) wire may be omitted at this time. Insert VR 1 to VR 3 with the PCB and wire up using the directions given in Figure 2 and the rear view of the panel shown in Figure 3. Connect a wire to the OV point on the PCB and take to both VR 4 and VR 5 (pin 1 in each case) and also to the ground point of the jack sockets.

d) Connect ±15V power supplies to the module (do not switch on); set TP 1 and TP 2 to their mid positions; connect a voltmeter to the output of Attenuator 1 (J6 to J9) ; turn VR 2 fully clockwise; switch power on and note the voltmeter is showing a reading of about +8 to +12V. If not, switch off and quickly check whether IC 2 is hot and then re-check wiring and component placement, including IC orientation if IC 2 was hot. When satisfactory check output of Attenuator 2 in the same way. To check functioning of the lag processor connect a low frequency waveform to the input and take the output to an amplifier (or oscilloscope). Gradually rotate VR 1 and note difference in tone (or waveshape).

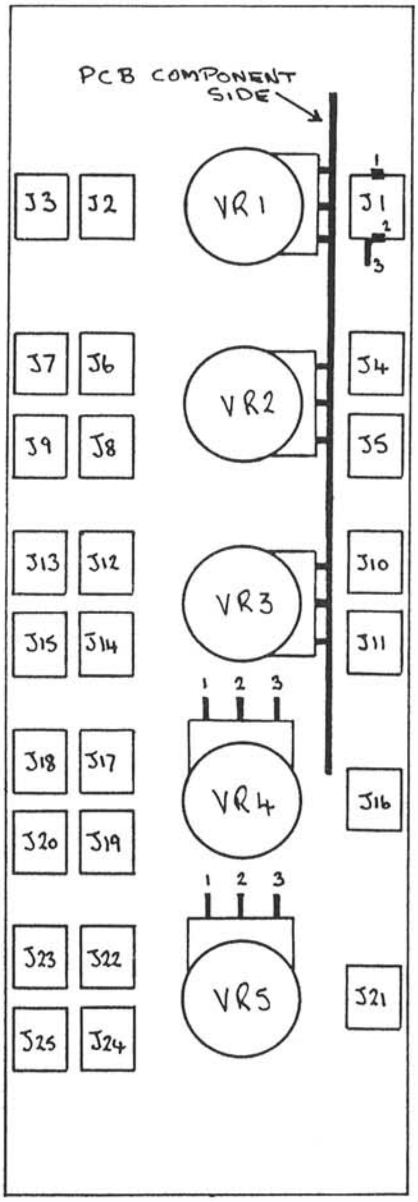


FIGURE 3. REAR VIEW OF 80-5 PANEL

4.CALIBRATION

The only calibration required is trimming the two inverters. Set up as described under construction and with VR 2 fully clockwise adjust TP 1 until exactly +10V is obtained at Jack Sockets 6 to 9. Repeat with VR 3 and TP 2 and outputs at J 12 to 15.

5. DESIGN

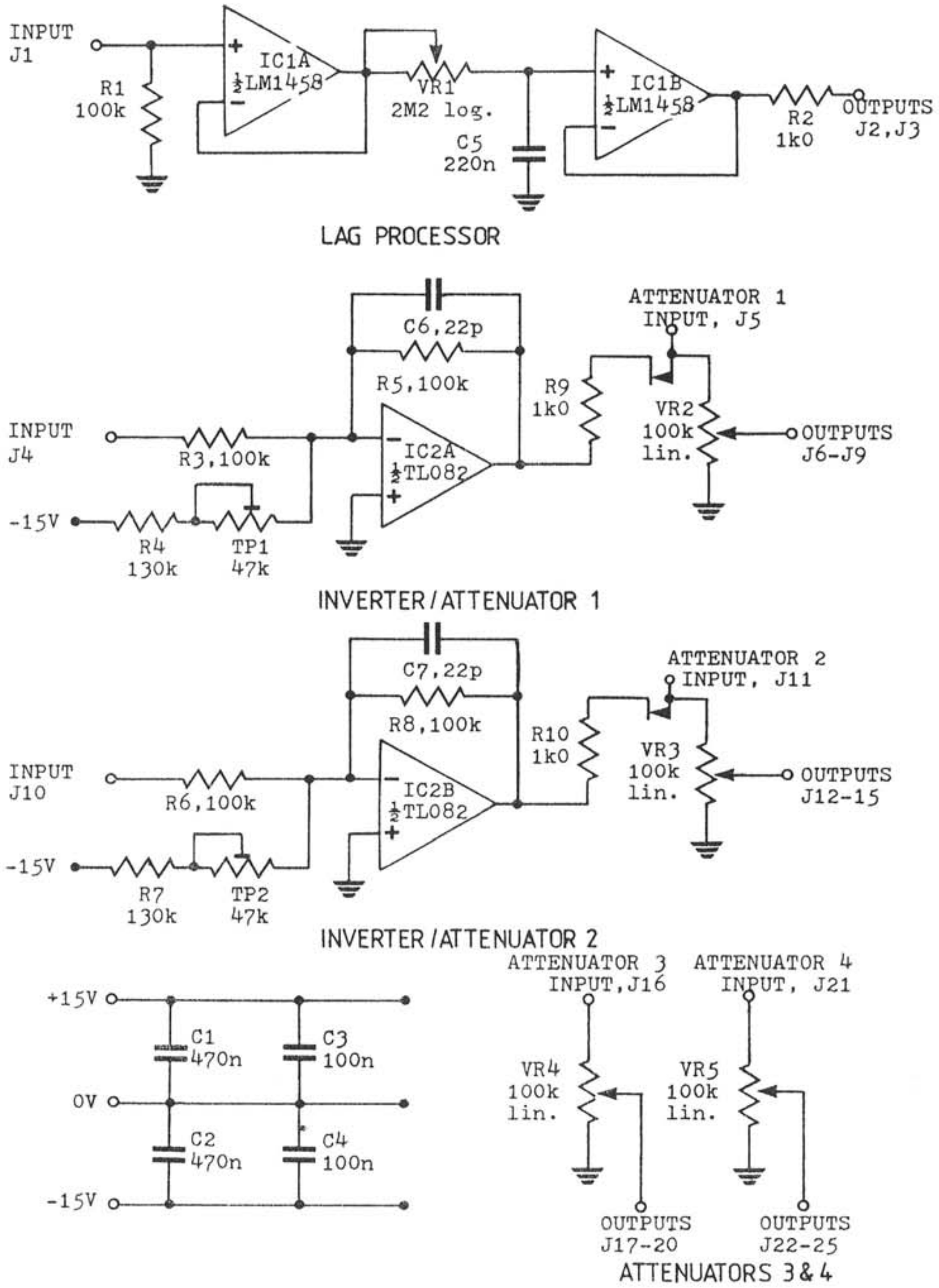


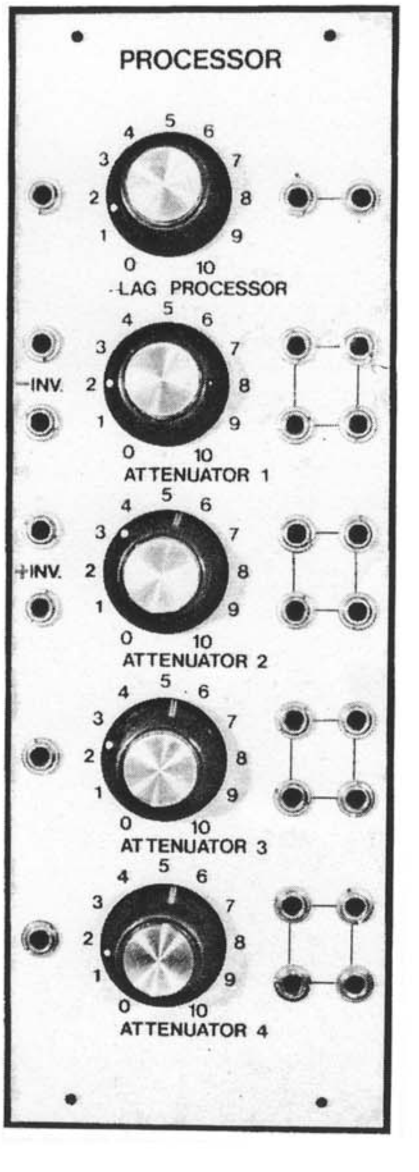
FIGURE 4.80-5 PROCESSOR

LAG PROCESSOR. IC 1A is a voltage follower and VR 1 is used to control the rate at which C5 is charged up. The voltage on C5 at any point in time is available at the output of IC 1B which is configured as a high impedance voltage follower.

INVERTERS. IC 2A (IC 2B) with R3/R5 (R6/R8) is a unity gain inverter and with only these components a voltage of, say, +10V would become -10V at the output. We are, however, injecting -10V into the summing node of the op. amp. via R4 and TP 1 (R7 and TP 2) which means that with 0V into R3 (R6) there is +10V at the output. In this situation +10V into R3 (R6) will result in 0V at the output. The outputs of the inverters may be disabled by a jack plug into J5 (J11) and VR 2 (VR 3) may then be used simply as attenuators without inversion - as is the case with VR 4 and VR 5.

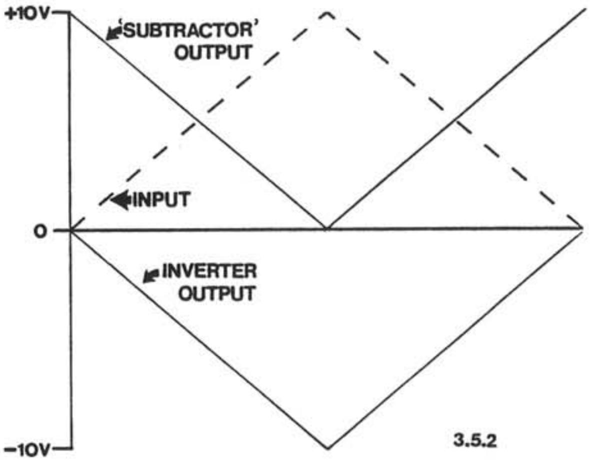
MODULE 80-5 PROCESSOR

The low output impedance and high input impedance of the DIGISOUND 80 modules allow one output to drive several inputs without overloading or introducing appreciable errors. In order, therefore, that a single output from a module may have individually adjustable levels to each of the modules that it is driving we have placed, whenever practical, attenuators on the inputs to modules. This arrangement also facilitates fading in of various effects, for example, if two modules are being modulated from another unit then one of the former two may be faded in and out without affecting the other. It was stated above that the attenuators are placed on the inputs whenever practical and in most cases it is the limitation of panel space which restricts the number that have been included. To overcome this problem of distribution we have included the 80-5 Processor module and this also includes a few other simple functions to aid synthesis.



Distribution of one output to four other inputs, with any level of attenuation on the combined output from the Processor, may be implemented in various combinations. For example, a single Processor may be used to distribute one signal to twelve other modules and with sets of three outputs at different levels of attenuation. For distribution purposes alone the Processor is invaluable and at least one is required for every ten other modules. Two of the distributors may also be used as 'inverters' or as a source of positive voltage for level shifting. To avoid confusion the term 'inverter' will be changed to 'SUBTRACTOR' since as constructed the effect is to subtract the input voltage from +10V. The main control signal in the DIGISOUND 80 is based on a 0 to +10V amplitude and thus if, say, the output of an envelope generator is taken via a 'subtractor' then the attack voltage will start off at +10V and decrease to zero instead of the normal response of going from 0 to +10V. The output from the 'subtractor' has an attenuator and thus the actual voltage excursion may be adjusted to the range desired. These 'subtractors' find wide application in synthesis patching, as is evident in the next section. It should be noted that signals are also inverted in phase when they pass through a 'subtractor'.

Commonly the term 'inverting', especially as applied to operational amplifiers, means that the input voltage is inverted in polarity, e.g., a +5V input becomes a -5V output. There are some patches in synthesis with the DIGISOUND 80 which require this voltage inversion and it is recommended that one of the 'subtractors' is modified to an 'INVERTER'. The latter is simply achieved by removing the 130k resistor and 47k trimmer connected to the inverting input on one side of the op.amp. In addition to other uses the inverter also allows negative DC voltages to be obtained for offset purposes. If this modification is made then we suggest you mark the panel accordingly and probably the simplest way is to prefix the 'INV' by an 'S' or 'P' to denote 'Subtractor-Inverter' or 'Polarity-Inverter' respectively. The two modes are illustrated in Figure 3.5.2.



3.5.2

The other facility included in this module is the 'Lag Processor' which as the name implies is a signal delay device akin to the conventional portamento control on the keyboard. Thus control signals may be made to glide from one step to another and provide a more subtle transition. An example of this effect is a sawtooth waveform from a LFO being used to sweep a voltage controlled module. The sharp transition as the sawtooth reaches its peak voltage can result in an obtrusive 'plop' and the effect may be reduced, or totally eliminated, by the Lag Processor without detracting too much from the intended effect.

The Lag Processor is in essence a low pass filter with a manually adjustable cut-off frequency in the lower frequency range. Thus attempts to delay high frequency signals will also result in a decrease in amplitude of the signal. Nevertheless it does find application as a low pass filter especially in the treatment of white and pink noise sources.

digisound

MODULE 80-6
VOLTAGE CONTROLLED FILTERS

1. INTRODUCTION

Four types of filter are available in the DIGISOUND 80-6 series, namely, low pass, high pass, band pass and allpass (phase shift). They are all four pole filters with one volt per octave control of their cut-off, or centre, frequency. The latter frequency may be set anywhere in the audio range. Voltage control of signal regeneration (resonance or Q factor) is also included.

Filters are normally used in three modes. Firstly, timbre modulation which allows more complex waveforms to be developed by continuously altering the partials present during the course of a note. The latter may be accomplished using a low frequency oscillator to vary the cut-off frequency of the filter. A widely employed variation of the latter technique is the use of an envelope generator, triggered by the keyboard, to control the cut-off frequency. In this way a tone may be produced, assuming a low pass filter, which begins at the fundamental followed by an increasing number of partials. Next, subtractive synthesis, in which partials are subtracted from complex waveforms to effect changes in tone quality. The third technique is resonant synthesis in which the regeneration of the signal is increased to a level which will cause the filter to oscillate on receiving a sharp impulse, e.g., from an envelope generator set to near minimum time constants or from a pulse waveform. This technique is widely used to produce percussive sounds. For additional information please consult the DIGISOUND 80 user's manual.

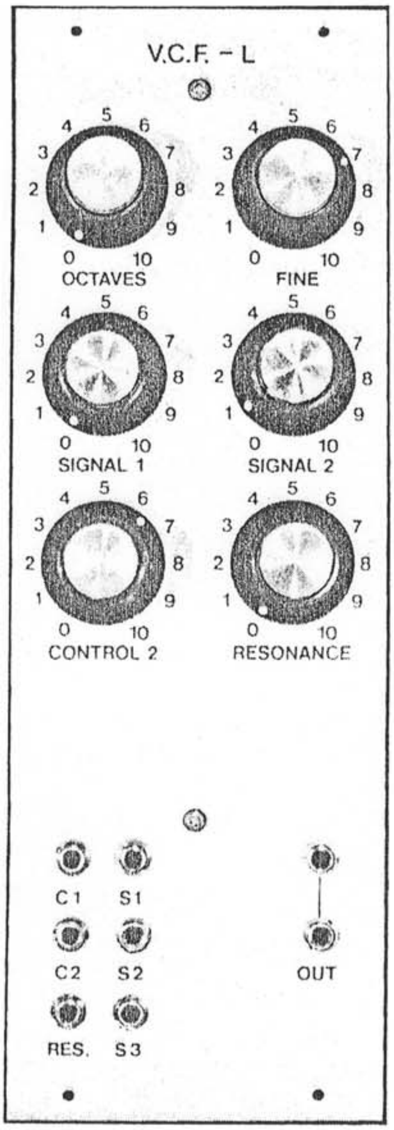


FIGURE 1. 80-6L PANEL

A brief description of the four types of filters is given below.-

LOW PASS FILTER. This will pass all frequencies up to the cut-off point and thereafter the frequencies are sharply attenuated at a rate of 24dB/octave. By increasing the resonance control, manually or using an external control voltage, a band of frequencies around the cut-off point are emphasised and the more regeneration used the more 'electronic' the sound becomes. Low pass filtering is useful in simulating the tonal characteristics of several conventional instruments.

HIGH PASS FILTER. This passes all frequencies above the cut-off point and the roll-off below this point is again 24db/octave. The effect of high pass filtering is therefore to remove the fundamental and lowest partials and leave only the weak upper partials. It does not, therefore, find widespread use in subtractive synthesis of waveforms although high pass filtering of a sawtooth waveform can produce bright sounds. The filter is also of use in filtering white noise. It is not normal to include resonance control with this type of filter, due to the limited application of the effect, but it has been included in the 80-6 design since it can be implemented for a low cost. Its effect is the same as with the low pass filter, namely, to emphasise a band of frequencies at the cut-off point.

BAND PASS FILTER. This will pass a band of frequencies centred on the pole frequency of the filter. The filter is derived from two poles of high pass followed by two stages of low pass filtering thus giving a 12dB/octave roll-off on either side of the centre frequency. The effect of the resonance control, or Q control, in this mode is to emphasise the cut-off frequency and so increase the rate of roll-off. This filter is used in imitative synthesis but more commonly a number of band pass filters need to be employed in order to achieve effective results.

PHASE SHIFT FILTER. This is an allpass filter with mixing of the original signal to create two deep notches. The effect of regeneration (resonance) is to sharpen the corners of the notches thereby increasing their depth. The 'phasing' effect is well known and most of the low cost phasers only have two notches and usually not as sharp as the 80-6P. A particular advantage of the design described below is its low noise but even so we believe that six notches are a minimum for a rich phasing effect.

2. DESIGN

The 80-6 filters utilise the CEM 3320 voltage controlled filter (VCF) IC from Curtis Electromusic Specialties. The functional block diagram and pin out of this device is shown in Figure 2. It will be seen that the CEM 3320 contains four independent filter stages which may be interconnected to provide a wide variety of filter responses. The pole frequency of the four stages is controlled by a single exponential generator which has a minimum range of ten octaves. The IC also includes a separate transconductance amplifier (pins 8 and 9) whose output is connected to the first

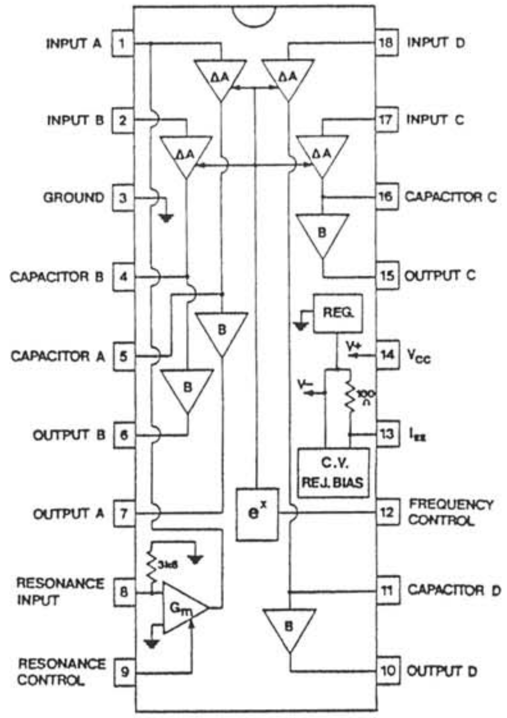


FIGURE 2. CEM 3320

filter stage. In the present design this amplifier is used to provide manual or external voltage control of regeneration (resonance). A major advantage gained by using the CEM 3320 is the low signal to noise ratio coupled with low distortion.

Figure 3 shows the complete circuit diagram for the 80-6L low pass filter. Note the letters A to E since within the area bounded by these points the components around the CEM 3320 vary in order to achieve the required filter response. Conversely, the components outside of these points, except for the value of R5, remains constant irrespective of filter type. Thus the modifications required for the 80-6H high pass filter, the 80-6B band pass filter and the 80-6P phase shift filter are shown in Figures 4, 5 and 6 respectively. The configuration of the filters follows the design principles given in the CEM 3320 data sheet and reference should be made to the latter if a full understanding of the design is required. Note, however, that the component values are based on using the filters with a +/-15V supply. Each filter is designed to accept 10V P-P signals and up to three signals may be mixed via the Signal 1 to 3 inputs which are summed by IC1. Since the Signal 3 input does not have have a potentiometer its gain has been reduced. Thus three 10V P-P signals may be mixed in equal proportions by setting RV1 and RV2 at 33% rotation. Similarly two 10V P-P signals may be mixed in any proportions via the Signal 1 and Signal 2 inputs so long as the combined percentage rotation of RV1 and RV 2 does not exceed 100%. The overall gain of the filters is nominally one but allowance has been made for peak voltages that occur at maximum resonance. The signal level within the CEM 3320 is determined by R5 which varies according to the type of filter and has been chosen such that no clipping will occur with a 10V P-P signal. The input signal is inverted by IC1 but re-inverted by the output stage configured around IC2b. Note that the CEM 3320 is a current-in current-out device and for proper operation with a +15V supply there will be a quiescent DC voltage of about 6V5 on each output buffer. This DC voltage is trimmed out by RV8 prior to amplification of the filtered AC signal by IC2b.

Regeneration (resonance) is obtained by feeding the filtered signal to the transconductance amplifier via C12 and R32. The amount of signal fed back to the input stage is controlled by a positive DC voltage applied to pin 9 of the CEM 3320. By using a jack socket this voltage may be applied manually or it may be obtained from an external control voltage source. In the manual mode, R6 connected to the +15V suppply forms an attenuating network with RV3 such that there will be approximately +10V at the input of RV3. If a jack plug is now inserted into the jack socket, J3, then R6 is disabled and RV3 may be used to attenuate the external control voltage. The value of R32 varies with different filter types and its value has been chosen such that +10V applied to R7 will cause the filter to oscillate.

A current limiting resistor, R32, is always required on the negative supply to pin 13 of the CEM 3320 and its value must be selected according to the supply voltage used - refer to data sheet for further information. By adjustment of the current limiting resistor an improvement in control voltage feedthrough is obtained, and this is the purpose of RV9.

The control scale for the CEM 3320 is 18mV/octave and this is applied to pin 12. The control inputs are summed by IC2a. The scaling components are R13, RV7, R14 and R15 and are such that after calibration a one volt control voltage applied to R10 or R11 will shift the cut-off frequency by one octave. The best control range is achieved with a voltage at pin 12 varying from -25mV to +155mV. Also an increasing positive voltage at this pin decreases pole frequency. The ranging is achieved with R12 connected to the negative supply and in order to obtain the usual convention of increasing positive voltage to increase frequency the voltages are inverted by IC2a. Normally Control Input 1 would be connected to the keyboard control voltage so that the

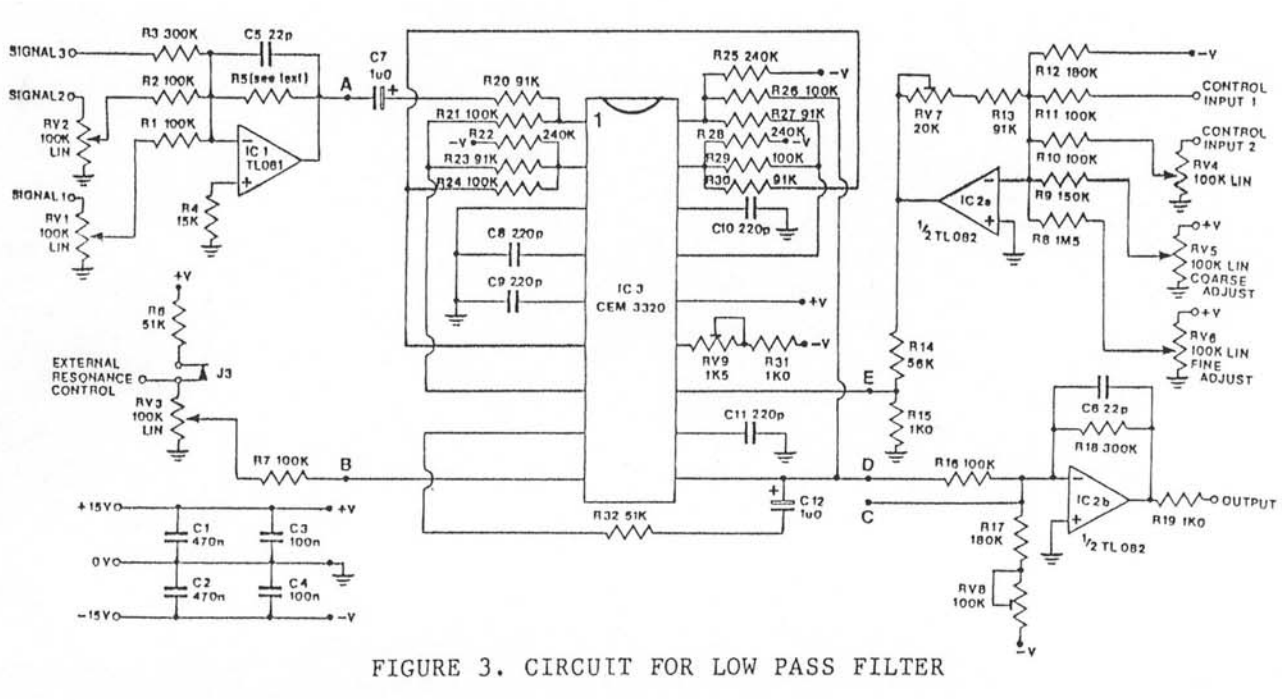


FIGURE 3. CIRCUIT FOR LOW PASS FILTER

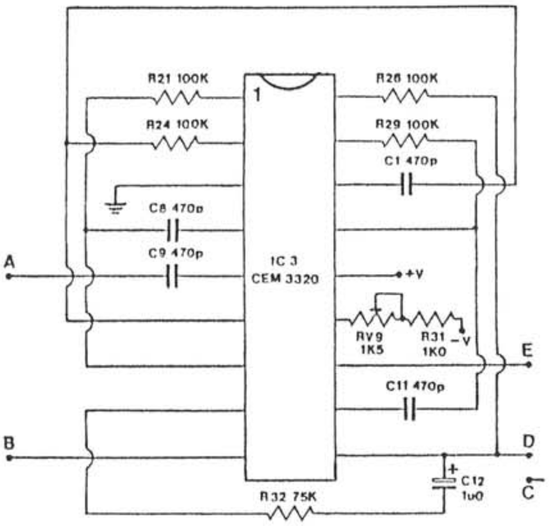


FIGURE 4. HIGH PASS FILTER

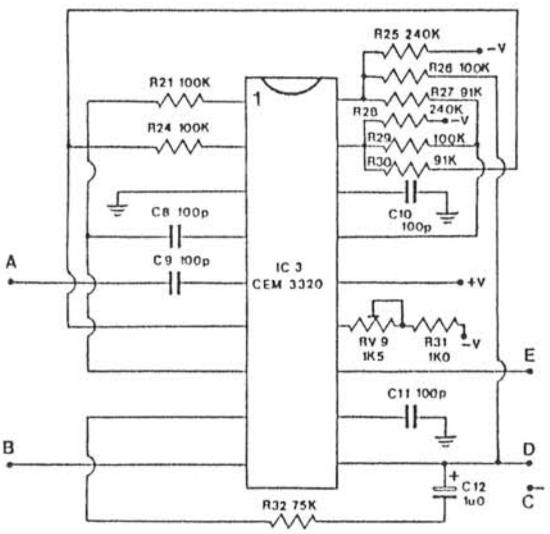


FIGURE 5. BAND PASS FILTER

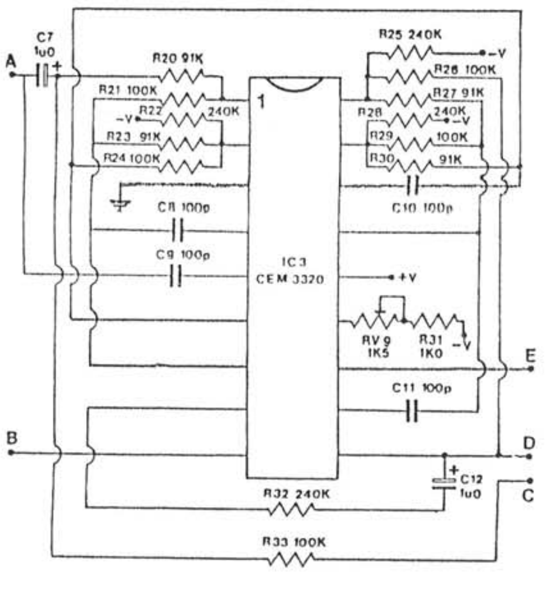


FIGURE 6. PHASE SHIFT FILTER

filter tracks the voltage controlled oscillator (VCO). Control Input 2 has an attenuator and the normal use for this input would be connection of an envelope generator to vary the cut-off frequency during the course of a note. This latter input is also suited to modulation from an external oscillator. The initial cut-off frequency is determined by the Coarse Control, RV5, and the Fine Control, RV6. Accurate adjustment of initial frequency is not usually available on synthesiser filters but it will be found useful for polyphonic applications, e.g., when used in conjunction with the ALPHADAC 16 microprocessor control system. RV5 allows adjustment over a range of ten octaves while the Fine Control, RV6, has a one octave span. The filter capacitors, C8 to C11, are chosen such that the filter covers a 1000:1 frequency range with the lowest frequency being typically in the range of 20 to 25Hz.

3. CONSTRUCTION

The 80-6 PCB has the component placement printed on which simplifies construction. This component overlay is reproduced in Figure 7. It should be noted, however, that not all of the component locations are used, as will be evident from examination of the circuit diagrams shown in Figures 3 to 6. Components used should therefore be checked against the components list. A more important point is that there are alternative locations for the polystyrene capacitors C8, C9, C10 and C11. The locations are either identified by the capacitor number or the number followed by the suffix 'H', e.g., C8H. The locations are listed below for the four filter types.-

i) For the 80-6L low pass filter the four capacitors, C8 to C11, are located in the positions where the capacitor number is NOT followed by the 'H' suffix.

ii) For the 80-6H high pass filter and the 80-6P phase shift filter then C8 to C11 ARE installed in the positions denoted by the 'H' suffix.

iii) In the case of the 80-6B band pass filter C8 and C9 are installed at the positions marked C8H and C9H while C10 and C11 are installed at the positions marked C10 and C11.

The usual care must be taken over the orientation of the electrolytic capacitors and the IC`S. Even when the latter are installed in their DIL sockets the number '1', denoting pin 1, should still be visible on the PCB. In any event compare the completed PCB against Figure 7 prior to applying power.

Wiring of potentiometers and other panel connections to the PCB are shown in Figure 8. This diagram illustrates the components when viewd from the rear of the panel. The arrows and associated letters indicate that a wire connection must be made from the position shown to the PCB. The latter has a connecting point on its front edge with letters corresponding to those shown in Figure 8.

The jack sockets in the diagram are of the type supplied by Digisound Limited. The top connection, as illustrated, is the connection which is made with the jack when the latter is inserted. The lower connection is disabled by insertion of a jack plug. Finally, the tab under the socket is the ground connection. It is recommended that all of these ground connections are wired to the 0V line since this facilitates connection of the filters to other equipment which may be using a separate power supply. The ground tabs may be joined together using tinned copper wire but other panel wiring should be made with insulated wire. 1/0.6mm insulated wire is ideal for panel wiring since it retains any shaping and thus allows a neat appearance to be obtained.

On completion of the construction stage carefully examine the underside of the PCB to ensure that all connections are properly soldered and that no solder bridges have been formed. Also check polarity of power supply before powering up.

4. CALIBRATION

A simple check may be made of the frequency control input and also that the wiring of RV5 and RV6 are correct. With IC3 removed, the power on and the coarse and fine controls fully anti-clockwise measure the voltage at the junction of R14 and R15 and adjust RV7 to obtain +155mV. Now turn RV5, coarse, or 'octaves', control fully clockwise and the voltage should be about -25mV. If you are able to make these measurements with a reliable voltmeter but do not obtain values which are close to those given then check the control input components and double check that there are no solder bridges on the foil side of the PCB.

Turn off power and insert IC3, power up, turn RV3 (resonance control) fully anti-clockwise and RV5 to about mid position. Measure the voltage at the output of the filter module and adjust RV8 until a zero reading is obtained. This cancels the DC voltage at the output from pin 10 of IC3.

The next step is to trim the control voltage feedthrough using RV9. Power up the filter and connect the +/-5V sine or triangle wave from the 80-2

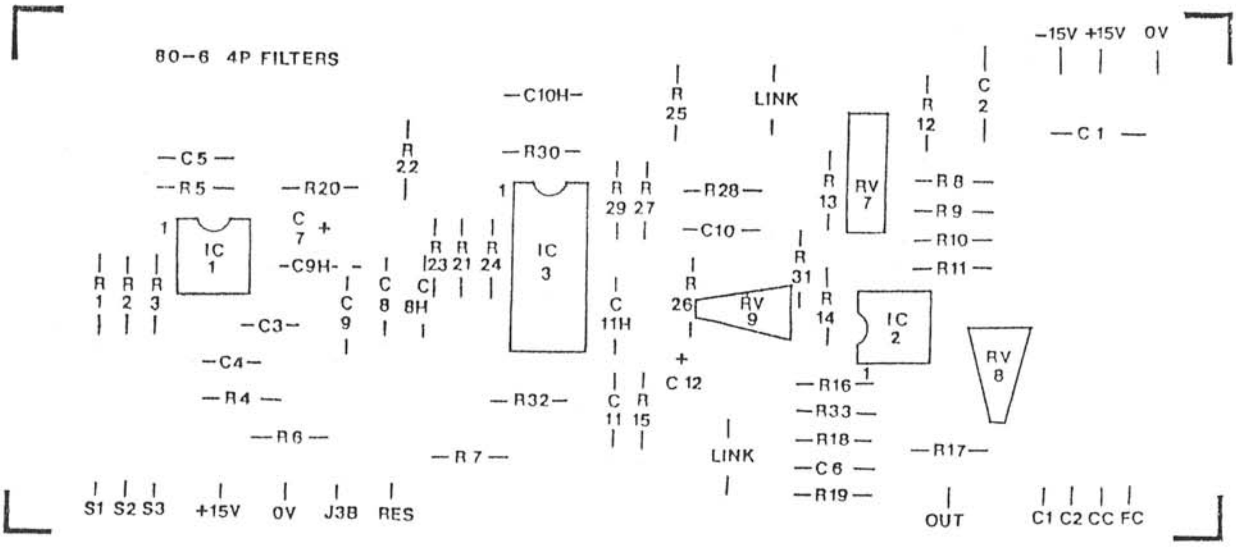


FIGURE 7. 80-6 COMPONENT OVERLAY

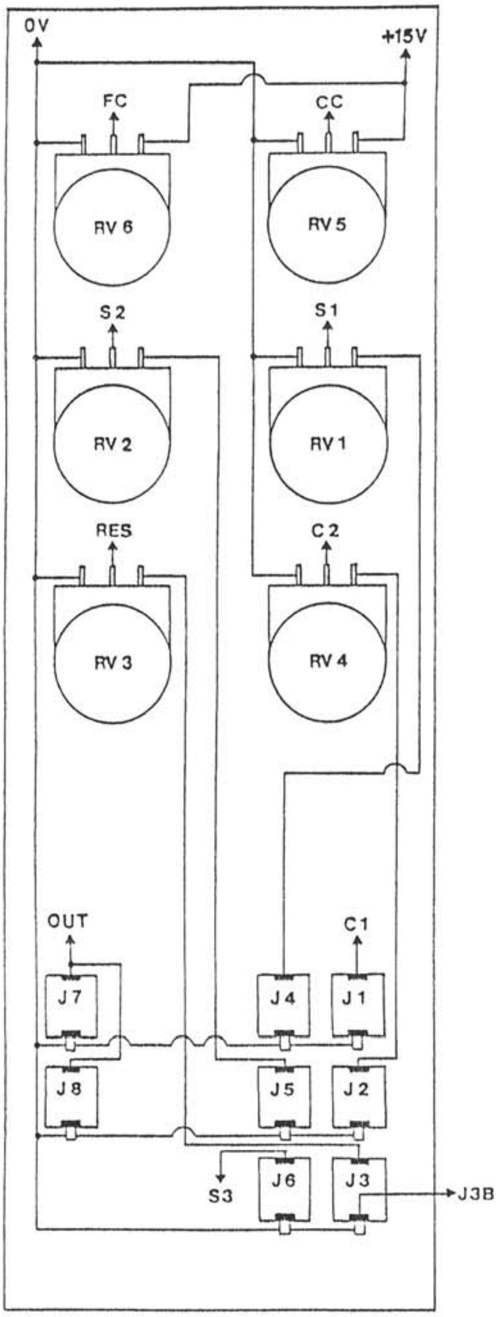


FIGURE 8. PANEL WIRING

VCO, or other oscillator with a similar output, to the Control Input 2. The oscillator should be at a frequency of about 50 to 100Hz. RV4 and RV5 should be set midway and all other potentiometers fully anti-clockwise. Connect the output of the filter to an amplifier and adjust RV9 for minimum output. Some adjustment to RV4 or the amplifier 'volume' may have to be made in order to obtain the optimum position for RV9. The value of RV9 or R31 must not be altered. If a suitable oscillator is not available then this step may be omitted and RV9 set to its mid position.

The last step is to calibrate the filter so that it will track 1V/octave oscillators, such as the DIGISOUND 80-2. The resonance control feedback resistor, R32, has been selected such that the filter will oscillate when some frequency control input is present, thus turning the filter into a low distortion sine wave oscillator. A calibrated VCO will normally be available at this stage in the construction of a synthesiser and this will simplifies calibration. Any of the following techniques may be used in addition to treating the filter as an oscillator and using the methods described for the VCO.

i). Beat frequency technique. Apply about 3V5 to Control Input 1, which will be the normal input for the keyboard, and adjust PR7 to give a frequency of about 250 to 300Hz when RV3 is rotated to the point where oscillation is sustained - as heard through one channel of an amplifier. Connect a calibrated VCO to the other channel of the stereo amplifier and apply an external control voltage to the VCO until there is no beat frequency. Increase voltage to both VCO and VCF frequency control inputs by exactly one volt and the adjust RV7 until no beat frequency is heard.

ii). Lissajous figures. Same procedure as (i) but outputs from VCO and VCF are coupled to the X and Y inputs of an oscilloscope to generate Lissajous figures. This method will probably be described in the handbook accompanying the oscilloscope. As an example, if both inputs are sine waves of equal amplitude then a stable circle will be generated when their frequencies are matched.

iii). Maximum signal amplitude.
Another approach with an oscilloscope is to apply a signal of about 250Hz from a calibrated VCO to Signal Input 1 of the filter and observe the output from the filter on the oscilloscope. Apply a voltage to Control Input 1 of the filter until the point where the output reaches its maximum amplitude is observed. A small amount of resonance will help. Increase voltage to both VCO and VCF by exactly one volt and adjust RV7 until maximum amplitude is restored.

5. COMPONENTS

COMPONENTS COMMON TO ALL FILTERS.

RESISTORS, 5%, 1/4w carbon film

R1,2,7,16,21,24,26,29	100k
R3,18	300k
R4	15k
R6	51k
R8	1M5
R9	150k
R17	180k
R19,31	1k0

RESISTORS, 1%, 1/4w metal film,100ppm

R10,11	100k
R12	180k
R13	91k
R14	56k
R15	1k0

POTENTIOMETERS, PRESETS

RV1,2,3,4,5,6	100k lin.
RV7	20k cermet multiturn
RV8	100k carbon
RV9	1k5 carbon

CAPACITORS

C1,2	470n polyester
C3,4	100n polyester
C5,6	22p polystyrene
C12	1u0 PCB electrolytic

SEMICONDUCTORS

IC1	TL 081
IC2	TL 082
IC3	CEM 3320

ADDITIONAL COMPONENTS FOR LOW PASS FILTER, 80-6L.

RESISTORS, 5%, 1/4w carbon film

R5	36k
R20,23,27,30	91k
R22,25,28	240k
R32	51k

CAPACITORS

C8,9,10,11	220p polystyrene
C7	1u0 PCB electrolytic

ADDITIONAL COMPONENTS FOR HIGH PASS FILTER, 80-6H

RESISTORS, 5%, 1/4w carbon film

R5	24k
R32	75k

CAPACITORS

C8,9,10,11	470p polystyrene

ADDITIONAL COMPONENTS FOR BAND PASS FILTER, 80-6B

RESISTORS, 5%, 1/4w carbon film

R5	33k
R25,28	240k
R27,30	91k
R32	75k

CAPACITORS

C8,9,10,11	100p polystyrene

ADDITIONAL COMPONENTS FOR PHASE SHIFT FILTER, 80-6P

RESISTORS, 5%, 1/4w carbon film

R5	12k
R20,23,27,30	91k
R22,25,28,32	240k
R33	100k

CAPACITORS

C8,9,10,11	100p polystyrene
C7	1u0 PCB electrolytic

MODULE 80-7A
V.C. STATE VARIABLE FILTER

1. INTRODUCTION

The DIGISOUND 80-7A Voltage Controlled State Variable Filter (VCSVF) follows the same lay-out as the earlier 80-7 version and has seven filter functions which are selectable by means of a rotary switch. The filters are: two pole (12dB/octave) and four pole (24dB/octave) low pass and high pass; one pole and two pole bandpass; and a band reject (notch) response. The filters operate over the audio range and the frequency of each filter is voltage controlled, manually and/or using external voltage, with a scale of one volt per octave. The resonance (Q) of the filters may be varied over a wide range and although self oscillation cannot be obtained under normal circumstances the maximum resonance level is sufficient to induce some ringing of the signal inputs. Resonance is controlled manually or by use of an external voltage and a linear input is converted to an exponential response which provides the right 'feel' for this control input.

The 80-7A is designed for 10V p-p signals and 0 to +10V control voltages.

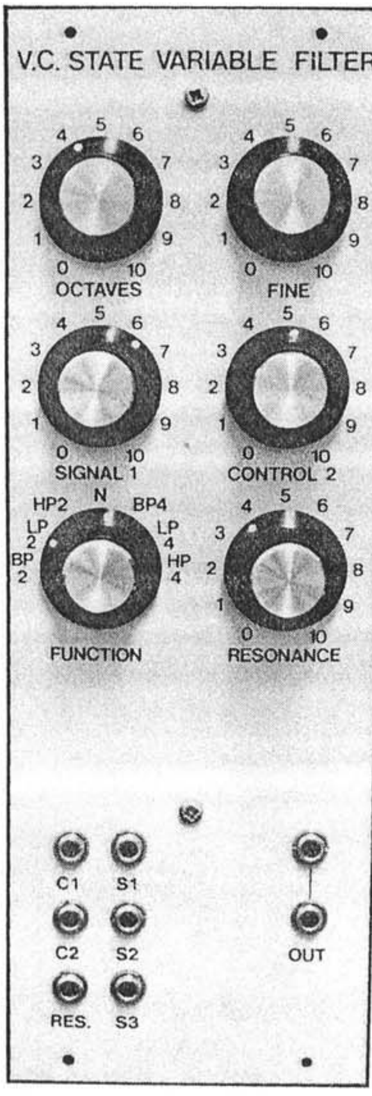


FIGURE 1. 80-7A PANEL

2. DESIGN

The main device in the 80-7A is the CEM 3320 voltage controlled filter IC and the complete circuit diagram is shown in Figure 2. If one examines the top part of the circuit diagram, around IC1, IC2a and IC2b, the basic text book elements of a state variable filter are evident. These are, a summer (IC1) and two integrators (IC2a and IC2b) with the required feedback path from the low pass output (IC2b) to the summer. In conventional designs, Q (resonance) is controlled by feeding back the bandpass output (IC2a) to the summer but this approach would introduce a number of complications in the present design - as should be apparent later. Consequently the Q control has been obtained by recirculating the bandpass output through a CEM 3335 voltage controlled amplifier, IC3a. The three outputs from this first stage are taken via switch S1b to the output amplifier built around IC6a.

To obtain the 24dB/octave low pass and high pass filters and the 2 pole bandpass the outputs from the first stage are switched by S1a to a second state variable filter built around IC5, IC2c and IC2d. These outputs are in turn switched by S1b to the output.

The notch, or band reject, output has been obtained in a different manner to usual, which it is believed makes it more useful. The notch output is obtained from the differential amplifier IC6b and inspection will show that the notch is derived by subtracting the bandpass output of the first stage from Signal Input 1. This allows the notch output level to be equal in amplitude to Signal Input 1 (or the level set by RV3) and the bandpass subtraction gives good control over notch shape. Normally a notch is obtained by combining the low pass and high pass responses but this arrangement results in peaking of the cut-off frequencies as Q is increased and this effect is often undesirable. Furthermore when using a notch filter one often requires the signal to be at unity gain and this is possible with the present design. The only slight drawback is that the notch is only effective on Signal Input 1.

The control inputs are shown at the bottom of Figure 2. For frequency control of the CEM 3320 there is: (a) Control Input 1 which will normally be used for a keyboard follow input and its 1V/octave response is obtained using IC7 summing amplifier attenuated by the divider R32/R33 which produces about 18mV to pin 12 of IC2 for each volt applied to a 100k resistor at the input to IC7. The exact scaling is obtained with RV2; (b) Control Input 2 with attenuator RV6 which will usually be used for modulation of the filter, for example by an envelope generator; (c) a coarse control, RV4, which will adjust the frequency of the filters over a ten octave range; and (d) a fine control, RV5, with a frequency adjustment of one octave and this control is useful when the 80-7A is used in a polyphonic set up. R26 sets the correct initial setting of the control scale of the CEM 3320 while R34 is the current limiting resistor for the IC and is the required value for a -15V input.

The manual control input for the resonance control is provided by R37, RV7 and the divider R38/R40. A jack socket is interposed between the supply voltage for RV7 such that when using an external control voltage RV7 may be used as an attenuator for this external input. R36 sets the

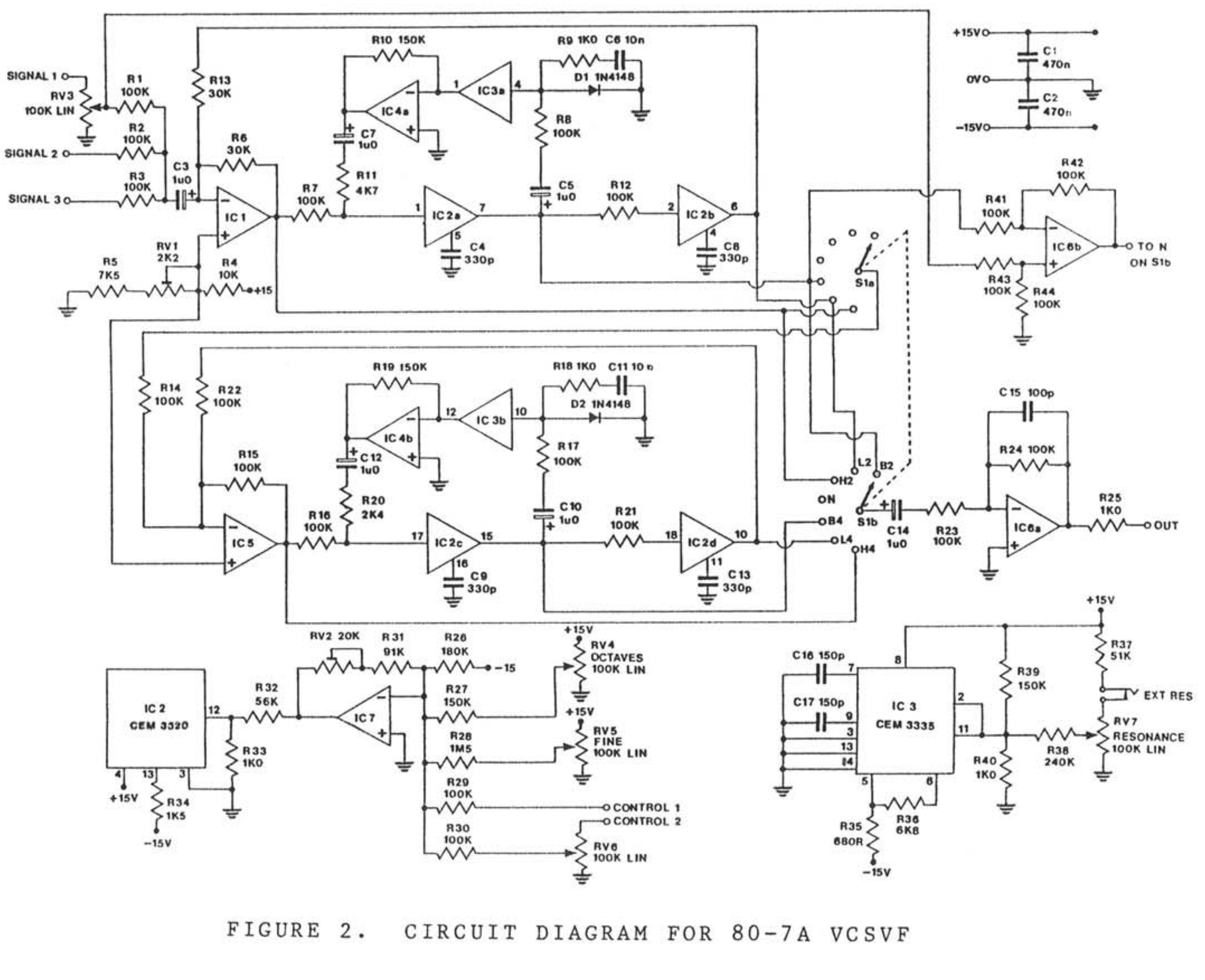


FIGURE 2. CIRCUIT DIAGRAM FOR 80-7A VCSVF

operating characteristics of the VCA while R35 is a current limiting resistor for operating the CEM 3335 from -15V. C16 and C17 are for compensation.

The control voltages for both frequency and resonance follow the normal levels for the DIGISOUND 80 modules and will normally be in the range 0 to +10 volts.

Reverting to the main circuit it should be noted that the CEM 3320 is a current in - current out device. For the first stage the required current is obtained via R7 and R12 and the voltage to obtain the current is obtained through the non inverting input of IC1 using the resistive divider R4, RV1 and R5. The voltage required is approximately seven volts but it has to be set quite accurately in order to obtain the full signal range through the cells IC2a and IC2b. Incorrect setting will result in clipping of the signals or even failure of the cells to function correctly. The setting is relatively easy to obtain and is discussed in Section 4.

The resonance control using IC3a (and IC3b for the second stage) is achieved by blocking the DC bias voltage with C5 and converting the signal to a current with R8. There will be a current out from IC3a proportional to the signal level and the voltage applied to the resonance control input discussed earlier. This output current goes to a current to voltage converter, IC4a, and this voltage is again converted back into a current prior to summing into IC2a. The latter voltage to current conversion may seem rather unecessary but it does simplify control over the filter without resorting to additional trimmers. Components R9, C6 and D1 are for compensation and to prevent latch up while C7 blocks any DC arising from IC4a.

The main difficulty with configuring both a two pole and four pole state variable filter is that of achieving comparable signal levels from all outputs, especially while providing adequate resonance. It is the peak signal level at maximum resonance which determines (because of clipping considerations) the signal level at maximum resonance and this situation is aggravated by added resonance in the second stage. In the original 80-7 the resonance level was quite low and even so the signal levels had to be 'normalised' by having a resistor for each output both between filter stages and between the filters and the output amplifier. This has been avoided in the present design and typical peak output voltages from a 10V p-p sinewave input are.-

Mode	Min. Res	Max. Res
BP2	4V4	7V2
LP2	4V6	7V2
HP2	5V6	8V6
BP4	2V2	9V0
LP4	3V0	10V4
HP4	7V2	12V8
N	equal to level set by RV3	

It should be noted that in the bandpass mode although the amplitude of the signal increases with resonance the actual 'volume' of sound does not necessarily increase since the effective bandwidth decreases. Likewise the 'volume' is decreased in the low pass and high pass modes as the cut-off frequency is varied and removes more of the input signal. The latter will be more noticeable in the 24dB/octave modes. Finally, although three signal inputs are provided the combined sum of the three inputs should not normally exceed 10V p-p otherwise clipping may occur within the CEM 3320, particularly as the resonance level is increased.

3. CONSTRUCTION

The 80-7A PCB is printed with a component overlay which aids the construction stage. The overlay is reproduced in Figure 3 to allow checking of component placement after the module has been completed.

Take particular care with the orientation of the IC's, the diodes and the electrolytic capacitors. The latter have their negative side clearly marked on their body with a band of '-' symbols. In the case of the diodes a band (vertical line) is shown on the PCB overlay (in front of identification 'D1' and after 'D2') and this band indicates that the band on the diode, denoting its cathode, should be towards the hole nearest the band on the PCB. For the IC's, even after the DIL sockets are installed the number '1', denoting pin 1, will still be visible on the PCB. In any event compare the completed PCB against Figure 3 prior to applying power to the module. Also before powering up check that the eight wire links have been made (uninsulated wire may be used for these) and inspect the foil side of the PCB for solder bridges and suspect joints.

The main panel wiring is shown in Figure 4 and this diagram illustrates the components when viewed from the rear of the panel. The arrows and associated letters indicate that a wire connection must be made from the position shown to the front edge of the PCB which has corresponding letters.

The filter selection switch, S1, is shown separately in Figure 5. The switch consists of a Lorlin RA Shaft Assembly and a separate 2pole 9way wafer. First, it is desirable to shorten the wafer securing shaft and this should be done by pulling it from its socket and cutting it while it is held in a vice. Attempting to cut this shaft in situ may damage its securing bush. The wafer securing bolts should also be shortened before attaching the wafer and this is best done by putting the 8BA nuts onto the bolts below the cutting point prior to sawing or cutting. Removing the nuts will then clean up the thread at the cut end. The rotary shaft at the front will also require shortening to suit the control knobs being used. Next remove the panel securing nuts and washers and adjust the end stop washer such that the switch will only rotate to the seven positions required. The switch should now be secured to the panel and the wafer installed. With the switch fully anti-clockwise the rear view of the wafer should appear as illustated in Figure 5 (only one side of the wafer has the contacts visible as shown in the diagram). The switch may then be wired up from the contacts shown to corresponding letters on the front edge of the PCB. Note the three connecting wires around the switch.

To wire up the notch filter one wire from 'S1' of Figure 4 goes to 'SN' on the PCB and another wire from 'B2' on Figure 5 goes to 'BN' on the PCB.

The jack sockets illustrated in Figure 4 are of the type supplied by Digisound Limited. The top connector, as shown, is the connection which is made with the jack plug when the latter is inserted. The lower connection is disabled by insertion of a jack plug and this is made use of to switch RV7 between manual and external

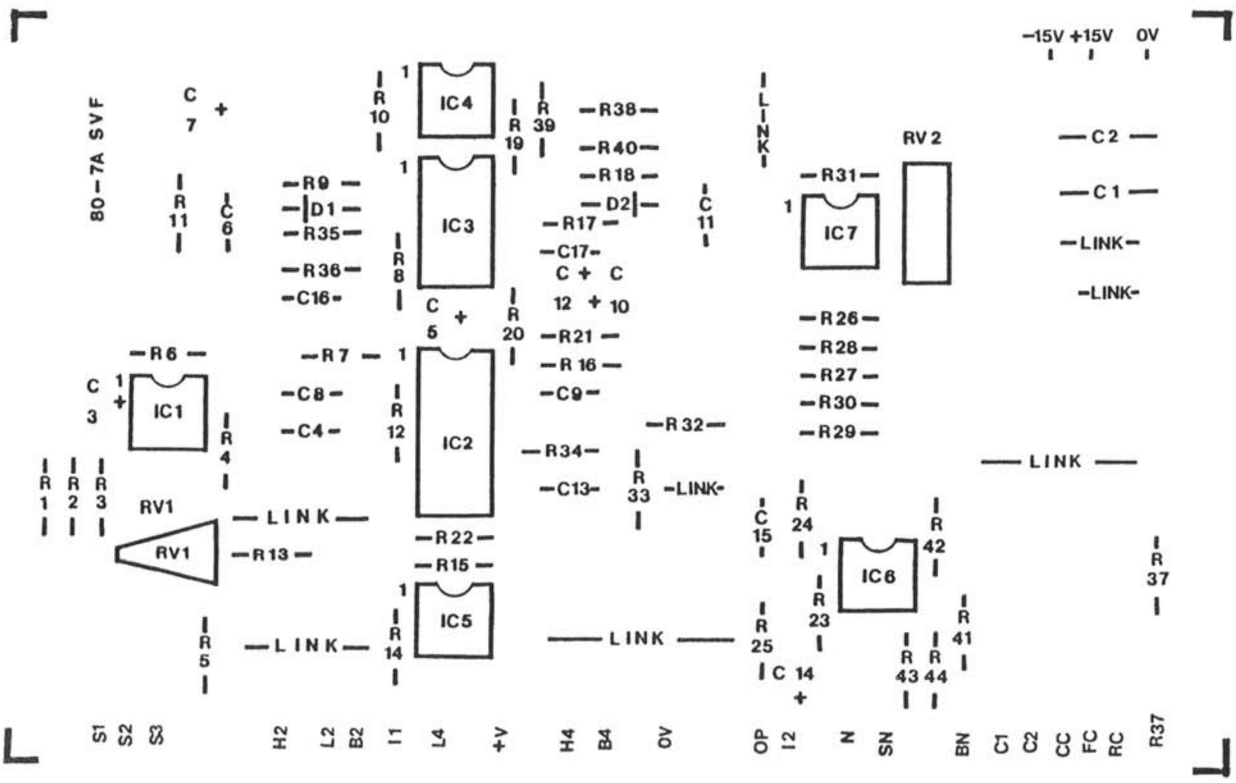


FIGURE 3. COMPONENT OVERLAY FOR 80-7A

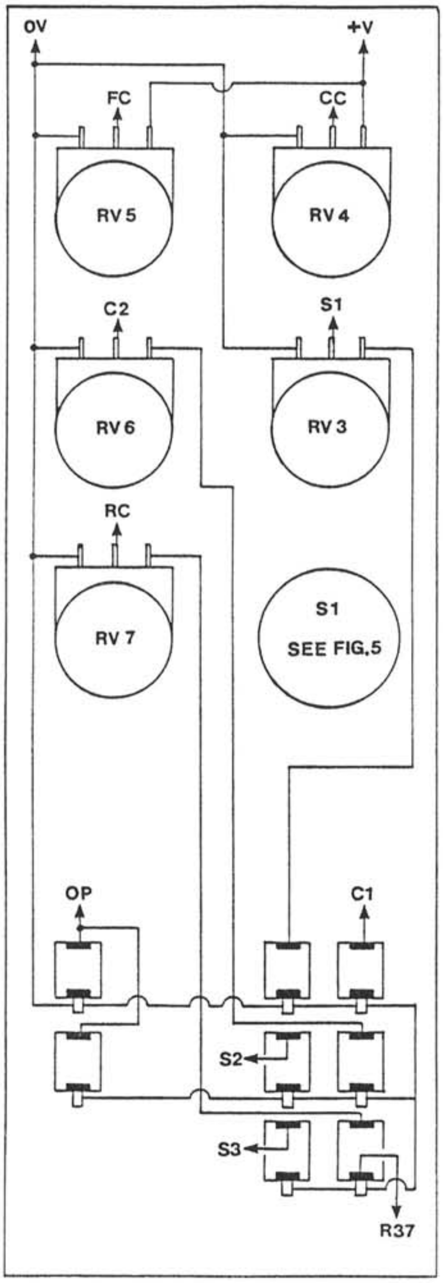


FIGURE 4. PANEL WIRING

resonance control. Finally, the tab under the socket is the ground connection. It is recommended that all of these ground connections are wired together and taken to the 0V connection on the PCB since this facilitates connection of the module to other equipment which may be operating from a separate power supply. The ground tabs may be soldered together using tinned copper wire but other panel wiring should be made with insulated wire. 1/0.6mm insulated wire is ideal for panel wiring since it retains any shaping and so allows a neat appearance to be obtained. It is also worth using a variety of colours to facilitate checking after assembly. Wires

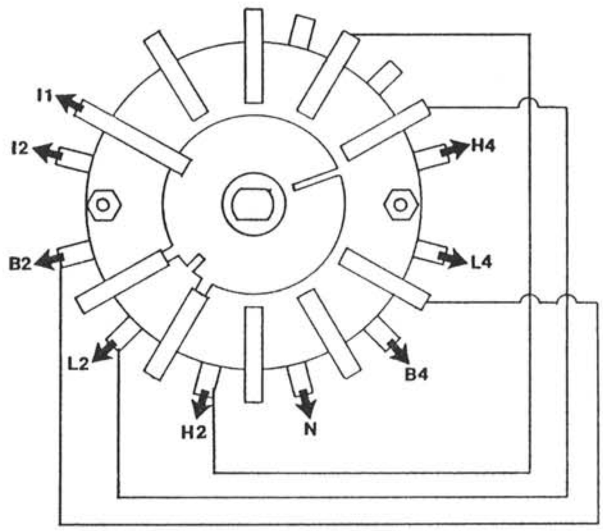


FIGURE 5. SWITCH S1 WIRING

between panel components and PCB should be kept as short as possible and this is quite important with the the large number of wires used for the 80-7A.

4. CALIBRATION

The first step is to adjust the bias voltage with RV1. Connect a sinewave of about 1kHz to one of the signal inputs (RV3 should be fully clockwise if Signal Input 1 is used); switch S1 to LP4 mode; turn RV7 fully clockwise (maximum resonance) and connect the output of the filter to an amplifier or an oscilloscope. Start with RV1 at mid position and turn RV4 (octaves) to obtain maximum amplitude. If no signal is heard (or seen) while RV4 is rotated then leave it set at mid position and adjust RV1 slowly, first one way then the other, until a signal is obtained. If nothing is heard (or seen) then switch off and check module for an assembly fault. When RV4, plus RV5 for fine tuning, have been set for maximum output then slowly turn RV1 to further increase amplitude without clipping. The latter will be evident using an amplifier by the harshnes of the tone and may also be accompanied by a decrease in output level. It is essential to turn RV1 slowly, that is, a small increment at a time and leave for a few seconds before making another adjustment since there is some time lag between adjustment and the filter settling down to a stable position.

Next apply a high harmonic waveform (sawtooth or pulse) to the signal input and switch to each filter response and each time alter RV4 to check that the tone varies as this potentiometer is adjusted. This is to determine that all filters are operating.

The last step is to adjust the control scale with RV2. It is assumed that by the time a constructor requires a filter they will possess a calibrated voltage controlled oscillator. The following text assumes the latter is a DIGISOUND 80-2 but it should be easy to interpret the instructions for other VCO's. The 80-7A may be made to resonate by the following means -
a. Switch to LP4 or HP4 mode;
b. Connect a patchcord between the output and Signal Input 2.
3. Connect a patchcord to the external resonance control input (RES) and connect +15V to the signal contact at the other end of the cord. Do not connect the +15V first and then insert into 'RES' otherwise the power supply will be momentarily shorted. Turn RV7 until the filter oscillates, that is, produces a sinewave whose frequency may be varied by RV4 and RV5.

Once oscillating the filter may be calibrated in the same way as a VCO. Connect a variable voltage source to Control Input 1, turn RV4 and RV5 fully anti-clockwise and then increase voltage to obtain about 200-300Hz. It is assumed for the moment that a digital frequency meter is available which is connected to the output of the 80-7A. Increase the applied voltage by exactly one volt and adjust RV2 such that the one volt change doubles the frequency of sinewave output. This step will have to be repeated a few times to achieve accurate calibration.

If a frequency meter is not available then connect the calibration voltage to both Control Input 1 on the 80-7A and the 80-2 VCO. Again increase voltage until a frequency of about 200-300Hz is obtained from the filter and if possible adjust the fine controls on one, or both, modules until the outputs are matched. If necessary, also use the 'octaves' control on the 80-7A to achieve matching of outputs. The matching may be determined by the Lissajous figures method if an oscilloscope is available or the beat frequency technique by taking the outputs from the two modules to a stereo amplifier. Now increase the calibrating voltage by exactly one volt and adjust RV2 of the 80-7A until the frequencies for the filter and VCO are again matched. Take care not to disturb any of the frequency control potentiometers during this latter step. The procedure should be repeated until a one volt increase doubles the output frequency from the filter - as it does with the VCO.

An alternative method with a calibrated VCO is the peak amplitude technique. Connect the sinewave from the VCO to Signal Input 2 of the 80-7A and the latters output to an amplifier or oscilloscope. Connect a calibrating voltage to Control Input 1 of both the 80-2 and 80-7A and increase the voltage until the output from the VCO is about 200-300Hz. Adjust the 'octaves' and 'fine' controls on the filter for maximum output signal. Next increase the calibrating voltage to both modules by exactly one volt and without touching any of the frequency adjusting panel controls adjust RV2 until the signal output is again at its maximum value. If an amplifier is being used then some adjustment of the 'volume' control may be required such that the point of maximum signal is clearly discernible. As with other techniques repeat a few times by decreasing calibrating voltage by one volt, obtaining maximum signal with RV2, increasing voltage by one volt and again adjusting RV2 for maximum signal. In fact the step is repeated until a one volt increase in applied voltage doubles the frequency and a reduction of one volt halves the frequency.

Precise calibration of filters is not essential in many situations but it becomes important if a number are used in a polyphonic system.

5. USING

The general application of filters will be well known to most constructors but for those unfamiliar with their application, as well as more unusual uses, they should study the relevant sections of 'Using The DIGISOUND 80 Modular Synthesiser'.

One particular advantage of the 80-7A, however, is the provision of both 12dB/octave and 24dB/octave low pass (and high pass) filters. While the 24dB types are usually employed in many synthesiser applications the 12dB filters are of value when creating 'electronic' sounds since they will leave a higher proportion of harmonics in the filtered signal. The variation in effect between the two filter types is worth exploring in most creative patches.

6. COMPONENTS

RESISTORS, 5%, 1/4w carbon film

R1,2,3,7,8,12,14,15,16,17, 21,22,23,24,41,42,43,44	100k
R4	10k
R5	7k5
R6,13	30k
R9,18,25,40	1k0
R10,19,39	150k
R11	4k7
R20	2k4
R28	1M5
R34	1k5
R35	680R
R36	6k8
R37	51k
R38	240k

RESISTORS, 1%, 1/4w metal film, 100ppm

R26	180k
R27	150k
R29,30	100k
R31	91k
R32	56k
R33	1k0

CAPACITORS

C1,2	470n polyester
C3,5,7,10,12,14	1u0 PCB electrolytic
C4,8,9,13	330p polystyrene*
C6,10	10n polyester
C15	100p polystyrene*
C16,17	150p polystyrene*

* may be substituted by polycarbonate

SEMICONDUCTORS

IC1,5	TL 081
IC2	CEM 3320
IC3	CEM 3335
IC4,6	TL 082
IC7	741
D1,2	1N4148

POTENTIOMETERS, TRIMMERS

RV1	2k2 carbon trimmer
RV2	20k cermet multiturn trimmer
RV3,4,5,6,7	100k lin. pot.

MISCELLANEOUS

S1	2p9w rotary switch

80-7A V.C. STATE VARIABLE FILTER

Please make the following corrections -

1. On the PCB component overlay two R27's are shown and the one nearest IC7 should be R26.
2. In the components list C10 (10n polyester) should be C11.
3. RV1 has been changed to 4k7.

ENVELOPE SHAPER

Tamper with your time constants. This Project 80 design by R.C. Blakey gives full control of Attack, Decay, Sustain and Release.

The envelope generator is based on the SSM2050, a voltage controlled transient generator produced by Solid State Micro Technology. Using this IC all that is necessary to vary the time constants for the Attack (A), Initial Decay (D) and Final Decay or Release (R) is a voltage applied to the appropriate pin via a scaling resistor. A minimum range of 2 mS to 20 S is available for each of the three timing functions. The voltage response is exponential which means that the most useful time range utilises the highest proportion of the associated control potentiometer. The attack output is nominally 0 to 10 V and the Sustain level (S) is simply a voltage applied to Pin 12.

It has separate gate and trigger inputs whereby a combined gate and trigger pulse will initiate a full ADSR response; a trigger applied after the first one and while the gate pulse is still present will restart the attack response and a gate pulse on its own will generate an AD contour. When the gate pulse is released the final decay commences, as is usual with ADSR and AD envelope shapers.

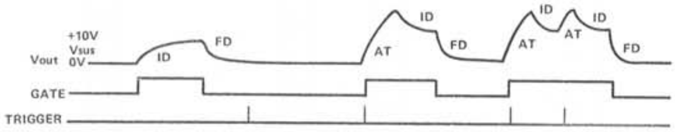
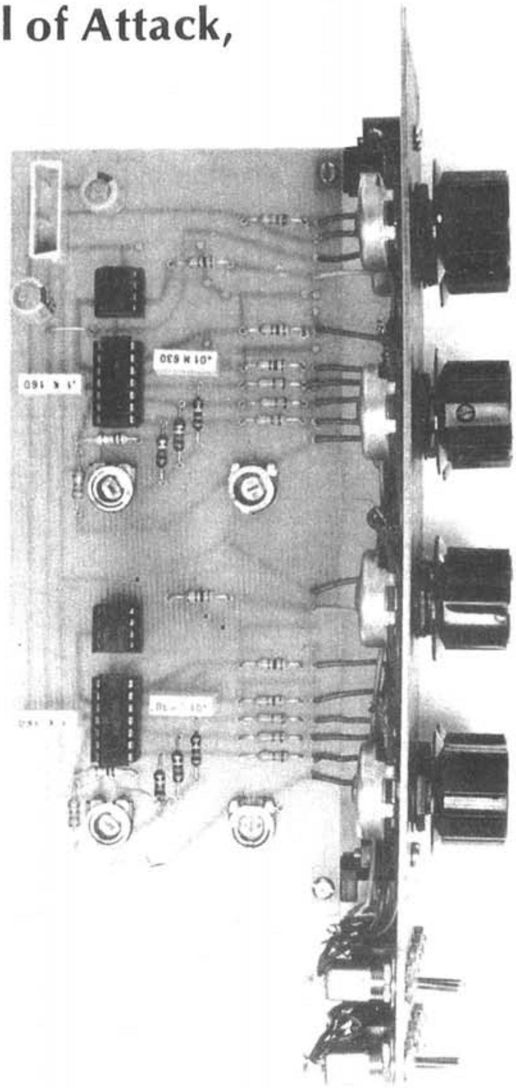


Fig.1. Wave forms associated with envelope processing.



The time constants may be trimmed so that any number of ADSR's can be adjusted to exactly the same scale. Also an adjustment to ensure that the sustain voltage accurately matches the peak attack voltage is provided. The output buffer in the SSM2040 is adequate for most practical purposes but to retain our 'plug in anything to anywhere' philosophy an external buffer has been added. Other features included are external initiation of the ADSR or AD contours, for example, from a manual push button, as well as provision to use gate and trigger pulses derived from TTL logic.

Construction

The PCB is designed to take two envelope generators and as usual will fit either a panel or the TEKA ALBA A23G case. If the latter is used then there is only sufficient panel space to sensibly install a single envelope generator.

Construction is very straightforward and the only points to note are the single wire link and the opposed orientation of the SSM2050 and the 741 buffer.

An on-off switch, SW1, is connected across the inputs marked 'TRIGGER (CMOS)' and 'GATE (CMOS)' so that when only single pulses are available, eg, manual gating, then both the ADSR (SW1 closed) and AD (SW1 open) responses can be obtained. The manual gating can be added by connecting a push to make switch between the PCB connections marked 'OUTPUT FOR MANUAL GATE' and 'MANUAL GATE'. The push button may be panel mounted but the preferred approach is to take the former connection to a jack socket and to use an external hand, or foot, switch connected to two jack plugs. These jack plugs go to the Gate (G) input and the Manual input (from R11). The option and type of switch is left to the constructor.

