

# TRANSCENDENT POLYSYNTH



**EXPANDABLE POLYPHONIC SYNTHESIZER**



*by* **POWERTRAN**

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# POLYSYNTH



The Transcendent Polysynth is no ordinary synthesiser — it's a family of them. Each of its voices is a complete synthesiser in itself with two VCOs, two ADSRs, a VCA and a VCF in addition to all the usual synthesiser functions. Design and development by Tim Orr.

The Polysynth is a four octave polyphonic music synthesiser. The standard unit has four voices making it possible to play up to four notes simultaneously. Each voice is a complete synthesiser in itself, having two VCOs, two ADSRs, one VCA and one VCF. The voices are totally voltage controlled which enables them to be ganged up in a bus system. So when the master sustain level is adjusted, the sustain level on all voices is set to the same value. However, by providing independent pitch and gate signals, it is possible to control each voice from the keyboard and yet have a common control over the other parameters. The machine can be expanded (with an extra mother board (PS4), four voice boards (PS7) and a panel board (PS6)) to a system with eight independent voices.

The design has minimal wiring (Fig.1), most of which is made with preformed ribbon cable links. Also, the four voice boards, which require nearly 50 signals each, plug into a Molex connector bus system.

All the common controls, such as ADSR parameters, modulation oscillators, volume and noise level are located on the left hand panel board (PS5).

The right hand board (PS6) handles the individual parameters of oscillator tuning and the voice on/off control. Both the panel boards deliver their signals to the mother board (PS4) which then distributes them to the voice slots. Pitch and gate parameters are independently fed to the voices, these signals being generated by PS1, 2 and 3.

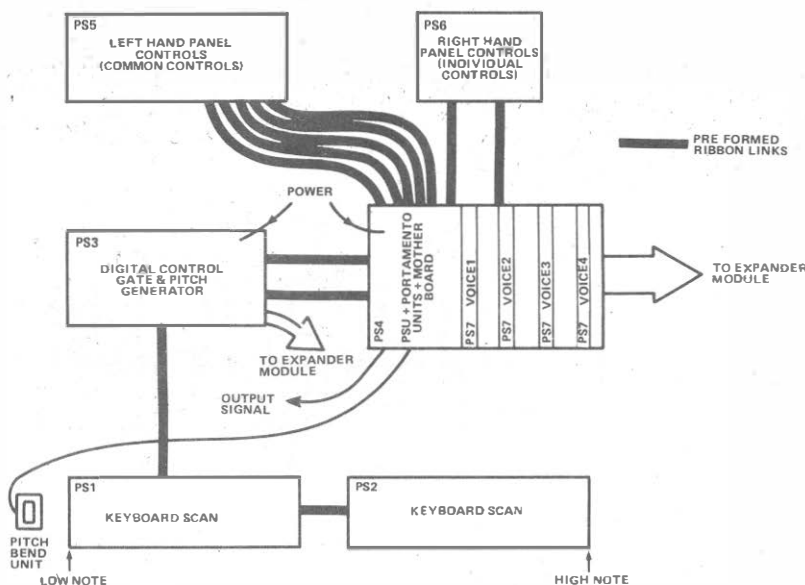
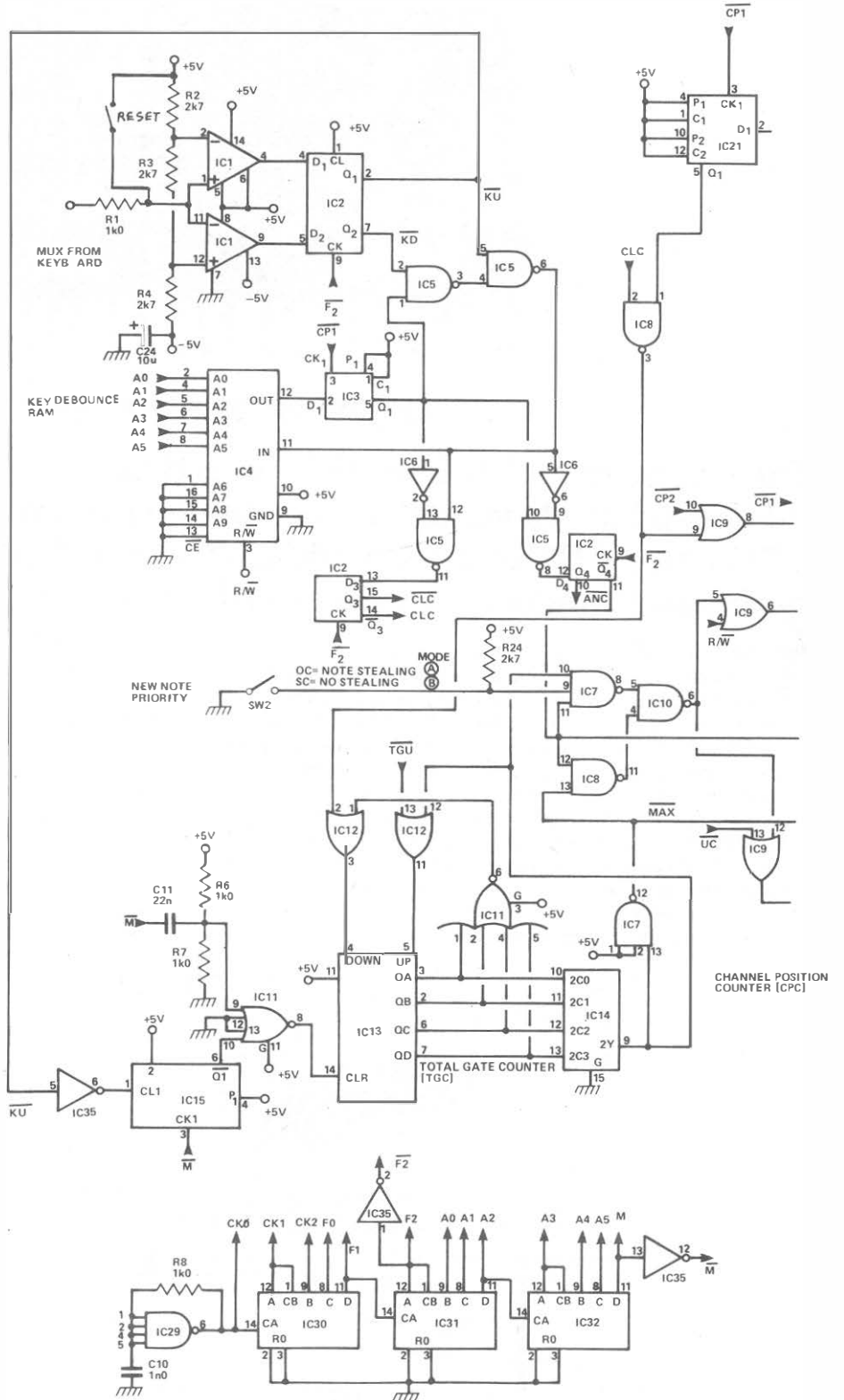
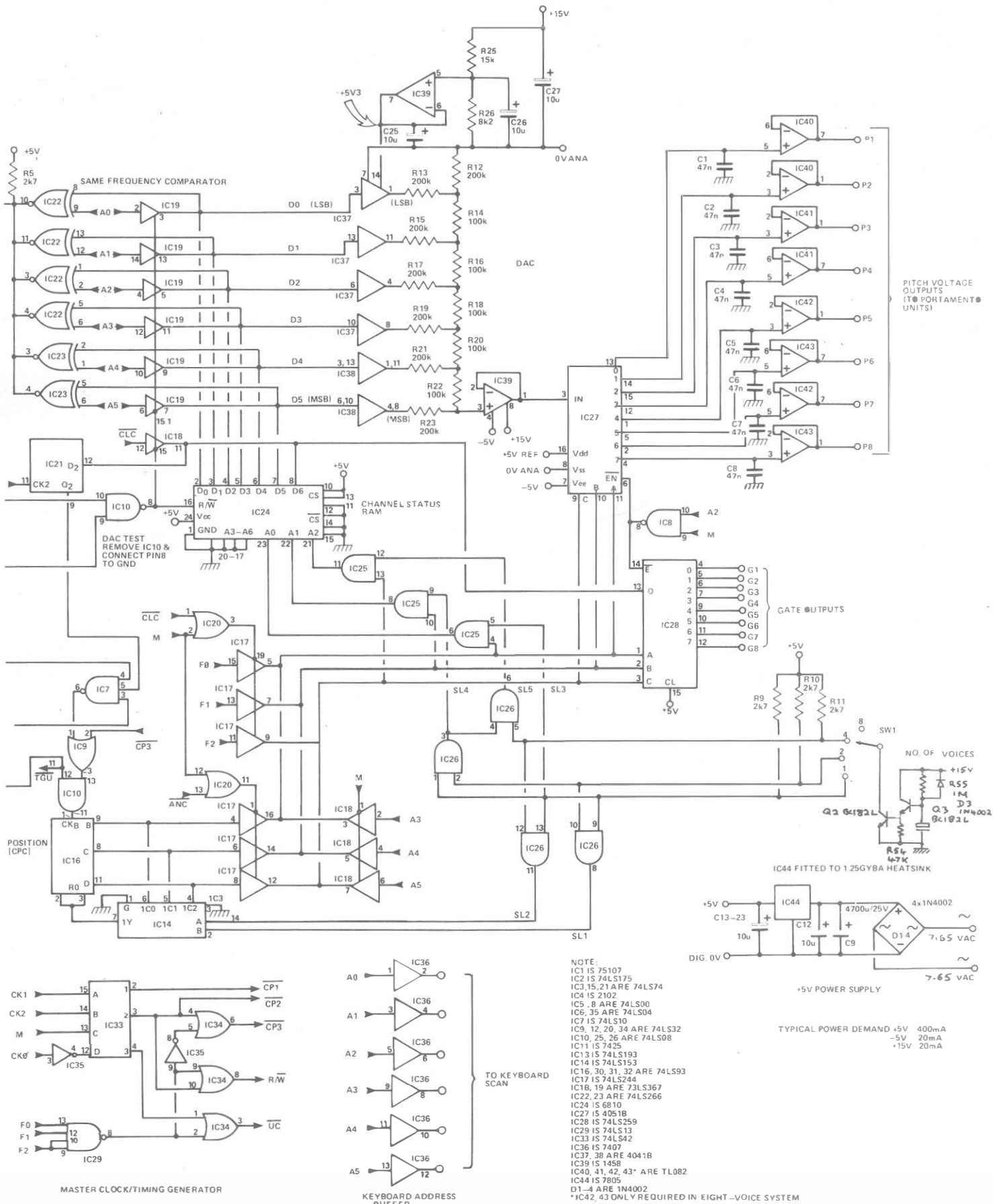


Fig.1 Board-to-board connections. The voice boards (PS7) simply plug into the mother board (PS4) using Molex connectors.

Fig.2 Circuit diagram of PS3, the digital board.





## HOW IT WORKS : DIGITAL CONTROL PS1, 2, 3

The synthesiser decides which notes are being played by digitally scanning the keyboard and analysing the received data. A high frequency oscillator (Fig.2, IC29 pin 6) generates the master timing for the system, which is then divided down by a 12 stage divider (IC30,31,32). The signals generated by this section are used to construct various timing waveforms. A0 to A5 are sent to the keyboard scan circuitry, PS1,2 (Fig.4). This is a 49-way multiplexer. The six bit code can address 64 locations, but as the keyboard only has 49 notes the other 15 locations remain unused. The bottom part of the code (A0,1,2) addresses each eight bit multiplexer whilst the top three bits (A3,4,5) are decoded by IC8. The outputs of IC8 then enable each of the seven multiplexers in turn so that a complete keyboard is produced. When a note is unpressed it generates - 5 V and when it is between contacts it is an open circuit.

Figure 5 shows a typical keyboard scan output. The unused top 15 addresses have been arranged to generate a key unpressed signal. A power clear circuit has been included to clear the system when the machine is powered up.

The keyboard output is fed to a voltage comparator, IC1, Fig.2. In-

tegrated circuits IC1,2,3,4,5 form a circuit block that decides when to assign a new channel to a voice (ANC) or to clear an existing one (CLC). The problem of channel assignment is very complex.

The synthesiser has only four voices, so the question to be answered is 'what should happen if more than four keys are pressed at once?' Should the system ignore the extra keys or should it reassign voices to the extra notes? If reassignment (referred to as note stealing) is to be used, then which notes should be taken, the first or last note played, or the highest or lowest in pitch?

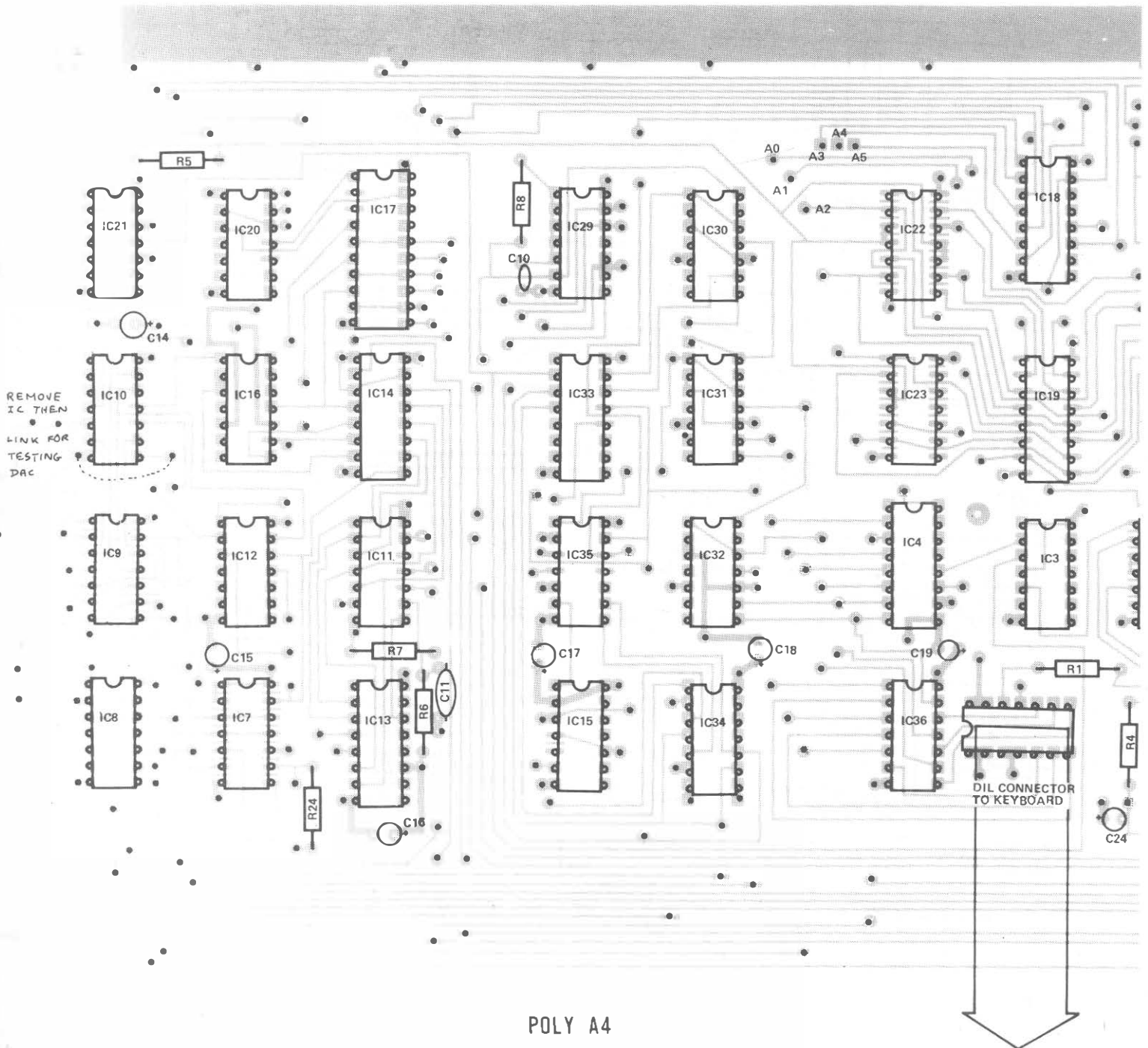
There is no 'correct' solution other than a voice per note.

Two modes of operation have been provided. Mode A permits note stealing, where the oldest channels are reassigned to new notes, when the selected number of notes is exceeded on the keyboard. Mode B does not allow more than the selected number of notes to be assigned. Extra notes are ignored.

The digital electronics is best considered as consisting of a series of modules.

The 'Total Gate Counter' is used to keep count of the number of channels in use at any point in time.

Fig.3 Component overlay of the master clock/timing generator. Dots indicate through-board connections on the double-sided PCB.



## HOW IT WORKS : DIGITAL CONTROL PS1, 2, 3

The 'Channel Position Counter' decides which will be the next channel to be assigned. In mode B, if a channel is already in use, then it will skip it and continue on until it finds a free one. In mode A, the channel position counter indicates the oldest channel which will then be reassigned if no free channel is available.

