Digital Delay Line

by POWERTRAN

DIGITAL DELAY EFFECTS UNIT

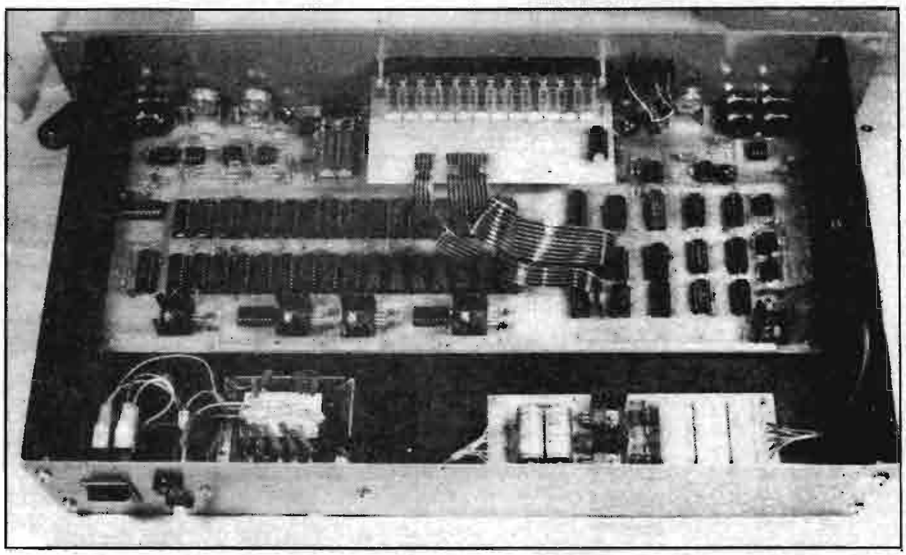
REPRINT OF ARTICLE PUBLISHED IN ELECTRONICS AND MUSIC MAKER (with corrections and amendments necessary to be consistent with kit)

by Tim Orr

- ★ **Digital encoding for studio quality results**
- ★ **Time delays from 0.625ms to 1.6 seconds**
- ★ **Produces all the popular time delay effects:**
- ★ **Phasing** ★ **Flanging** ★ **ADT and chorus**
- ★ **Echo (including 'freeze' for infinite repeats)**
- ★ **Time domain vibrato, etc.**

Many musical effects such as echo boxes, flanging pedals etc. use a time delay as part of their circuitry. The cheaper units, aimed at the stage musician, offer only one or two effects per box; in addition, they use analogue delay components whose sound quality deteriorates considerably as the delay increases. High quality delay units for studio applications, in contrast, use digital techniques offering theoretically unlimited delay times; however, they are very expensive, often with four figure price tags.

Now, the E&MM Digital Delay Effects Unit offers you the best of both worlds; it gives all the time delay effects, with digital quality, all for the price of a high quality analogue unit but with much superior specifications). the nost popular effects are shown in Figure 1, along with the various ways of producing them. The E&MM Delay is shown in block diagram form in Figure 2, and by manipulating its variables all the effects in Figure 1 may be obtained. These are introduced here; the project will be concluded next month, with full circuit and construction details.



Phasing

Phasing is produced by mixing an audio signal with a delayed version of itself. The frequency response this produces is known as a comb filter. Feedback is sometimes used to make the frequency response more peaky, which in turn produces a more noticeable colouration of the sound. By slowly modulating the time delay, the notches in the comb filter expand and contract, producing an interesting musical effect. Phasing effect pedals use a phase shift filter rather than a time delay line, although the effect is the same. Phasing is characterised by having very few notches within the audio band, typically 2 to 5. This is equivalent to time delays between 0.2ms and 0.5ms.

Flanging

Phasing and flanging are often confused, which is not surprising as the two effects are produced in a similar way. To obtain a flanging

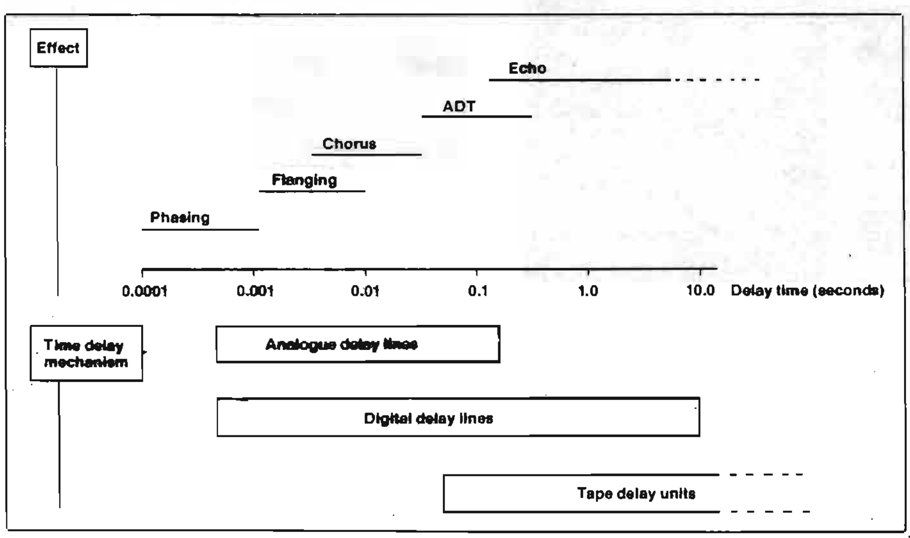


Figure 1. Effects obtainable with time delays.