Resistors R12, R13 and R14 are not part of the basic kit but are to be installed by constructors who are using TTL logic to derive gate and trigger pulses. Also in this case the switch, SW1, is connected across the PCB con-

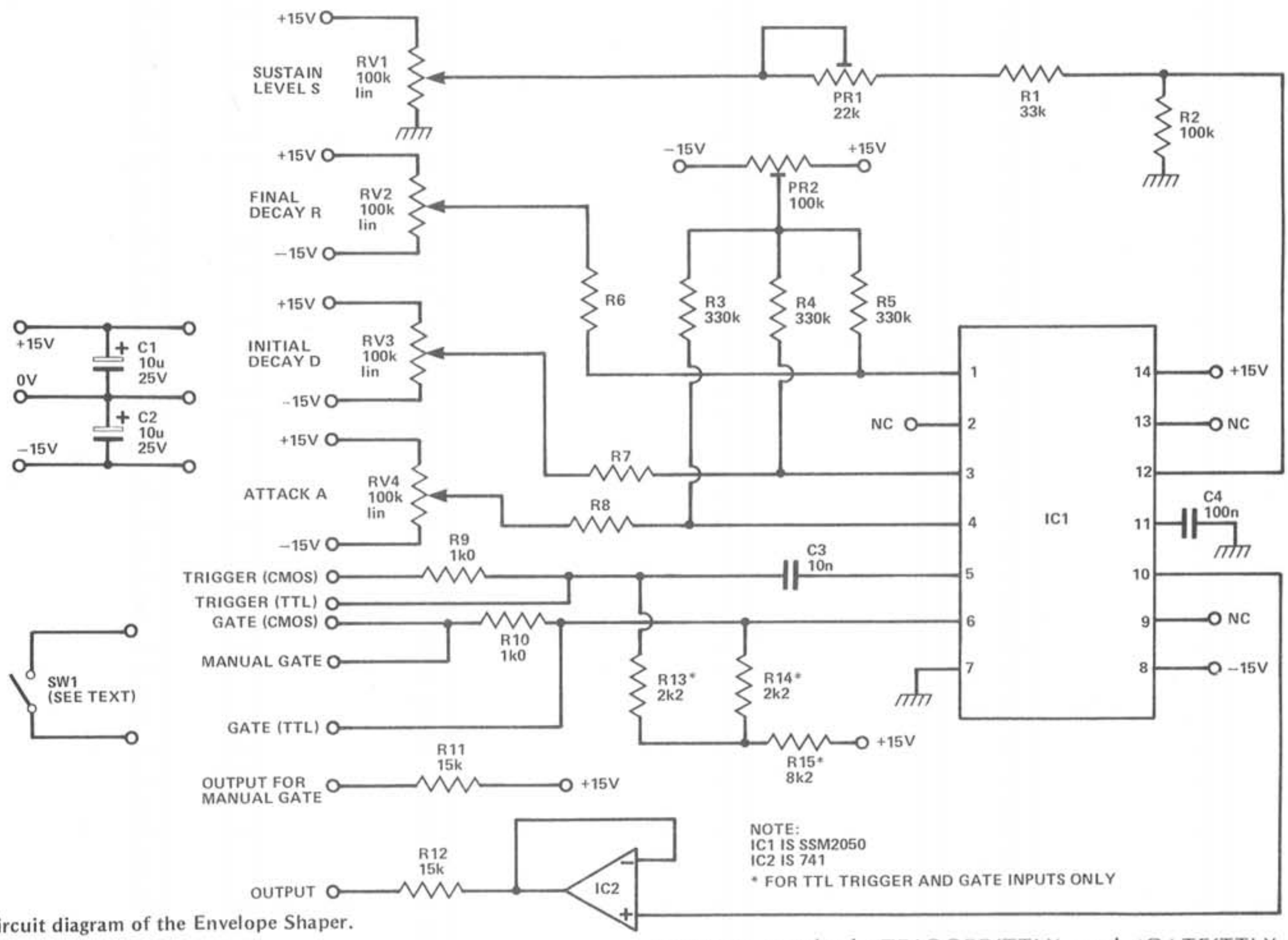


Fig.2. Circuit diagram of the Envelope Shaper.

HOW IT WORKS

The SSM2050 Voltage Controlled Transient Generator contains a voltage controlled resistor to generate the nominally exponential slopes and various logic devices to define the states. An attack flip-flop (AF/F) is set by the trigger pulse and reset by either NOT GATE or the attack comparator determining that the output has reached +10 V. Thus ATTACK = GATE and AF/F; INITIAL DECAY = GATE and NOT AF/F; FINAL DECAY = NOT GATE. Each state is characterised by a nominally exponential approach to a characteristic voltage; these being +13 V, sustain voltage and 0 V for attack, initial decay and final decay respectively.

The input stages of the SSM2050 logic inputs have a lateral PNP structure which protects them from excess voltages. Their sensitivity is 750 uA or 1V5 max., these being the minimum current and voltage required to trigger the SSM2050. For 5 V, 10 V and 15 V CMOS gate and trigger inputs these requirements are met using 1k, 10k and 15k resistors respectively to these inputs.

The attack, initial decay and final decay inputs have a nominal impedance of 3k1 and a time constant sensitivity of 18 mV/octave with a 100n timing capacitor (C4). An increasing positive voltage increases the time constant. Thus R6, 7 and 8, connected to +15 V via the rotary controls RV2, 3 and 4, will have nominal values of about 300k to achieve a five decade timing range from 2 mS to 20 S. The input impedance, however, varies by up to +25% between devices. Fortunately the impedance may be measured with a high input impedance ohmmeter as the resistance between pins 1 and 7 and so the appropriate scaling resistor may be selected by multiplying this resistance by 100 and adding 10k. The nearest E24 resistor is chosen and more precise adjustment of timing is achieved by injecting a small offset voltage via PR2 and R3, 4 and 5. The attack voltage may vary between 10 and 11 volts and PR1, R1 and R2 provide a means of matching the maximum sustain voltage to the peak attack voltage. The sustain level can then be varied from 0 to 100% of attack voltage using RV1.

As an additional safeguard the output of the SSM2050 has been buffered by IC2 configured as a voltage follower.

nections marked 'TRIGGER(TTL)' and 'GATE(TTL)' to provide the same function as before. For manual gating with TTL a push to break switch should be connected between 'GATE(TTL)' and OV, since the gate and trigger pulses are held high by the additional resistors.

Setting Up And Calibration

Provide a means of manually gating the envelope generator as described in the previous section and the switch may be constructed from two strips of metal, if necessary. Connect the output to a voltmeter set to a DC range of 15V and turn Attack control (RV4) to about 3 o'clock position and all other external controls to zero. Put SW1 in the ADSR position (gate and trigger commoned), turn PR1 fully anti-clockwise and PR2 about mid position. Apply power to the module, depress the manual button and keep held down while observing the voltmeter. The voltage should steadily rise and will probably take between 5 and 20 seconds to reach about 10V. Since the module is not calibrated the time taken may be outside of the range stated. The important point is that the voltage increases to a maximum of about 10V and then drops sharply to zero. If this response is observed then set Sustain control (RV1) to mid position and RV2, 3 and 4 to about the 3 o'clock position (a little less if the time to reach 10V was greater than 10 seconds in the previous step or a little more if the time was less than 5 seconds). Press button and hold down as before. The voltage should now rise to about 10V and then decay at the same rate to a voltage of approximately 5V and remain steady. On releasing the button there will be a final decay to about 0V. Finally, open switch SW1 to check

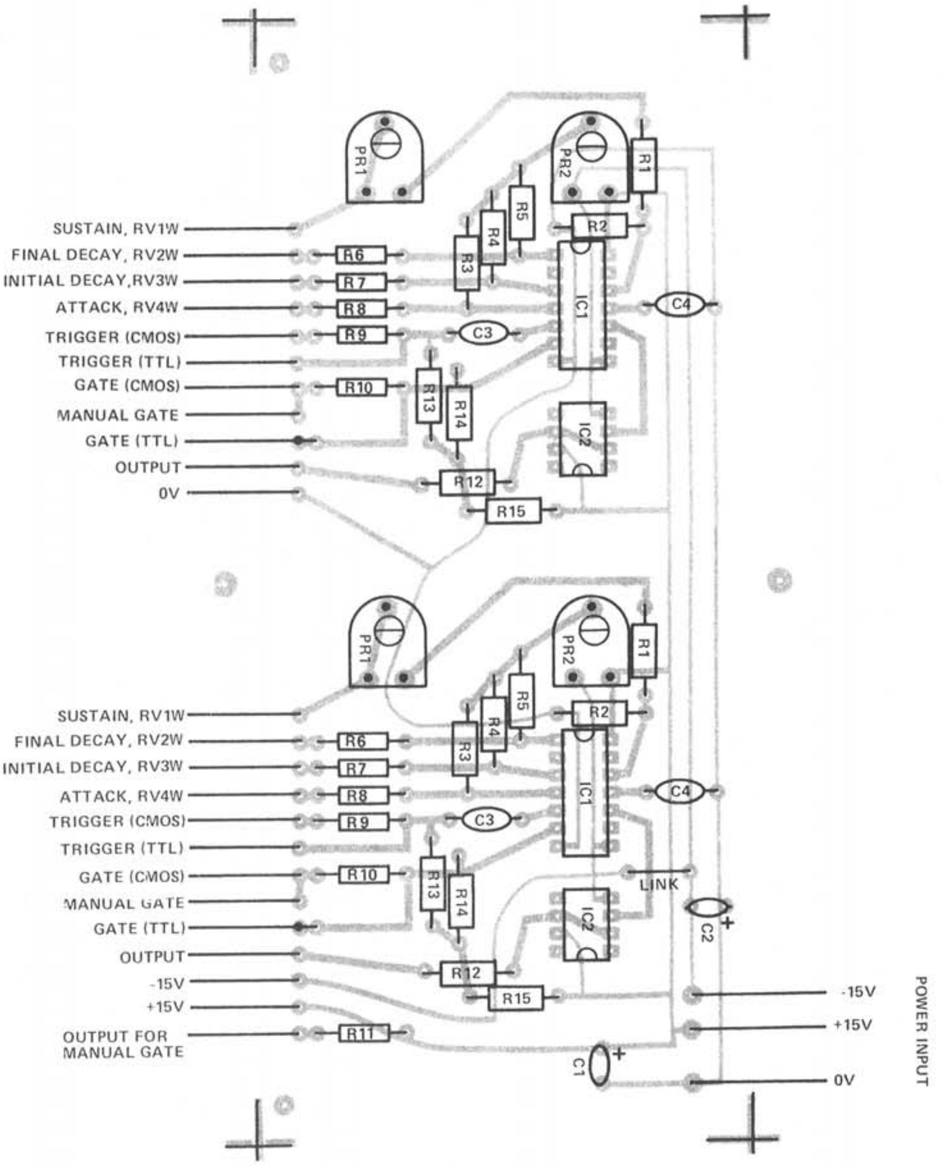


Fig.3. Component overlay.

PARTS LIST

Resistors ¼W, 5% Carbon film	
R1	33k
R2	100k
R3,4,5	330k
R6,7,8	see text
R9,10,12	1k0
R11	15k
Potentiometers	
RV1,2,3,4	100k linear
PR1	22k carbon
PR2	100k carbon
Capacitors	
C1,2	10u 25V electrolytic
C3	10n polyester
C4	100n polyester
Semiconductors	
IC1	SSM2050
IC2	LM741CN, or equivalent
Miscellaneous	
SW1	Sub Min SPST (or SPDT) switch

the AD response and repeat the last step. This time the voltage should rise to about 5V and maintain this value until the button is released which will initiate the decay to about zero. Note that in the AD mode the Initial Decay control (RV3) determines the attack time and the Sustain level controls the amplitude of the AD contour. The above demonstrates that all functions are operational.

The next step is to adjust the sustain voltage to match the peak attack voltage. Close SW1; set RV4 to about 3 o'clock; RV1 fully clockwise; RV2 and RV3 to the zero. Depress the manual button, observe the voltmeter and note whether there is a discernible drop in voltage after the attack has reached its peak. If so, turn PR1 clockwise and repeat the last step. Repeat until peak attack voltage and sustain level are matched. The adjustment to PR1 must be made in small increments so as to avoid having a higher sustain voltage than the attack voltage, otherwise malfunction of the SSM2050 can occur. It is therefore better to err on the safe side and wait until the envelope shaper is connected to the VCA at which time any mismatch between the two voltages can be checked by ear and a minor adjustment made to PR1 to correct it, if necessary.

The final step is to adjust the time constants and this calibration is only required for the Attack time control (RV4). The module should be in the ADSR mode (SW1 closed) and all other control pots set to zero. If an oscilloscope with a triggered sweep is available then the gate and the oscilloscope can be simultaneously triggered and PR2 adjusted to give an attack time of 2 mS when RV4 is at zero. An alternative method is to time the attack period, for example, by observing a voltmeter connected to the output and measuring the time between pressing the manual push button and the voltage dropping sharply. With the latter method adjust RV4 so that there is 10V0 at its wiper, trigger the module and adjust PR2 until the time taken is 9 S (slightly more than less). When this time is obtained turn RV4 to zero and check that a fast response time is obtained.

BUYLINES

A single 80-8 module with PCB and all components shown on the circuit diagram for CMOS inputs is available from Digisound Limited for £9.83 and a dual unit for £17.02, both inclusive of postage and VAT.

ETI

MODULE 80-9
DUAL VOLTAGE CONTROLLED AMPLIFIER

1. INTRODUCTION

The DIGISOUND 80-9 Dual Voltage Controlled Amplifier (VCA) is primarily intended for use in conjunction with envelope generators which will shape the signals entering the VCA to produce the desired sound contour. The design is based on the CEM 3330 to provide a module with excellent characteristics in respect of noise, feedthrough and distortion. The 80-9 will accept signal inputs up to +/-10V p-p and the controls are designed for voltages of 0 to +10V.

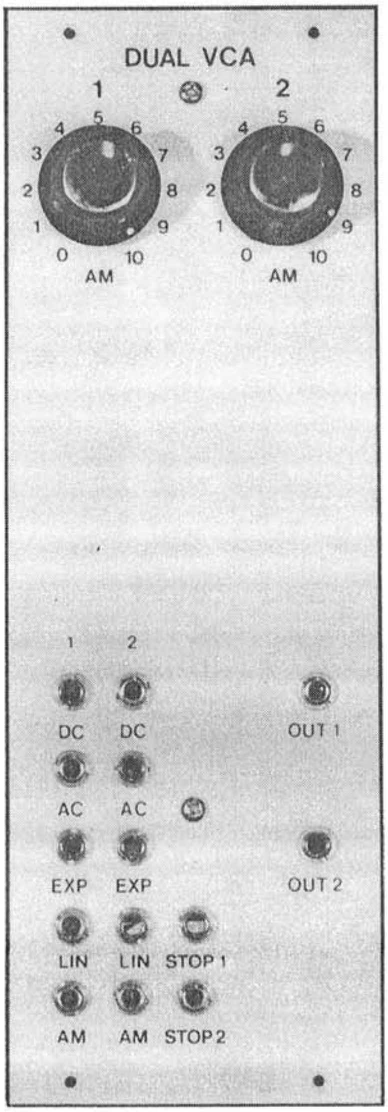


FIGURE 1. 80-9 PANEL

Figure 1 illustrates the panel lay-out of the Dual VCA. The signals into each half may be AC or DC coupled. Thus in its normal application the signal will be applied to the AC input whereas if the module is being used to control the amplitude of a control voltage the latter would be applied to the DC input. The control response may be either exponential (nominally 9dB/volt) or linear (approx. 10% per volt). When used with an envelope generator which has exponential contours (e.g., the DIGISOUND 80-10 or 80-18) the logical choice would be the linear control input and indeed this is useful for some settings of the envelope generator. In general, however, the exponential control input generally provides better sound contours. This input is also preferred when the envelope generator has significant control voltage feedthrough. The user should, however, experiment with both control inputs. A 0 to +10V control voltage when applied from, say, a LFO to the AM input will allow 0 to 100% amplitude modulation as the related potentiometer is rotated from 0 to 10. The other feature provided is a 'STOP' input which allows the sound output to be cut off sharply. This is particularly useful in live performance since it allows the sound to be terminated in time with the rest of the group.

When a signal is applied to the module it will pass to the output until a jack plug is inserted into either of the control inputs. This facility is useful when tuning the synthesiser.

These construction notes relate to PCB's marked DIGISOUND 80-9A which have a few minor modifications compared to the original 80-9. The two versions are completely compatible since the changes are: (a) a different method of compensating the linear control input; and (b) removal of the distortion trim which many users found difficult to adjust and gives little improvement over the typical 0.2% distortion obtained with the present design.

2. DESIGN

The CEM 3330 IC contains two voltage controlled amplifiers each of which consists of a variable gain cell and a log converter. The gain cell is the current-in, current-out type and has simultaneous linear and exponential controls. The log converter generates the logarithm of the linear control input current while the exponential control input is transmitted unchanged to its output.

Reference to the circuit diagram shown in Figure 3 and pins 1 to 9 of IC1 (also refer to the block diagram of the CEM 3330 shown in Figure 2) illustrates the basic features of the design. The signal input (pin 4) is a summing node and can, therefore, accept multiple signal inputs by simply adding more resistors although only one is employed in the 80-9 and with R16 = 100k signals should be kept within +/-10V p-p. R16 converts the signal voltage into a current while

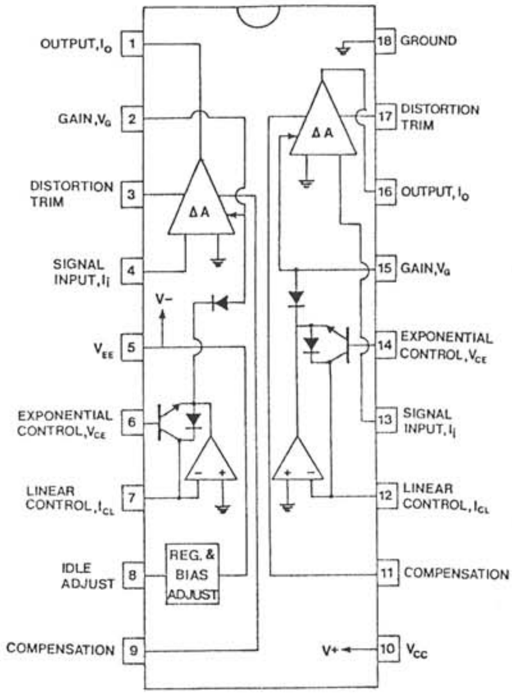


FIGURE 2. CEM 3330 IC

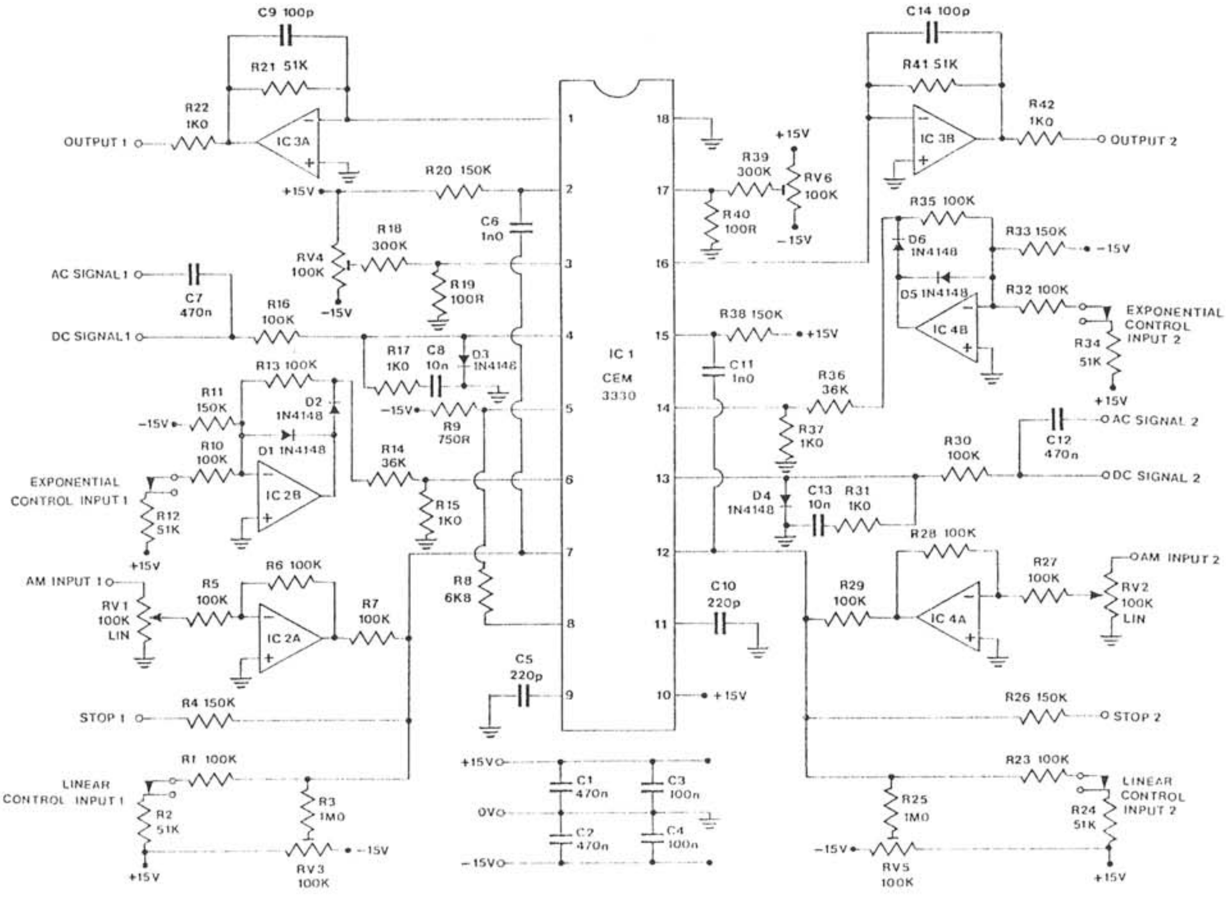


FIGURE 3. CIRCUIT DIAGRAM OF 80-9A DUAL VCA

components R17, C8 and C5 provide compensation and D3 prevents any latch-up problems. R20 connected to +15V produces a reference current for the gain cell and the 100uA developed produces good scale accuracy.

Both linear (pin 7) and exponential (pin6) control inputs are provided and the control voltages should range from 0 to +10V for full gain. Because the overall gain of the VCA is determined by both the linear and exponential inputs some voltage must be applied to one before the other will have any effect, that is, if one is at zero volts the gain will be zero regardless of the voltage on the other. Hence both control inputs are connected to +15V through resistors (R2 and R12) using the jack sockets of the module. For the exponential control via R10 the control voltage is inverted by IC2B to obtain the normal control sense of increasing gain with increasing control voltage. The linear input is a summing node and will accept multiple inputs - three are used in the 80-9. The normal linear control is applied to R1. The amplitude modulation (AM) operates by inverting the control voltage through IC2A while trimmer RV3 corrects for component tolerances and may be adjusted such that +10V applied to R5 will give zero output with RV1 fully clockwise (minimum resistance). The other linear input is the 'STOP' facility and with -15V applied to R4 the VCA will effectively be cut-off. The linear control input also requires compensation and with the method shown in the current data sheet (dated April 1980) there is a risk of the log converter breaking into oscillation when the gain of the linear control input is below -80dB. This is avoided by using a 1n0, or larger, capacitor (C6) between the linear control input and the reference current input.

One of the unique features of the CEM 3330 is the ability to vary the operating characteristics of the amplifiers between Class A and Class B by placing a resistor (R8) between the I_{EE} pin and the idle current adjust pin. The value of 6k8 provides a nominal standby current of 7uA (Class AB) which for the 80-9 application provides a good balance between distortion, bandwidth and control voltage feedthrough. The latter is further reduced by injecting a small voltage, using RV4, R18 and R19, into the distortion trim pin (pin 3). RV4, when adjusted, will also reduce distortion but note that the points for minimum feedthrough and minimum distortion are not identical. Diodes D1 and D2 have been added to the exponential control input to block negative voltages to pin 6 which would increase feedthrough.

The CEM 3330 requires a current limiting resistor when operated with negative supplies greater than -7V5 and this is provided by R9 which is the correct value for a -15V supply with the idle current employed.

The current output from the CEM 3330 is converted to a voltage by IC3A which also restores the polarity of the input signal. Since the VCA will normally be used as the last stage of a synthesiser the maximum gain has been set at 0.4 by R21 and this may be altered to suit users specific requirements.

3. CONSTRUCTION

The 80-9A PCB is printed with a component overlay which aids the construction stage. The overlay is reproduced in Figure 4 to allow checking of component placement after the module has been completed.

Take particular care with the orientation of the IC's and diodes. For the IC's, even after the DIL sockets have been installed the number '1', denoting pin 1, will still be visible on the PCB. With the diodes a band (line) is shown on the PCB overlay either before or after the diode number and this band indicates that the band on one end of the diode, denoting its cathode, should be towards the hole nearest the band on the PCB.

Before applying power to the module compare the completed PCB against Figure 4 and also carefully check the foil side for solder bridges and suspect joints.

The panel wiring is shown in Figure 5 and this diagram illustrates the components when viewed from the rear of the panel. The arrows and associated letters indicate that a wire connection must be made from the position shown to the front edge of The PCB which has a corresponding marking.

The jack sockets illustrated in Figure 5 are of the type supplied by Digisound Limited. The top connector, as shown, is the connection which is made with the jack plug when the latter is inserted. The lower connection is disabled by insertion of a jack plug and use is made of this for the control inputs to keep the VCA 'open' by applying voltages to these inputs via resistors R2, R12, R24 and R34. Finally, the tab under the socket is the ground connection. It is recommended that all of these connections are taken to the 0V connection on the PCB since this facilitates connection of the module to other equipment which may be operating from a separate power supply. It also provides the ground when interconnections are made with leads made with screened cable. The ground tabs may be soldered together using tinned copper wire but other panel wiring should be made with insulated wire. 1/0.6mm insulated wire is ideal for panel wiring since it retains any shaping and so allows a neat appearance to be obtained. Wires between panel components and PCB should be kept as short as practical.

Utilising the 'STOP' facility on each side is optional. One method would be to connect a footswitch to the jack socket such that when the switch is depressed it will switch -9V (battery within the footswitch) to R4, or R26, whose value should be changed to 91k. Alternatively, the jack socket may be replaced by a sub miniature push to make button switch. One side of the switch is connected to -15V, provided at the edge of the PCB, and the other side to R4, or R26.

4. CALIBRATION

Two adjustments are required for each side of the Dual VCA. These are.-

(i) To balance the AM input control voltage against the linear control input via R2 (or R24). Turn the AM control, RV1 (or RV2), fully clockwise and apply a +10V VCO signal to the DC input. Apply exactly 10 volts to the AM input socket, using a potentiometer as a voltage divider, and either examine the output of the VCA being adjusted with an oscilloscope set to maximum sensitivity or else listen to the output by connecting to an amplifier. Turn RV3 (or RV5) until the signal is seen, or heard, then reverse direction until the signal is _just_ cut off.

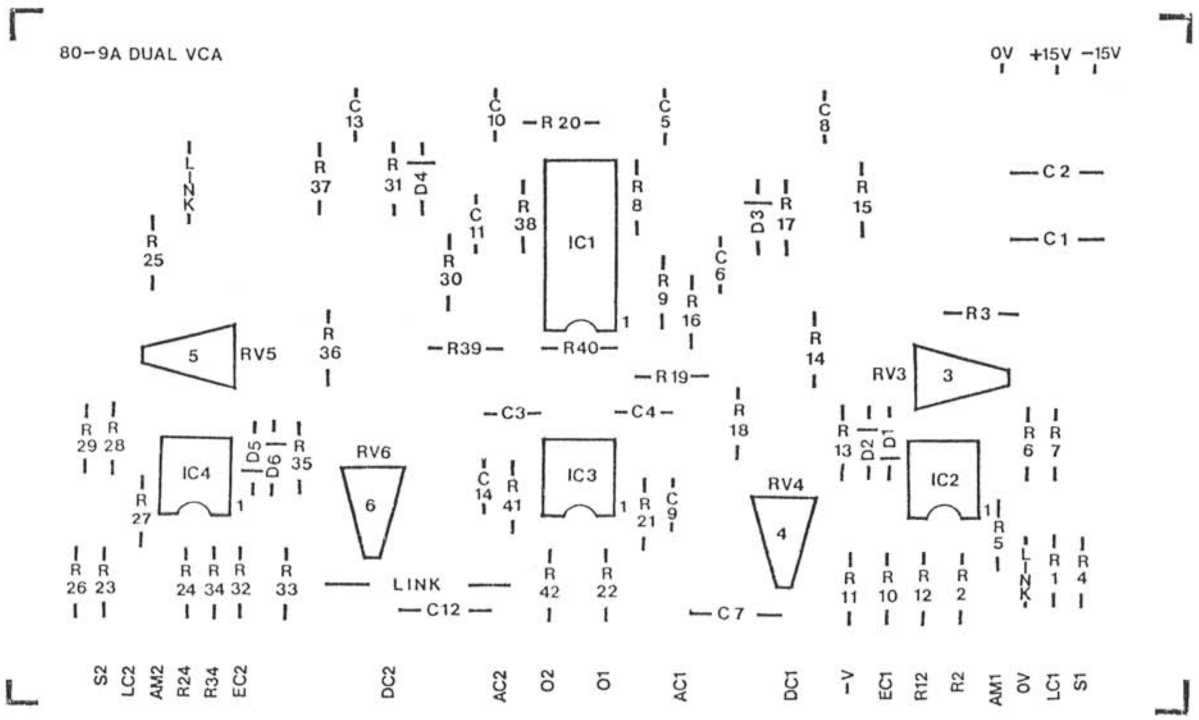


FIGURE 4. 80-9A COMPONENT OVERLAY

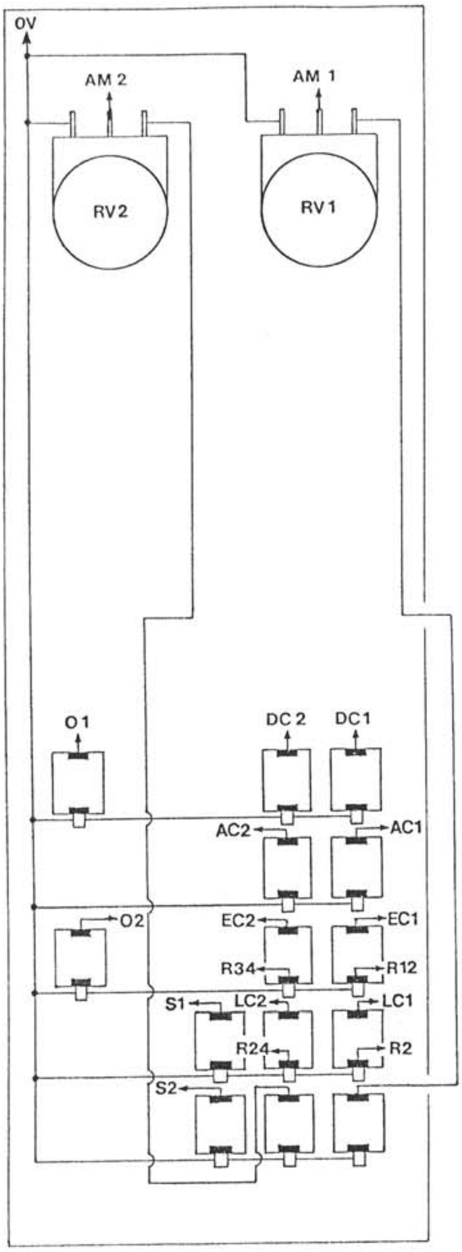


FIGURE 5. PANEL WIRING

(ii). Trimming control voltage may be done in two ways. First, with no connections to any VCA inputs adjust RV4 (or RV6) to give exactly 0V at the appropriate output. Better results will, however, be obtained by adjusting for minimum output voltage while a voltage to either control input is varied between 0 and +10V.

5. COMPONENTS

RESISTORS, 5%, 1/4w carbon film

R 1,5,6,7,10,13,16,23,27,28,29,30, 32,35	100k
R 2,12,21,24,34,41	51k
R 3,25	1M0
R 4,11,20,26,33,38	150k
R 8	6k8
R 9	750R
R 14,36	36k
R 15,17,22,31,37,42	1k0
R 18,39	300k
R 19,40	100R

POTENTIOMETERS/TRIMMERS

RV 1,2	100k lin.
RV 3,4,5,6	100k min. hor. carbon

CAPACITORS

C 1,2,7,12	470n polyester
C 3,4	100n polyester
C 5,10	220p polycarbonate*
C 6,11	1n0 polycarbonate*
C 8,13	10n polyester
C 9,14	100p polycarbonate*

*may be replaced by axial polystyrene

SEMICONDUCTORS

IC 1	CEM 3330
IC 2,4	LM 1458
IC 3	TL 072
D 1,2,3,4,5,6	1N4148

ENVELOPE SHAPER

Is it Mantovani or a Project 80 synthesiser? You can't tell the difference with this VCES designed by R.C. Blakey

Conventional ADSR envelope generators are adequate for most practical purposes since they are capable of providing a reasonable simulation of the amplitude envelopes of many musical instruments. The Project 80 Voltage Controlled Envelope Shaper (VCES) is provided for those who wish to obtain more realistic simulation or to obtain dynamic control over envelope shape. It is also a useful tool for innovative synthesis. The design incorporates the following features; bending of the standard exponential attack, decay and release curves to other shapes; alteration of attack, decay and release times by an external voltage thus allowing the envelope to be altered in proportion to the note played; the use of non linear sustain; built-in timer for re-triggering to create dual peak envelopes and also the generation of a delayed AD envelope.

Design and Application

The VCES is based on the CEM 3310 Voltage Controlled Envelope Generator produced by Curtis Electromusic Specialties. While it is well suited for use as a conventional ADSR envelope generator for both monophonic and polyphonic synthesisers the facilities provided on chip also make it ideal for configuring a complex envelope generator. The attack (A), decay (D), and release (R) parameters have a scale sensivity of 60 mV/decade (18 mV/octave) while sustain level (S) is linearly proportional to the voltage applied to pin 9. To facilitate generation of complex shapes each of the four inputs has been buffered by an op amp configured as a summer and our standard 0 to +10 V control voltages allow the A,D and R times to be varied from 2 mS to greater than 20 S. Likewise for the sustain input a voltage of 0 to +10 V varies the sustain level from 0 to 100% of the peak attack voltage which has also been normalised to +10 V.

The A, D and R responses follow an exponential curve. These characteristic curves may easily be altered in this design by taking a proportion of the output from the module and feeding it back to the appropriate input for the attack curve. The greater the amount of feedback the more convex the response and, although the overall time constant will increase, this may be adjusted over a wide range with the manual control provided. If the output is inverted prior to feedback then the attack curve will become concave in shape. Some of these curves are closer approximations to conventional instruments while others offer some novel responses. The shape of the decay curve, or the release curve, may be similarly altered and thus the VCES offers virtually unlimited scope for generation of envelope shapes. The use of low frequency waveforms to modify the time constants is also practical but setting up to obtain useful results is quite time consuming. Two attenuators, with or without inversion, are provided and the 80-5 Processor module may be used for distribution and attenuation when more complex patching is required.

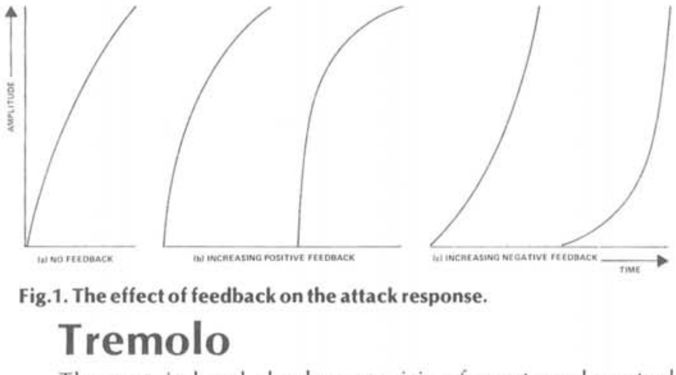


Fig.1. The effect of feedback on the attack response.

Tremolo

The sustain level also has provision for external control and one application is to apply a low frequency waveform to this input in combination with an attenuator and perhaps the manual control to produce a varying sustain. If this envelope is now used to control a VCA the effect is a tremelo only during the sustain part of the note. In the design both the upper and lower levels of the sustain control have been clamped for protection.

Another application for voltage control of envelope shape is the automatic alteration of the time constants or sustain level while the instrument is being played.

Time and Time Again

A simple timer has been incorporated in the design which allows re-triggering, or initial trigger delay, for periods up to about 2.5 S. The effect of re-triggering is to produce an envelope with two peaks, which is a transient effect exhibited by a number of conventional instruments. Often, however, as such instruments reach their peak output the

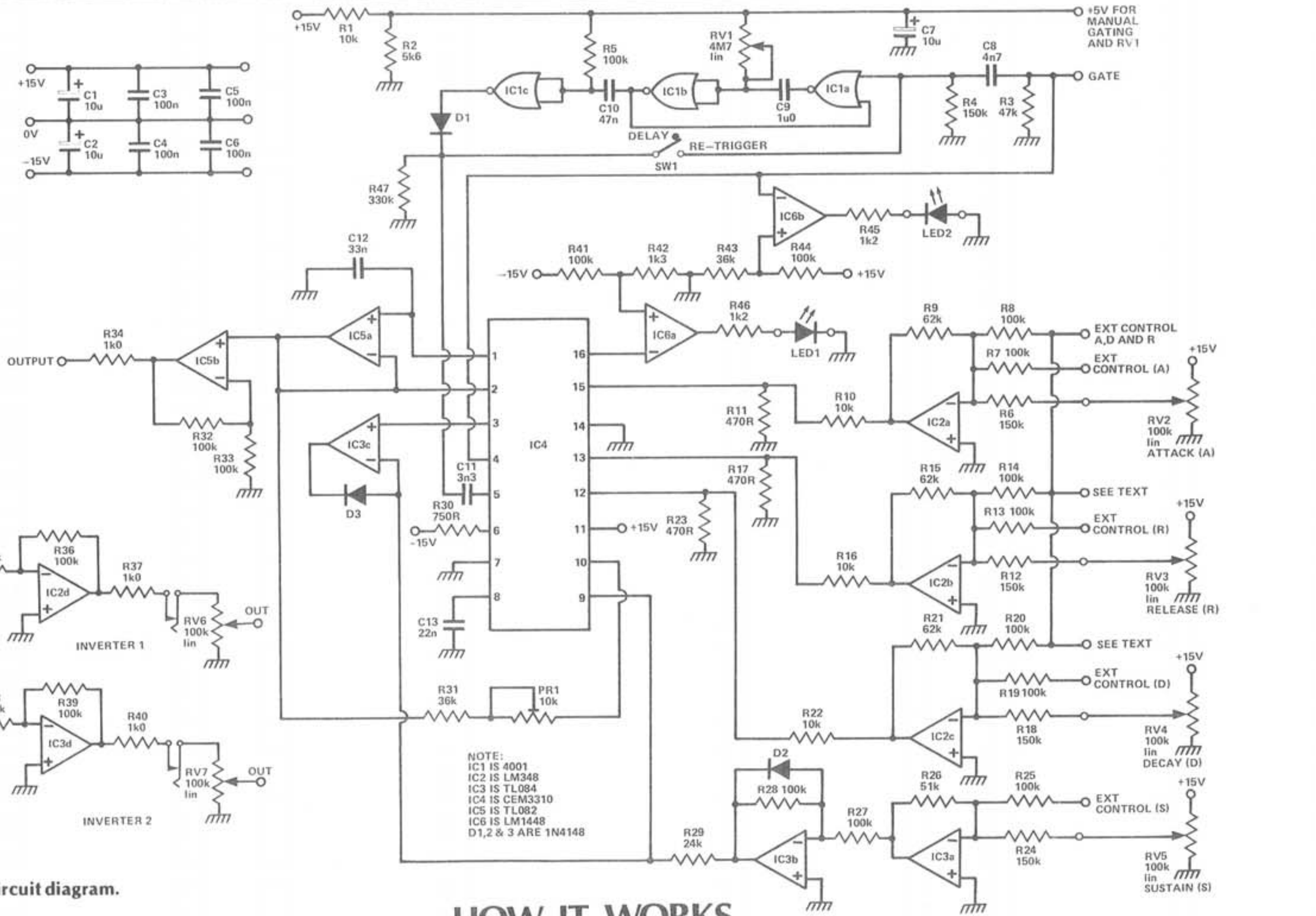


Fig.2. Circuit diagram.

HOW IT WORKS

The attack, decay and release control inputs (pins 15, 12 and 13 respectively) have a control sensitivity of 18 mV/octave and a 10k/470R attenuating network is place ahead of these inputs. The time constants of the attack, decay and release curves are determined by RxCx times the exponential multiplier, exp. $(-V_C/V_T)$; where Rx is R31 plus part of PR1; $C_X = C12$; $V_C =$ the control voltage at the appropriate pin; and $V_T = kT/q$. The values of Rx and Cx have been chosen to minimise errors and to retain the very low voltage feedthrough which is a feature of the CEM 3310. With the values used an increasing negative voltage at the 10k/470R attenuator will increase the time constant and a voltage from 0 to about −5V6 will give a range of 2 mS to 20 S for each time control input. To obtain both external and manual control of the time constants an inverting summer with a nominal gain of 0.62 has been placed ahead of the attenuating resistors; in the case of the attack control, using IC2a and associated resistors R6 to R9 ahead of attenuator R10/R11. Thus increasing positive voltages up to +10 V will now give the same control range. R7,8 are for external control voltages while RV2 via R6 provides manual control over the same range. R8 is connected with R14 (release control) and R20 (decay control) such that an external voltage applied to R8 will simultaneously change all three time constants.

Sustain level on the CEM 3310 is determined by the voltage at pin 9 and a voltage from 0 to +5 V will change the sustain level from 0 to 100% of the peak attack voltage. To obtain both manual and external control of sustain and retain the control polarity this input is preceded by IC3a and IC3b configured as two inverting summers with an overall gain of 0.5. Thus 0 to +10 V at R25 will produce the 0 to 100% sustain level control. Manual control is obtained with RV5 and R24. If the sustain voltage were to exceed [redacted] the peak attack voltage then the envelope will ramp up to this higher voltage level with undesirable results. Pin 3 of the CEM 3310 outputs the peak attack voltage and so the sustain level and pin 3 are connected to IC3c arranged as a precision peak follower to prevent the aforementioned situation.

The output buffer within the CEM 3310 (pin 2) has adequate drive capability for most applications but in this design it may be used to drive several inputs and overloading of the buffer will result in a loss of performance. The internal buffer has been bypassed by IC5a and the output increased to the Project 80 standard of +10 V by a non inverting amplifier, IC5b, which has a gain of two.

The attack output pin (pin 2) provides a voltage of between −0V4 and −1V2 only during the attack phase to provide a visual indication of the attack phase using LED 1. Pin 16 is connected to IC6a arranged as a comparator. IC6b is also a comparator and will turn on LED 2 when a gate voltage is present.

The CEM 3310 requires both gate and trigger pulses to generate an ADSR envelope. The 80-10 module is designed to operate with a gate voltage of +5 V and the trigger is generated by differentiating the gate pulse using C8 and R4 which is then applied to pin 5 via C11 (SW1 closed). To obtain the re-triggering and delay facilities IC1a and IC1b are used to form a monostable. The time delay is determined by the charging time of C9 via RV1 and when the voltage on C9 exceeds the threshold voltage of IC1b its output will go low and reset the monostable. IC1c is used to generate a second trigger pulse whose short duration is determined by the time required to charge up C10, to the threshold voltage level, via R5. With SW1 closed two trigger pulses are therefore generated when RV1 provides sufficient resistance to produce a noticeable delay. With SW1 open only the delayed trigger pulse is presented to pin 5 of IC4, which will allow it to operate in an AD mode.

To operate the CEM 3310 from ±15 V supplies it is necessary to place a series current limiting resistor on the negative supply line to pin 6. The value of R20 has been chosen to comply with the general equation $R_{EE} = (V_{EE} - 7.2)/0.010$. For the timer a nominal +5 V is derived from the voltage divider formed by R1,2 since power supply to IC1 is not critical. These latter components may be changed to suit other gate voltages.

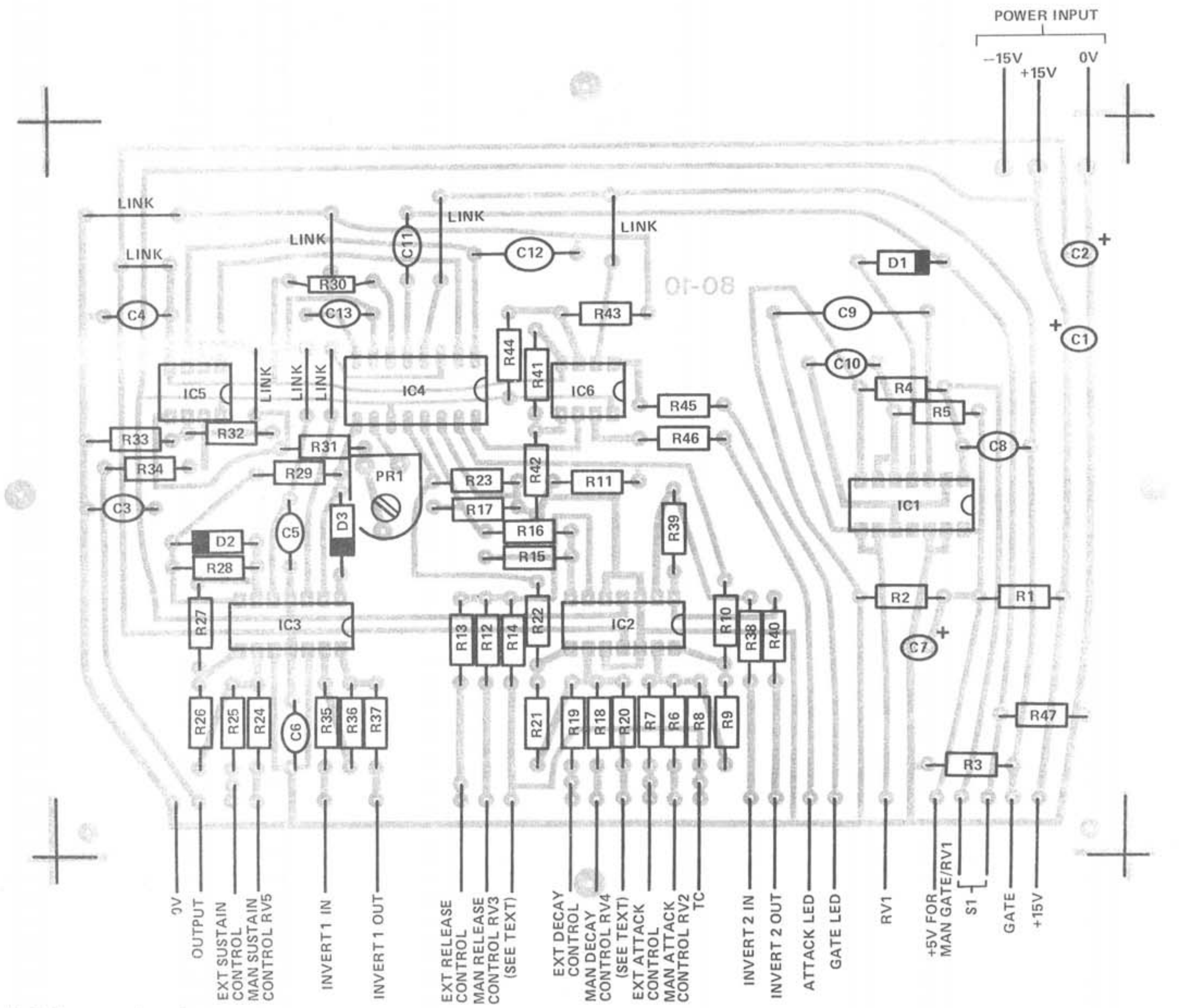


Fig.3. Component overlay

PARTS LIST

Resistors, ¼w, 5% carbon film unless specified.

R1	10k
R2	5k6
R3	47k
R4,6,12,18,24	150k
R5,7,8,13,14,19,20,25,27, 28,32,33,35,36,38,39, 41,44	100k
R9,15,21	62k
R10,16,22	10k 1% metal film
R11,17,23	470R 1% metal film
R26	51k
R29	24k
R30	750R
R31,43	36k
R34,37,40	1k0
R42	1k3
R45,46	1k2
R47	330k

Potentiometers

PR1	10k carbon
RV1	4M7 linear
RV2,3,4,5,6,7	100k linear

Capacitors

C1,2,7	10u 25 V PCB electrolytic
C3,4,5,6	100n polyester
C8	4n7 polycarbonate
C9	1u0 polyester
C10	47n polyester
C11	3n3 polycarbonate
C12	33n polycarbonate
C13	22n polyester

Semiconductors

IC1	4001B
IC2	LM248N
IC3	TL084CP
IC4	CEM3310
IC5	TL082CP
IC6	LM1458N
D1,2,3	1N4148
LED 1	Red LED
LED 2	Green LED

Miscellaneous

SW1	SPST sub-miniature toggle switch
PCB, case, etc.	

sound alters due to the presence of noise and complex waveforms in the transient. A better simulation of this effect is obtained by using two envelope generators, two sound sources, a dual VCA and mixing the outputs from the latter together. In this example the VCES timer is in the delay mode and will initiate an AD envelope when the trigger occurs. It should be noted that only AD envelopes are practical in the delay mode since if the sustain level is above zero the voltage will ramp up to the set level when the gate pulse is received.

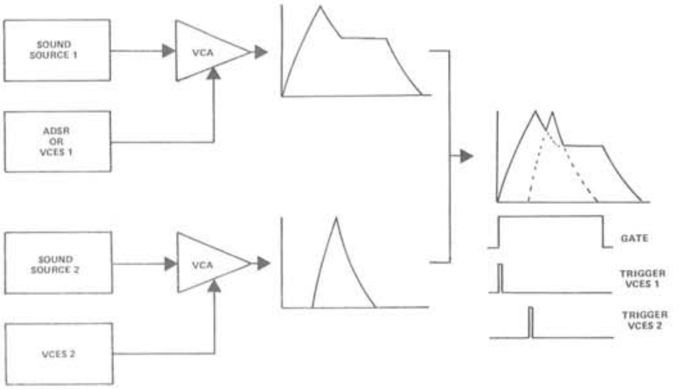


Fig.4. Patch for obtaining realistic transient effects.

Construction

In common with other Project 80 modules the Voltage Controlled Envelope Shaper may be panel mounted or installed in a Teka Alba A23G case. The latter, however, does not have sufficient panel area to neatly accommodate all of the facilities provided. In the cased module illustrated we have omitted the two inverters and controls RV6 and RV7.

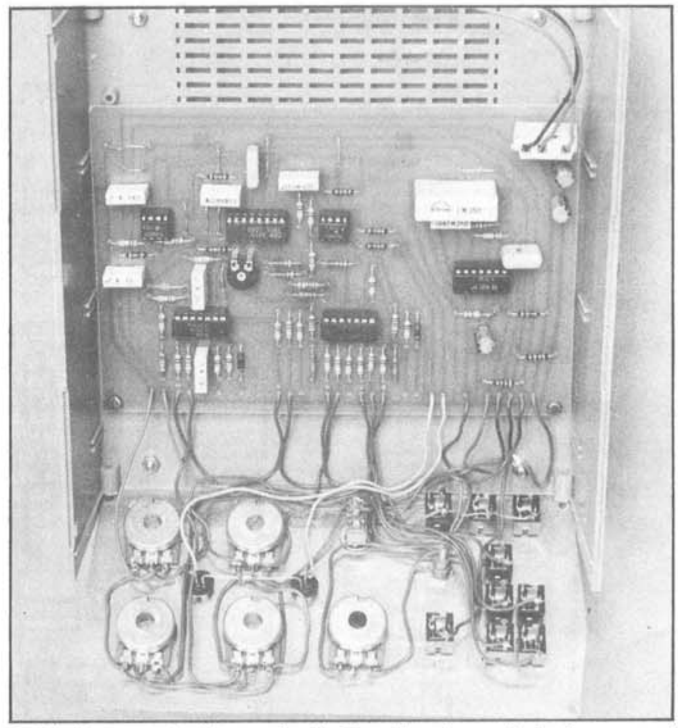
The panel markings for the inverters are −, + and A1 (or A2) with the latter being associated with the attenuating potentiometer RV6 (or RV7). Taking Inverter 1 as the example: R35 is wired to the jack socket marked − at the connection which makes contact with the jack plug; the output of the inverter (R37) is wired to the jack socket marked + but to the connection disabled when a jack plug is inserted, whereas the other connection on this socket is wired to RV6; finally the wiper of RV6 is wired to the make connection on the A1 jack socket. This allows a jack plug into the − socket to access the inverter and the output is obtained at A1, with attenuation when required. For non-inverted voltages which require attenuation these are obtained via the + socket with the output at A1.

One external control of attack, decay and release times is commoned and accessed on the PCB at R8. This allows all three time constants to be altered simultaneously and is connected to jack socket marked TC, denoting time constants. If required, however, the constructor may obtain two independent controls for each time constant by cutting the PCB tracks that join up the inputs of R8, R14 and R20. PCB connections are provided at R14 and R20 to cater for this modification.

The module may be manually gated by connecting a push-to-make switch from the +5 V line to the gate input.

Calibration and Testing

The attack, decay and release manual controls are numbered 0-10 for reference purposes since once external voltages are applied a time calibration becomes meaningless.



Tackle the control wiring methodically, otherwise you're in for a case of the wiring jungles.

PR1 allows more than one module to have the same time constants for a given input voltage. For precise calibration a triggered timing device is required but in most instances the following technique is adequate. Set PR1 to mid-position and connect a voltmeter between ground and the junction of R6 and R9. Turn RV2 until a voltage of −5V6 is obtained. Set all other control pots to zero. Gate the module manually and time the attack time as shown by the attack LED being on. The manual push button is held down until the LED goes off. Gate the unit several times to allow all components to stabilise for this long attack time and then commence adjusting PR1 to give an attack time of 20 S.

Note that the gate LED is only on while the manual button is depressed. Next set the attack time for a short duration, switch to re-trigger mode, turn RV1 fully clockwise and manually gate the module and keep push button held down until the test is complete. The attack LED should come on when the button is first depressed and again about 2.5 S later when the unit re-triggers. Keep settings the same but put the switch into delay mode. In this test the attack LED should come on about 2.5 S after manually gating the unit. Finally connect the output to a VCO which is in turn connected to an audio amplifier and set the attack, decay, sustain and release controls to about mid-position. Gate the module and release the push-button when a steady note is obtained. The test is a simple means of checking that the A,D,S and R functions are all operational. The functioning of the inverters may also be checked in the same way by putting the output from the VCES through the inverters without attenuation prior to the VCO. In this test the envelope will be inverted, that is, the frequency starting high, decreasing, holding steady and then finally going high again.

BUYLINES

ETI

80-10. Voltage Controlled Envelope Shaper (V.C.E.S.)

The errors noted in the September ETI are: Fig.2, IC 6 should read LM 1458; Components List, IC 2 should be LM 348N; and some text repetition in the 'How It Works' section.

Construction Advice:

a) RV 1 wired up as follows.-

b) LED's. Attack LED - lead adjacent to flat on lens goes to ground and other lead to R 46; Gate LED - R 45 to lead adjacent to flat on lens and other lead goes to ground.
c) As with the 80-8 we recommend use of a 2mm socket for the manual gating facility or the installation of a push button switch.
d) The panel is quite crowded and not shown in ETI. Constructors making their own panel can obtain the lay-out on receipt of a S.A.E. When using this panel lay-out, or the panel available for the kit, the attack and sustain pots should be mounted with their pins facing the edges of the panel so as to leave sufficient access to the LED's. The LED's supplied have a tight fitting mounting kit, which is beneficial, and require some patience.

NOTE:

Since submitting the project to ETI we felt that such a versatile module should have an independent trigger input. That is, an input that allows re-triggering from external sources. This addition does not affect any of the text in ETI. The external trigger input is located on the right hand side of the PCB input marked 'GATE LED' in Figure 3. This track goes to an additional diode (IN 4148), D4, which is sited over the hyphen of 80-10. The band of the diode faces the rear of the board. To avoid any confusion the diode placement is marked on the PCB. External re-triggering is obtainable with the switch in the 'delay' position and RV 1 fully anti-clockwise.

digisound

MODULE 80-11A
DUAL RING MODULATOR

1. INTRODUCTION

The DIGISOUND 80-11A Dual Ring Modulator is compatible with the earlier 80-11 and is AC coupled; will accept 10V p-p signal inputs and has a nominal unity gain. Signal feedthrough via the 'X' input, or so-called carrier input, may be suppressed to better than -50dB while the 'Y' input may be suppressed to -40dB, or better. The 80-11A is based on the MC 1496 (or LM 1496) balanced modulator IC, which is probably the most cost effective device to achieve the performance levels stated above.

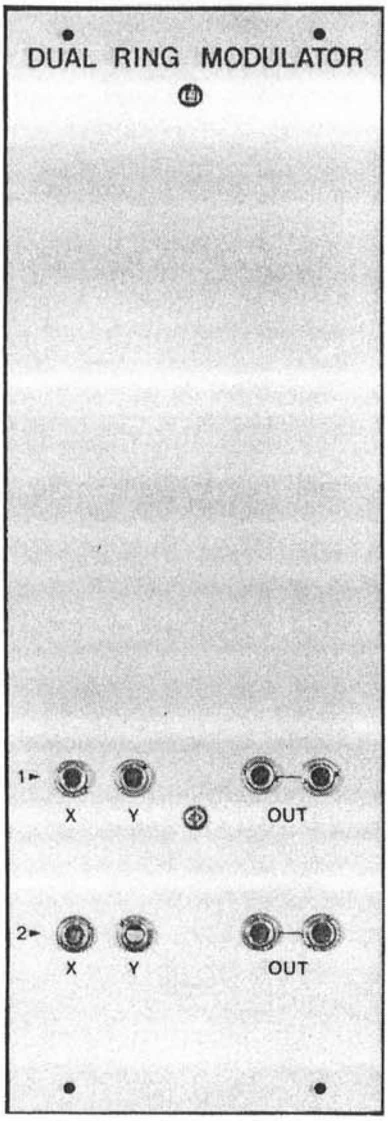


FIGURE 1. 80-11A PANEL

2. DESIGN

The two ring modulators in the 80-11A are completely independent, except for the common power supply and decoupling capacitors. The circuit diagram of each ring modulator is shown in Figure 2. The 1496 integrated circuit is designed for a variety of communications applications in which its output voltage is a product of a signal input (the 'Y' input) and a switching function, referred to as the carrier ('X' input).

The circuit is quite simple with the 'X' input being attenuated by R1 and R2 and the 'Y' input by R8 and R9. The signal to the latter may be much higher than the carrier input without adversely affecting feedthrough. The other resistors around the input provide the bias requirements for the IC while RV1 and RV2 and associated resistors allow trimming of the signal feedthrough. The output of IC1 goes to the differential amplifier built around IC2 which restores the signal to its original level, that is, two 10V p-p inputs will yield approximately a 10V p-p output. If, however, the module is used for frequency doubling by applying the same signal to both 'X' and 'Y' inputs then the output will be half the input signal.

3. CONSTRUCTION

The 80-11A PCB is printed with a component overlay which aids the construction stage. The overlay is reproduced in Figure 3 to allow checking of component placement after the module has been constructed. The separate ring modulators are also indicated since the same component numbers are used for both units.

Take the usual care with orientating the electrolytic capacitors (their negative wire is clearly indicated by a band of '-' symbols on the capacitor body) and also the integrated circuits. Even after installing the DIL sockets the number '1', denoting pin 1, will still be visible on the PCB. In any event compare the completed PCB against Figure 3 and also carefully check its foil side for solder bridges before applying power.

The panel wiring is shown in Figure 4 and this diagram illustrates the components when viewed from the rear of the panel. The arrows and associated letters indicate that a wire connection must be made to the PCB from the position shown. The PCB has a connecting point on its front edge with letters corresponding to those in Figure 4.

The jack sockets in the diagram are of the type supplied by Digisound Limited. The top connection, as illustrated, is the connection made with the jack plug when the latter is inserted. The lower connection is disabled by insertion of a jack plug. Finally, the tab under the socket is the ground connection. It is recommended that all of the ground connections are wired to the 0V line since this facilitates connection of the module to other equipment which may be operating from a separate power supply. The ground tabs may be soldered together using tinned copper wire but other panel wiring should be made with insulated wire. 1/0.6mm. insulated wire is ideal for panel wiring since it retains any shaping and allows a neat appearance to be obtained. Wiring should be kept as short as practical.

4. CALIBRATION

Setting up the ring modulators is simple but needs to be done carefully. Connect a sinewave (or triangle) of about 1kHz from a VCO to the 'X' input of ring modulator 1 and connect its output to an oscilloscope; or a sensitive AC voltmeter; or to an amplifier. Adjust RV2 until the signal being measured, or heard, is at a minimum. This setting is quite critical and hence the use of multi-turn trimmers. Now connect the signal to the 'Y' input and adjust RV1 for minimum signal. Repeat these steps for the second ring modulator. Some minutes will now have elapsed since adjustment of the first unit was made and so repeat the whole procedure for both ring modulators. Mainly due to the biasing requirements of the 1496 it is desirable to operate the 80-11A from a stable power supply, such as the DIGISOUND 80-1 or 80-1A.

The importance of the above steps lies in the fact that the points of minimum feedthrough correspond with the correct balancing of the inputs for modulation.

The level of feedthrough obtained varies somewhat between IC's but typically with a 10V p-p signal the feedthrough of the 'X' input may be trimmed to between 10mV (-60dB) and 20mV (-54dB). Typical values for the 'Y' are between 40mV (-48dB) and 80mV (-42dB).

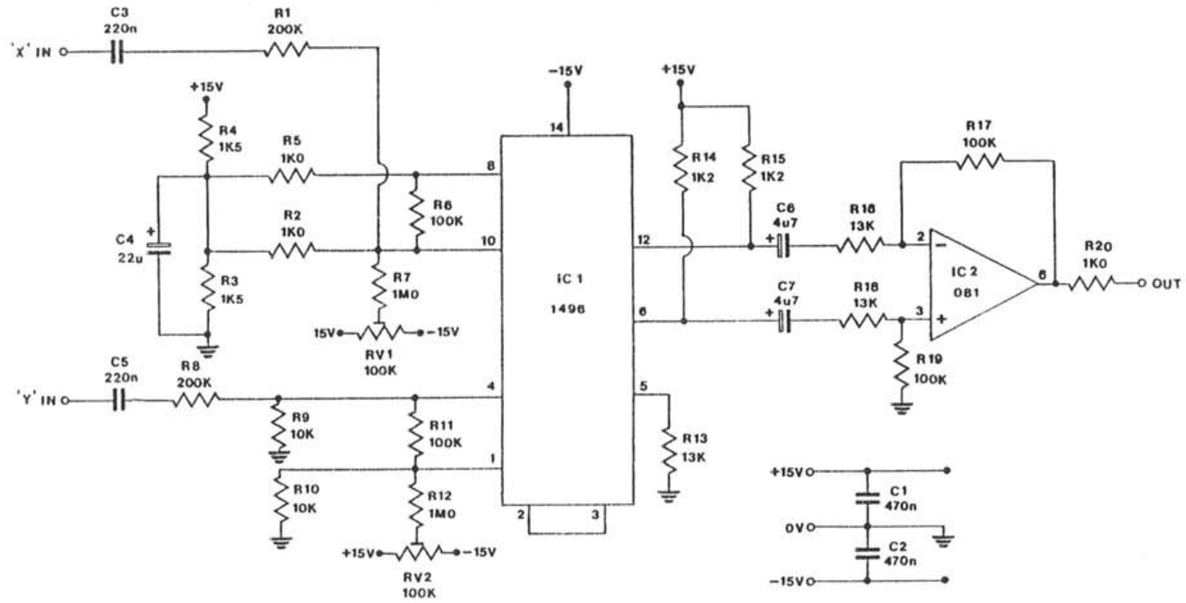


FIGURE 2. CIRCUIT DIAGRAM FOR ONE 80-11A RING MODULATOR

5. USING

The ring modulator has been widely used as a sound modifier right from the birth of electronic music. The popularity of the device lies in the fact that it produces the sum and difference of the frequencies of two signals applied to it. When these signals have high harmonic content the resultant combination of frequencies is extremely complex and bell-like in character. While a tolerable imitation of bells, gongs and chimes may quickly be obtained, a realistic synthesis reqires a great deal of patience and often the use of more than one ring modulator.

As inferred earlier, a ring modulator may also be used as a frequency doubler by applying the same signal to both the 'X' and 'Y' inputs. If, however, the signal is other than a pure sinewave the intermodulation of the harmonics will produce complex outputs.

In normal use, i.e., with two independent signals, it does not matter which signal is connected to the 'X' or 'Y' input - the end result is the same. The only point to note is that the original signal will be suppressed most when applied to the 'X' input and so this may govern selection in some instances. Signals should be within the normal audio range of 20Hz to 20,000Hz.

Further information on using ring modulators will be found in 'Using the DIGISOUND 80 Modular Synthesiser'.

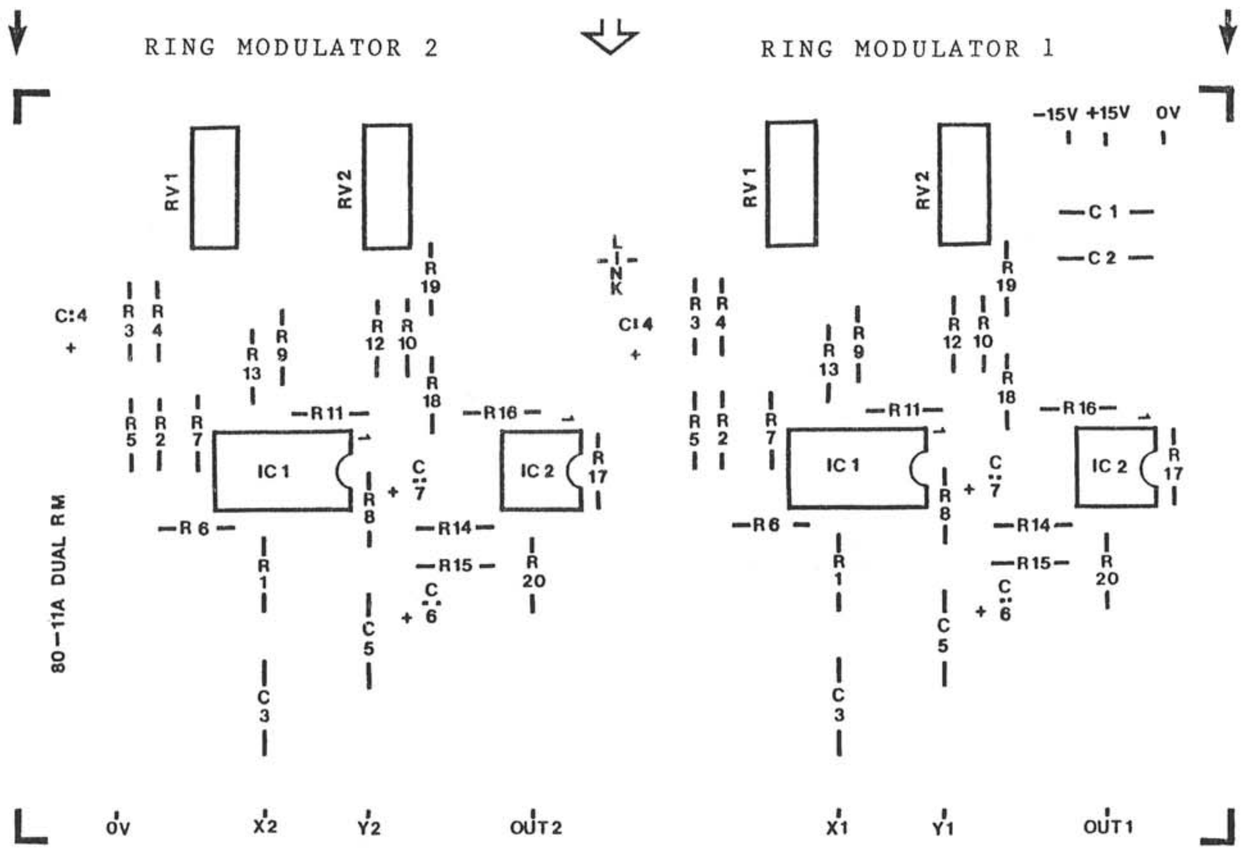


FIGURE 3. COMPONENT OVERLAY FOR 80-11A

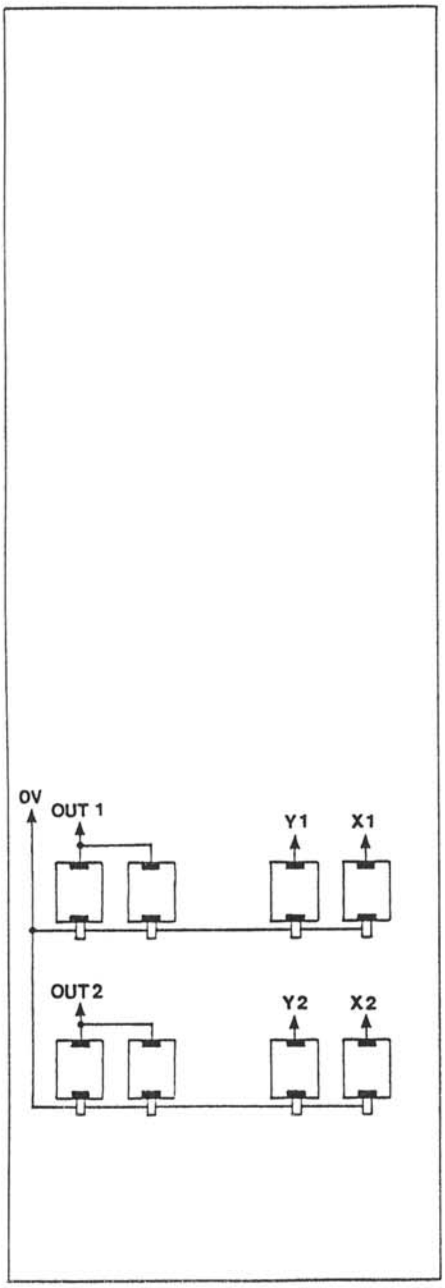


FIGURE 4. PANEL WIRING

6. COMPONENTS

A kit will consist of two sets of the following components (plus DIL sockets), except for C1 and C2 which are common to the two units.

RESISTORS, 5%, 1/4w carbon film	
R1,8	200k
R2,5,20	1k0
R3,4	1k5
R6,11,17,19	100k
R7,12	1M0
R9,10	10k
R13,16,18	13k
R14,15	1k2

CAPACITORS	
C1,2	470n polyester
C3,5	220n polyester
C4	22u PCB electrolytic
C6,7	4u7 PCB electrolytic

TRIMMERS	
RV1,2	100k cermet multiturn

SEMICONDUCTORS	
IC1	MC 1496P (or LM 1496N)
IC2	TL 081

digisound DUAL RING MODULATOR 80–11

INTRODUCTION

The ring modulator, or four quadrant multiplier, has been widely used as a sound modifier right from the birth of elect--ronic music. The popularity of the device lies in the fact that it produces the sum and difference of the frequencies of two signals applied to it and when these signals have a high harmonic content, e.g., sawtooth waveforms, the resultant combination of frequencies is extremely complex and bell-like in character. While a tolerable imitation of bells, gongs and chimes can be quickly obtained a realistic synthesis requires a great deal of patience and invariably the use of more than one ring modulator.