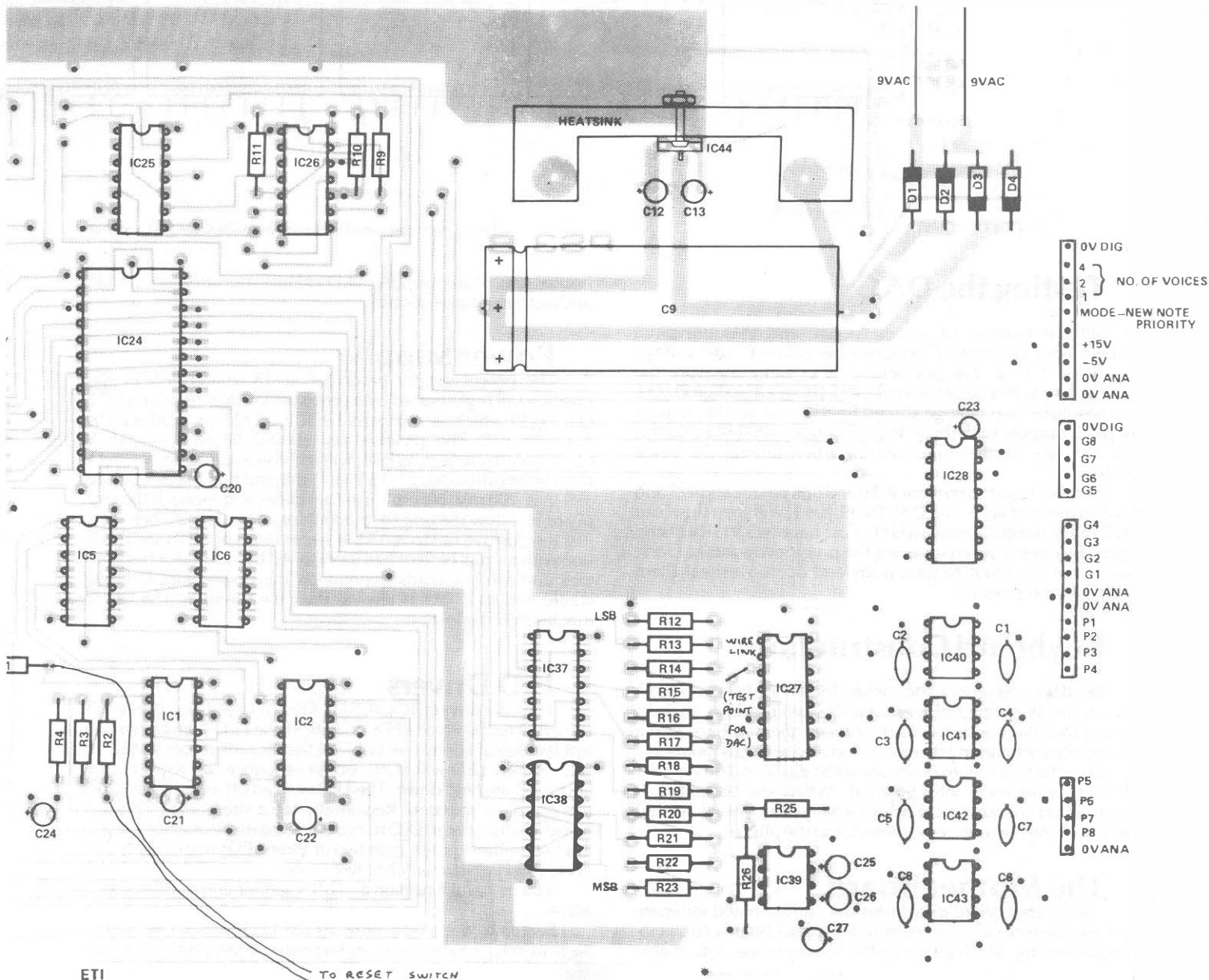
The 'Channel Status RAM' stores the pitch and gate parameters, the number of parameters stored being determined by the selector switch SW1. Data bits D0 to D5 generate pitch and bit D6 is the gate signal. The 'Same Frequency Comparator' comes into operation when a note is released. When this occurs, the circuit removes the gate signal from the respective memory location but rewrites the pitch data which would otherwise be cleared!

The electronics in Fig.2 generate gate and pitch signals which are fed to the synthesiser voices. Some typical waveforms are shown in Fig.9 for a four note selection. Each note pressed generates a gate and pitch signal and assigns a voice channel. When a note is released the gate signal is lost, but the pitch value remains until the channel is reassigned. The gate signals are distributed via an eight way addressable latch, IC28. The pitch signals are distributed via an eight way multiplexer driving

eight sample and hold units (IC27,40,41,42,43). In a four voice system IC42,43 are omitted.

The pitch data is generated by a precision DAC (IC37,38,39). This converts the six bit code into an analogue voltage using an R/2R network.

A typical DAC error is shown in Fig.11. When the MSB of the code changes from 0 to 1 the step size is too small, which results in an error for all codes where the MSB is 1. Generally the worst errors are generated at the changeover point of the high bits of the code, Fig.12. The synthesiser is exceptionally sensitive to errors of this nature and so the DAC must be accurate to ten bits even though it is only converting six bits. A ten bit accuracy will give a worst error of 1 part in 32 (3% of a semitone) per step, which is only likely to occur when bits D5 and/or D4 change state. To obtain this performance 0.1% tolerance, 25 ppm/°C metal film 100k and 200k resistors are used. For superior performance the resistors can be matched for a 2 to 1 ratio using the best matched pairs at the MSB end of the DAC. Using this technique worst step errors of about 1.5% can be obtained. The DAC is powered from a +5V3 reference voltage, which results in a +1V/octave output.



## PARTS LIST : PS1, 2

Keyboard Multiplexer		Semiconductors	
Resistors all 2% M.O.		IC1-7	4051
R1,2	100k	IC8	74LS42
R3	1k0	Q1	BC212L
R4	5k6	D1	1N4148
R5	4k7		
R6-18	2k7		
Capacitors		Miscellaneous	
C1-5	10u 16 V tantalum	PCBs PS1 and PS2, 8 off 16 pin D1L sockets, 14 pin D1L socket, 49-note keyboard, 49 keyboard contacts (single pole), full length bus bars, self-adhesive cable clamp.	

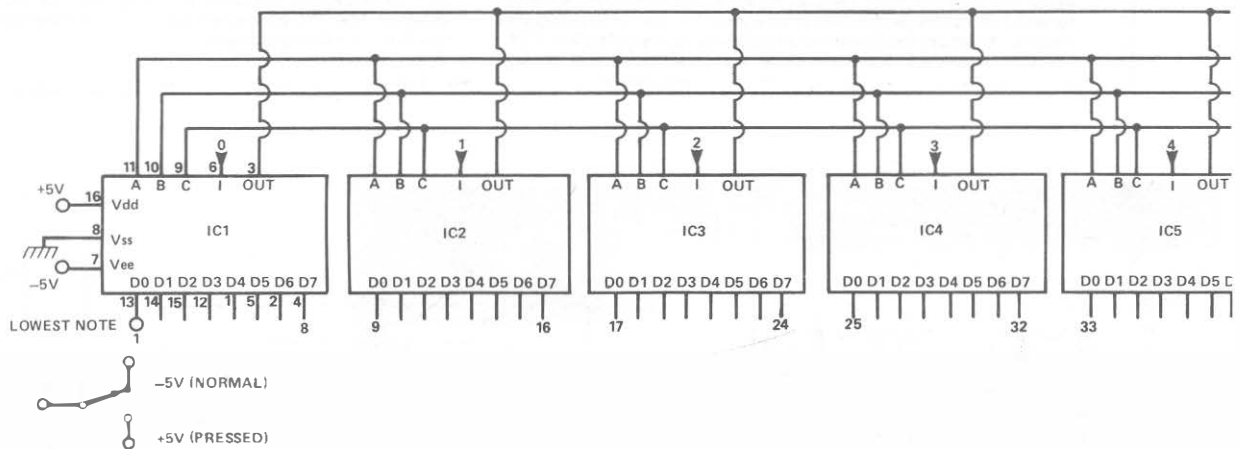


Fig.4 Circuit diagram of the keyboard multiplexer. (PS1 and PS2).

## Testing the DAC

Set the number of voices to one and measure pitch voltage one. Use note C to generate octaves. The voltage should be  $1\text{ V} \pm 2\%$  per octave. If possible measure the voltage for each note of the keyboard, using a  $4\frac{1}{2}$  digit DMM. The semitone step change should be  $83.3\text{ mV} \pm 3\%$ . Repeat for pitch outputs two, three, four. If you are unable to do this you can rely upon a musical ear when driving the voice modules!

Another test is to remove IC10 and link pin locations 7 and 8 with a piece of wire. The DAC (IC39, pin 1) will then draw out a full range ramp (64 steps) which must have NO VISIBLE step errors. Be careful when inserting the precision resistors. Don't bend them too close to their body and don't overheat them when soldering them in.

## Keyboard Construction

Bend the end wire on the contact blocks as shown in Fig.13. Thread the 49 contact blocks onto the two bus bars, making certain that the contact wire is between the two bars. Use a clean cloth for handling them to avoid grease contamination.

Glue the blocks into position on PS1 and 2 and then solder the bent wire ends into position. When the assembly is mounted on the keyboard adjust each wire contact so that there is a 1 mm clearance between it and the plunger.

## The Mother Board

The mother board distributes the common and independent synthesiser parameters from the panel boards to the individual voices, which plug into the 50-way voice slots. It also

houses the power supply, LED drivers, portamento circuits and output volume control.

## Power Supply

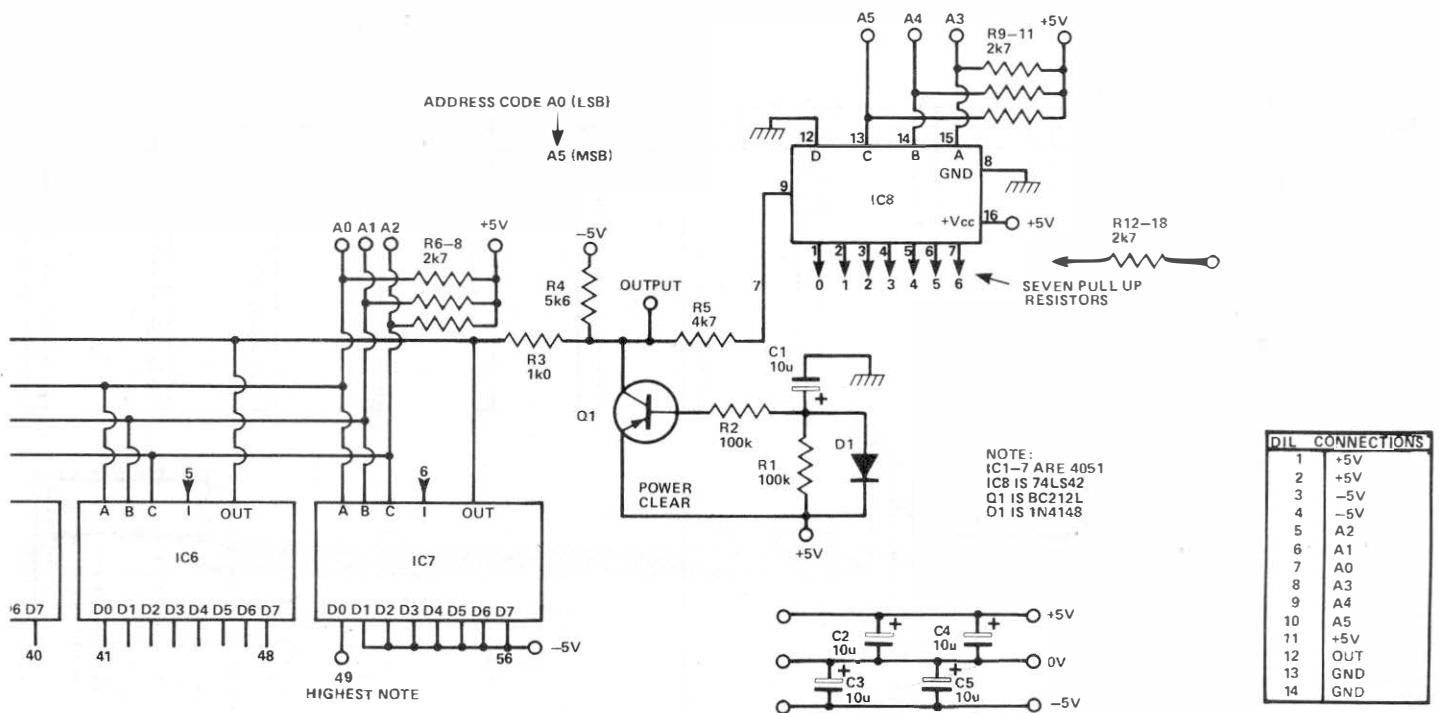
The power supply produces  $\pm 15\text{ V}$  and  $-5\text{ V}$ . The positive rail is generated by a precision voltage regulator (IC1, Q4, Fig.7) which is mirrored by IC2, Q2, Q3 to produce the negative rail. The positive rail should be set to  $+15\text{ V} (\pm 10\text{ mV})$  by adjusting PR1, with no voices plugged in. Check that the negative rail is  $-15\text{ V} \pm 75\text{ mV}$  and that the  $-5\text{ V}$  rail is  $-5\text{ V} \pm 200\text{ mV}$ . Now plug in the voices and recheck the supply rails. Allow the unit to 'burn in' for 24 hours and then readjust the positive rail to  $+15\text{ V}$  if necessary. Lock PR1 into position with a small blob of nail varnish. All the oscillator frequencies and pitch spreads depend on the power supply being stable and so if PR1 is altered then the tuning of the whole machine will be lost!

## LED Drivers

The LED drivers (IC4, IC5 Fig.8) buffer the TTL gate signals from PS3 to the voice LEDs on PS6. These LEDs can be used to test the digital control section. Set the 'Nos. of voices switch' to four. A new LED will come on, in sequence, as notes on the keyboard are held down. The LEDs will go off when the respective notes are released. Repeatedly tap a single note. This will cause an illuminated LED to cycle around the four voices. If you tap two or three notes, then two or three LEDs respectively will cycle around. Now switch to two voices.

Voices 1 and 3 turn on and off together and so do voices 2 and 4.

Switch to one voice. Now all the LEDs will act in unison. Next switch to eight voices and repeatedly tap a note. The four



## PARTS LIST : PS3

Resistors all 2% m.o. unless otherwise stated

R1,6,7,8	1k0
R2,3,4,5,9,10,11,24	2k7
R12,13,15,17,19,21,23	200k 0.1% metal film 25 ppm
R14,16,18,20,22	100k 0.1% metal film 25 ppm
R25	15k 1% metal film 25 ppm
R26	8k2 1% metal film 25 ppm

### Capacitors

C1-8	47n polycarbonate
C9	4700u 25 V electrolytic
C10	1n0 ceramic
C11	22n polycarbonate
C12-27	10u 16 V tantalum

### Semiconductors

IC1	75107
IC2	74LS175
IC3,15,21	74LS74
IC4	2102
IC5,8	74LS00
IC6,35	74LS04
IC7	74LS10
IC9,12,20,34	74LS32
IC10,25,26	74LS08

IC11	7425
IC13	74LS193
IC14	74LS153
IC16,30,31,32	74LS93
IC17	74LS244
IC18,19	74LS367
IC22,23	74LS266
IC24	6810
IC27	4051B
IC28	74LS259
IC29	74LS13
IC33	74LS42
IC36	7407
IC37,38	4041B
IC39	1458
IC40,41,42,43*	TL082
IC44	7805
D1-4	1N4002

\*IC42,43 only required in eight-voice system

### Miscellaneous

PCB PS3, 5 off 8 pin DIL sockets, 28 off 14 pin DIL sockets, 9 off 16 pin DIL sockets, 20 pin DIL socket, 24 pin DIL socket, DIL connectors and ribbon cable, Redpoint 1.25GYBA heatsink, cable clamp.

LEDs will come on in sequence and then there will be a four note gap whilst the unit addresses the other four voices that live in the expander module.

Now the mode selection. Select four voices and mode A. Play and hold down a four note chord with the left hand. Now press four notes at once with the right hand, and then release the right hand. The LEDs will go out, indicating that the right hand 'stole' the four voices and then released them, in doing so turning them off. Repeat this procedure in Mode B. The LEDs will remain on.

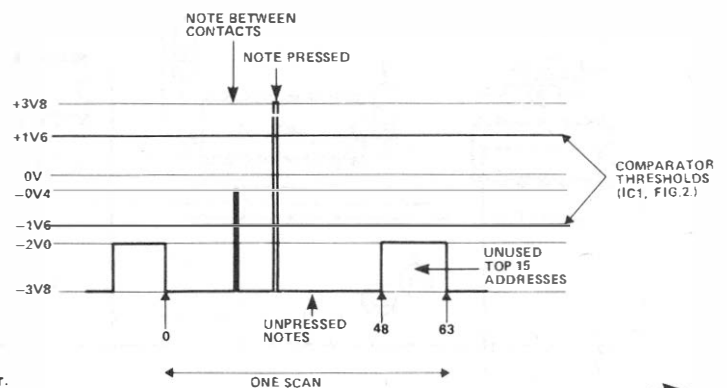


Fig.5 Output of keyboard multiplexer.