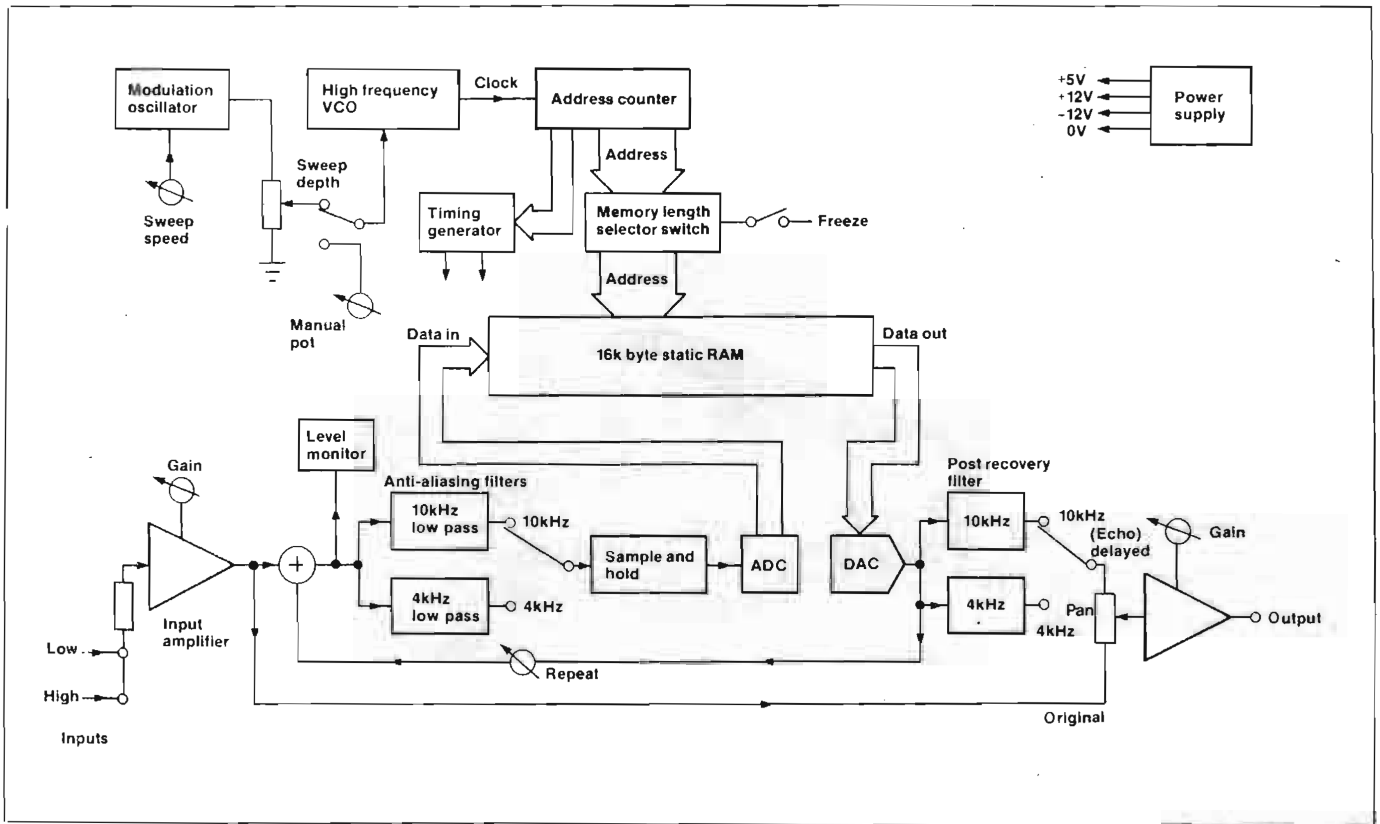
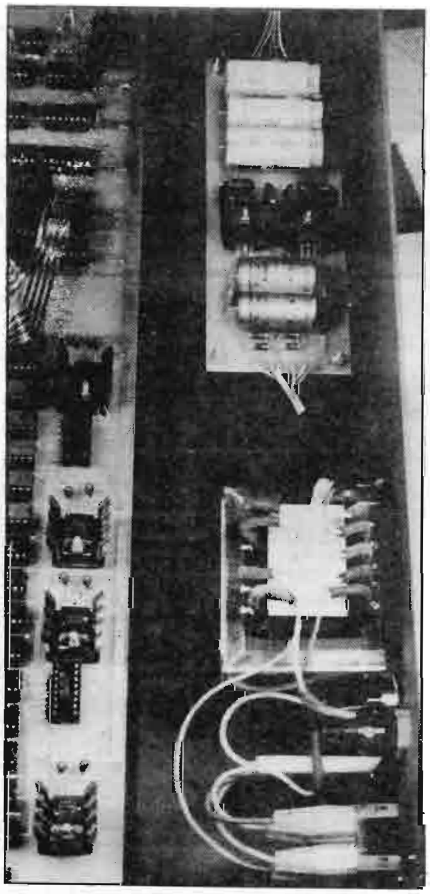
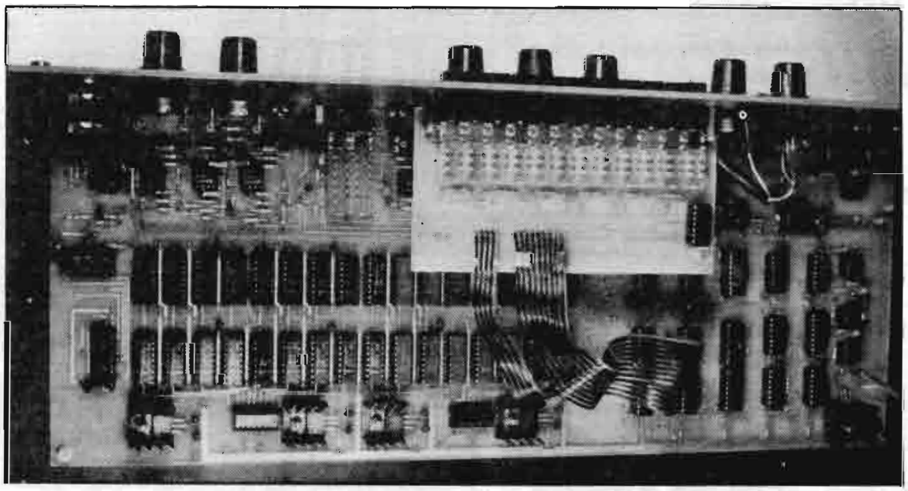


Figure 2. Block diagram of the digital delay line.





effect use a time delay varying between 1 and 10ms. A 10ms delay will produce a comb filter with 100 notches (over a 10kHz bandwidth). Flanging often uses strong feedback which produces a heavy colouration of the sound.

ADT & Chorus

ADT (Automatic Double Tracking) and chorus are both very similar effects. The chorus effect uses a time delay that is slowly modulated, and the original and the delayed signal are mixed together producing a 'spacey' effect. ADT uses a longer time delay to simulate a very short echo, short enough to give the impression of two sound sources.

SPECIFICATIONS

Size: 2 unit (3.45") high, 19" rack mounting, 10" deep.
Delay time: 0.625ms to 0.64s at 10kHz bandwidth.
1.6ms to 1.6s at 4kHz bandwidth. Both time delays can be halved using the manual control pot.
Modulation oscillator: Triangle sweep; rate, 0.025Hz to 17Hz.
Memory size: 16K bytes, 128K bits.
Input impedance: Low 1k5
High 28k5
Output impedance: 220 ohms.
Output level: +3dBm, with 30dBs of available attenuation.
Typical signal to noise ratio: 75dB (this is not signal to quantisation noise).
Overload LED: Turns on 6dB before clipping.

Echo

Time delays greater than 30 or 40ms become noticeable as distinct echoes. Time delays of around one second are very useful for building up melodies with several repeats. Also, it is possible to freeze the sound in the digital memory and have it continuously recirculate without degeneration. This repeating sound may then be used as a sequencer-like backing, or transposed using the delay time controls.

Vibrato

Vibrato can be produced on any time delay setting, but best results are obtained on the 40ms delay with 10kHz bandwidth. A modulation speed of 3 to 7Hz with a small modulation depth should do it. **E&MM**

The E&MM Digital Delay Line is obtainable as a complete kit of parts from Powertran Electronics, Portway Industrial Estate, Andover, Hants SP10 3WW.

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Part 2 concludes the project with full constructional details.

Circuit Description

The complete circuit of the unit is shown in Figures 3 and 4, and operation may be clearer if the block diagram (Figure 2, published last month) is studied at the same time.