Another sound synthesis technique that may be explored with two ring modulators is that of intermodulation. For this a sound source is modulated by one oscillator and the output goes both to a tape recorder and a second ring modulator where it is further treated by another oscillator prior to mixing at the tape recorder. Pleasing results can be obtained and various signal sources, such as voice and traditional musical instruments, should be experimented with. The modulating source does not have to be an oscillator operating at a fixed frequency and interesting effects may be obtained using rhythmic patterns generated via the keyboard or other means.

Ring modulators may also be used as octave shifters since if the same signal is applied to both inputs then the output will be double the input frequency.

DESIGN FEATURES

The 80-11 utilises the SSM 2020 Dual VCA since it will accept large signals and thus maintain a high signal to noise ratio with a low total harmonic distortion.

The two inputs, X and Y, on each side of the SSM 2020 have been AC coupled so as to accept a wide variety of signal sources with amplitudes up to 10V p-p. The circuit is designed for approximately unity gain in the ring modulation mode but with frequency doubling the gain will be about 0.5.

A ring modulator produces the sum and difference of the frequencies applied to the X and Y inputs and the original signal frequency should be suppressed. The degree of suppres--sion achieved with the SSM 2020 in this design varies somewhat between devices. For the X input the results are usually comparable to, or better than, widely used four quadrant multi--pliers and is typically between -50 and -60dB. For signals applied to the Y input the suppression lies in the range -40 to 50dB.

CONSTRUCTION AND SETTING UP

The circuit diagram for the Dual Ring Modulator is shown in Figure 1 and the component overlay for the PCB is illustrated in Figure 2. The essential point to note in construction is that IC 2 is reversed in orientation to IC's 1 and 3.

Setting up requires adjustment of the pre-sets, PR1 to PR4, so as to optimise the suppression of the original signals and achieve accurate multiplication. First set PR 1 to 4 at their mid positions and apply power. Next apply a 10V sinewave of about 1 kHz from a VCO to the the Y input of Ring Modulator 1 (Y 1) with the X 1 input grounded and connect RM 1 output to an amplifier or oscilloscope. PR 2 is then adjusted for minimum feedthrough. The signal is then applied to X 1 with Y 1 grounded and PR 1 adjusted for minimum output. Leave for a few minutes with the signal connected and repeat the last two steps. The setting of the pre-sets is very precise for minimum feedthrough. Repeat the procedure for RM 2, i.e., a signal applied to Y 2 input with X 2 grounded and adjusting PR 4 followed by signal to X 2 input with Y 2 grounded and adjusting PR 3. Again repeat these two steps after a few minutes.

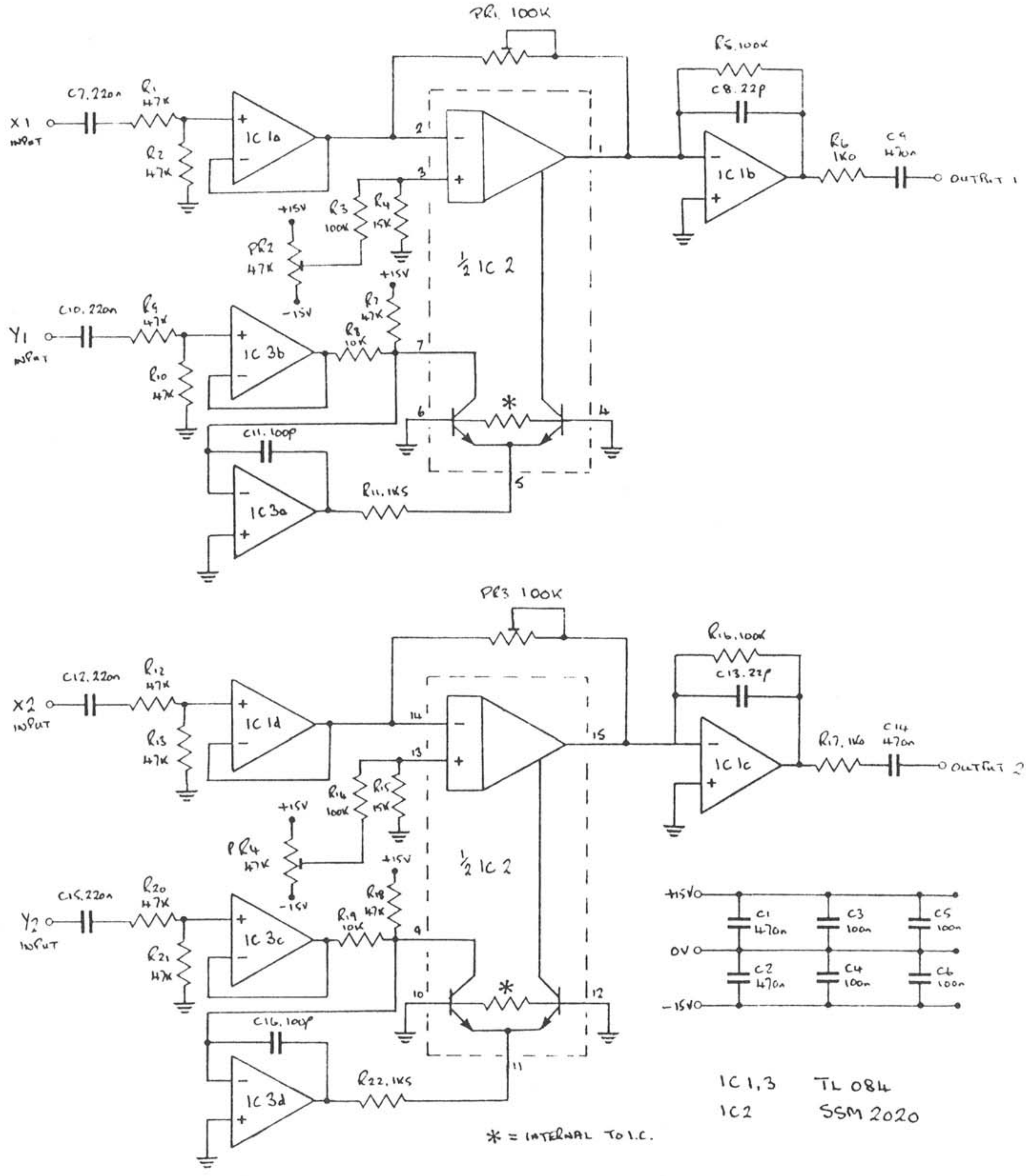


FIGURE 1

80-11. DUAL RING MODULATOR

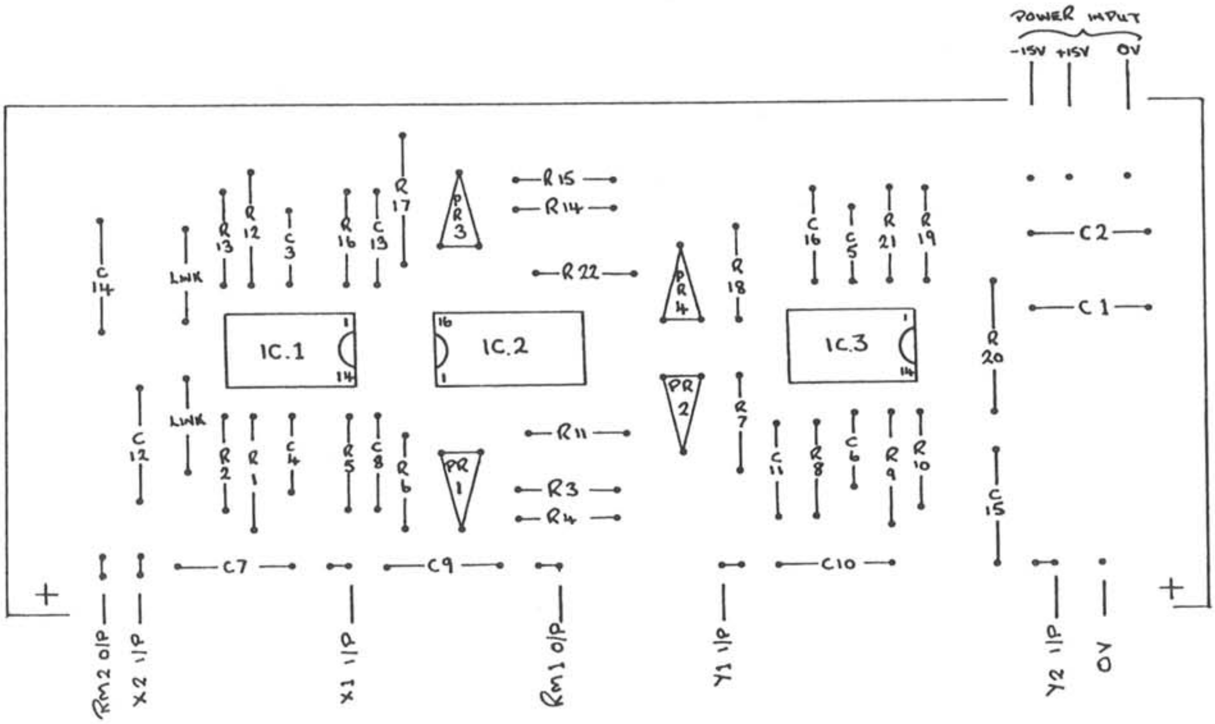


FIGURE 2. COMPONENT OVERLAY

COMPONENTS FOR 80-11

RESISTORS, ¼W, 5%, carbon film

R 1,2,7,9,10,12,13,18,20,21	47k	R 6,17	1k0
R 3,5,14,16	100k	R 8,19	10k
R 4,15	15k	R 11,22	1k5

TRIMMERS

PR 1,3	100k carbon	PR 2,4	47k carbon

CAPACITORS

C 1,2,9,14	470n polyester
C 3,4,5,6	100n polyester
C 7,10,12,15	220n polyester
C 8,13	22p polystyrene
C 11,16	100p polystyrene

SEMICONDUCTORS

IC 1,3	TL 084CN
IC 2	SSM 2020

INTRODUCTION

Noise sources are essential for the synthesis of many sounds. These range from musical instruments, such as pipe organ and percussive devices; to natural sounds like wind, rain and surf; and to man-made sounds as occur with steam engines, explosions and gunfire. The noise sources provided on the 80-12 module are white noise which has the characteristic hissing sound, pink noise which is deeper in intensity, and a low frequency noise which is sometimes referred to as red noise. How the latter sounds depends a great deal on the size and frequency response of the speakers used but it can literally bring the house down! The low frequency noise may also be used as a random modulating source.

A Sample & Hold circuit is also incorporated in the 80-12 which allows external sound sources to be sampled and converted into control voltages for, say, operating a VCO. The Sample & Hold unit therefore provides a means by which the synthesiser plays itself. For example, if the sound source is a sawtooth waveform then the Sample & Hold will convert this waveform to a series of discrete voltage steps with the amplitude of each step being related to the amplitude of the sawtooth while the duration of each step is determined by the sampling rate. Virtually any external signal may be used although in many instances these will have to be pre-amplified by the external input module (80-13). The low frequency noise source may be used directly to produce a completely random set of control voltages and is one of the reasons for incorporating the two circuits within the same module. Some extremely interesting effects may be obtained with the Sample & Hold unit and these will be discussed in 'Using the Synthesiser' which is incorporated in the published booklet.

DESIGN FEATURES

White noise is obtained by reverse biasing of an NPN trans--istor followed by two stages of amplification to obtain an output of between 5 and 10V p-p.

White noise has equal energy per cycle whereas pink noise has equal energy per octave. To obtain the latter from white noise a -3dB/octave filter must be used. Normally such a filter requires a high component count but a good approximation has been obtained with just two resistors and two capacitors in the active filter built around IC 1c (Figure 1). The amplitude of the pink noise is also in the range of 5 to 10V p-p.

Low frequency noise is obtained by low pass filtering of the pink noise using a 6dB/octave filter with a cut off frequency of about 16Hz. Its amplitude is 10V p-p.

The Sample & Hold circuit uses the principle of gating a FET, Q2, on and off and storing the sampled voltage on a capacitor, C 16, which is buffered by a voltage follower, IC5b. An internal clock, based on a CMOS 555 timer, allows incoming signals to be sampled at rates between 1 cycle per 4.5 seconds and 25Hz by varying RV 1. The signal input may be attenuated by RV 2 and so this will provide a means of controlling the voltage range from the Sample & Hold output. Provision is made for using an external clock and the pulse output from the 80-2 VCO, or 80-3 VCLFO, is suitable. To obtain discrete voltage output steps, however, it is necessary to use a narrow pulse width by adjusting the PWM control on the VCO's. Varying the pulse width does nevertheless produce an additional effect. For example, if the signal source is a sawtooth waveform then a short pulse will give discrete steps and if this output is taken to another VCO then its frequency will vary stepwise. If the pulse width is now increased then part of the rising amplitude of the saw--tooth will appear at the output and the VCO will begin to glide from one frequency to another. Achieving the desired result depends on the clock rate and the amplitude of the signal being sampled and the latter may be attenuated by RV 2. The module also has an output from the internal clock but due to the narrow width it is only suitable for synchronising purposes or as a trigger for the 80-10.

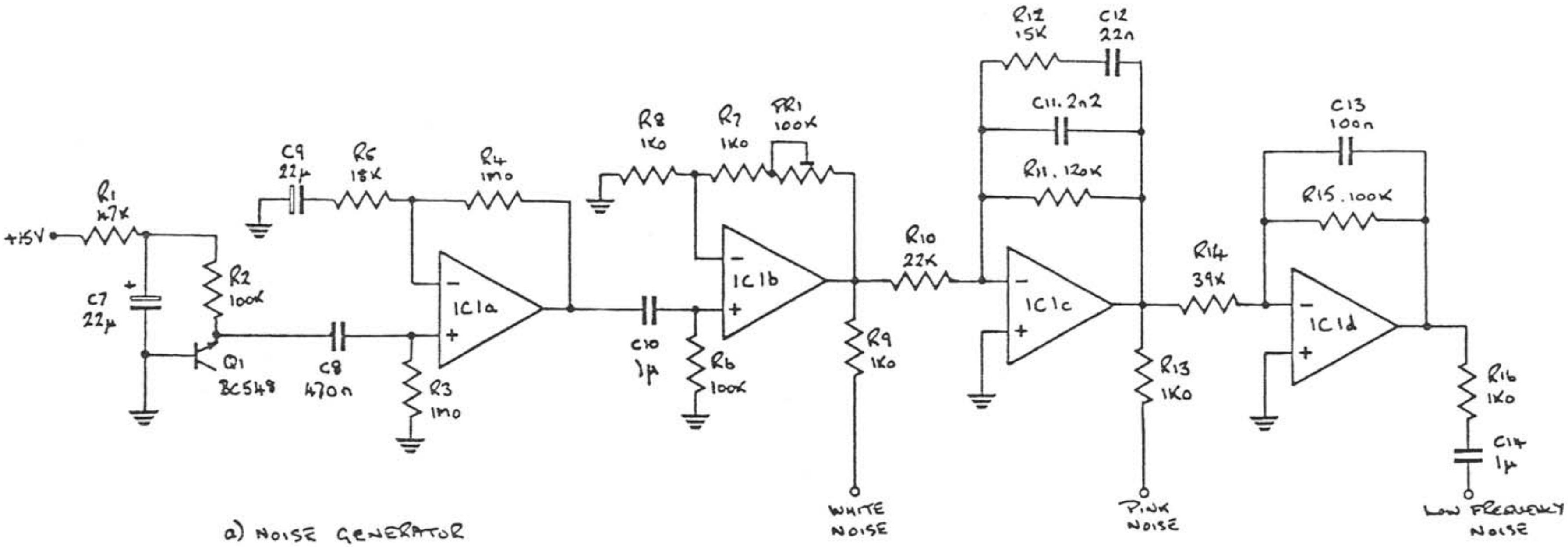
CONSTRUCTION AND SETTING UP

The complete circuit diagram is shown in Figure 1 and the component placement on the PCB is illustrated in Figure 2. The usual precautions on orientation of the I.C.'s, transistors, diodes and electrolytic capacitors must be observed.

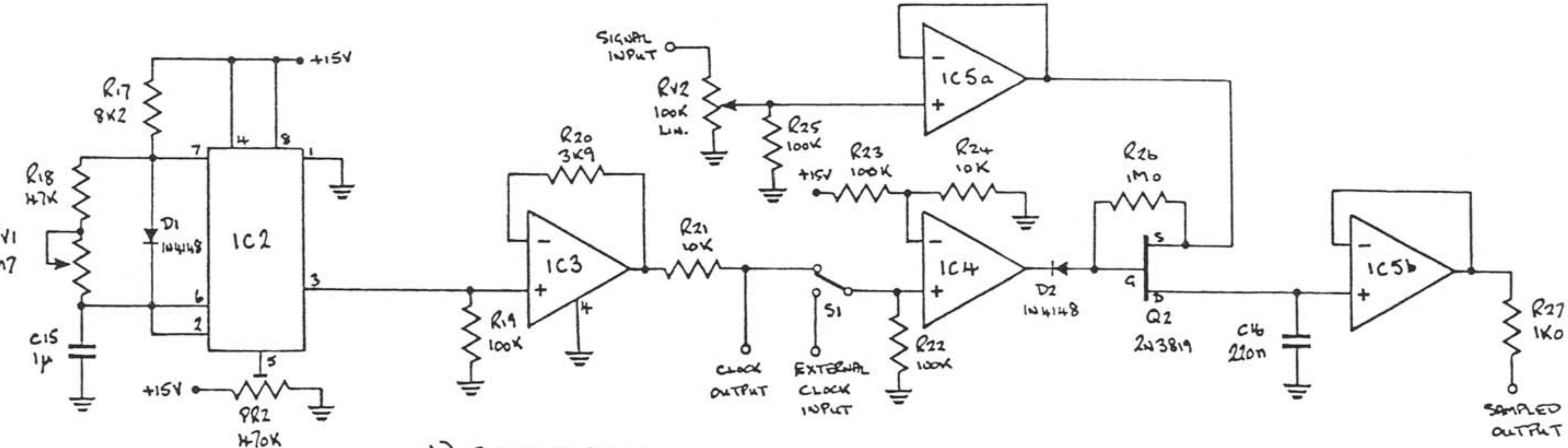
Setting up the noise generator is simply achieved by adjust--ing PR 1 to obtain the desired amplitude. This may be done in three ways: (a) using a DMM on the white noise output and adjus--ting PR 1 until a reading of about 1V8 to 3V2 is obtained on the AC range; (b) using an oscilloscope and observing the white noise output while adjusting PR 1 to obtain a peak to peak output of 5 to 10 volts; or (c) connecting the output of the low frequency noise at R 16 to a DC voltmeter and adjusting PR 1 until occasional readings of about -7V are obtained when observed over a period of a few minutes. With most analogue meters the above readings will be approximately halved. With some trans--istors it may be possible to obtain the correct amplitude for the white and pink noise using PR 1 but the amplitude of the low noise will be too high. In the latter circumstances the best approach is to try another NPN transistor rather than alter the value of R 14. A transistor socket is provided with the kit to facilitate this change should it be necessary.

Only one adjustment is required for the Sample & Hold section, namely, to adjust PR 2 to obtain a short pulse output from the internal clock based on IC 2. This adjustment should be made in the following sequence -

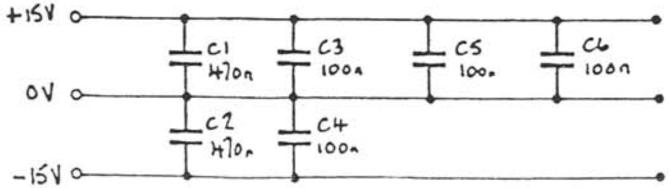
a) Connect the low frequency noise output to the signal input of the Sample & Hold with RV 2 fully clockwise.
b) Connect the output of the S & H to a VCO with the coarse frequency control set to mid position and one of the waveforms connected to an amplifier.
c) Set RV 1 (clock rate) to about mid position.
d) Turn PR 2 fully anti-clockwise and switch power on.
e) After a few seconds delay slowly turn PR 2 clockwise until the VCO output starts changing frequency in a step-wise manner.
f) Finally check that the VCO continues to change step-wise over the full range of the internal clock (RV 1) and if necessary turn PR 2 further clockwise until this condition is achieved.



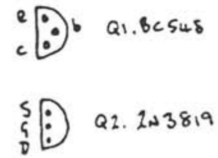
a) NOISE GENERATOR



b) SAMPLE AND HOLD



IC1	LM348
IC2	ICM 7555
IC3,4	CA3140E
IC5	TL082



PIN VIEWS

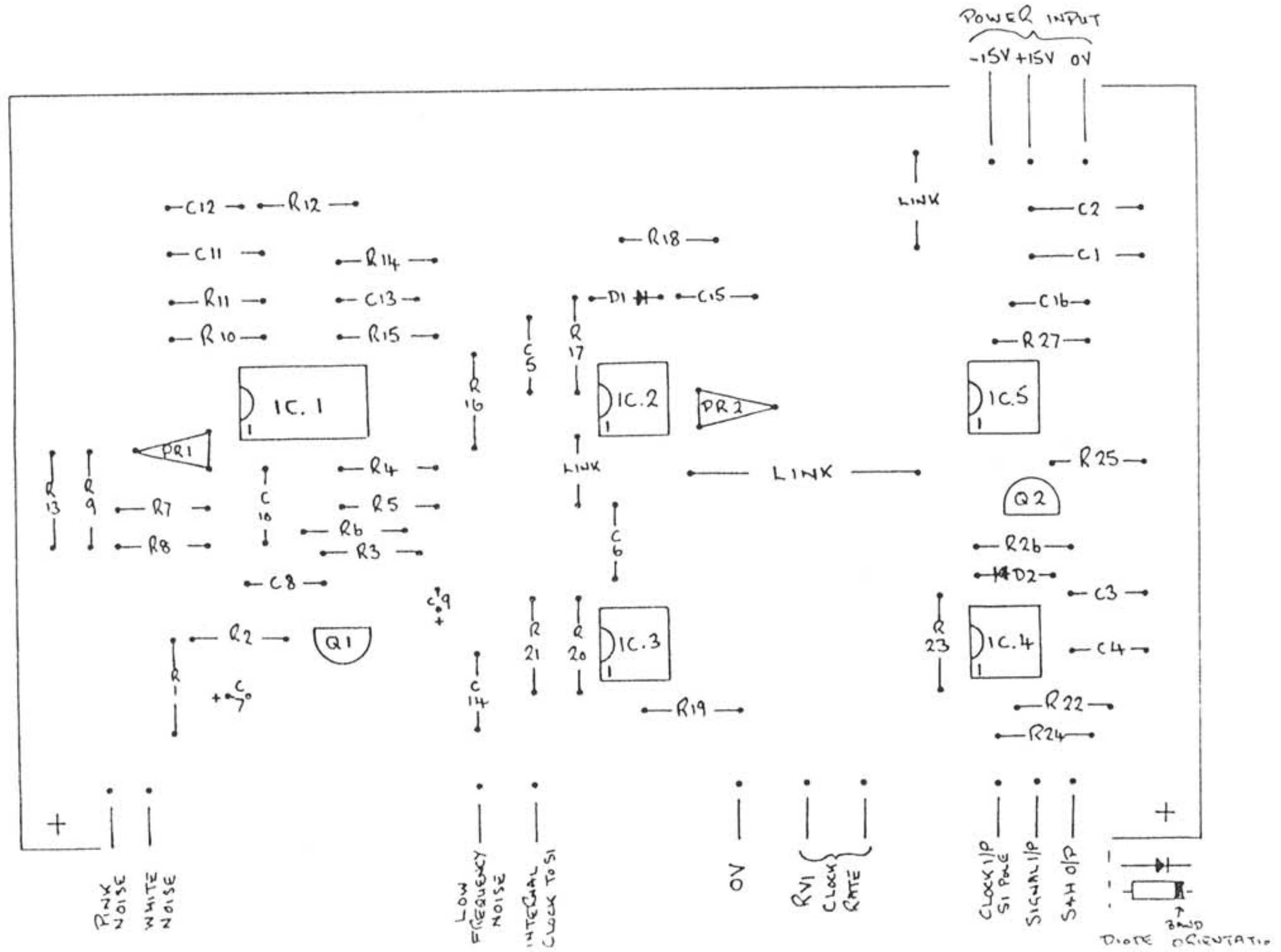


FIGURE 2 COMPONENT OVERLAY

80-12 COMPONENTS

RESISTORS, 5%, ¼w, carbon film

R 1, 18	47k	R 11	120k
R 2,6,15,19,22,23,25	100k	R 12	15k
R 3,4,26	1M0	R 14	39k
R 5	18k	R 17	8k2
R 7,8.9,13,16,27	1k0	R 20	3k9
R 10	22k	R 21,24	10k

CAPACITORS

C1, 2	470n polyester	C10,14,15	1u MKH polyester
C3,4,5,6,13	100n polyester	C11	2n2 polystyrene
C7,9	22u, 25V PCB elect.	C12	22n polyester
C8	470n MKH polyester	C16	220n MKH polyester

POTENTIOMETERS/TRIMMERS

RV 1	4M7 lin.	PR 1	100k carbon
RV 2	100k lin.	PR 2	470k carbon

SEMICONDUCTORS

IC 1 LM348N; IC 2 ICM 7555IPA; IC3,4 CA 3140E
IC 5 TL 082CP; D1,2 IN 4148 ; Q1 BC 548; Q2 2N3819

MISC: S 1 SPDT min. toggle. ; T 1 Transistor holder.

digisound EXTERNAL INPUT 80–13

INTRODUCTION

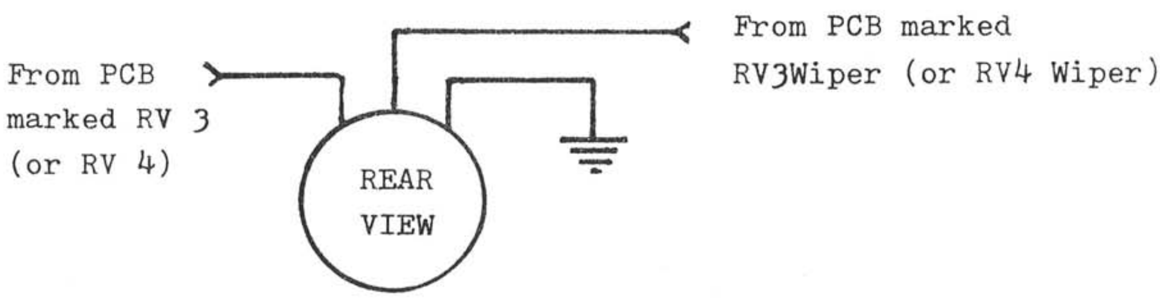
The 80-13 External Input Module allows interfacing of the Digisound 80 synthesiser with external sources, such as other live musical instruments or with pre-recorded material. It contains high quality pre-amplifiers to bring both microphone and line signals up to a level where they can be treated in the synthesiser, for example, by filtering or modulation. With or without filtering the sound envelopes from external sources may then be extracted, using a full wave rectifier principle, and these envelopes used to drive a voltage controlled amplifier whose sound source is derived from the synthesiser. The external source, or its envelope, may be connected to the gate/trigger extractor in order to control ADSR's (80-8 or 80-10) so as to generate different envelope shapes as an alternative approach for modifying the external sound source. The ADSR's may also be used to control VCF's and this combination of envelope following and dynamic treatment of the synthesiser sound has wide application.

CONSTRUCTION AND SETTING UP

The circuit diagram for the External Input Module is shown in Figure 1 and the component overlay illustrated in Figure 2. Orientation of I.C.'s, diodes, zeners and the electrolytic capacitors requires the usual care and the only other point to note is the use of screened wire to the inputs of the pre-amplifiers. This wiring should be kept as short and neat as possible and one end of the screen connects to the jack socket and the other end to the points marked 'screen' on the PCB overlay. To save too many short external connections between the miniature jack sockets the output of the envelope follower may be wired direct to the jack socket connection going to the peak detectors. The wiring is to the connection of the latter input which is disabled on insertion of a jack plug.

The only setting up required is to ground the input to the microphone pre-amplifier (IC 1) and adjust PR 1 for zero voltage at its output. The latter voltage should be measured between R5 and C9.

Note that RV 3 and RV 4 should be 'reverse' wired, as illustrated below, so as to provide increased sensitivity with clockwise rotation of the potentiometers.



COMPONENTS FOR 80-13.

RESISTORS, ¼w, 5%, carbon film

R 1,12,15,18	10k	R 14	51k
R 2	22k	R 16	5k6
R 3,7	1M0	R 20,25	3k9
R 4,5,9,17	1k0	R 21,26	2k7
R 6	150k	R 22,27	150R
R 10,11,13	20k	R 23,28	2k2
R 8,19	100k	R 24,29	1k2

CAPACITORS

C 1,2	47u,25V PCB electrolytic
C 3,4,5,6,	100n polyester
C 7,9,10,12,13,14	470n polyester
C 8,11	22p ceramic
C 15,16	10u, 25V PCB electrolytic

POTENTIOMETERS/TRIMMERS

RV 1	10k log.
RV 2	100k log.
RV 3,4	1M0 lin.

SEMICONDUCTORS

IC 1	NE 5534AN
IC 2	NE 5534N
IC 3	LM 1458
IC 4,5,6,7	CA 3140E
Q 1,2,3,4	BC 548
D 1,2	IN 4148
ZD 1,2	6V8 400mW zener diode
D 3,4	Red LED

MISCELLANEOUS: ¼in. mono jack sockets (2)

NOTES ON THE 80-13

The line input has a nominal impedance of about 60k and is therefore suitable for connecting to the line output (tape,etc) of most audio equipment. Maximum gain =11.

The microphone input has a nominal impedance of about 5k and will work with most microphones. Maximum gain = 1000

In the envelope follower the value of C 13 has been chosen to suit general applications and its value is a compromise between smoothing and time lag. If normally used for a specific application then the value can be altered. For example, when interfaced to a drum pick-up the value may be reduced.

The dual gate/trigger detector may cause most confusion. The design is based on selecting signal peaks as a proportion of the input signal so as to avoid many of the problems arising from signals of varying amplitude. With complex envelopes it is best to make use of the smoothing effect of the envelope follower prior to passing the signal to the peak detectors. In this mode the outputs from the peak detectors would go to an envelope generator controlling a VCA and the envelopes of the original sound will thus be transformed. A more complex patch is to use the envelope follower to reproduce the sound envelope of the input signal and to use the peak detectors to gate an ADSR connected to a VCF which is treating sounds generated by the synthesisers VCO's.

The two peak detectors are annotated 'GATE' and 'TRIGGER' but

CONTD. ON PAGE 4

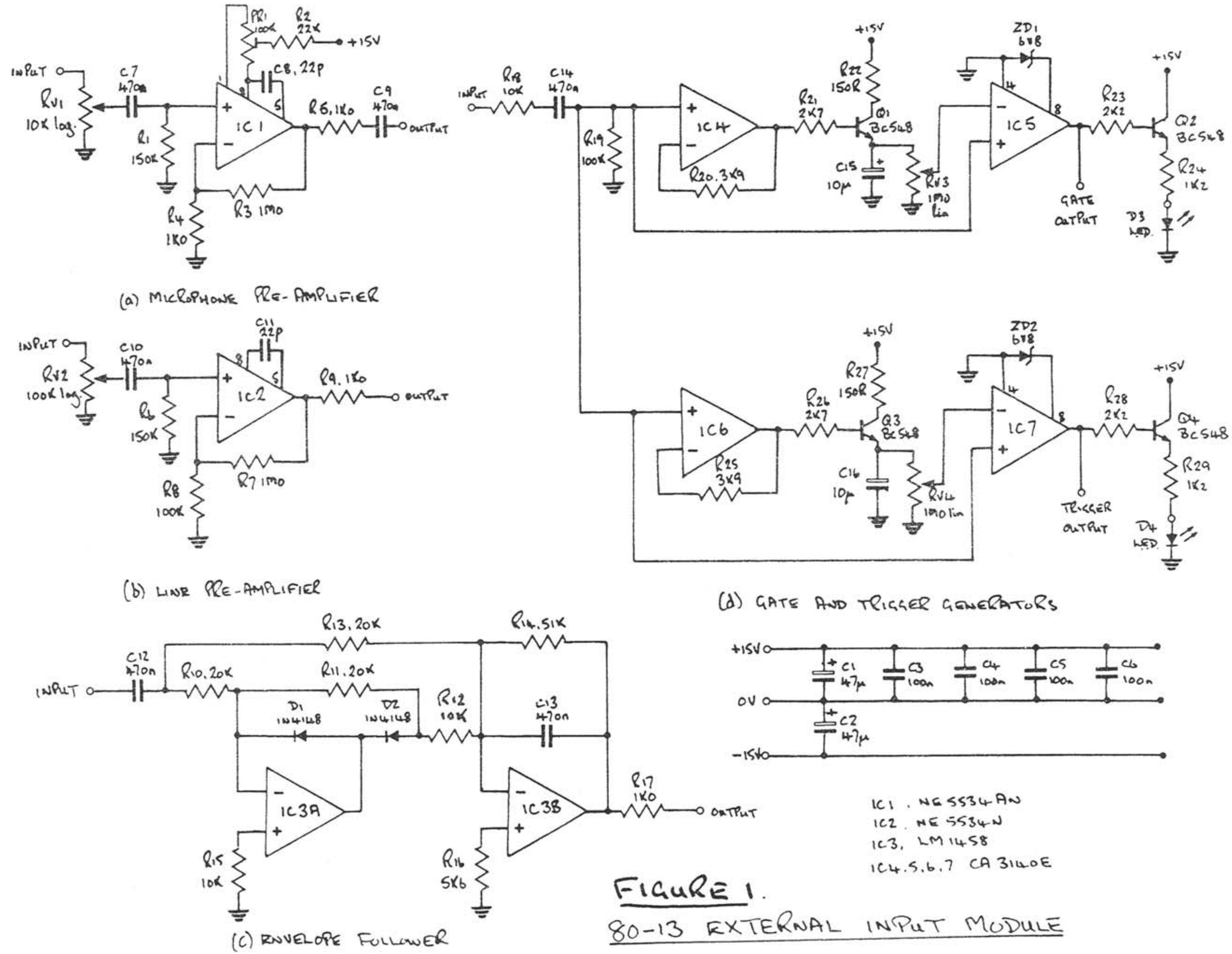


FIGURE 1.

80-13 EXTERNAL INPUT MODULE

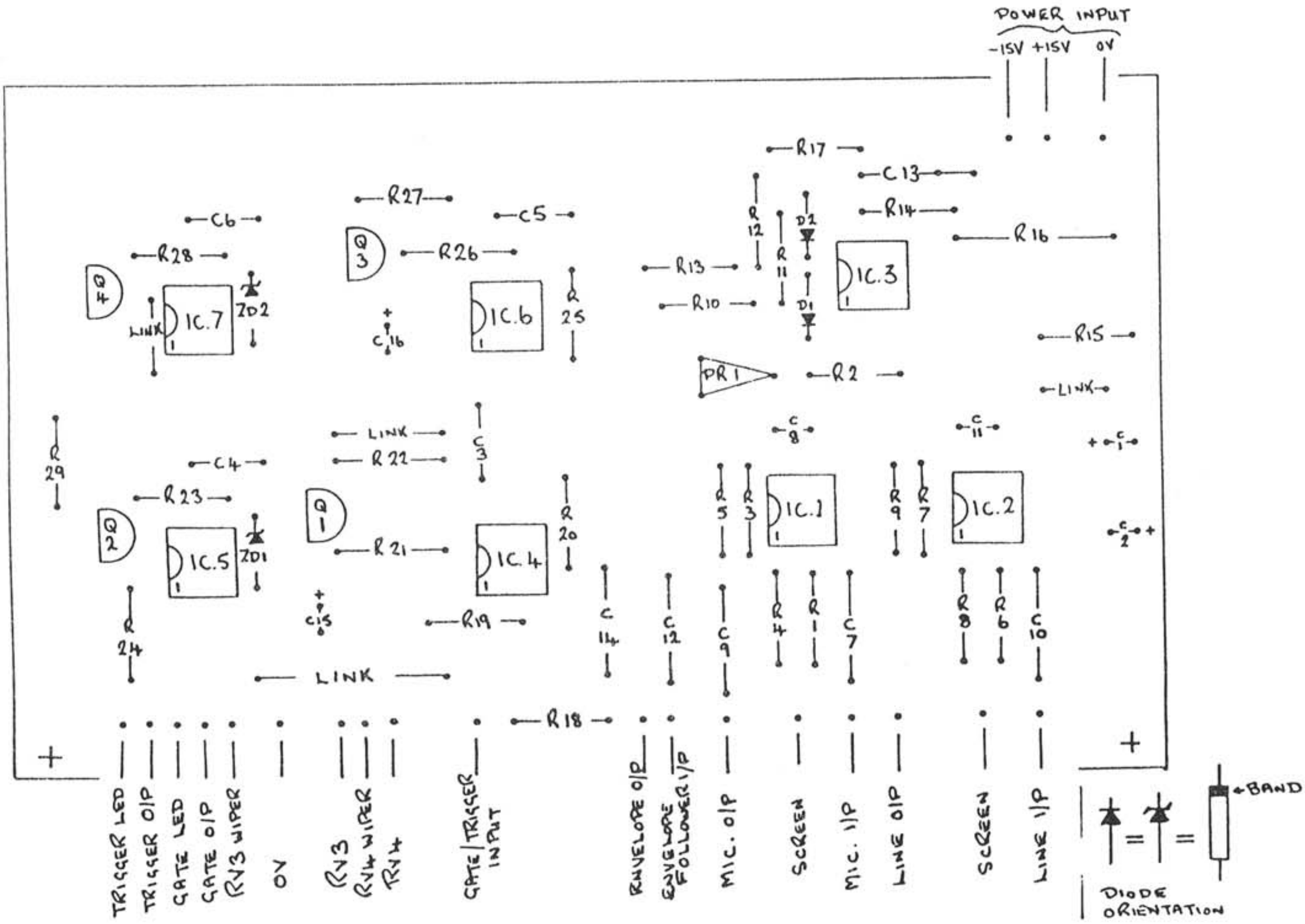


FIGURE 2. COMPONENT OVERLAY

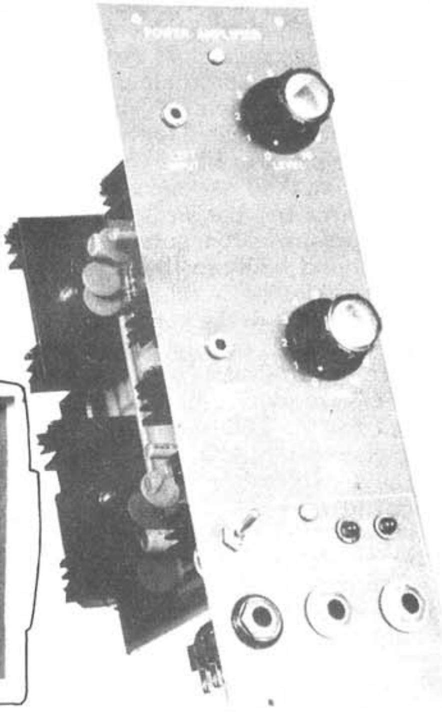
NOTES ON 80-13 CONTD.

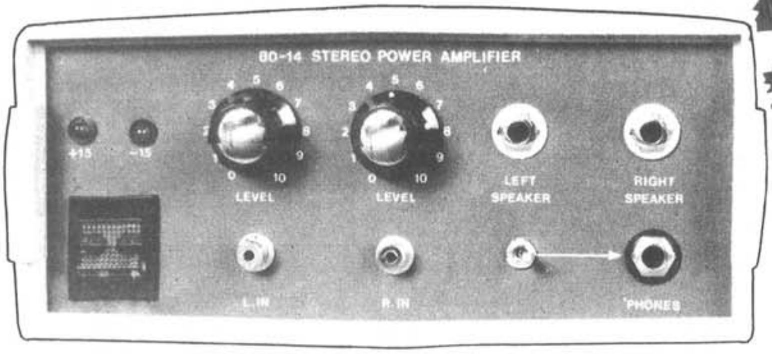
they are not totally independent, i.e., they are peak detecting the same signal input. Thus in the context of the logic applic--able to the 80-8 and 80-10 envelope generators the gate and trigger concept will only operate satisfactorily if the trigger transients are greater in amplitude than the voltage generating the gate pulse. The advantage of two detectors is, however, the ability to select two levels of sound, such as from two instrum--ents, and using the outputs to gate two separate ADSR's.

THE LED's CAN BE CONFUSING. With a weak signal the LED's may illuminate but the output voltage swing for the ADSR's may not be adequate to gate the latter units satisfactorily. So if you do not get adequate gating (audibly obvious or as shown by LED indicators on 80-10) then increase the input voltage. ALSO the LED's may illuminate when the unit is not in use due to stray voltages charging up capacitors C 15 and/or C 16. Swapping the CA 3140E's around may cure it or transfer the fault from one output to another but since it does not affect the proper operation of the module these changes are not necessary.

Learning to use the module is best accomplished by trying it out with some pre-recorded music which should either be a straight percussion piece or music with a distinct percussion beat. The latter may be filtered to more clearly separate the percussion.

MONITOR AMPLIFIER





Sound out your collection of Project 80 synthesiser modules with this 10 W per channel stereo power amplifier. Design by Charles Blakey

The 80-14 Stereo Power Amplifier is designed to be used in conjunction with the Project 80 synthesiser but with a few component changes it may also be used as a compact general purpose amplifier. The amplifier has an input impedance of 100k and an output of 10 W per channel into 8R with a maximum distortion of 0.5% (typically 0.1%). For use with the synthesiser the input sensitivity is 1V2 RMS for the rated output and for the general purpose version the sensitivity is 250 mV RMS. A switched headphone output is incorporated, suitable for use with low impedance headphones.

The design is based on the TDA2030 which is a Class B amplifier with low harmonic and cross-over distortion. It incorporates power limiting circuitry, giving short-circuit protection, in addition to a conventional thermal shut down system. The choice is based on experience with the TDA2030, the fact that 10 W per channel is adequate for domestic or monitoring use, the need to keep heat generation to an acceptable level and, not least, to provide a compact module.

Power Regulations

To obtain the 10 W per channel output it is necessary to use a 30 V supply; the maximum rating of the device is 36V. Furthermore, synthesiser applications in particular can generate peak current demands and these factors dictated the use of a regulated +15 V supply and DC coupling of the speakers. The components for rectifying, smoothing and regulating the power supply have been incorporated on the same PCB as the amplifier. When used with the synthesiser this allows the same mains switch to be used as for the +15 V module supply (Project 80-1) and for the fuse and transformer for the amplifier to be housed in the keyboard case. A miniature three pole connector may then be used to couple the 15-0-15 V unregulated supply to the module housing allowing the module to be rapidly removed from the case when required. Also by having the capacitors on the PCB the ground returns from the speakers are kept short.

All Change

The component values shown in the circuit diagram are for the synthesiser version. For the general purpose amplifier the following component changes are required.

R3,8	wire links
R4,6,11,13	100k
R5,12	3k3
C6,11	4u7 PCB electrolytic

Construction

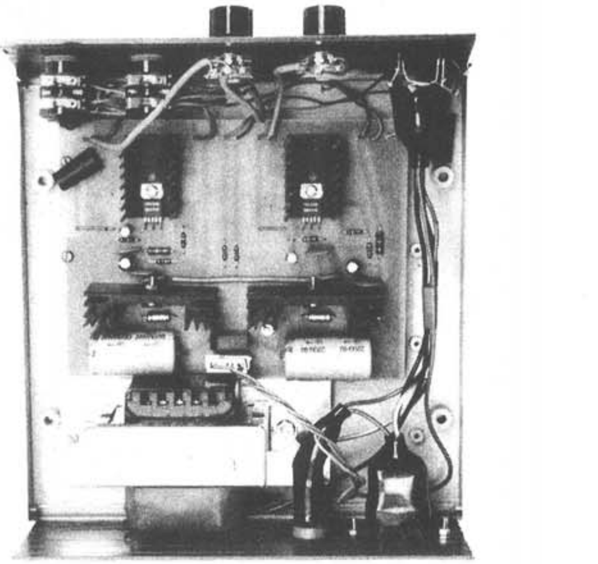
The module will fit onto the standard 9 x 3 inch panel and it can also be installed in a Teka Alba A23G case but as with the panel version the transformer will have to be external to the case. For the self-contained amplifier a case with minimum internal dimensions of width 220 mm, length 250 mm and height 90 mm is required.

Construction should be carried out in the following sequence. Make the one wire link with insulated wire then solder in the resistors, capacitors and the bridge rectifier. Next install the TDA2030s. Slide the heatsink under the TDA2030 and, after checking that the pins are still in place, bolt the IC and the heatsink to the PCB. Do not move the

→

heatsink once the IC has been soldered since this will stress the pins. The voltage regulators are now bolted to their heatsinks (the pins should protrude from the side having the greatest distance from mounting hole to edge) and the combined heatsink and IC held firmly against the PCB while the regulator is soldered in place. There is no need to isolate any of the ICs from their heatsinks, but it should be noted that the heatsinks for the negative regulator and the TDA2030s will be at negative potential. A small amount of heatsink compound between the IC and their respective heatsinks is desirable.

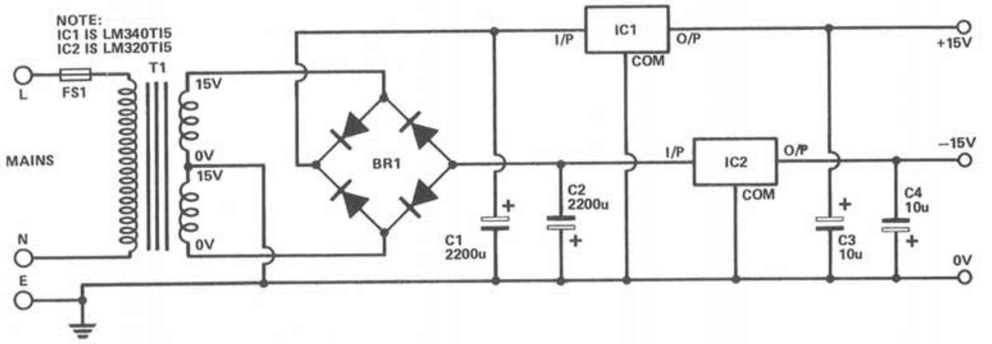
Next wire the PCB to the panel components. Screened wire should be used for the input leads which go from the input jack sockets to the rotary potentiometers and from the latter to the PCB. Do not common ground connections at the panel (except for the LEDs), but take them back to the appropriate connection hole on the PCB. Keep wiring as short and neat as possible. For the speaker leads it is preferable to use wire of at least equivalent to 16/0.2 mm. R8 and 15 can be soldered direct to the switch and a lead taken from the other end to the headphone socket while R9 and 16 should be soldered direct to the headphone socket. Remember to take a ground return from this socket to the PCB.

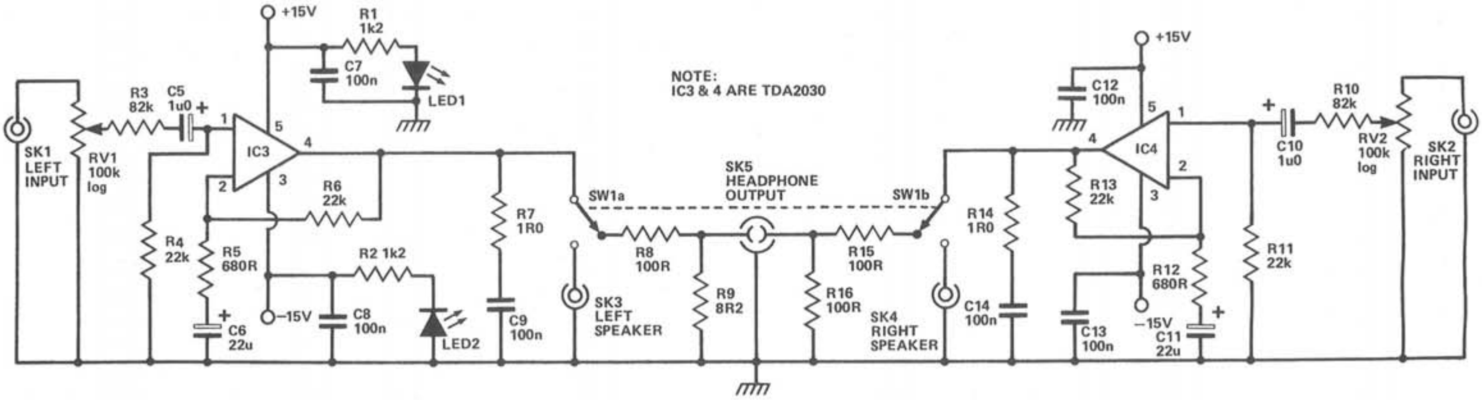


Inside the Project 80 Monitor Amplifier. Straightforward design and PCB layout makes for simple construction.

Fig.1 (right) Circuit diagram of the power supply. BR1 is a 2 A bridge rectifier.

Fig. 2 (below) Circuit diagram of the stereo Monitor Amplifier (synthesiser version). See text for alterations necessary to make a general purpose amplifier.





HOW IT WORKS

The TDA2030 power amplifiers (IC3 and IC4) require few external components and the function of the latter for the left input is described. C5 AC couples the amplifier while R3 and R4 form an attenuating network to reduce the sensitivity for use with the Project 80 synthesiser and, in the absence of RV1, determine the input impedance. RV1 provides manual adjustment of attenuation. R5 and R6 set the closed loop gain and for the general purpose version (R4 = 100k; R6 = 100k; R5 = 3k3) the voltage gain is approximately 30 dB. C6 is for DC decoupling of the inverting input and adjusts the low frequency cut-off. R7 and C9 increase frequency stability while C3, C4 (power supply) together with C7 and C8 are bypass capacitors which also reduce the risk of oscillation.

SW1 allows selection of speaker or headphone outputs and for the latter R8 and R9 attenuate the output to a level suitable for low impedance headphones.

The power supply is a conventional regulated supply with a nominal + 15 V and 1A5 per rail which is sufficient for 8R speakers at peak output of the amplifiers (about 13 W with 10% distortion). The regulators will also cope with 4R speakers in combination with a suitable transformer. R1, R2 together with LED 1,2 give a visual indication of supply voltage to the amplifiers.

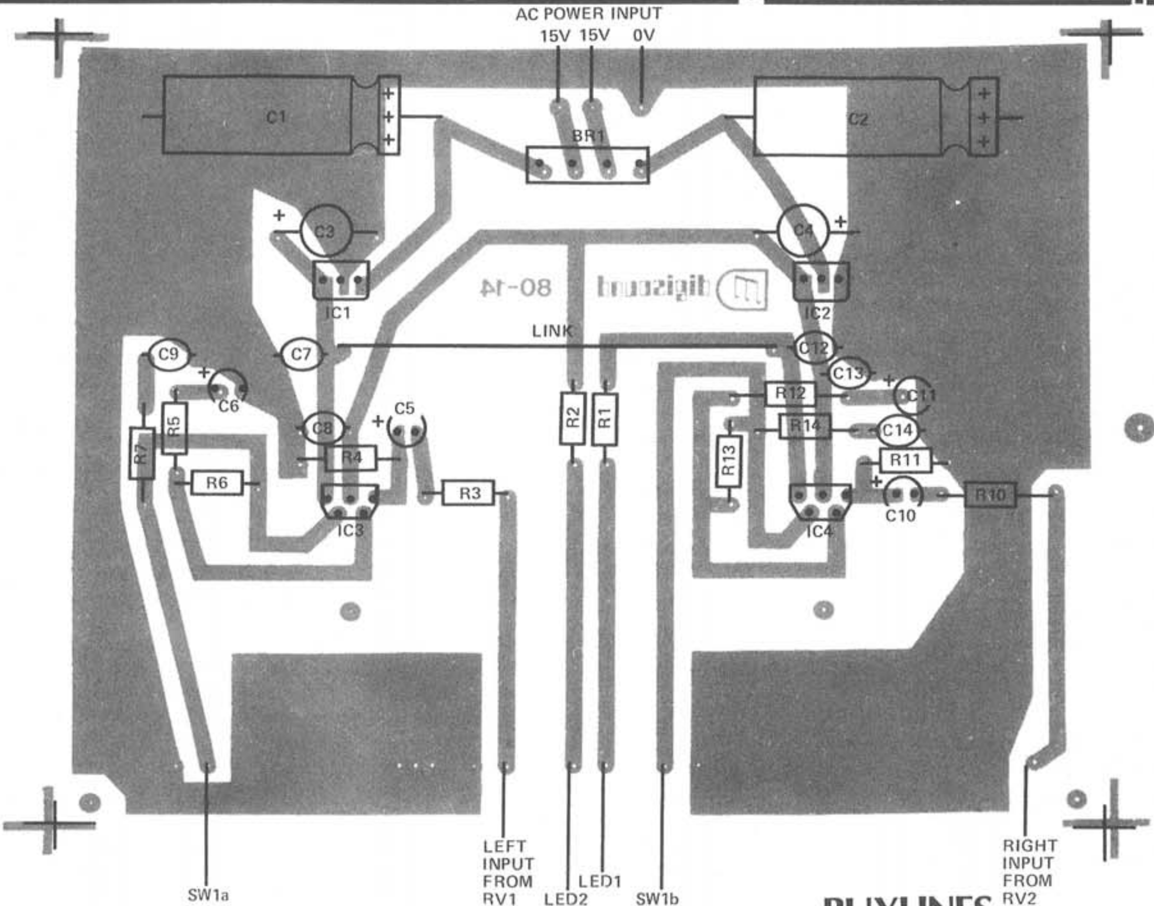


Fig.3 Component overlay.

BUYLINES

A kit for the stereo amplifier is available from Digisound Limited, 13 The Brooklands, Wrea Green, Preston, Lancs PR4 2NQ for £22.20 inclusive of postage and VAT. The kit includes the PCB and all listed components except transformer and case/panel. Please specify synthesiser or general purpose version.

PARTS LIST

(Synthesiser version, 8R speakers)

Resistors ¼ W carbon except where stated	
R1,2	**1k2**
R3,10	**82k**
R4,6,11,13	**22k**
R5,12	**68R**
R7,14	**1R0, ½ W**
R8,15	**100R, ½ W**
R9,16	**8R2, 2W5 wirewound**
Potentiometers	
RV1,2	**100k logarithmic**
Capacitors	
C1,2	**2200u 40 V electrolytic**
C3,4	**10u 25 V electrolytic**
C5,10	**1u0 100 V PCB electrolytic**
C6,11	**22u 25 V PCB electrolytic**
C7,8,9,12,13,14	**100n disc ceramic**
Semiconductors	
IC1	**LM340T-15**
IC2	**LM320T-15**
IC3,4	**TDA2030H**
LED1,2	**Red LED**
B1	**2 A bridge rectifier**
Miscellaneous	
SW1	**DPDT subminiature switch**
SK1,2	**3.5 mm jack sockets (phono sockets for GP version)**
SK3,4	**0.25 inch mono jack sockets**
SK5	**0.25 inch stereo jack socket**
T1	**50 VA transformer, dual 15 V secondaries in series or 15-0-15 V type.**
FS1	**Chassis fuse holder with 1 A fuse.**
Heatsinks for IC1,2,3, and 4	

Ironing

A final point to note is that comparatively heavy currents will flow through many of the connections and it is essential that they are properly soldered. The connections requiring most care are those to ground where the large foil area acts as a heatsink. This is eased by using a tinned PCB but even then it is necessary to place the soldering iron adjacent (not touching) to the lead to be soldered and allow the area to heat up sufficiently prior to heating the lead and applying solder to it.

After construction connect the transformer and switch on. Gently touch each IC in turn. These should remain cool, since the TDA2030 quiescent current to each is only of the order of 50 mA. The LEDs will indicate whether the power supply is functioning. If any of the ICs run hot at this stage check the component placement and condition of soldered joints. Next connect the speakers and if any hum is evident check the wiring from PCB to panel components. Finally connect the amplifier to an audio source to determine that the module is functioning correctly.

Conventional Hi-Fi speakers should not be used in conjunction with a synthesiser since single frequency tones of more than a few watts can damage treble speakers. For most purposes full range speakers with a nominal impedance of 8R and a rating of 15 W will prove adequate.

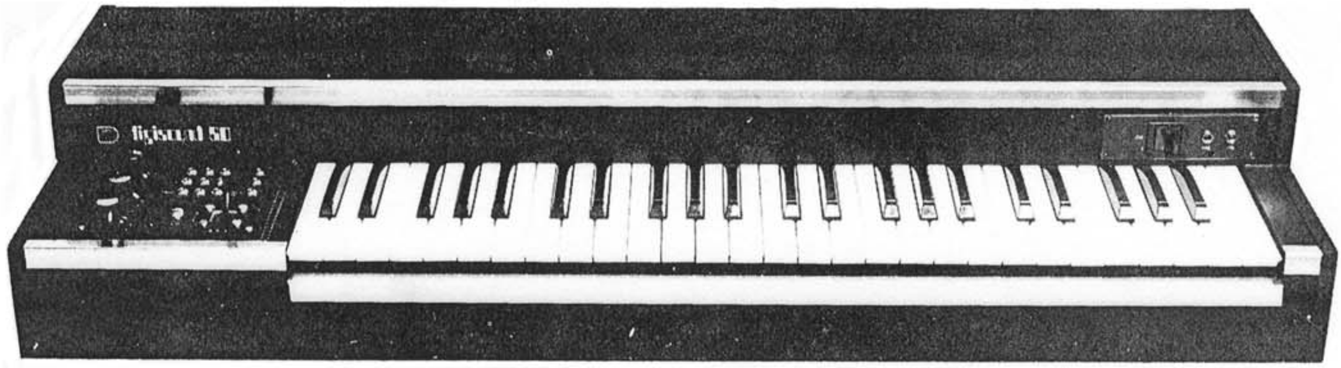
80-14 STEREO AMPLIFIER. AMENDMENTS/CORRECTIONS

The kit supplied is the ~~more sensitive~~ GENERAL PURPOSE version with components listed on p.79 of ETI article. Note the following corrections.-

1. p.79. R3,8 should read R3,10
2. Parts List. R5,12 should be 680R as circuit diagram
3. Circuit diagram. R16 is 8R2 as parts list.

When building this kit first check that power supply functions correctly and then instal one amplifier at a time and check operation. DO NOT strain the pins of the TDA 2030's.

KEYBOARD CONTROLLER



Bring your Project 80 Modular Synthesiser to life with the final components — the keyboard assembly and keyboard controller. Design and development by Charles Blakey.

The design is based on the principle of digitally scanning each key in sequence so that the controller may be readily interfaced with a microprocessor to accomplish sequencing, composing, multi-voices and polyphonic control. The synthesiser can, therefore, be expanded to suit the needs of the user. The digitally scanned keyboard is equally well suited to a monophonic synthesiser and the design presented will suit keyboards having up to 64 keys. It is stable, accurate and readily adjusted to suit various volts per octave relationships. One of its best features is that it dispenses with the analogue 'Sample & Hold' (known to many as 'Sample & Droop'). With the digital version the key voltage can be held constant for days, although this is not very interesting musically!

Both the 80-8 and 80-10 envelope shapers have been designed to operate in the ADSR mode without an independent trigger voltage and so a trigger output voltage coincident with the gate voltage is not included. Instead a touch activated trigger is provided which may be connected to the independent trigger output of the 80-10 module to produce complex envelopes.

Keyboard and Scanner

Assume that the diode and key contact, which will connect Column 0 to Row 0, is the first key of the keyboard, Column 1 Row 0 is then the second key, Column 2 Row 0 the third key and so on until one gets to Column 7 Row 7 which would be the 64th key. IC3 is sequentially scanning the columns by outputting a positive voltage and so, when Q0 of IC3 is high this voltage will be present on all diodes of Column 0, that is on one side of the contacts for keys 1, 9, 17, 25, 33, 41, 49 and 57. IC4 is an identical sequential scanner to activate the rows and the step from row to row take place when IC3 has completed a scan of the eight columns. The controller is, therefore, scanning the keyboard sequentially from first key to last key with an effective first key down priority. By way of an example let us follow what happens when key 25 is pressed. A logic '1' signal will be present on Row 3 and when Q3 of IC4 goes high the combined '1 + 1' logic signals at the IC5d AND gate will generate a '1' output. This may sound rather slow but in practice a 49 note keyboard will be completely scanned at a rate of about 100 times a second. A '1' from IC5d (or any AND gates) will cause the eight input NOR gate (IC7) to go low and further logic around IC8 generates two outputs when a key is pressed, namely $\overline{\text{KEYDOWN}}$, which is a logic '0' signal, and KEYDOWN which is a '1' output. In the monophonic arrangement KEYDOWN is connected to the INHIBIT line of the clock built around IC1 so that the clock stops and will stay off for the duration the key is held down. The KEYDOWN signal is, therefore, also used to form the gate voltage for the envelope shapers. Clearly with the clock stopped keyboard scanning ceases. Nevertheless the data on the output lines of the counter (IC2) used to drive IC3 and IC4 and which stopped simultaneously with the clock is still present at its outputs Q0 to Q5 (ignore Q6). These six data bits identify the key being held down and for the 25th key the binary code output will be 011000. A few additional codes for a 49 note C-C keyboard are listed below and for those not familiar with the binary code it will be worth the effort to make a complete listing. The listing assumes starting at Column 0 Row 0.

Key No.	Note	Binary Code
1	CC	000000
2	CC #	000001
25	C	011000
26	C #	011001
27	D	011010
28	D#	011011
29	E	011100
49	C	110000

When the key is released scanning recommences until another key is found down.

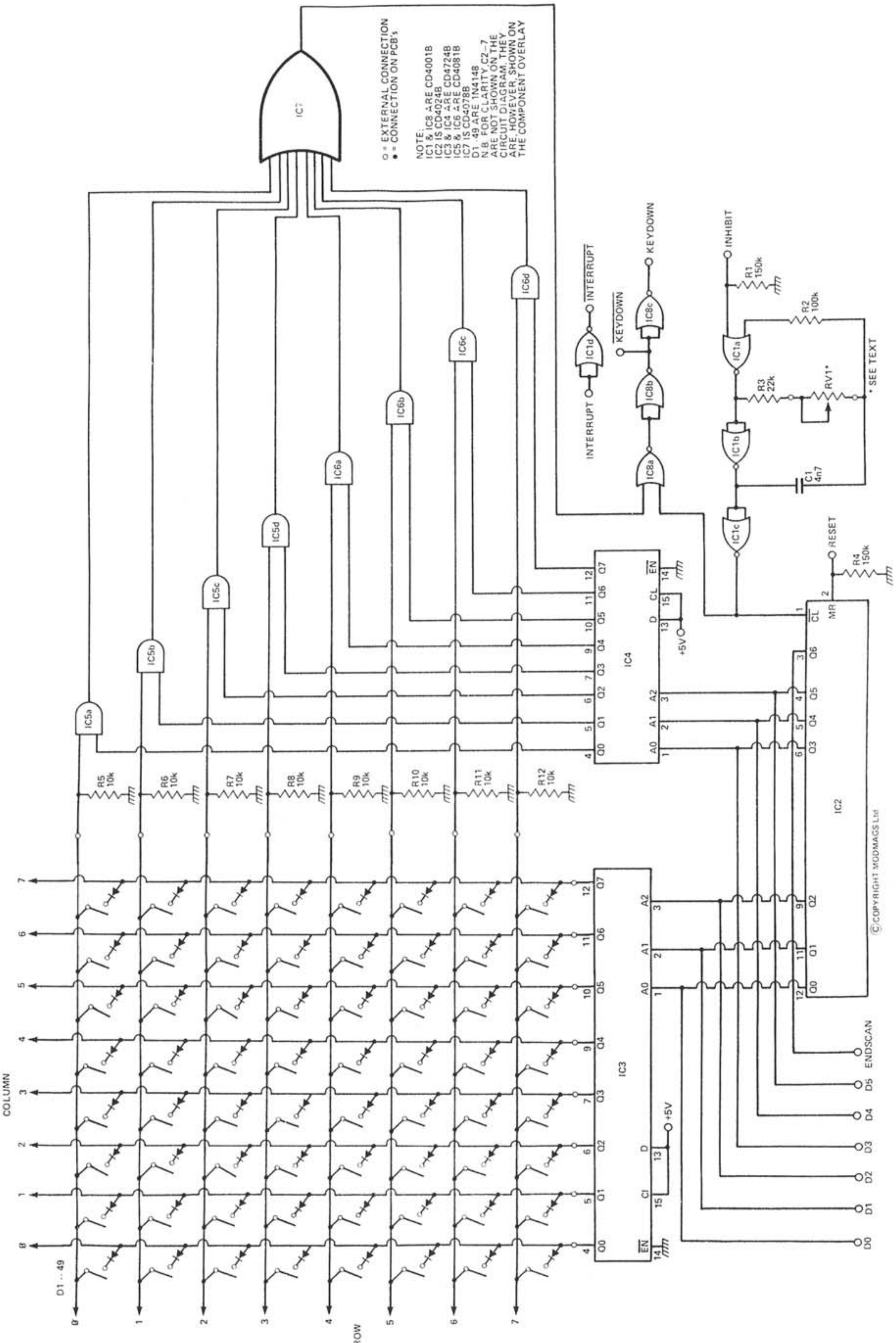
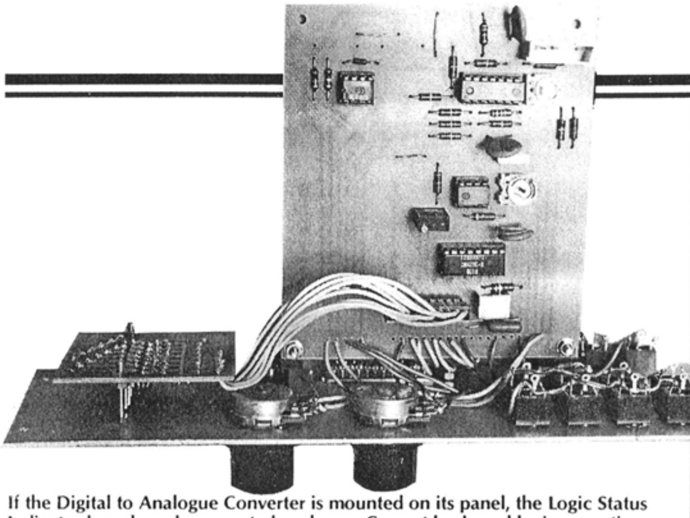
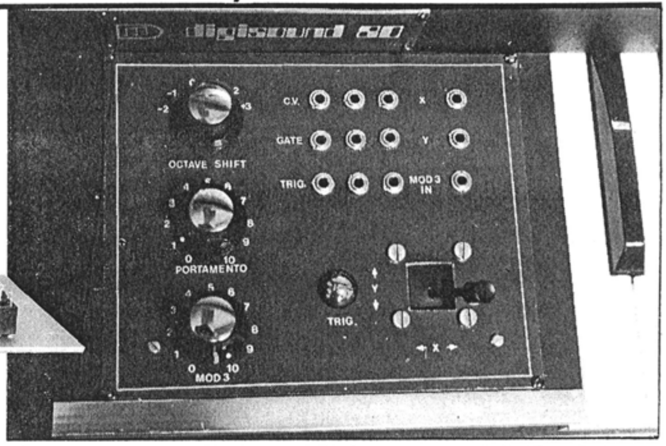


Fig.1 Circuit diagram of the keyboard matrix and scanner.

Keyboard Controller



If the Digital to Analogue Converter is mounted on its panel, the Logic Status Indicator board can be mounted as shown. Correct keyboard logic operation can then be checked by watching the LED patterns on the front panel (see text).



The option A layout, with Octave Shift, Portamento control X-Y control, Touch Trigger, etc, all in the keyboard case. (See Fig.14).

HOW IT WORKS

KEYBOARD CONTROLLER: The circuit diagrams comprise four parts: an 8 x 8 diode matrix mounted on the keyboard — or at least part of it equivalent to the number of keys on the keyboard; scanning and logic circuit for the keyboard; a digital to analogue converter; a +5 V power supply sufficient to cope with the logic circuits. In the scanner circuit an astable is made from the NOR gates IC1a, b and c which will have a clock period of about 2.2R3C1 and so with the components used the clock rate is about 4.4 kHz. the clock may be stopped by a logic '1' at the input marked INHIBIT, which is normally pulled low by R1 connected to ground. Provision for adjusting the clock rate using RV1 has been included but not installed in the monophonic version. The astable is used to clock a seven stage counter, IC2, and the lower three bits (Q0 and Q2) are in turn used to drive an eight-bit addressable latch, IC3, which is arranged as an active high eight channel demultiplexer. Thus the output of IC3 is a series of logic '1' pulses which is sequentially scanning the eight columns of the diode matrix. The next three bits from the IC2 counter go to IC4 which is another eight channel demultiplexer. When all eight columns of the diode matrix have been scanned by IC3 then IC4 increments one step and puts a logic '1' on one input of the AND gates (IC5,6) which are associated with specific rows of the diode matrix. When a key contact is closed the '1' output from IC3 will be transmitted to the appropriate row bus and will take the other input of the AND gate high — which is normally held low by a resistor (R5-12) connected to ground. With two highs at the AND gates they will output a '1' and this is detected by the eight-input NOR gate, IC7, which goes low. Now the digital code for each of the available 64 key contacts will be defined by the first six bits of the IC2 counter. When IC7 goes low a KEYDOWN output (logic '0') is produced, which is used for latching the D to A Converter and also a complimentary KEYDOWN output which is wired to the INHIBIT line of the clock. Thus, whenever a key contact is closed both the clock and counter are stopped and the data present on the Q0 to Q5 outputs of the latter is the binary code for the particular key pressed. These data bits, D0 to D5, are connected to the Digital to Analogue Converter IC10. IC10 is an eight bit D to A Converter but only the first six bits are used. The converter may also be latched via pin 4 and so this pin is connected to the KEYDOWN signal from the scanner. When a key contact is closed KEYDOWN is low and this makes the D to A transparent to the data at its inputs, namely the binary code for the particular key pressed, and converts the data to a voltage. When the key is released then KEYDOWN goes high and the data present at the D to A is latched so that the output voltage remains until the next key is pressed and the sequence is repeated. IC10 includes an internal voltage reference which is utilised in this design with R16 deriving a reference current to the internal zener diode while C12 is for stabilising and decoupling. The output of IC10 needs to be buffered by a high impedance op amp having a low input bias current and this is provided by IC11. Also, when the internal reference voltage is used the output from IC10 will be about 10 mV per data bit whereas to obtain the required 1 V/octave an output of 83.3 mV/bit (83.3 mV/semitone) is necessary and so the gain of IC11 is made adjustable around a nominal gain of about eight. IC10 will not drive a high RC load directly and so the portamento components RV2 and C13 are taken from the output of IC11 and buffered by IC12a arranged as a voltage follower. The control voltage, with or without slewing (portamento), now goes to IC12b which is a summing amplifier with unity gain where it may be mixed (voltages added or subtracted) with other control voltage sources via R20/RV3, R21 or R23. The output of IC12b will be inverted and this is corrected by IC12c which is a unity gain inverter. PR3, R27, R28 and R29 are used to remove the offset voltages arising from IC12.