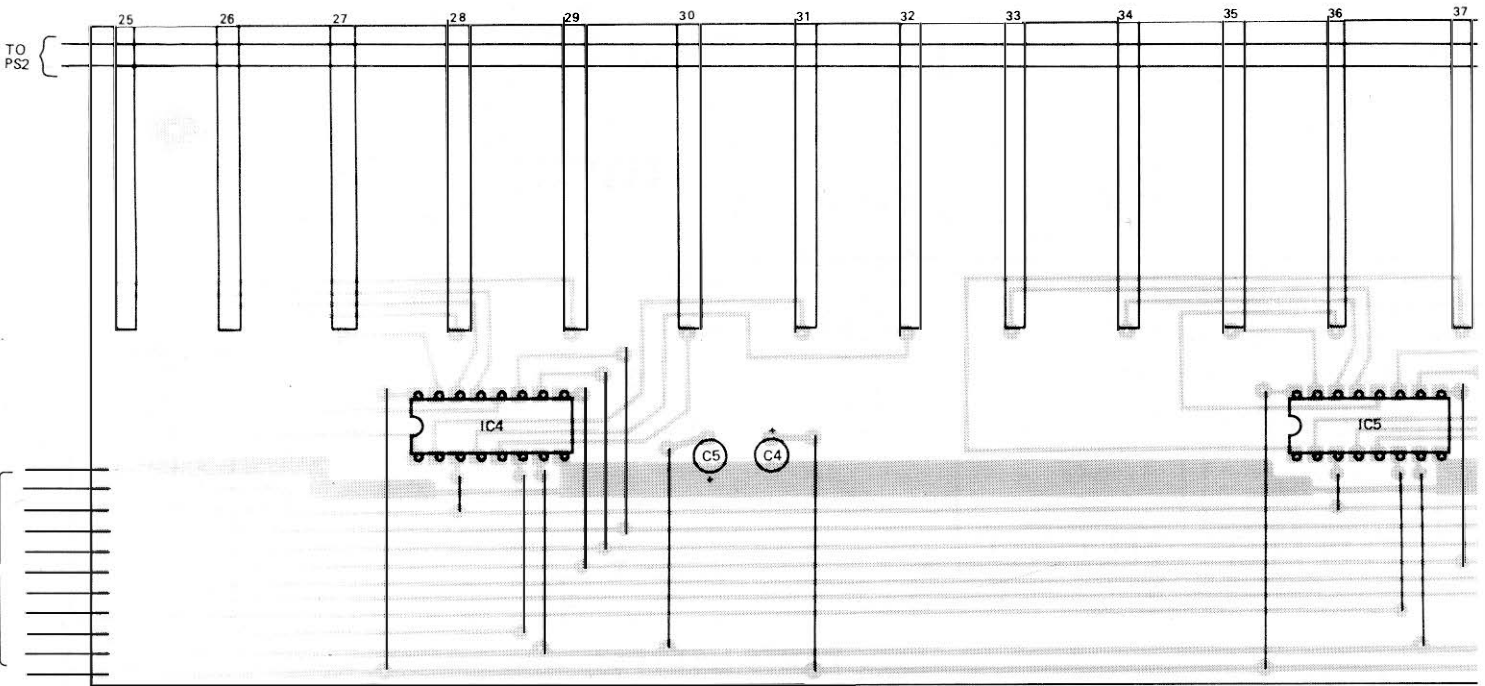
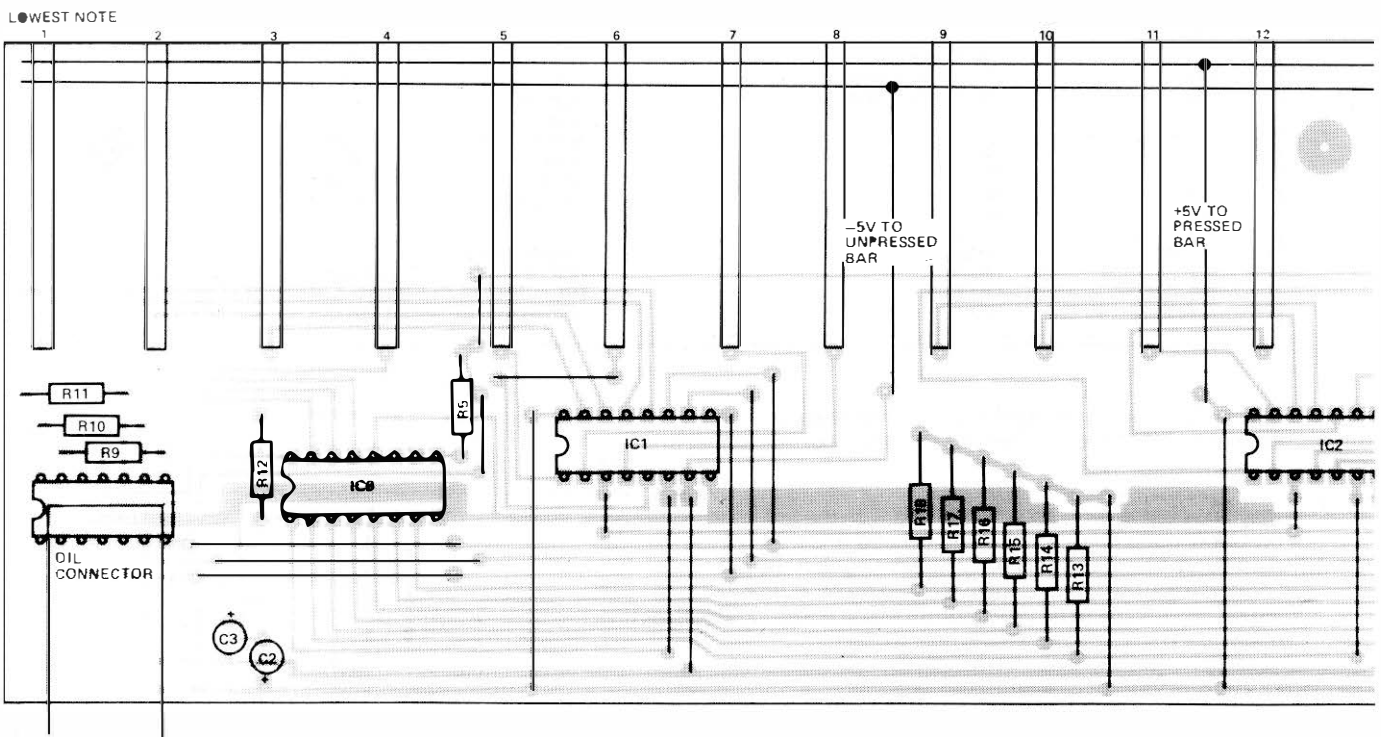
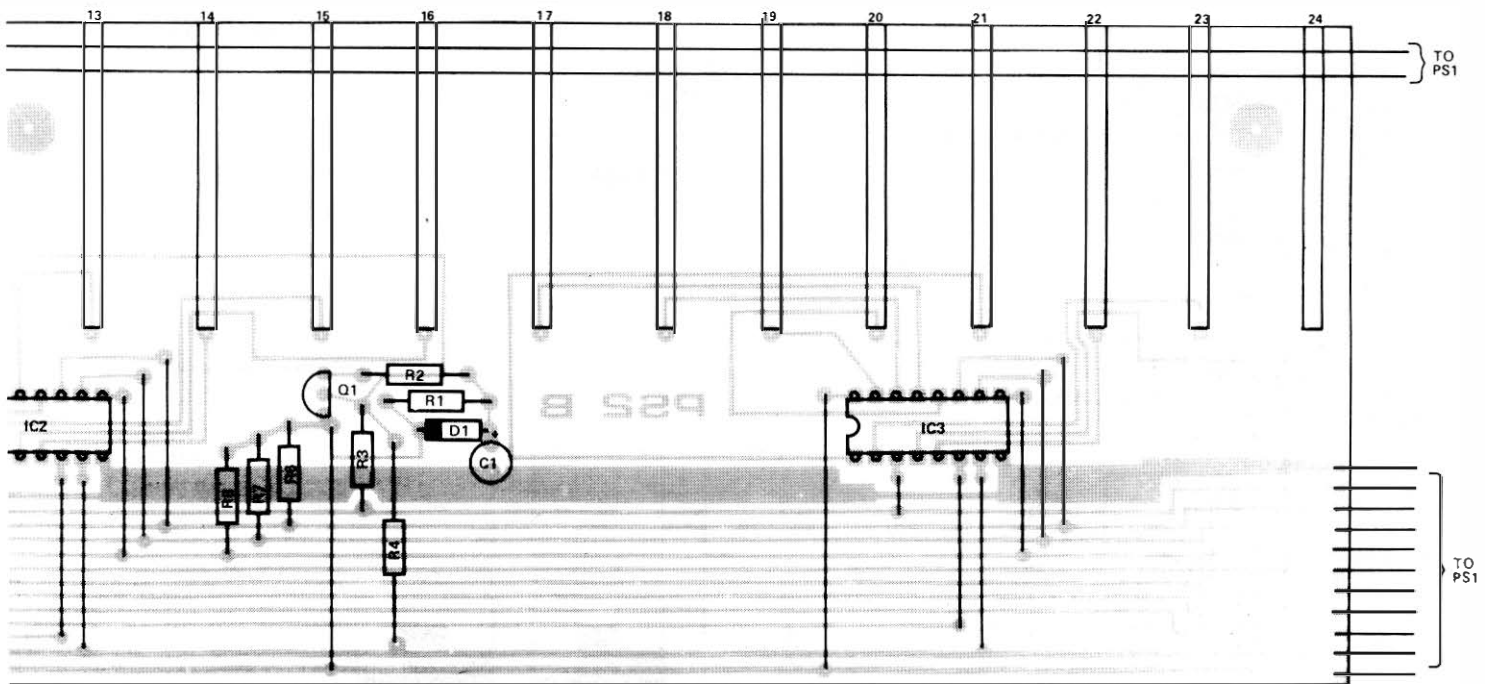
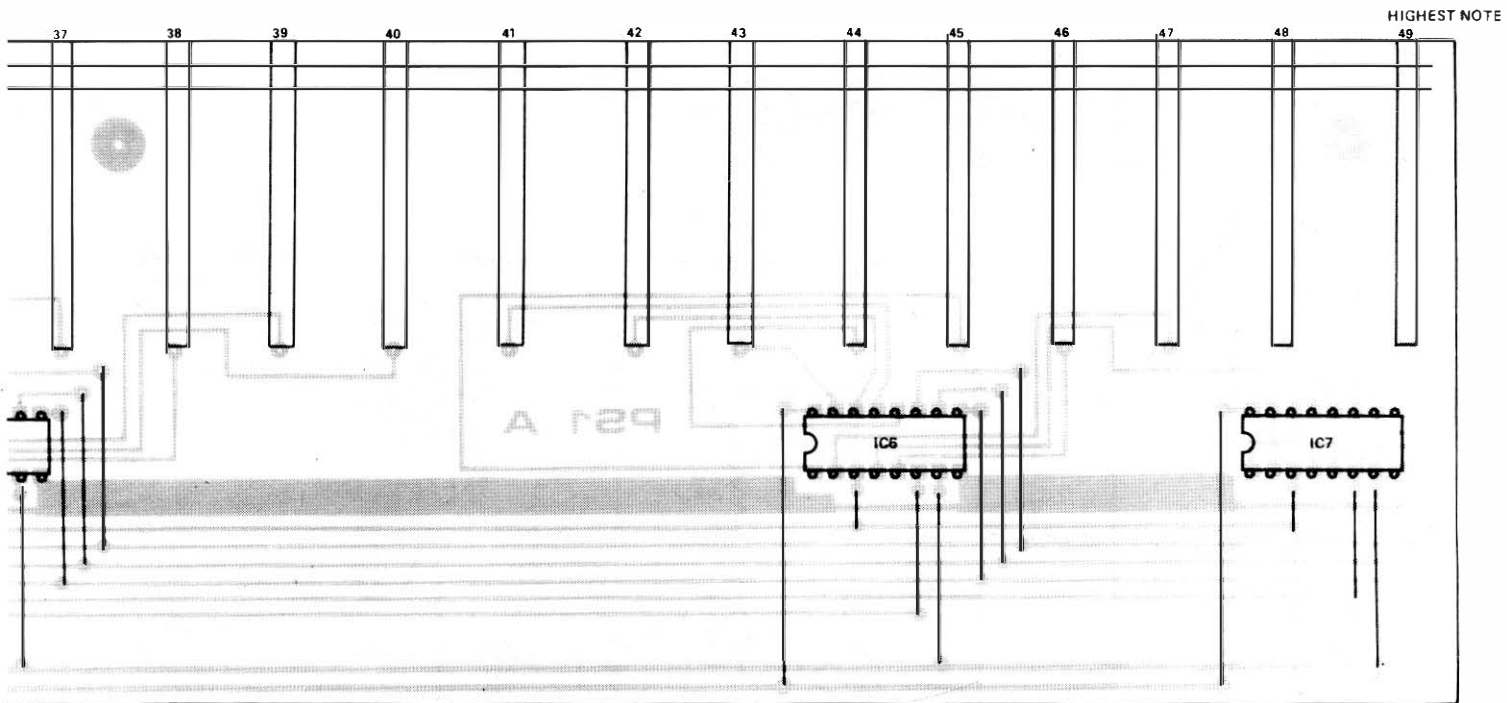
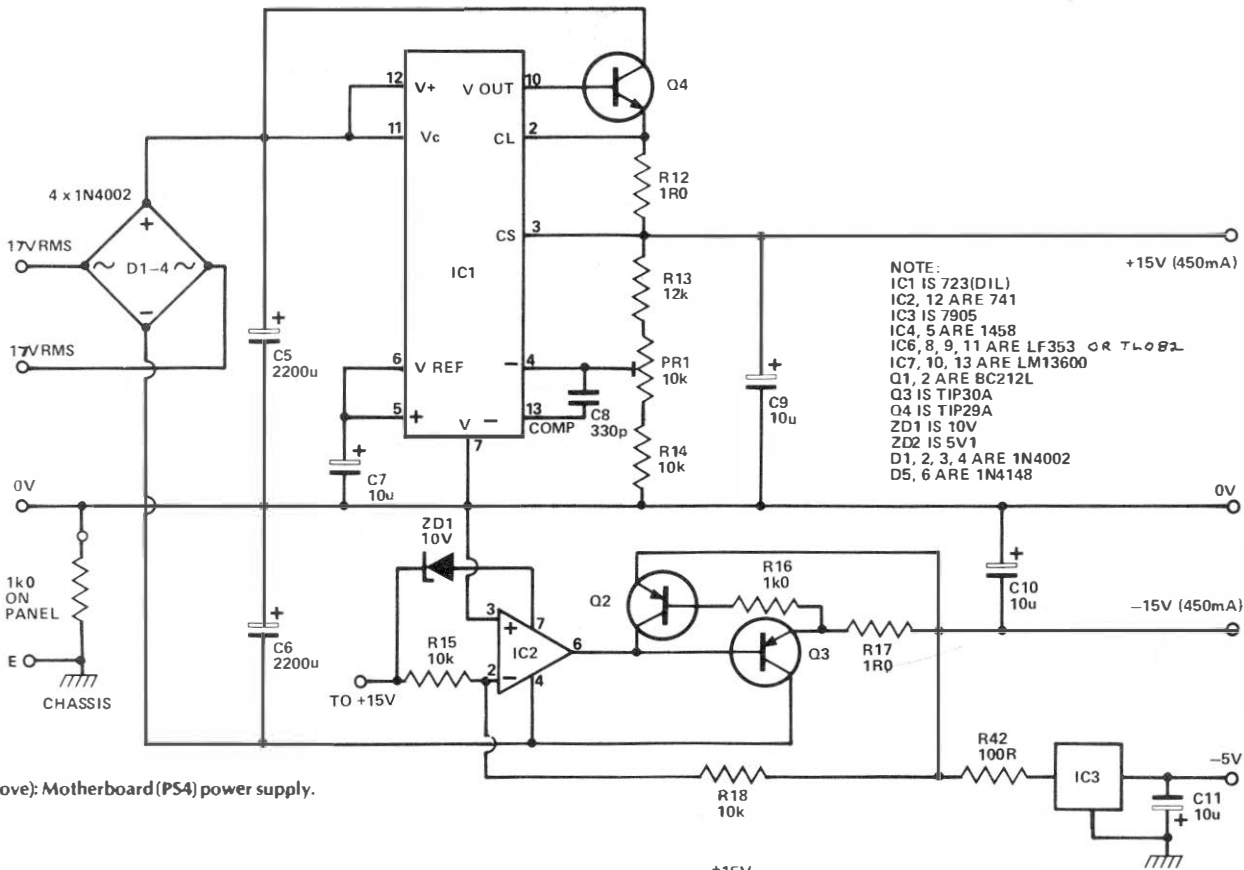


Fig.6 PS1 (above) and PS2 (below) together form the multiplexed keyboard.

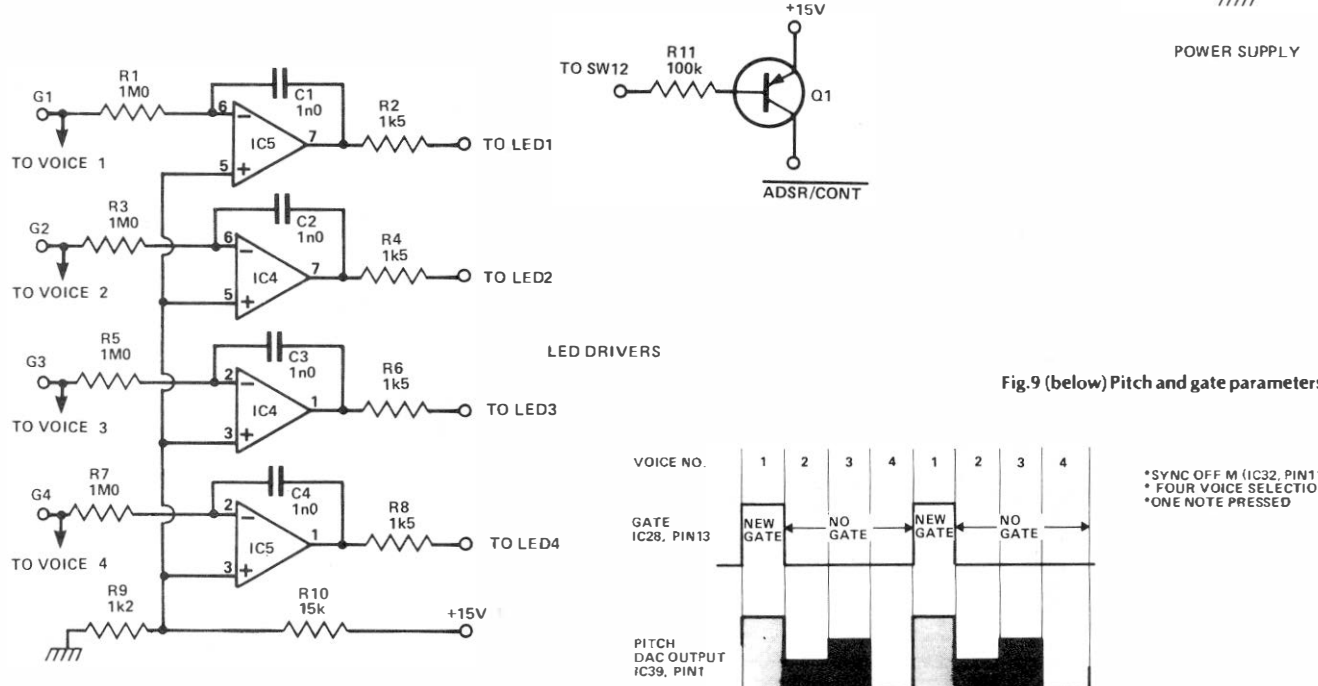






NOTE:  
 IC1 IS 723(DIL)  
 IC2, 12 ARE 741  
 IC3 IS 7905  
 IC4, 5 ARE 1458  
 IC6, 8, 9, 11 ARE LF353 OR TLO81  
 IC7, 10, 13 ARE LM13600  
 Q1, 2 ARE BC212L  
 Q3 IS TIP30A  
 Q4 IS TIP29A  
 ZD1 IS 10V  
 ZD2 IS 5V1  
 D1, 2, 3, 4 ARE 1N4002  
 D5, 6 ARE 1N4148

Fig.7 (above): Motherboard (PS4) power supply.



LED DRIVERS

Fig.9 (below) Pitch and gate parameters.

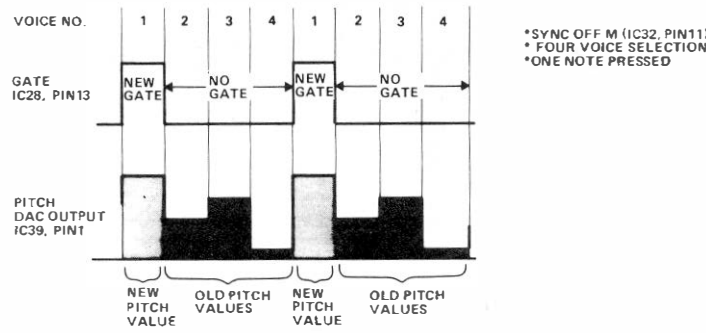
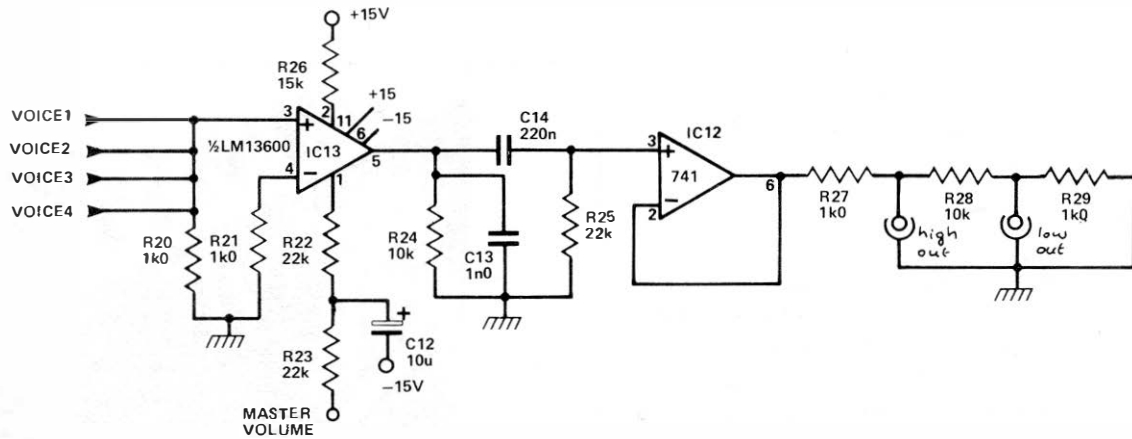


Fig.8 Motherboard LED drivers.