The input signal is amplified by the input amplifier IC21, to bring the signal up to a suitable operating level. Next the signal is filtered by two low pass filters, one of which has a 4kHz cut-off, and the other a 10kHz cut-off. The 4kHz or 10kHz operation is selected by S16. These are known as anti-aliasing filters; aliasing is an effect that sounds like ring modulation and is caused by harmonics of the input signal interacting with the analogue to digital conversion. If these harmonics are greater in frequency than ½ of the conversion frequency, then side bands will be generated that will fall within the audio spectrum: to prevent this from happening, the input signal is low-pass filtered to remove these high frequency harmonics.

The signal is then fed into the ADC (analogue to digital converter). This section continuously samples the analogue waveform, and measures the instantaneous amplitude which it describes with an 8-bit digital word. This word is then stored in the digital memory. In order to convert the analogue waveform into a digital word it must be 'frozen' long enough to allow the ADC to perform the measurement. The freezing is done with a sample and hold device, IC10. The 'lumpy' output of the sample and hold unit (TP8, see Figure 8) still represents the input signal; if it were low-pass filtered then the original waveform would be recovered, as in fact it is when it is reconstructed by the digital to analogue converter (DAC) after the selected time delay (TP10, Figure 8).

The ADC consists of three main sections, a comparator IC14, a successive approximation register (SAR) IC17, and a DAC IC13. Their purpose in life is to measure the input voltage and to describe it with an 8-bit digital word. The SAR produces a binary code which it sends to the DAC; this generates an output voltage which the comparator compares with the input signal. The result of the comparison determines whether the MSB of the digital word is a 1 or a 0. The SAR then tests the next bit of the code, and then the next, until all 8 bits have been determined; the conversion is then complete. The 8-bit word causes the DAC to produce a voltage equal in magnitude to the input signal, therefore the word is a measurement of the input sample.

The DAC is in fact a companding DAC, and not a linear one; it can be operated in both compression and expansion modes. In the ADC (IC13) it compresses the signal, and in the DAC (IC33) it expands the signal thus giving an overall linear transfer function. The performance of the DAC and ADC can be described in several ways. First, the dynamic range: this is the ratio between the largest signal that the system can handle and the smallest. The dynamic range is 72dB which is quite good.

The signal to noise ratio is the ratio between the largest signal that the system can handle, and the output noise with no input signal. This may be better than 72dB, but it is not that important, because the system only generates digital noise (quantisation noise) when it is converting signals. The noise only appears when a signal output is being generated; this noise is related to the

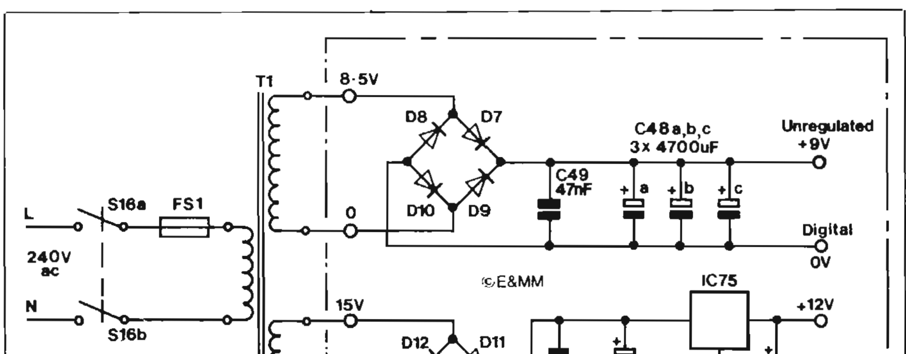


Figure 3a. Circuit diagram of PSU.

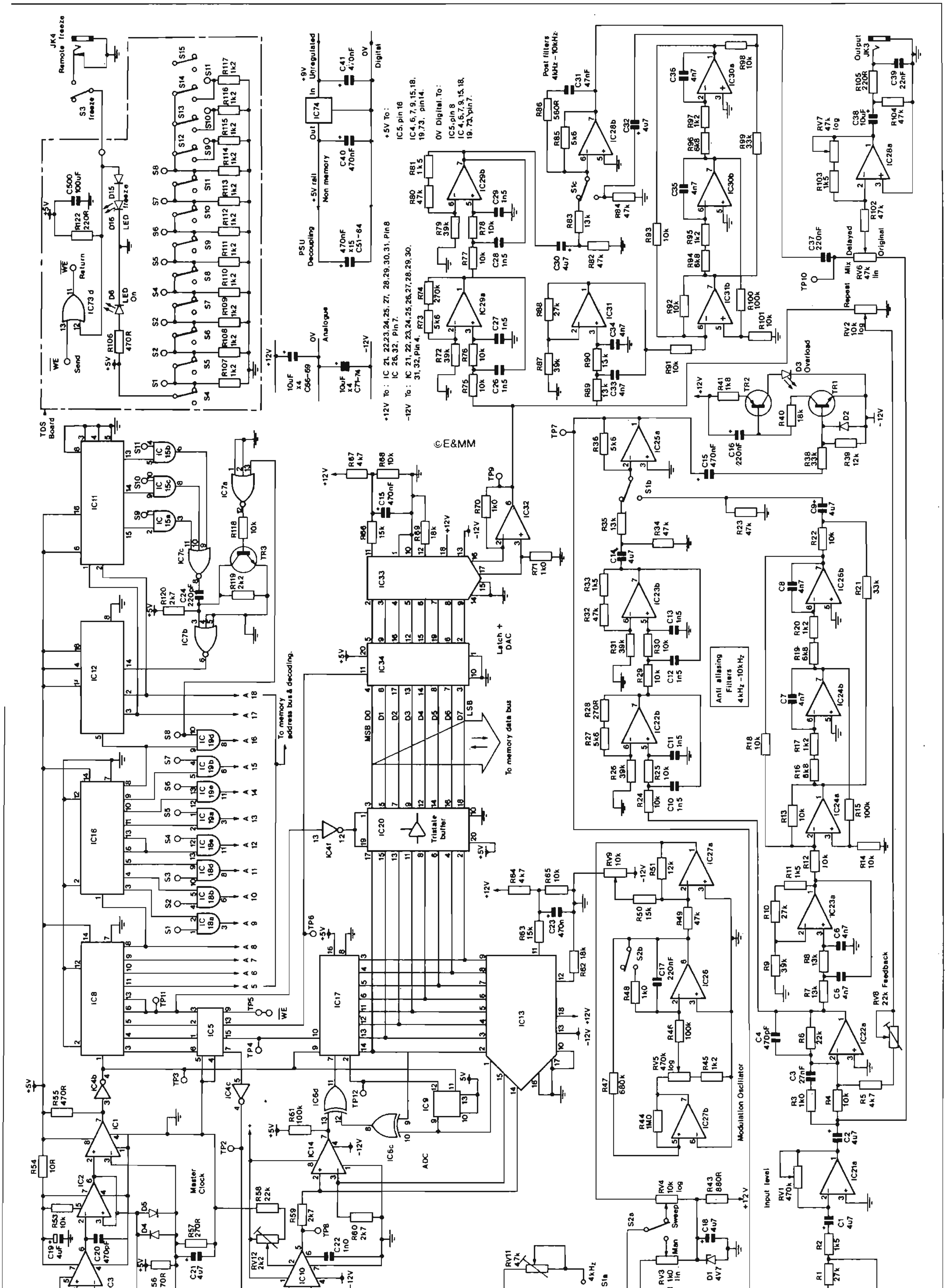


Figure 3. Main circuit of the Digital Delay Effects Unit.