The KEYDOWN signal from the controller remains high for the duration the key is held down and is, therefore, the conventional gate signal which defines the duration of the sustain in ADSR envelope generators. For practical purposes this CMOS output needs to be buffered and this is done with IC13 arranged as a comparator. IC13 needs to be a high slew rate op amp capable of going back to near ground potential so as to be compatible with the internal logic of customised envelope generators. Another advantage of the CA3140E is that its output voltage may be clamped by placing a zener diode between pins 4 and 8 and the output voltage will be approximately equal to the zener voltage less two diode drops. This is the arrangement shown for use with the ETI 80 envelope generators. Other gate voltages may, however, be derived by using the resistive dividers R33/34 and R35/36 to operate IC11 from a variety of voltages.

A means of generating the +5 V supply required for the CMOS devices and the D to A Converter is included on the scanner PCB. The input voltage for this section is taken from the rectified and smoothed positive supply of Module 80-1 and the voltage dropped across R13/14. This is then regulated by IC9 and will provide the 30 mA or so required for this part of the project.

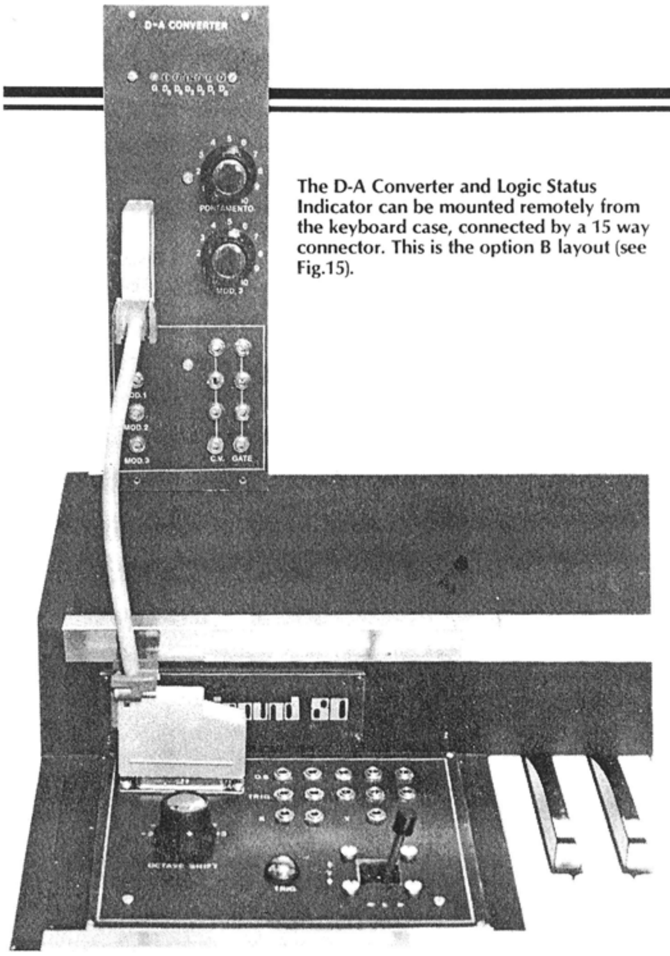
OCTAVE SHIFT, X-Y CONTROLLER, TOUCH TRIGGER: For the octave shifter IC14 is a programmable current source which, by adding D50 and making the sum of R38,39 and 40 equal to about 10 times that of R37, will be insensitive to temperature changes. The voltage increments developed at the junctions of R38,39 and 40 are taken via a rotary switch, SW1a, to a non-inverting amplifier whose gain may be adjusted with PR4 to give +3 V from the junction of IC14/R38 junction; +1 V at the R39/40 junction; and zero from the junction of R40 with ground. PR5 cancels the offset voltage in IC15. The positive outputs are made available at the pole of SW1b. The +2 V and +1 V outputs are inverted by IC16 and PR6 allows accurate adjustment to compensate for the small errors introduced by R44,45. PR7 is for offset adjustment of IC16. These negative outputs are also selectable by SW1b.

For the X-Y controller it was also convenient to use the reference voltage derived at the junction of IC14 and R38 and this voltage is buffered from IC17a and IC17c arranged as voltage followers. For the 'X' potentiometer of the joystick, RV4, the voltage is amplified by IC17c which is an inverting amplifier. The gain of the later is equal to

$$\frac{R49}{RV4 + R47}$$

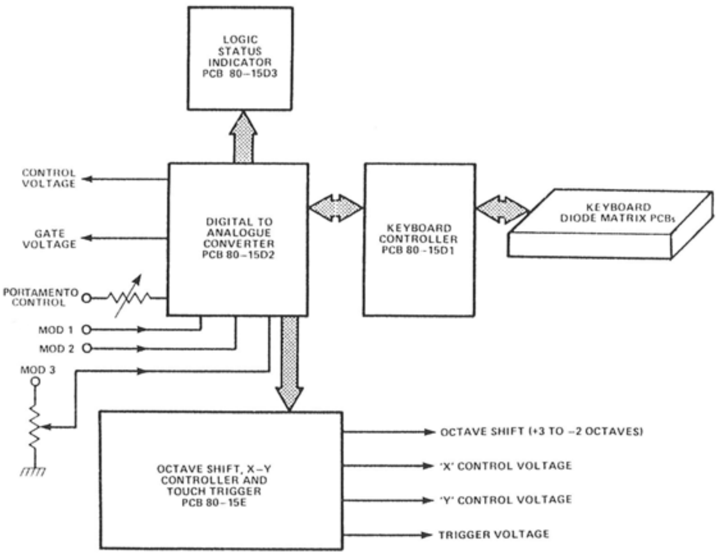
and thus the output voltage will vary from −2.7 X Vin to −0.84 X Vin and since Vin is about 0V5 this gives a range of approximately −0V4 to −1V4. PR8 and R48 are used both to convert these outputs into positive voltages and also to adjust the output to 0 V when the 'X' potentiometer is in the extreme left hand position. The net effect is a control voltage which is variable from zero to about 0V95. This range may be easily adjusted by altering the value of R49 and the general arrangement used allows a wide variety of values for the resistive range of the joystick potentiometer. The circuit for the 'Y' controller is identical to the 'X' controller.

A trigger pulse is derived from touching an insulated metal contact and is based on IC18 arranged as a comparator. R56 and R57 set up the switching level, which is about 150 mV. The impedance of the output is determined by R55 and on touching the metal conductor mains hum will be picked up and cause IC18 to switch between zero and about +13 V at around 50 Hz. The output is, therefore, smoothed sufficiently by C17 to be used as a trigger pulse for the 80-10 envelope generator, which has a fully independent trigger input. The sensitivity of the touch trigger may be changed by altering the value of R55 and/or adjustment of the comparator voltage set by R57 and R57.



The D-A Converter and Logic Status Indicator can be mounted remotely from the keyboard case, connected by a 15 way connector. This is the option B layout (see Fig.15).

Fig.2 (below) Block diagram of the keyboard circuits.



These are the basic features of the keyboard controller and the salient points to remember are: (a) only a single key contact is required to generate both the key code, which is subsequently converted to a control voltage, and the gate signal: (b) Pressing two keys in error will not cause problems because of the decoding arrangement used; (c) Key bounce is not a problem; (d) The only inputs and outputs of interest at this time are KEYDOWN, $\overline{\text{KEYDOWN}}$, (clock) INHIBIT, and the six data bits D0 to D5. Other connections are for future expansion and it is evident that their inclusion has only added a few pence to the cost.

Digital To Analogue Converter

The six data bits, D0 to D5, from the scanner have to be converted to an analogue control voltage and also scaled correctly, which for the ETI 80 and most commercial synthesisers is 1 V/octave. The essential component is IC10 which is a self-contained eight-bit digital to analogue converter. An advantage of using the ZN 428E-8 is that the device is designed to be microprocessor compatible and has a latch at pin 4. When the logic signal at this pin goes from '0' to '1' the data present when it was at '0' is latched into the IC and the output voltage remains constant. From the earlier discussion on the controller it will be recalled that when a key is pressed a $\overline{\text{KEYDOWN}}$ signal is generated. This signal is connected to pin 4 of IC10. Thus when a key is pressed the clock stops, $\overline{\text{KEYDOWN}}$ goes to logic '0', and the data bits D0 to D5 are available to IC10 which converts them into a voltage directly related to the binary code.

When the key is released KEYDOWN goes high and so the data from the last key pressed is held in IC10 and the voltage output remains constant.

Since the control voltage may be used to control a number of VCOs, perhaps set at different intervals, it is more convenient in some cases to modify the control voltage to all modules simultaneously. This may be done through any of the three inputs marked MOD. 1 to MOD. 3. The latter incorporates an attenuating potentiometer, RV3, and could be used for, say, frequency modulation.

The keyboard scanner is implemented in CMOS and so could be powered by +15 V, but the D to A converter, IC10, is designed to operate from a nominal +5 V supply (INPUTS IN EXCESS OF +7 V MAY DESTROY THE DEVICE) and in any event this 5 V logic level is used by most microprocessors. IC1-10 only require about 30 mA at +5 V and so there is no need for a separate power supply. The required power is taken from the 80-1 power supply and, to prevent wasting the fully regulated +15 V supply, the supply voltage is taken across C1 on the 80-1 PCB. THE +5 V POWER SUPPLY OF THE KEYBOARD SCANNER PCB MUST NOT BE USED FOR POWERING ADDITIONAL ICs OR EQUIPMENT.

A useful diagnostic and learning tool is provided by a logic status indicator. There are six red LEDs, one for each of the six data bits D0 to D5 and a green LED for the gate output (KEYDOWN). With the keyboard scanning and no keys pressed all the red LEDs will be on and the gate LED off. When a key is pressed, the binary code for the key will be displayed and the gate LED will turn on.

In electronic synthesis stability and accuracy of control voltages is essential. The octave shift generator uses a current generating IC, which can be operated with minimum temperature drift and is substantially insensitive to supply voltage. The voltages derived can be added to the control voltage via MOD. 1 or MOD. 2 inputs on the D to A Converter PCB and a shift of +3 V, +2 V, +1 V, 0, −1 V, or −2 V is available from a selector switch. These are equivalent to a shift from +3 to −2 octaves with the Project 80 modules.

X-Y Controller

As configured, the range available is about 0V9 (0.9 octaves) and with the joystick parked in the bottom left hand corner the output is zero. This latter arrangement is preferred to X-Y controllers with a centre off position, since rarely does the pot go *exactly* to the centre position and the chance of an offset voltage is increased because the pot is in mid range. The X-Y controller may be wired directly to one of the MOD. inputs on the D to A Converter, although this is not advised, since the unit does not have the same voltage stability as other inputs. The best approach is to patch the 'X' and 'Y' outputs directly to the control inputs as required.

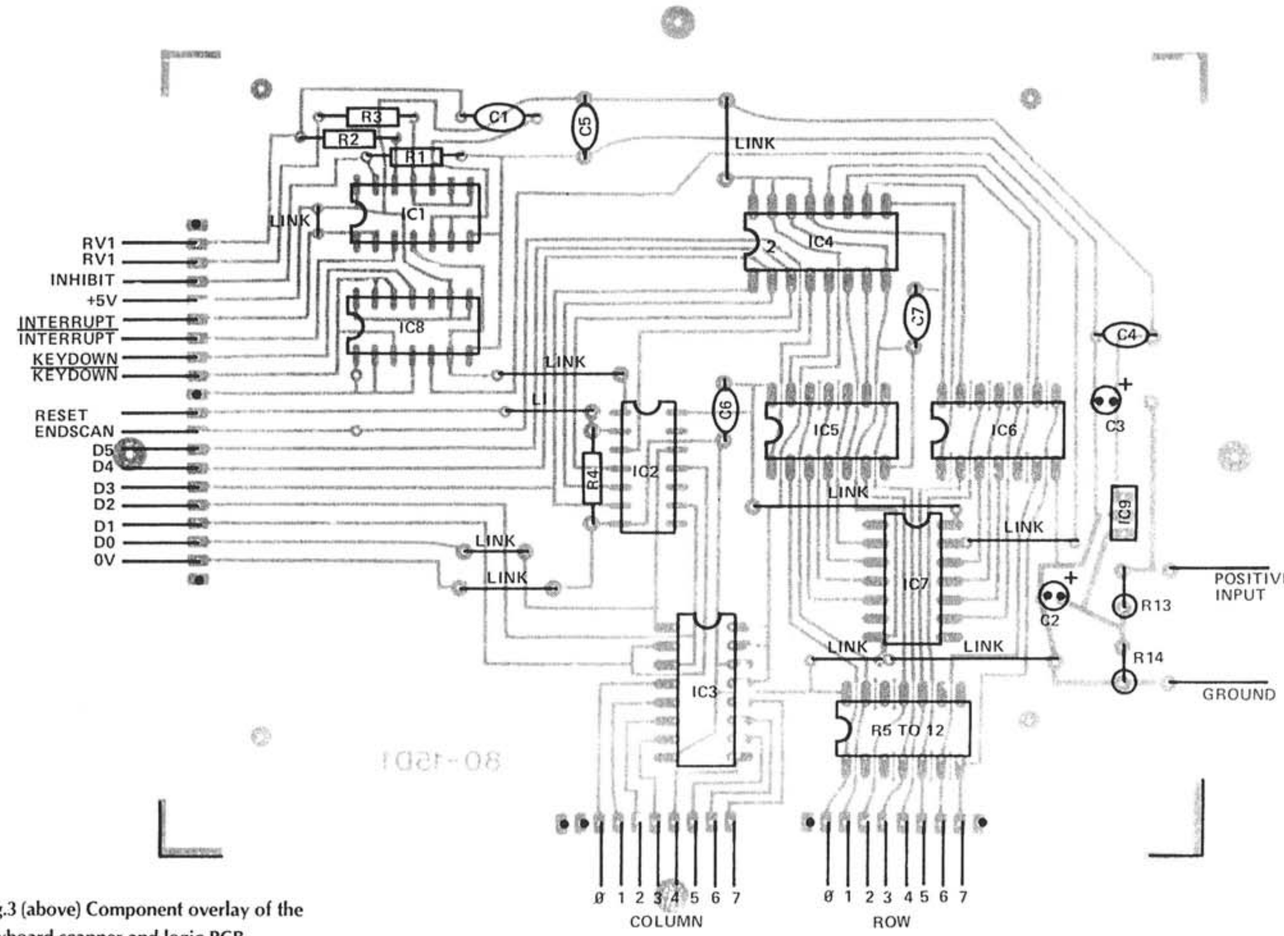


Fig.3 (above) Component overlay of the keyboard scanner and logic PCB.

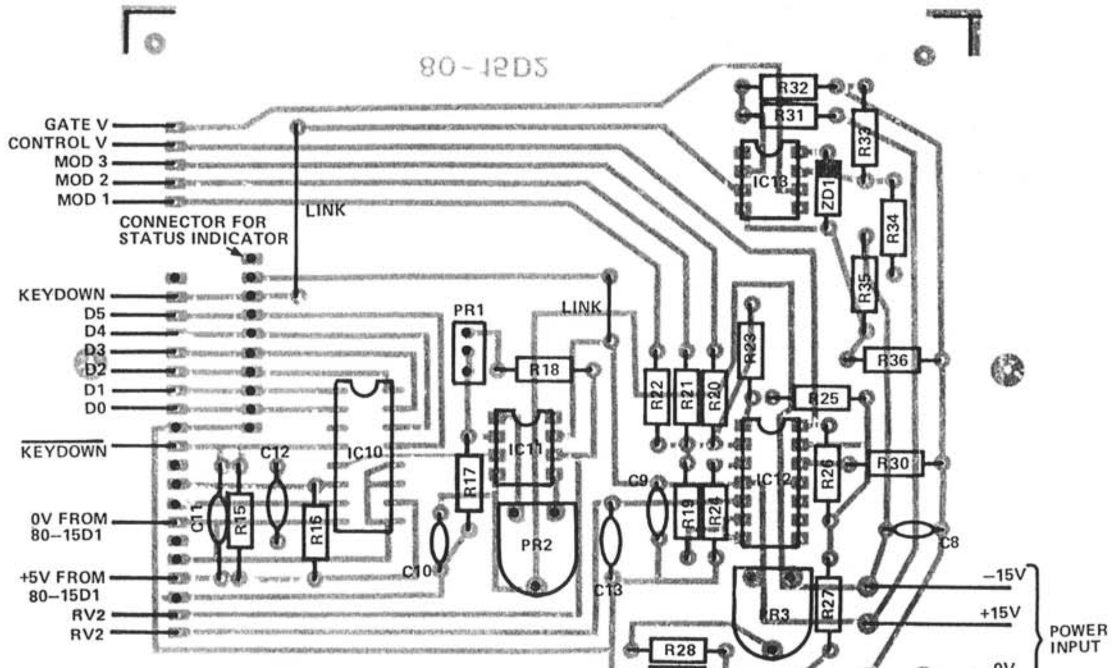
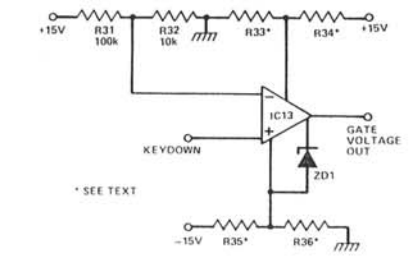
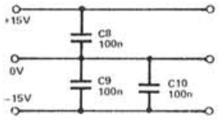


Fig.4 Component overlay of the digital to analogue converter.





Custom ICs for envelope generators generally require both a gate and a trigger voltage to produce a full ADSR envelope. To make the 80-8 and 80-10 envelope generators versatile and also allow manual generation of envelope signals the necessary trigger pulses are derived within these modules. The 80-10, however, also has an independent trigger output and so a method of providing trigger pulses at will is a useful addition to the keyboard controls.

For a 49 note keyboard there are four PCBs. The main reason for this is to make the PCBs acceptable for different makes of keyboard. They have been used with both the Kimber Allen SKA and the keyboard marketed by Clef Products. The first step is to identify the arrangement of PCBs (Fig. 10).

Turn the keyboard so that the keys face downwards and firmly support it in such a way that none of the keys are pressed down. Now arrange the four PCBs as outlined. Using some of the key contacts, position each PCB in turn so that the contacts line up with the solder connections on the PCB and also with the

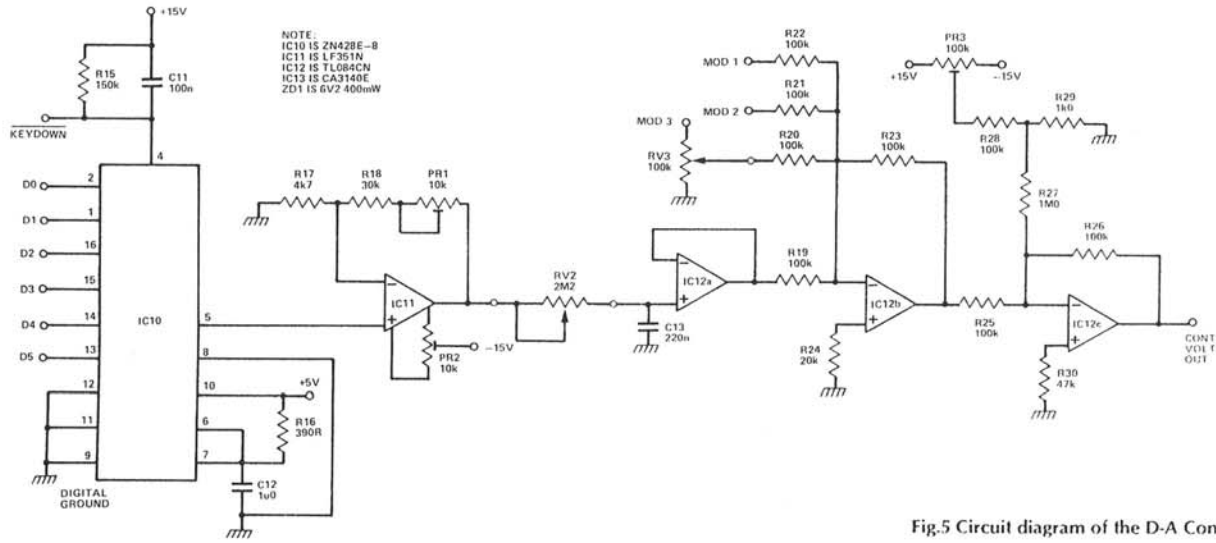


Fig.5 Circuit diagram of the D-A Converter.

PARTS LIST

49 NOTE KEYBOARD WITH CONTROLLER AND D TO A CONVERTER

Resistors

Carbon Film ¼W 5%	
R1,4,15	150k
R2,31	100k
R3	22k
R24	20k
R30	47k
R32	10k

Metal Film ¼W 1% 100ppm TC	
R16	390R
R17	4k7
R18	30k
R19,20,21,22,23,25,26,28	100k
R27	1M0
R29	1k0

Others	
R5-12	10k DIL resistor network (RS 140-057)
R13	150R, 4 W wirewound
R14	330R, 4 W wirewound

Potentiometers	
RV1	for future expansion
RV2	2M2 logarithmic
RV3	100k linear

PR1	10k cermet multiturn
PR2	10k cermet
PR3	100k cermet

Capacitors	
C1	4n7 polycarbonate
C2	47u 25V PCB electrolytic
C3	10u 40V PCB electrolytic
C4-10	100n ceramic disc
C11	100n polyester
C12	1u0 MKH polyester
C13	220n MKH polyester

Semiconductors	
IC1,8	CD4001B
IC2	CD4024B
IC3,4	CD4724B
IC5,6	CD4081B
IC7	CD4078B
IC9	LM78L05ACZ
IC10	ZN428E-8
IC11	LF351N
IC12	TL084CN
IC13	CA3140E
D1-49	1N4148
ZD1	6V2 400 mW zener

Miscellaneous

49 note keyboard; 49 GJ SPCO key contacts; sundry PCB connectors.

key plunger. Keeping the PCB held down scribe its location onto the keyboard frame. The PCBs may now be glued to the frame with Araldite Rapid adhesive. The next step is to make connections between the tracks. In most cases there will be sufficient gap between the PCBs to allow the use of proprietary PCB connectors, suitably shortened, to make the connections between the tracks.

If possible, PCB connectors should be used when positioning the PCBs on the frame. They greatly simplify joining up. Cut the pins so that there is sufficient metal to bridge the gap between the PCBs, but will not interfere with the placement of the key contacts. Also, when soldering them in place, ensure that excess solder does not flow sideways as this may subsequently prevent the key contacts from seating properly.

Ensure that the keys are aligned properly. With the recommended type the top edge of the contact will line up with the top edge of the PCBs. After all keys have been glued allow the adhesive to set overnight. Then the connections can be soldered to the PCBs as shown in Fig. 11. The diodes are now soldered in place. D1 is connected betweeen the track for Column 0 and the pad for key 1; D2 from the track for Column 2 to key 2 and so on until D8 which connects Column 7 track with key 8.

The final step is to connect the keyboard to the scanner PCB. The use of Molex connectors on the latter PCB is recommended while the other end of the wire is soldered directly to the tracks. For the monophonic keyboard Column 0 and Row 0 go to the first key, since this will result in a zero voltage from this key when pressed and the output is used for calibration purposes. Furthermore the 80-2 VCO is set such that 0 V at the Control Input will produce a frequency of 65.41 Hz, which relates to the first key.

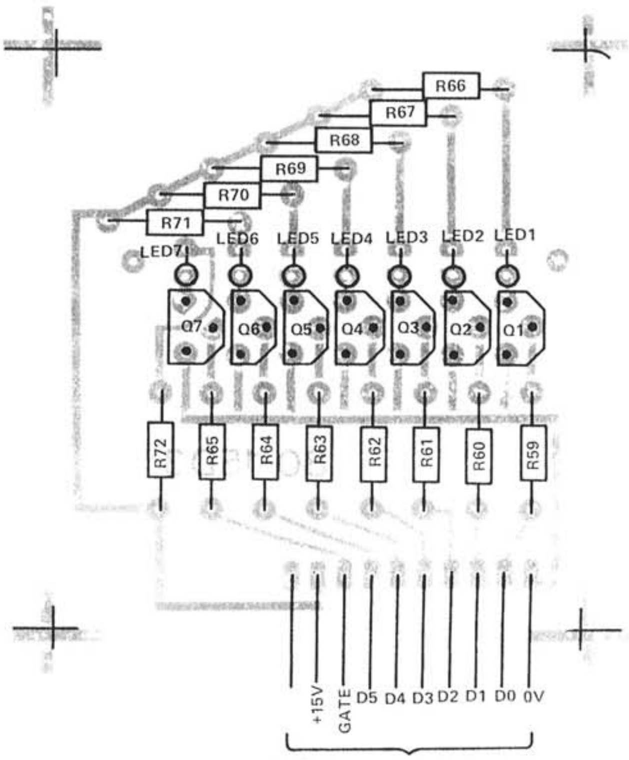


Fig.6 Component overlay of the keyboard controller status indicator.

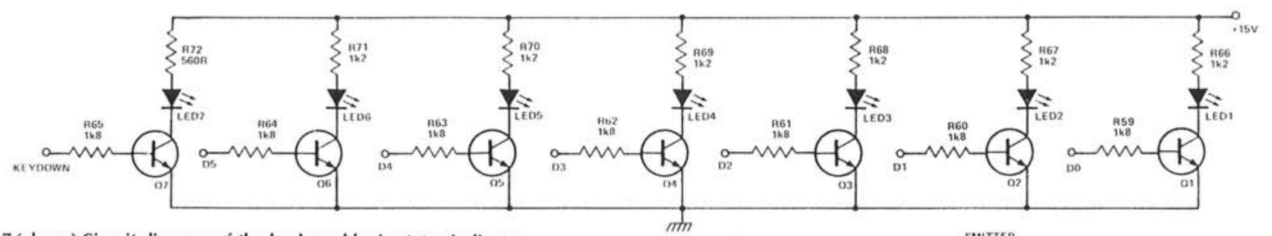
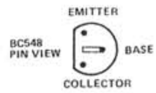
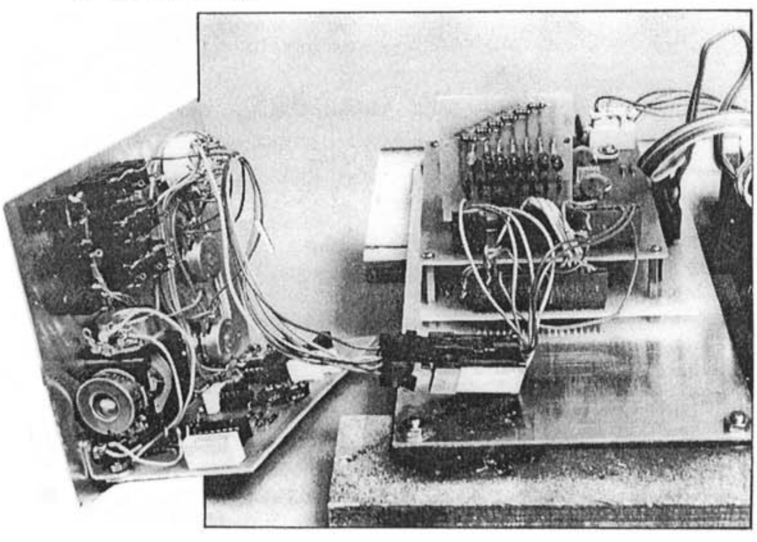


Fig.7 (above) Circuit diagram of the keyboard logic status indicator.

NOTE:
Q1–7 ARE BC548
LED1–6 ARE 3mm RED
LED7 IS 3mm GREEN



In the option A layout (below), the D-A Converter, Keyboard Scanner and Logic Board and Logic Status Indicator are incorporated in the keyboard case. (See also Fig.14).



BUYLINES

The following component packs are available from Digisound Ltd, 13 The Brooklands, Wrea Green, Preston, Lancs PR4 2NQ. All prices are inclusive of P&P and VAT.

1. Set of PCBs, connectors and diodes for 49 note keyboard — £11.25
2. PCB and components for Keyboard Logic Controller, D to A Converter and +5 V supply parts — £29.90
3. PCB and components for Logic Status Indicator — £3.90
4. PCB and components for Octave Shift, X-Y Controller and Touch Trigger — £16.10

PARTS LIST

LOGIC STATUS INDICATOR

Resistors
Carbon Film ¼ W 5%

R59,60,61,62,63, 64,65	**1k8**
R66,67,68,69,70,71	**1k2**
R72	**560R**

Semiconductors

Q1-7	**BC548**
LED1-6	**3mm red LED**
LED7	**3mm green LED**

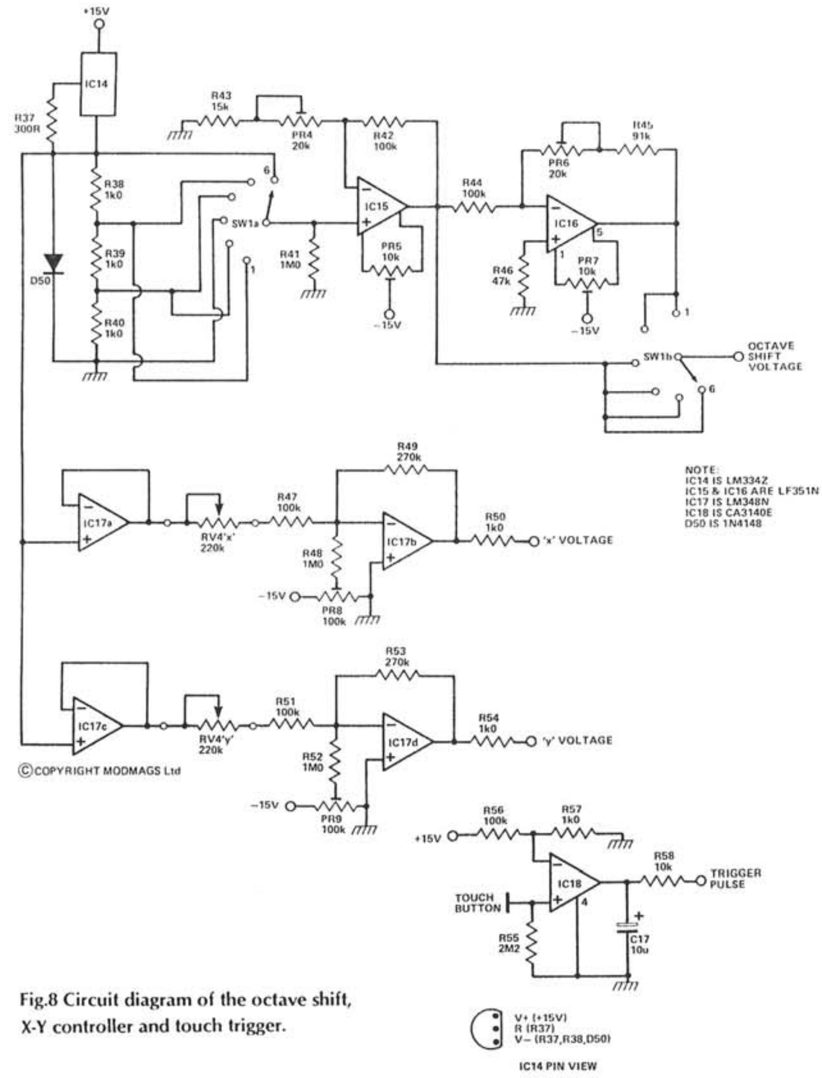
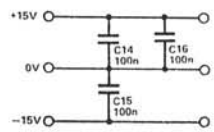


Fig.8 Circuit diagram of the octave shift, X-Y controller and touch trigger.

PARTS LIST

OCTAVE SHIFT: X-Y CONTROLLER: TOUCH TRIGGER

Resistors	
Carbon Film ¼W 5%	
R41,48,52	1M0
R46	47k
R47,51,56	100k
R50,54,57	1k0
R55	2M2
R58	10k
R49,53	270k
Metal Film ¼ 1% 100ppm TC	
R37	300R
R38,39,40	1k0
R42,44	100k
R43	15k
R45	91k
Potentiometers	
RV4	X-Y Joystick, 220k pots
PR4,6	20k cermet multiturn
PR5,7	10k cermet
PR8,9	100k carbon
Capacitors	
C14,15,16	100n polyester
C17	10u 40V PCB electrolytic
Semiconductors	
IC14	LM334Z
IC15,16	LF351N
IC17	LM348N
IC18	CA3140E
D50	1N4148
Miscellaneous	
SW1	2 pole 6 way rotary switch



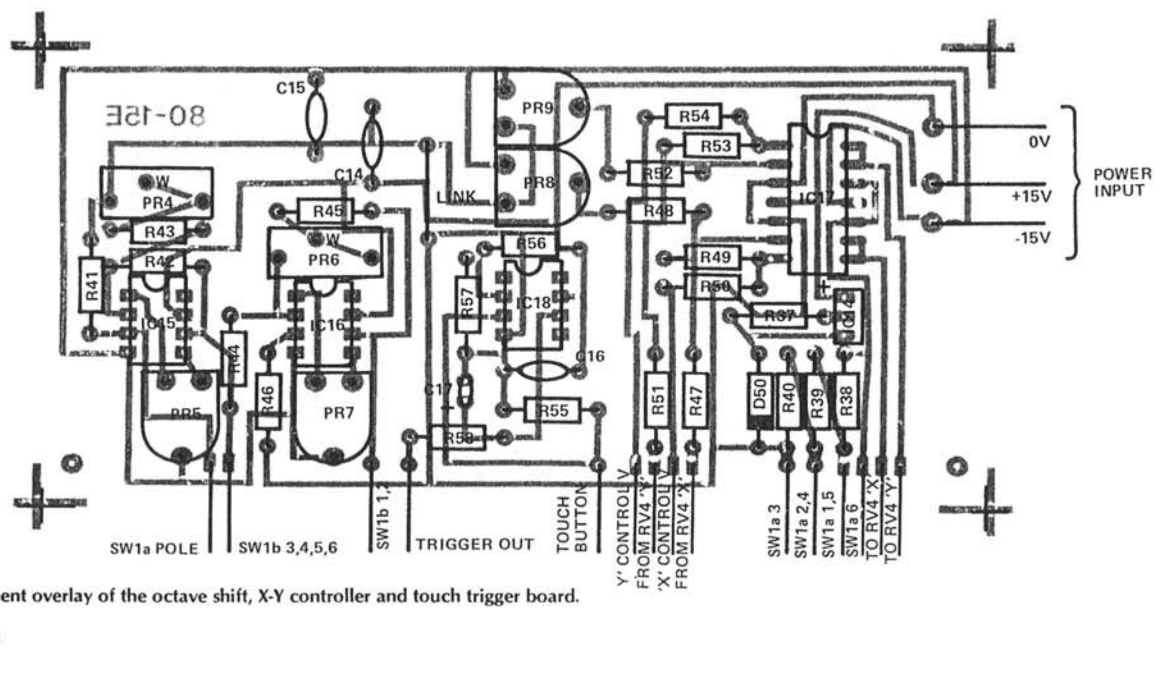


Fig.9 Component overlay of the octave shift, X-Y controller and touch trigger board.

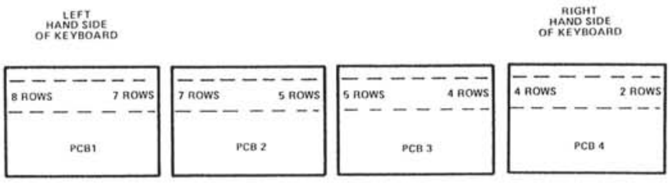


Fig.10 Arrangement of the keyboard PCBs.

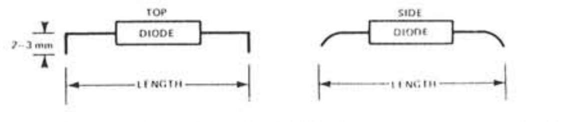


Fig.13 Shaping the diodes to avoid leads bridging across Column tracks.

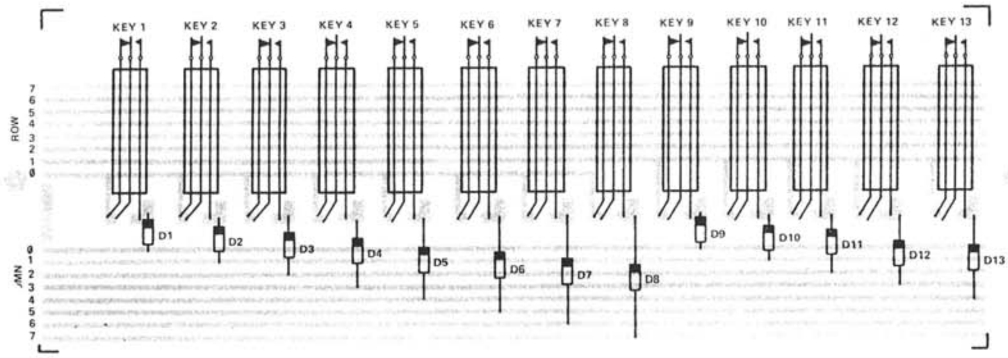


Fig.11 Keyboard component overlay.

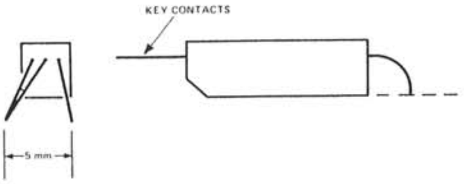


Fig.12 Shaping the key contacts. Each key should be firmly gripped at its sides when splaying the contacts, to avoid breaking the rivets. The leads should be level with the contact base. Before soldering, place the key contact on a flat surface to check that the leads are level. The key contacts can now be glued to the PCB with the same epoxy resin adhesive as before.

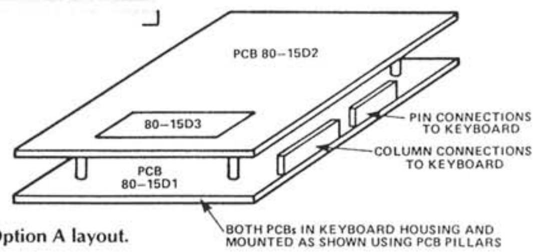


Fig.14 Option A layout.

Controller And D To A

There are two options for the connections between these two PCBs and for the general arrangement of the synthesiser controls. In Fig. 14 both the scanner PCB and the D to A Converter are housed within the keyboard case and the appropriate control voltages made available from the control panel sited to the left of the keyboard. The preferred arrangement is that in which the control data from the scanner is taken to a 15-way connector on the control panel and to house the D to A converter in a separate module.

When the 80-15D1 PCB is complete with the power supply components there will be three unused holes in the PCB, one adjacent to C17 and two adjacent to pin 1 of IC8. These should be ignored. On the 80-15D2 PCB to make the gate voltage compatible with the ETI-80 envelope generators put wire links in the places shown for R34 and R36 and ignore R33 and R35. Note that for option A (all PCBs within the case) both the 80-15D1 and 80-15D2 boards should be fitted with Molex PCB pins and 80-13D3, if used, will require a Molex edge connector. With option B the D to A converter only requires Molex pins for the 80-15D3 and the latter does not require an edge connector.

The last step is to make the connection to the 80-1 power supply. Constructors with a regulated +5 V supply which can be conveniently used will not have to install components R13, R14, C2, C3 and IC9 on the 80-15D1 PCB and the external supply goes to the positive and ground inputs shown in Fig. 3, but with a wire link made further down the 'positive' track and to the hole which would have been occupied by the positive lead of C3.

CONNECTIONS:

1. BETWEEN OUTPUTS OF 80-15D1
a) Connect the two RV1 leads together
b) Connect KEYDOWN TO INHIBIT
2. BETWEEN 80-15D1 AND 80-15D2
a) D0 to D5 connections
b) 0V connections
c) +5V connections
d) KEYDOWN connections
3. BETWEEN 80-15D2 AND 80-15E
a) MOD. 1 to octave shift output
4. BETWEEN 80-152 AND CONTROL PANEL
a) Control voltage output
b) Gate voltage output
c) Portamento control RV2 (two leads)
d) Mod. 3 to RV3

Setting Up And Calibration

Keyboard Controller and D to A Converter: Using the logic status indicator press each key in turn and note that a stable keycode is displayed and the gate LED comes on. The best approach is to list any keys that do not function correctly when the key is lightly pressed. If only a few fail to function then most likely the key contacts require a slight adjustment. If more fail, look for patterns of eight which then may correspond to connections on a particular Column or Row, for example, if there is no response to keys 9 to 17 this indicates a fault on Row 1. The status indicator will also show that there are no crossed wires in the Column and Row wiring since, pressing the keys in the sequence lowest to highest will generate the six bit binary code in logical order.

If the status indicator is not used then connect a voltmeter to the gate output from the D to A Converter and note that about +5 V is obtained for each key pressed. Next connect the voltmeter to the control voltage output and note that the voltage increments as each key is pressed, from left to right.

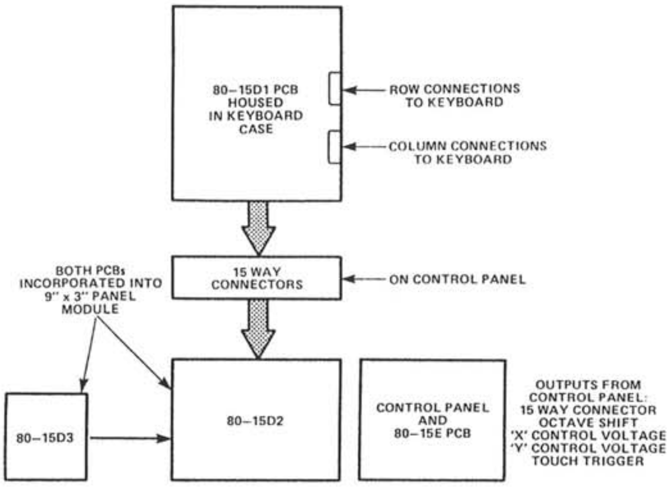
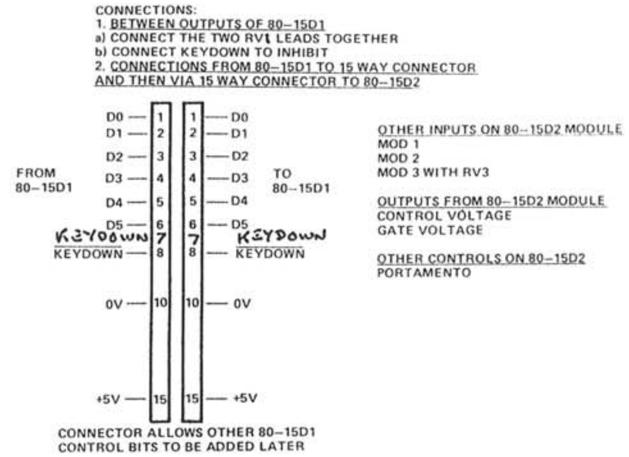


Fig.15 (above) Option B layout.



With the keyboard logic functioning and key contacts adjusted where necessary the last step is to calibrate the D to A Converter. Press the first note and keep it held down and ensure that the octave shift switch is on zero and the portamento control is fully anti-clockwise. Connect a voltmeter to R18 at the end which connects pin 6 to IC11 and adjust PR2 for 0V000. Next connect the voltmeter to the control voltage output and adjust PR3 to obtain 0V000. With the voltmeter still connected to the control voltage output release the first key, hold down key 49 and adjust PR1 to give 4V000. Lastly press all the C keys to determine that they produce 0, 1, 2, 3 and 4 V output. Some minor adjustments may have to be made to PR1 to obtain these values within a few millivolts of their correct values. Next check that the portamento control is functioning. ie, as the control is turned clockwise the time for the voltage to reach its true value will increase.

Place the octave shift switch on way 3 (0 V) and measure the voltage at the output. Adjust PR5 to give 0 V. Leave the switch in the same position and measure the voltage at R45 where it connects to pin 6 of IC16 and adjust PR7 for zero output. Turn switch to +2 octaves and adjust PR4 to give 2V000 at the output. Lastly turn the switch to −2 octaves and adjust PR6 to give −2V000 at the output.

To adjust the X-Y controller keep the joystick in the bottom left hand position and first adjust PR8 for 0 V from the 'X' output and the PR9 for 0 V from the 'Y' output. Place the joystick in the top right hand corner and check that a voltage in excess of 0V9 is available from both 'X' and 'Y' outputs.

Keyboard Housing

Detailed construction for a housing is not provided since dimensions will vary according to the keyboard used.

1. Make the keyboard housing about 35inches (94cms) long so that a module housing containing a row (or rows) of 12 modules will seat neatly on top. This length also provides adequate space at the left hand side for the keyboard scanner PCB and the control panel.
2. The control panel is 162 x 115 mm and the hole in the keyboard casing should only be slightly smaller so as to allow this panel to be passed through the hole when assembling or dismantling.
3. The overall depth of the case is about 29 cm which provides a firm base for the module as well as leaving sufficient space behind the keyboard for the power supplies. Connections from these power supplies to the module housing can be made with miniature three-pole connectors and these should be clearly identified, or preferably of different design, to prevent wrong connections which will damage the modules.
4. The case should have a 7-8 cm step above the level of the key. This serves two purposes: (a) it allows the power supply PCB to be mounted vertically for easy access to the trimmers; and (b) it provides a valuable gap between the keyboard and the modules and avoids having patchcords drooping onto the keys.
5. The rear panel should be preferably ventilated.

PROJECT 80 MODULAR SYNTHESISER

Project 80 began in February 1980 and continued throughout 1980. The modules appeared as follows:

ETI February	1980	VCO, PSU and VCLFO
ETI March	1980	Voltage Controlled Mixer
ETI May	1980	Low-pass, High-pass, Band-pass and State Variable Filters
ETI July	1980	State Variable Filter and Envelope Shaper
ETI August	1980	Dual VCA
ETI September	1980	Envelope Shaper
ETI October	1980	Monitor Amplifier
ETI April	1981	Noise Generator

For the latest information on prices of kits for these modules, contact Digisound Ltd, 13 The Brooklands, Wrea Green, Preston, Lancs PR4 2NQ. Tel: Kirkham (0772) 683138. Backnumbers of these issues of ETI are available for £1 per copy inclusive of post and packing. Send your cheques and POs (made payable to Modmags Ltd) to:

Backnumbers,
Modmags Ltd,
145 Charing Cross Road,
London WC2H 0EE.

KEYBOARD CONTROLLER. ADDITIONAL CONSTRUCTION NOTES

A. LOGIC CONTROLLER. 80-15D1

1. Wire links on PCB are denoted by solid lines between PCB holes, note the small link between IC 1 and the edge. Ignore links marked 'A' and 'B'.
2. If a +5V supply is available then this is connected between the inputs on rear of PCB marked 0V and +5V. Alternatively this external +5V supply may be connected at the appropriate inputs on the front edge of the PCB. When the supply is taken from the 80-1 PCB the positive input must be connected to input at rear marked '+V' and components R13, R14, C2 and IC9 installed. Install these before inserting the other IC's and check that about +5V is obtained. The 0V line can be picked up at the wire link going from the track to the right of C2 and the +5V is available at the link to the left and below IC4. This +5V supply is only for the keyboard controller kits and no external equipment must be attached to it.
3. RV 1, to alter clock rate, is not installed at this time and is not included in the kit.
4. Refer to construction notes regarding other edge connections.

B. D-A CONVERTER. 80-15D2

1. Resistors R33,34,35 and 36 are not included. Insert wire links ONLY in place of R34 and R36 which will produce a +5V gate pulse.
2. Other wire links shown by solid line between PCB holes.
3. Mount potentiometers on panels with pins facing downwards.
4. Both 0V and +5V must come from the logic controller supply via the 15 way connector.
5. The PCB can be hard wired between the 15 way socket and the PCB.
6. The PCB shows a capacitor, C_x, in parallel with R26. This is to avoid oscillation when a microprocessor is installed and is 22pF. It is included in the kit and should be installed now.

C. LOGIC STATUS INDICATOR. 80-15D3

1. This may be connected to 80-15D2 PCB using a Molex connector on the latter PCB or it may be hard wired.
2. The resistors and transistors should be soldered in place first. The top of the transistors should be about 9 to 10mms above the PCB surface.
3. Short lead of LED's towards the BC 548 transistors. The LED's must all be the same height which can be achieved by placing a strip of thin cardboard between the LED pins. The strip to be about 11 or 12mms wide, i.e., the base of the lens will be about 11/12mms above the PCB surface when finished. Nuts and bolts are provided for attaching PCB to proprietary panel.
4. Note the position of green LED before soldering LED's.

D. OCTAVE SHIFT, ETC. 80-15E

1. The X-Y potentiometer (RV4) is 100k and R49,53 become 470k which produces about a 1V swing on both X and Y axes.
2. The X-Y potentiometer should be mounted as illustrated in Figure 2 which is viewed from the top and can be adjusted to give zero X and Y outputs when paddle is in bottom left hand corner.
3. A black knob is included in the kit which can be glued to the paddle. The long bolts are for mounting the X-Y pot to the panel.
4. The touch trigger is constructed as shown in Figure 3. Put the sleeved grommet into the panel hole. Screw the pillar to the chrome head and push it through the grommet. Note that the thread on the chrome head is not very accurate and sometimes the pillar will slip over the thread. In all cases, however, we have found that the pillar will lock tight. Finally bolt the solder tag to the pillar.
5. The PCB is joined to the panel with L brackets provided and the foil side should face the front edge of the panel.

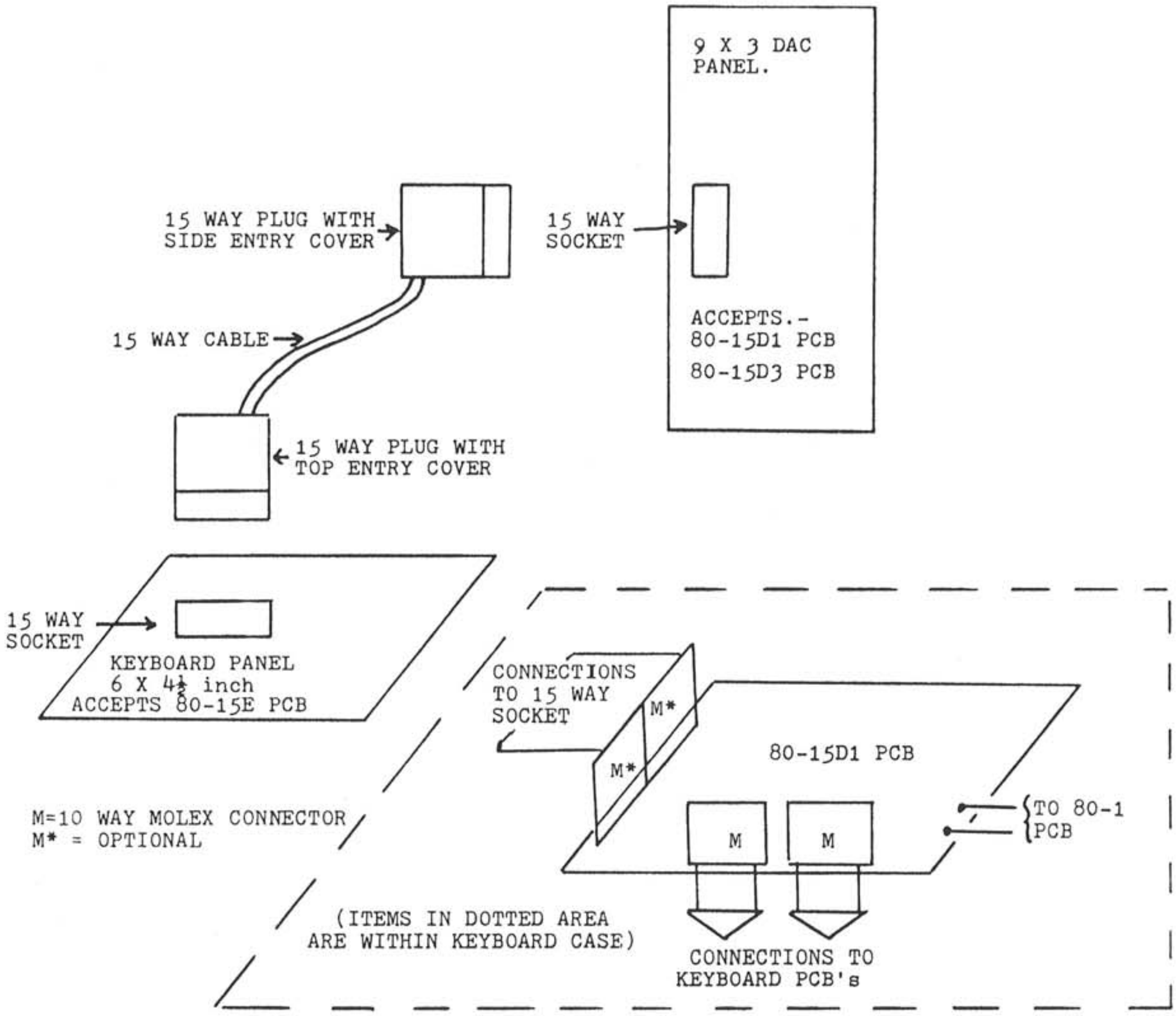


FIGURE 1. ARRANGEMENT OF KEYBOARD CONTROLLER. NOT TO SCALE. PANEL DETAILS NOT SHOWN.

The above arrangement simplifies future expansion when the 80-15D1 PCB must be accessed for changes to links and additional wires to the 15 way socket. It allows for a microprocessor to be interposed between 80-15D1 PCB and a digital to analogue converter(s). See price list for details of keyboard controller requirements.

NOTE: 1. On both panels the 15 way connector socket socket should be mounted with the flange behind the panel; 2. Keep logic control wires, i.e., to and from 15 way sockets and the plug link as short as possible.

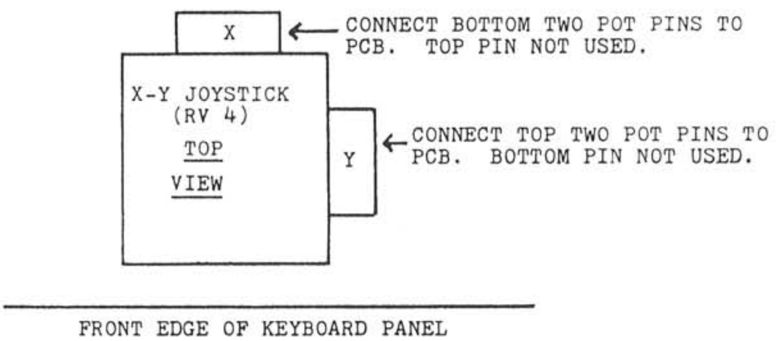


FIGURE 2. WIRING OF X-Y JOYSTICK. (RV 4)

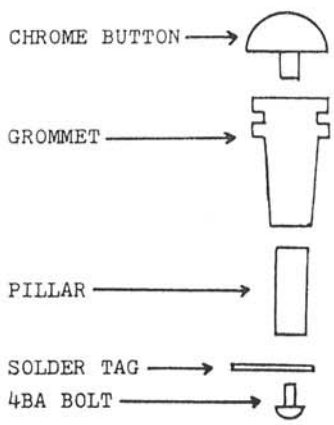


FIGURE 3. ASSEMBLY OF TOUCH TRIGGER

digisound

MODULE 80-16
DUAL RESONANT FILTERS

1. INTRODUCTION

The main application of the DIGISOUND 80-16 Dual Resonant Filters (DRF) is to boost specific bands of frequencies as an aid to more closely simulating traditional musical instruments. The technique is, however, applicable to the modification of 'electronic' sounds. The frequency range of the filters is typically 30 to 3,500Hz and their Q may be varied from 0.5 to 10. The selected frequencies may be boosted by about 13dB above the original signal and a particular advantage of the current design is that this boost is available at all Q levels without having to worry about possible overvoltage and distortion. In fact the peak outputs are within +/-1dB, irrespective of Q, and the 'gain' is adjusted by attenuating the bandpass outputs.

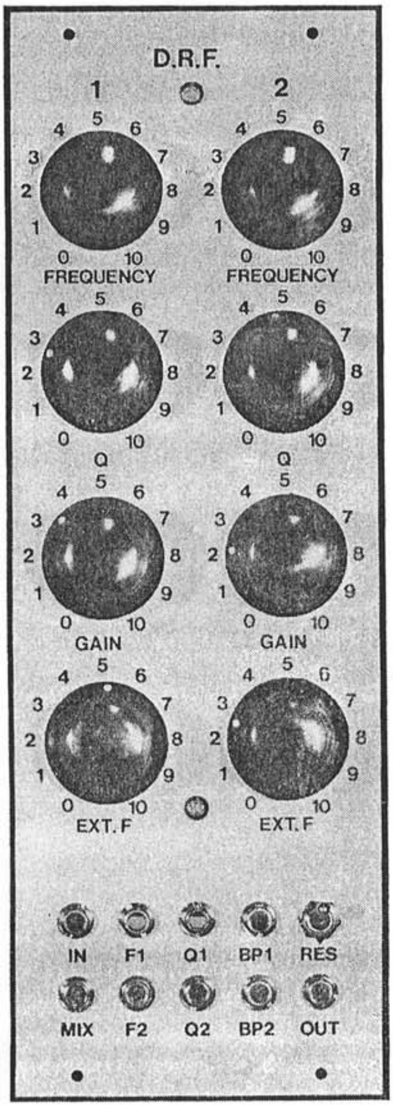


FIGURE 1. 80-16 PANEL

The design is based on the CEM 3350, from Curtis Electromusic Specialties, and this IC is a Dual Voltage Controlled State Variable Filter. As a result of using this device it becomes inexpensive to add external voltage control of both frequency and Q. The 80-16 module therefore provides this capability plus independent band pass outputs for each filter. These latter outputs are inverted with respect to the input signal and are convenient for the combination of two modules in parallel when more than two resonant peaks, or bands, are required. The bandpass outputs are also available with the same polarity as the input signal and the module is converted to resonant filters by switching in a proportion of the original signal. The 80-16 may therefore be used as dual bandpass filters and the external voltage control allows the generation of swept bandpass and other 'voicing' effects.

2. DESIGN

The complete circuit diagram for the 80-16 DRF is shown in Figure 2. It is readily apparent that it is centred around IC2, the CEM 3350, whose pin out and functional block diagram is shown in Figure 3.

Most of the design is best understood by reference to specific features of the CEM 3350. First, although the IC accepts a wide range of power supplies it is not guaranteed to withstand a total voltage greater than 26V across the V_{CC} and V_{EE} pins. To avoid this situation the CEM 3350 is operated from +/-12V supplies by using the 100mA 12V regulators, IC3 and IC4.

Next, the transconductors are the usual NPN differential pairs with current mirror active loads, similar to the CA3080 and LM13600 operational transconductance amplifiers (OTA's).

In common with these latter devices the signal levels to the CEM 3350 must be attenuated to a low level, typically 20 to 80mV, to avoid distortion. The inverting input stage built around IC1A provides an initial 33% signal reduction and further attenuation is obtained with R3/R4 on one side and R21/R22 on the other. Thus for a 10V p-p input signal the input to the CEM 3350 will be about 20mV. The input signal are connected to pins 4 and 14 which are termed the Variable Gain Inputs, V_{IV}. These inputs allow a fairly high signal levels without the filter going into 'jump resonance' at high Q settings because the peak gain can never exceed unity at any gain setting. It is the use of these inputs which allows the amplitude of the output signals to be kept constant irrespective of the frequency and Q settings so long as the latter are within the range specified in the Introduction. Obviously higher frequencies and Q levels are obtainable but the output level will begin to fall outside of the specified +/-1dB and the frequency and Q range provided is satisfactory for resonance filters.

As a matter of interest the CEM 3350 also has what is termed 'Fixed Gain Input' at pins 2 and 12 and in most applications of the device the signal will be shared between the Fixed and Variable Gain Inputs in order to use moderate signal levels at high Q levels.

Both low pass and band pass outputs are available simultaneously from the IC and for this application the outputs are naturally taken from the band pass outputs at pins 5 and 15. The filter outputs are of high impedance and they should be connected to high impedance buffer-amplifier with a low input bias current in order to minimise loading of the filter outputs. IC 5A and IC5B are arranged as non-inverting amplifiers and their gain set to restore the signal to the same level as the input signal, e.g., 10V p-p.

Reference to Figure 3 shows that there is a separate transconductor for both frequency and Q control on the two halves of the filter. These cells are all connected to the I_{ref} pin (pin 1) and the value of this current

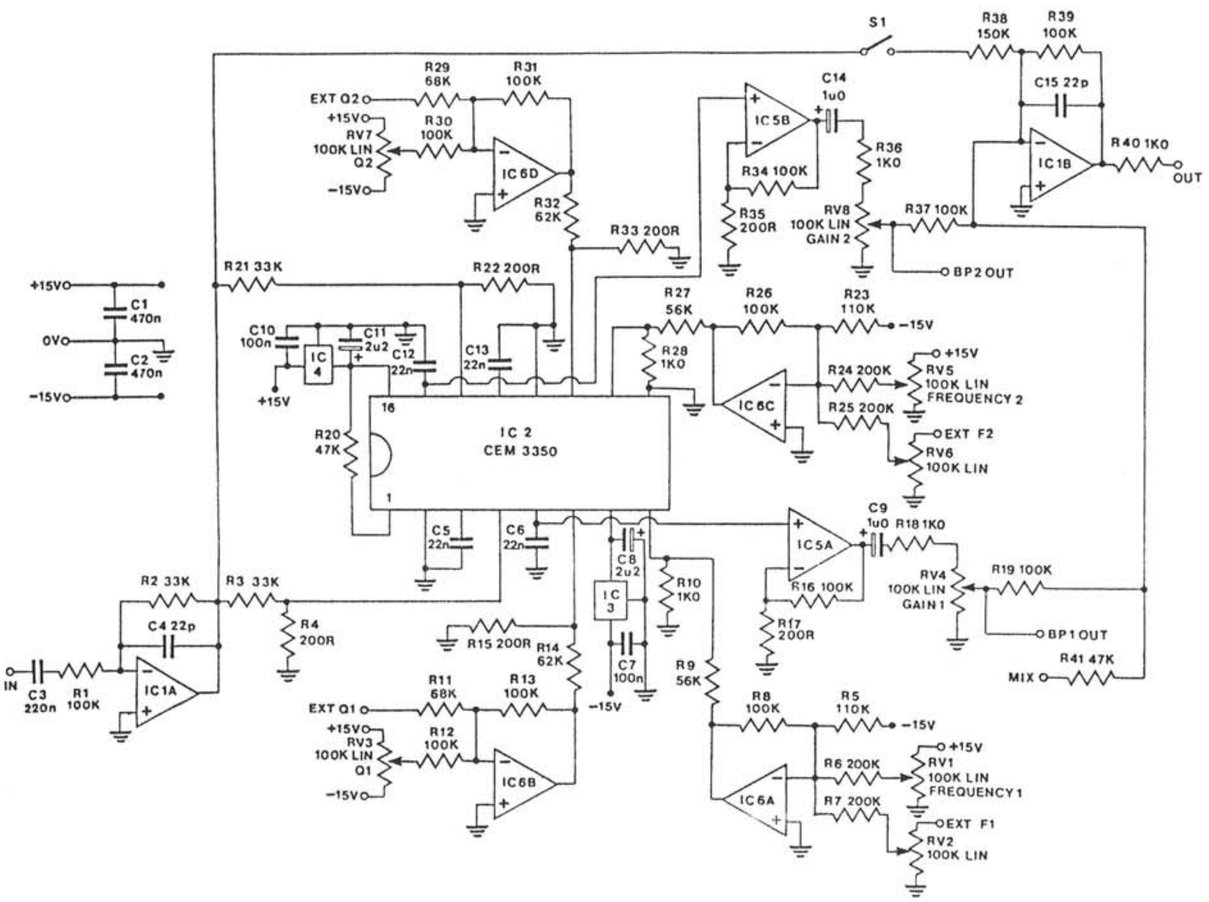


FIGURE 2. CIRCUIT FOR 80-16 DRF

determines the response of the transconductor cells. With a current of 50-200uA the cells will respond to control voltages in a linear fashion whereas with I_{ref} at 400-600uA the response is exponential. R20 connected to the +12V supply produces a nominal 400uA. The frequency and Q controls on both halves are preceeded by inverting amplifiers derived from the quad op-amp IC6. These serve to bring the controls into the usual response, namely, an increasing positive voltage applied to the appropriate op-amp inputs will increase frequency or Q. They also allow external voltage control inputs and for the external frequency control attenuating potentiometers, RV2 and RV6, have been incorporated into the panel. As stated earlier the manual Q controls, RV3 and RV6, allow Q to be varied between 0.5 and 10 with an exponential control response while RV1 and RV4 allow manual setting of frequency in the range of about 30Hz to 3,500Hz. The frequency range is, of course, also governed by the timing capacitors C5, C6, C12 and C13.

If frequency and Q is kept within the specified range then the outputs from the buffer amplifiers, IC5A and IC5B, will be bandpass outputs close to unity gain. These outputs are connected to attenuating pots RV4 and RV8 which allow adjustment of gain prior to the mixing stage built around IC1B. Note that from the wiper of these potentiometers the bandpass outputs, BP1 and BP2, are also available but their signal will be inverted with respect to the input signal. This polarity inversion is not a disadvantage for many bandpass applications. They also provide an easy method of combining two modules in parallel via the 'mix' input, R41, as discussed in the applications section. From the attenuators the two bandpass outputs are combined in the inverting summer, IC1B. With S1 open the two bandpass outputs are present at the output with the same polarity sense as the input signal. Closing S1 allows part of the original input signal to pass through and thus the bandpass outputs are then simply boosting specific frequencies present in the original signal. Starting with a 10V p-p signal which is reduced to 3V3 at IC1A the original signal is further reduced by R38/R39 to 2V2. The maximum output from either of the

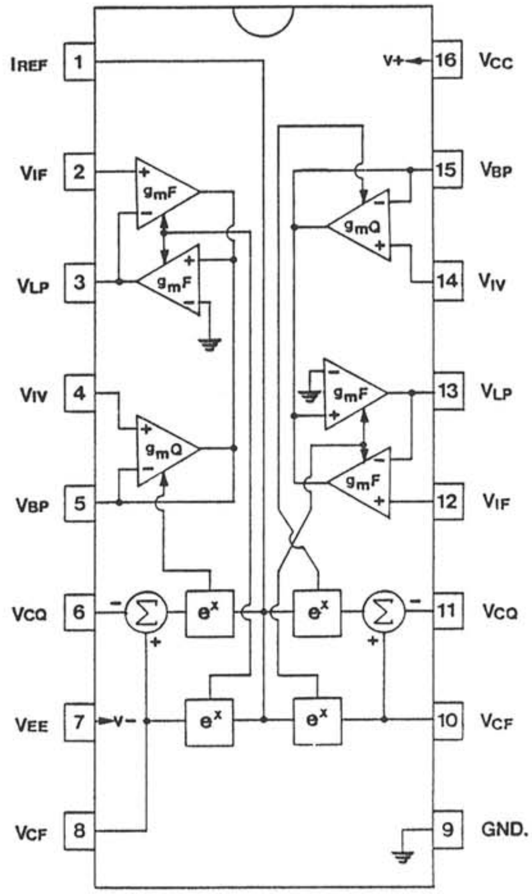


FIGURE 3. CEM 3350 IC

bandpass filters is 10V p-p and thus their maximum boost compared with the original signal is 4.55 times, or +13dB. To alter the module for other signal levels it is usually only necessary to alter the value of R2 to restore this ratio but care should be taken to ensure that the signal into the CEM 3350 is about 20mV. For much lower signals, of the order of a few volts, it would be more advantageous to reduce the value of R3 and R21 so as to maintain 20mV at the V_{IV} pins of IC2.

3. CONSTRUCTION

The 80-16 PCB is printed with a component overlay which simplifies the construction stage. The overlay is reproduced in Figure 4 to allow checking of component placement after the module has been constructed.

Special care should be taken regarding orientation of the electrolytic capacitors and the IC's. For the latter even when the DIL sockets have been installed the number '1', denoting pin 1, will still be visible on the PCB. In any event compare the completed PCB against Figure 4 before applying power. Also ensure that the wire links have been made and that the foil side of the PCB is free from solder bridges.

The panel wiring diagram is shown in Figure 5 which illustrates the components when viewed from the rear of the panel. The arrows and associated numbers and letters indicate that a wire connection must be made from the position shown to the PCB which has the corresponding identification mark on its mounting edge. Note, however, that the wipers of RV4 and RV8 are also connected to jack sockets J4 and J8 respectively. The jack sockets in the diagram are of the type supplied by Digisound Limited. The top connection, as illustrated, is the connection which is made with the jack plug when the latter is inserted. The lower connection is disabled by insertion of a jack plug. Finally, the tab under the socket is the ground connection. It is recommended that all of these ground connections are wired to the 0V line since this facilitates connection of the DRF to other equipment which may be powered from a separate supply. The ground tabs may be joined together using tinned copper wire but other panel wiring should be made with insulated wire. 1/0.6mm insulated wire is ideal for panel wiring since it retains any shaping and thus allows a neat appearance to be obtained. Wires should be kept as short as practical.

Before inserting IC2 check that the 12V regulators are producing the required voltage, about +12V from IC4 and about -12V from IC3. Turn off power before inserting IC2, the CEM 3350.

No calibration of the 80-16 DRF is required.

4. USING

In common with other DIGISOUND 80 modules the 80-16 Dual Resonant Filters are designed to accept 10V p-p signals with sufficient headroom to avoid distortion with greater signals. Usually the DRF will be located after the VCA, or after the mixer in a polyphonic system. A difficulty begins to arise as the original 10V p-p signal from the VCO is treated by several modules, that is, the signal amplitude is decreasing and if it is too low when it reaches the DRF then signal to noise problems may arise, particularly if used in the band pass mode at high Q settings. If this does occur then the normal signal level reaching the DRF should be checked and R2 adjusted accordingly, as described earlier. For example, for an average signal input of 5V p-p then R2 should be changed to 68k.