PORTAMENTO

	RA	RB	RC	CA	CB
1	R30	R31	R32	C15	C16
2	R33	R34	R35	C17	C18
3	R36	R37	R38	C19	C20
4	R39	R40	R41	C21	C22

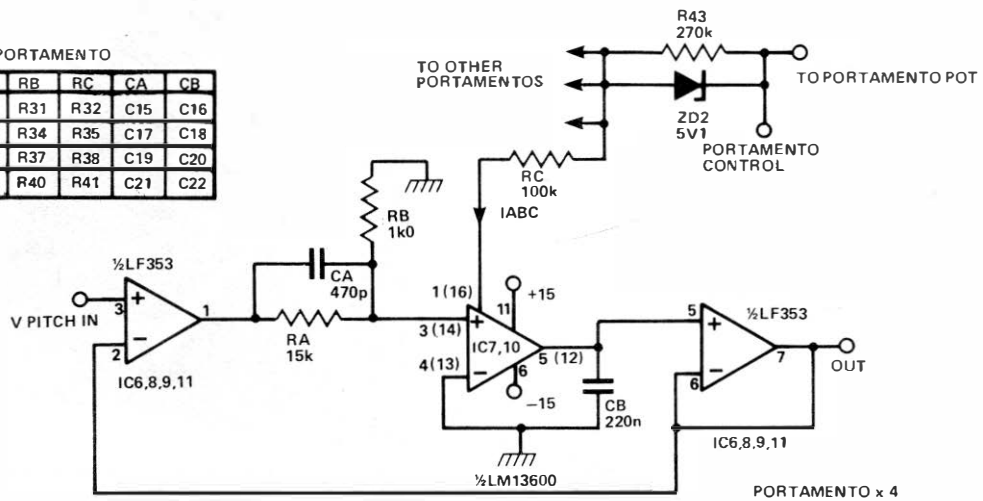


Fig.10 Motherboard portamento, master volume and pitch bend controls.

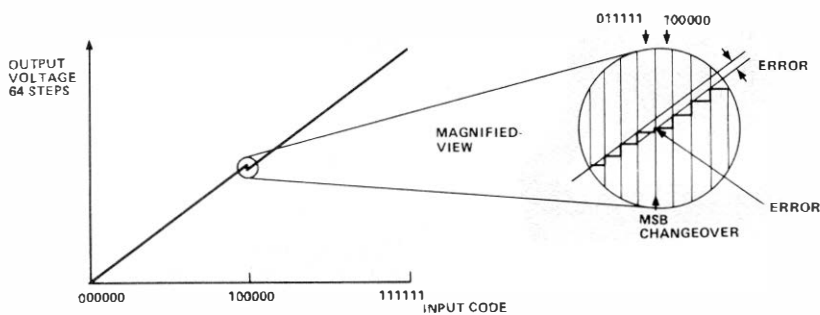
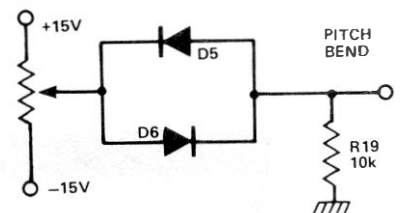


Fig.11 (left) Typical DAC error.



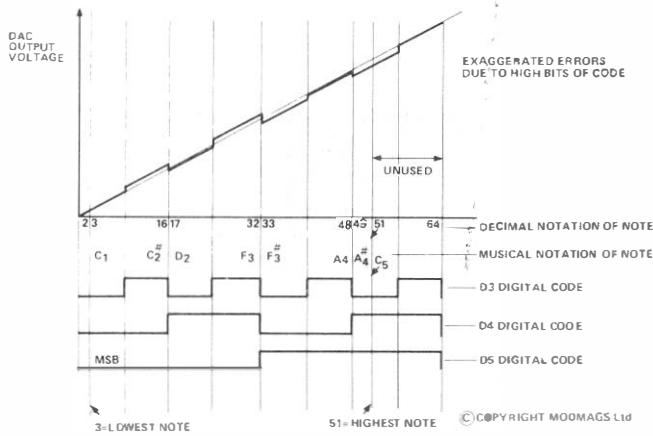
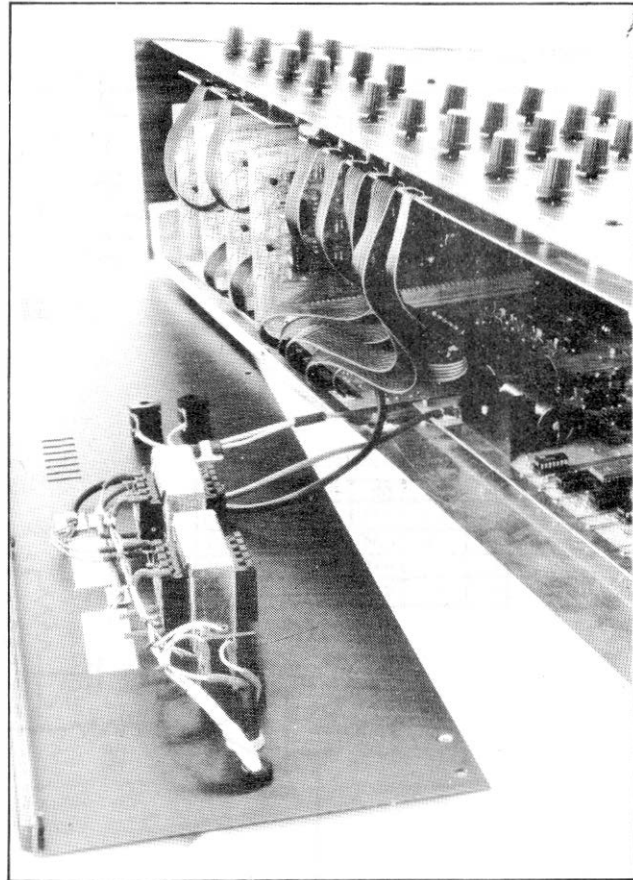


Fig.12 Points of worst error in the DAC



The power supply is mounted on the rear panel.

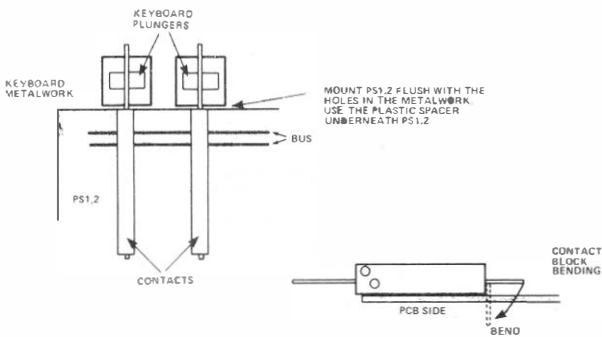


Fig.13 Mechanical assembly of keyboard plungers.

ADJUST FOR 1mm CLEARANCE WITH KEY NOT PRESSED USING LONG NOSED PLIERS.

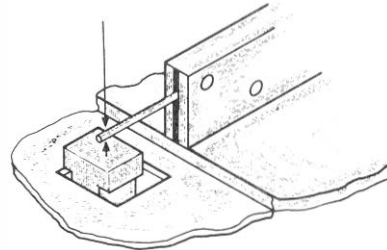
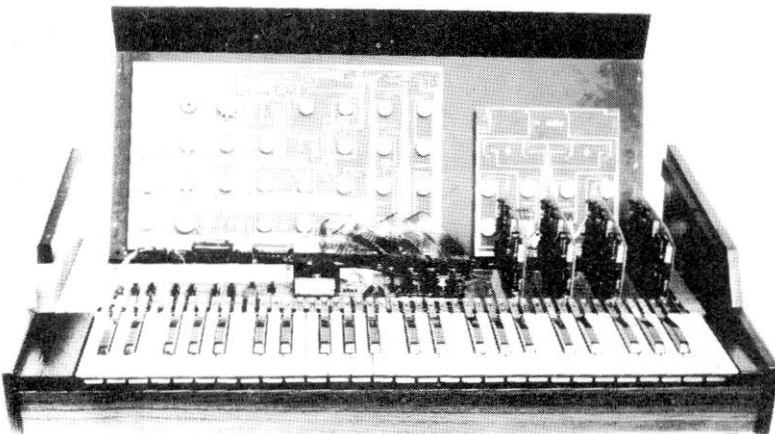


Fig.14 Adjustment of 1mm gap above keyboard plunger.



The top panel can be raised to show control board PS5 and PS6.

Next month we conclude the Polysynth project with details of the control and voice boards. (PS5, 6 and 7).

# POLYSYNTH

**Part 2: This month we continue the Polysynth project with details of the motherboard and control boards. Design and development by Tim Orr**



**P**olyphonic portamento has been provided by using a voltage controlled slew limiter for each pitch voltage (Fig. 10 p. 11, Fig. 12 p. 14). The slew rate is controlled by the  $I_{ABC}$  current and the timing capacitor  $C_B$ .

$$\text{Slew Rate} = \frac{I_{ABC}}{C_B} \text{ VS}^{-1}$$

An LM13600 OTA is used to charge and discharge  $C_B$ . The OTA is contained in a feedback loop of two op-amps to reduce the differential offset voltage change between the input and the output. The circuit generates a voltage change of about 100  $\mu\text{V}$  over its full range of 0 to  $-4 \text{ V}$  at all slew rate settings.

With the portamento pot fully clockwise, the pitch of the notes will not be noticeably slewed. As the pot is rotated anticlockwise the slew limiting will become very obvious. At its slowest rate, a four octave transition will take about 2 S.

## Volume Control

The audio outputs from all voices are mixed into IC13, Fig. 10. This is a voltage controlled amplifier. With the master volume control anticlockwise the output will be fully off.

## Pitch Bend

The pitch bend joystick is shown in Fig. 12. Note that the offset lever has to be cut short to avoid fouling the pitch bend plate. This lever is adjusted so that with the joystick in the centre, the voltage on the pot wiper is zero.

## Panel Boards

Board PS5 generates all the common synthesiser parameters and PS6 the individual parameters. Most of the panel controls merely generate DC voltages. However, there are three modulation oscillators which are used to produce VCO vibrato, VCF sweep and pulse width modulation of the VCO square wave output. Note that the pulse width modulation pots (RV2,4) are dual function.

Clockwise they are connected to the modulation oscillator and anticlockwise they produce a manual control ranging from one-to-one square wave to a very thin pulse. In the centre of the pot movement there is a mechanical 'click'.

A pseudo-random noise source (IC5,6) with a low frequency boost has also been included as a sound source. Boards PS5 and PS6 connect to PS4 via preformed 12" flexible jumper links. Make certain that these links are clamped to the boards with self-adhesive ribbon cable clamps, as these will take the strain of the cables as they are flexed.



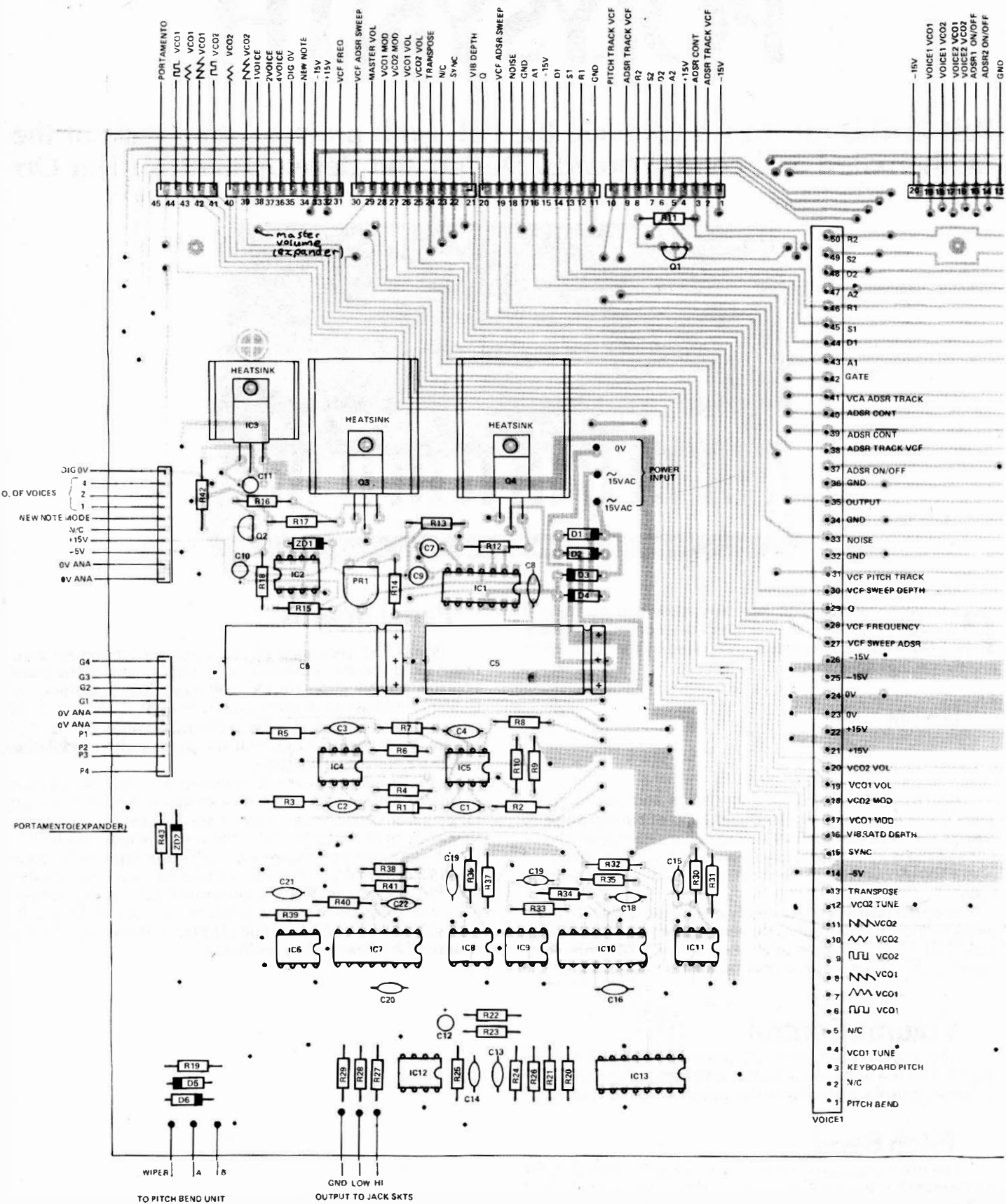
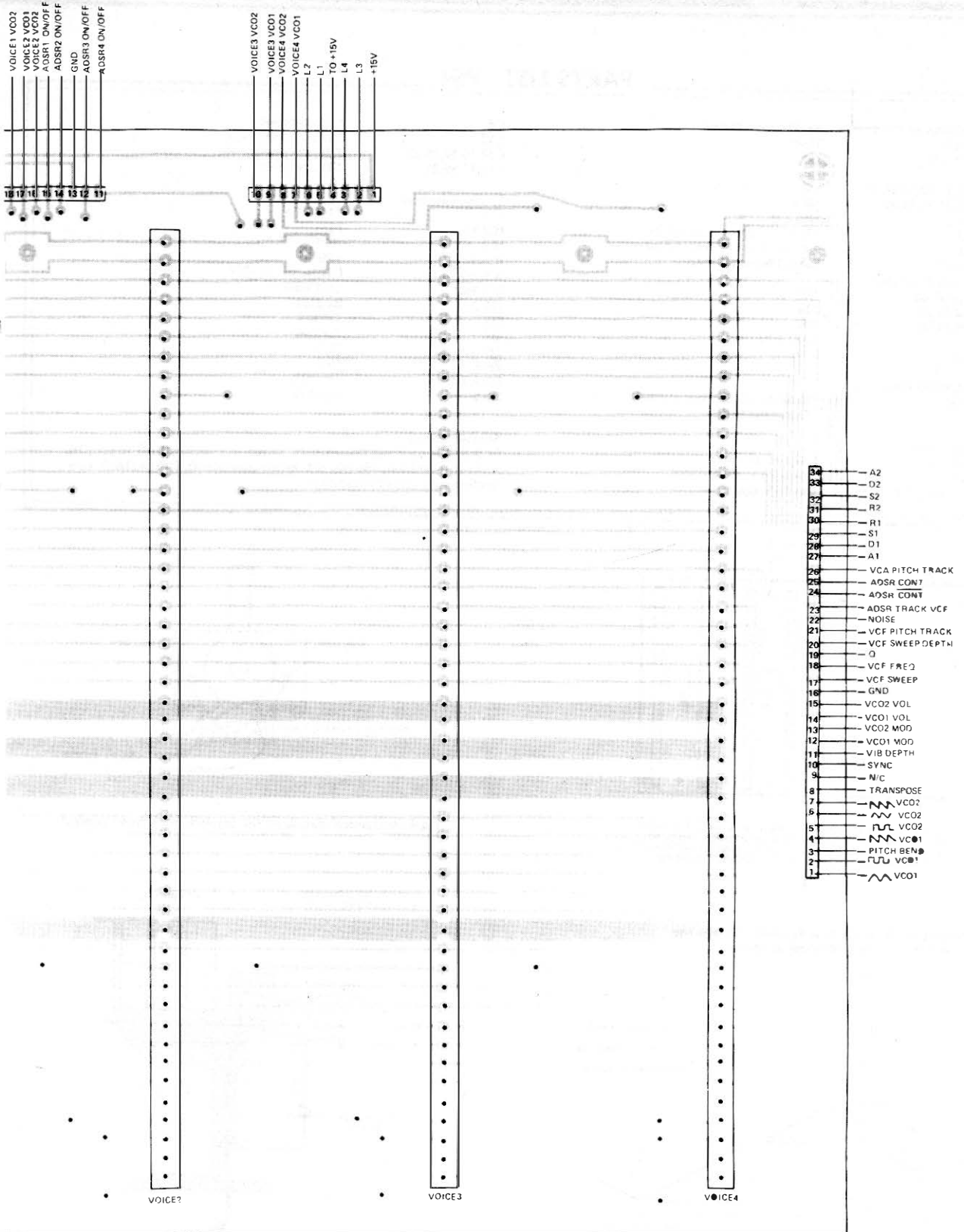


Fig.1 Component overlay of the motherboard (PS4). The circuit diagram was published last month.