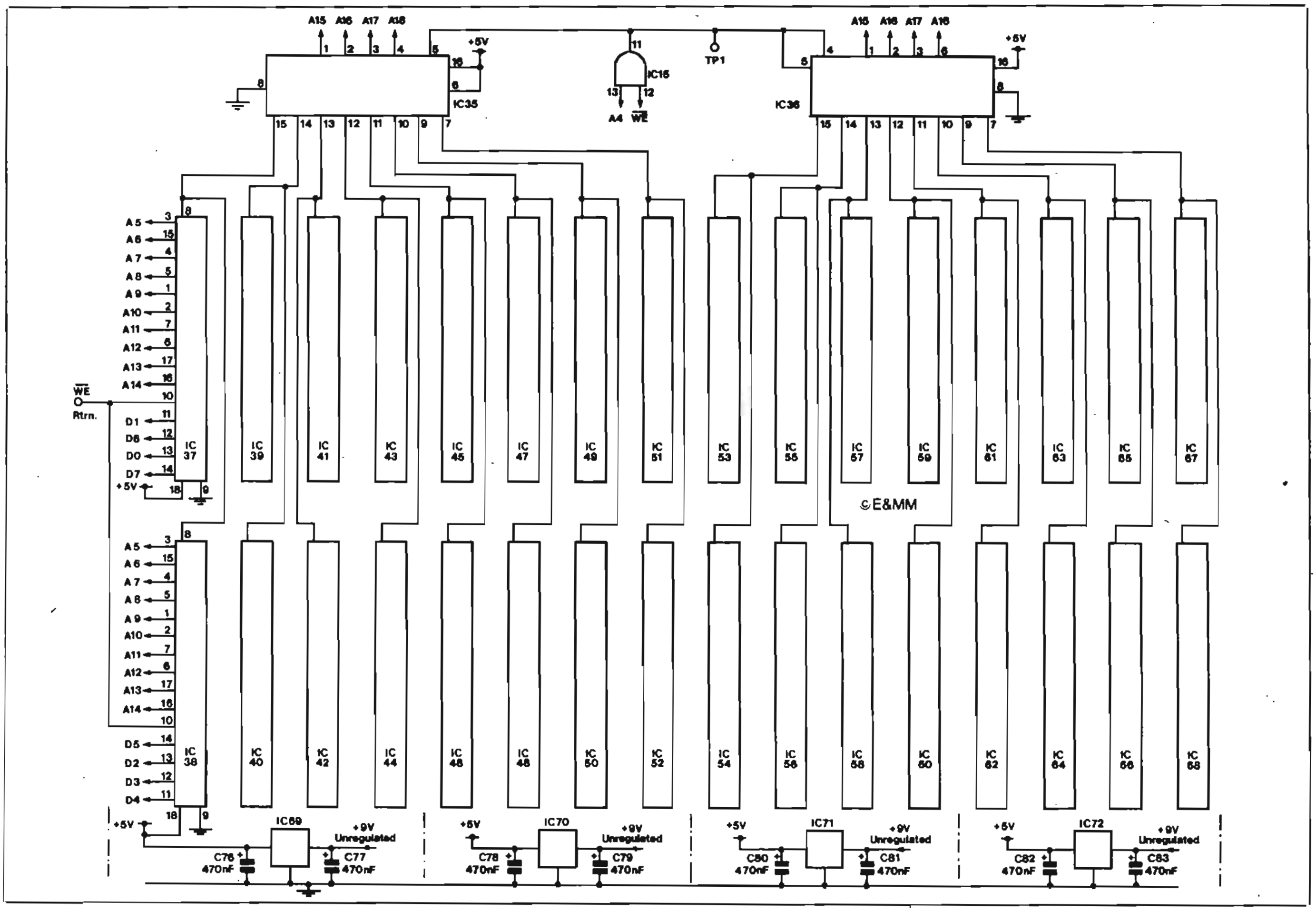


Figure 4. Circuit of the memory section.

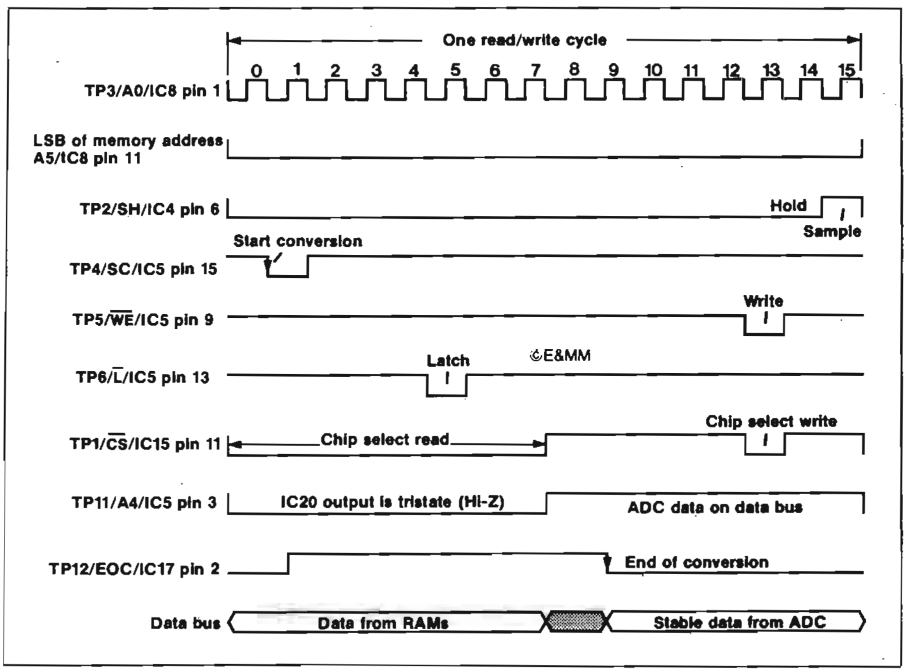


Figure 5. Timing diagram for one read/write cycle.

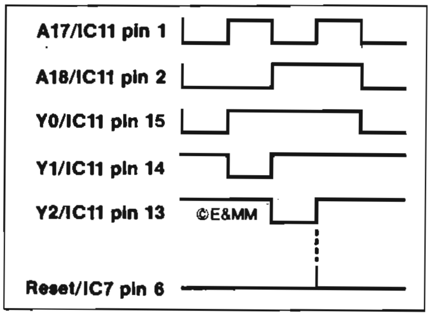


Figure 6. Memory reset timing, shown with ¾ memory selected.

amplitude and spectrum of the input signal.