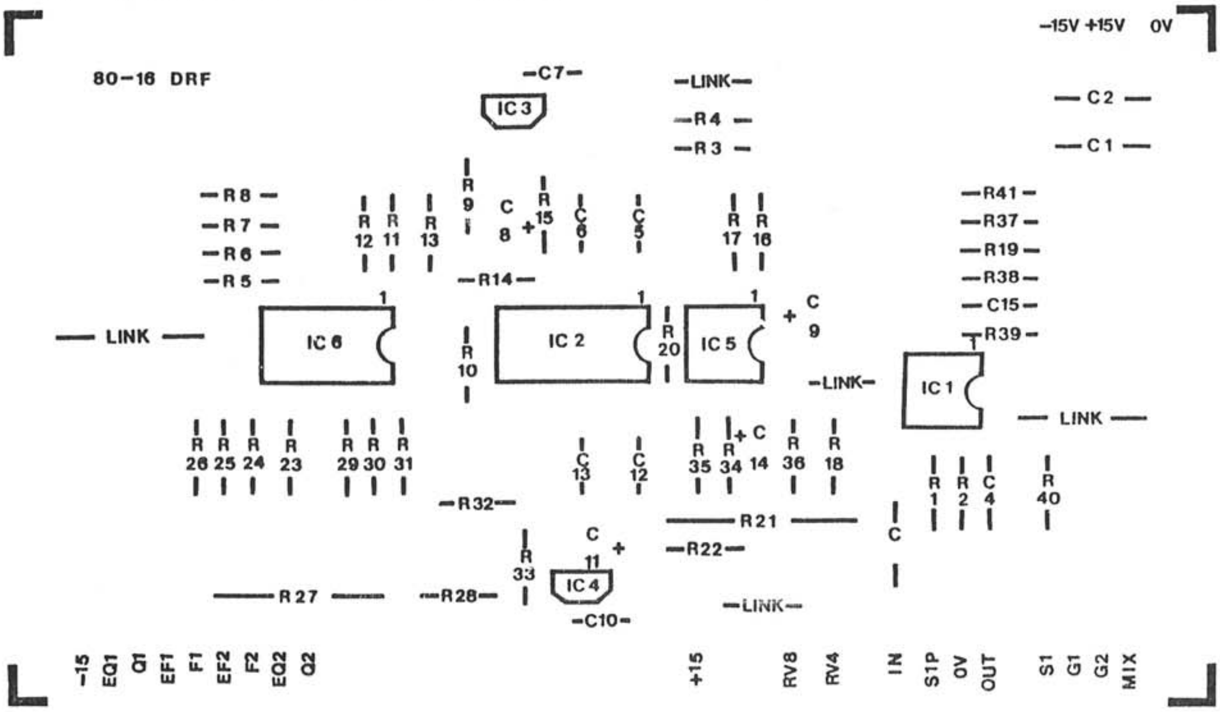


FIGURE 4. COMPONENT OVERLAY FOR 80-16 PCB

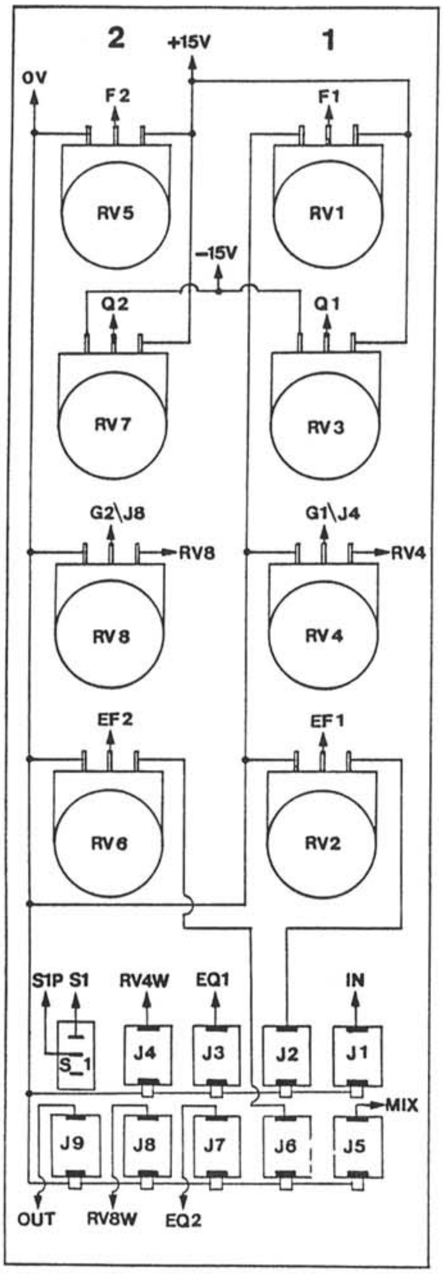


FIGURE 5. PANEL WIRING

Next one should become familiar with the frequency scale of the module. For resonance filtering the effect is best judged by ear and so accurate frequency scaling is uneccessary which is the reason for using 5% resistors and the absence of trimmers in the design. The manual frequency controls are scaled 0 to 10 and as a rough guide the frequency at each whole number will be 30, 50, 90, 150, 250, 420, 720, 1250, 2100 and 3500Hz. The actual values will vary somewhat on each half and also between modules.

The effect of Q setting should also be fully understood and this may be achieved by experimenting with the filters in the bandpass mode. Figure 6 shows the effect of two Q settings (1 and 10) on the signals passed when the frequencies are set at 1kHz and 3kHz. It will be obvious from examination of the response that as Q is increased then the apparent volume will decrease. Thus high values of Q are used to boost a particular harmonic while low Q values will boost a broad band of frequencies. The Q control is also exponential and so RV3 and RV7, which have a 0 to 10 scaling, will result in approximate Q values at the whole numbers as follows: 0.5, 0.7, 0.9, 1.2, 1.7, 2.2, 3.0, 4.1, 5.5, 7.5 and 10.

In the resonant filtering mode, i.e., with S1 closed, the main use of the DRF is to simulate the formants of traditional instruments. The formants are the resonant frequencies of the instruments arising from their mechanical construction and remain constant irrespective of the note played and so there is no need for the filters to track the keyboard control voltage. There are discrepancies

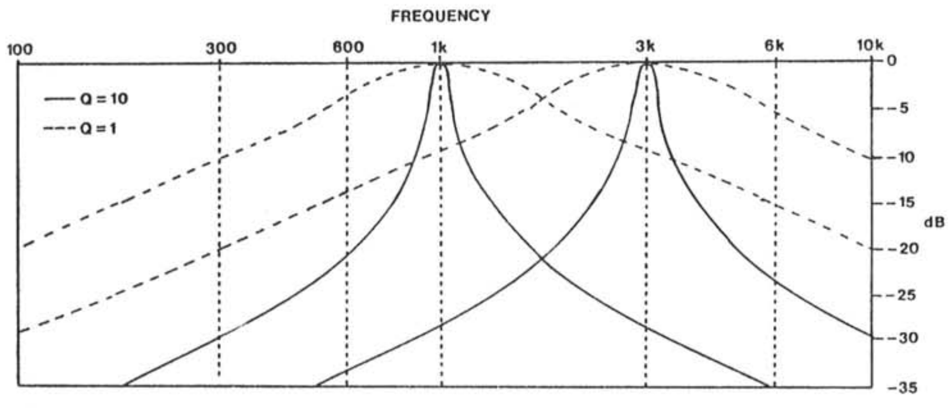


FIGURE 6. EFFECT OF Q ON BANDPASS RESPONSE

between various sources on the formant frequencies of traditional instruments but good results can quickly be obtained by experimentation and some thought given to the sound being imitated, or created. For example, a 'bass' instrument such as the tuba, bass guitar and some drums will obviously have their formants at the lower frequency settings, say, below 300Hz. On the other hand the more high pitched instruments such as woodwind instruments will have formants at about 1kHz or above. In almost every instance one will be using a high harmonic waveform and a moderate Q, say 3 to 5, would be a good starting point.

Using the DRF for the creation of 'electronic' sounds requires rather more in the way of experimentation. for frequency modulation effects attenuating potentiometers have been included on the panel which reduces the amount of patching required. One will have to remember to set the initial frequency, using the manual controls RV1 and RV5, at a point where the external control voltage will not put the frequency of the filter(s) much beyond 4kHz. As this value is exceeded the amplitude will decrease although this can be a useful effect. Likewise when using the external voltage control of Q the manual Q setting should not be more than about number 5 on the 0 to 10 scale otherwise the Q will go so high that very little sound will be obtained, particularly in the bandpass mode. the application of the external control facilities, as well as resonant filtering, is discussed in 'USING THE DIGISOUND 80 MODULAR SYNTHESISER'.

Whether the DRF is being used for formant filtering or for electronic effects it is often beneficial to use two modules and so give enhancement, or treatment, from up to four filters. The additional module will have to be connected in parallel with the first since the input signal is attenuated as it leaves the module. In practice accurate external mixing of two modules in parallel can be quite cumbersome and so the DRF has provision for combining two modules. of four filters are required then the use of a joined patchcord, as shown in Figure 7, connected to the two bandpass outputs, BP1 and BP2, will

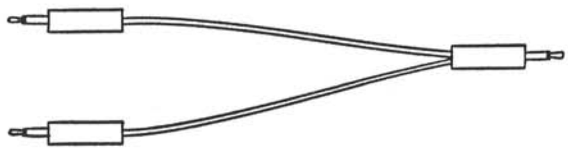


FIGURE 7. 'MIXING' PATCHCORD

produce the average of these two signals. If the two gain controls were at maximum rotation and the bandpass outputs at two separate frequencies (normal case) then the two outputs would be at half the input signal level. The combined output is connected to the 'mix' input and resistor R41 has been chosen to restore their level to unity gain. The two signals are also re-inverted to the same polarity as the input signal and mixed with the other bandpass outputs of the second module and, when in the resonance mode, the correct proportion of the original signal. Care must be taken when attenuating the bandpass outputs from the first module, especially if different levels of attenuation of the two signals are required. The effect of anti-clockwise rotation of the gain control is more accentuated than in normal operation and the reduction of one signal will cause an increase in the other signal. This latter effect is most evident near maximum gain. The other point to observe is that neither gain potentiometer should be at, or close to, zero. Thus if only three filters are required it should be one on the mixing module which is attenuated to zero. While this mixing may also seem cumbersome it is in practice quite easy to obtain the desired levels and it saves having to use a separate mixing facility.

5. COMPONENTS

RESISTORS, 5%, 1/4w carbon film

R1,8,12,13,16,19,26,30,31,34,37 and R39	100k
R2,3,21	33k
R4,15,17,22,33,35	200R
R5,23	110k
R6,7,24,25	200k
R9,27	56k
R10,18,28,36,40	1k0
R11,29	68k
R14,32	62k
R20,41	47k
R38	150k

POTENTIOMETERS

RV1,2,3,4,5,6,7,8	100k lin.

CAPACITORS

C1,2	470n polyester
C3	220n polyester
C4,15	22p polystyrene
C5,6,12,13	22n polycarbonate
C7,10	100n polyester
C8,11	2u2 PCB electrolytic
C9,14	1u0 PCB electrolytic

SEMICONDUCTORS

IC1	TL 082
IC2	CEM 3350
IC3	79L12
IC4	78L12
IC5	TL 072
IC6	LM 348N

MISCELLANEOUS

S1	SPDT sub. min toggle switch

digisound

MODULE 80-17
REVERBERATION

1. INTRODUCTION

The DIGISOUND 80-17 Reverberation module is a solid state replacement for the lower cost spring line reverberation units. We believe the sound is more natural and pleasing than that obtained from the latter and also provides other advantages, such as freedom from vibration and unnatural resonant peaks, low power demand, plus the ability to vary the reverberation time. On the other hand the 80-17 does share one shortcoming, namely a limited treated bandwidth of about 3.6kHz in order to produce a maximum useful reverberation time of about three seconds.

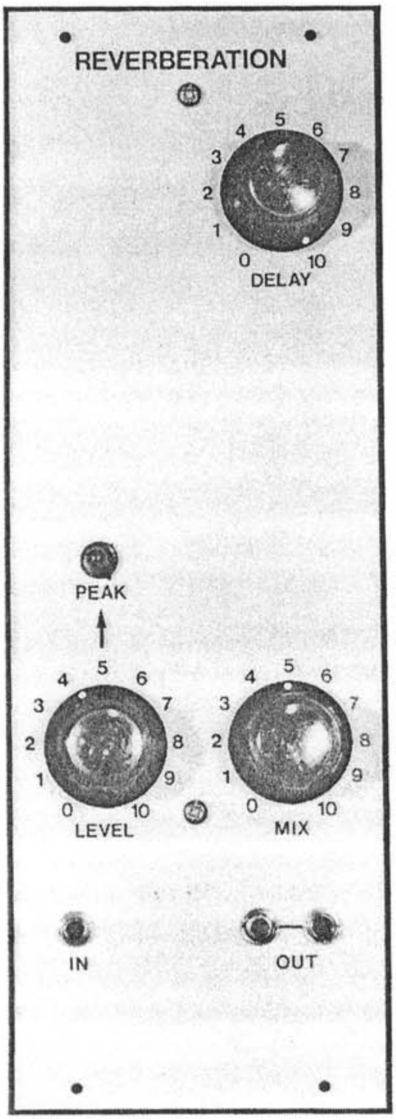


FIGURE 1. 80-17 PANEL

The design is based on a bucket brigade device which is essentially an analogue shift register for delay purposes. These devices are widely used for well-known electronic treatments such as phasing and flanging and often claims are made with some of these effects units that they provide reverberation. The latter effect is, however, very precisely defined and until recently it was impractical to implement reverberation at low cost using bucket brigade IC's. The other point to note is that reverberation is not an electronic effect. When listening to music in a large hall one is barely conscious of reverberation but in contrast when listening in a small room or in a venue with poor acoustics one does becomes very conscious of the thinness, or unnatural character, of the sound.

Sound emitted in an enclosed space will be subjected to both simple and multiple reflections from internal surfaces. Furthermore since these surfaces are at varying distances the time for these reflections to occur and decay will vary. The result is highly complex but the delays cause phase cancellation and reinforcement which produce the pleasing effect known as reverberation.

To implement the latter effect with conventional bucket brigade devices requires at least three dual 512-stage types together with some means of varying the delay from each section so as to avoid a 'fluttering' effect as a result of combining equal delays. These difficulties are overcome by using the Matsushita MN 3011 BBD which is a 3328-stage device having six tapped delays of unequal spacing.

2. DESIGN

The reverberation module utilises the MN 3011, which at the time of writing is the latest in a series of bucket brigade devices for audio applications to come from National Panasonic (Matsushita Division). They are all fabricated in P-MOS and one should start by ignoring much of what is currently in print regarding the disadvantages of P-MOS BBD's. It is a fact that they are somewhat limited in clocking speed, typically 10kHz to 100kHz, and also have a limited bandwidth of about 10 to 12kHz. The latter, however, is not usually a limitation since the bandwidth is often limited by the designers wish to obtain long delay times. What makes the series ideal for audio applications is their low insertion loss, low distortion and excellent signal to noise ratio and for the MN 3011 the specified values are 0dB, 0.4% and 76dB respectively.

Bucket brigade, or charged coupled, devices are analogue shift registers which operate by sampling the input signal at a rate determined by an external clock. The signal level at the time of sampling is stored on an internal capacitor. This charge is then clocked down a series of capacitors by means of internal switches and the transfer process is accomplished by a dual clock whose outputs are in antiphase and so are alternately opening and closing adjacent switches. It will be apparent that the slower the clock speed the longer the delay. Since the device operates at relatively high clocking speeds the input signal are faithfully reproduced at the output.

The functional block diagram and pin out of the MN 3011 is shown in Figure 2. The first thing to note is that it is a 12 pin IC but in a standard 18 pin DIL package. As is normal with such devices it requires two power supplies, V_{dd} and V_{gg}, and the former may be up to -18V while V_{gg} should be +1V higher than V_{dd}. The most interesting feature of the MN 3011 is that it has six tapped delays and the number of stages for each tapping is shown. The tappings are not evenly spaced since otherwise the reverberation effect would have a distinct flutter. If the device is clocked at 10kHz then the delays from the outputs 1 to 6 would be 19.8, 33.1, 59.7, 86.3, 139.5 and 166.4 milliseconds respectively. If these delays are multiplied by 0.33 then one obtains the equivalent room path length for one trip. Thus the longest delay is equivalent to a room length of 55 metres (181 feet). The delay time is not the same as the reverberation time

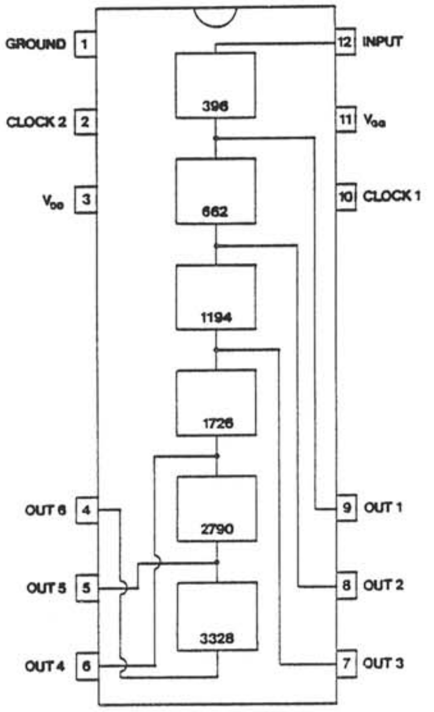


FIGURE 2. MN 3011 IC

since, as will be clear later, the delayed signal is recycled around the MN 3011 and produces a new set of delays. This is an ongoing process with new delays being recycled as older ones are decaying - the reverberation process! The 80-17 has a measured reverberation time of about 3 seconds, that is the time for a signal to decrease by 40dB with the delay set to maximum. This is not the usual definition of reverb. time which would give a longer time but we feel the signal reduction by 40dB is realistic.

The complete circuit diagram for the 80-17 module is shown in Figure 3. IC1A provides an input buffer with the output going to a peak voltage detector built around IC1B; to IC1C via the mix control, RV2, which allows a proportion of the untreated signal to be mixed with the treated signal; and to the MN 3011 via a filter section. The module will accept input signals with a minimum level of about 140mV rms, while higher signals are attenuated by RV1. In fact any signals found within the DIGISOUND 80 synthesiser may be used without damaging the MN 3011 but they must be attenuated to react to the LED peak indicator in the correct manner, as discussed in the Using section.

The signal input passes through two 12dB/octave filter sections, built around IC2A and IC2B, which have a cut-off frequency of 3.6kHz. The object of these filters is to remove high frequencies which may give rise to foldover distortion by creating low frequency cross products with the clock frequency. The dual op. amp. is operated from a single supply voltage and thus excessive inputs signals will simply be clipped at ground and V_{cc} and thus prevent damage to the MN 3011. Since BBD's are in practical terms operating from a single supply it is necessary to introduce a bias voltage so that bipolar signals will not be clipped. In this design the same applies to the filter IC's and thus the bias is introduced via the non inverting inputs using components R39, R40 and RV4. The filtered signal enters pin 12 of IC3, the MN 3011.

It was stated earlier that the BBD is a P-MOS device operating from a negative supply. Voltages are, however, relative to ground and so by connecting +15V to the ground pin of the MN 3011 and ground (0V) to the V_{dd} pin the effect is the same. This also allows the filter IC's to be operated from +15V. R1 and C5 connected between +15V line and IC3 prevent clocking pulses feeding back to the supply lines. The clock driver for IC3 is a purpose designed IC, a MN 3101, which in addition to providing the antiphase clock also has a V_{gg} output for the MN 3011. It is thus very cost effective and space saving. Note that it is also powered in the same way as the BBD, namely +15V to the ground pin and 0V to the V_{dd} pin. With the components shown connected to pins 5, 6 and 7 the potentiometer, RV3, allows the clock rate to be varied between 10kHz and 100kHz.

The six delay outputs from the MN 3011 are summed by the resistor network formed by R14 to R25 with the shorter delay being subject to less attenuation. From the longest delay, pin 4, the signal goes via R26 back to the input of the filter and thus provides recycling of the delayed signal in order to generate a true reverberation effect. This single return was found to be better than recycling of all delays which tend to make the sound rather 'muddy'. The reverberated signal is filtered by two more active filters constructed around IC4 and these have the same characteristics as the input filters. With both filter sections some passive filtering has been added to increase the roll-off and the signal loss in these filters is compensated by increasing the gain of the following active filters.

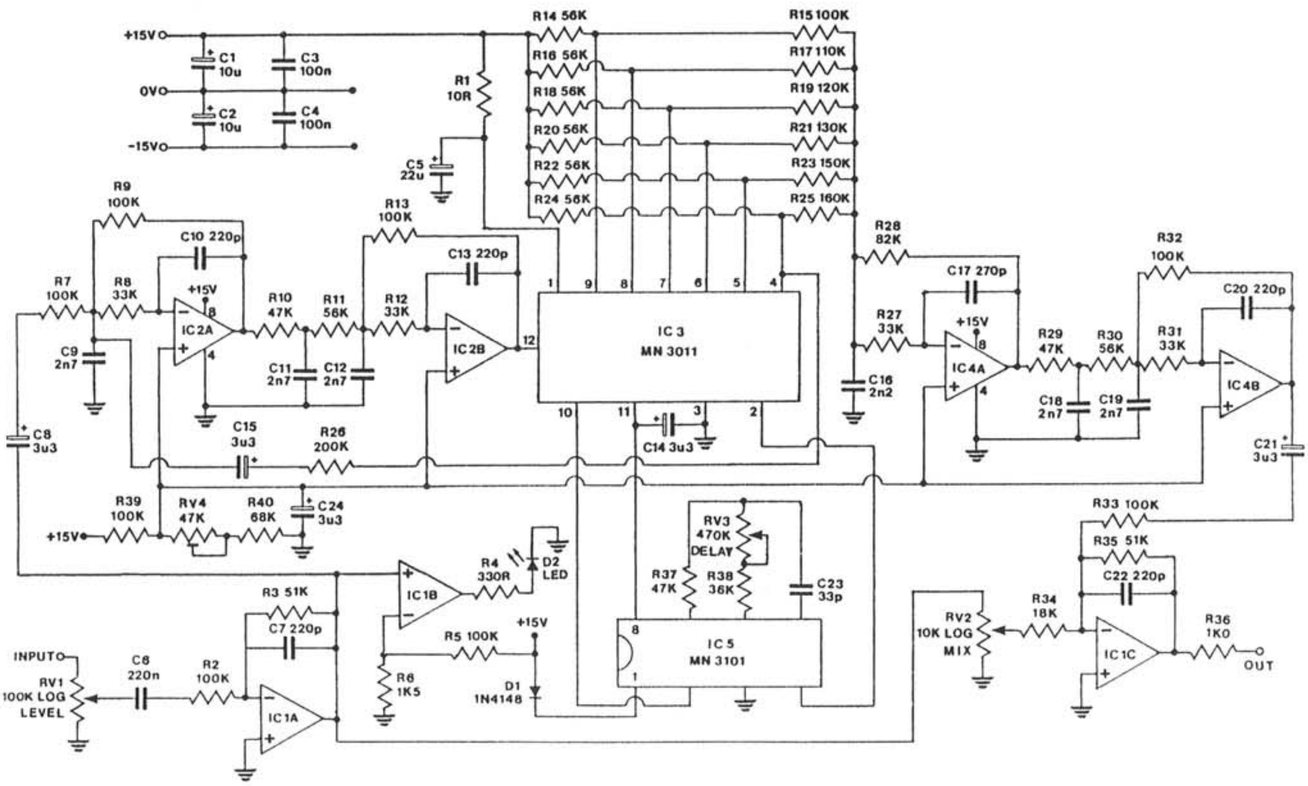


FIGURE 3. CIRCUIT FOR 80-17 REVERBERATION

The output from the reverberation section goes to IC1C where it may be mixed with a proportion of the original signal. Mixing of the untreated signal is not a dilution of the effect. As previously mentioned the reverberated signal has numerous signals which are delayed in phase with respect to the untreated signal. The effect of mixing is, therefore, to enhance and subtract harmonics from the untreated signal and so the whole signal, at least within the bandwidth of the filters, undergoes change. The main disadvantage of simple mixing is that the signal within the bandwidth of the filters is enhanced in comparison with the higher frequencies present in the original signal.

3. CONSTRUCTION

The 80-17 PCB is printed with a component overlay which simplifies the construction stage. The overlay is reproduced in Figure 4 to allow checking of component placement after the module has been constructed.

Clocked devices in general can be troublesome when brought into proximity with analogue circuits and various precautions have been taken to minimise such problems. The main one being to provide a ground plane on the component side of the PCB. This does not pose any real construction difficulties but the following points should be observed: (a) use a piece of insulated wire to make the one link on the PCB; (b) insert a piece of wire through the hole marked 'JOIN' and solder on both sides of the PCB so as to connect the 0V line to the ground plane; (c) use anti-wicking DIL sockets with mylar insulating strips - as provided in the kit; and (d) although the risk is small there may be a little insulation missing from the resistors, etc. which could short them to the ground plane. As a precaution insert a piece of card, such as postcard, between the PCB and component while the latter is being soldered in place. After soldering remove the card which will then leave a small gap.

Another special point is the handling of the MN 3011 and MN 3101 which are CMOS devices. With the advent of 'B' series CMOS we have all become careless on handling such IC's. For the MN types, however, take the precaution of working on a grounded metal surface, such as a piece of aluminium foil. Furthermore do not insert the IC with the power on or use a soldering iron on the PCB with the IC installed. These are all normal precautions and remember that even if your experience with CMOS has been a carefree one the damage caused by static may not prevent the IC working but it may impair its performance.

Special care should be taken regarding orientation of the electrolytic capacitors and the IC's. For the latter even when the DIL sockets have been installed the number '1', denoting pin 1, will still be visible on the PCB. The band on diode, D2, should face pin 1 of IC5, as indicated

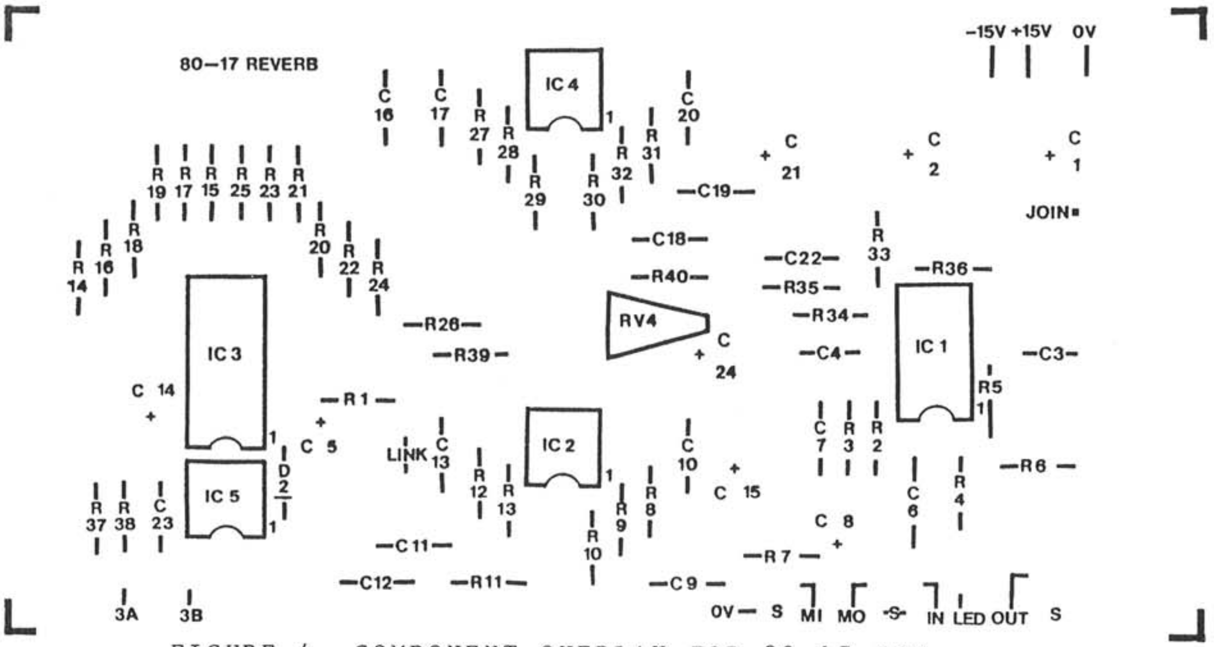


FIGURE 4. COMPONENT OVERLAY FOR 80-17 PCB

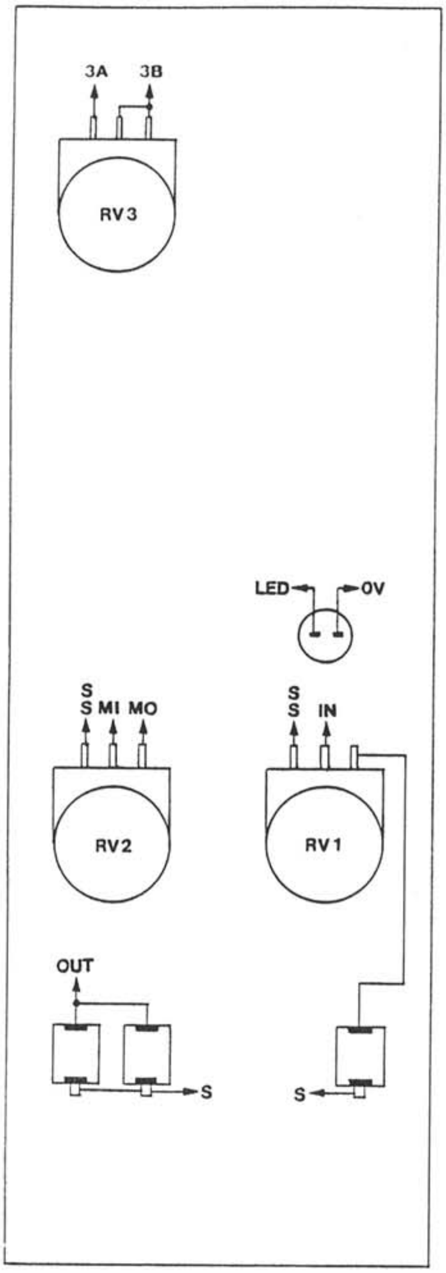


FIGURE 5. PANEL WIRING

by the band below the 'D2' on the overlay. For the LED, D1, ground (0V) is connected to the short lead. On completion of the construction then compare the component placement against that shown in Figure 4 and also check that no solder bridges have been formed on either side of the PCB. **Remember the MN 3011 is expensive and it will have been checked prior to despatch.**

The panel wiring diagram is shown in Figure 5 which illustrates the components when viewed from the rear of the panel. The arrows and associated numbers and letters indicate that a wire connection must be made from the position shown to the PCB which has the corresponding identification mark on its mounting edge. Care has been taken with the lay-out of the panel to keep the RV3 wires short and as far away as possible from the signal wires. It is, however, worth using miniature screened cable for the signal wiring and screen connections, marked 'S', are provided on the PCB adjacent to the appropriate hole for the signal wire. 'S' marked on the panel wiring diagram indicates the termination of the other end of the screen surrounding the wire.

The jack sockets shown in Figure 5 are the type supplied by Digisound Limited. The top connection, as illustrated, is the connection which is made with the jack plug when the latter is inserted. The lower connection is disabled by insertion of a jack plug. Finally, the tab under the socket is the grounded connection. In most modules we recommend the latter is connected to the 0V line but with the 80-17 this grounding will be made by soldering the screen to the socket and the 0V track on the PCB.

Finally, it has been found beneficial if the panel itself is grounded. The insulating material on the steel usually prevents the jack sockets from making a ground connection but the non slip washer for the potentiometers invariably cuts through the plastic. Gently clean a small area on RV1 or RV2 with emery paper or a file and solder a wire to this patch and connect the other end with one of the screen points or the 0V connection.

4. CALIBRATION

The only setting up required is to adjust RV4 to the optimum bias voltage. The most accurate method is to connect a swept sine wave to the 80-17 input and examine the output with an oscilloscope. With RV4 initially at mid position gradually increase the applied signal by clockwise rotation of RV1 until clipping or distortion of the top or bottom of the waveform is observed. Reduce the signal as necssary and adjust RV4 so that clipping or distortion is symmetrical. A less precise, although adequate, approach is to measure the voltage at the junction of R39 and RV4 and adjust the latter in the range of 6V3 to 6V5.

5. USING

The module may be located anywhere in the signal chain but usually it will be after a VCA; or after a mixer in a polyphonic system. Despite the precautions taken against radiated noise the physical location of the module may have to be selected such that it does not cause interference.

The main requirement during use of the DIGISOUND 80-17 module is that the input signal is maintained at a sufficient level to keep the signal to noise ratio acceptable. The LED peak indicator is arranged such that from the point where it is illuminated by the signal to the point where distortion of the signal commences is +6dB, that is a doubling of signal level. Under these circumstances RV1 should be set such that the LED is ON. This point may be quickly established and as stated above there is adequate headroom for peak excursions. Even with slightly more than a +6dB excursion it will only be an occasional peak that is distorted or clipped.

Generally, the 'mix' control, RV2, will be fully clockwise to ensure that higher frequencies in the original signal are not being filtered out. The amount of mixed signal is, however, a matter of personal preference and if a higher level is considered desirable then the value of R35 may be decreased.

With the 80-17 module the delay (reverberation) time may be varied and it is worth experimenting with this control on different types of sound since in some instances a shorter reverberation time can be very effective.

A switch may be incorporated into one of the 'out' holes on the panel in order to switch the effect in and out but this was not considered essential for a patchable system. Finally, another technique often employed with spring lines is to connect the output to a VCA so as to have voltage control of reverberation.

6. COMPONENTS

RESISTORS, 5%, 1/4w carbon film except where stated.

R1 (1/2w)	10R
R2,5,7,9,13,32,33,39	100k
R3,35	51k
R4	330R
R6	1k5
R8,12,27,31	33k
R10,29,37	47k
R11,30	56k
R26	200k
R28	82k
R34	18k
R36	1k0
R38	36k
R40	68k

RESISTORS, 1%, 1/4w metal film

R14,16,18,20,22,24	56k
R15	100k
R17	110k
R19	120k
R21	130k
R23	150k
R25	160k

CAPACITORS

C1,2	10u PCB electrolytic
C3,4	100n polyester
C5	22u PCB electrolytic
C6	220n polyester
C7,10,13,20,22	220p polystyrene
C8,14,15,21,24	3u3 PCB electrolytic
C9,11,12,18,19	2n7 polystyrene
C16	2n2 polystyrene
C17	270p polystyrene
C23	33p polystyrene

POTENTIOMETERS, TRIMMERS

RV1	100k log
RV2	10k log
RV3	470k lin
RV4	47k horizontal carbon

SEMICONDUCTORS

IC1	TL 084
IC2,4	LM 358
IC3	MN 3011
IC5	MN 3101
D1	5mm Red LED
D2	1N4148

digisound

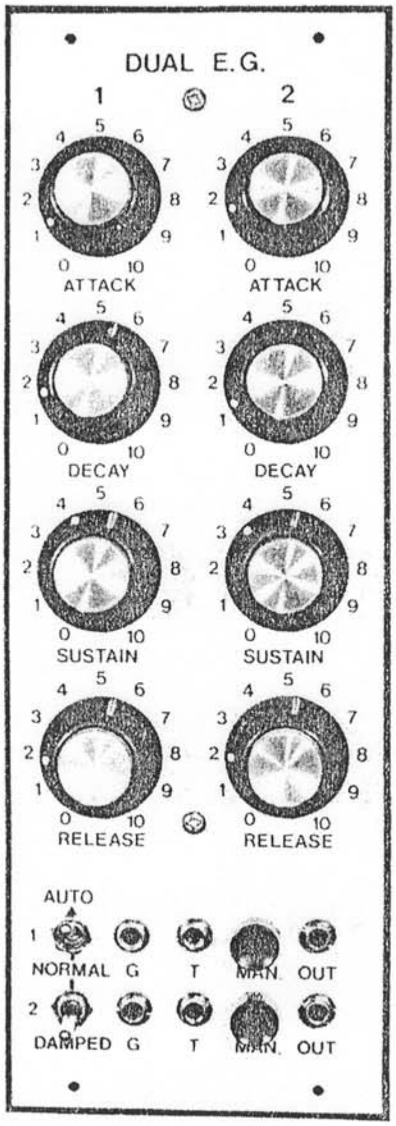
MODULE 80-18
MULTI-FUNCTION ENVELOPE GENERATOR

1. INTRODUCTION

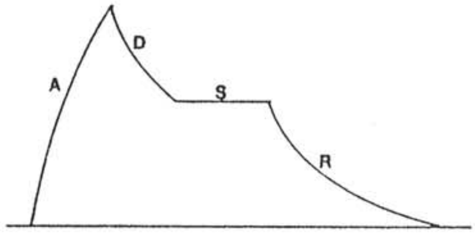
The DIGISOUND 80-18D Dual Envelope Generator consists of two independent generators constructed on the same PCB. Each envelope generator may be operated in one of three modes. These are:-

1. NORMAL. This is the conventional ADSR type of envelope, illustrated in Figure 2, in which the duration of the sustain period is determined by the presence of a gate voltage which in turn is equal to the period a key is depressed.

2. AUTOMATIC. In this mode a short pulse will cause the envelope to

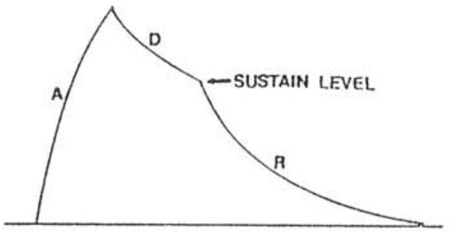


1. 80-18D PANEL



2. NORMAL ADSR ENVELOPE

cycle through a complete ADR envelope of the type illustrated in Figure 3. This mode is useful when the module is used in conjunction with programmable sound generators which normally only output a short pulse coincident with the start of a note. It will also be found useful by less skilled keyboard players since pressing a key momentarily will provide a complete envelope and one does not have to get the sustain period timing correct. It is also applicable to situations where long envelope times are set, since the user will have both hands free to manipulate the synthesiser while the contour is progressing through its cycle.

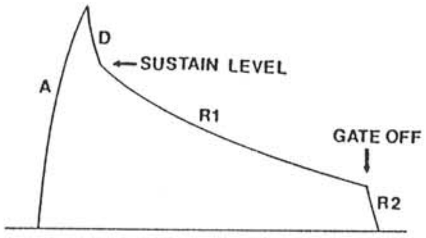


3. AUTOMATIC ENVELOPE

The AUTOMATIC mode is particularly beneficial when envelopes are being initiated from non-keyboard sources, for example, an LFO or the internal clock of the 80-12 Noise Generator/Sample & Hold module. A short pulse will now generate a complete ADR envelope and, by adjustment of the time constants, this type of envelope can be made to approximate the ADSR type, as is evident from Figure 2. Usually these external sources would only generate a limited AD type of envelope.

3. DAMPED. The objective of this mode is to more closely simulate the piano envelope which has a sharp attack, a brief initial decay, a long release and finally a very short release as the damper is applied to the string. This ADRR envelope is illustrated in Figure 4. In this

mode release of the key, which is the end of the gate pulse, causes the final release, R 2, to occur. In other words releasing the note has the same action as applying the damper on a piano.



4. DAMPED ENVELOPE

Other features of the 80-18 Envelope Generator are:-

i. Independent trigger input for re-triggering and generating multiple peak envelopes in the NORMAL and DAMPED modes.

ii. Gate and trigger pulses within the range of +3V to +15V are acceptable.

iii. Wide range of time constants. Typically 2 milliseconds to 20 seconds and a minimum of 10 seconds. If higher time constants are required then the upper limit is best extended by increasing the value of C 9.

iv. 0 to +10V peak attack output. This is user adjustable to other values if required.

v. 0 to 100% sustain level.

vi. The time constants may be trimmed to match other 80-18's in polyphonic applications.

vii. Low control voltage feedthrough which means low residual voltage when the envelope cycle is completed thus ensuring that the VCA is 'off'.

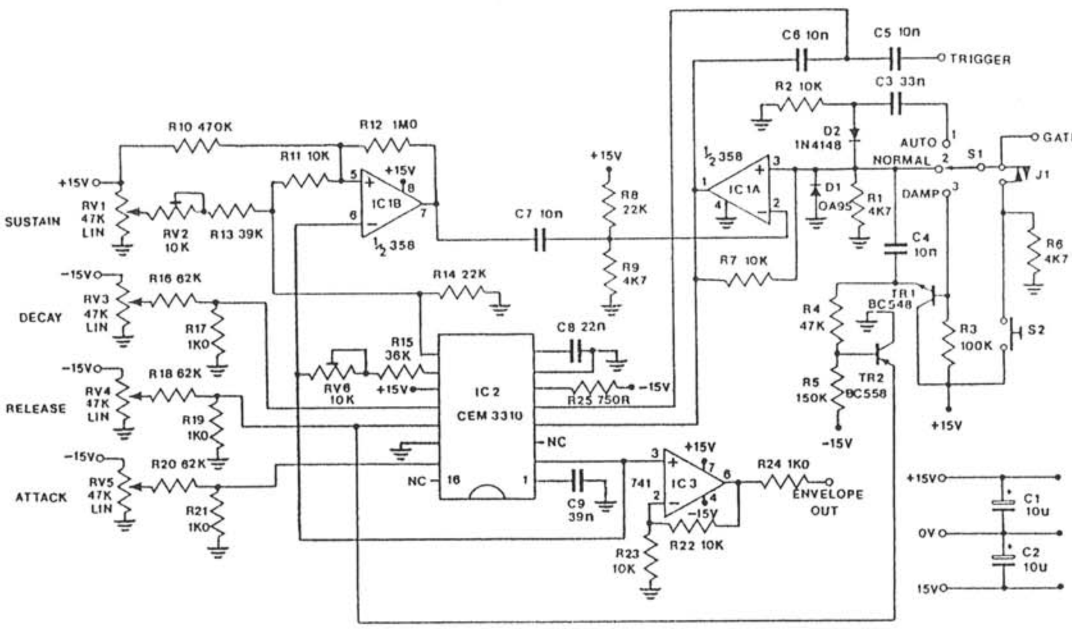
viii. Manual gating facility.

2. DESIGN

The 80-18 originates from a design by Doug Curtis, the president of Curtis Electromusic Specialties, and the heart of the module is the CEM 3310 Voltage Controlled Envelope Generator from this company. The CEM 3310 is also used in the versatile DIGISOUND 80-10 VCEG.

The complete circuit diagram for the 80-18 is shown in Figure 5 where it will be observed that the three operating modes are obtainable by simply switching S 1 to the appropriate position.

The main circuitry around IC 2, CEM 3310, is conventional and full details of using this device are to be found in the Data Sheet for this IC. RV 1, 3, 4 and 5 and associated resistive dividers are used to adjust the sustain level and the time constants for attack, decay and release. In the sustain line there is a trimmer, RV 2, which allows accurate matching of the peak attack voltage to the peak sustain voltage. The latter may be accomplished automatically with the CEM 3310, as in the 80-10 module, but to keep cost to a minimum a manual trimmer has been used in the present circuit. The



5. CIRCUIT DIAGRAM

time constants are also governed by C 9 plus the feedback components RV 6 (for matching units in a polyphonic system) and R15. With the components listed the A. D and R time values may be varied between two milliseconds and about twenty seconds or at least delete ten seconds. IC 3 is solely to convert the +5V output of the CEM 3310 to a +10V output which is compatible with the rest of the DIGISOUND 80. If a +5V output is adequate for use in another system then IC 3 may be by-passed, or omitted. Likewise the gain of IC 3 may be modified for other peak output voltages.

The unusual part of the circuit is built around IC 1, TR 1 and TR 2. In the NORMAL mode a gate pulse, in the range of approximately +3V to +15V, will simply switch the output of IC 1A high and this output is connected to the gate pin (pin 4) of the CEM 3310. A simultaneous trigger pulse will also be generated and applied to pin 5. The CEM 3310 requires both pulses in order to generate useful envelopes but the trigger pulse is readily obtained by differentiating the gate pulse.

If an independent trigger pulse is also available then this may be applied to the TRIGGER input, C 5, while the gate is high and so initiate a new attack cycle for generation of multiple peak envelopes.

In the AUTOMATIC mode IC 1A will again go high and produce the required gate and trigger pulses. Note, however, that IC 1A is acting as a set-reset flip-flop and in this mode the length of the gate pulse at the input, S 1, is of no consequence and is solely used to switch IC 1A high. The envelope then progresses through the attack and decay phases and when the latter is within about 100mV of the sustain level, as determined by R10 and R11 connected to IC 1B, the output of IC 1B goes high which will then reset IC 1A low and cause the envelope to go into its release phase. In other words when IC 1A goes low as the decay voltage matches the sustain voltage then as far as the CEM 3310 is concerned the action is the same as when the normal gate pulse is removed.

When S 1 is switched to the DAMPED mode then the gate pulse is applied to the base of TR 1, switching it on and producing a positive pulse to IC 1A which, as before, initiates the envelope. Again in this mode an independent trigger may be applied, if required. With IC 1A high the cycle will normally follow the same procedure as the automatic mode and it will be reset low by IC 1B when the decay more or less matches the sustain level which has been manually set by RV 1. When the gate pulse is removed then TR 2 will be turned on and since its emitter is connected to the release control of the CEM 3310 it will short out this input and cause the second release phase (denoted by R2 in Figure 4) to go to zero in several milliseconds. There are two points which should be apparent from this description. First if the gate pulse is removed earlier than indicated in Figure 4 then the cycle will end before the piano type envelope is obtained. Likewise if one allows the first release period to go to zero, or approaching same, then the damper action will obviously not be effective.

The circuit also includes a manual gating facility using the push button, S 2, and this facility is disabled by J1 when external signals are in use.

3.CONSTRUCTION

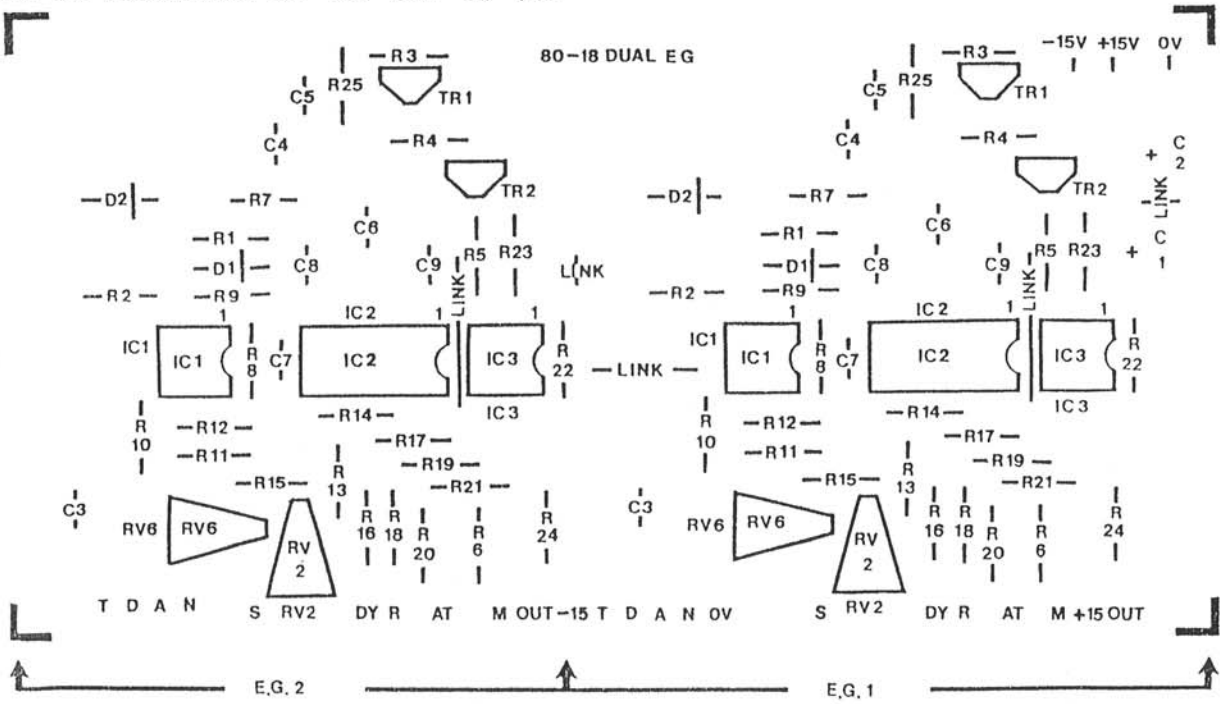
Construction is greatly simplified since the PCB has a component overlay printed on it. A reproduction of this overlay is shown in Figure 6 and may be used for reference when the PCB markings become obscured by the components. The main point to note is that the PCB accepts two envelope generators with the same component numbers for each. The two are, however, well separated and this separation is indicated in Figure 6. Take the usual precautions regarding orientation of semiconductors and the electrolytic capacitors. In the case of diodes a line is shown after the diode number and this line indicates the orientation, i.e. the band on the body of the diode should face the hole nearest the band on the PCB.

The panel wiring diagram is shown in Figure 7 and again the two envelope generators are clearly marked. Note the link on switch S 1 which converts it to a single pole 3 way type. The jack sockets illustrated represent the type supplied by Digisound and the top connection is the normal make contact while the bottom one is the connection which is broken on inserting a jack plug. The latter has been used on J1 to disable the manual gate when an external gate is in use. Finally, the white tab underneath the socket represents the ground terminal which should be connected to the OV line. Bare solid wire (1/0.6mm type) may be used for connecting all the ground tabs together but insulated wire should be used elsewhere Grounding the jack sockets allows the module to be readily connected to other equipment which is powered from a different supply. The arrows in Figure 7 indicate connections to the PCB and the markings correspond with those printed on the PCB.

After checking component orientation, absence of solder bridges and the wiring then the unit may be powered up from a $\pm$15V supply. The three operating modes may be checked using the manual gating button, S 2. There are two adjustments which may be made although the unit will function perfectly well if the two trimmers, RV 2 and RV 6, are set to their mid positions.

RV 2 allows matching of the peak sustain, RV 1 fully clockwise, to the peak attack voltage. That is to ensure that a sudden jump or drop does not occur at the end of the attack time when RV 1 is set to its maximum. The simplest way of adjusting this is to set the attack time to about 5 seconds and the sustain to maximum and observe the output of the envelope generator with an analogue voltmeter while keeping the manual push button depressed. Adjust RV 2 until there is no sudden change in voltage at the end of the attack cycle. Alternatively the 80-18 may be connected to an 80-2 VCO via Control Input 2 and by attenuating the input sufficiently a sudden change in tone will be discernible at the end of the attack period if peak attack and maximum sustain are not matched. Another alternative is to observe the envelope with an oscilloscope.

As inferred above, for normal use RV 6 may simply be left in its mid position. If, however, you wish to obtain the shortest possible time constants then turn RV 6 fully anti-clockwise. If the longest maximum times are of greater interest then conversely turn RV 6 fully clockwise. RV 6 is primarily for matching units in a polyphonic system and the most reliable method of matching is using a triggered oscilloscope. With this method the attack potentiometer, RV 5, should be set to its minimum and RV 6 adjusted to give an attack time of two milliseconds. An alternative method is to connect up the polyphonic system and gate the 80-18's simultaneously and then progressively match them in pairs by listening to the outputs from the appropriate VCA's. Again the matching is carried out using the attack time (all other potentiometers fully anti-clockwise) and it will be found best to set the attack pot. at some specific point on the dial. RV 6 is then adjusted until the peak output is reached simultaneously from a pair of 80-18's. One of this pair is then used to match a third unit and so on. This method is quite adequate since if one cannot discern a difference between units then the calibration is satisfactory.



6. PCB COMPONENT OVERLAY

COMPONENTS:

RESISTORS, $\frac{1}{4}$w, 5% carbon film

R1,6,9	4k7
R2,7,11,22,23	10k
R3	100k
R4	47k
R5	150k
R8,14	22k
R10	470k
R12	1M0
R13	39k
R15	36k
R16,18,20	62k
R17,19,21,24	1k0
R25	750R

POTENTIOMETERS

RV1,3,4,5	47k lin.
RV2,6	10k carbon preset

CAPACITORS

C1, 2	10u/35V PCB elect.
C3	33n MKH
C4,5,6,7	10n MKH
C8	22n MKH
C9	39n MKH

SEMICONDUCTORS

IC 1	LM 358
IC 2	CEM 3310
IC 3	741
TR 1	BC 548
TR 2	BC 558
D 1	OA 95
D 2	IN 4148

MISCELLANEOUS

S 1	1p3w sub-min. toggle
S 2	push to make

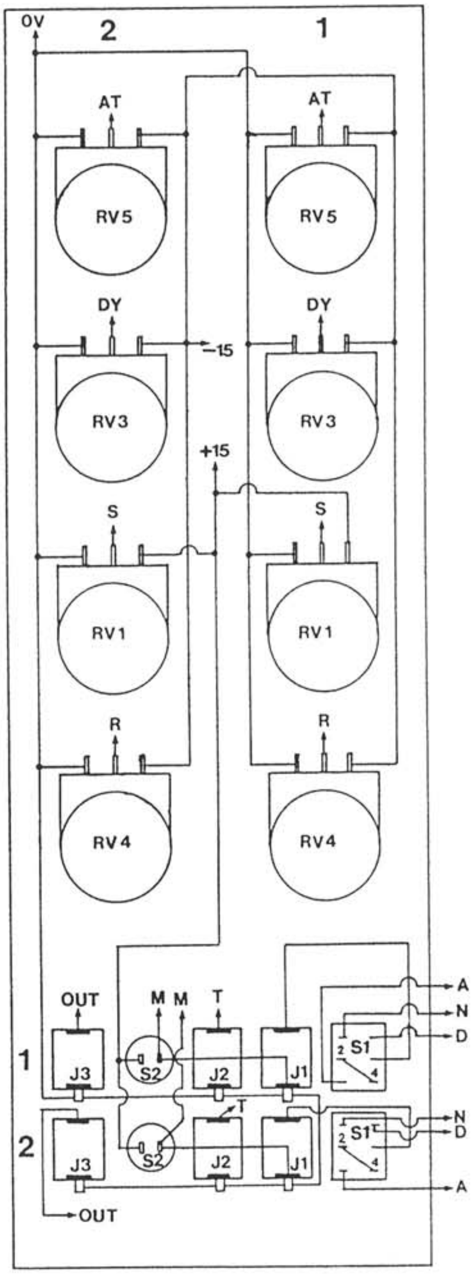


FIGURE 7. PANEL WIRING

4. USING

The DIGISOUND 80-18 may be used for all envelope shaping applications and one should refer to 'Using the Digisound 80 Modular Synthesiser' manual for comprehensive information of such applications. The additional AUTOMATIC and DAMPED modes will obviously be useful in specific patches and these should be obvious.

There are two points which should be noted regarding the 80-18:

i. In the AUTOMATIC mode the system may latch up if the sustain control is at or very near its maximum. To some degree this depends on the impedance of the gate source but in any event a latch up situation is quickly cured by switching to the NORMAL mode and back again.

ii. The input impedance of the gate

input is relatively low, typically 4k7 in the NORMAL and DAMPED modes and 10k in the AUTOMATIC mode. Thus simultaneous gating of more than two 80-18's from another module with a nominal output impedance of 1k0 may cause loading of this external module. This is not, however, a problem with the standard gate outputs from the 80-15D2 or from 'ALPHADAC 16' since these gate outputs are of near zero impedance. The loading problem referred to will simply reduce the amplitude of the output from the module used for gating, for example, with an 80-3 VCLFO the amplitude of the waveform will gradually be reduced, and possibly degraded, as more 80-18's are connected up. This may not affect the gating of the 80-18 but if the waveform is also being used elsewhere then one should be aware of the loading effect. This situation could have been avoided by installing a buffer on the gate input but it is unlikely that the low impedance of the module will cause any problems in practical situations.

Other features of the 80-18 concern its use with the ALPHADAC 16 operating in the arpeggiation modes. First, the NORMAL (ADSR) envelope must be selected. More important is the fact that the very short pulses generated in the staccato mode will not re-trigger the envelope generator. The 80-18A should therefore be used for the monophonic voice (split keyboard) when, or if, the staccato effect is required.

digisound

MODULE 80-18A
ADSR ENVELOPE GENERATOR

These construction notes must be read in conjunction with those for the DIGISOUND 80-18 Dual Multi-Function Envelope Generator.

1. MODIFICATIONS FOR 80-18A

The 80-18A provides the 'NORMAL' envelope function, that is, a standard ADSR envelope. Two envelope generators may be installed on a single PCB. The same panel and PCB is used for both the 80-18 and 80-18A and the latter is obtained by simplifying the input stage. This is illustrated by the circuit diagram of the 80-18 shown in Figure 1 and for the 'A' version nearly all of the components within the dotted area are omitted. The two components which remain are:-

i) R1 which becomes 10k

ii) D1 which gives fast protection against negative voltages, for example, if the module is being gated by a ±V pulse or other waveform.

To simplify subsequent construction you should now cross out, from the 80-18 list, those components not required for the 'A' version. Also amend the value of R1. Make these alterations in pencil since you may wish to convert the module to a multi-function type later.

2. CONSTRUCTION

For construction, in addition to omission of certain components, as described above, the other changes required are:-

i) Install a wire link on the PCB across the holes which would have been used for R7.

ii) The 'make' contact from the GATE jack socket is taken direct to the NORMAL (N) input on the PCB.

If the DIGISOUND panel is being used then it is suggested that a jack socket be installed in the hole prepared for the multi-function switch. This is solely to improve appearance.

Other construction and setting up procedures are the same for the 80-18.

3. USING

The impedance of the GATE input is 10k but otherwise most of the information listed for the NORMAL mode of operation of the 80-18 is still applicable. The exception being that the 80-18A will be re-triggered in the staccato mode of the ALPHADAC 16 arpeggiation routines.

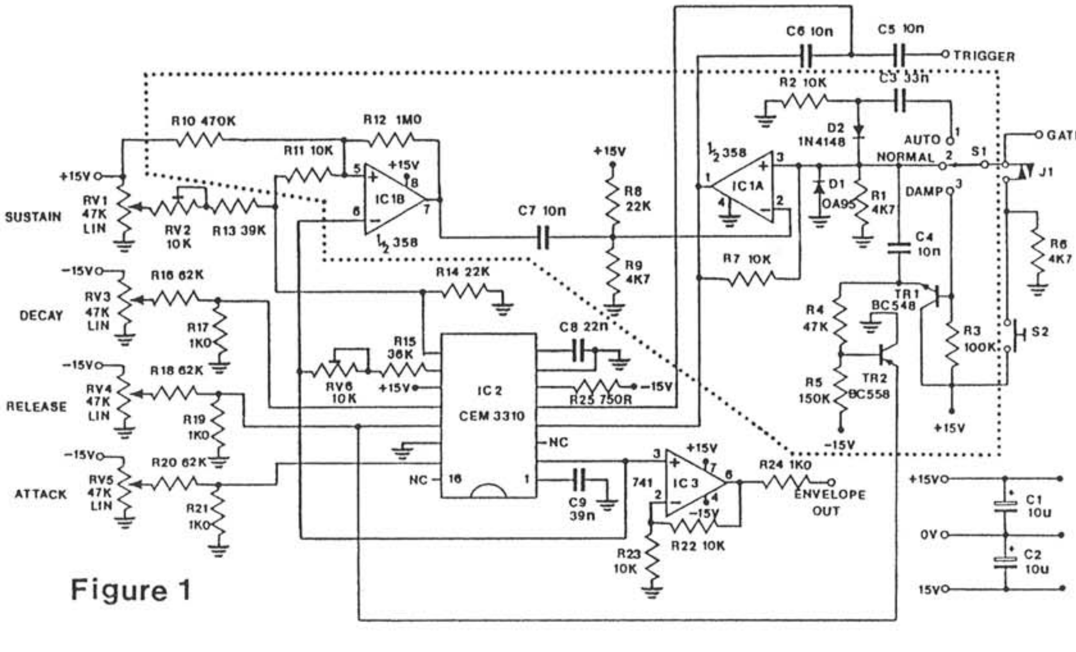
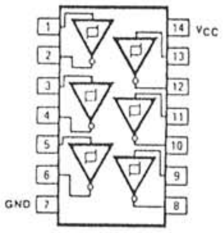
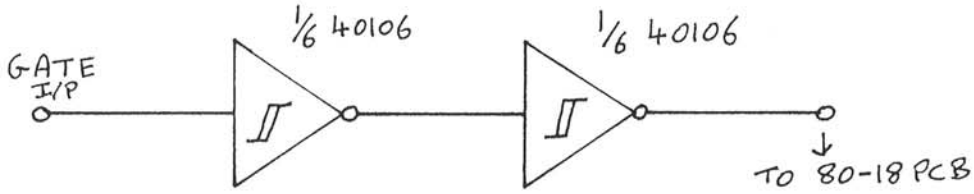


Figure 1

80-18 MODIFICATION - MAY 1985

Due to the lowered gate impedance produced by the additional circuitry of the expanded 80-18 module, some users have experienced gating problems (manifested by repeated Attack/Decay triggering), especially when the unit is gated from less than +15V. This problem may be overcome by the inclusion of a Schmitt trigger prior to the gate input as shown in the diagram below. Each gate input requires the use of two gates in series from a 40106 IC in order to remove inversion, so a dual unit will require four gates from a 40106. The IC may be mounted on a small piece of stripboard and power taken from the 80-18 Chiri connector. With this addition the module should gate perfectly with any signal in excess of +9V (CMOS 'on' threshold at +15V).

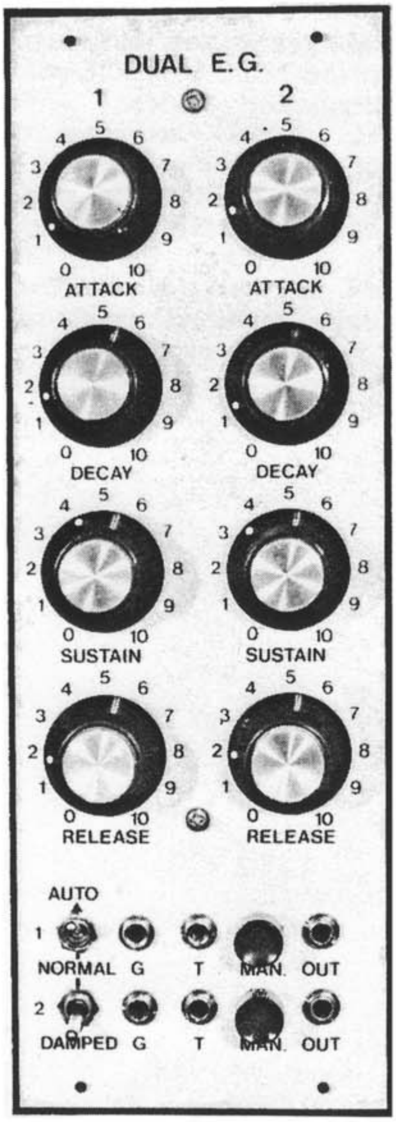




MODULE 80-18
MULTI-FUNCTION ENVELOPE GENERATOR

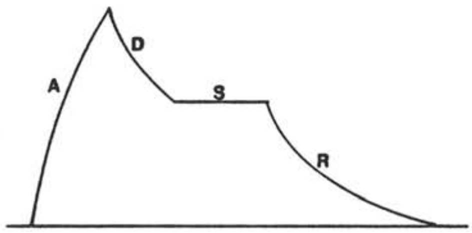
Few synthesists appear to recognise the value of envelope generators other than for the usual applications of obtaining a sound contour when used in conjunction with a VCA or modifying the timbre during the course of a note when connected to a control input of a VCF. It should be noted that the combination of envelope generator plus VCA is often referred to as an envelope shaper. Envelope generators are, however, one of the most useful sources of control voltages, particularly since a single gate pulse can initiate a complex pattern of control voltages. It is hoped that this manual will illustrate some of the more diverse applications.

In view of the above there is a need for relatively low cost envelope generators to encourage their widespread use. At the same time, however, quality is important and especially the ability of the envelope output to return to near zero at the end of its cycle, irrespective of the settings used. For this latter reason the DIGISOUND 80-18 and 80-18A modules have been introduced to replace the 80-8. The 80-18 is a more versatile unit and has three operating modes which are simply selected via a single pole three-way switch. The three modes are:-



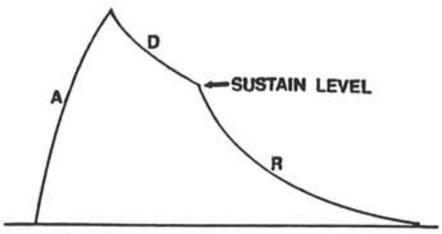
1. 80-18D PANEL

1. NORMAL. This is the conventional ADSR type of envelope, illustrated in Figure 2, in which the duration of the sustain period is determined by the presence of a gate voltage which in turn is equal to the period a key is depressed.



2. NORMAL ADSR ENVELOPE

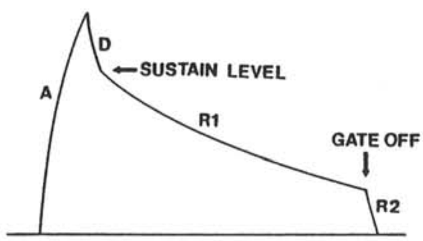
2. AUTOMATIC. In this mode a short pulse will cause the envelope to cycle through a complete ADR envelope of the type illustrated in Figure 3. This mode is useful when the module is used in conjunction with programmable sound generators which normally only output a short pulse coincident with the start of a note. It will also be found useful by less skilled keyboard players since pressing a key momentarily will provide a complete envelope and one does not have to get the sustain period timing correct. It is also applicable to situations where long envelope times are set, since the user will have both hands free to manipulate the synthesiser while the contour is progressing through its cycle.



3. AUTOMATIC ENVELOPE

The AUTOMATIC mode is particularly beneficial when envelopes are being initiated from non-keyboard sources, for example, an LFO or the internal clock of the 80-12 Noise Generator/Sample & Hold module. A short pulse will now generate a complete ADR envelope and, by adjustment of the time constants, this type of envelope can be made to approximate the ADSR type, as is evident from Figure 2. Usually these external sources would only generate a limited AD type of envelope.

3. DAMPED. The objective of this mode is to more closely simulate the piano envelope which has a sharp attack, a brief initial decay, a long release and finally a very short release as the damper is applied to the string. This ADRR envelope is illustrated in Figure 4. In this mode release of the key, which is the end of the gate pulse, causes the final release, R 2, to occur. In other words releasing the note has the same action as applying the damper on a piano.



4. DAMPED ENVELOPE

The three timing functions (A, D and R) have ranges between two milliseconds and ten seconds, or more, and are exponentially scaled. The latter results in the most useful time ranges utilising the highest proportion of the associated control potentiometers. The attack voltage rises to +10V and the sustain level is also adjustable from zero to +10V.

The envelope generator has separate gate and trigger inputs and their respective jack sockets are marked G and T on the panel. A trigger pulse is not required to initiate any of the envelope modes but in the NORMAL and DAMPED modes the application of an external trigger voltage while the gate pulse is still present will re-start the attack cycle and thus allow generation of multiple peaked envelopes. The module accepts ground referenced gate and trigger inputs within the range of +3V to +15V. The gate input is of low impedance and thus one should avoid gating more than two 80-18's from an external module, such as the 80-3 VCLFO, whose outputs have a nominal impedance of about 1k0. The low impedance is not a problem for the normal gating sources, i.e., the keyboard via 80-15D2 or the 'ALPHADAC 16' since the output impedance of these is near zero.

In the AUTOMATIC mode very high sustain levels, about 90% or more, may cause the ADR cycle to latch-up in some circumstances. What will happen is the output will stay at the maximum sustain level. If this does occur then simply switch to NORMAL, which will release the cycle, and then back to AUTOMATIC having reduced the sustain level slightly.

The time constants may be trimmed to enable accurate matching of units in a polyphonic system.

The module may be manually gated using the push button marked 'MAN' and this facility is disabled when external gate sources are connected to the 'G' socket.

Other features of the 80-18 concern its use with the ALPHADAC 16 operating in the arpeggiation modes. First, the NORMAL (ADSR) envelope must be selected. More important is the fact that the very short pulses generated in the staccato mode will not re-trigger the envelope generator. The 80-18A should therefore be used for the monophonic voice (split keyboard) when, or if, the staccato effect is required.

MODULE 80-18A
ADSR ENVELOPE GENERATOR

The DIGISOUND 80-18A is a direct replacement for the 80-8 module but with improved quality in terms of control voltage feedthrough. This improvement ensures that the envelope returns to near zero irrespective of the settings used for the ADSR envelope.

The PCB and panel are identical for the 80-18 and 80-18A modules and so the latter may subsequently be converted to a multi-function envelope generator, if required. A jack socket placed in the hole used for the function switch will improve the appearance of the module while in its simplified form.

The gate impedance is 10k but other characteristics are the same as the 80-18 operated in the NORMAL mode. Reference should therefore be made to the description of the latter module.

The 80-18A will be re-triggered in the staccato mode of the ALPHADAC 16 arpeggiation routines.

digisound

MODULE 80-19
DUAL V.C. LOW FREQUENCY OSCILLATOR

1. INTRODUCTION

The DIGISOUND 80-19 dual voltage controlled low frequency oscillator provides a wide range of features not normally available from conventional designs. First, a frequency range in excess of 50,000:1 and in the prototypes this was greater than 150Hz down to less than 1 cycle every 10 minutes. The actual range may be adjusted by changing one component per oscillator. The frequency of each VCLFO is adjusted using coarse and fine manual potentiometer controls and it may also be changed, or swept, by using an external control voltage. Three output waveforms are provided: triangle, sawtooth and pulse with the latter being adjustable from almost 0 to 100% duty cycle. The output waveforms are nominally 0 to +10V at full gain but each oscillator has a voltage controlled amplifier in its output path to control the gain, either manually or with an external control voltage.

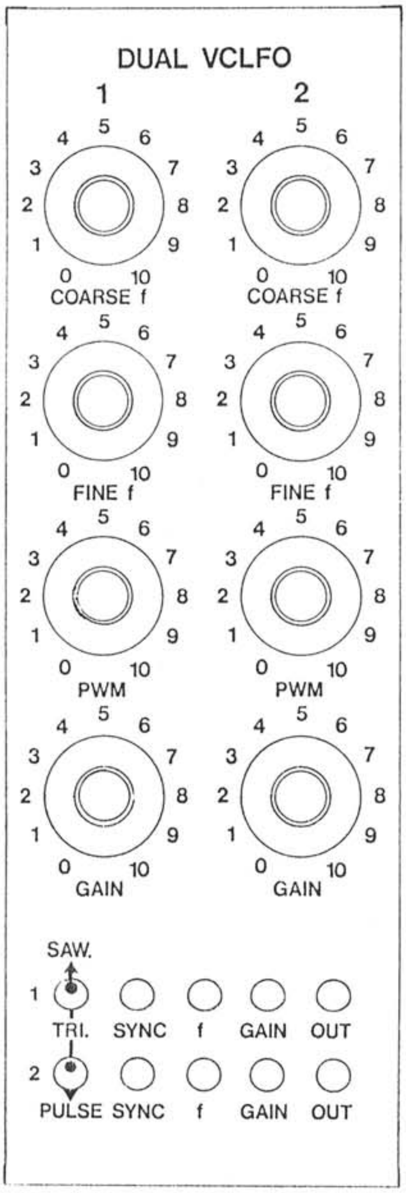


FIGURE 1. 80-19 PANEL LAY-OUT

The design utilises a CEM 3374 Dual Voltage Controlled Oscillator and a CEM 3360 Dual Voltage Controlled Amplifier, both from Curtis Electromusic Specialties Inc. The CEM3374 allows two modes of synchronisation. On one side a positive edge will cause the oscillator to reset and start from zero, which is valuable for a wide range of sweeping and synchronisation effects. On side 2 a positive edge will cause the waveform to reverse direction and may therefore be used to provide more complex modulating waveforms. These sync. features combined with voltage control of both frequency and gain allow many effects which have hitherto required several modules.

2. DESIGN

The complete circuit diagram for the 80-19 Dual VCLFO is shown in Figure 2. Apart from synchronisation both of the oscillators are identical and so the discussion will be confined to VCLFO 1 derived from pins 1 to 9 of the CEM 3374 (IC 2).