# PROJECT: Polysynth





## PARTS LIST : PS4

Resistors all 2% M.O. unless otherwise stated

R1,3,5,7	1M0
R2,4,6,8	1k5
R9	1k2
R10,26,30,33,36,39	15k
R11,32,35,38,41	100k
R12,17	1R02 W
R13	12k 1% 50ppm metal film
R14	10k 1% 50ppm metal film
R15,18	10k 0.5% metal film
R16,20,21,27,29,31, 34,37,40	1k0
R22,23,25	22k
R19,24,28	10k
R42	100R 1 W
R43	270k

Potentiometers

RV33	5K or 10k linear (single axis joystick)
PR1	10k cermet preset

Capacitors

C1,2,3,4	1n0 ceramic
C5,6	2200u 25 V electrolytic
C7,9,10,11,12	10u 16 V tantalum

C8

C13

C14,16,18,20,22

C15,17,19,21

330p ceramic

1n0 polycarbonate

220n polycarbonate

470p ceramic

Semiconductors

IC1

IC2,12

IC3

IC4,5

IC6,8,9,11

IC7,10,13

Q1,2

Q3

Q4

ZD1

ZD2

D1,2,3,4

D5,6

723 (DIL)

741

7905

1458

LF353 or TL082

LM13600

BC212L

TIP30A

TIP29A

10 V

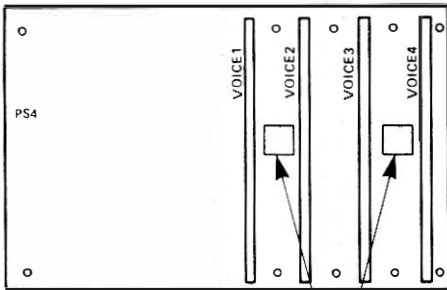
5V1

1N4002

1N4148

Miscellaneous

PCB PS4, 8 off 8 pin DIL sockets, 14 pin DIL socket, 3 off 16 pin DIL sockets, 2 off heatsinks (RS 401-964 or equivalent), TV5 heatsink, connecting pins, self-adhesive clips.



STICK TWO SELF ADHESIVE RUBBER FEET ONTO THE METAL BASE OF THE SYNTHESIZER UNDERNEATH PS4. THESE WILL HELP TO LIMIT THE CONSIDERABLE FORCES GENERATED WHEN THE VOICE BOARDS ARE INSERTED OR REMOVED FROM PS4

Fig.2 Plugging in the voice boards could damage the motherboard if it is not cushioned as shown.

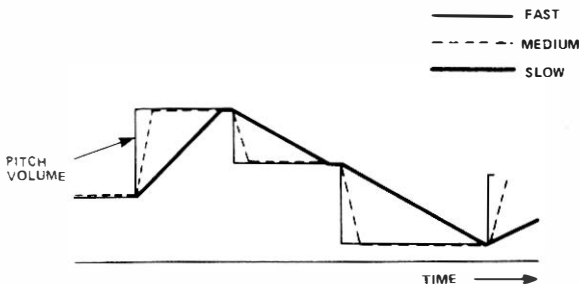


Fig.3 The effect of variable slew limiting (portamento) on pitch voltage.

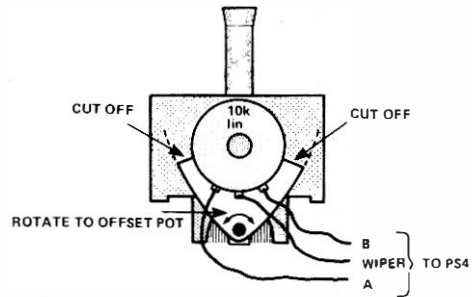


Fig.4 Mechanical details of the joystick pitch bend control.

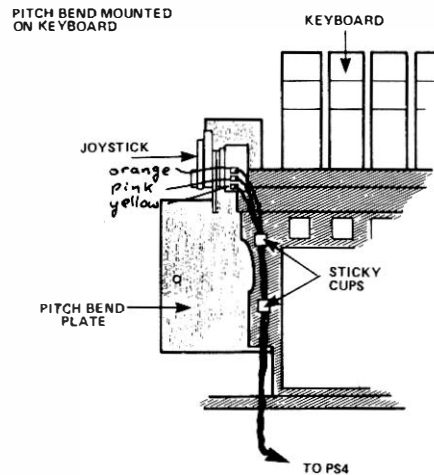


Fig.5 Mounting the joystick pitch bend control (RV33) on the end of the keyboard.

# PROJECT: Polysynth

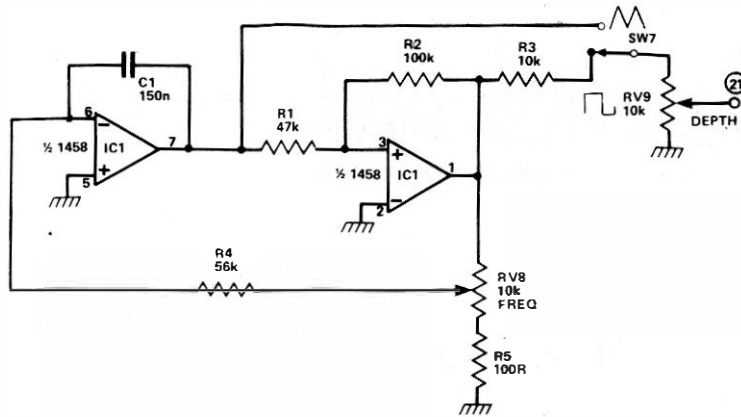


Fig.6 (left) Circuit diagram of the vibrato oscillator on Control Board PS5 (0.5 Hz - 50 Hz).

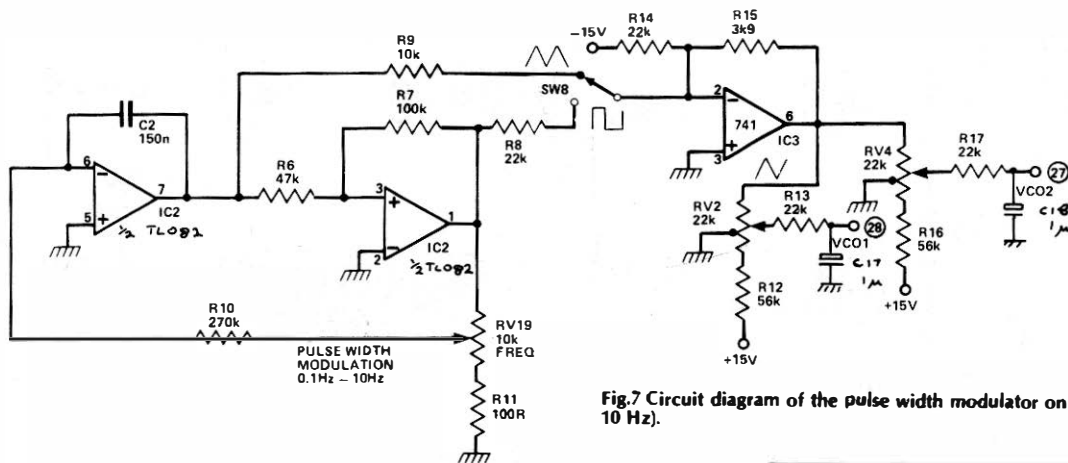
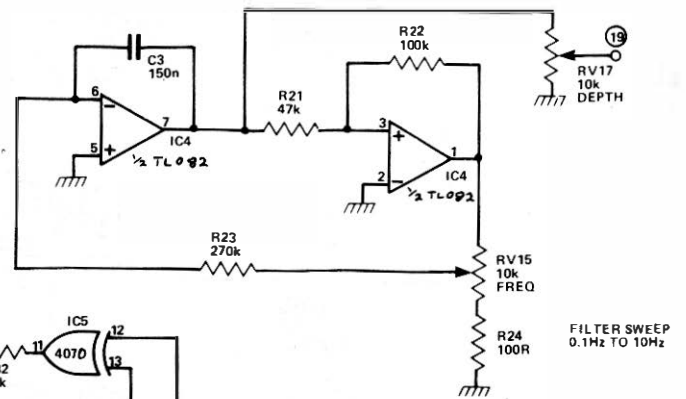


Fig.7 Circuit diagram of the pulse width modulator on PS5 (0.1 Hz - 10 Hz).

Fig.8 (right) Filter sweep circuit on PS5 (0.1 Hz - 10 Hz).



FILTER SWEEP 0.1Hz TO 10Hz

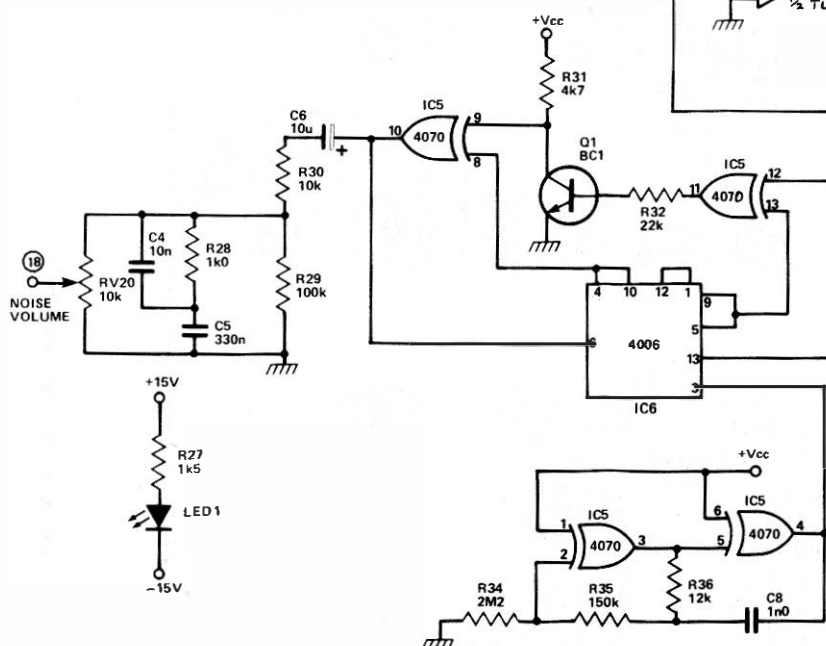


Fig.9 (left and below) Noise volume and resonance control networks on PS5.

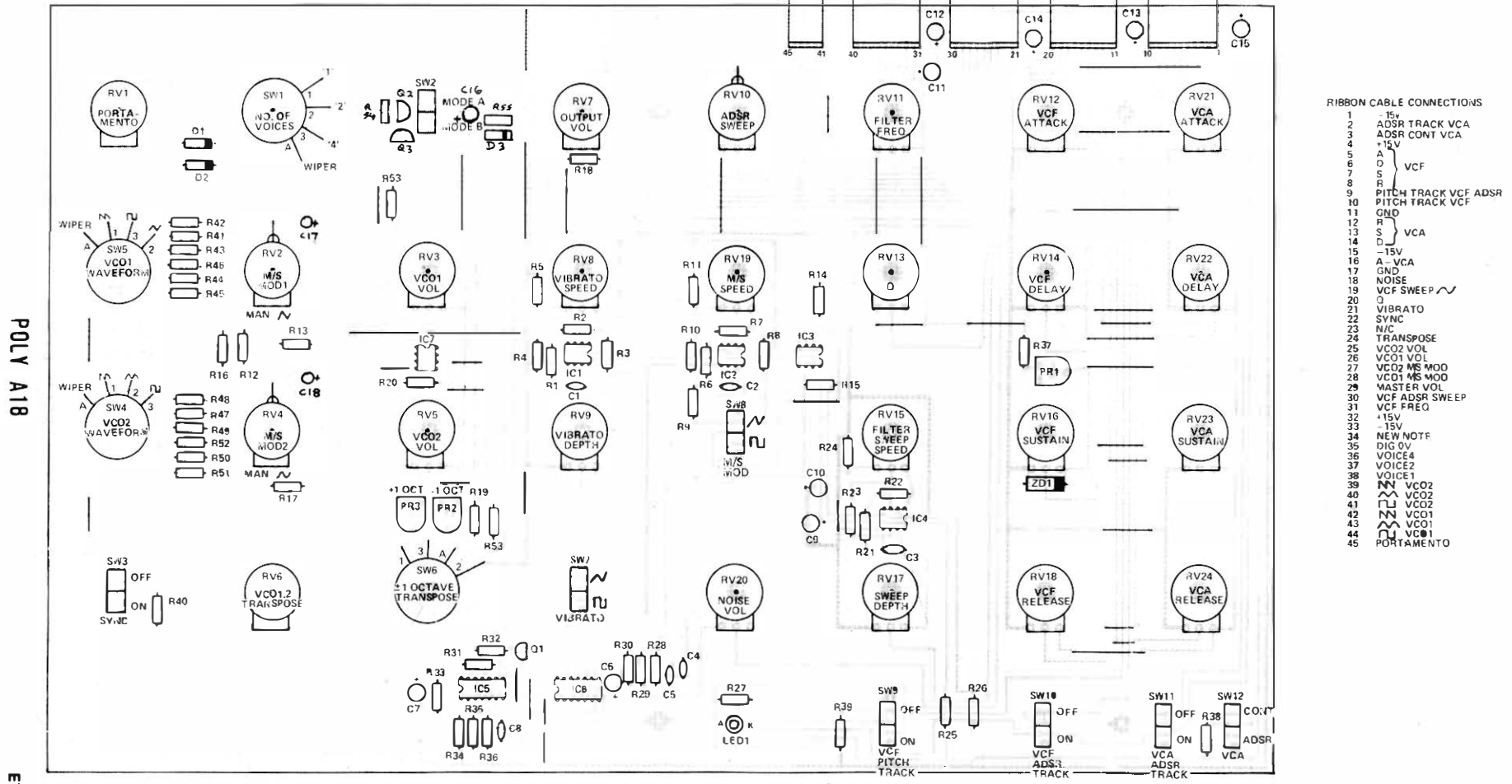


Fig.10 Component overlay of control board PSS.

**PARTS LIST : PS5**

Resistors all 27. P.D. unless otherwise stated

R1,6,21,39,54	47k
R2,7,22,29,38,40-52	100k
R3,9,30	10k
R4,12,16	56k
R5,11,24,33	100R
R8,13,14,17,32	22k
R10,23	270k
R15	3k9
R18,31	4k7
R19	100k 1% metal film 50ppm
R20	10k 1% metal film 50ppm
R25,26,53	47R
R27	1k5
R28	1k0
R34	2M2
R35	150k
R36	12k
R37	6k8
R55	1M

Potentiometers

RV1,8,15,19,20	10k logarithmic
RV2,4,10	22k linear with central click and central tapping

RV3,6,5,7,9,11,13,16,17,23	10k linear
----------------------------	------------

RV12,14,18,21,22,24	4k7 linear
---------------------	------------

PR1	10k cermet preset
PR2,3	100k cermet preset

Capacitors

C1,2,3	150n polycarbonate
C4	10n polycarbonate
C5	330n polycarbonate
C6,7,9,10-16	10u 16 V tantalum
C8	1n0 polycarbonate
C17,18	1u tantalum

Semiconductors

IC1	1458
IC3,7	741
IC5	40708
IC6	40068
Q1	BC182L
D1,2	1N4148
ZD1	5V1
LED1	0.2" green
LED2,4	T.C.02
D3	1N4002

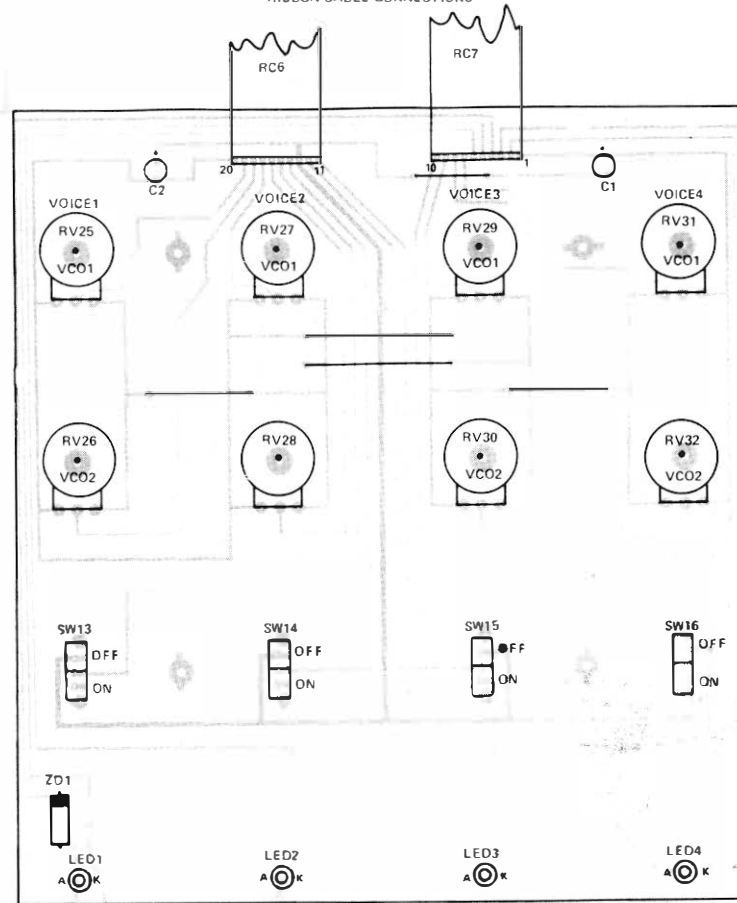
Switches

SW1	3-pole 4-way rotary
SW2,3,7,8,9,10,11,12	Single-pole double-throw slide
SW4,5,6	4-pole 3-way rotary

Miscellaneous

PCB PS5, connectors, self-adhesive clips

RIBBON CABLE CONNECTIONS



RIBBON CABLE CONNECTIONS

1	+15V
2	LED3
3	LED4
4	+15V
5	LED1
6	LED2
7	RV31
8	RV32
9	RV29
10	RV30
11	SW15
12	SW14
13	GND
14	SW13
15	SW12
16	RV28
17	RV27
18	RV26
19	RV25
20	-15V

**PARTS LIST : PS6**

Potentiometers

RV25-32	10k linear
---------	------------

Capacitors

C1,2	10u 16 V tantalum
------	-------------------

Semiconductors

ZD1	10 V
LED1-4	0.2" red

Switches

SW13-16	Single-pole double-throw slide
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Miscellaneous

PCB PS6, ribbon cable connectors, self-adhesive cable clamps

Fig.11 Component overlay of control board PS6.