Now consider the signal to quantisation noise ratio. If the delay line is processing speech, then the quantisation noise is hardly noticeable; the noise is masked by the rapidly changing information in the speech. If the input signal is high in frequency compared to the selected bandwidth, then again the quantisation noise is lost, this time having been removed by the output filters. However, if the input signal is a low frequency pure tone then quantisation noise can be heard sizzling away in the background! This problem is overcome by giving the input signal a treble lift from 600Hz up to 6kHz (R3 and C3 give pre-emphasis) and by providing a treble cut at those same frequencies on the output signal (R86 and C31 give de-emphasis). The overall frequency response is flat, and the quantisation noise is selectively filtered out. The energy spectrum of most natural sounds falls off with increasing frequency, and so the pre-emphasis does not produce any signal overload problems.

The memory is 16K bytes long, being constructed from 2114L static RAMs (4 bits by 1K). The read/write cycle is as follows: the memory address is set up, and data is read from that memory location by being clocked into a latch (IC34) that drives the DAC (IC33). Next, data is written into the same memory location, the data being obtained from the ADC. The address is then incremented by one bit. If the full memory length were being used, then the address would have to count in a full circle (16K) to retrieve the data that had just been entered.

The output data is converted to an analogue voltage by the DAC (IC33). This voltage is quantised into steps, and it needs filtering to remove the unwanted harmonics that constitute these steps. Again a 4kHz and a 10kHz low pass filter (IC28, 29, 30, 31) are used.

The master clock for the system is generated by a high frequency voltage controlled oscillator, IC1, 2 and 3. IC2 and 3 form a standard Schmitt trigger/integrator oscillator, the frequency of which is controlled by the current into pin 5 of IC3. IC1 is used as a buffer to drive the subsequent TTL stage.

IC8, 12 and 16 are binary dividers which generate the memory addresses A0 to A18.

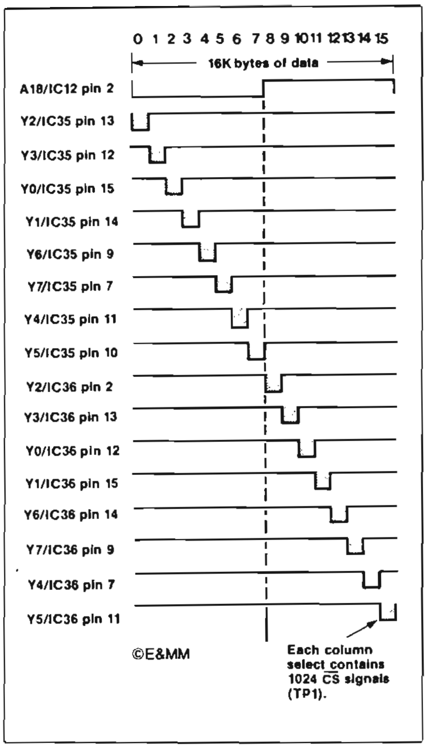


Figure 7. Memory column select timing, with full memory.

A0 to A4 are used to generate timing signals, such as read, write, start conversion, sample, and A5 to A18 are used as the memory address. Shorter time delays are obtained by using smaller sections of the memory, by progressively disabling the memory addresses using IC15, 18 and 19. The memory is sectioned into four quarters (see memory options in parts list) and so the top four time delay selections have equal time increments, but the lower eight selections provide time delays in octave increments. The master clock oscillator frequency may be manually controlled by RV3, or modulated by the low frequency triangle oscillator IC26, 27.

Test points

The timing diagram for one conversion read/write cycle is shown in Figure 5. All the waveforms will be clearly visible at the indicated test points (TP1-12). The memory reset timing is shown in Figure 6. A18 has a period of 1.6 seconds or 0.64 seconds, depending on the selected bandwidth. The reset pulse has a period of less than one micro-second, so don't be surprised if you cannot see it! Figure 7 shows the memory column select decoding. The number of columns selected will depend on the time delay selected by SW12, 13, 14, 15.

Construction

Most of the components, including con-

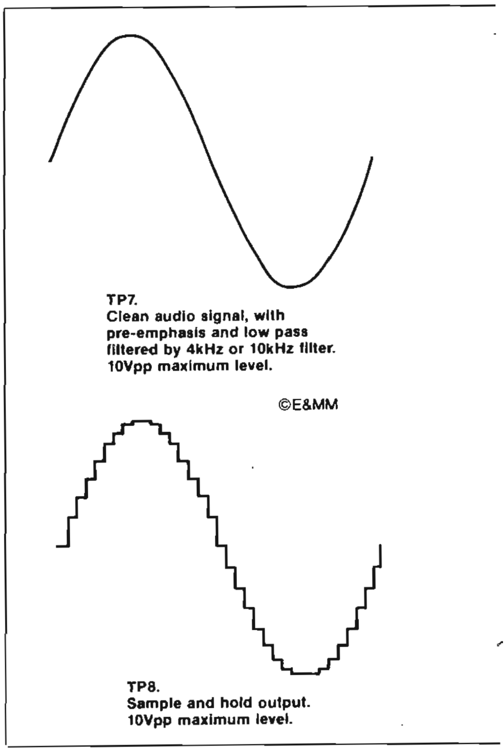


Figure 8. Waveforms for TP7-10, showing ADC and DAC operation.