The CEM 3374 contains two completely independent precision voltage controlled oscillators. Each has an exponential control input (pins 7,12) as well as a linear control input at pins 8 and 11. Triangle (pins 5 and 14) and sawtooth (pins 2 and 17) waveforms are simultaneously available on chip and the timing capacitors are connected to pins 6 and 13. In order

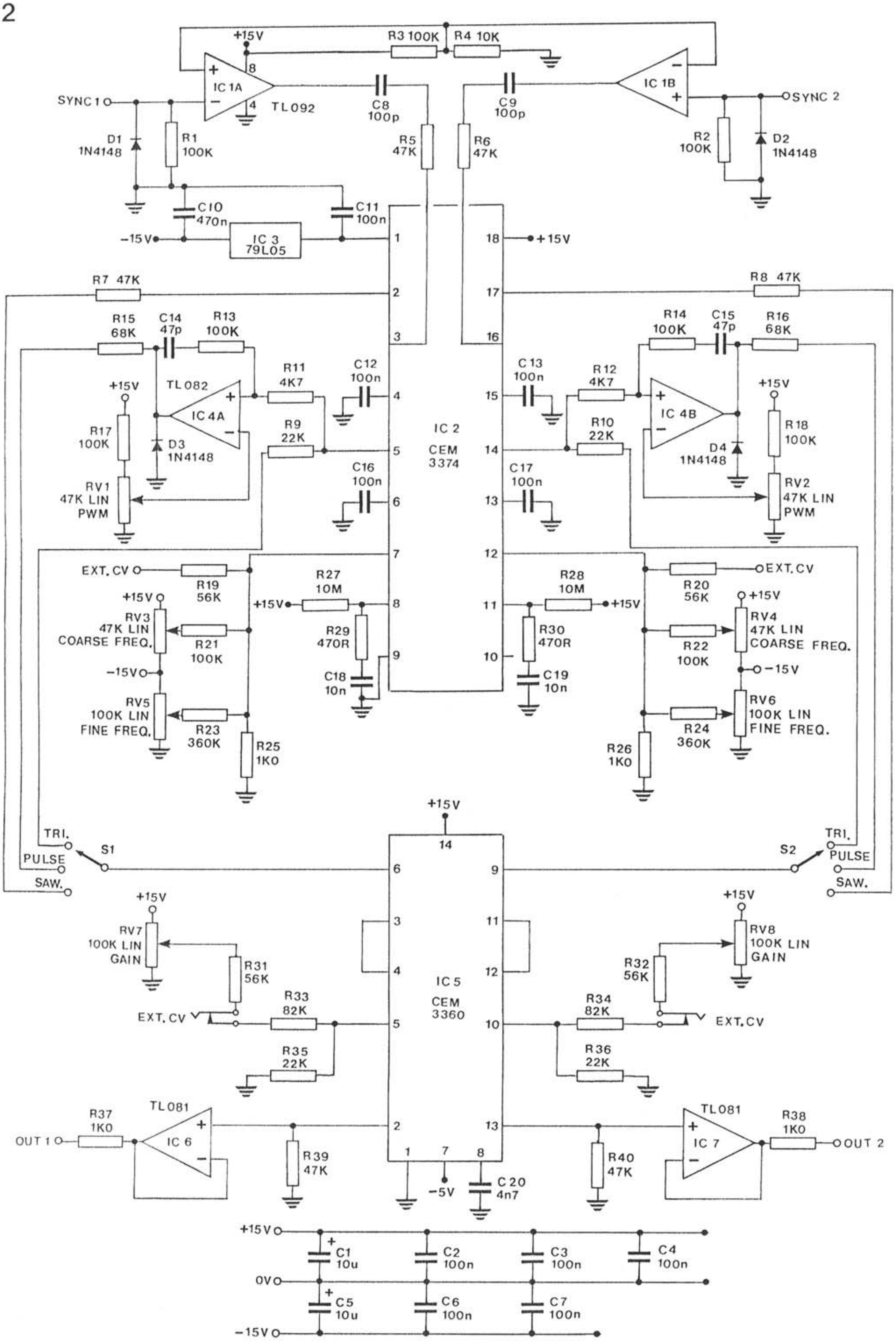


FIGURE 2. 80-19 DUAL VCLFO CIRCUIT DIAGRAM

to provide all of these features in a single IC the temperature compensating circuitry incorporaterd into the CEM 3340 VCO has been omitted. The CEM 3374 does, however, contain a temperature sensor which outputs from pin 10 a nominal 2V5 having a temperature coefficient of +3300 parts per million per degree Centigrade. This output voltage is commonly required as a reference voltage for many digital to analogue converters and so by using it as the reference and deriving the exponential control voltage from the same D to A one may obtain excellent oscillator stability.

The synchronisation input at pin 3 (VCLFO 1) allows the output to reset to zero while the sync at pin 16 causes the waveform to reverse direction (as the hard sync input on the CEM 3340 -please refer to the CEM 3340 data sheet, the 80-2 construction notes, or the DIGISOUND 80 Users manual).

The CEM operates with a positive supply in the range of +10 to +16 volts and a negative supply between -4V5 and -7 volts. To comply with these requirements a -5V supply to pin 1 is derived from the -15V supply using the regulator IC 3.

Turning now to the main circuit, then IC 1 is used for the external sync inputs which will respond to a positive going pulse from about +1V5 to +15V. The inputs are arranged so that synchronisation occurs on the positive edge of the pulse. C8/R5 (C9/R6) differentiate the output from IC 1 and also provide current limiting to the sync input pins on IC 2.

The exponential frequency control input at pin 7 has three inputs, viz., an external control voltage with a nominal 1V/octave scale via R19; a coarse control using RV3 and R21; and a fine adjustment using RV5 and R23. The polarity of the input is such that a positive input increases frequency. Two points should be observed. First, 5% tolerance resistors have been used throughout and so accurate scaling is not available and there is likely to be some variation betwen two VCLFO's. Accurate components were not considered worthwhile in the absence of temperature compensation or for typical LFO type applications. Secondly, the minimum specification for frequency sweep on the CEM 3374 is 50,000:1 and the design utilises this range (and more, if available). The result of the latter is that if both rotary controls are fully clockwise (highest frequency) and one then inputs a voltage to the external control input the output frequency is likely to increase but it may be far from the expected 1V/octave. Thus the external input should be kept within the designed frequency range of between 150Hz to 1 cycle per 10 minutes (for reasons already stated these values may vary somewhat between units). The timing capacitor C16 (C17) in the standard design is a 100nF polycarbonate and changing this component will alter the frequency range. A lower frequency limit (higher capacitor value) is not recommended since the total range may be reduced. Halving the capacitor value will double the frequency levels.

The linear input, pin 8, is used to inject a reference current into the IC via R27 and for this low frequency application the current is kept at a very low level. For a typical VCO application R27 would be 3M0 and C16 would be 1n0. Components R29, C12 and C18 are for compensation purposes.

The waveform outputs for the VCLFO's are selected by means of a switch - S1 for VCLFO 1 - and a current limiting resistor is selected in relation to the output level of the waveform. The 0 to +10V sawtooth goes to S1 via R7 while the 0 to +5V triangle goes via R9. The triangle output is also used to generate a pulse waveform using the comparator, IC4A. Varying the comparators reference with RV1 allows the pulse width to be varied from about 0% to almost 100% duty cycle. The output of the comparator ramps between about +/-13V and the negative portion is removed (limited to about -0V6) with diode D3. The pulse then goes to S1 via R15.

The waveforms are switched to a CEM 3360 Dual Voltage Controlled Amplifier (VCA). This IC is of the current-in, current-out type and has both exponential and linear control inputs. The linear control has been chosen in this design since the user will find it easier to judge output level if this is linearly proportional to the input. For example, the outputs are at a nominal level of +10 volts at full gain and so when RV7 is at half rotation, setting 5, the output will be about +5 volts. The manual control input, RV7, may be by-passed and a 0 to +10V external control voltage will adjust the gain over the full range. A useful feature is that (as designed) the CEM 3360 requires a few millivolts of control voltage, from RV7 or externally, before the output level begins to increase. Thus the VCLFO may be left connected to a VCO or other sensitive module without fear of modulation breakthrough when the gain control is set to zero. This superior performance in comparison with the more common VCA's has resulted in the widespread use of the CEM 3360 in synthesisers and other equipment. Another useful feature is that no adjustments are required for most applications. The output of the VCA is converted to a voltage by R39 and buffered to provide a low impedance output by IC 6.

3. CONSTRUCTION

The PCB component lay-out is shown in Figure 3 and is quite straigtforward and unambiguous. The best approach to construction is to install components in order of increasing height, e.g., the four wire links followed by diodes, resistors, DIL sockets and so on. The main points to watch at this stage are the orientaion of the IC's, diodes, the electrolytic capacitors C1 and C5 and the -5V regulator but all of these are clearly shown in Figure 3. One other point to observe is the location for C8 and C9 which have holes to accommodate other sizes of capacitors. Ensure that C8 and C9 are installed such that they bridge over the gap, i.e., use the holes nearest IC 1.

In comparison with many other modules there is quite a lot of wiring and the connections are shown in Figure 4. The letters, etc., shown in this diagram indicate that a wire must be taken from that point to the PCB having the same identification mark -

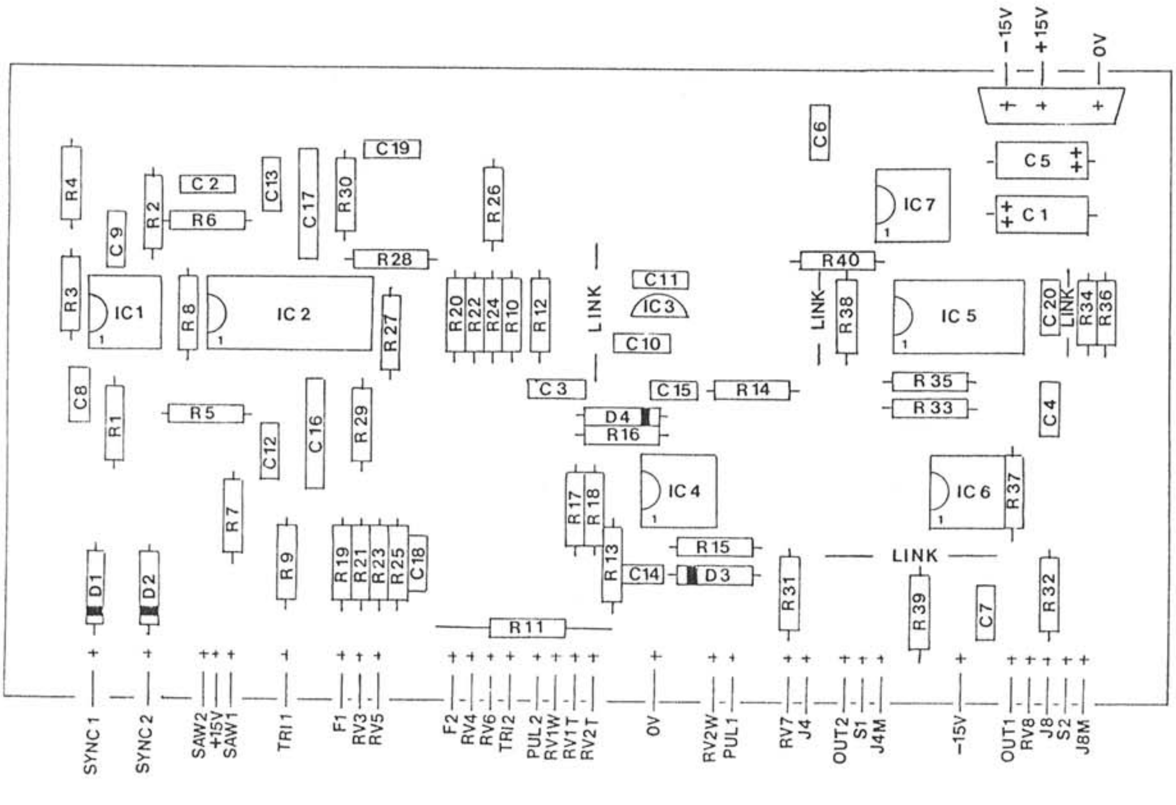


FIGURE 3. 80-19 PCB COMPONENT LAY-OUT

refer to Figure 3 for the latter. Thus 'RV1W' shown on RV1 in Figure 4 is wired to 'RV1W' on the PCB as shown in Figure 3. Keep the wires neat and as short as possible and 1/0.6mm wire is recommended since it enables the wire to be neatly moulded to shape. The use of several colours will assist with subsequent checking.

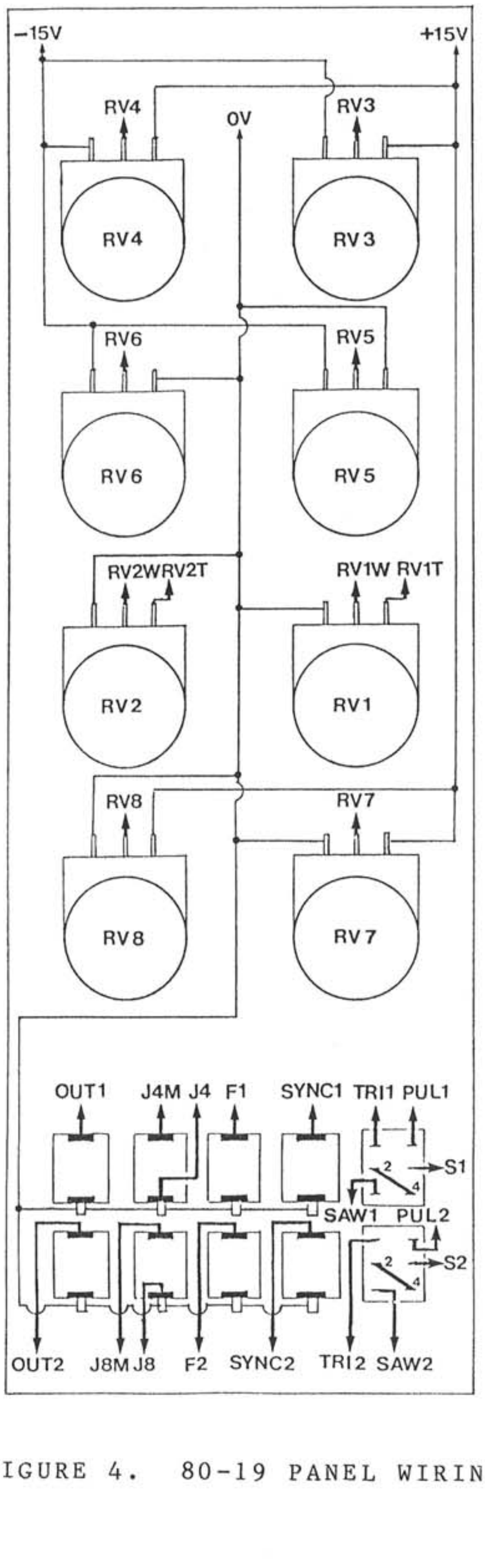


FIGURE 4. 80-19 PANEL WIRING

The jack sockets illustrated in Figure 4 are the type supplied by Digisound Limited. The top connecting tab is the connection made when a jack plug is inserted while the lower connection is broken on insertion of a plug. There is also a ground tab underneath the socket and we recommend these are connected to the 0V line so that the module may be used with equipment which may be separately powered.

On the switches ensure that a wire link is installed between pins 2 and 4 and keep to the orientation shown. After assembly has been completed then carefully check the wiring as well as the component placement and orientation on the PCB. Also carefully examine the foil side of the PCB to check for solder splashes and bridges which may be shorting tracks together. Additionally keep the underside of the PCB as clean as possible, especially around C16 and C17, since the capacitor charging and discharging currents are exceedingly low and will be affected by solder flux and other extraneous matter. These will not affect the operation of the VCLFO's but may well influence the lowest frequency attainable.

After careful checking - some parts are expensive! - the unit may be powered up and put into operation since there are no adjustments to make.

4. USING

Because the 80-19 is voltage controlled it will substitute in most of the applications shown in 'Using The Digisound 80 Modular Synthesiser'. In fact, in some applications the built-in atttenuators (voltage controlled amplifiers) will simplify many patches. The additional uses of the synchronisation inputs will be obvious to the user.

5. COMPONENTS

RESISTORS, 5%, 1/4w carbon film

R1,2,3,13,14,17,18,21,22	100k
R4	10k
R5,6,7,8,39,40	47k
R9,10,35,36	22k
R11,12	4k7
R15,16	68k
R19,20,31,32	56k
R23,24	360k
R25,26,37,38	1k0
R27,28	10M0
R29,30	470R
R33,34	82k

CAPACITORS

C1,5	10u axial electrolytic
C2,3,4,6,7,11,12,13	100n polyester
C8,9	100p ceramic
C10	470n polyester
C14,15	47p ceramic
C16,17	100n polycarbonate
C18,19	10n polyester
C20	4n7 polyester

POTENTIOMETERS, SWITCHES

RV1,2,3,4	47k lin. rotary
RV5,6,7,8	100k lin. rotary
S1,2	1p3w sub. min. toggle

SEMICONDUCTORS

IC1	TL 092
IC2	CEM 3374
IC3	79L05
IC4	TL 082
IC5	CEM 3360
IC6,7	TL 081
D1,2,3,4	IN4148

80-19 MODIFICATION - MAY 1985

Following a suggestion by Mr. R. King, we can now advise of a small modification to reduce current consumption and thus remove unwanted modulation present on the PSU rails when using the 80-19 Dual VCLFO module. The modification involves removal of the feedback components on both halves of IC 4 (R13/C14 and R14/C15) and repositioning the diodes D3 and D4 as shown in the modified circuit diagram and overlay below. The series combination of D3/R15 (D4/R16) should occupy the two holes previously used for R15 (R16). The modified unit should still function exactly as before, but without IC 4 running hot.

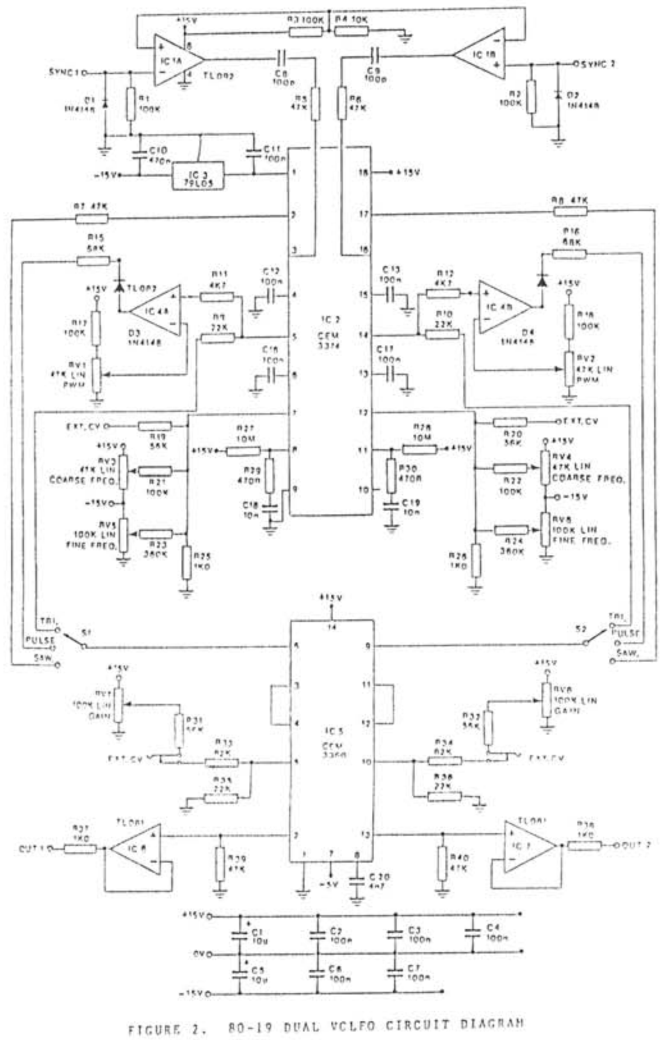


FIGURE 2. 80-19 DUAL VCLFO CIRCUIT DIAGRAM

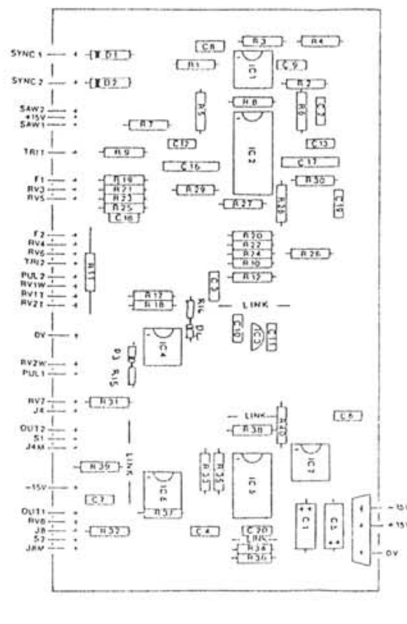


FIGURE 3. 80-19 PCB COMPONENT LAY-OUT

WAVEFORM MULTIPLIER

A single VCO on a synth is, to be honest, pretty boring. Generate rich multiple oscillator sounds by hooking it up to one of our multiplier boards. Design by David Ward-Hunt.

Many synthesisers, both mono and polyphonic, utilise two or more VCOs, slightly detuned, to generate a rich chorusing sound. Chorus or phasing treatment of a single VCO can go some way towards livening up the sound, but these tend to suffer from a rather repetitive sweep and on some units a considerable amount of background noise during periods of silence — not to mention aliasing when used with high frequency high harmonic content waveforms which any decent synthesiser is capable of producing.

The beauty of using two or more slightly detuned oscillators is that in addition to producing a full chorusing sound, the problems of background noise and bandwidth are eliminated. However, multiple VCOs don't come cheap! An alternative method of achieving a 'multiple oscillator sound' is to generate additional waveforms from the existing VCO output. If each of these 'new' waveforms is out of phase with the original and with each other, then a fuller sound will be heard. However, the richness of the sound from multiple oscillators comes not from the fact that they are out of phase with each other, but from the fact that the phase difference is continually changing — the ear perceives phase *change* rather than phase difference. Therefore it is necessary not only to have additional out-of-phase waveforms, but their phase differences should be continually moving with respect to each other and the original.

A Passing Phase

The circuit described here does just that. It will accept sawtooth, triangle or sine wave inputs, though with the latter two the output will bear little resemblance to the original waveform due to the circuit action: however, they are still useful sounds to experiment with. The circuit has been used successfully to treat the VCO sawtooth outputs from a number of synthesisers including the Transcendent, Digisound '80 and PE Minisonic; it has also been used with a Korg Sigma and Roland SH02 (see the interfacing notes below). The one disadvantage of the circuit (there has to be one, doesn't there?) is that for setting-up purposes, constructors will need access to a scope or a second VCO with which to adjust the circuit to produce the correct waveforms.

The picture shows how the prototype was mounted in a Teko Alba A23 case, but this is not essential and most people will build the boards into their synth.

Using The Multiplier

The multiplier board is fed with the output from your existing VCO. With a sawtooth waveform fed to the circuit, the output is a series of six sawtooth waveforms each individually phase modulated and mixed with the original sawtooth from the VCO. One multiplier PCB (generating six 'new' waveforms) is used with each VCO. If you do have two or more VCOs each feeding a separate multiplier board the effect is outstanding, especially when the VCOs are tuned to form a chord. The output from the multiplier(s) is then fed back to the synthesiser and treated by the VCF and VCA in the normal way.

Construction

The project consists of two PCBs. The first holds the modulation oscillators for phase modulating the multiplier; the second PCB holds the multipliers and associated circuitry. The reason for splitting the project into two PCBs is that one modulation PCB is sufficient to drive up to four multiplier PCBs. (In fact there is no reason why it wouldn't drive more; however, we believe that if you intend to use more than four multiplier boards, the small additional expense of another modulator is well worth it for adding an even richer sound.)

All the components are mounted on the two PCBs with the exception of two diodes which are mounted on a switch (thereby saving two wires from the PCB to the switch). The only external connections required are the VCO input and the output from the unit plus the power supply connections (see below) which ideally should

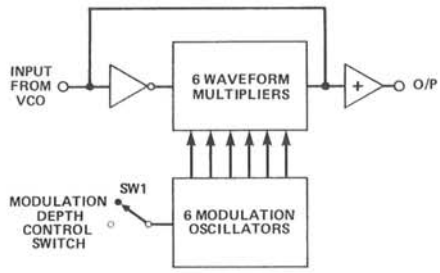


Fig. 1 Block diagram of the ETI Waveform Multiplier. Should you require an even richer sound, there's no reason why more than six multipliers shouldn't be used.

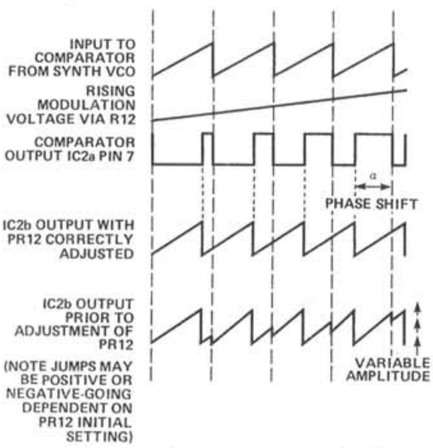


Fig. 2 The waveforms associated with various sections of the circuit. This indicates the operation of the unit and will also guide those people who are setting up the unit with an oscilloscope.

come from the same power supply as the VCO being input to the unit.

Interwiring between the PCBs should be clear from the diagram, as should the wiring up of the switch with its associated zener diodes. The switch is a DPDT with centre off and it is essential (for setting-up purposes if nothing else) to have this 'off' position (see Buylines).

A note is in order here about the component numbering. In order to make the numbering clearer and logically the same for each of the six multipliers on the PCB, each resistor or preset is designated by a two-figure number. The first figure indicates which of the six multipliers it is associated with, and the second is the 'relative number' of the component. For example, R11 is 'R1' on the first multiplier, R21 is 'R1' on the second multiplier, R35 is 'R5' on the third multiplier and so on.

With regard to the modulation PCB, though all the capacitors are of the same value, take care not to mix up the resistors associated with the two oscillators, as each oscillator should have all three of its resistors the same value. If not, the modulation output waveform will take on a pulse form at the output associated with the first op-amp in each oscillator.

PARTS LIST

MODULATION BOARD

Resistors (all $\frac{1}{4}$W, 5%)	
R1–3	1M8
R4–6	2M2
Capacitors	
C1–8	100n polyester
Semiconductors	
IC1	LM324
IC2	LM1458
ZD1,2	6V2 400 mW zener
Miscellaneous	
SW1	DPDT with centre off (see Buylines)
PCB	(see Buylines)

MULTIPLIER BOARD

Resistors (all $\frac{1}{4}$W, 5%)	
R1–3, 11,12,13,14, 15,16,21,22, etc to 66	100k (39 in total)
R4	18k
Potentiometers	
PR11–61	100k miniature horizontal preset (six in total)
PR12–62	220k miniature horizontal preset (six in total)
Capacitors	
C1,2	47u 16 V PCB electrolytic
C3–6	100n polyester
Semiconductors	
IC1	LM1458
IC2–4	LM324
D1–6	1N4148

Miscellaneous

PCB (see Buylines); case to suit (Teko Alba A23); sockets to suit.

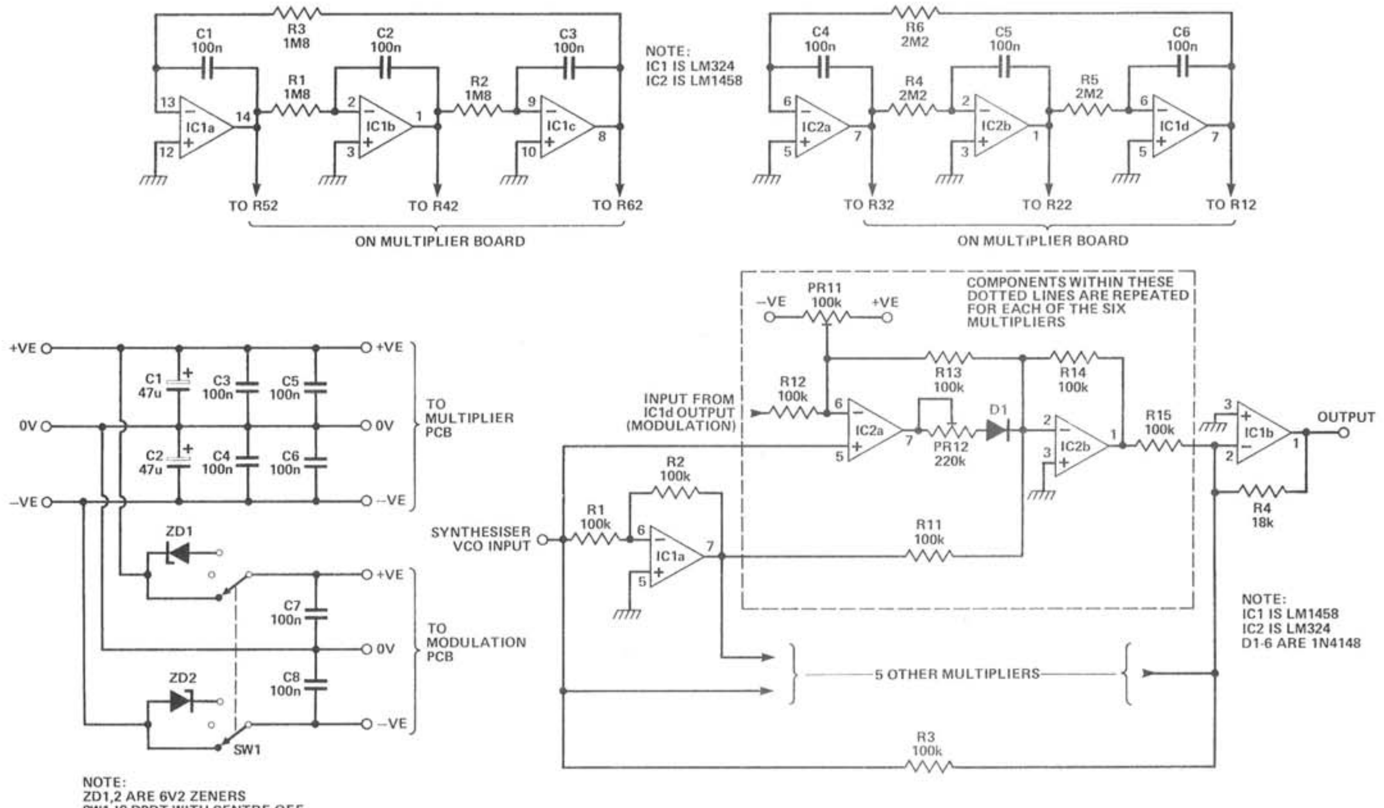


Fig. 3 Complete circuit diagram of the Waveform Multiplier.

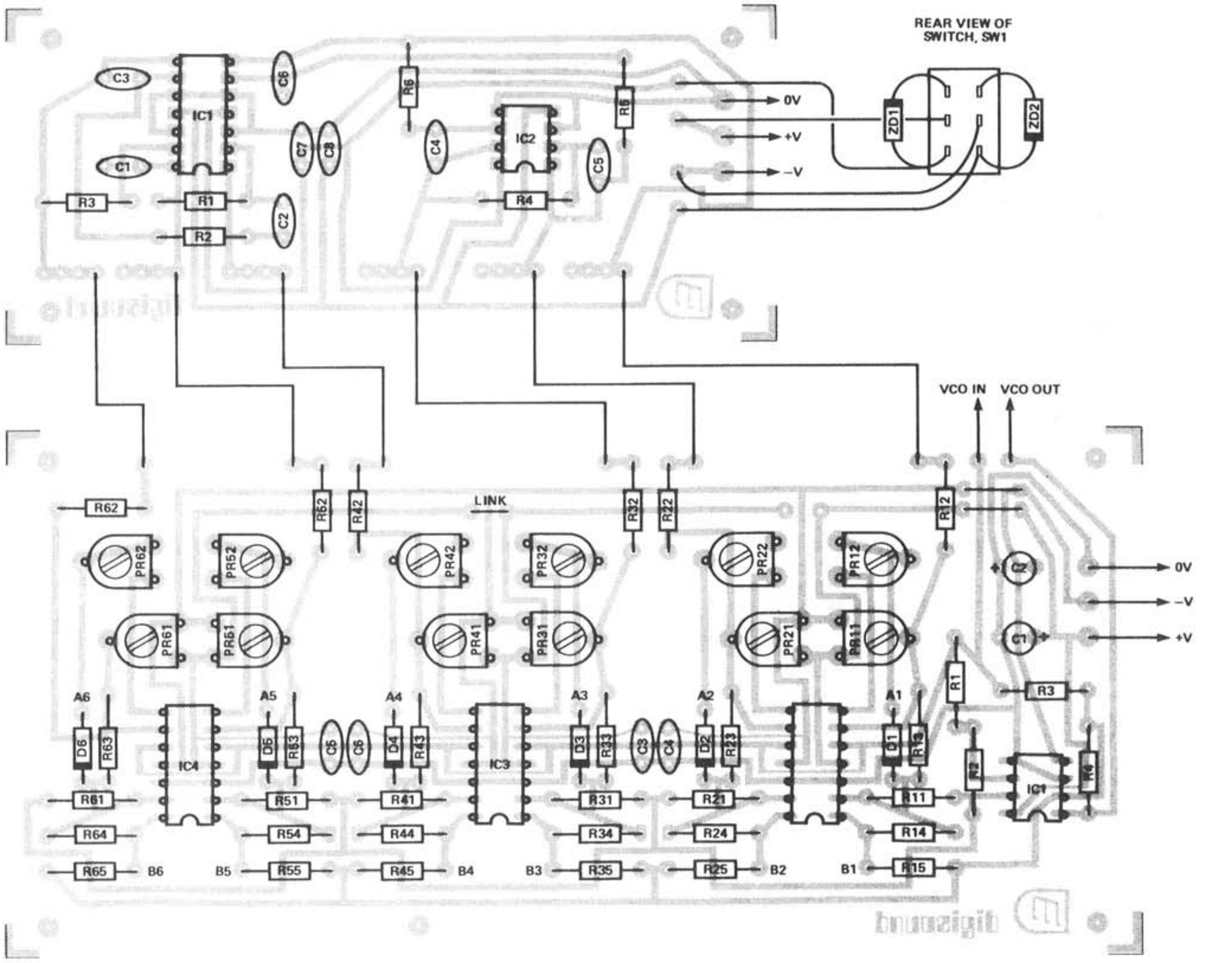


Fig. 4 Component overlays for the modulator board (top) and the multiplier board (bottom), and the interwiring required. Several multiplier boards may be driven from one modulator board.

HOW IT WORKS

The modulation oscillators are based on a standard three phase oscillator; two of these are used. Each of the three integrators in the loop outputs a waveform that is one-third of a cycle behind the others. The speed of the two oscillators is set at about 0.6 Hz and 0.4 Hz respectively. This modulation rate was found to give the best simulation of a number of oscillators running in free phase over a wide keyboard span.

It should be noted that the output of these oscillators is, in fact, a trapezoid shape waveform and it might be thought that some filtering would be required to produce a waveform more akin to a sine wave. This was tried in the development stage, but in practice the trapezoid waveform gave a better randomness to the overall output, whereas with a sine wave modulation a more definite sweep could be detected on long sustained notes.

Referring now to the multiplier circuit, the output from the synthesiser VCO is taken to IC1a configured as an inverter/buffer; the VCO waveform also goes to one input of the comparator IC2a. The other input of the comparator is fed with a voltage set up on PR11 together with the modulation voltage via R12. With a positive-going sawtooth, the point at which the comparator goes high is determined by the sum of the fixed voltage from PR11 and the modulating voltage via R12. As shown in the waveform diagram, with a rising modulation voltage the width of the pulse at the comparator's output increases. However, the comparator will, of course, always reset at the same moment as the VCO sawtooth. Thus the comparator's reset is synchronised with the original sawtooth, whereas its positive-going excursion can be voltage-controlled to any point within one cycle of the input waveform.

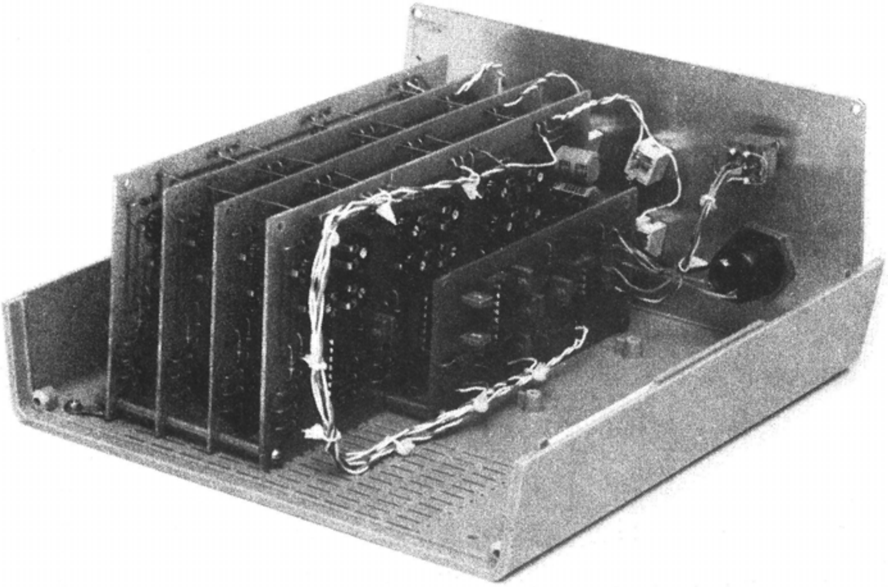
The output of the comparator is rectified by D1 and summed with the inverted sawtooth from IC1a via R11. These voltages (actually currents) result in a new sawtooth whose reset point is determined by the positive-going edge of the comparator. Thus, as shown in the waveform diagram, this new sawtooth is phase-shifted from the original and the amount of phase shift is dependent on the comparator pulse width. The output of the summing amplifier IC2b is taken via R15 and mixed with the other multiplier outputs plus the original sawtooth (via R3), where IC1b acts as the mixing amplifier.

Setting Up

As stated earlier in the article, setting up requires access to an oscilloscope; you should have two VCOs and some method of listening to their output, but hopefully these requirements will not be a problem for most constructors.

Once all the components have been installed, ensure that the ICs, diodes and the two polarised capacitors are correctly orientated. All the ICs should be facing the same way and the diodes mounted on the PCB should have their band pointing towards pin 1/pin 14 of the ICs. Turn all the presets to their mid-position.

The setting up procedure is reasonably straightforward, though repetitive, since each of the six multipliers has two presets which need to be adjusted; however, your efforts will be rewarded! Constructors with access to a scope will find it easier to use the scope to see the various waveforms mentioned in the text. Those of you



A view inside the prototype showing how it is configured to provide four independent channels driven by one modulator board. Miniature jack sockets are used for input and output (see also the lead photograph).

without a scope will be using your ears rather than your eyes, so make sure you have some means of listening handy. Either an amplifier and speakers, or headphones of reasonable quality will do, but remember we are listening to a 'raw' signal so don't stick it straight into your favourite hi-fi amp!

The first step is to power up the circuit and apply a VCO sawtooth waveform running at about 500 Hz (for ease of listening — so with a scope it doesn't matter). Ensure that the modulation switch is in the off position. Attach a lead to the output of IC1b (junction of pin 7 and R2). This should be outputting the VCO sawtooth (inverted). Next move the scope probe or audio lead to the junction of D1 and PR12 (marked A on the component overlay). This is the comparator output from which we want to get an approximate square wave output. At first you may hear no sound at all, but as PR11 is rotated you should hear a square wave break in with the pulse width varying as you turn the preset. Adjust PR11 for an approximate 50% duty cycle. (For those unfamiliar with this sound, adjust PR11 for the 'loudest' sound. When the duty cycle is less than about 30% or more than about 70% a 'thin' spiky sound predominates, so set PR11 midway between these two — an accurate 50% is not important.) Repeat this step on each of the five other multipliers.

With the comparators now set up, we need to adjust the presets on the comparator outputs so as to give a smooth sawtooth waveform. PR12 is the preset associated with this adjustment. Attach your lead to the output of IC2b (junction of pin 1 and R15 — point B on the overlay). Until PR12 is correctly adjusted the sawtooth will contain a jump in it as shown in the waveform diagram. Using a scope, all you need to do is turn PR12 until this jump is smoothed out. Without a scope you will need two VCOs in order to accomplish this adjustment.

First, slow down the VCO which is connected to the circuit, to its lowest setting (somewhere around one cycle per second is fine). Now attach a lead from the aforementioned junction of pin 1 and R15 (point B) and run it to the FM input of your second VCO. If you now listen to the output of this second VCO it will be frequency-modulated by the sawtooth wave that we need to smooth out. So all you need to do now is turn PR12 until this second VCO gives a smooth upward frequency sweep followed by a sudden return to the starting frequency. This may sound difficult to perform but in fact it is surprising how easily the ear can detect any jumps in the modulating waveform. However, if you are unsure of what a VCO sounds like when it is being frequency-modulated by a slow running sawtooth, listen to this effect on its own before trying to set up PR12. Again this adjustment is carried out on each of the six multipliers.

With all the multipliers now set up, speed up the VCO connected to the circuit, attach a lead to the output and you should see/hear a multiple sawtooth waveform. Now switch the modulators on (either setting of SW1) and you should hear a 'phasing' sound as the modulation builds up. Finally, check that the two 'on' positions of SW1 give different depths of modulation.

If, when the modulation switch is set to 'full' you find you can hear a distinct pause or break in one of the new waveforms, this indicates an incorrect setting of the preset associated with the comparator. To determine which one needs a finer adjustment, check each of the six multipliers by attaching an audio lead or probe to point B and adjust PR11, 21, 31 etc as appropriate, so that even with full modulation depth the comparator (and thus the sawtooth) remain within the range of one full cycle of the original sawtooth.

Interfacing Notes

The original circuit was developed for a Digisound 80 modular synthesiser; this has ± 15 V supplies and a positive-going 0–10 V VCO output. However, the circuit is fairly tolerant of a wide range of conditions and can be adapted to almost any synth with possibly some small additional components. The most likely problem concerns the ratio of the VCO output amplitude when compared with the voltage supply rails. The minimum and maximum supply voltages to the circuit described are ± 5 V to ± 16 V. Bearing in mind these two limits, the waveform to be treated should be equal to or greater than one-third the sum of the voltage rails.

If the input waveform does not meet these requirements the obvious course is to add an op-amp to the front end of the circuit to bring the waveform up to the required level. For instance, with the PE Minisonic, which is pretty much a worst case, the power supply is ± 9 V, the VCO output is 1 V peak-to-peak. In this case a simple $\times 6$ amplifier would do, but so as to allow plenty of headroom a $\times 8$ amplifier was made up. **ETI**

BUYLINES

A kit of parts for this project, comprising glass fibre PCB and all electronic components, is available from Digisound Ltd, 14-16 Queen Street, Blackpool, Lancs FY1 1PQ. The prices, inclusive of VAT and postage, are £10 for the multiplier and £6.05 for the modulator. Housing the project is entirely up to you and most people will simply fit the boards inside their synth, but we put ours into a Teko Alba A23 case to match the rest of the Project 80 modules. If you want to do likewise, the case is available from West Hyde, Unit 9, Park Street Industrial Estate, Aylesbury, Buck.

WAVEFORM MULTIPLIER

ETI ARTICLE. JANUARY 1983
No errors have been noted and in Figure 4 it is quite obvious that the unmarked IC on the Multiplier Board is IC2.

GENERAL
With the co-operation of David Ward-Hunt the Waveform Multiplier has been added to the DIGISOUND 80 range. While the cost for a single unit is over half that of an 80-2 VCO the advantage of the multiplier approach is that it produces a 'new' and consistent sound over a wide range.

DIGISOUND 80 users should, however, note the following points -
1. The power connections are not our standard CHIRI type but the PCB's will accept a screw connector shown on the price list as 'Mains 3-pin PCB connector'. One reason for this is that the power connections on the Multiplier Board do not follow our usual convention (also note that power connections are not in the same order on the modulator board).
2. At this time we do not have any plans to produce a panel for this kit. As can be seen from the cased unit in the article one only requires three holes for a single multiplier and the amount of printing is minimal. We do, however, supply blank panels and it would be quite practical to mount two multipliers on a single 3 inch wide panel. To simulate lettering used on our printed panels use 14pt (3.9mm) Helvetica Medium {Letraset Sheet No.729} for the main heading and 12pt (3.0mm) Helvetica Medium {Letraset Sheet No.1568} for the other lettering. Subsequent spraying of the panel with Letraset Matt will prevent scuffing of the lettering. When drilling the panel the best approach is to gradually increase the size of the hole. 1/4 inch or 6.5mm holes are required for both switch and 3.5mm jack sockets.

For customers with multiple VCO's it may be advantageous to have a larger panel - please let us know whether you wish to have 6 X 9 and 9 X 9 inch blank panels added to the range.

digisound

MODULE 80-21

VOLTAGE CONTROLLED DIGITAL OSCILLATOR

1. INTRODUCTION

A conventional VCO can produce several waveforms, rich in harmonics, which may be filtered in order to alter the timbres. This is quite satisfactory for a wide range of musical requirements but the small range of waveforms available (usually sawtooth, square and triangle) and the coarse effects of analogue filters mean that it is impossible to produce many of the delicate, natural sounds which are so characteristic of modern digital synthesis. This module adds some exciting new possibilities to existing synthesisers by combining the flexibility of analogue voltage control with the clarity and realism of digitally generated waveforms.

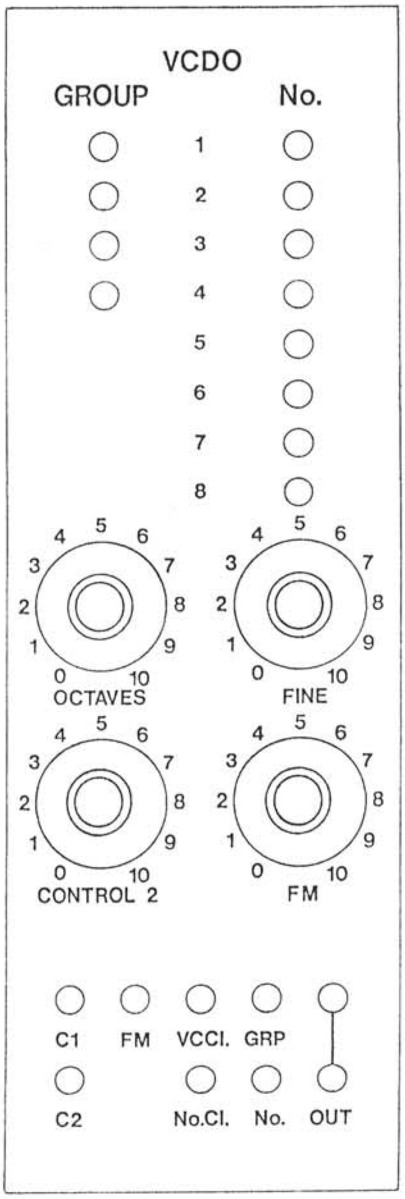


FIGURE 1. 80-21 PANEL LAY-OUT

As a unit, the voltage-controlled digital oscillator (VCDO) may be regarded as an ordinary VCO, but with a far greater range of waveforms. The design is fully compatible with existing synthesiser systems (1V/octave frequency control, 10V peak-to-peak output, linear and exponential modulation inputs) and offers the versatility of 32 different waveforms covering a wide variety of sound textures. A particular waveform can be selected either with push-button switches using a simple incremental system or by a combination of a push-button switch and suitable electronic pulses to the input provided. The module has a wide frequency range (approximately 30Hz - 10kHz) which allows it to be used as either an audio or modulation source.

2. DESIGN

The VCDO works on a very simple principle. The 32 waveforms are encoded in a 2716 EPROM with each waveform being represented as a series of 64 8-bit numbers (a wavetable). A binary counter is made to step through the waveform data at a frequency controlled by a VCO and each item of data is subsequently converted into an analogue voltage by a simple 8-bit DAC.

The complete circuit diagram for the 80-21 VCDO is shown in Figure 2. IC 1 is a CEM 3340, which with the addition of a few external resistors and capacitors functions as a high quality VCO, featuring accurate exponential and linear control of frequency. Manual control of frequency is possible by means of potentiometers RV6 and RV7, providing coarse and fine adjustment respectively. External voltage control of frequency is

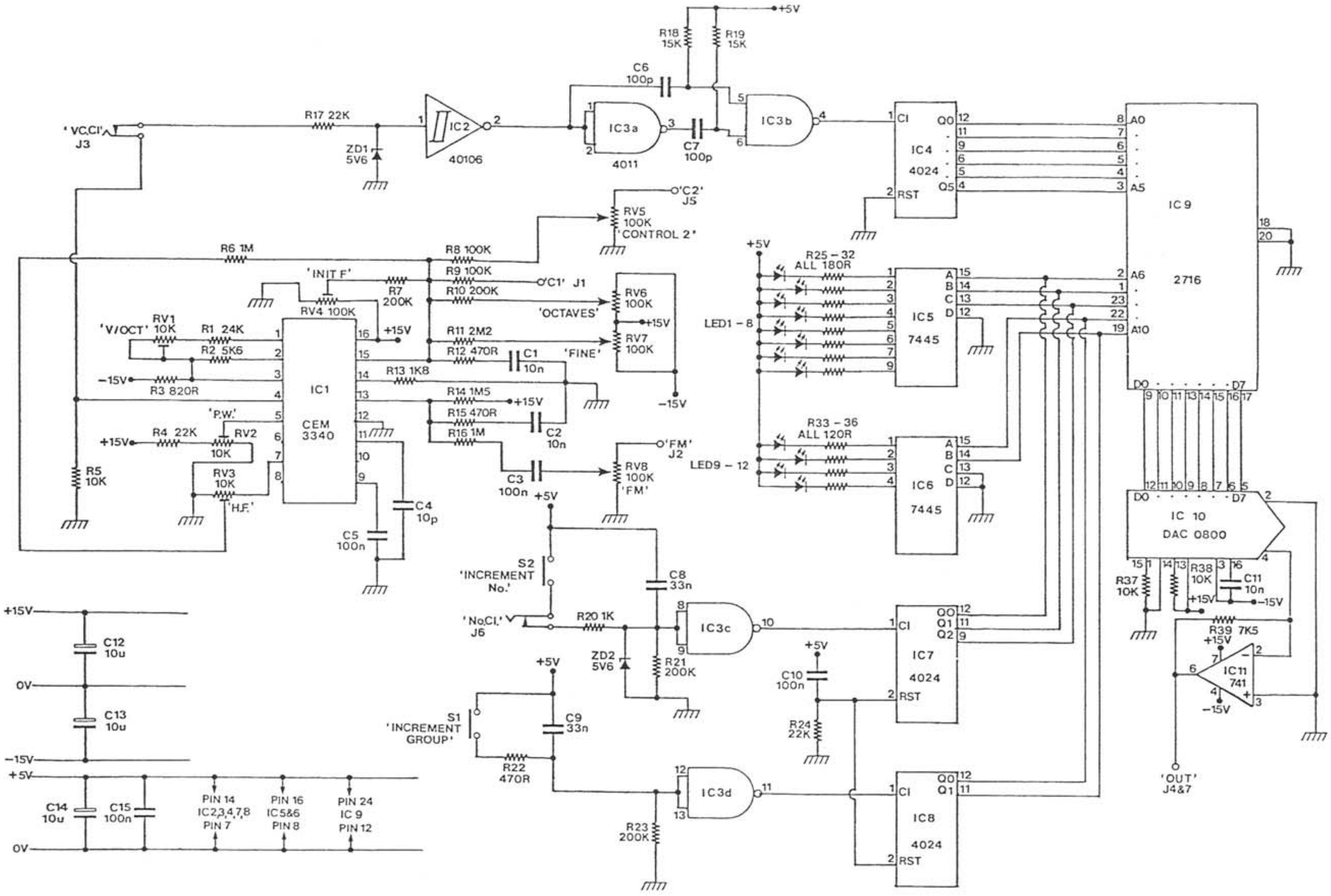


FIGURE 2. 80-21 VCDO CIRCUIT DIAGRAM

possible by connection to C1 and C2 (exponential response) and also via a linear frequency modulation input. The C2 and FM inputs are equipped with attenuating potentiometers, RV5 and RV8. Three output waveforms are provided (triangle, sawtooth and pulse), but in this application only the pulse output is required, which is available at pin 4. A positive-going control voltage to pin 5 allows adjustment of the duty cycle of the pulse wave from approximately 0% to 100%. Frequency control is by means of timing capacitor C4 and multiple voltage control via resistors R8-11 to pin 15, which is a virtual earth summing node. In this module, the frequency range has been shifted upwards compared to a normal VCO by altering the value of C4 from 1nF to 10pF. Additionally, pin 13 may be employed as a linear frequency control input, providing the facility of linear frequency modulation. The VCO is configured such that it may be calibrated for an accurate +1V/octave response using presets RV1 and RV3. Provision has also been made for connection to an external VC clock which, if permanently connected, allows the removal of the CEM 3340 and associated circuitry.

The pulse output is suitably attenuated to 5V by R17/ZD1 and is further processed by a Schmitt trigger (1/6 of IC 2). Squaring of the pulse output is necessary as at extremely high frequencies an unacceptable amount of slewing is present, which inhibits operation of the next circuit block, a frequency doubler. The frequency doubling circuitry configured around IC 3a and IC 3b is included to provide an extra octave's range. It functions by separately differentiating both edges of the square wave - C6/R18 differentiate negative edges and C7/R19 differentiate positive edges. The output of IC 3b is then a series of narrow pulses corresponding to both edges of the original square wave clock signal. Ripple counter IC 4 steps through the lower six address bits of IC 9, a 2716 EPROM suitably programmed with wavetables. The data outputs at pins 9-17 of the 2716 go directly to IC 10, which is a high speed multiplying digital-to-analogue converter (DAC 0800). The data is thus converted to an analogue voltage which is buffered by IC 11. The same IC also scales the output to 10V peak-to-peak.

The 32 waveforms are subdivided into 4 Groups of 8 Waveforms and ripple counters IC 7 and IC 8 are used to select the required waveform Number and Group respectively. Their clock inputs (pin 1) are fed by IC 3c and IC 3d which invert and debounce the switches S1 and S2. Additionally, an external input is provided so that a suitable waveform or pulse train may be used to advance the waveform Number in a particular Group. ZD2/R20 are included to limit an incoming externally generated pulse to +5V. R24 and C10 form a power-on reset network to take the reset inputs of the select counters high at switch-on in order to start at waveform Number 1 in Group 1.

IC 5 and IC 6 are BCD to decimal converters and LED drivers, displaying two decimal equivalents present on the upper five address lines of the 2716. Thus the two highest address lines (A_9 and A_{10}) are decoded to light one of four green LEDs representing the waveform Group whilst control lines A_6 to A_8 light one of eight red LEDs representing the waveform Number.

Power supply requirements to the VCDO are +/-15V at approximately 40mA per rail and a separate +5V rail at 500mA.

3. CONSTRUCTION

The PCB component lay-out is shown in Figure 3. There are a number of wire links to be made on the board and these should be inserted first. The rest of the components should then be fitted onto the PCB in order of increasing height (i.e. zener diodes, resistors, IC sockets, presets and capacitors). Note the orientation of the electrolytic capacitors and ensure that all the ICs are inserted as shown on the component overlay as they do not all have the same orientation. The use of a PCB solvent cleaner to remove residual flux is recommended.

Off the board, there are 12 LEDs, 4 potentiometers, 7 jack sockets and 2 push-button switches to be wired up. These components may be mounted on a front panel as shown in Figure 1, or in any other format that individual constructors may wish to use. The connections to be made between the front panel and the PCB are shown in Figure 4.

The PCB has a space for a four pin CHIRI-type connector which may be used for the power supply connections rather than hardwiring them to the board.

4. CALIBRATION

Once construction is complete and the unit has been carefully checked, set all presets to mid-position and power up. Calibration of the VCO circuitry is by way of four presets and is carried out as follows.

Firstly, RV2 is adjusted so that the unit operates over a frequency range from approximately 30Hz up to 10kHz. The correct setting of RV2 is likely to be slightly anti-clockwise from mid-way and can be recognised when the frequency may be increased (e.g. by RV6) without any noticeable sudden jumps.

The two multiturn presets, RV1 and RV3, are used to achieve a precise 1 volt/octave CV to frequency relationship and may be calibrated in a number of ways. The most convenient method is to use a previously calibrated keyboard, but failing this a variable voltage source which can be increased by precisely one volt may be used. Also required is some means of observing the output frequency. The simplest way is to take the output through an amplifier and speaker and to calibrate it by ear, providing the ear concerned has had some musical training. Alternatively, a frequency meter or oscilloscope may be used to visually display the frequency. Once the wiper of RV3 has been grounded, calibration may proceed by increasing the output of the variable voltage source/keyboard by precisely 1 volt whilst it is connected to control input C1. RV1 is then adjusted to

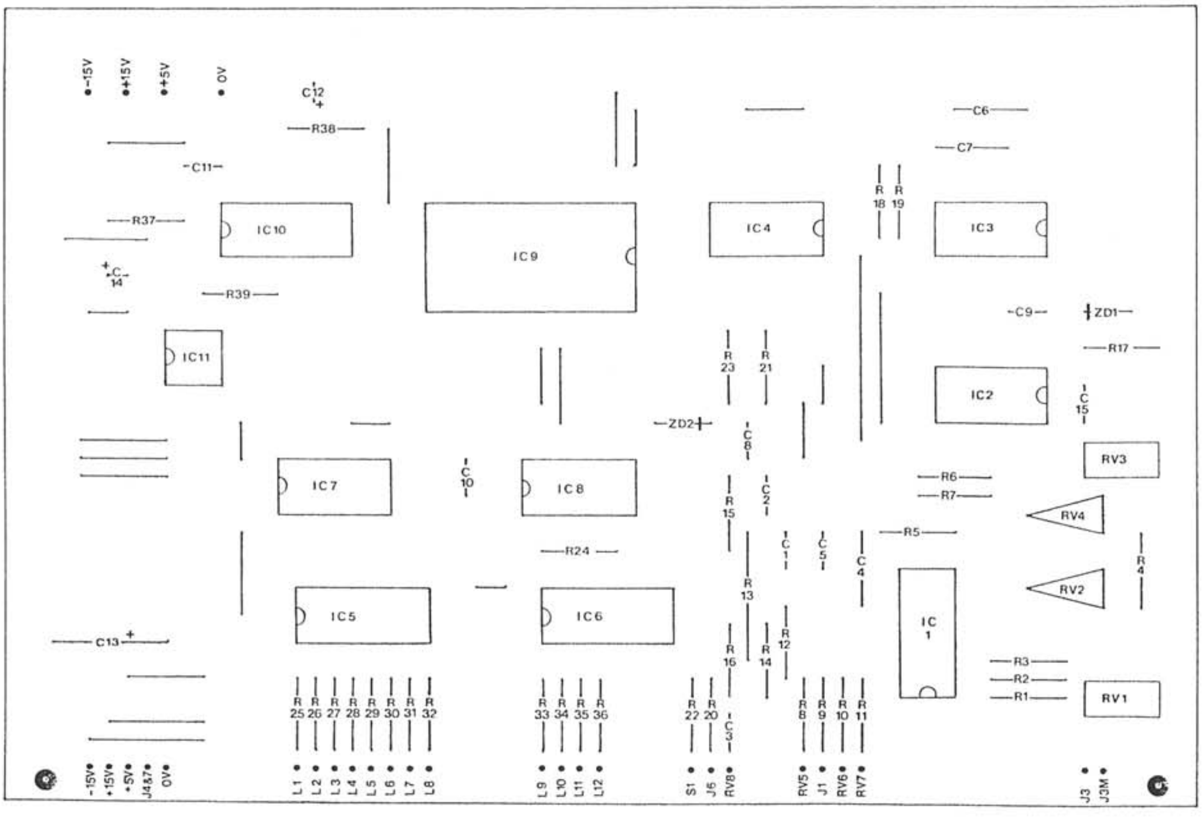


FIGURE 3. 80-21 PCB COMPONENT LAY-OUT

produce an exact doubling of the output frequency. This procedure is best repeated several times over a frequency range of 150-500Hz (this may be varied using RV4, 6 and 7).

This procedure is then repeated using an initial frequency of about 5kHz and adjusting RV3 to achieve an exact

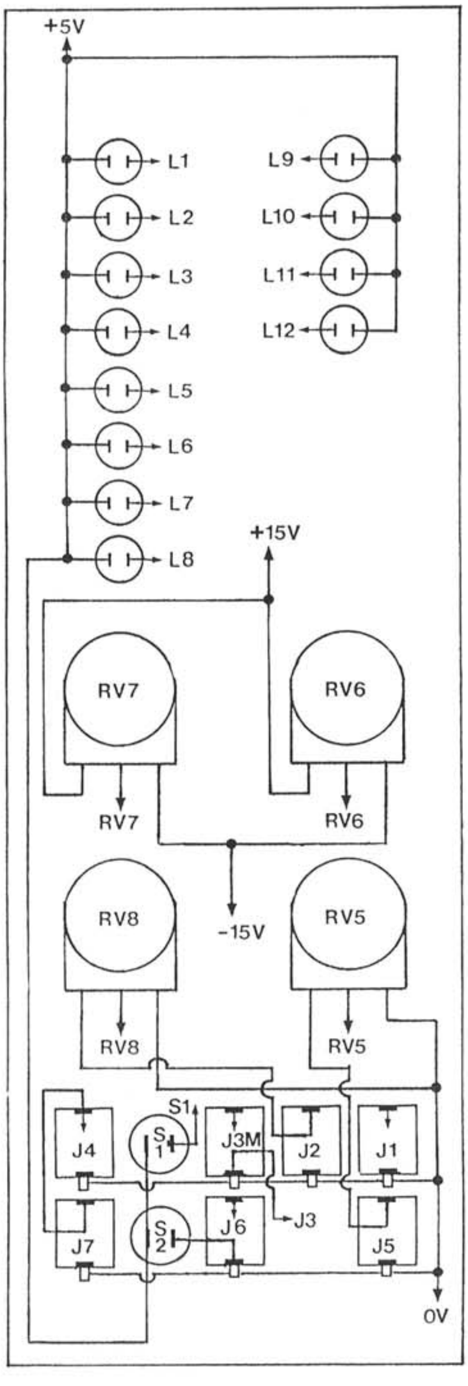


FIGURE 4. 80-21 PANEL WIRING

doubling of frequency when the applied voltage is increased by precisely 1 volt. The module should then accurately track over its entire range.

The final step in the calibration sequence is to adjust RV4 to give a convenient initial frequency when no inputs are connected, which to a large degree is a matter of personal taste. It may, for example, be set to 65.4Hz, which is the lowest note on a 4 octave C-C keyboard.

5. IN USE

The VCDO is supplied with a pre-programmed EPROM containing the data for 32 64-byte waveforms. Organised in 4 groups, these are as follows:-

1) Starting as a sine wave, this group progresses with the addition of extra harmonics in varying quantities, though none above the sixth are added.

2) The waveforms of this group contain some higher harmonics, and as a result sound brighter.

3) With lots of high harmonics and subdued lower harmonics and fundamental, these waveforms sound characteristically sharp and metallic.

4) This group contains some of the basic waveforms to be found on a conventional VCO (sawtooth, square, triangle, pulse etc.) plus one or two more unusual waveforms.

Plots of the 32 waveforms present on the EPROM are to be found in the Table attached.

With suitable filtering and envelope shaping, a wide variety of sounds can be produced, both imitative and innovative. On the imitation side, Groups 1 and 2 can provide some very good church organs as well as xylophone, electric piano etc. Group 3 is ideally suited for bells, gongs, chimes and so on. Group 4 enables you to use the VCDO for conventional synthesis but it also includes some unusual waveforms unavailable on a standard VCO. As might be expected, the use of several VCDOs in a polyphonic system sounds especially impressive.

One or two unusual modes of operation yield some novel effects. Use of a linear FM patch produces sounds similar to those obtained from the recently popularised FM synthesisers. The waveform select input provides the possibility of cycling through any particular group, which can be quite dramatic when free-running or in time with the EG trigger from a sequencer/arpeggiator.

Additionally, the VCDO can operate as a modulation source. However, the output is stepped, and if being used as a frequency modulator for a VCO, for example, some form of filtering should be used in order to "smooth out" the waveform. This would be unnecessary for amplitude modulation.

6. COMPONENTS

RESISTORS, 5%, 1/4w carbon film

R3	820R
R4,17,24	22k
R5	10k
R11	2M2
R12,15,22	470R
R18,19	15k
R20	1k
R21,23	200k
R25-32 (8 off)	180R
R33-36 (4 off)	120R
R39	7k5

RESISTORS, 1%, 1/4w metal film, 100ppm

R1	24k
R2	5k6
R6,16	1M
R7,10	200k
R8,9	100k
R13	1k8
R14	1M5
R37,38	10k

POTENTIOMETERS, SWITCHES

RV1,3	10k min. multiturn, side adj.
RV2	10k horizontal preset
RV4	100k horizontal Cermet preset
RV5-8	100k lin. rotary
S1,2	push to make

CAPACITORS

C1,2,11	10n polyester
C3,5,10,15	100n polyester
C4	10p polystyrene
C6,7	100p polystyrene
C8,9	33n polyester
C12,14	10u PCB electrolytic
C13	10u axial electrolytic

SEMICONDUCTORS

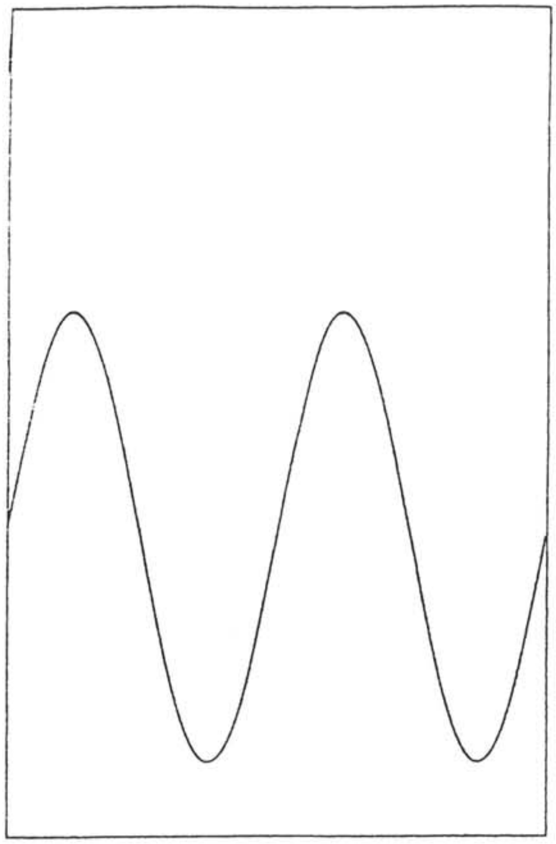
IC1	CEM 3340
IC2	40106
IC3	4011
IC4,7,8	4024
IC5,6	7445
IC9	2716
IC10	DAC0800
IC11	741
D1-8	5mm red LED
D9-12	5mm green LED
ZD1,2	5V6 400mW zener

TABLE 1 - WAVEFORM PLOTS

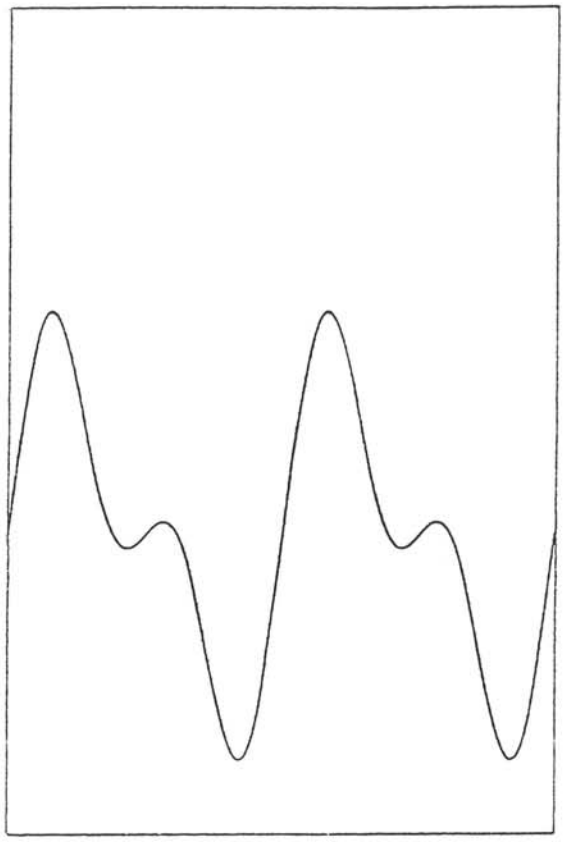
The diagrams in this table show the waveforms present in the first 3 Groups, stored in the 2716 EPROM. These plots have been filtered in order to produce continuous rather than stepped waveforms, but because of this they do not all exactly match the stepped waveforms that can be observed directly from the output on an oscilloscope. However, they do give a good guide to the type of harmonic content present in each waveform.