# PROJECT: Polysynth

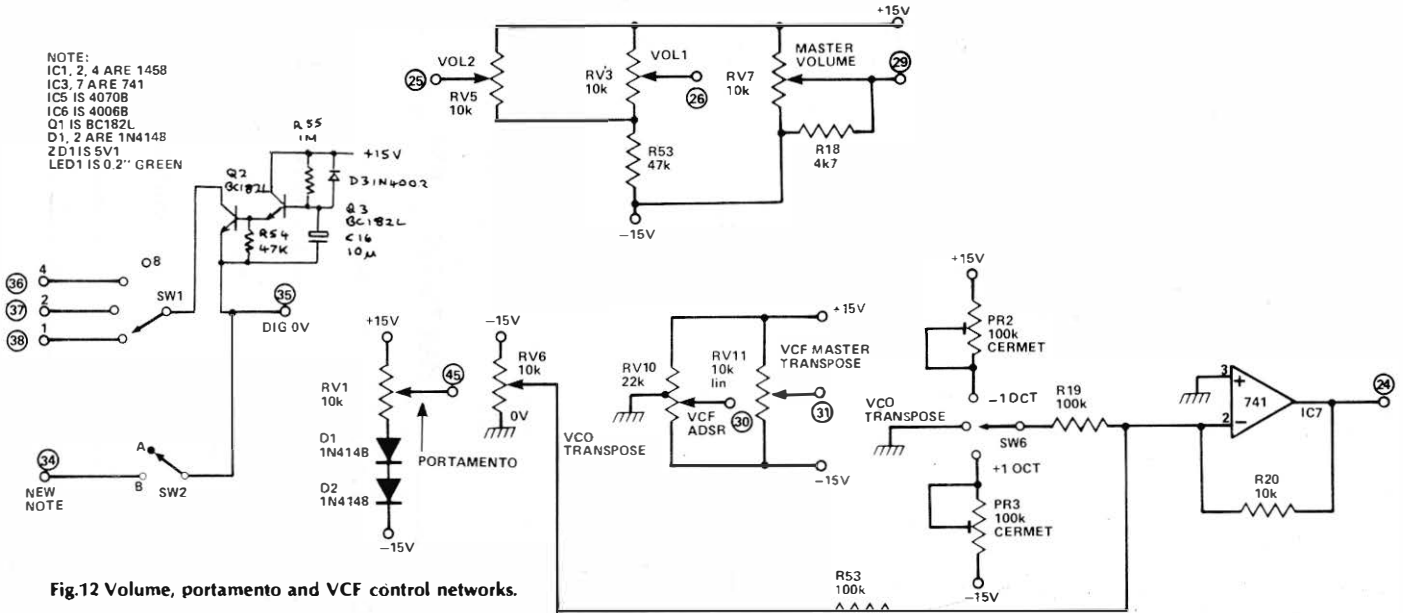


Fig.12 Volume, portamento and VCF control networks.

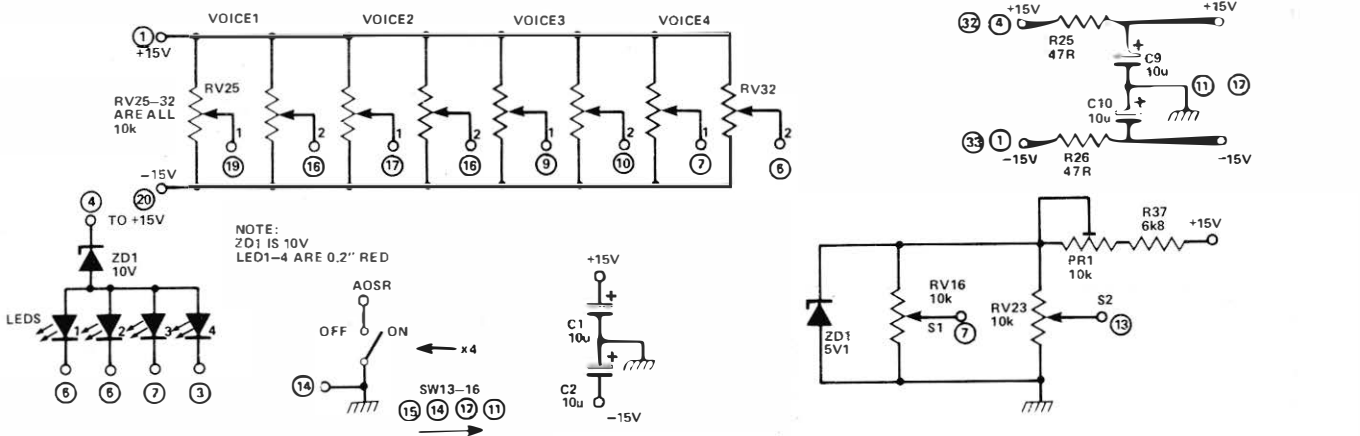


Fig.13 RV25 to RV32 are the VCO1,2 controls on control board PS6.

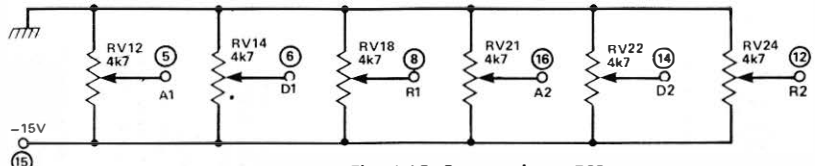


Fig.14 ADSR networks on PS5.

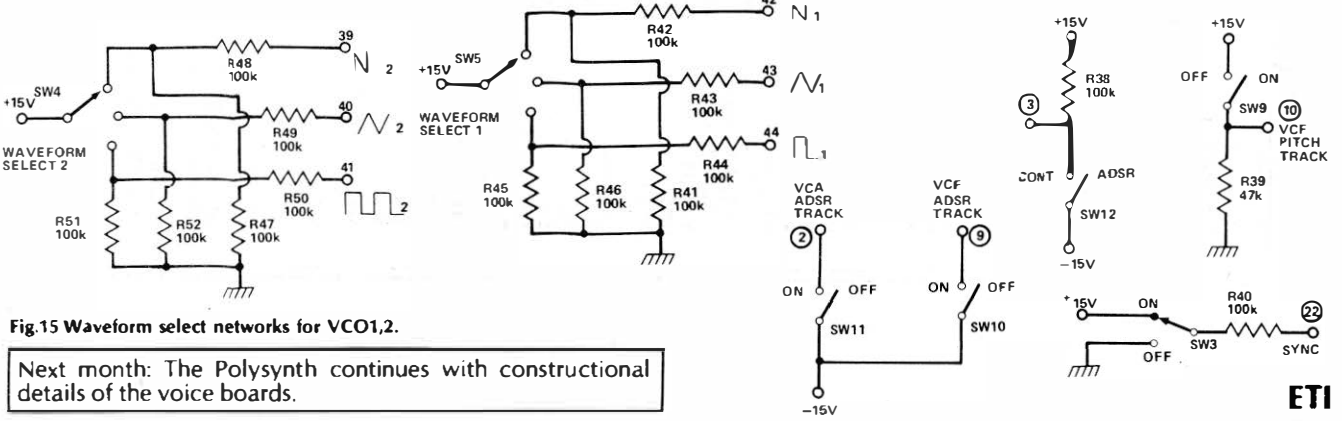


Fig.15 Waveform select networks for VCO1,2.

Next month: The Polysynth continues with constructional details of the voice boards.

# POLYSYNTH



Part 3 of the Polysynth describes the voice boards and power supply, and shows how the four-voice expander unit may be built from the basic boards. Design and development by Tim Orr.

The Polysynth can generate up to four individual voices or eight voices when using the expander unit. Each voice contains the equivalent electronics of a medium-sized monophonic synthesiser. Dedicated music synthesiser integrated circuits have been used to enable voltage control of the VCO and ADSR units.

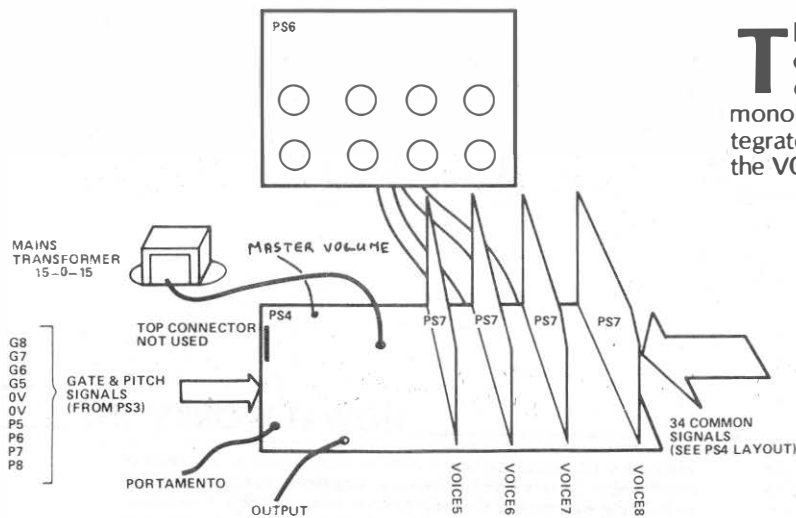


Fig.1 (Above) Block diagram showing the boards and connections needed for the four-voice expander unit.

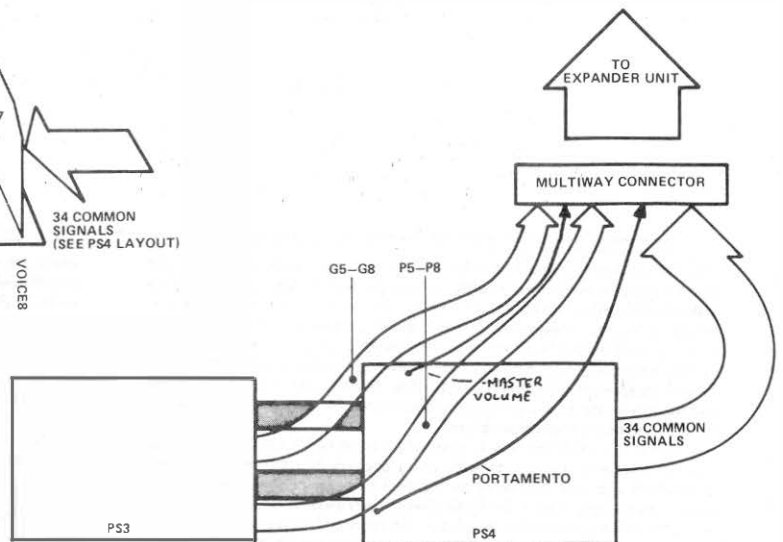
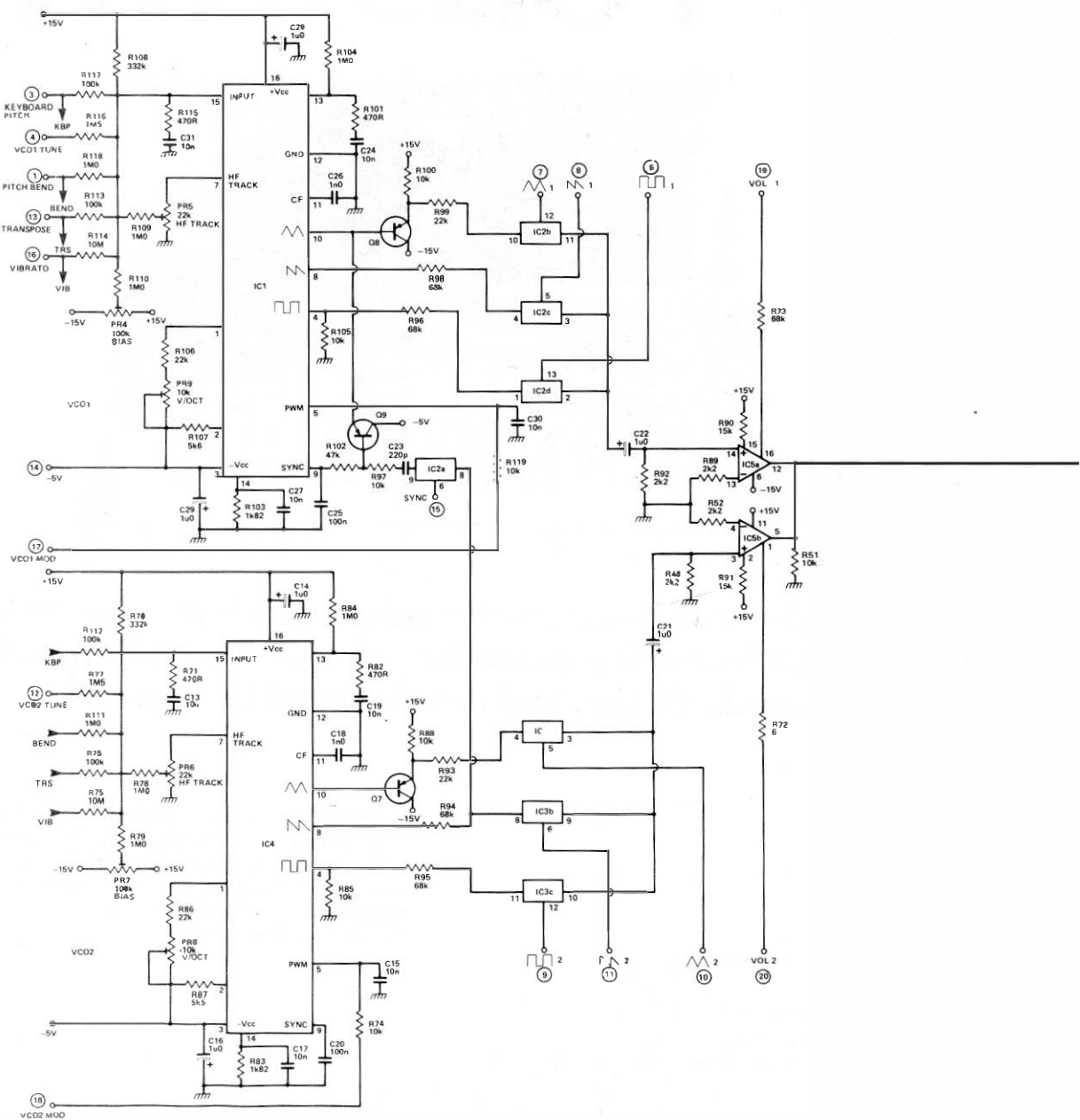


Fig.2 (Right) The control signals for the expander are taken from the Polysynth via a multiway connector.



## HOW IT WORKS : PS7

The VCOs are Curtis CEM3340 devices (IC1,4). These can generate three output waveforms; a ramp, a triangle and a square wave with a voltage-controlled mark/space ratio. The VCO has an internal exponentiator that converts the linear input voltages of 1 V per octave into musical intervals of one octave. The frequency control input is a virtual earth amplifier and so other frequency control voltages may be mixed resistively into this point. The device also has an internal temperature compensation circuit that minimises frequency and scale factor/temperature drift problems.

It is a very difficult task to produce a bank of VCOs that will track over a wide musical range (about seven-and-a-half octaves in this case) and which will not drift in pitch relative to each other or against absolute frequency. The CEM3340 devices perform this task as well as any other VCOs currently available. To ensure that the VCOs remain in tune when they are transposed it is necessary to match the 'transpose' and 'keyboard voltage' resistor pairs. These resistors are R117 and R113 (for VCO1) and R112 and R76 (for VCO2). The resistors

used are 0.1% tolerance and *if possible* they should be matched to 0.01%. Also, be very careful when inserting these resistors. Use pliers to hold the wire next to the resistor body when bending it and don't take too long soldering it in. Thermal and mechanical stress can change the resistor value.

An analogue switch (IC2,3) is used to select the output waveforms from the VCOs. These signals are fed to IC5 which is used as a voltage-controlled amplifier. The VCO outputs are mixed together and fed into the VCF. This is known as a two-pole state variable lowpass filter with exponential frequency control and voltage-controlled resonance (Fig. 6). Two OTAs (IC10) are used as variable gain integrators. The gain is linearly proportional to the  $I_{ABC}$  current flowing into pins 1 and 16 of IC10. The filter cut-off frequency is linearly proportional to this gain. Therefore, a change in the  $I_{ABC}$  current will result in a similar change in the VCF frequency. Transistors Q5,6 convert the transpose, keyboard, frequency pot and sweep voltages into an  $I_{ABC}$  current (the

# PROJECT : Polysynth

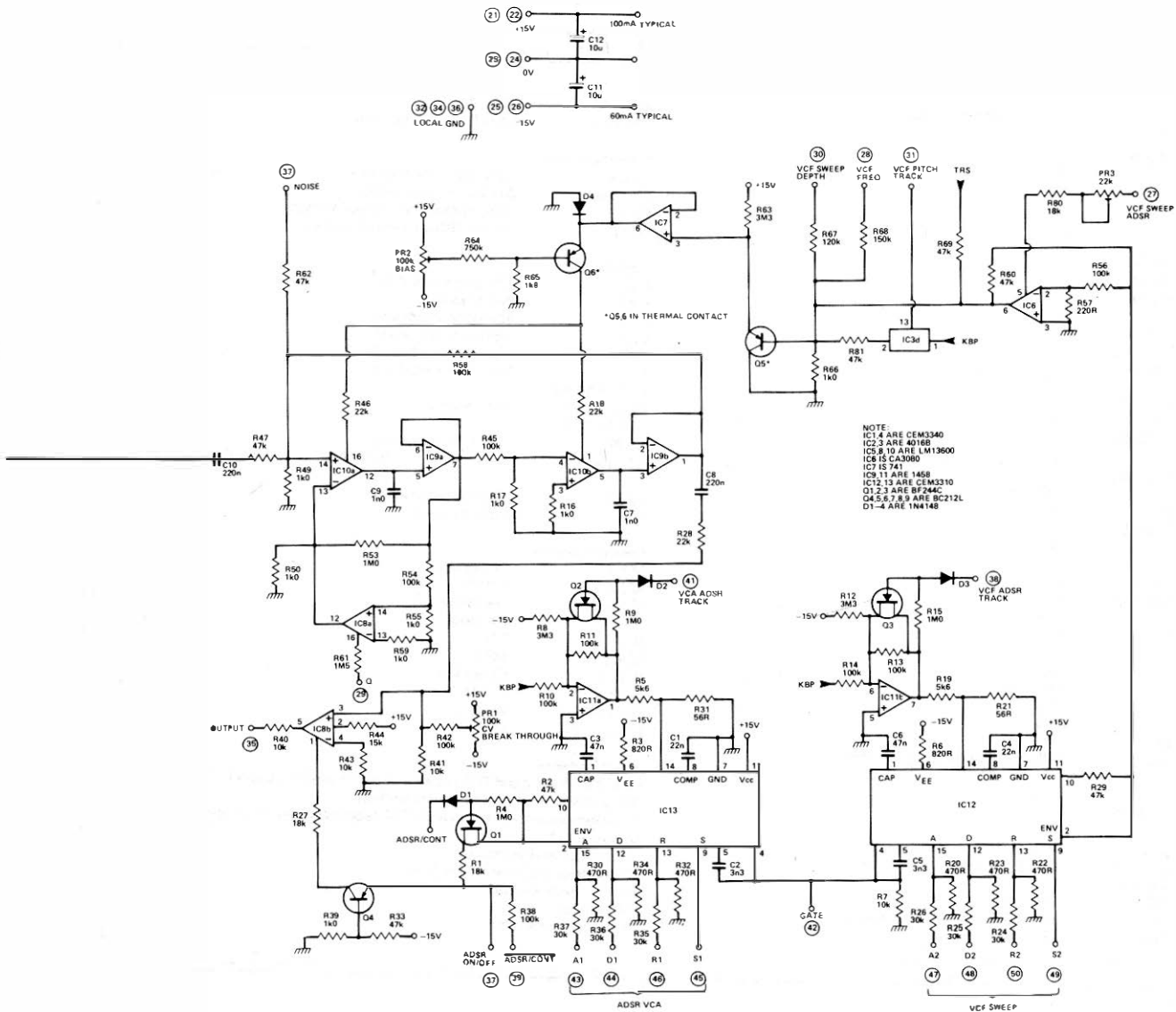


Fig.6 Complete circuit diagram for one voice of the Polysynth (PS7). The ringed numbers refer to the edge connector sockets (see Fig.7).

collector current of Q6) which is exponentially proportional to the sum of these as seen at the base of Q5. IC8 (pins 12, 13, 14, 16) is used to voltage-control the Q factor of the filter. When the  $I_{ABC}$  current to this device (pin 16) is zero, the Q factor is determined by R50 and R53, resulting in a high Q response. IC8 provides a negative feedback route so that as its  $I_{ABC}$  current is increased, more negative feedback is applied. This damps the filter which lowers the Q factor.