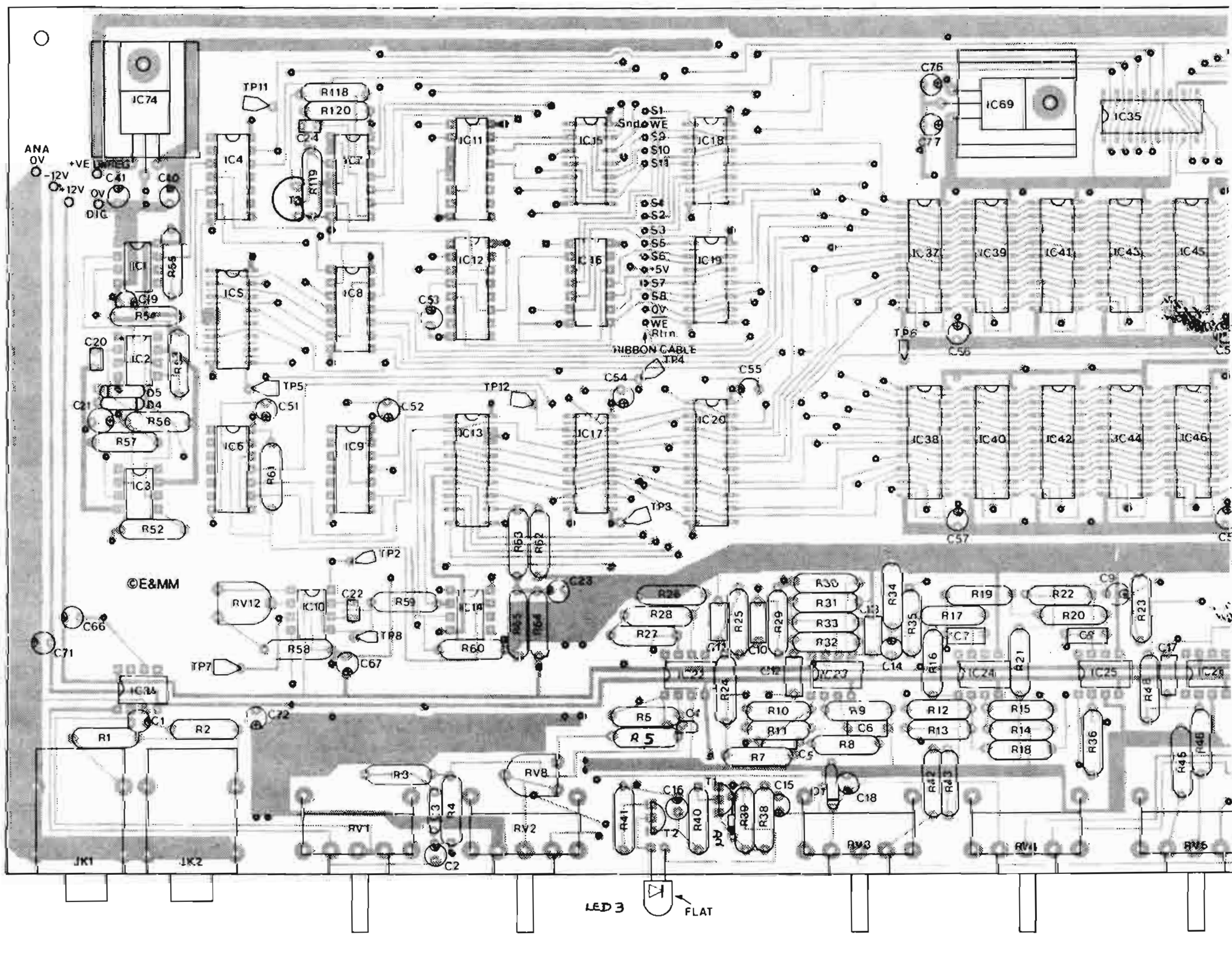
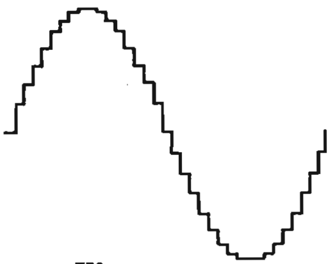


Figure 9. Component layout for the main PCB.



TP9.
DAC output same as TP8 but delayed in time.
3.5Vpp maximum level.

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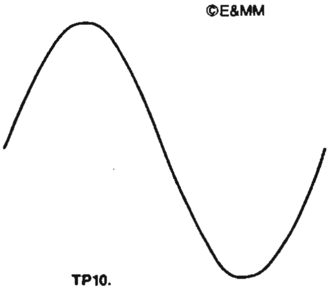


TP10.
Filtered audio output with pre-emphasis removed.
3.0Vpp maximum level.

trols, are mounted on one large double sided PCB, whilst two smaller single sided boards carry the time delay selector switches and the power supply components (with the exception of the transformer). Powertran's PCBs will not carry a printed component legend, but all the component positions are identified in Figures 9, 10 and 11 and construction should be straightforward.

Sockets are recommended for the ICs, and again, these are provided in the kit. As always, take special care with the soldering, and check for dry joints, solder splashes and correct component orientation before switching on.

There is very little wiring to be done. The switch board is linked to the main board with two lengths of ribbon cable, as shown on the component overlays; the PSU board and transformer wiring is shown in Figure 12. The connections to the freeze switch and footswitch socket are on the switch board diagram.

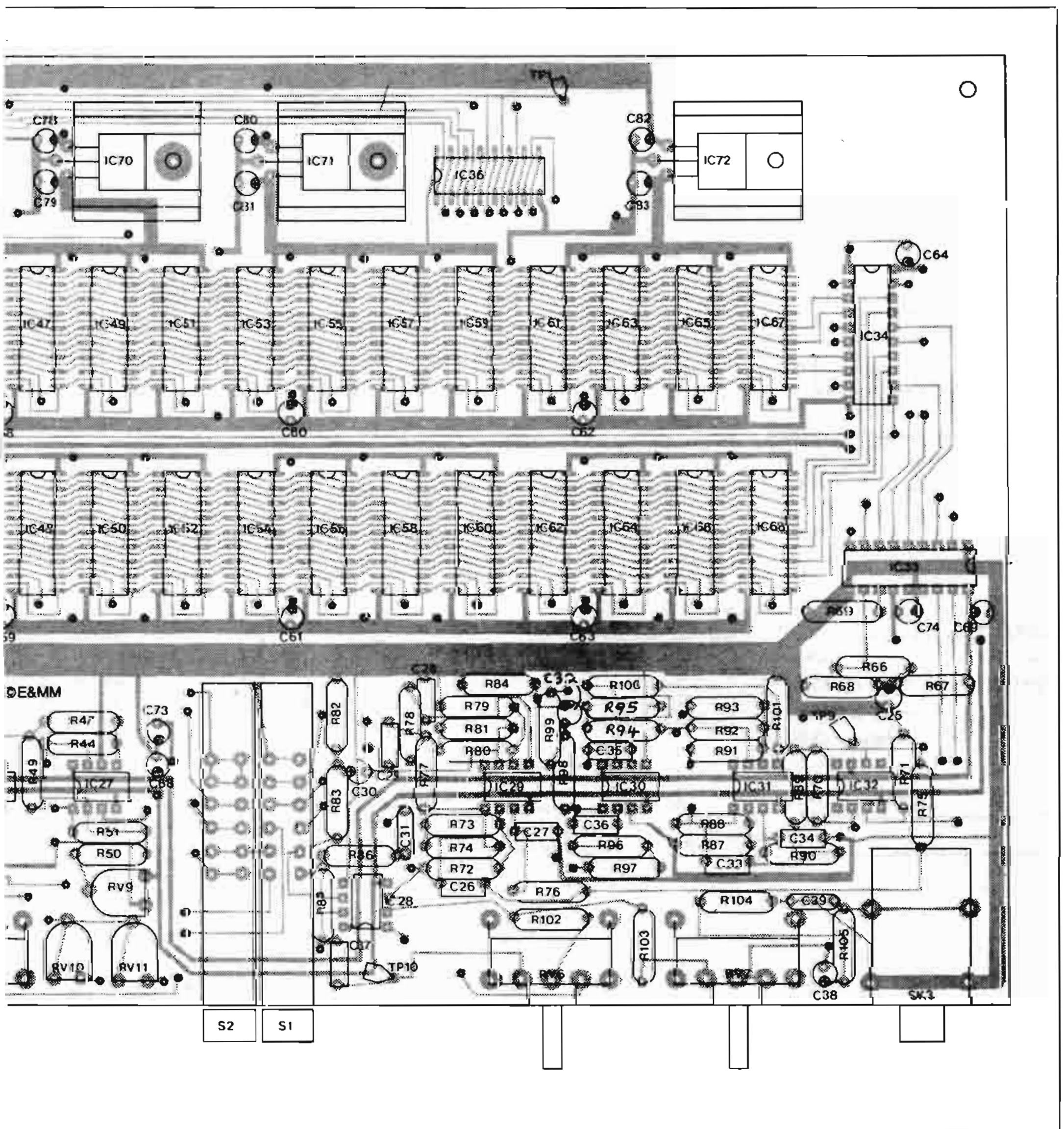
If required, the delay unit may be built with ¼, ½ or ¾ memory to begin with, and this is simply a matter of omitting some of the components: the parts list gives details.