The waveforms from Group 4 are not included as they consist of 6 "conventional" waveforms and 2 "random" waveforms. The details are:-

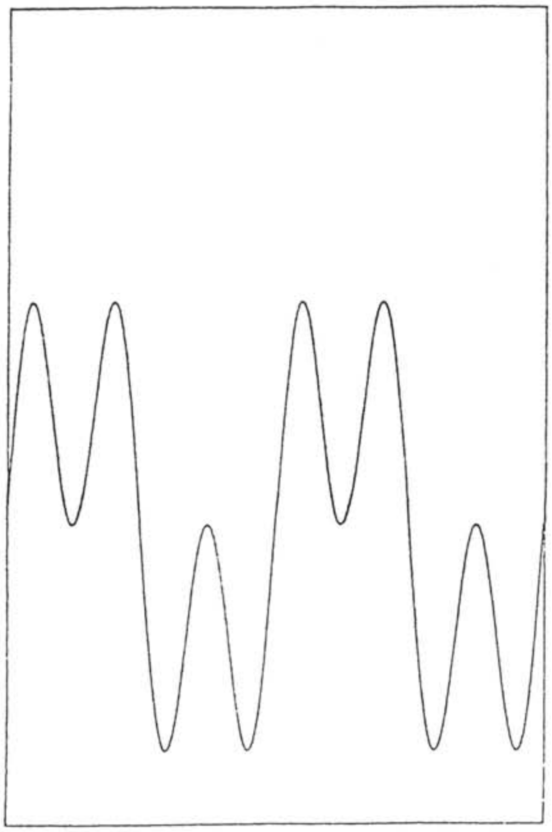
1 Triangle
2 Square
3 Sawtooth
4 2 x Sawtooth
5 Pulse (3:1 mark/space)
6 Random
7 Square & Sawtooth
8 Random



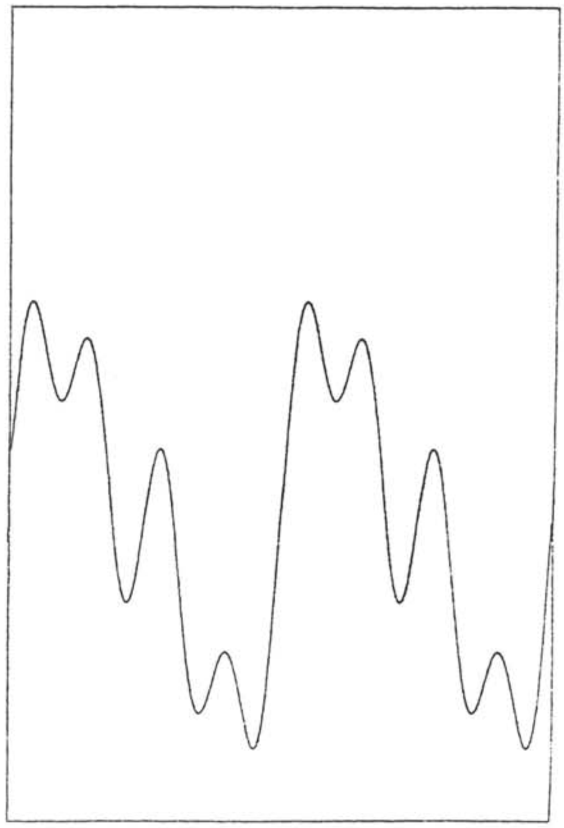
GROUP 1 WAVEFORM 1



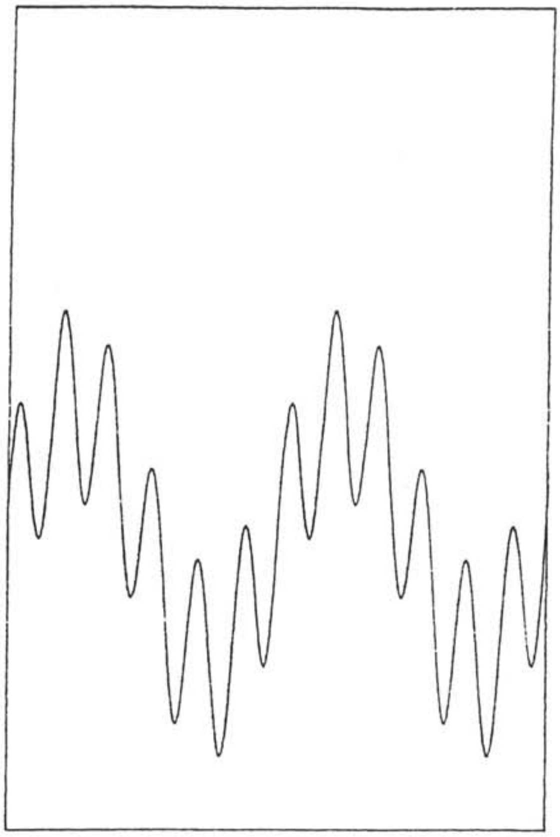
GROUP 1 WAVEFORM 2



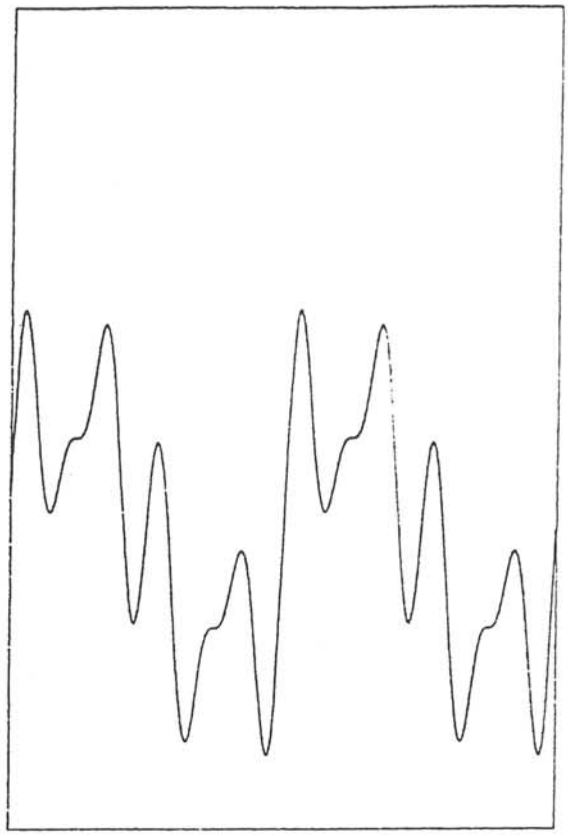
GROUP 1 WAVEFORM 3



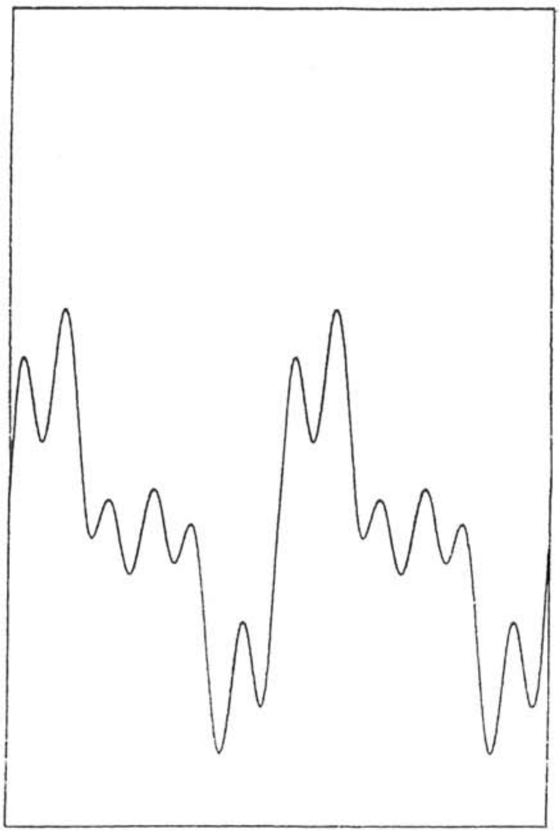
GROUP 1 WAVEFORM 4



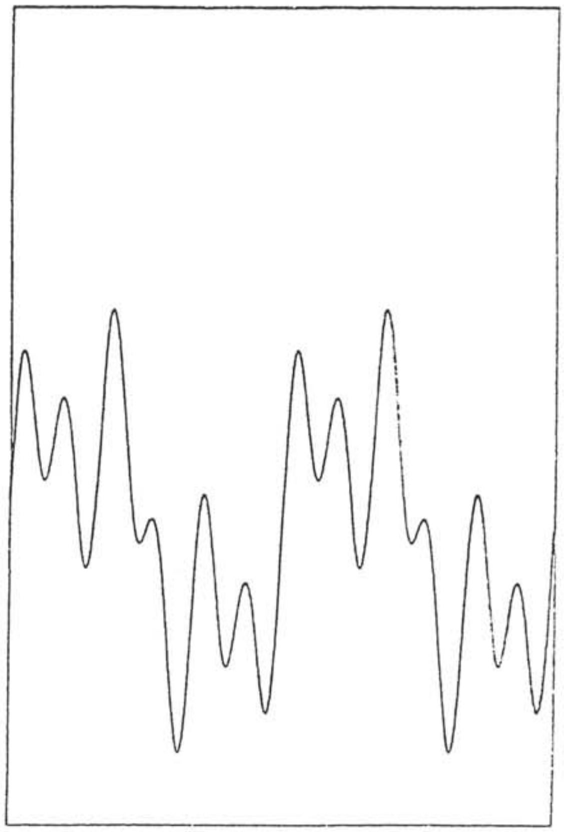
GROUP 1 WAVEFORM 5



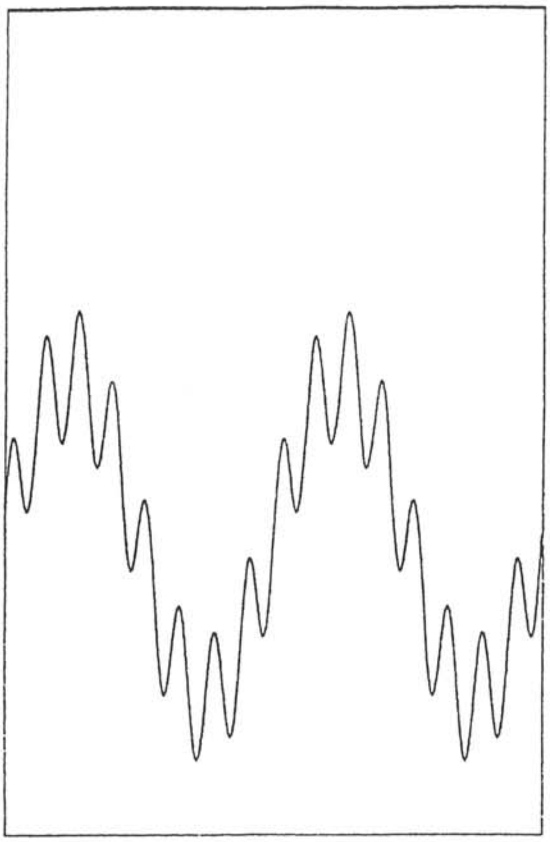
GROUP 1 WAVEFORM 6



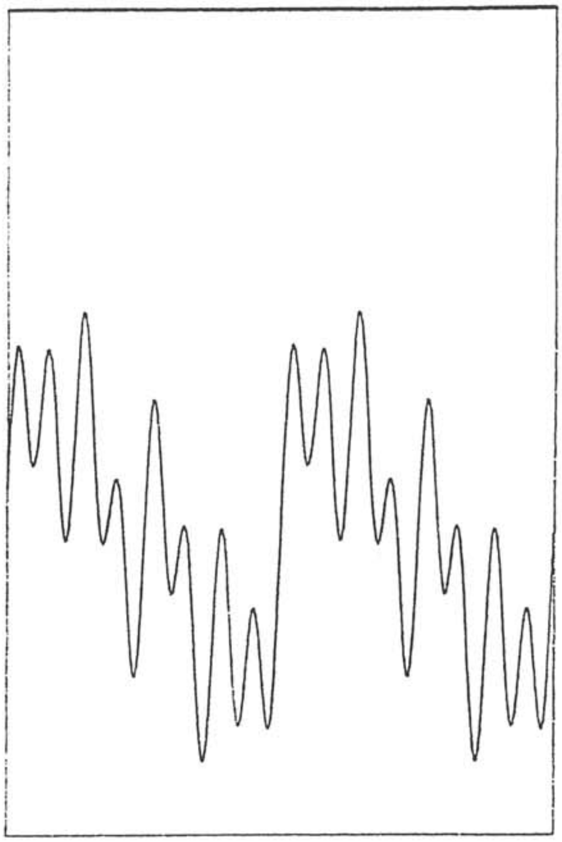
GROUP 1 WAVEFORM 7



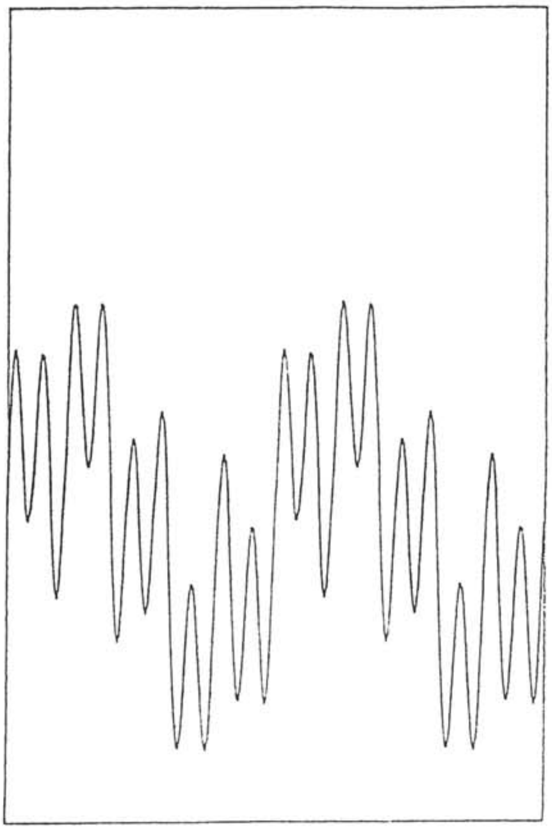
GROUP 1 WAVEFORM 8



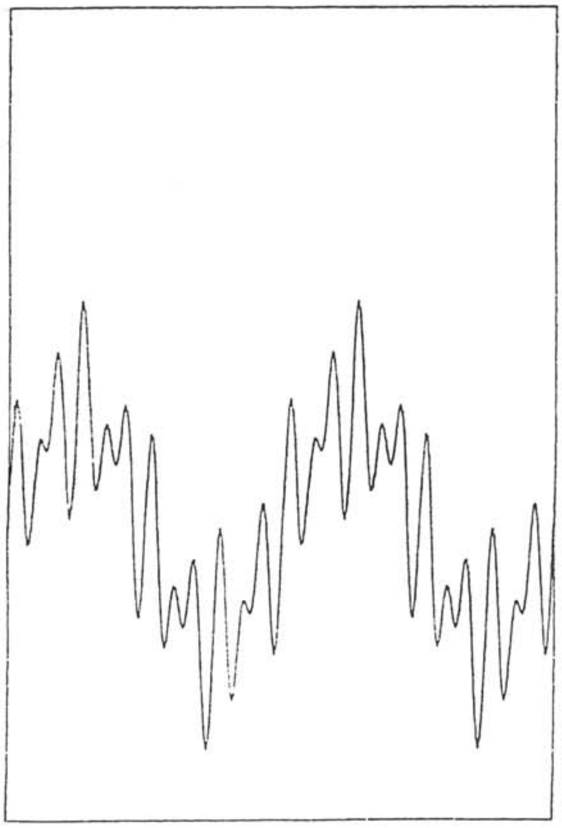
GROUP 2 WAVEFORM 1



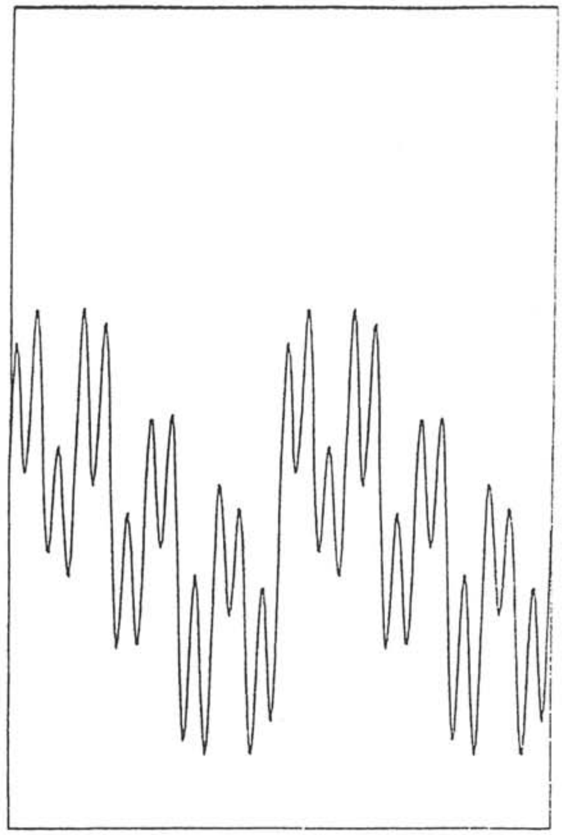
GROUP 2 WAVEFORM 2



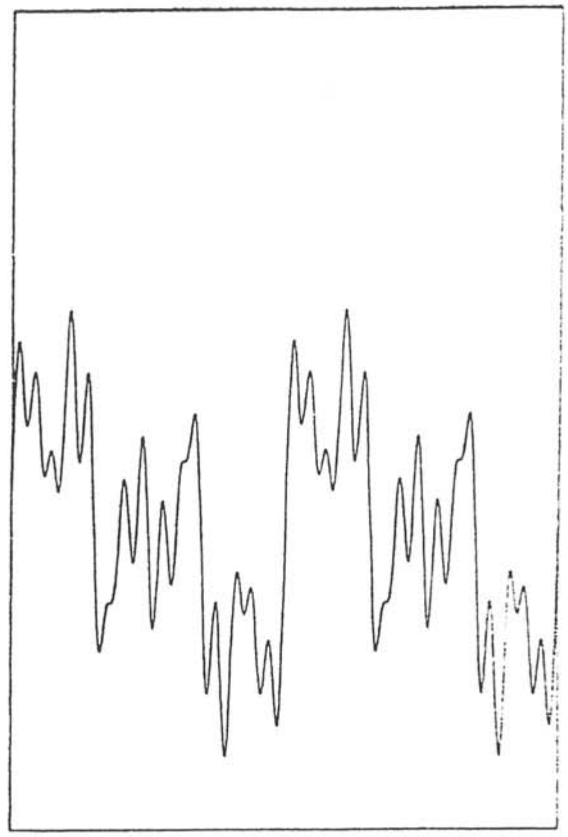
GROUP 2 WAVEFORM 3



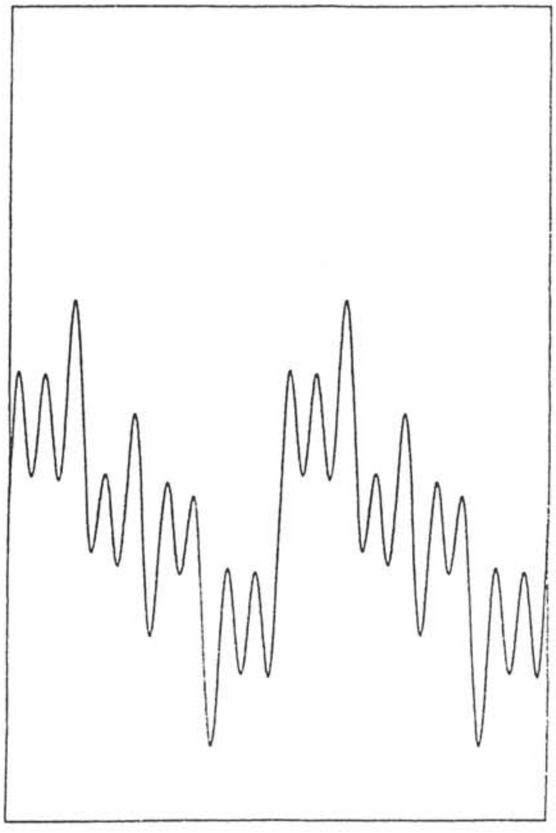
GROUP 2 WAVEFORM 4



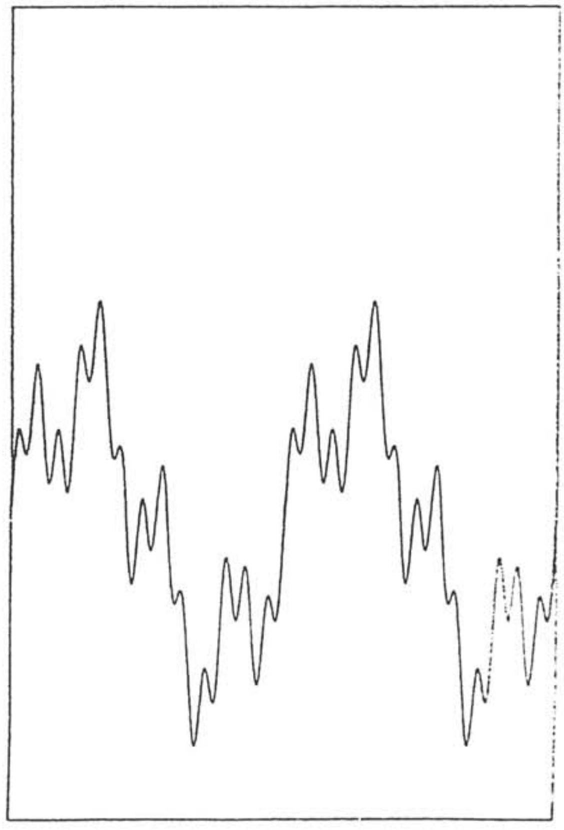
GROUP 2 WAVEFORM 5



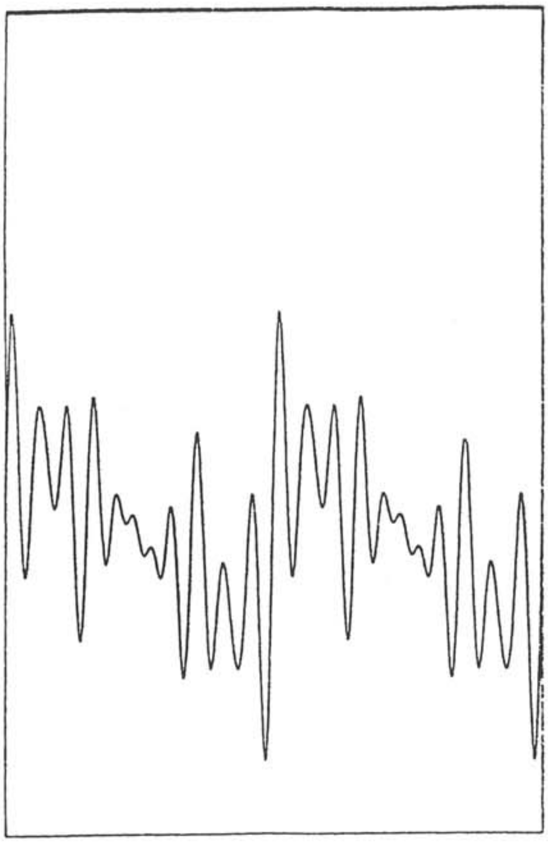
GROUP 2 WAVEFORM 6



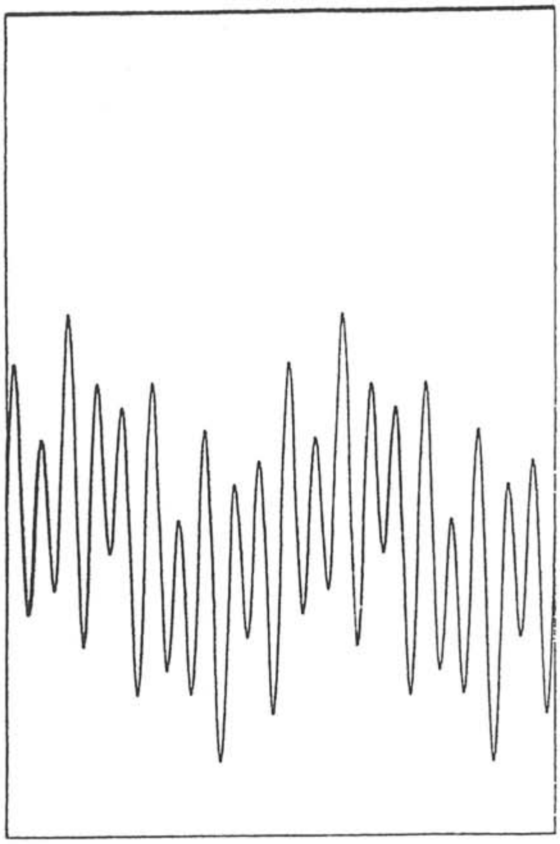
GROUP 2 WAVEFORM 7



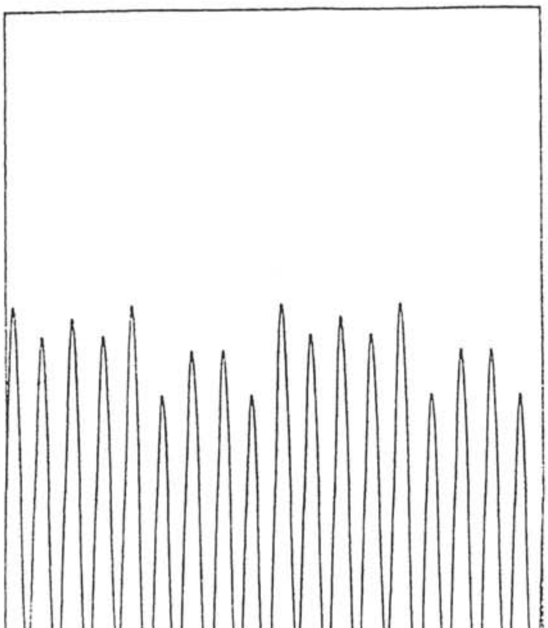
GROUP 2 WAVEFORM 8



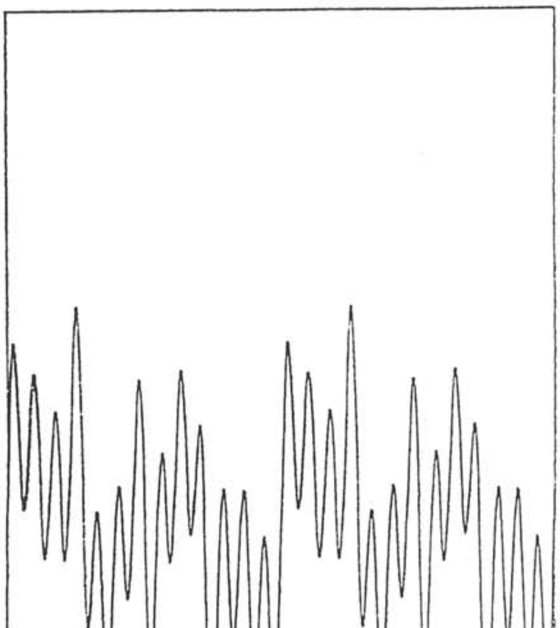
GROUP 3 WAVEFORM 1



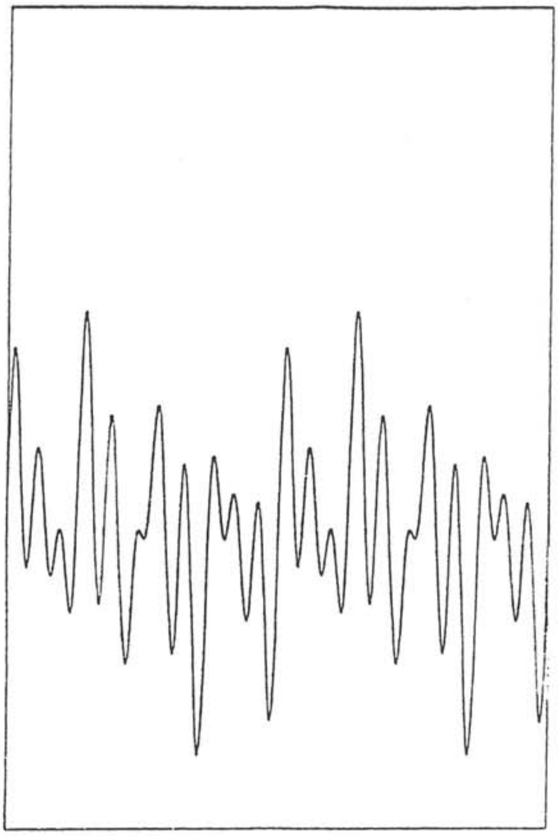
GROUP 3 WAVEFORM 2



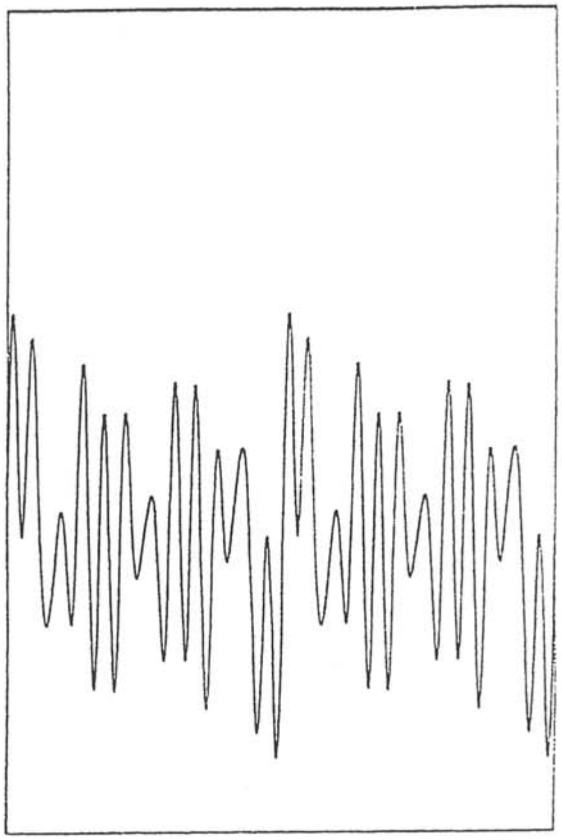
GROUP 3 WAVEFORM 3



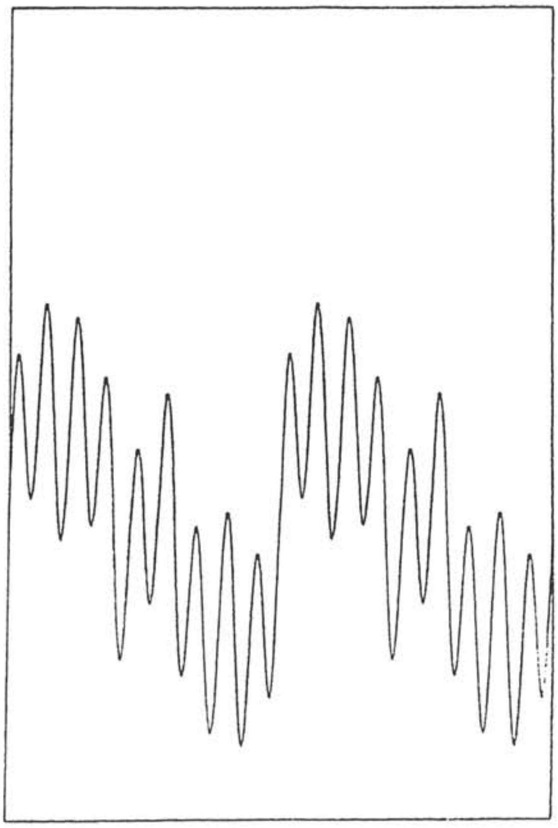
GROUP 3 WAVEFORM 4



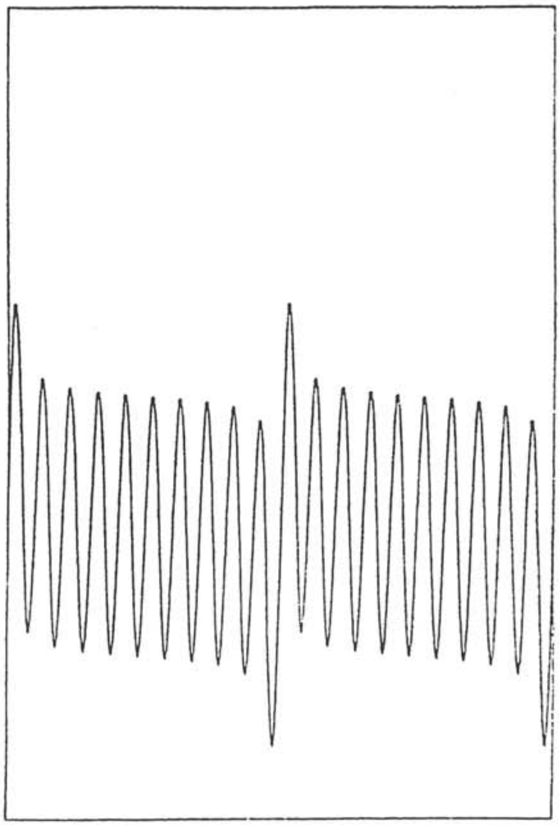
GROUP 3 WAVEFORM 5



GROUP 3 WAVEFORM 6



GROUP 3 WAVEFORM 7



GROUP 3 WAVEFORM 8

INTRODUCTION

This simple design has been developed to allow preselected voltage levels to be instantly recalled, but may be used in many ways, e.g. as part of an LFO or a simple sequencer. We are sure that we have not thought of all of its possible uses, so there is plenty of scope for you to use your imagination!

DESIGN

The circuit diagrams for the two parts of this unit are shown in Figure 1. Taking first the "preset board" (80-22B), the design centres around the 4051 CMOS multiplexer IC. This device functions as a 1 pole 8 way electronic switch, selecting 1 of 8 DC voltage inputs, the level of which is adjusted using the appropriate preset. The 1 of 8 function is selected by a 3 bit address input so that 000 (binary 0) selects input number 1, 001 enables input number 2 etc. The voltage out of the 4051 (pin 3) is then buffered by an op-amp (1/2 of a 1458) which may be configured as either a simple non-inverting buffer or as an inverting op-amp with a gain of approximately one. In the latter case the maximum output level is limited to -12.6V due to circuit action.

The input to the 4051 is linked to the controller board (80-22A) which simply provides the facility to increment the preset number selected. This is achieved using a 4024 binary ripple counter (IC 2) with the clock input fed by 1/4 of IC 1 (4011) which suitably debounces and inverts the switching performed by S1 or an external clock signal. Thus, each time the switch is pressed, the binary number present on pins 12, 11 & 9 of IC 2 increases by one. This three bit number drives a 7445 BCD to decimal converter and LED driver which lights one of eight red LEDs to display the currently selected preset group. Simultaneously, the three bit number is brought up to +15V logic levels by the three inverting transistor buffers (TR 1-3), in order to be compatible with the 4051 address inputs. Because of this inverting action, preset selection number 1 will close switch position number 8 on the 4051 - in other words the system counts backwards, which is of little consequence in this application. It should also be noted that on power-up, position number 1 is selected as a power-on reset function is performed by R3 and C2.

CONSTRUCTION

The first step prior to construction is to decide which, if any, of the op-amp buffers should be made to invert and the position of the two resistors and link noted. The printed circuit mounting components should be assembled according to the PCB overlays (Figures 2 and 3) in order of increasing height. Figure 3 depicts two positive going buffers (top pair) and two inverters (bottom pair), but this is only an example - different arrangements can be readily determined from this diagram. Check the orientation of polarised capacitors and semiconductors prior to soldering. Once assembly is complete, it is advisable to clean the PCBs using a solvent cleaner and to inspect them closely for dry joints and solder bridges.

It is recommended that at this stage the units are temporarily connected and powered up to check for correct operation (see Calibration section). Once they are functioning correctly, the constructor must decide on the physical positioning of the two types of PCB. It is suggested that the controller PCB is panel-mounted so that the LEDs, the switch and the jack socket are easily accessible. To this end, a punched (but not printed) panel is

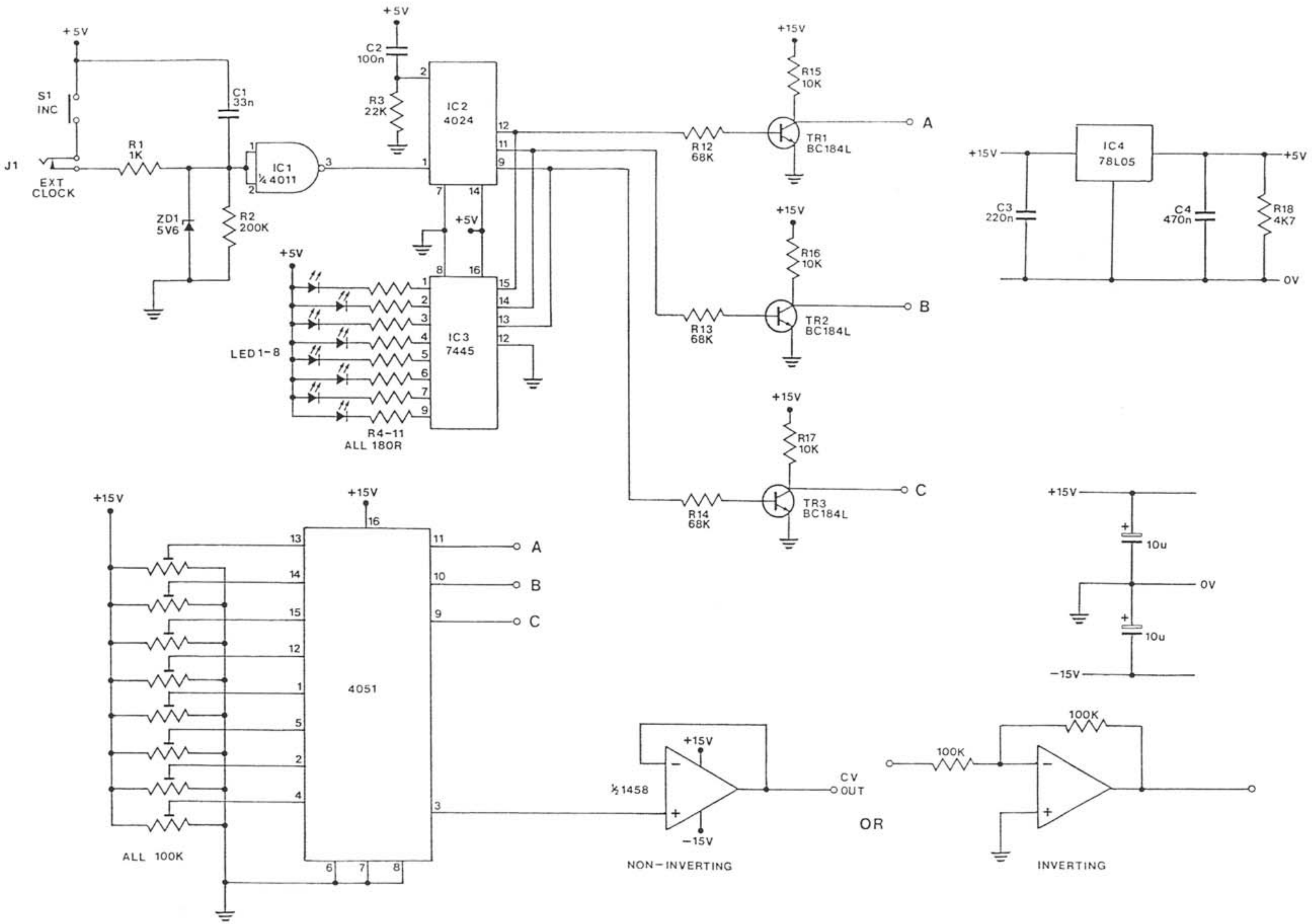


FIGURE 1. 80-22 CIRCUIT DIAGRAM

available. The preset PCB is probably best positioned near to the CV inputs it will control (i.e. behind the front panel of the relevant module). In this type of arrangement, the three control lines (A, B & C) from the 80-22A PCB will have to be distributed to all of the 80-22B PCBs that are being used. However, this may be done in a series manner rather than taking all 80-22B boards back to a single 80-22A board.

The power supply requirements for a 1 + 1 combination are +15V at 75mA and -15V at 7mA. Both circuit boards will accept our standard three way "mains" PCB connector for these connections.

CALIBRATION

No calibration is necessary on either board, except of course for the adjustment of up to 32 presets on each 80-22B board! However, these are best set with the unit connected to the appropriate CV input(s). With power applied to both PCBs, LED number 1 should be lit and the voltages corresponding to the position of all number 1 presets should be present at their respective outputs. A quick check of this may be made by connecting a voltmeter across the output and checking that as the program number is incremented, adjustment of the appropriate preset (as shown in Figure 3) produces a corresponding variation in output voltage. Once this has been done, the unit may be permanently installed and adjusted to the desired preset levels.

IN USE

The Patcher provides a simple yet reliable method of retaining preset CV levels. Each patch control unit (80-22A) can select 1 of 8 "programs" and may control an almost unlimited number of preset boards (80-22B). Thus, with a basic system (i.e. one of each board), it is possible to preset 4 different control voltages on 8 programs. The output voltages could then be hardwired to the break connection of any control input jack socket without losing any flexibility as the preset voltages are disabled on insertion of a jack plug. Alternatively, it may be preferable to use the Patcher to substitute for specific settings of a potentiometer. In this case preset number 1 should be omitted and the wiper of the pot. taken to the wiper position of the omitted preset. In this way preset position number 1 will correspond to normal pot. control, while positions 2 to 8 may be any preset voltage. This type of arrangement is only possible with positive going CVs as a voltage below V_{EE} (0V) will destroy the 4051 IC. However, negative going CVs may be generated by reconfiguring the output buffer to invert the preset voltage (i.e. a preset voltage of +5V becomes -5V etc.).

As a working example, some users may wish to preset seven envelopes together with normal manual envelope control on an 80-18 module. Referring to the construction notes for this module, it can be seen that the Attack, Decay and Release controls require negative going control voltages to increase these three time constants. This can be accomodated by firstly rewiring the three pots. to the +15V PSU rail rather than the -15V rail. The three wiper connections should then be disconnected and taken to the wiper positions of three number 1 presets. With three corresponding 4051 buffers arranged to invert (i.e. using 6 100k resistors) the incoming, now positive, pot. controlled CVs will be negated. This negative output voltage may then be taken back to the A, D & R points on the PCB. It should be apparent that the sustain control may be similarly adapted but without rewiring the pot. and using a non-inverting output buffer configuration.

With the rewiring complete, the 80-18 module will function normally in preset position number 1 (apart from a slightly limited A, D & R range due to the inverting action of the output buffer).

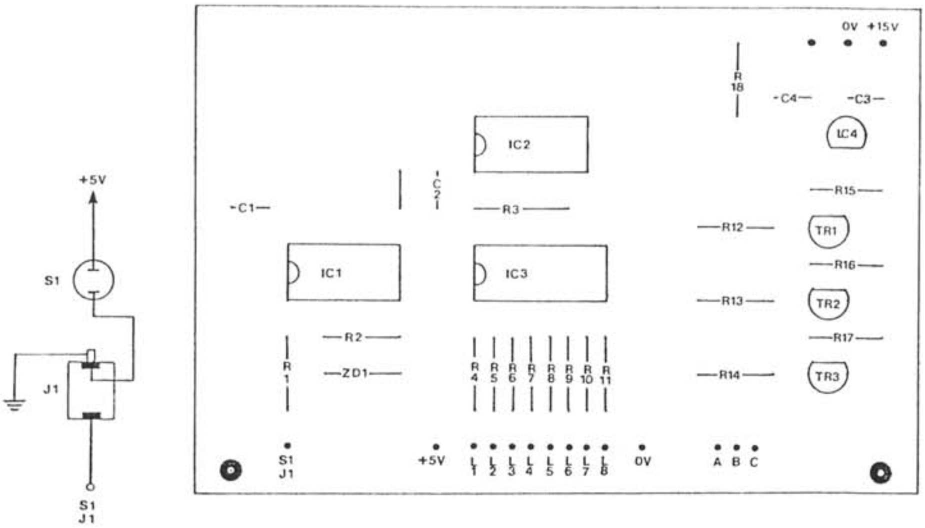


FIGURE 2. 80-22A COMPONENT OVERLAY

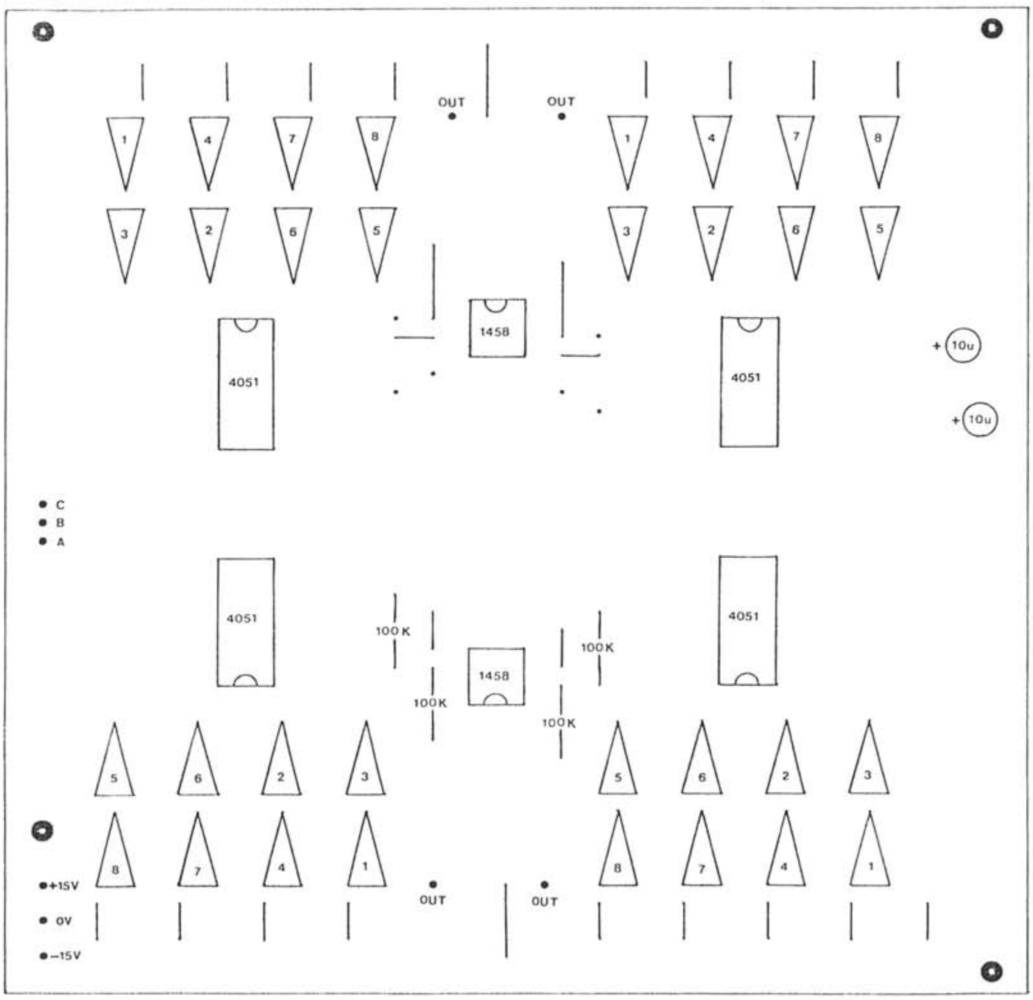


FIGURE 3. 80-22B COMPONENT OVERLAY

Potentiometers which have one track end wired to the negative PSU rail and the other end wired to the positive rail (e.g. VCO frequency) are best adapted to purely positive going CVs, although this will result in a halved control range.

It is worth noting that some particularly interesting (and useful) "sequencer effects" may be generated by using the external clock facility. As an example, the previously described modified 80-18 module could be connected such that the same gate input clocks the program number. In this way a different envelope would result each time a new note is played (repeating after eight notes).

Another, rather different mode of operation involves the use of the Patcher with a clock oscillator. In this situation the combination of one control board and one preset board can function as a simple 4 channel, 8 step/note sequencer, where additional channels may be added in groups of 4. The gate pulse would be the clock input, and if this were a pulse wave with variable pulse width (as from an 80-19 Dual VCLFO), the "gate-on" time is continuously variable.

In addition, the 1 + 1 combination with an oscillator can be used as a four phase, multi-waveform LFO where the frequency is the clock frequency divided by eight. The only drawback to this approach is that the waveform outputs are stepped. This may be alleviated by inserting a slewing capacitor between the 4051 output (pin 3) and ground. The value of this capacitor is dependent on the frequency and degree of smoothing required.

A further mode of operation might be as a signal routing network. In this case, 8 different waveforms would be linked to the 8 positions that the preset wipers would normally occupy (no presets are needed in this application).

The 80-22B PCB may also be adapted for microprocessor control. The three control lines (A, B & C) could be connected to parallel I/O ports and may then be switched in any order. Also, multiple boards may be used with each under individual control rather than bussed together as in the above descriptions. The only modification required would be the use of 7407 O/C buffers to bring the +5V computer logic levels up to +15V.

Almost any combination of control boards and preset boards may be used, depending on the particular requirements of individual users. However, as either a preset memorizer or an interesting modulation source, the possibilities are endless!

COMPONENTS

80-22A

RESISTORS, 5%, 1/4w carbon film

R1	1k0
R2	200k
R3	22k
R4-11 (8 off)	180R
R12,13,14	68k
R15,16,17	10k
R18	4k7

CAPACITORS

C1	33n polyester
C2	100n polyester
C3	220n polyester
C4	470n polyester

SWITCH

S1	push to make button

SEMICONDUCTORS

IC1	4011
IC2	4024
IC3	7445
IC4	78L05
TR1,2,3	BC 184L
L1-8	5mm red LED
ZD1	5V6 400mW zener

80-22B

RESISTORS, 5%, 1/4w carbon film

8 off	100k

CAPACITORS

2 off	10u PCB electrolytic

POTENTIOMETERS

32 off	100k horizontal preset

SEMICONDUCTORS

4 off	4051
2 off	1458

digisound

MODULE 80-23

QUAD LOW FREQUENCY OSCILLATOR

1. INTRODUCTION

The 80-23 Quad LFO is provided as a low cost complement to the 80-19 Dual VCLFO, and is intended for use in applications that do not require voltage control of LFO frequency or precision modulation signals, merely a cyclically changing control voltage. Four LFOs of identical design have been readily incorporated onto a single PCB. Each LFO output has six switchable waveforms:- square (SQ), triangle (TR), sawtooth (SA) and pulse (PU) together with inversions of sawtooth and pulse (SA & PU respectively).

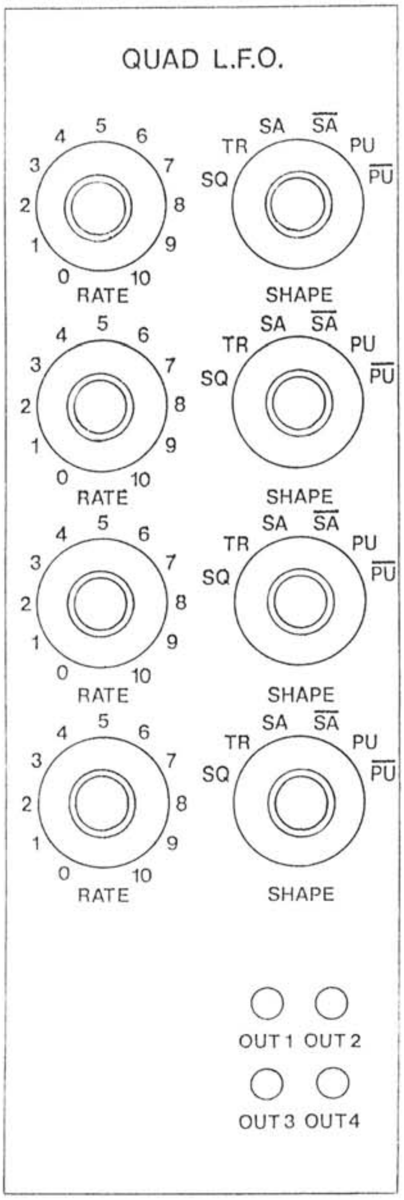


FIGURE 1. 80-23 PANEL LAY-OUT

2. DESIGN

The circuit diagram for one of the four LFOs is shown in Figure 2. The design is conventional for LFO circuitry and utilises a two op-amp configuration which simultaneously generates square and triangle waveforms. The rate of charge of capacitors C1 and C2 (forming, in effect, a non-polarised electrolytic) is governed by the setting of RV1 and the value of R1. However, using switch S1, diodes D1 and D2 may be switched in circuit to force rapid discharge of an otherwise symmetrical waveform. The direction of D1 or D2 selects either a falling or rising ramp waveform or one of two state mark/space pulse waveforms. The selected output waveform is present at pole A of S1, where each waveform is selectively amplified by 1/4 of IC 3 to bring levels up to about +/-5V P-P, with resistors R5 and R6 providing the necessary scaling factors. The four pulse and sawtooth waveforms are a little below the triangle and square levels due to the diode drop (about 0.6V) caused by either D1 or D2. It was felt that depth controls were not needed on this module as most other modules have an attenuatable control input suitable for an LFO signal.

Different ranges of "rate" may be obtained by adjustment of selected components. Changing the value of R3 will alter the maximum rate attainable, with lower values producing a higher rate. The values of C1 and C2 can be altered such that larger values produce longer sweep times. The value of RV1 on the circuit diagram is 1M Lin, which provides a good range of sweep rates. However, the spread is a little uneven - ideally a 1M reverse Log pot should be used but these seem very difficult to obtain at the present time.

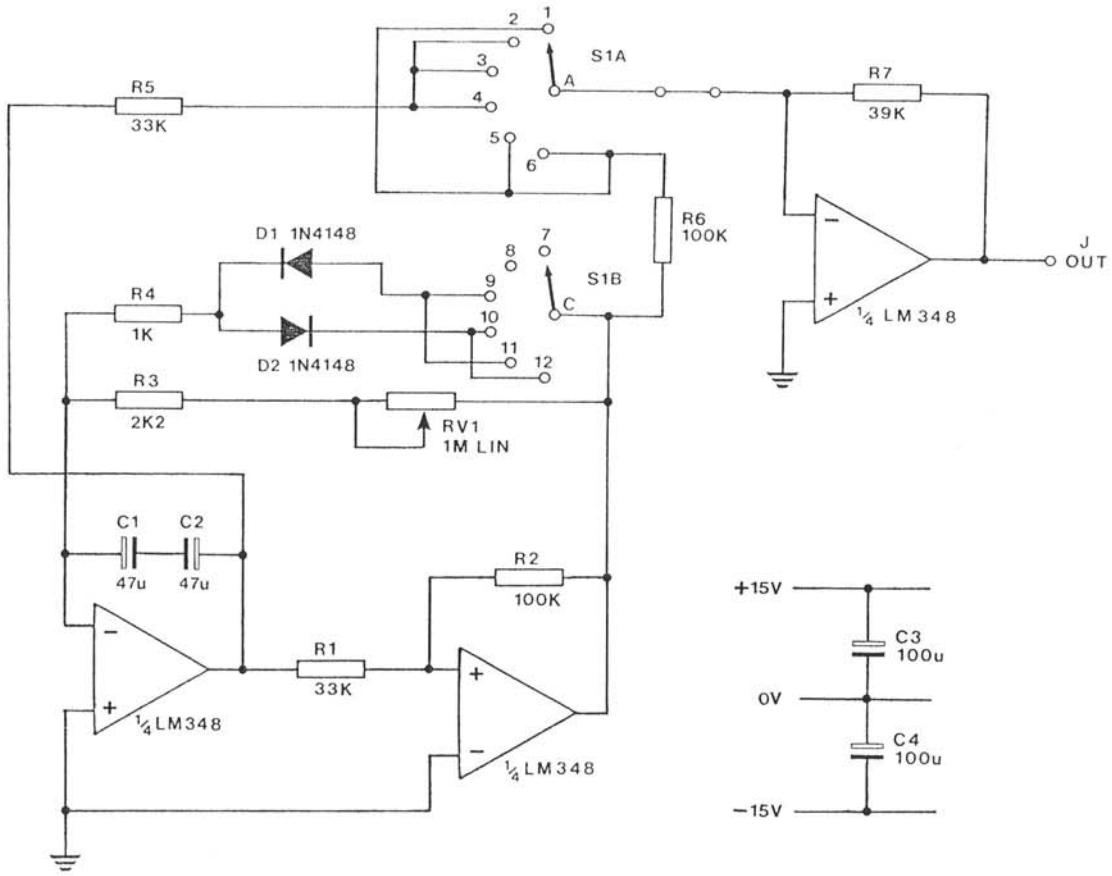


FIGURE 2. 80-23 QUAD LFO CIRCUIT DIAGRAM

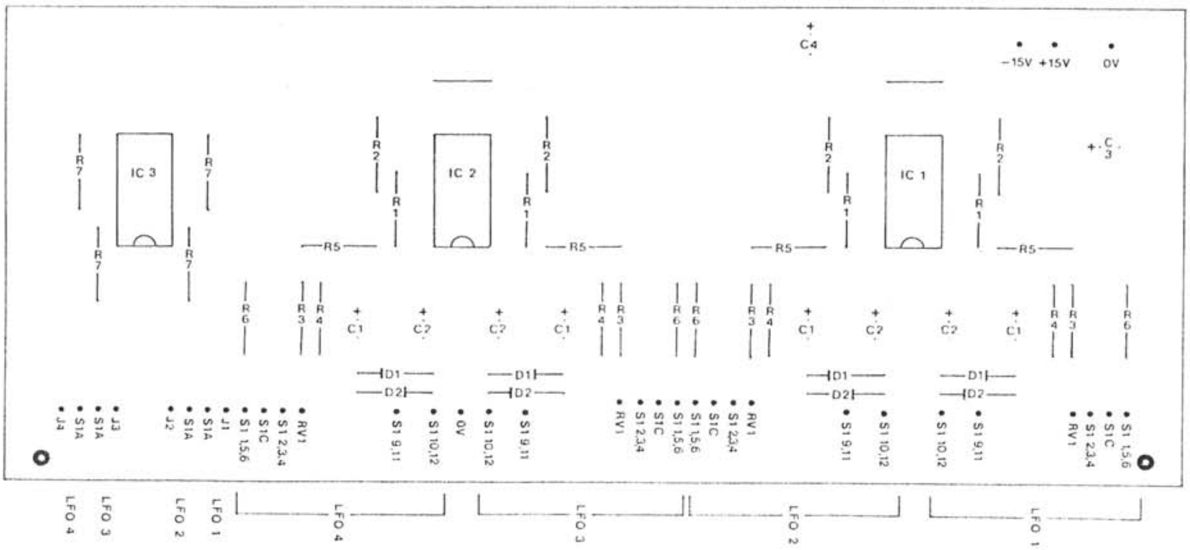


FIGURE 3. 80-23 PCB COMPONENT LAY-OUT

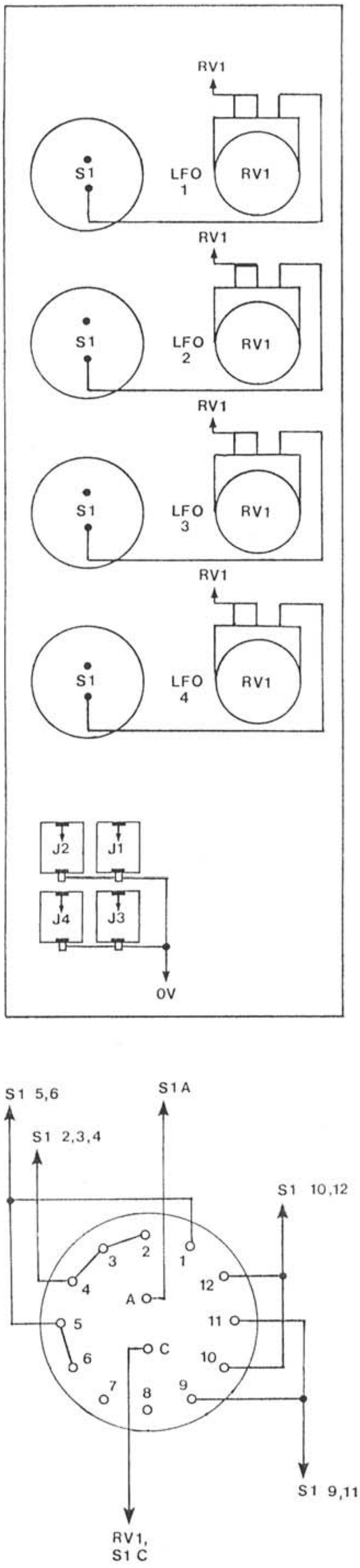


FIGURE 4. 80-23 PANEL/SWITCH WIRING

3. CONSTRUCTION

The PCB component layout is shown in Figure 3. Construction is very straightforward and should begin with the circuit board assembly in the normal manner. The poteniometers, rotary switches and output sockets may be mounted on a front panel as shown in Figure 1. The connections to be made between the front panel and the PCB are shown in Figure 4.

4. IN USE

The uses of an LFO within a modular synthesiser are many and varied. For instance, the four LFOs provided by this module may be connected to four different parameter inputs to produce a constantly varying sound. Where more than one input per parameter is provided, waveforms may be combined producing a less predictable source of modulation. Alternatively, a number of LFOs may be mixed (in a VCM) and their composite output used as a complex modulation signal. It should be noted that although the outputs are set at about +/-5V as opposed to 0V to +10V, most synthesiser functions can still be affected. Where both a control potentiometer and a CV input socket exist for one function, the former may be used as an adjustable bias control.

5. COMPONENTS

RESISTORS, 5%, 1/4w carbon film

R1,5 (x4)	33k
R2,6 (x4)	100k
R3 (x4)	2k2
R4 (x4)	1k
R7 (x4)	39k

POTENTIOMETERS, SWITCHES

RV1 (x4)	1M lin. rotary
S1 (x4)	2P6W rotary

CAPACITORS

C1,2 (x4)	47u PCB electrolytic
C3,4	100u PCB electrolytic

SEMICONDUCTORS

IC1,2,3	LM 348
D1,2 (x4)	1N 4148

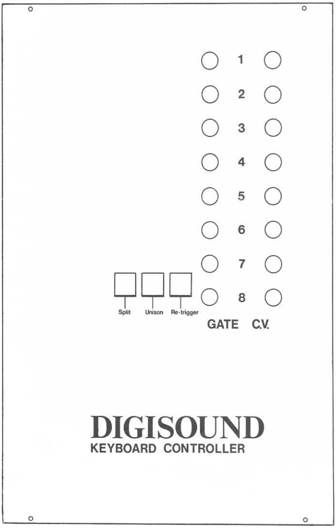
DIGISOUND 80-24

8 NOTE POLYPHONIC KEYBOARD CONTROLLER

1. INTRODUCTION.

The Digisound 80-24 unit is an 8 note polyphonic keyboard controller with separate CV and Gate outputs. It has a split keyboard facility with a fully variable split point and last note priority, together with a note allocation system that prevents erratic VCO swapping. The design incorporates a 16 note buffer to stop 'overflow' notes being lost.

The unit can be used with anything from 1 to 8 voices and the controller automatically detects which outputs are being used. There is also an 8 note Unison mode to aid tuning of the VCOs, and also to create rich sounds with all VCOs being played monophonically. In addition the unit has a switchable staccato / legato re-triggering function.



A +5V power supply is required for the controller at about 350 mA. Most of the components fit onto the main PCB, leaving just 16 3.5mm jack sockets to fit onto the two small connector PCBs which connect to the larger board. Therefore the only external wiring needed is two wires for the power supply, and a 16 way ribbon cable for connection to the keyboard.

The 80-24 Controller operates directly from the standard Digisound 80-15 keyboard set up using the keyboard PCBs and either GJ or GB gold contacts, so this polyphonic controller can replace the 80-15 D1, D2, D3 and E monophonic units used before, without any modification of the keyboard itself.

2. CONSTRUCTION.

The printed circuit board has a component overlay silk screened onto it to aid assembly. First of all, insert the track pins from side A of the board at the positions marked with a plain circle on the overlay, and solder both sides. Then insert all the diodes (not LEDs yet) and resistors; solder and crop. Please note that all the components are on side A of the board. Insert and solder IC sockets and all the capacitors ensuring the right polarity of the electrolytic ones. Special care should also be taken with R35-42 and C24-30 so that there is enough clearance for the jack sockets which mount on two separate PCBs; The capacitors will have to be bent over at 90 degrees for this, prior to soldering. Then, insert the push button switches making sure they are square to the PCB.

Prepare the two jack socket strips by soldering 8 jack sockets onto each and then solder a length of tinned copper wire into each of the vacant holes on the jack socket strips, crop on the soldered side and bend the free ends down sharply. Locate the wires from the socket strips onto the main PCB so that the strips are fully home, solder and crop the wires.

The two 8 pin single line connectors m be fitted to the 'row' and 'column' inputs on the PCB, or ribbon cable soldered directly to the board depending on your preferred method of fixture to the keyboard unit being used.

The LEDs should be fitted just prior to installing the front panel. D9-19 must all be inserted in their correct positions on the PCB. The panel can then be temporarily placed over the PCB with a few jack socket nuts securing it. The whole assembly should then be inverted, ensuring that all LEDs drop neatly into their respective front panel holes. Solder and crop the LEDs. The panel can then be removed ready for PCB inspection and calibration.

Check the assembly thoroughly and 1 . the ICs into their sockets and check orientation of all polarised components (i.e. electrolytic capacitors, diodes, LEDs, and ICs). Take anti-static precautions as many of the devices are static sensitive.

3. CALIBRATION.

To calibrate the unit, set unison mode and measure the voltage on the CV outputs with an accurate voltmeter (they should all give identical readings). Press the bottom C key and check that the output voltage is within 20mV of 0V. Press the C key 3 octaves up from bottom C, and adjust the preset (VR1) for 3.00V.

4. POLY KEYBOARD OPERATION.

The bottom note on the keyboard produces 0v, above which the scaling is 1v/octave. The Gate signal is 0v (untriggered) to +5v (triggered).

As soon as power is applied all LEDs will turn on for a short while, then all off. This action automatically resets the microprocessor and its associated circuitry so that the unit is ready for use.

The controller detects which outputs are in use by sensing if plugs are present in the Gate sockets. Output 1 is always assumed to be in use.

Notes are allocated to the outputs which are in use in ascending order.

If more keys are pressed than there are sockets in use, the pitch of the last key is fed to the right most output in use, replacing the pitch which is currently being output. If RETRIGGER is on, then the gate signal to that output is turned off for a fraction of a second to re-trigger the envelope shaper.

When a key is pressed, if that pitch has been output in the recent past, then it will be routed to the same output, even if lower order sockets are free. This prevents notes swapping to different VCOs when, for instance, the same chord is played repeatedly.

When the SPLIT function is turned on, the SPLIT LED will flash until a key is struck. This key will then be used as the split point. This key and all those above it will be routed to outputs 5-8 whereas the keys below it will be allocated to outputs 1-4. For the split function to operate usefully, at least one of the high order outputs (5-8) should be in use.

The UNISON function causes the keyboard to operate in monophonic mode where all the outputs follow the same single pitch, and all the gates operate together. SPLIT and UNISON may not be used simultaneously, and pressing either will turn the other off.

5. COMPONENTS.

Resistors - 1/4W 5%

R1	1M	x1
R2	1K2	x1
R3	4K7	x1
R4-19	10K	x16
R20-29,34,43	330R	x12
R30	56K	x1
R31,33	27K	x2
R32	6K8	x1
R35-42	82R	x8
VR1	10K Horiz Preset	x1

Capacitors

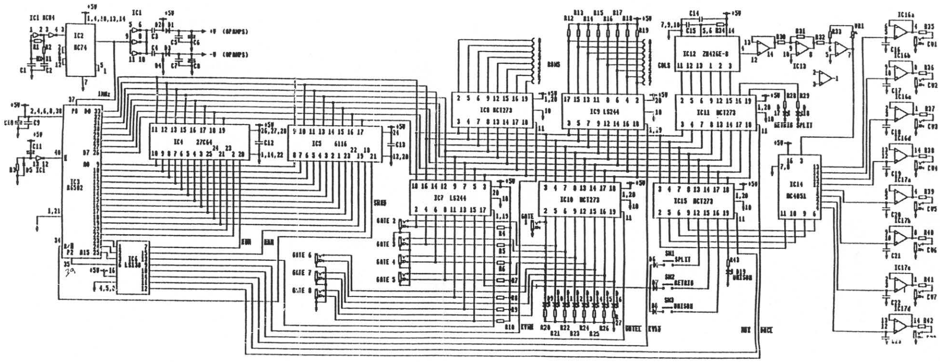
C1,2	33pf Ceramic	x2
C3-5,7,9,12-15	100nf Ceramic	x9
C6,8,10,11	47uf 16v PCB Elect.	x4
C16-31	22nf Polyester	x16

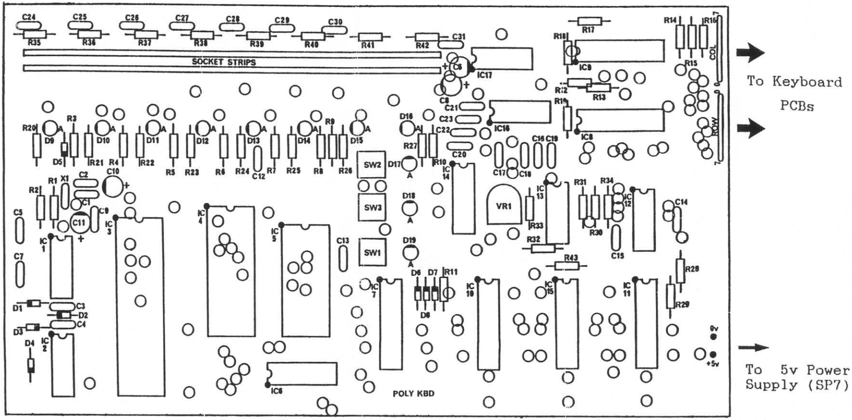
Semiconductors

X1	4MHz Crystal	x1
D1-8	1N4148	x8
D9-16	Red 3mm LED	x8
D17-19	Green 3mm LED	x3
IC1	74HC04	x1
IC2	74HC74	x1
IC3	R6502 Processor	x1
IC4	27C64 Prog. EPROM	x1
IC5	6116 RAM	x1
IC6	74LS138	x1
IC7,9	74LS244	x2
IC8,10,11,15	74HCT273	x4
IC12	ZN426E-8	x1
IC14	74HC4051	x1
IC13,16,17	TL064	x3

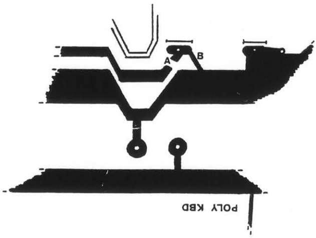
Miscellaneous

Track pins	x143
14 Way IC Socket	x6
16 Way IC Socket	x2
20 Way IC Socket	x6
24 Way IC Socket	x1
28 Way IC Socket	x1
40 Way IC Socket	x1
Push button Switch	x3
Switch Cap	x3
3.5mm PCB Jack Socket + Nuts	x16
Main PCB	x1
Socket PCB	x2
10 Way PCB Connector	x2
9" x 6" Front Panel	x1


IC3 R6502
IC4 27C64
IC5 6116
IC7 LS244
IC8 HCT273
IC9 LS244
IC10 HCT273
IC11 HCT273
IC12 ZN426E-8
IC14 HC4051
IC15 HCT273
IC2 HC74
ROWS
COLS
SPLIT
UNISON


To Keyboard
PCBs
To 5v Power
Supply (SP7)
SOCKET STRIPS
POLY KBD
COL
ROW
0v
+5v

Important PCB Modification.



Please make the following modification on the PCB track leading to pin 1 of IC 4 on the underneath of the board:

Cut the track at 'A'

and

Make a fresh link at 'B'

according to the above diagram.

MIDI-CV CONVERTER CONSTRUCTION NOTES

1. Insert all the track pins from side A of the PCB at the positions marked with a plain white circle on the PCB overlay, and solder both sides.

2. Insert diodes and resistors; solder and crop (NB. all components are mounted on side A of the PCB). Resistors R4-R11 also need to be soldered on the top side of the PCB at one end.

3. Insert and solder IC sockets. Some parts of some sockets may need slight trimming so that they fit neatly over the track pins.

4. Insert and solder capacitors, 4MHz crystal, and presets.

5. Insert and solder transistor and all sockets ensuring the MIDI sockets are screwed down before soldering. Bolt down the 32 way connector before soldering and ensure correct orientation.

6. Follow the 'modifications notes' regarding further assembly on the main PCB.

7. Insert and solder components and wire links onto the panel PCB. Bolt the 32 way right angle connector onto the foil/solder side of the panel PCB and carefully solder all the connections taking care not to short out any PCB tracks under the connector. This connector is the only component to be soldered to the underside of the panel PCB.

8. Insert and solder the rotary switch and push button switches ensuring correct alignment, and make up the socket PCB as in the 'modifications notes'.

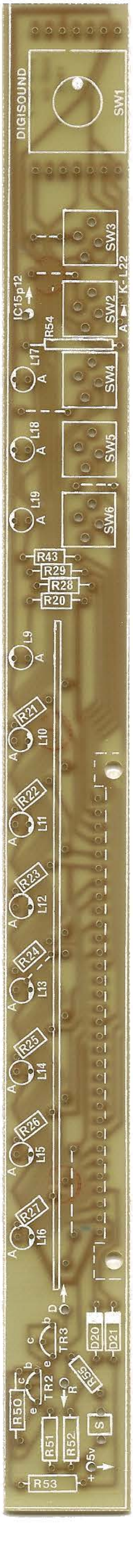
9. When complete the socket PCB should be soldered (foil side up - component side down) to the panel PCB via the bent over resistor leads and tinned wire connection.