It is possible to sweep the filter with an ADSR waveform via IC6. When the ADSR sweep pot (molex pin number 27) is at 0 V, PR3 is adjusted so that an ADSR waveform produces no movement at IC6, pin 6. When the ADSR sweep pot is then set to -15 V, IC6 is turned off and so an ADSR waveform will generate a positive change at pin 6 via resistor R60. When the ADSR pot is set to +15 V, IC6 is turned fully on and so the ADSR waveform generates a negative change at pin 6.

This circuit generates the characteristic synthesizer-swept filter sound.

There are two ADSR units, IC12, 13. These are Curtis CEM3310 devices. They have a 50,000:1 time constant control range, with voltage control of all parameters and a true RC exponential envelope shape. Also, the voltage control of the time-constants is exponential. Every 18 mV increase at pins 15, 12 and 13 halves the A, D and R time-constants respectively. The time-constants of the ADSR units can also be transposed by injecting a voltage at pin 14. A +18 mV increase at this point will double the time-constants. All natural instruments have attack and decay times that are frequency-related. The top note on a piano dies away very quickly but the bottom note continues for a long time. It is possible to simulate this in the voice unit by injecting the keyboard pitch voltage into pin 14 of the CEM3310. This ADSR pitch tracking may be turned off by using FETs Q2 and Q3 to short out the keyboard pitch voltage.

IC12 is used to provide a sweep voltage for the VCF; IC13 generates the amplitude envelope for the output VCA (IC8 pins 1, 2, 3, 4, 5).



## PARTS LIST : PS7

Resistors all 2% M.O. unless otherwise stated

R1,27,80	18k	
R2,33,47,60,		
62,102,29	4.7k	
R3,6	820R	
R4,9,15,53,		
78,109	1M0	
R5,19	5k6	
R7,40,41,43,		
51,74,85,88,		
97,100,105,		
119	10k	
R8,12,63	3M3	
R10,11,13,14,		
38,42,45,54,		
56,58	100k	
R16,17,39,49,		
50,55,59	1k0	
R18,28,46,93,		
99	22k	
R20,22,23,30,		
32,34	470R *	
R21,31	56R	
R24,25,26,35,		
36,37	30k *	
R44,90,91	15k	
R48,52,89,92	2k2	
R57	220R	
R61,77,116	1M5	
R64	750k	
R65	1k8	
R66	1k0 *	
R67	120k	
R68	150k *	
R69,81	47k *	
R70,108	332k 0.5% metal film 25ppm	
R71,82,101,		
115	470R	
R72,73,94,95,		
96,98	68k	
R75,114	10M	
R76,112,113,117	100k 0.1% metal film 25ppm	
R79,84,104,		
110,111,118	1M0 *	
R83,103	1k82 0.5% metal film 25ppm	
R86,106	22k *	

\* 1% 50ppm metal film

R87,107	5k6 *
R113,117	100k 0.1% metal film

<b>Potentiometers</b>	
PR1,2	100k horizontal preset
PR3,5,6	22k horizontal preset
PR4,7	100k cermet horizontal preset
PR8,9	10k multiturn cermet preset

<b>Capacitors</b>	
C1,4	22n polycarbonate
C2,5	3n3 ceramic
C3,6	47n polycarbonate
C7,9	1n0 polycarbonate
C8,10	220n polycarbonate
C11,12	10u 16 V tantalum
C13,15,17,19,	
24,27,30,31	10n ceramic
C14,16,21,22,	
28,29	1u0 35 V tantalum
C18,26	1n0 5%, 30ppm ceramic multilayer

C20,25	100n polycarbonate
C23	220p ceramic

<b>Semiconductors</b>	
IC1,4	CEM3340
IC2,3	4016B
IC5,8,10	LM13600
IC6	CA3080
IC7	741
IC9,11	1458
IC12,13	CEM3310
Q1,2,3	BF244C
Q4,5,6,7,8,9	BC212L
D1-4	1N4148

**Miscellaneous**  
PCB PS7, 4 off 8 pin DIL sockets, 2 off 14 pin DIL sockets, 7 off 16 pin DIL sockets, 5 off 10 way Molex sockets.

Transformer with 17 0-17 and 0-7.65 tapings (Powertran special)

Mains switch

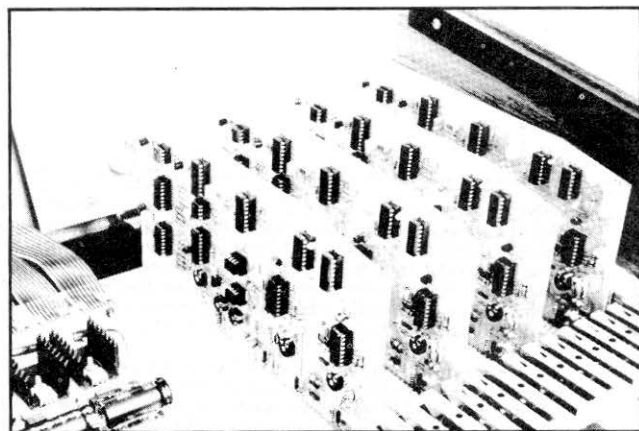
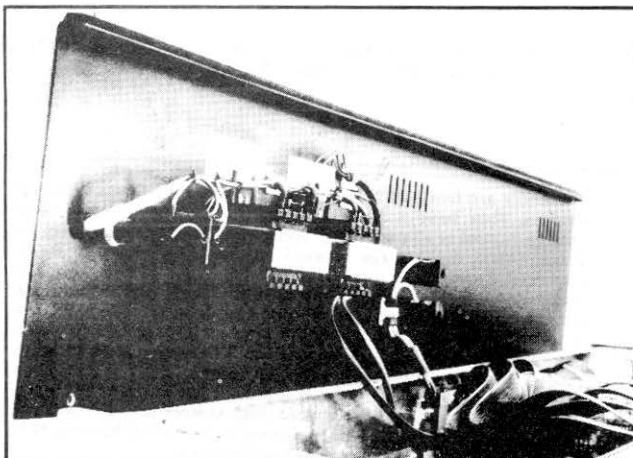
Mains lead and grommet

20 mm fuse and fuseholder (1 A)

2 off 1/4" jack sockets

1k0 resistor

36 instrument knobs



Above: prototype back panel. Production models may differ.  
Right: the voice boards plug directly into the mother board.

Next month: We conclude the Polysynth project with full setting up and test details.

# PROJECT : Polysynth

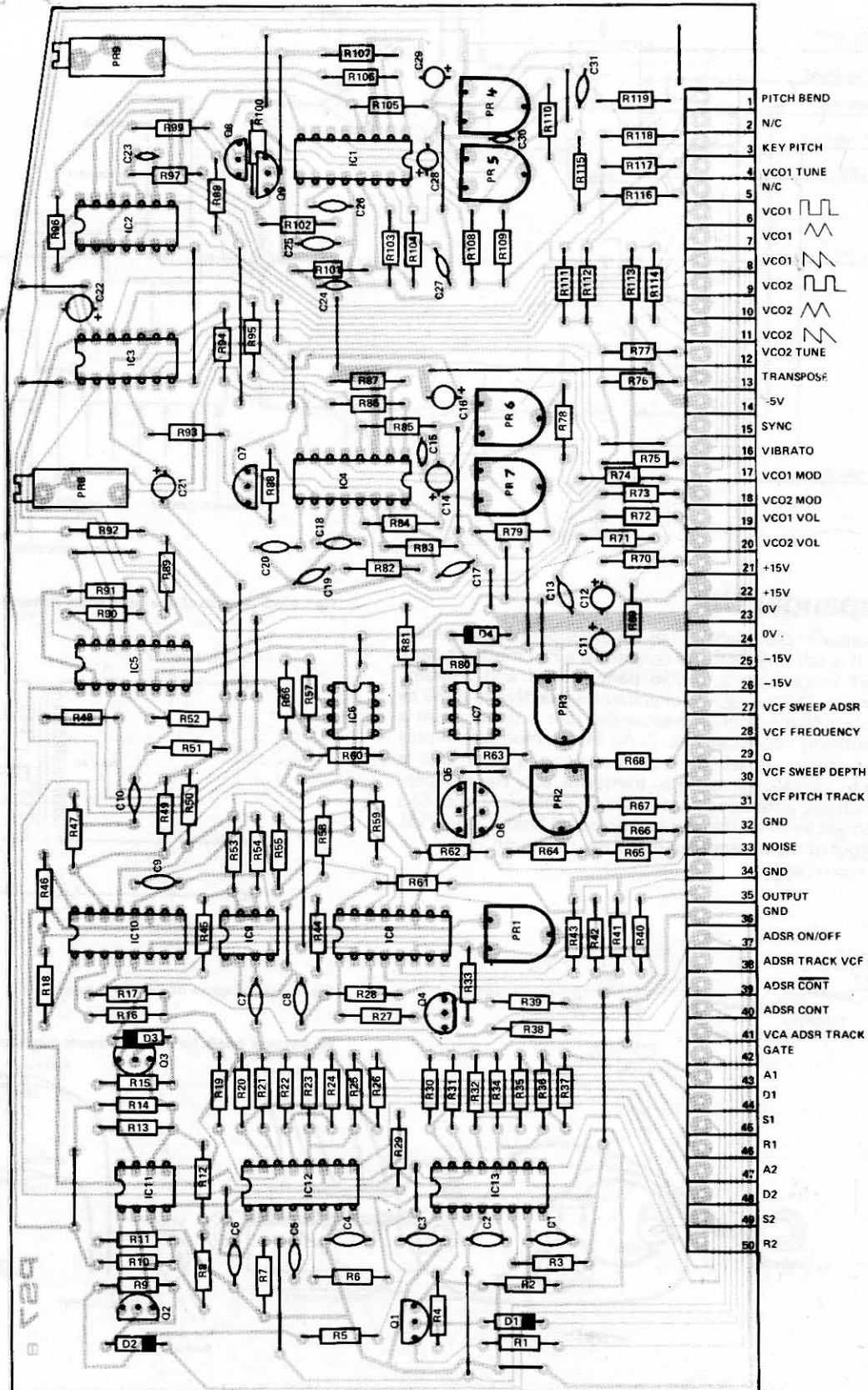
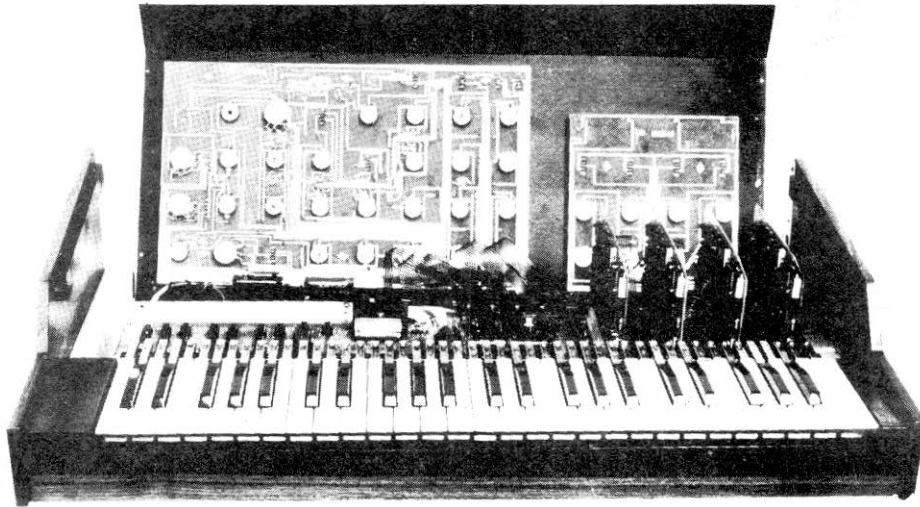


Fig.7 Component overlay for the PS7 voice board. Take care with the orientation of the ICs, especially IC1 and IC4. One corner of the board is cut off to avoid fouling the panel boards.

ETI

# POLYSYNTH



We conclude the Polysynth project with the final setting up and alignment procedure. Design and development by Tim Orr.

Assuming that the rest of the synthesiser has been checked out and found to be working then the voice boards can be tested and aligned. When inserting or removing the voice boards make sure that the power is always turned off. Set up the panel as shown in Fig. 4. Insert a voice into slot number four with the component side facing the centre of the machine, the copper track side facing the wooden end. Make certain that the ICs are the correct way around, in particular IC1,4. Turn on the power and check the  $\pm 15$  V and -5 V rails on the voice board. Both VCOs should be oscillating. Check pins 4,8 and 10 for square, ramp and triangle waveforms. Next look at IC2 pin 2 and IC3 pin 3 and check that the two waveform selectors function properly. Also check that the two VCO tuning pots control their respective VCO frequencies. When the machine has been calibrated, these pots will have a two octave tuning range.

Check that the two transpose controls affect the VCO frequency. Move the pitch-bend lever; this will slightly change the VCO pitch. Also check that the keyboard controls the

pitch, although it will not yet be in tune. Test out the three vibrato controls. Turn off the vibrato and tune the two VCOs to the same frequency. They should slowly beat with each other. Look at IC 5 pin 5 (the top of R51). Check that the level controls for each VCO operate correctly. Turn both of them on. Turn on the sync switch. VCO1 should lock onto the frequency of VCO2. Turn off the sync and turn off the volume to VCO1. Select a square waveform from VCO2. Test the VCO2 MS (mark/space) control pot. With the pot anticlockwise the waveform will be square. As the pot is rotated to its central position the square will turn into a thin pulse. Clockwise of centre the pulse width is controlled by the mark/space oscillator. Check out the mark/space speed and waveform controls. Repeat for VCO1.

Select a 100 Hz ramp waveform from VCO1. Turn VCO1 level to maximum, and VCO2 level to off. Look at the VCF output, IC9 pin 1 (the left hand side of R58). The VCF frequency pot will vary the filter cut-off frequency, and the resonance control will vary the Q factor (Fig. 5). Press a note on the keyboard. This will generate the ADSR sweep waveform as shown in Fig. 2. Adjust PR3 so that with the ADSR sweep pot in its central position there is no VCF sweep. Now rotate the

Fig.1 VCF frequency response.

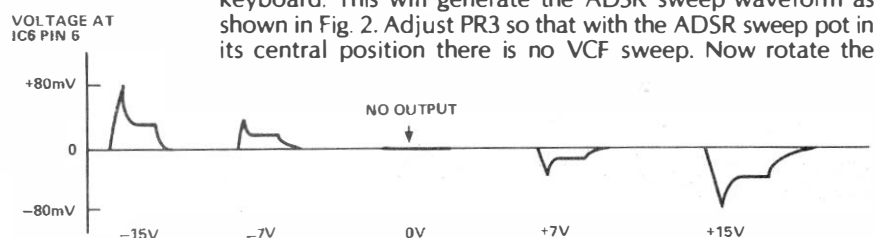
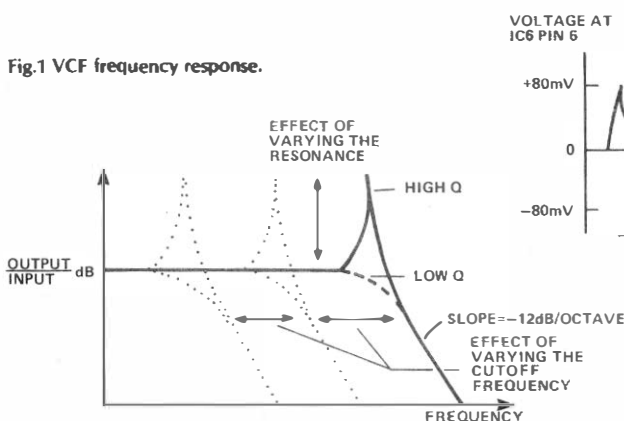


Fig.2 Waveforms associated with the ADSR sweep potentiometer.

ADSR sweep pot anticlockwise. When the note is pressed it should be possible to get a filter sweep that sounds like a 'DOW' noise. In the clockwise position the sound is a 'WAH'. Check out the VCF ADSR controls. They should behave as shown in Fig. 3. Also test the TRACK switch. This will generate

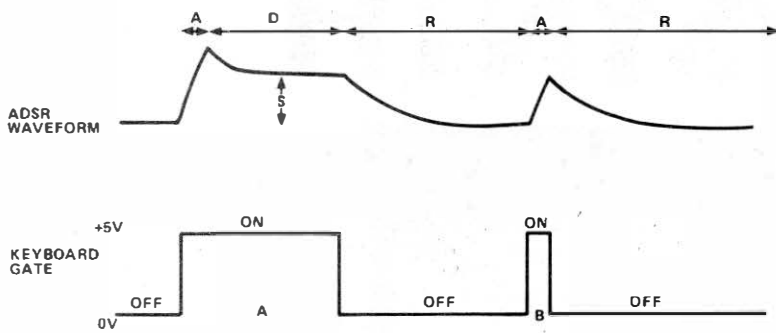


Fig.3 ADSR operation.

fast time-constants at the top end of the keyboard and slower ones at the bottom end. Now check out the VCF TRACK switch. Turn it on and play notes up and down the keyboard. The shape of the waveform at the VCF output will remain roughly the same as the frequency varies. But with the VCF TRACK off, the high notes will be sinusoidal, but the low notes will contain a strong harmonic content. Turn the VCF TRACK switch on. Turn up the noise level to test that it makes it to the filter.

The next and last section is the VCA. Turn off both the VCOs, the noise source and the VCF sweep. Set up the VCA ADSR as shown in Fig. 4. Press a note on the keyboard. This will start the ADSR which generates a fast envelope contour, causing a click at the VCA output, IC8 pin 5. Adjust PR1 until this click reaches a minimum. Turn on VCO1 so that the VCA has a signal to modulate. Test the VCA ADSR controls and the TRACK switch. When the note is released and the ADSR waveform has decayed away the output of the VCA will die away completely. Turn the ADSR/CONT switch to CONT. The sound will return and will be unaffected by the VCA ADSR. Now turn the relevant voice ON/OFF switch to OFF. The voice will now be off.