Once all the soldering is done, do not insert any ICs except those in the power supply. Power up and test the regulated supply rails (the unregulated rail voltages only refer to a fully loaded power supply). Insert the ICs, in lots of 10, and then power up and check the regulated supply rails. Do this until all the ICs are inserted. Don't forget to turn off the power when you are putting them in! Having completed a successful power up you can now test the unit.

Connect a signal and check all is functional. If not, then check to see if all the TP waveforms are being generated correctly. Also look at all 19 address lines. If you experience a regular repeating fault in the memory section then you may have a non-functional area of memory. Check out the address lines, the data bus and the column decoding. If these are all OK then it is probable that a memory chip is faulty. This can be located by a process of substitution. Finally, set up the presets as follows:

1) Set up the unit for a long echo, and set REPEAT to maximum. Adjust RV8 so that repeats continue for a long time, but not so that they grow in amplitude.

2) Measure the voltage on the positive end of D1; it should be +4.7V. Monitor IC27 pin 1, and adjust RV9 so that the triangle waveform is offset so that its bottom point is at +4.7V. If you don't have a 'scope, a voltmeter may be used, but turn down the



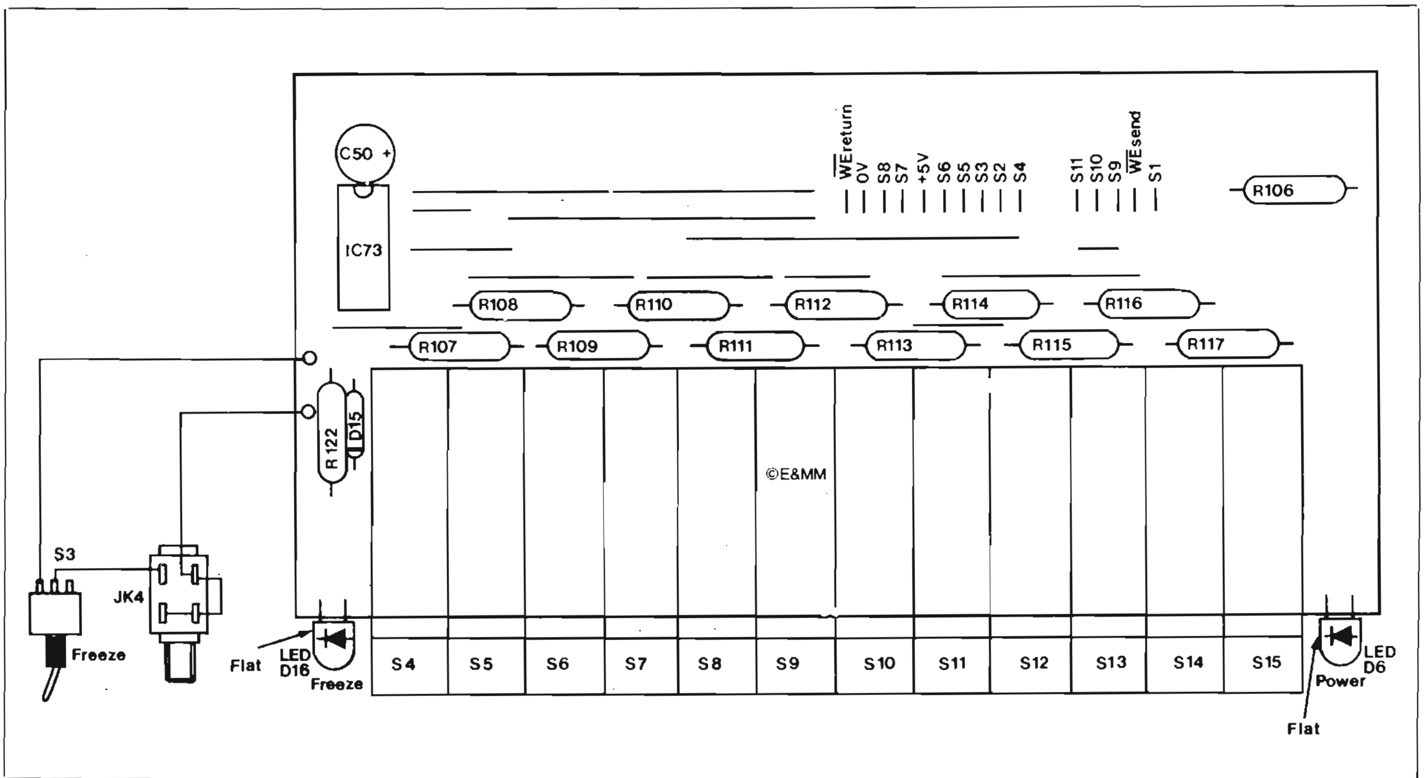


Figure 10. Component layout for the switch PCB.

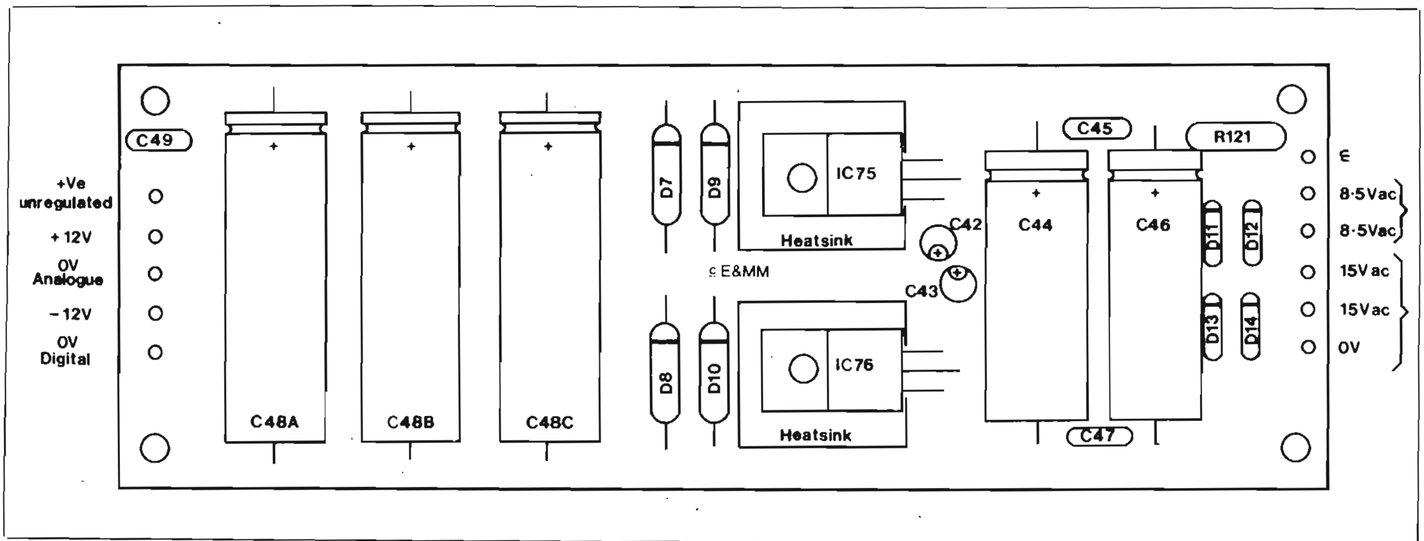


Figure 11. Component layout for the PSU board.