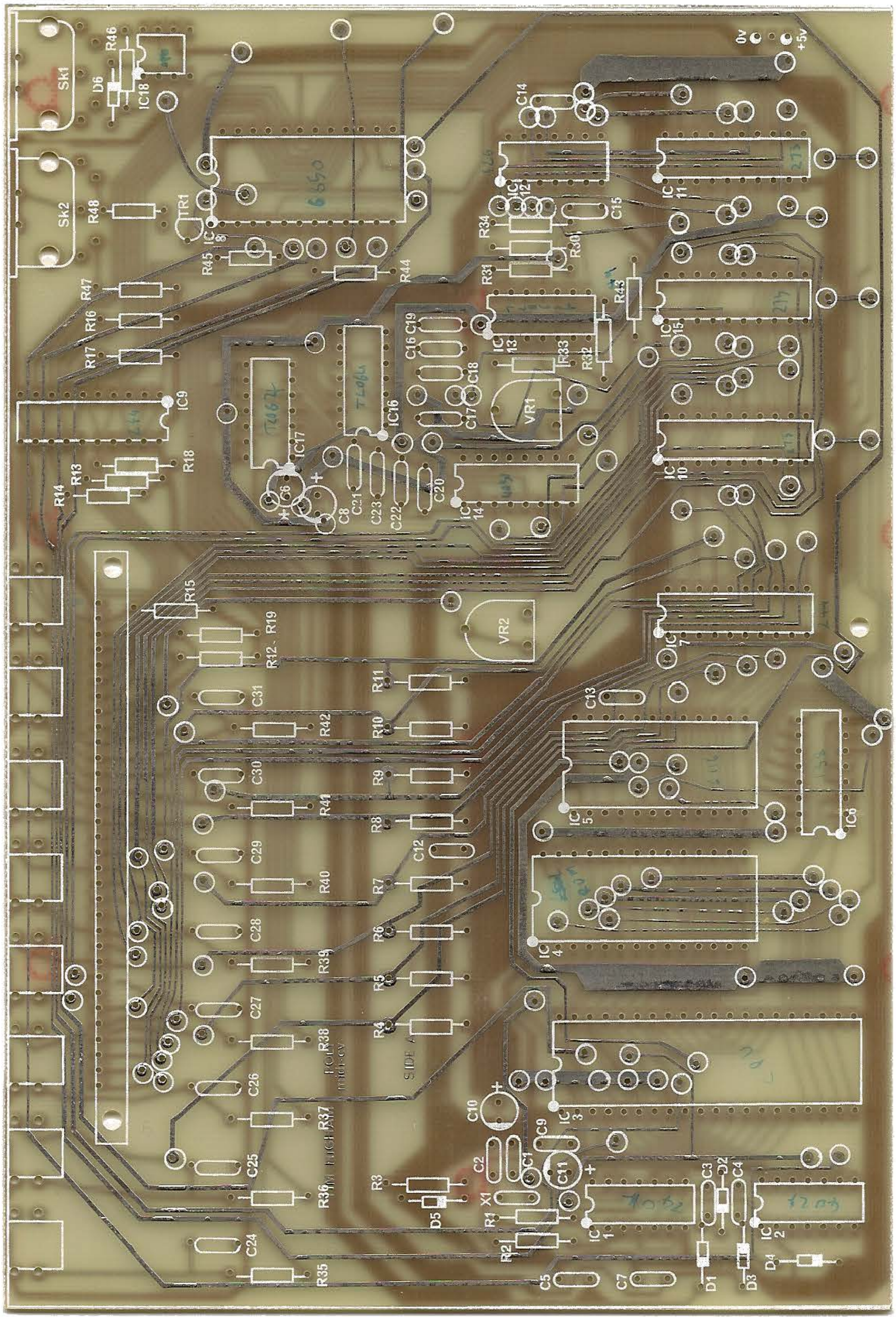
10. When all connections have been thoroughly checked and ICs mounted correctly the panel PCB can be placed on the main PCB via the 32 way connectors and all LEDs placed in their appropriate holes (LED anode 'A' is the longer lead), the entire assembly can be fixed to the front panel and securing nuts put on. Then align all LEDs flush with the front panel and solder them in.

11. CHECK EVERYTHING THOROUGHLY.

12. To calibrate: a) apply power, connect MIDI keyboard, select 'Unison' mode, and connect a volt meter to CV output 1. 0v should be read when the bottom note on the keyboard is played. Adjust the left hand preset VR1 to give 5v when the top note on a 5 octave keyboard is played. Check all outputs give the same reading in 'Unison' mode. b) VR2 should be adjusted so that no pitch change is heard when toggling the 'Bend' switch between the on and off positions.

NB. Take anti-static precautions with all ICs since most are static sensitive.


DIGISOUND
SW1
SW2
SW3
SW4
SW5
SW6
R43
R29
R28
R20
R21
R22
R23
R24
R25
R26
R27
R50
R51
R52
R53
R55
TR2
TR3
D20
D21


Sk1
Sk2
0v
+5v
VR1
VR2
TR1
SIDE A

MIDI-CV MODIFICATIONS NOTES

1. The mains rocker switch should be placed in the front panel with the wider spaced solder tags at the top and the mains supply going to the top of the switch. The two leads from the bottom tags should go to the mains transformer side of the SP7 5v supply as in diagram a. The use of heatshrink insulating tubing is recommended on all bare mains terminals within the unit. The mains earth lead (green and yellow striped wire) must be firmly soldered to the solder tag supplied and this bolted to the case ensuring good electrical connection.

2. Cut the PCB track leading from IC4 pin 1 and add a new link as in diagram b.

3. The two 1N4148 diodes and 4k7 resistor should be added to the main PCB so as to make up the circuit in diagram c. The two diodes must be connected in series between pin 16 of IC14 (4051) and the +5V supply rail. The resistor must be taken from the same point (pin16 IC14) to the +V supply found at pin 4 of any one of the TL064 ICs. Placing the components through the appropriate hole in the PCB with the unwanted track pin removed, will aid this procedure.

4. The 16 way rotary switch is supplied with a slightly melted hole in its base - this is due to the end stop pin being removed so that the switch becomes a continuous rotation version. The switched should be mounted so that the round notch on the shaft base is on the right as shown on the PCB. About 1cm of the shaft will of course need to be cut off so that the collet knob lies flush with the front panel and rotates smoothly.

5. Two short pieces of copper wire should be soldered into the holes marked LED 'L22' on the panel PCB and bent downwards. They can be soldered to the LED (below the 'Velocity/Mode' switch) after it has been aligned in the front panel hole. Care should be taken with the polarity of the LED - the longer lead is the anode.

Please read and understand all instruction notes before beginning construction.

6. A wire should be connected from the point marked 'D' on the panel PCB to the diode bus on the socket PCB. The diode bus and other components are shown in diagram e. Some form of insulation on the bent over resistor leads is required to stop them shorting out. On the 3.5mm GATE jack sockets only (on the small socket PCB), looking from the front of the socket with the pins pointing down, the front left pin should be bent inwards or cut off (see diagram d) so as not to be connected to the PCB.

7. A wire must be taken from IC15 (the middle 74HCT273 IC) pin 12 to the hole marked on the panel PCB 'IC15 pin 12'.

8. Connect a wire from IC1 (74HC04) pin 13 to the reset switch point marked 'R' on the panel PCB.

9. The reset button is best left until it is aligned in its front panel hole to the left of the 3.5mm sockets. It should then be carefully soldered onto the PCB, ensuring that it can be fully depressed with the red cap on the switch front. The solder tags on the switch will not protrude very much through the PCB.

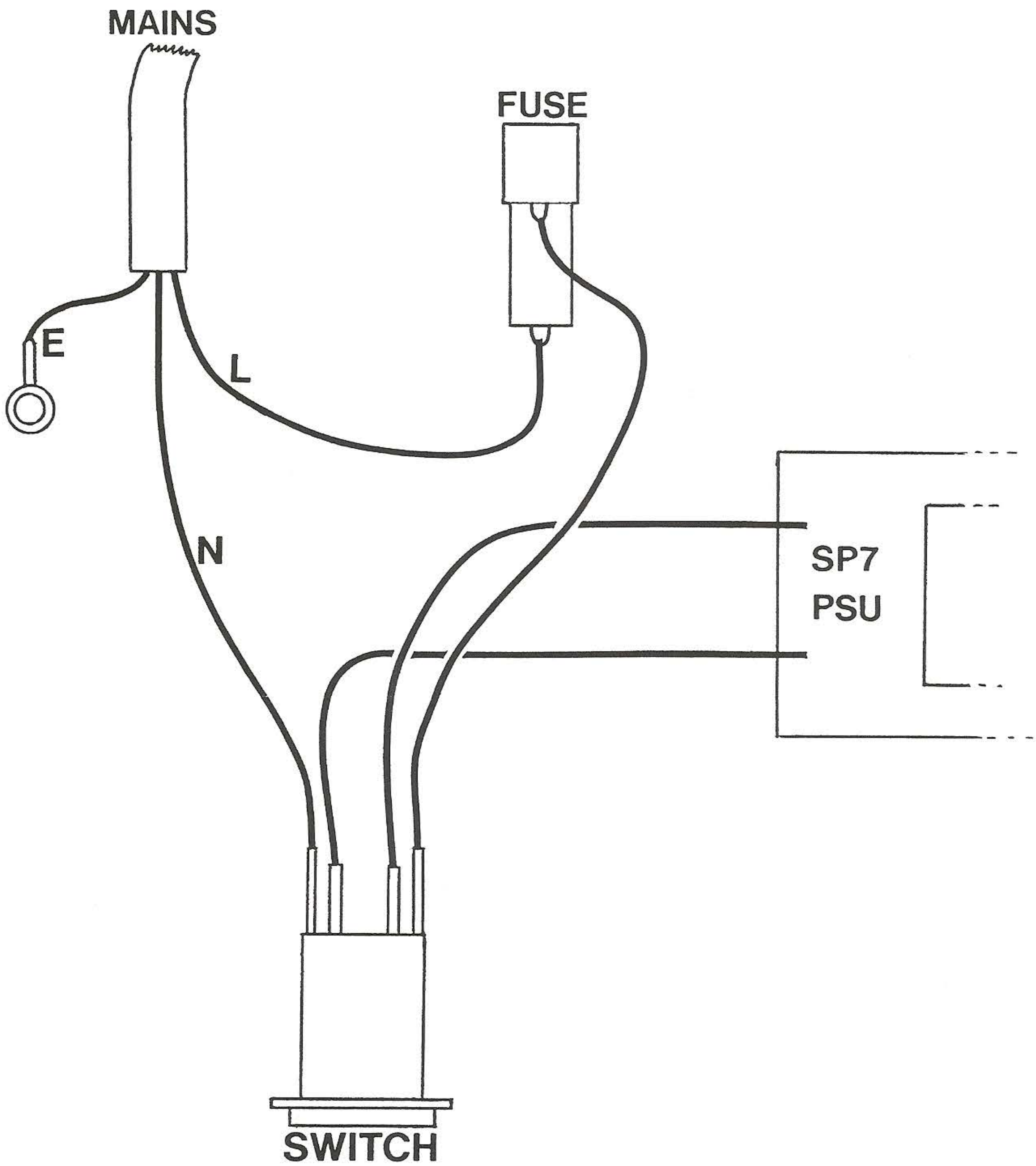
10. A piece of 2 way ribbon cable can be used to connect the 2nd MIDI IN socket to the rear of the unit if required. It is simply connected in parallel with the front panel PCB IN socket. The 2 wires are soldered to pins 4 and 5 of the MIDI socket (as numbered on the back - the middle left pin and the middle right pin). the 2 wires are then taken to the same pins on the PCB mounted MIDI IN socket. From the top of the PCB these are the two back-most connections, which again correspond to the middle left and the middle right pins on the socket.

11. Both PCBs should be mounted with the plastic cards underneath so that they are insulated from the case. The case being earthed with the solder tag and the mains earth lead.

12. R43 on main PCB should read R49.

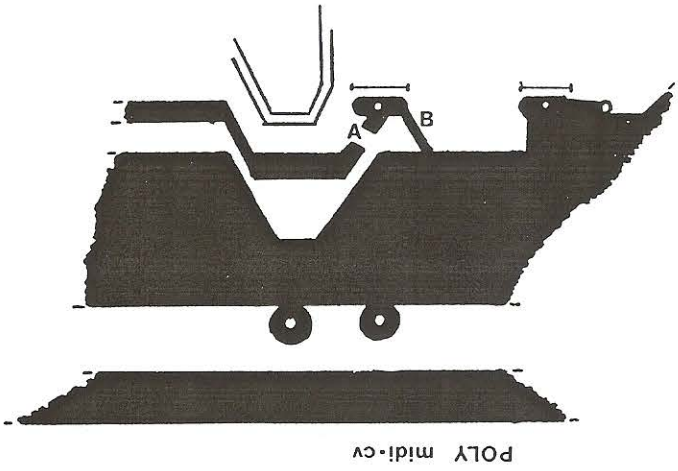
NB. You may wish to check correct operation of the unit before mounting on the front panel and case work. This is recommended since fault finding is difficult with the metal work attached. The LEDs will need to be lightly soldered in place for this, but will need to be desoldered to ease the front panel mounting and then soldered again flush with the panel as in the construction notes.

(a)



Important PCB Modification.

(b)



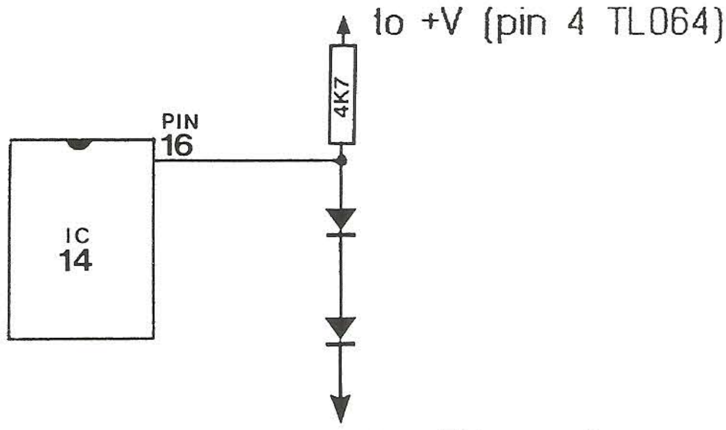
Please make the following modification
on the PCB track leading to pin 1 of
IC 4 on the underneath of the board:

Cut the track at 'A'

and

Make a fresh link at 'B'

according to the above diagram.



(c)

d

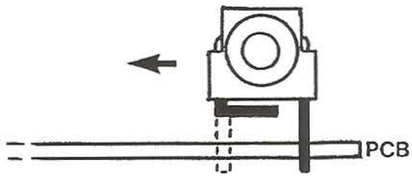


Diagram view from front of Gate sockets on socket PCB. Bend or cut off the front left pin only.

---

e

Diagram view from back with sockets removed for clarity. Solder only cathode (band) side and leave lead long to form the diode bus connecting all diodes to 'D' on panel PCB.

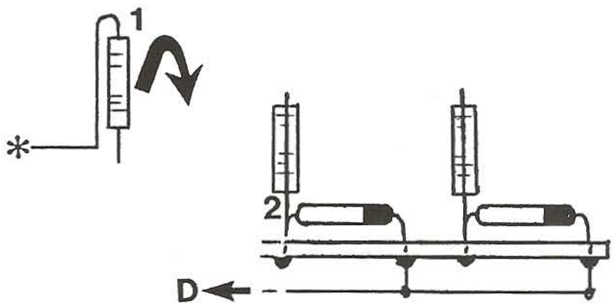


Diagram view from back again, form resistor as in 1, and solder in same hole as diode anode as in 2.

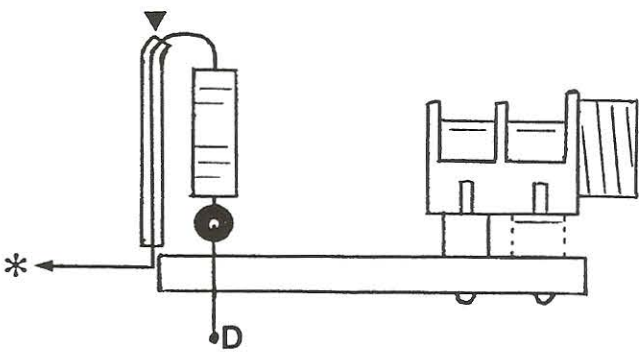


Diagram side view. Note insulating sleeving on resistor lead.▼

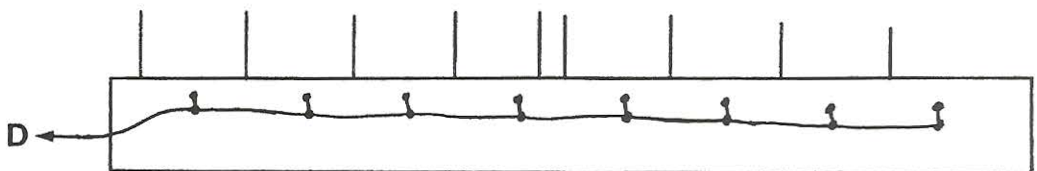


Diagram bottom view. Note diode bus bar.

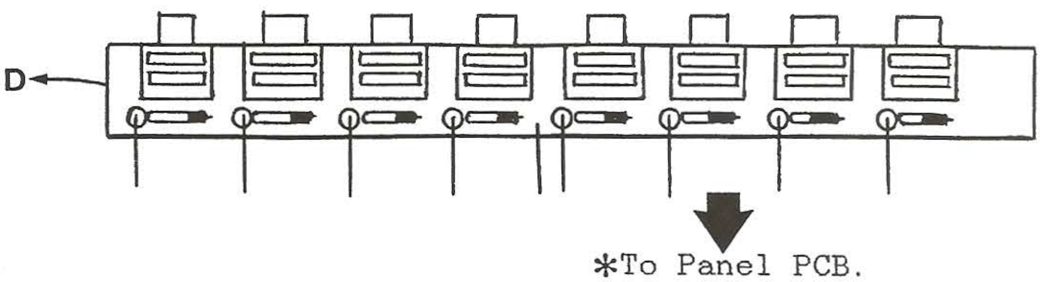


Diagram top view showing all the components and leads to be connected to panel PCB.

*To Panel PCB.

MIDI-CV COMPONENTS.

Resistors - 1/4W 5%

R1	1M	x1
R2,44,45	1K2	x3
R3	4K7	x1
R4-19	10K	x16
R34	330R	x1
R30,32	5K6	x2
R31	2K2	x1
R33	27K	x1
R35-42	82R	x8
R46-48	220R	x3
R49	68K	x1
VR1	10K Horiz. Preset	x1
VR2	22K Horiz. Preset	x1

Capacitors

C1,2	33pf Ceramic	x2
C3-5,7,9,12-15	100nf Ceramic	x9
C6,8,10,11	47uf 16V PCB Elect.	x4
C16-23	220nf Polyester	x8
C24-31	22nf Polyester	x8

Semiconductors

X1	4MHz Crystal	x1
D1-6	1N4148	x6
TR1	BC558	x1
IC1	74HC04	x1
IC2	74HC4024	x1
IC3	R6502 Processor	x1
IC4	27C64 Prog. EPROM	x1
IC5	6116 RAM	x1
IC6	74HCT138	x1
IC7,9	74HC244	x2
IC8	MC6850	x1
IC10,11,15	74HCT273	x3
IC12	ZN426E-8	x1
IC14	74HC4051	x1
IC13,16,17	TL064	x3
IC18	CNY173	x1

Miscellaneous

Track pins	x146
6 Way IC Socket	x1
14 Way IC Socket	x6
16 Way IC Socket	x2
20 Way IC Socket	x5
24 Way IC Socket	x2
28 Way IC Socket	x1
40 Way IC Socket	x1
32 Way PCB straight connector	x1
3.5mm PCB Jack socket + nuts	x8
5 pin PCB DIN socket	x2
Main PCB	x1

R20-29,43,54,55	330R 1/4W 5% Resistor	x13
R50-53	8K2	x4
R56-63	1K	x8
D9-16	3mm Green LED	x8
D17,19,22	3mm Red LED	x3
D18	3mm Yellow LED	x1
TR2,3	BC184L	x2
D20,21,23-30	1N4148	x10

Min.PCB Reset button	x1
" PCB button Red Cap	x1
32 WAY R/Angle PCB connector	x1
PCB Push button switch	x5
Push button cap	x5
16 WAY PCB Binary rotary switch	x1
Rotary knob	x1
3.5mm PCB Jack socket + nuts	x8
10mm sleeving	x11
Front Panel PCB Mk.2.	x1
Socket PCB	x1

6BA Bolt	x4
6BA Nut	x2
4BA Bolt	x5
4BA Nut	x10
4BA Washer	x5
Solder	x10m
Red thin wire	x1m
Black thin wire	x1m
19" Rack Case + printed front panel	x1 ea.

R1	4K7 1/4W 5% Resistor	x1
C1	2200uf 25V PCB Elect.	x1
C2	100nf Polyester	x1
C3	470nf Polyester	x1
IC1	7805 Regulator	x1
B1	2A Bridge	x1
T1	6VA, 2x6V Transformer	x1
F1	500mA quick blow fuse	x1
H1	Heatsink	x1

Miscellaneous

Panel mount fuseholder	x1
Mains cable	x3m
Mains plug with 2A fuse	x1 ea.
Solder	x1m
Mains rocker switch	x1
Conclamp	x1
Grommet	x1
Red medium wire	x1m
4BA Bolt	x1
4BA Nut	x2
4BA Washer	x2
4BA Solder tag	x1
Heatshrink tubing	x1/4m
Panel mount 5 pin DIN socket	x2
Purple thin wire	x1m
Orange thin wire	x1m
Cable tie	x1

Pitch Bend Mod.	1N4148 diode	x2
	4K7 resistor	x1

digisound

MODULE 80-C9 VOICE

1. INTRODUCTION

The DIGISOUND 80-C9 voice card provides the complete sound processing circuitry for a monophonic synthesiser voice. On-board design includes two VCOs, a four pole low pass VCF, two ADSR envelope generators and five VCAs which together comprise a relatively elaborate synthesiser voice. Some of the interconnections between the above circuit blocks are hardwired, but a degree of flexibility has been maintained by the use of electronic analogue switches. The routing of these enables the most popular synthesiser patches/effects to be selected. The circuit functions are controlled either by means of manual potentiometers which are disabled on insertion of a jack plug into the appropriate CV input socket or by independent potentiometers and external CV input sockets.

A schematic diagram of the voice card circuitry is shown in Figure 2. The

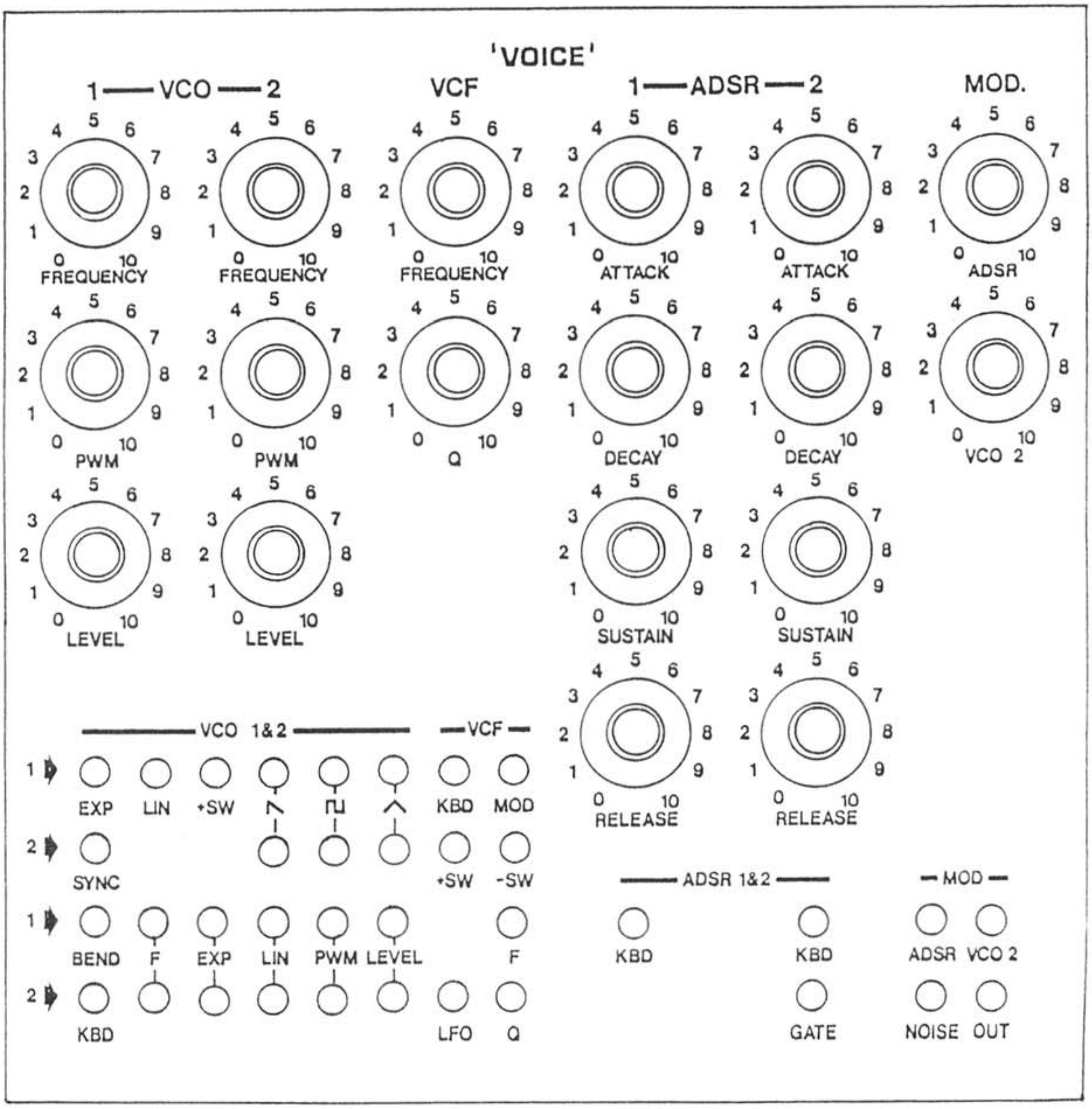


FIGURE 1. 80-C9 PANEL LAY-OUT

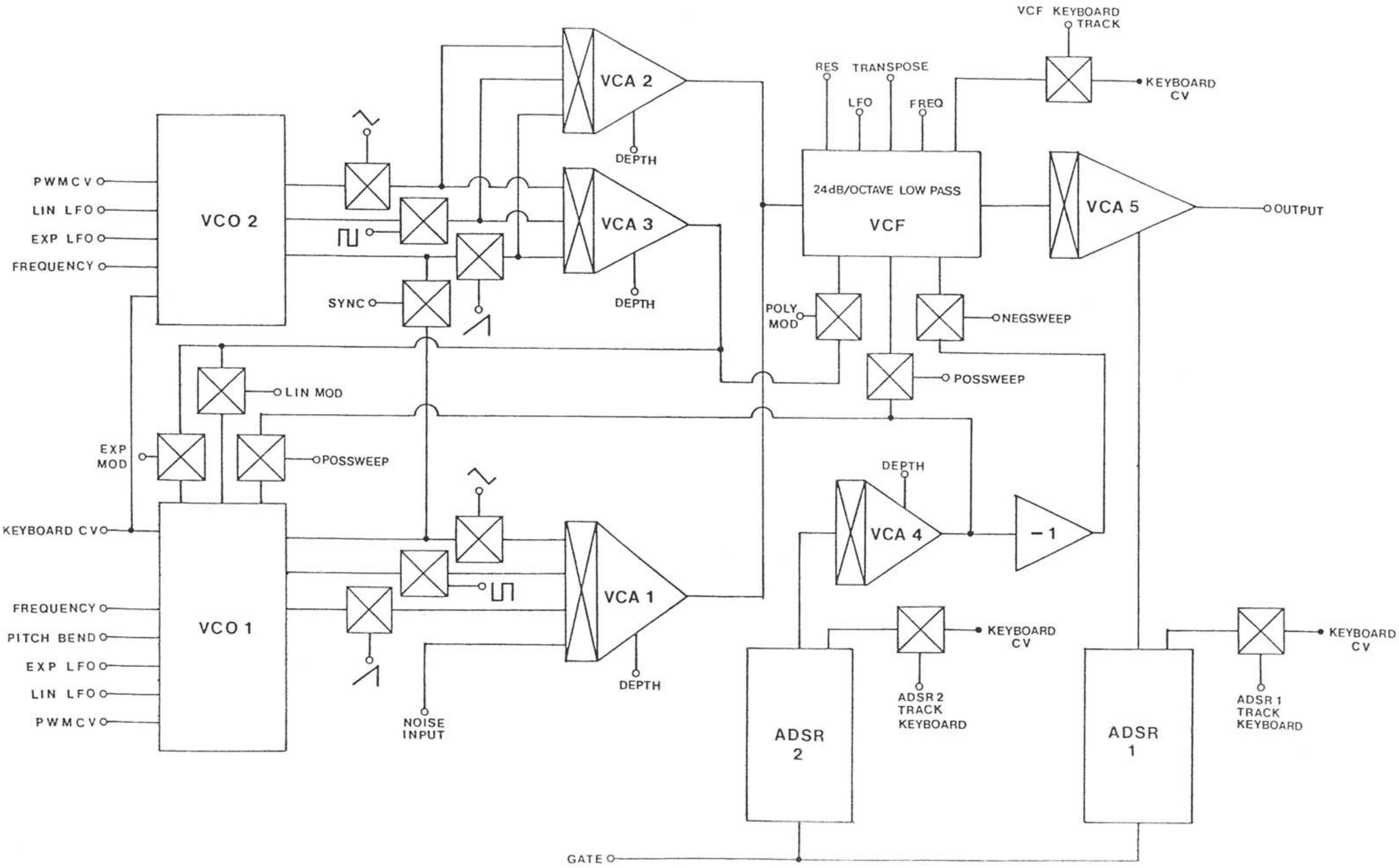


FIGURE 2. VOICE SCHEMATIC

design utilises two CEM 3310 VCTGs, two CEM 3340 VCOs, a CEM 3360 dual VCA and a CEM 3372 signal processor which together form a high quality voice requiring the minimum of calibration. An accurately scalable keyboard control input is provided and may be calibrated to the usual +1V/octave control voltage standard. Inputs are also provided to enable connection to external equipment such as LFOs, sample and hold networks and other control voltage generators. An audio input is also available, allowing an external noise source (or additional oscillators) to be mixed with the audio output of VCO 1.

2. DESIGN - GENERAL

The complete circuit diagram for the 80-C9 voice card is shown in Figure 3. The circuit design is centred around six custom music ICs from Curtis Electromusic Specialties. The use of these devices has enabled a highly compact dual VCO synthesiser voice to be built on a standard 100 x 200mm Eurocard.

The design utilises all electronic internal switching/patching (controlled by panel mounting SPDT sub min toggle switches S1 to S16) and is arranged for maximum flexibility. Connections to and from the PCB are grouped along one long edge of the board and may be effected either by hardwiring or by the use of Molex connectors and a small motherboard. The first of these options will be discussed here. There are 57 connections in all which may be subdivided into the following groups:- control voltage inputs (26), electronic analogue switch control lines (16), audio inputs (1), audio outputs (1), gate inputs (1), power supply connections (5) and reserved for future expansion (7). All the control inputs provided will accept voltages in the range -5V to +5V (see Table 1) making interfacing to micro-computers far simpler.

A. VCO 1 & 2

Both VCOs are based upon the CEM 3340 and are configured in a similar manner. For this reason detailed analysis of only one of these oscillator blocks is necessary, with any differences being mentioned at the relevant point. The main frequency control input is at pin 15, configured as a summing stage thus allowing multiple independent frequency control. In this application, resistors R1-8 effect the required voltage control with two on-board CV inputs and four independent CV inputs for connection to external equipment.
The on-board CV inputs are:-
a) Possweep - ADSR output via VCA and to CEM 3340 via R4
b) Internal exponential modulation - from VCO 2 via a separate VCA.
The other inputs are:-
a) Frequency - via R1 to effect tuning of the oscillator
b) Keyboard CV - via R6 for connection to a musical keyboard (commoned to both VCOs)
c) Pitch bend - allows detuning of VCO 1 or connection to joystick controllers etc.
d) Exponential LFO - for connection to an external LFO whose output will vary frequency in an exponential manner
Pin 14 of the CEM 3340 is used as a scaling factor. Since the current gain of the internal multiplier is set near unity, 100K input resistors and a 1K8 scaling resistor (R16) produce the standard 1V/octave response and about 18mV at the base of Q1 (internal to IC). R8 and RV19 together with an incoming reference voltage (in this case +15V) set the initial frequency of the oscillator and have been chosen such that with no external voltage applied the frequency may be adjusted to 65.406 Hz (the lowest note of a four octave keyboard).

For greatest accuracy of the internal multiplier the current out of pin 2 should be close to that out of pin 1. This balance is achieved via RV20.

The exponential generator in the CEM 3340 is capable of delivering a current for charging and discharging the timing capacitor (C4) from greater than 500uA to less than the input bias current of the buffer which gives a typical frequency range of 500000:1. For synthesiser applications the most accurate portion of this range (50-100uA) is used. With a 1nF timing

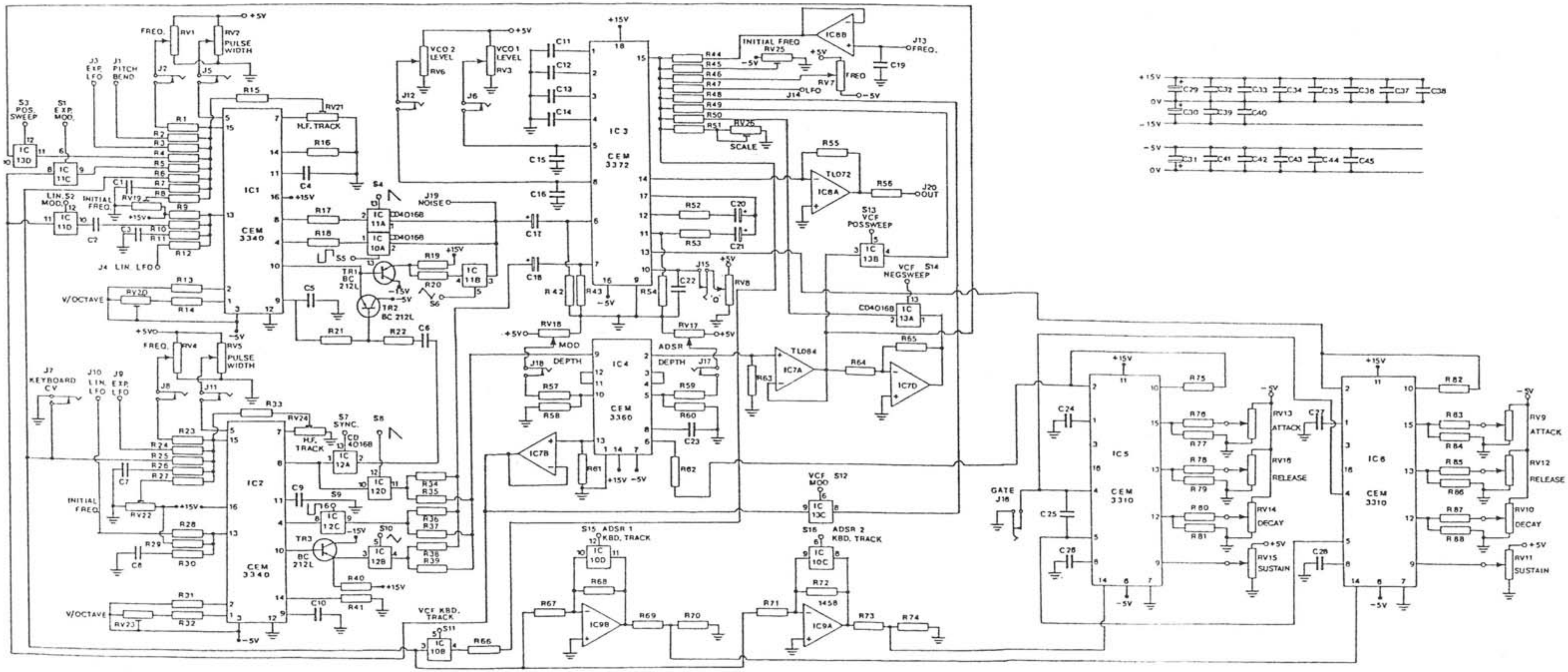


FIGURE 3. 80-C9 VOICE CIRCUIT DIAGRAM

capacitor and a 15V positive supply the most accurate range is therefore between 5Hz and 10KHz.

For accurate operation of the exponential generator in the CEM 3340 a reference voltage is required. In this application the +15V supply rail is suitable, providing the PSU is well regulated (as in Digisound modules 80-1/80-1A). This is connected to pins 15 (via RV19 & R8) and 13 (via R9). In the former case this provides a stable initial frequency while the latter establishes a reference current for the exponential generator which is derived via R9. Pin 13 is also available for use as a linear frequency control input with R12 present for connection to an external LFO. R12 produces a 10% change in frequency per volt at this input.

Similarly, the inclusion of R10 and the associated electronic switch (IC 11D) connects pin 13 to the output of VCO 2 in the same way (and to the same modulation bus) as the internal exponential modulation input.

It should be noted that if the output from a suitable LFO is not capacitively AC coupled, any DC offset present will cause detuning of the oscillators. Also, a negative current at pin 13 which is in excess of the reference current will gate the oscillator off.

One of the biggest drawbacks of exponential VCOs has been their temperature sensitivity which results in frequency drift as the device warms up. The CEM 3340 however incorporates temperature compensating circuitry. This is achieved by multiplying the current sourced into the control pin (pin 15) by a coefficient directly proportional to the absolute temperature. This coefficient is produced by the "Tempco Generator" and the cancellation is nearly perfect.

Included as part of the CEM 3340 is a method of overcoming high frequency tracking error, which is an effect resulting in a slight flattening of the frequency at the extreme high end of the audio range. This effect is caused by the bulk emitter resistance within the transistor based exponential generator circuitry becoming significant as current, and therefore frequency, is increased. To compensate for this undesirable effect, pin 7 outputs a current which is one quarter of the exponential generator current. This is converted into a voltage across RV21 (RV24 for VCO 2), a proportion of which is fed back to the frequency control input at pin 15 via R15 (R33 for VCO 2). Thus, as the frequency is increased this feedback voltage will sharpen the control voltage scale, but only at the high end of the frequency spectrum. This produces a reliable voltage to frequency conversion over the entire audio range.

Three simultaneously available waveform outputs are produced by the CEM 3340, namely sawtooth, pulse/square and triangle at pins 8, 4 and 10 respectively. The sawtooth and pulse/square outputs pass via R17 and R18 to CMOS analogue switches (IC 11A & IC 10A) for later mixing at the input of the VCF. However, the triangle output has a finite resistance and requires buffering to maintain VCO performance. In the worst case, neglect of this buffer may result in a lowering of the frequency due to loading effects. The buffer used is a single NPN transistor (TR1), which passes the triangle wave output via R20 to a third analogue switch (IC 11B).

Pulse width control is achieved by direct injection of a 0V to +5V analogue voltage to pin 5 allowing pulse width to be varied from 0 to 100% values of mark/space ratio, via RV2 or an external CV into J5.

The main difference between the two VCOs is the sync. arrangement. In this design VCO 1 becomes the "slave" oscillator and VCO 2 the "master". The method of sync. used (the CEM 3340 permits two types) is soft synchronisation which on selection causes the triangle upper peak to reverse direction prematurely with the result that the oscillation period is an integral multiple of the pulse period. This sync. technique is achieved by closing analogue switch IC 12A, thus supplying negative pulses to pin 9 of the "slave" oscillator (VCO 1). Capacitors C5 (VCO 1) and C10 (VCO 2) prevent unwanted sync. or waveform instability from any voltage spikes on the positive supply line.

The only other difference between the external circuitry of the two VCOs is that VCO 2 allows less resistive mixing of control voltage inputs namely possweep, int. exp. mod., int. lin. mod. (all three being internal to the PCB and derived from the output of VCO 2) and external pitch bend.

Power supply to both CEM 3340s is a stable +15V to pin 16 and -5V to pin 3. The use of well regulated supplies is essential to maintain oscillator performance.

B.VCF & VCA

The VCF in this design is based upon the recently available CEM 3372 signal processor. The heart of this IC is a four pole low pass voltage controllable filter employing open-loop design to give an enhanced "rich" quality to the processed input. The filter response is of the Butterworth type with a sharp 24dB/octave roll-off characteristic, ideal for electronic music applications. Internal to the CEM 3372 is a VCA to allow overall signal feedback, and hence resonance or "Q" value to be voltage controlled. The passband gain remains constant as the resonance is varied and this eliminates the drop in volume as resonance increases, commonly found in this type of filter. The CEM 3372 also features low noise, low control feedthrough and temperature compensated transconductors for cut-off frequency stability.

Audio input to the VCF has been made more versatile with the inclusion of a two channel voltage controlled mixer with independent control voltage inputs for each channel (pins 5 & 8 for channels 1 & 2 respectively). The two audio inputs (pins 6 & 7) are suitable for use as signal summing nodes and in this design mix the three outputs of both VCOs.

Also included on the chip is a VCA of the current in, current out type allowing ready mixing of multiple inputs. In this design the input to the VCA is direct from the output of the VCF so no mixing is required. The VCA also features extremely low noise and exceptionally low control feed-through without the need for additional trimmers. This makes it ideal for control by fast transient waveforms such as those produced by an ADSR envelope generator.

In this design the two channels of the voltage controllable input mixer are each fed by the three electronically switched outputs from the CEM 3340s. It is at this point that the attenuation of the output from both VCOs takes place in order to present their respective channel inputs with a total signal amplitude of approximately 80mV peak-to-peak. This is necessary to maintain low distortion, typically 5% THD. Attenuation on channel 1 is via resistors R17, R18 and R20 forming a potential divider network with R42. Channel 2 attenuation is via resistors R34, R36, R38 and R43. In addition, but on channel 1 only, there is a noise input to the VCF allowing a noise source to be mixed into the audio bus of the voice card. The two composite signal sources first pass through capacitors C17 and C18 (channels 1 and 2 respectively) in order to block any DC offset present on either of the VCO outputs. The signal levels of channels 1 and 2 are controlled by positive going control voltages (0V to +5V) to pins 5 and 8 respectively. Capacitors C15 and C16 shunt any AC voltage to ground.

Within the CEM 3372 the four independent filter stages are hard-wired to low pass configuration. Filter capacitors C11 - C14 are chosen such that the filter frequency range covers the entire audio range (less than 20Hz to greater than 20KHz) whilst control of the cut-off frequency is by means of voltage control at pin 15. In order that multiple voltage sources may simultaneously adjust the filter frequency, resistors R44 to R51 effect a summing stage. The voltage to frequency scale is adjustable to precisely +1V/octave by the combination of RV26 and R51. Provision has been made for connection to an external LFO via R47. Transposition of filter frequency by potentiometer RV7 is possible via R46. An initial cut-off frequency may be set up via RV25 and R45 such that with no frequency applied to pin 15 the VCF passes no audible frequencies (individual circumstances will dictate the most useful position of this trimmer). A suitable connection to an external CV jack socket (J13) may be made via R44 which is preceded by a standard sample-and-hold network (IC 8B and C19) included for future expansion. Resistor R48 is hardwired to the internal modulation bus which is sourced by VCO 2, thus implementing an internal VCLFO function. Resistors R49 and R50 carry the ADSR signal from the "sweep" bus, the former connection being positive going (hence possweep) and the latter being an inverted version of this. These two control voltages are selected by control of analogue switches IC 13B and IC 13A. If both are selected no change in filter frequency will occur.

The signal output of the VCF (pin 17) is routed via capacitors C20 and C21 (which provide a DC blocking action) to pins 11 and 12. Pin 11 is the input to the signal regeneration (resonance) VCA, control of which is by external voltage (0V to +5V via RV8 or external CV into J15) to pin 10 (capacitor C22 shunts AC to ground). This network allows resonance to be varied from a Q factor of 0.7 up to oscillation. Pin 12 is the signal input to the fixed VCA, the gain control of which is buffered, brought out to pin 13 and hardwired to the output of ADSR 1. The output of the fixed VCA is at pin 14 of the CEM 3372 and passes via a low noise BI-FET op amp to J20, the audio output.

C. ADSR 1&2

The production of suitable control envelopes is achieved by the use of two CEM 3310 envelope generators. In this design both ICs are gated simultaneously from a common gate input at pin 4. This gate voltage should be positive going and between 3V and 15V. The gate input jack socket (J16) is wired such that with no plug inserted the gate input is held at 0V. If this connection is omitted the gate input will float high. The CEM 3310 is equipped with separate gate and trigger inputs such that during the gate on/sustain period (ie after A,D) a positive voltage to the trigger input, pin 5, will restart the initial attack and decay cycles. This feature has not been used in this design and pins 4 and 5 are capacitively coupled together by C25 producing a differential gate pulse to the trigger input.

Pins 15,12,9 and 13 permit voltage control of attack, decay, sustain and release respectively. The attack, decay and release inputs are suitably attenuated by six potential divider networks formed around resistors R76 - R81 and R83 - R88, all of which are part of encapsulated SIL resistor network packages. These three inputs require negative going control voltages between 0V and -5V, whereas the sustain input requires a control voltage between 0V and +5V. In all cases the greater the deviation from 0V, the larger the A,D,S or R contour produced.

ADSR 1 is hardwired to control the final VCA (internal to the CEM 3372) whilst ADSR 2 controls the frequencies of VCO 1 and the VCF. The latter signal is first processed by a VCA, the depth (and therefore ADSR amplitude) of which is controlled by either potentiometer RV17 or an external CV via J17 (0 to +5V). This attenuatable ADSR signal is subsequently buffered by IC 7A and split so that two electronic switches allow the signal to alter both the pitch of VCO 1 and the cut-off frequency of the VCF. This same signal from IC 7A is also inverted by IC 7D providing an inverted sweep to change the VCF cut-off frequency. These three effects are selectable by closure of the relevant analogue switches whose controls are situated at S3(possweep VCO), S14(negsweep VCF) and S13(possweep VCF).

Simultaneous control of the three time constants (A, D and R) is possible by means of injecting a control voltage at pin 14 of the CEM 3310. In this design these control voltages are derived from the keyboard control voltage (J7) using the two op amps within IC 9. On closure of the two electronic switches IC 10C and IC 10D, feedback resistors R68 and R72 are effectively shorted out and thus the keyboard track ADSR functions are disabled. It can therefore be seen hat normally these control inputs will be at +15V and the effects may be selected by grounding these switch control inputs with toggle switches S15 and S16 (which are therefore wired in a reverse manner compared to the other 14 switches). Use of the keyboard CV to alter the A, D and R time constants produces larger envelopes at the lower end of the keyboard and short, staccato-like envelopes at the higher end. This is a common feature of many conventional instruments such as the piano.

Power to the CEM 3310s is +15V and -5V at pins 11 and 6 respectively.

).ADSR & MODULATION VCAs

As already mentioned, the output from ADSR 2 is passed through a VCA to allow attenuation of envelope amplitude. This VCA is one of a pair of independent VCAs available within the CEM 3360. The two VCAs are of the current in/current out type with both linear and exponential control of gain over greater than a 100dB range. The VCAs feature very low noise and wide bandwidth, making them ideal for precise signal processing.

The ADSR output is fed via current converting resistor R62 to pin 6 of IC 4 (signal input to VCA 1) and is vailable again at pin 2 (signal output). Control of signal amplitude is possible by positive going (0V to +5V) CV to pin 5 which allows access to the internal logarithmic converter. Thus an incoming linear control voltage (in this case potentially divided by resistors R59 and R60) will modify gain in an exponential manner. The output then passes to buffer IC 7A and inverter IC 7D as previously described.

The second VCA is utilised for modulation attenuation. Modulation waveforms are produced by VCO 2 and resistively mixed by resistors R35, R37 and R39 into pin 9 of IC 4, the modulation waveshape being selectable y IC 12B, IC 12C and IC 12D. Exponential control is at pin 10 (again potentially divided by resistors R57 and R58) by means of a control voltage ranging from 0V to +5V. The output is at pin 13 and is buffered by IC 7B before being bussed to the exponential and linear frequency control inputs of VCO 1 and the cut-off frequency control input of the VCF. These three routes are switchable by IC 11C, IC 11D and IC 13C respectively.

3. CONSTRUCTION

The 80-C9 voice card has been engineered to be built on a double sided 100 x 200mm PCB, the two sides being linked by track pins. It will be seen that component placement is extremely dense and for this reason extra care must be taken in assembly. Because of this high component density, groups of copper tracks run parallel to each other with minimal bare fibre-glass between them, thus providing increased opportunity for solder splashes to bridge two or more tracks. Therefore careful regulation of the amount of solder per joint is recommended, particularly for the track pins. If too little solder is used the track pin may not electrically connect to the component side, but if too much solder is used one risks bridging two track pins together or to a top track. The possibility of under soldering a joint may be alleviated by applying the solder between pin and track such that when heat is applied the solder seeps round under the pin rather than forming a useless "blob" on top of the track pin. Examination of each joint with a magnifying glass is strongly recommended.

Components are best assembled onto the board in the following order:-
a) Track pins
b) Resistors
c) IC Sockets
d) SIL Resistors
e) Trimmers
f) Capacitors
g) Transistors
h) ICs

Track pins should be located in <u>all</u> component side holes that terminate in a solder pad; any other holes are for normal components. Push the track pin firmly through the hole and snap off

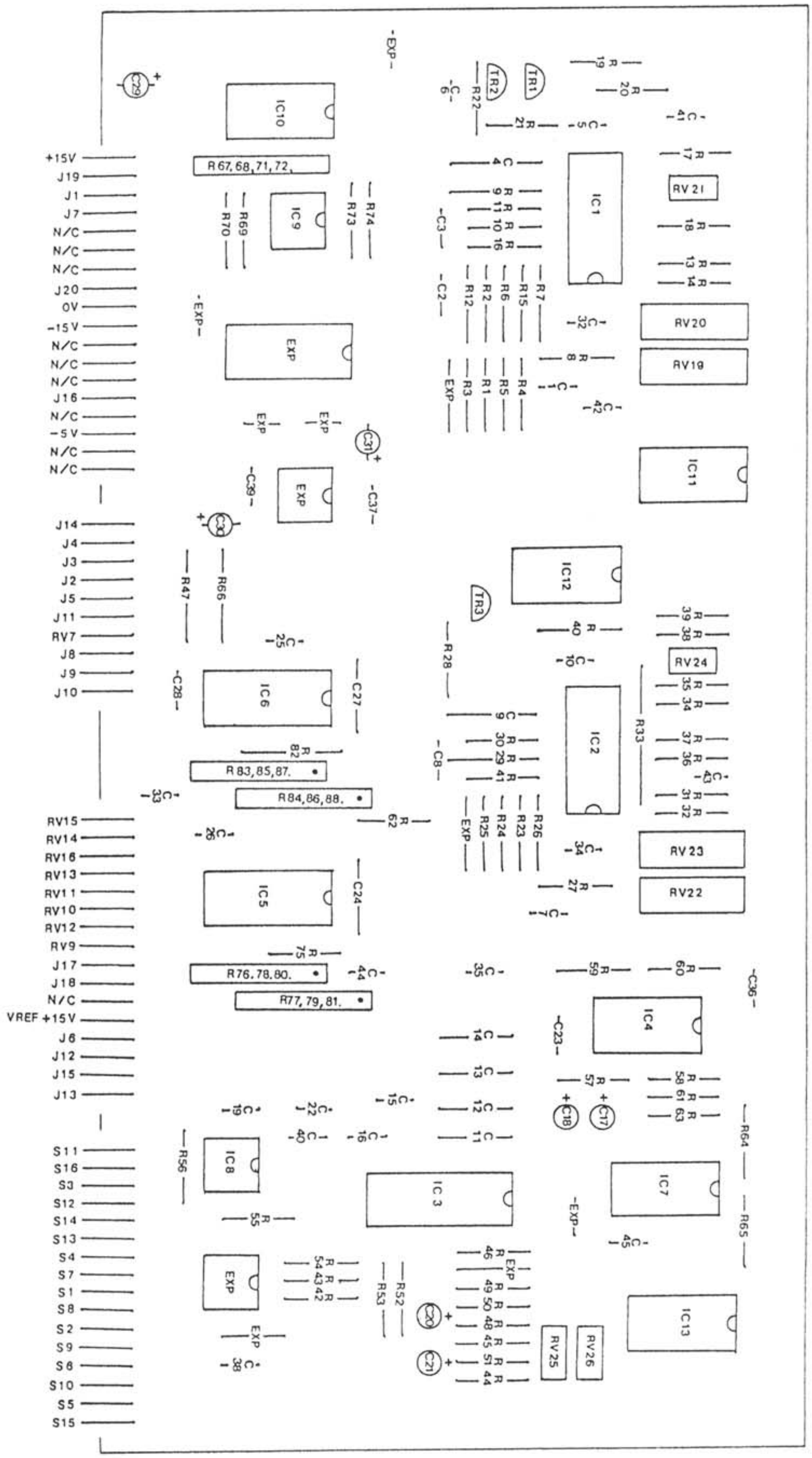


FIGURE 4. 80-C9 PCB COMPONENT LAY-OUT

from the strip of pins provided. Solder the top side of all pins first and then solder the underside, being careful to examine each joint (particularly on the component side).

Having pinned through the entire board, select and preform the resistors. Resistor leads should be bent such that no excess lead is evident, as on the component side this could give rise to a dry short with a nearby length of track. Component placement is detailed in Figure 4 and should be adhered to for all listed components. After installation of the components, the use of a PCB solvent cleaner is strongly recommended. This should be applied liberally with a fine real bristle paint brush, and removes all residual flux which may degrade circuit performance or even produce a short circuit.

The voice card PCB design is such that many options are available to suit individual users. As already mentioned, this board together with a keyboard and keyboard circuitry may be utilised as a complete monophonic synthesiser. It could also be one of several voices in a complex (possibly computer programmable) polyphonic system. The wiring for these two options is quite dissimilar, the main difference being the routing of CV inputs to either external jack sockets and potentiometers or to micro-processor based interface circuitry with a common bus. These notes concentrate on the former option.

The simplest option for using a voice card as a synthesiser in its own right requires most CV inputs to be hard-wired to potentiometers with the remainder being fed by dedicated LFOs or brought out of circuit to jack socket connectors. This type of adaptation should prove simple for most constructors and Figure 5 shows the wiring necessary to connect the PCB to the 9 x 9 inch panel supplied by Digisound Ltd. Constructors wishing to make use of this design in their own format should refer to Figures 4 and 5 and Table 1, from which it is possible to extrapolate all relevant wiring information.

To aid construction and connections to panel hardware, a PCB mounting bracket is used. This bracket is secured to the front panel by means of potentiometers RV3, RV6, RV11 & RV15 and the PCB is held in place by two PCB slides. The bracket is used in such a manner that the plastic-coated side is face up and parallel to the PCB, thus providing added insulation in the unlikely event that the PCB and bracket should touch. This method ensures that the wires between the PCB and externally mounted components remain as short as possible. Connections from the PCB may be made in one of two ways; either single sided terminal pins or standard Molex plug and socket arrangements. The former of these options effectively "hardwires" the PCB to the panel, while the latter facilitates its removal. Whichever method is chosen, a neat physical appearance may be achieved using either 1/0.6 single strand wire or multistrand 7/0.2 wire harnessed by a number of light-duty cable ties. Connections to the switches and 3.5mm jack sockets are best achieved by passing the appropriate wires through the two holes in the mounting bracket provided for this purpose. Before the toggle switches S1-S16 are wired up it is necessary to install a 4k7 pull down resistor (R89 - R104) on each one. Each resistor has one lead connected to 0V and the other to the switch pole i.e. the appropriate "S" point.

Panel wiring is arranged in such a manner that control potentiometers RV1, RV2, RV3, RV4, RV5, RV6, RV8, RV17 & RV18 are disabled after insertion of a jack plug into the appropriate socket, thus providing the facility for <u>either</u> potentiometer <u>or</u> external voltage control. Control of the functions of both envelope generators is by means of potentiometers only (RV9 - RV16), while VCF frequency may be controlled by both RV7 <u>and</u> externally via J13. Therefore the wipers of potentiometers RV7 and RV9-RV16 go directly to the appropriately marked points on the PCB, while all other connections from the PCB link to jack sockets (J1 - J20) and switches (S1 - S16).

As previously mentioned, the PCB is fixed onto the mounting bracket by two self-adhesive PCB slides. If terminal pins have been fitted it is

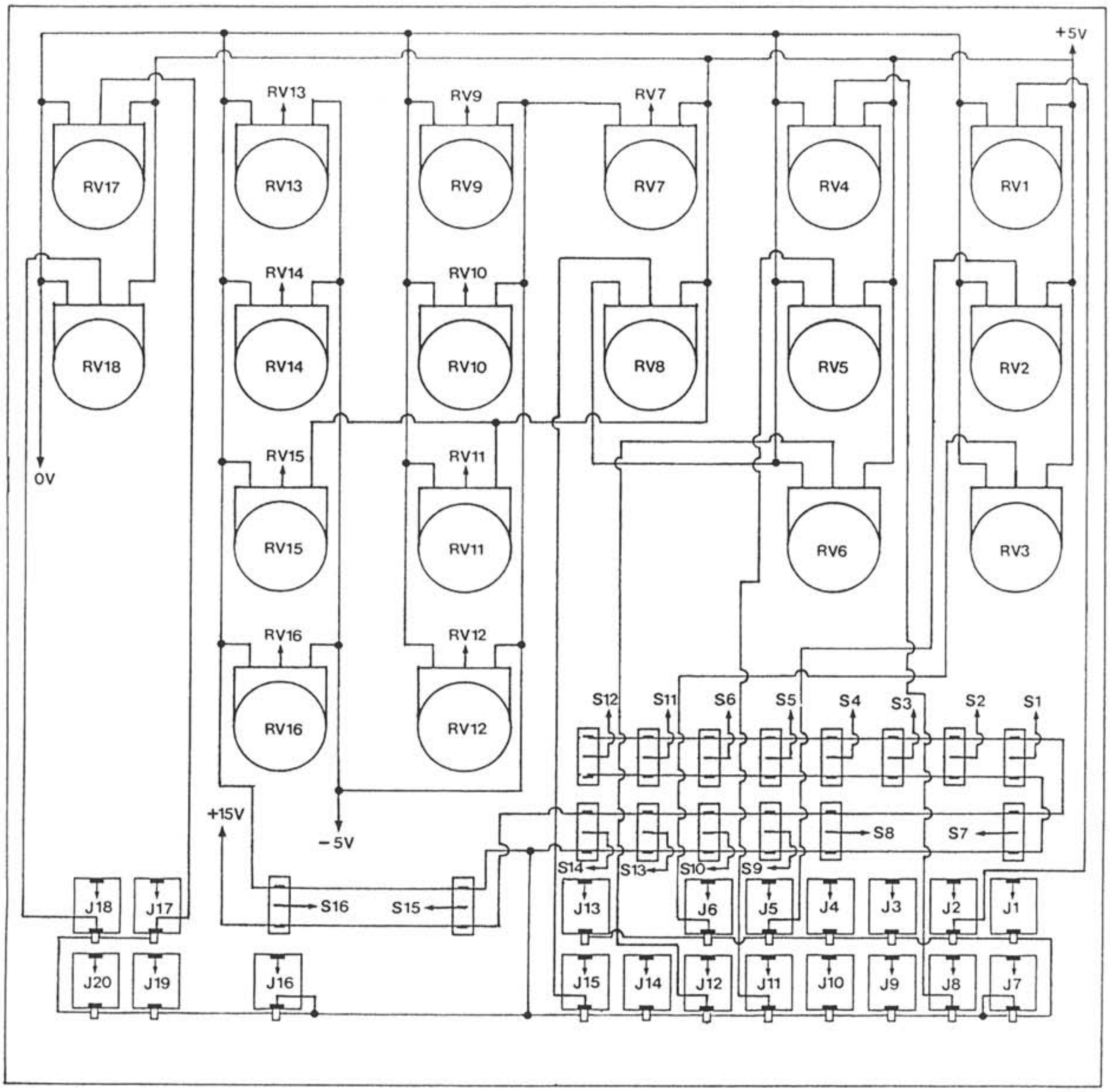


FIGURE 5. 80-C9 PANEL WIRING

recommended that the two slides be attached to the PCB but not stuck down onto the bracket until the board has been tested. This is because removal would require either the slides to be replaced or all the connections to be unsoldered. If however Molex plugs and sockets have been employed, the PCB may be unplugged and slid out for subsequent inspection.

Power supplies for a single voice card are a stable +/-15V at 300mA per rail and +/-5V at 100mA per rail. Note that the +5V rail is not connected to the PCB but is required to source a number of control potentiometers on the front panel. Other power supply connections are best taken direct to the PCB and from there to panel mounted components. Ultimate connections to the PSUs may be hardwired or use an appropriate "in-line" plug/socket connector (e.g. 5 pin DIN).

4. CALIBRATION

Calibration has been kept to a minimum by the use of specialised ICs and is a relatively sraightforward procedure for a design of this complexity. Trimming of the required functions is by way of eight multiturn presets, three for each of the oscillators and two for the single low pass filter, with all other circuit blocks requiring no calibration whatsoever. The procedure for calibration may be separated into two distinct types of adjustment. The first is the adjustment of a convenient starting frequency (pitch of the VCOs and cut-off frequency of the VCF). The second is calibration of the response of the VCOs and VCF to an incoming keyboard control voltage, i.e. the relationship between control voltage and frequency.

Taking VCO 1 first, RV19 has to be adjusted to provide a convenient starting pitch for the oscillator. This to some degree is a matter of personal taste, but it is suggested that this preset be adjusted such that with no input voltages the oscillator will be tuned to the lowest frequency of a four octave keyboard, i.e. 65.406Hz assuming a C-C keyboard. For VCO 2 it is recommended that the initial frequency is set the same as for VCO 1 so that the frequency potentiometers behave in a similar manner. However, such a starting frequency is too high for modulation purposes and it may be desirable to wire the switch on the exponential modulation control input socket (J9) to -5V. In this way VCO 2 will operate five octaves below VCO 1 until insertion of a jack plug into this socket. Alternatively RV22 may be set to a frequency to compromise performance as both an audio and modulation oscillator (i.e. 3 to 5 octaves below that of VCO 1).

In order to perform the calibration of initial frequency it is necessary to hear (or see) the oscillators. This is achieved by selection of any waveform and allowing the VCO signals to pass to the output sockets (i.e. provide a constant gate and turn VCO level, VCF frequency and ADSR sustain controls fully clockwise with all other controls fully anticlockwise).

The next step is to adjust both oscillators so that they accurately track an incoming keyboard control voltage, normally a +1V/octave relationship. For this, both oscillators are calibrated identically except that once VCO 1 is calibrated it may prove simplest to calibrate VCO 2 using VCO 1 as a reference oscillator. There are a number of ways to achieve this +1V/octave scaling. One is to use a previously calibrated oscillator (as above) and make the calibration using the beat frequency technique. Another approach is to employ two stable fixed frequency oscillators and use them in conjunction with an oscilloscope to generate Lissajous figures. Whichever method is chosen, a calibrated voltage source (e.g. from a keyboard) and an accurate voltmeter or oscilloscope will greatly simplify this process. Calibration may now proceed by firstly grounding RV21 (RV24 for VCO 2) such that pin 7 of the CEM 3340 is at 0V, and applying a positive control voltage to the keyboard CV input. This voltage is increased until a frequency of about 200Hz is produced by the oscillator. Then increase this voltage by one volt (as accurately as possible) and adjust RV20 (RV23 for VCO 2) until the frequency is exactly double that of the initial frequency. This step should be repeated several times in order to achieve an exact doubling of frequency per volt applied and it is recommended that the incoming control voltage be varied such that the calibration is carried out in the general range of 150 to 500Hz.

This procedure should now be repeated using an initial frequency of about 5kHz (i.e. increase the calibration voltage by between 4 and 5 volts) and adjusting RV21 (RV24 for VCO 2) until a doubling of frequency is obtained when the applied voltage is increased by exactly one volt. This is basically the same technique as used for calibrating RV20/RV23 but at a frequency 4 to 5 octaves higher. This adjustment is the previously mentioned high frequency track adjustment and is only possible after accurate calibration of RV20/RV23. Once the RV21/RV24 calibration has been carried out, recheck the low frequency calibration and observe that the VCOs now track correctly over the entire audio range. Both adjustments of the presets are best repeated after the voice card has been powered up for several hours.

Once RV20, RV21, RV23 & RV24 have been accurately adjusted, both oscillators have been accurately calibrated for an incoming keyboard control voltage. Other inputs will behave in a very similar manner but may be less accurate due to resistor tolerances. It is now necessary to repeat the adjustment of the initial frequency (as previously described via RV19 and RV22) since adjustment of the keyboard tracking presets will have altered the preset starting frequency.

The final stages of calibration involve similar techniques but relate to the voltage controlled filter. In this case RV25 will set an initial cut-off frequency and RV26 allows adjustment of the voltage to cut-off frequency scale. Calibration is best achieved by allowing the filter to oscillate. Then adjustment may be performed in a similar manner to that of a VCO. Sustained oscillation can be maintained by setting RV8 fully clockwise and adjusting RV7 such that the frequency of oscillation is within the audio range. (Note that to hear the filter oscillate both envelope generators must be constantly gated on with ADSR sustain controls fully clockwise). Once the sound of the oscillating filter has been identified, connect a variable voltage source or previously calibrated keyboard to J7 (as for calibration of the VCOs). It should now be possible to hear the frequency of oscillation vary with a change in voltage at J7. Calibration may now proceed by adjustment of RV26 in the same way as for the VCOs, but it is recommended that for adjustment of tracking the beat frequency method be employed using one or both of the previously calibrated VCOs. If this procedure is used then it is preferable to compare similar sounding waveforms. This is possible by using the triangle wave outputs. Once the VCF has been adjusted so that the cut-off frequency changes at +1V/octave, the initial frequency may be established by adjustment of RV25.

5. IN USE

The primary use of a synthesiser voice card is fairly obvious, as with the addition of a noise source and keyboard circuitry the user possesses a "complete" analogue synthesiser. However, it is anticipated that many users will wish to further process the audio signal using existing Digisound 80 modules, with for example the addition of external LFOs, ADSRs etc. This is easily achieved with reference to Table 1 which shows the relevant CV limits. A suitable output is produced by modules 80-2 (VCO) and 80-3 (VCLFO), but all other modules having outputs between 0V and +10V may be used if they are attenuated prior to connection to the voice inputs. For this we recommend use of module 80-5 (processor) which provides for attenuation of 4 independent channels. Alternatively, if it is envisaged that connections to +10V outputs will be made frequently, it may be preferable to construct a potential divider on the relevant input jack sockets. This is done using two 47k resistors joined together at one end. One free end is connected to the jack socket, the other free end to 0V and the join of the two resistors to the PCB input.

6. COMPONENTS

RESISTORS, 5%, 1/4w carbon film	
R7,11,26,30	470R
R17,34	470k
R18,36	620k
R19,22,40	10k
R20,38	270k
R21,39,55,61,62,63	47k
R35,64,65	100k
R37	130k
R42,43,54,56	1k
R52	24k
R53	27k
R57,59	30k
R58,60	20k
R69,73	5k6
R70,74	56R
R89-104 (16 off)	4k7

RESISTORS, 1%, 1/4w metal film, 100ppm	
R1,2,3,4,5,6,23,24,25	100k
R8,27	200k
R9,29	1M5
R10,12,15,28,33	1M
R13,31	5k6
R14,32	24k
R16,41	1k8
R44,45,46,47,48,49,50,66	56k
R51	910R
R75,82	27k

OTHER RESISTORS

R67,68,71,72 100k SIL, 4 individual 1 off)

R76,78,80,83,85,87 10k SIL, 7 commoned (2 off)

R77,79,81,84,86,88 470R SIL, 7 commoned (2 off)

CAPACITORS

C1,3,7,8,25	10n polyester
C2,5,10,32-45 (17 off)	100n polyester
C4,9	1n 1% polystyrene
C6	220p ceramic
C11	330p polycarbonate
C12,13,14,24,27	33n polycarbonate
C15,16,22	2n2 polypropylene
C17,18	1u PCB electrolytic
C19	1n polypropylene
C20,21	2u2 PCB electrolytic
C23	4n7 polyester
C26,28	22n polyester
C29,30,31	1u tantalum bead

PRESETS

RV19,22	100k horizontal multiturn
RV20,23	10k horizontal multiturn
RV21,24	10k min. multiturn, side adj.
RV25	50k min. multiturn, side adj.
RV26	500R min. multiturn, side adj.

SEMICONDUCTORS

IC1,2	CEM 3340
IC3	CEM 3372
IC4	CEM 3360
IC5,6	CEM 3310
IC7	TL 084
IC8	TL 072
IC9	LM 1458
IC10,11,12,13	4016B
TR1,2,3	BC 212L

POTS, SWITCHES (PANEL MOUNTING)

RV1-18	10k lin. rotary
S1-16	SPDT sub. min. toggle

TABLE 1.

Notation on PCB overlay	Parameter	Control voltage limits
+15V	Power supply input - to +15V rail of PSU	
J19	Noise/Audio input	
J1	VCO 1 Pitch bend/Frequency control	(-5V - +5V)
J7	Keyboard control voltage input	(-5V - +5V)
N/C		
N/C		
N/C	Reserved for future expansion	
J20	Audio output	
0V	Ground connections - to panel and to 0V rail of PSU	
-15V	Power supply input - to -15V rail of PSU	
N/C	Reserved for future expansion	
N/C	Reserved for future expansion	
N/C	Reserved for future expansion	
J16	Gate input for ADSR 1 & 2	
N/C	Reserved for future expansion	
-5V	Power supply input - to -5V rail of PSU	
N/C	Reserved for future expansion	
N/C	Reserved for future expansion	
J14	VCF Cut-off frequency CV input	(-5V - +5V)
J4	VCO 1 Linear frequency CV input	(-5V - +5V)
J3	VCO 1 Exponential frequency CV input	(-5V - +5V)
J2	VCO 1 Frequency CV input to RV1	(0V - +5V)
J5	VCO 1 Pulse width modulation CV input to RV2	(0V - +5V)
J11	VCO 2 Pulse width modulation CV input to RV5	(0V - +5V)
RV7	VCF Cut-off frequency control potentiometer	(-5V - +5V)
J8	VCO 2 Frequency CV input to RV4	(0V - +5V)
J9	VCO 2 Exponential frequency CV input	(-5V - +5V)
J10	VCO 2 Linear frequency CV input	(-5V - +5V)

RV15	ADSR 2 Sustain CV input	(0V - +5V)
RV14	ADSR 2 Decay CV input	(0V - -5V)
RV16	ADSR 2 Release CV input	(0V - -5V)
RV13	ADSR 2 Attack CV input	(0V - -5V)
RV11	ADSR 1 Sustain CV input	(0V - +5V)
RV10	ADSR 1 Decay CV input	(0V - -5V)
RV12	ADSR 1 Release CV input	(0V - -5V)
RV9	ADSR 1 Attack CV input	(0V - -5V)
J17	ADSR Modulation (VCA 4) depth CV input	(0V - +5V)
J18	VCO 2 Modulation (VCA 3) depth CV input	(0V - +5V)
N/C		
VREF	Voltage reference for VCOs 1 & 2 - to +15V rail of PSU	
J6	VCO 1 Output level (VCA 1) CV input	(0V - +5V)
J12	VCO 2 Output level (VCA 2) CV input	(0V - +5V)
J15	VCF Resonance/"Q" CV input	(0V - +5V)
J13	VCF Cut-off frequency CV input	(-5V - +5V)
S11	Keyboard CV track VCF cut-off frequency	
S16	Keyboard CV track ADSR 2 time constants (inv. function)	
S3	ADSR 2 Output to VCO 1 frequency control input	
S12	VCO 2 Modulate VCF cut-off frequency	
S14	ADSR 2 Inverted output to VCF cut-off frequency	
S13	ADSR 2 Output to VCF cut-off frequency	
S4	Select VCO 1 sawtooth output to VCA 1	
S7	Synchronise VCO 1 to VCO 2	
S1	VCO 2 Output to exp. frequency control input of VCO 1	
S8	Select VCO 2 sawtooth output to VCA 2 & 3	
S2	VCO 2 Output to lin. frequency control input of VCO 1	
S9	Select VCO 2 pulse output to VCA 2 & 3	
S6	Select VCO 1 triangle output to VCA 1	
S10	Select VCO 2 triangle output to VCA 2 & 3	
S5	Select VCO 1 pulse output to VCA 1	
S15	Keyboard CV track ADSR 1 time constants (inv. function)	

TABLE 1 CONT.

CORRECTIONS **13th NOVEMBER 1984**

1) Page 7, Column 1, Line 2 - the second word should be potential not frequency.

2) Page 14 - resistors R76, 78, 80, 83, 85, 87 are contained in two 10k SIL, 4 individual (not 7 commoned) networks. This also means that they may be inserted either way round into the PCB.