This concludes the initial alignment and debugging of the voice. Repeat all of this process for voices 3,2 and 1 until all four voices are plugged in and working. Allow the machine to 'burn in' for 24 to 48 hours, then retest all the functions.

The next section deals with aligning the VCF and VCOs for frequency and tuning.

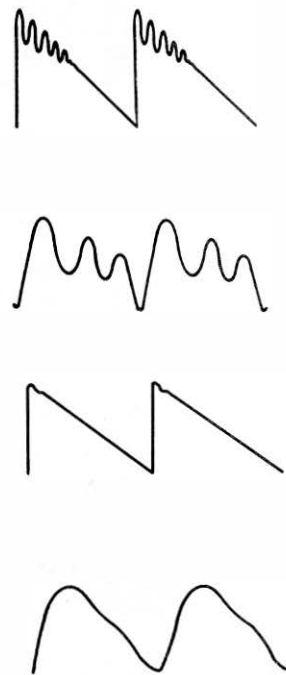


Fig.5 VCF response. From top to bottom: high frequency, high Q (resonance control clockwise); low frequency, high Q; high frequency, low Q (resonance control anticlockwise); low frequency, low Q.

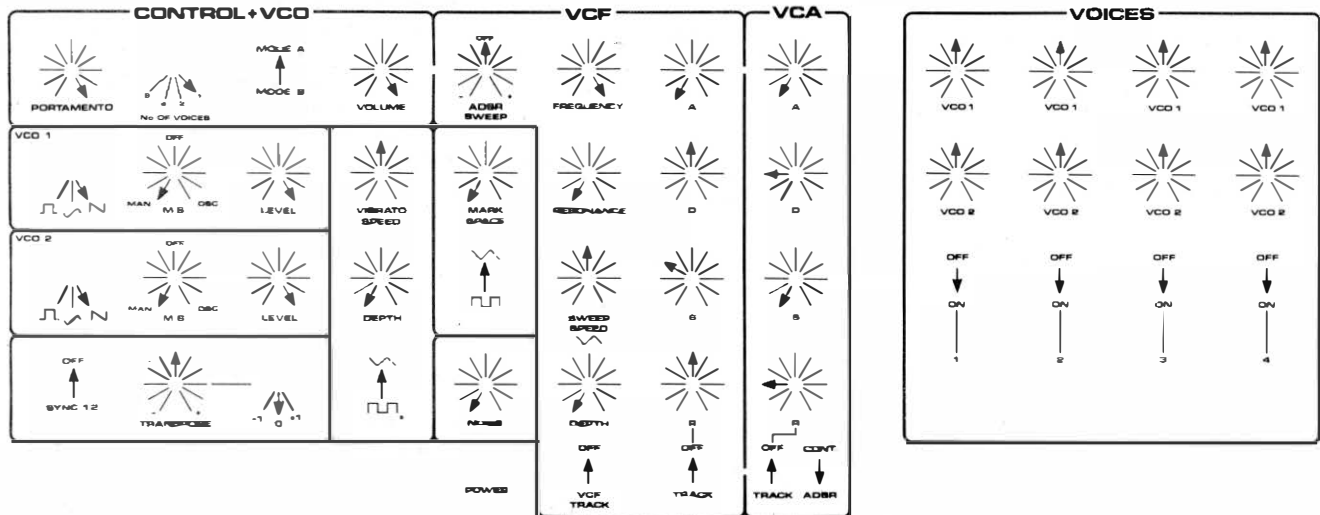


Fig.4 Front panel control positions for setting up procedure.

# PROJECT : Polysynth

The best method is to tune the VCOs relative to a fixed tone. I use a crystal oscillator divided down to 400 Hz. You can mix this with the VCO output so that you can hear the beats, or even better you can display the two frequencies on a dual beam oscilloscope.

## Oscilloscope Method

Display the 400 Hz fixed reference squarewave on one beam and sync from it. Display the VCO to be aligned on the other beam. Press the bottom note on the keyboard and set the VCO to 100 Hz. The VCO output will remain almost stationary relative to the reference squarewave. It will drift slowly to the left or to the right, which should be corrected by fine tuning the VCO. Play a note one octave up and adjust the PR9 preset so that the VCO output is stationary (ie 200 Hz) relative to the reference signal. Now go back to the bottom note. The pitch of this will have been changed by PR9, and so retune the VCO (to 100 Hz) with the fine tuning pot. Repeat the process again and again until the VCO interval converges to one octave. When altering PR9 it is best to overcorrect as you will then converge more rapidly. Now the tuning can be more finely set up by repeating the process for higher octaves. When finally set up the VCO output will be almost static relative to the reference tone on all five octave notes. Best results are obtained by tuning the VCO to be static relative to the reference tone at the top end of the keyboard. In fact when tuning up a synthesiser, musicians always tune up the VCOs for unison at the top of the keyboard. Then any pitch spread errors will cause minimum beat frequencies. Tuning the VCOs at the bottom of the keyboard generates maximum beats.

Repeat the entire tuning process for VCO2 using PR8 to adjust the pitch spread. Then tune the other voices. If all the VCOs track relative to a fixed reference then they will track with each other. Select one-voice mode, using both VCOs. Turn on all four voices and press the top note on the keyboard. Tune all the VCOs to 1600 Hz so that they are slowly beating together (a total of eight VCOs). Now play the lower notes down the keyboard. If the VCOs track then they will continue to slowly beat. The pitch spread tuning should be such that over the keyboard's range the beat rate does not exceed 1 to 2 Hz. If the VCOs track properly the synthesiser can now be switched to four voice polyphonic operation.

## Octave Transpose Switch

Set the octave transpose switch to 0. Tune a VCO to 200 Hz so that it slowly beats with the reference 400 Hz. Turn the octave transpose switch to +1 and adjust the preset (PR3 on board PS5) for an exact one-octave increase. Now set the switch to -1 and adjust the other preset (PR2 on board PS5) for an exact one-octave decrease. Set the number of voices to one, the octave transpose to 0 and turn on and tune all eight VCOs to be in unison. Now try the effect of the transpose switch and pot. All the VCOs should be transposed without a significant or objectionable increase in the beat rate. If the beat rate does become objectionable it will be because of an inaccurate transposition in one or more of the VCOs. This is due to the mismatch of resistor pairs R117/R113 or R112/R76 on the voice board, which should be matched to 0.01% for optimum results.

## VCO Bias

Set the transpose pot and switch to maximum. Set all the tuning pots to maximum and play the top note on the keyboard. Adjust PR4 (for VCO1) and PR7 (for VCO2) on each voice board for a VCO pitch of 4 kHz. This is the maximum frequency of operation for the machine.

## VCF Bias

Select one-voice operation. Tune all the VCO1s to 400 Hz (ramp waveform). Turn on the VCF TRACK switch. Set the AD-SR SWEEP pot to off and the VCF frequency pot to its central position. Turn the resonance control to maximum. Adjust PR2 so that the VCF rings with the eighth harmonic of the ramp (3200 Hz). Repeat this for the other voices. Now try altering the filter frequency. The VCF on each voice should generate the same tone.

## HF Track

Set the bottom note of the keyboard to 200 Hz and tune it against the 400 Hz reference note. Now play the top note (3200 Hz). The VCO may have gone slightly flat in which case adjust the HF TRACK preset to restore the high frequency tuning. Repeat this for every VCO. The Curtis data sheet recommends aligning the HF tracking at 10 kHz. This is, however, outside the tuning range of the Polysynth. At 4 kHz (the maximum frequency of the machine) it may not be necessary to use the HF track. The HF presets are PR9 for VCO1 and PR6 for VCO2.

## Drift

Both the absolute frequency and the pitch spread drift with time and temperature. There is a turn-on drift caused by the warming up of the VCO chips and the power supply. The -5 V rail will change slightly as it warms up and this causes a frequency and pitch spread change. The same is true for the  $\pm 15$  V rails but to a lesser extent. The VCO bank should be finally aligned after the chips have been burnt in for 24 hours and after the unit has been powered up for at least 10 minutes.

Long-term drift is caused by the ageing of the ICs and precision components and the voltage references in the power supplies. This will probably necessitate slight recalibration of the unit every six to twelve months.

## Portamento

The portamento circuits are designed to generate virtually zero voltage change between input and output at all portamento speeds. If eight voices are set up to play in unison they will track over the keyboard range when the portamento is set to fast or slow. On the slow setting, a full-range keyboard transition will take place about 2 S.

Set the synthesiser up for four voice operation and tune up the voices on the top note. Now with the portamento slow (anticlockwise) play a four-note chord in the bottom octave. The four VCOs will shoot off from the top note and zoom down and land exactly on the chord. Lots of wild sounds can be generated using this polyphonic portamento facility. **ETI**

## Pitch Spread

The pitch spread adjustment is very sensitive, but in order to obtain a musically useful synthesiser it must be properly set up. When two or more VCOs are being controlled from the keyboard it is imperative that they track. If they do not then objectionable frequency beating will occur as the keyboard pitch is altered. The Polysynth can have up to 16 VCOs in operation at once and so the pitch adjustment must be spot on. The VCOs have an exponential transfer function, which is musically very useful. It enables linear voltage changes from the keyboard to generate musical intervals from the VCOs. Also you can transpose VCO1 relative to VCO2. This relative tuning is maintained as the VCO pair is moved in pitch by the keyboard voltage.

This is a very powerful feature of the music synthesiser but it relies on the transfer function of all the VCOs being a perfect exponential curve. If one VCO deviates from this curve then it will never track with the other VCOs. If all the VCOs have the same curve but it is not an exact exponential then they will not track when transposed (VCO1 relative to VCO2) or when played in the polyphonic mode. If all the VCOs have exactly the same true exponential curve and yet the digital pitch generator has significant errors then the VCOs will not track in the polyphonic mode. However, if all these problems are properly resolved then you end up with a marvellous polyphonic music synthesiser. Figure 7 shows the VCO transfer function on a log/lin graph. Here a perfect exponential is shown as a straight line. The VCOs tend to go flat at high frequencies, which is caused by the accumulation of timing errors in the oscillator plus the effect of bulk resistance in the exponentiating transistor. However, the CEM3340 has a high frequency tracking adjustment to improve the top end tuning.

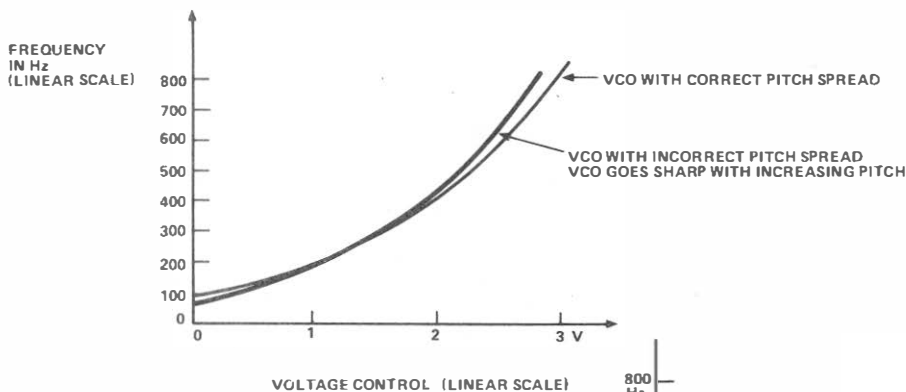


Fig.6 Exponential transfer function of a musical VCO.

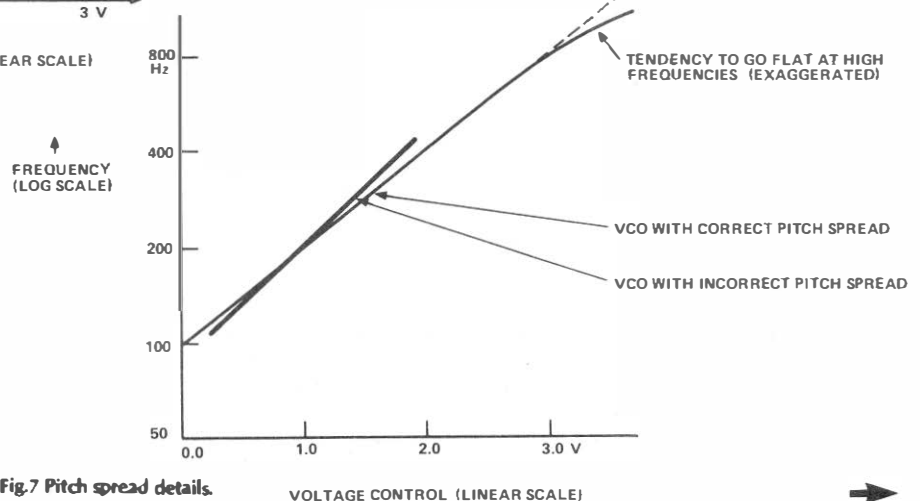


Fig.7 Pitch spread details.

## VCO Pitch Spread Adjustment

Turn the unit on and let it warm up for 10 minutes. The digital pitch generator must be working properly with a resolution of about 10 bits. If it cannot obtain this accuracy then it will not be possible to align the VCOs. Look at the VCF output (the left hand side of R58). Turn off VCO2 and select a sawtooth from VCO1. Turn all high frequency track presets (PR5,6) anticlockwise (this turns them off). Select one-voice operation and remove all modulations and sweeps. Turn off the sync. Set the VCF to maximum frequency and resonance to minimum. Play the bottom note on the keyboard and bias the VCO to 100 Hz. Now play the note one octave up. It should shift the VCO by an octave, but it won't! This is because the pitch spread trim is wrong. The pitch spread trim for VCO1 is PR9. Turning it clockwise gives more VCO octaves per keyboard octave; it gives the VCO a sharper tuning. Turning it anticlockwise gives less VCO octaves per keyboard octave; it gives the VCO a flatter tuning. So if VCO1 is sharp one octave up turn PR9 anticlockwise. However, adjusting the preset also alters the bottom note. This makes the tuning of the VCOs rather difficult unless you have a good musical ear. If you are blessed with this then it is possible to tune the VCO by playing scales or octaves, listening to the VCO output and making suitable changes to the preset.

For those who were born with tin ears a more technical approach should be employed. A frequency meter can be used to set the VCOs to give 'almost' octave intervals. As the frequency meter gate is asynchronous to the VCO then the reading will be slightly different every time. A frequency meter with a 1 S gate will give 1 Hz accuracy for a 100 Hz signal. A 10 S gate will give 0.1 Hz accuracy but 10 S is a long time to wait for two gate periods (20 S). A frequency meter is useful to give you the tuning to within a fraction of a percent.

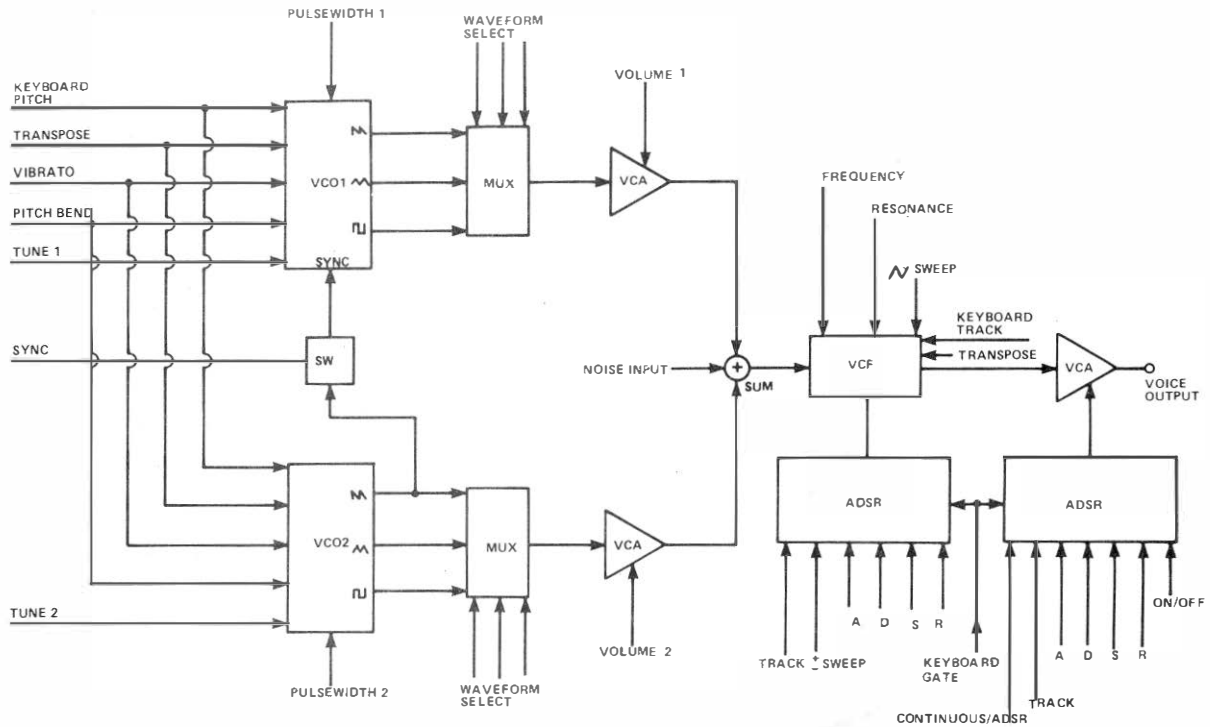


Fig.3 Block diagram of one voice of the Polysynth (PS7).

## Expander Unit

In order to play eight voices, an expander unit is needed. (Fig. 1). It is self-powered and consists of a PS4 mother board, four PS7 voice boards, a PS6 panel board and a mains transformer. The tuning and alignment is exactly the same as for the first four voices. The expander unit is driven from a large multiway connector (Fig. 2). All 34 common signals plus the portamento are wired up on a one-to-one basis, that is vibrato to vibrato, transpose to transpose, etc. The independent pitch and gate signals are obtained separately from PS3 (don't forget to insert the extra sample and hold ICs on PS3). The output of the expander unit and the Polysynth are mixed resistively.

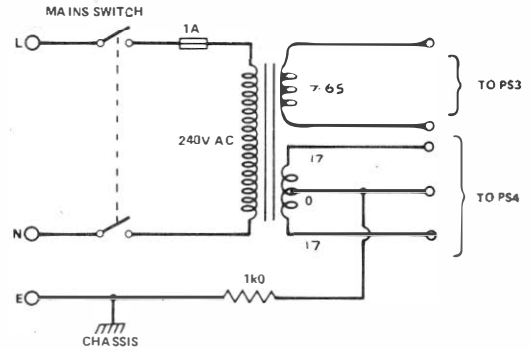


Fig.4 Circuit diagram of the power supply.

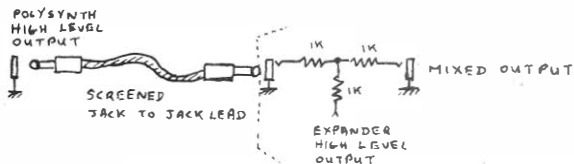


Fig.5 Wiring details for the components that mount on the back panel.