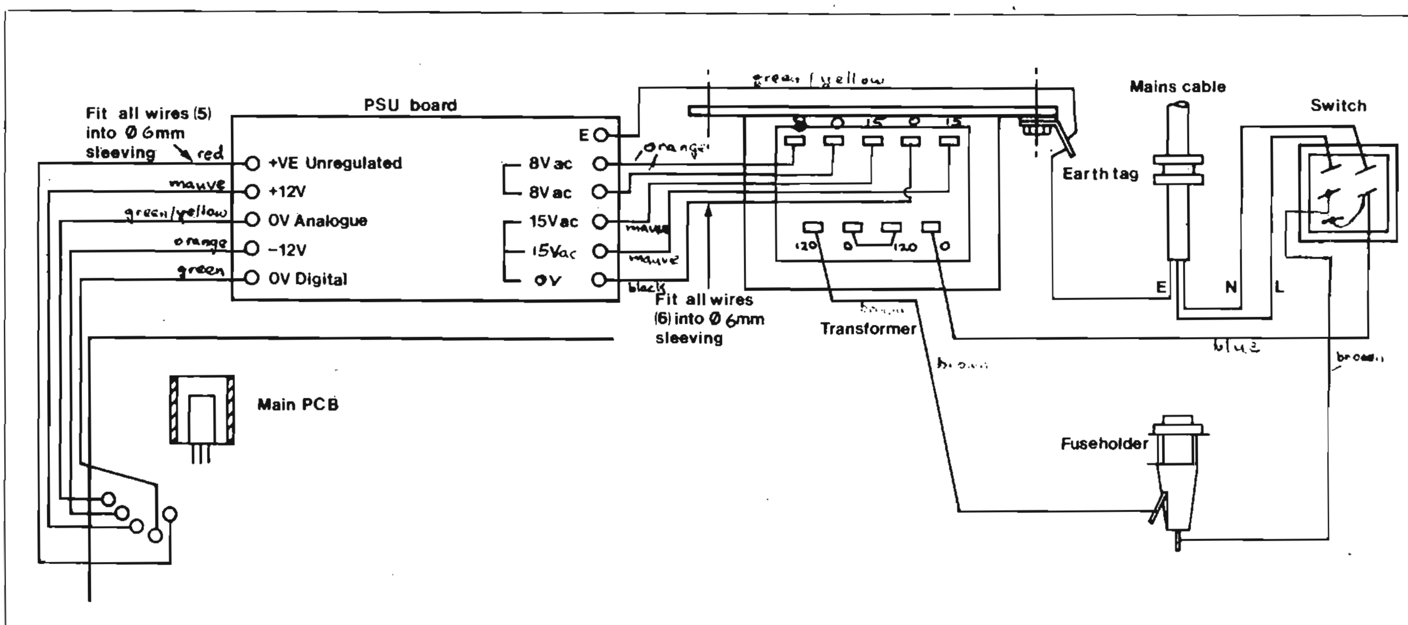


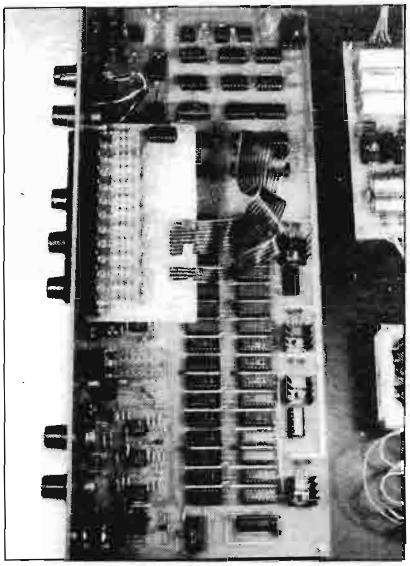
Figure 12. Wiring to the PSU board and transformer.

sweep speed to avoid misleading readings.

3) Turn the time delay pot anti-clockwise, and select manual control of time delay. Select 10kHz bandwidth, and measure the frequency at test point 3 (TP3). Adjust RV10 so that the frequency is about 400kHz. Now select 4kHz bandwidth, and adjust RV11 so that the frequency is about 160kHz. These frequencies can be set without instruments by entering a short signal, freezing it and setting the pre-sets so that the delay times on the longest setting are 0.64 secs and 1.6 secs (time 10 repeats, i.e. 6.4 secs and 16 secs).

4) The sample and hold offset adjustment RV12 only produces a small DC shift, which when compared to the 10Vp-p audio signal level at this point is not significant. However, if the ADC is dithering between two quantised states (and hence producing 1 LSB of dithering noise) then the DC offset can be used to shift the analogue voltage by just enough to prevent this. Listen to the delayed output, and adjust RV12 for minimum noise. Find the best pre-set position by manually adjusting the delay time. The delay effects unit is now ready for use.

E&MM



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DIGITAL DELAY UNIT PARTS LIST

Resistors — 5% ½W carbon unless specified

R1,10,88	27k	3 off
R2,11,33,81,103	1k5	5 off
R3,48	1k0	2 off
R4,12-14,18, 22,24,25,29, 30,53,65,68, 75-78,91-93, 98,101,118	10k	23 off
R5,52,64,67,121	4k7	5 off
R6,58	22k	2 off
R7,8,35,83,89,90	13k	6 off
R9,26,31,72, 79,87	39k	6 off
R15,46,61,100	100k	4 off
R16,19,94,96	6k8	4 off
R17,20,45,95,97, 107-117	1k2	16 off
R21,38,99	33k	3 off
R23,32,34,49,80, 82,84,102,104	47k	9 off
R27,36,73,85	5k6	4 off
R28,56,57,74	270R	4 off
R39,51	12k	2 off
R40,62,69	18k	3 off
R41	1k8	
R42,43	680R	2 off
R44	1M0	
R47	680k	
R50,63,66	15k	3 off
R54	10R	
R55,106	470R	2 off
R59,60	2k7 1%	2 off
R70,71	1k0 1%	2 off
R86	560R	
R105,122	220R	2 off
R119	2k2	
R120	2k7	
RV1,5	470k log pot PCB mounting	2 off
RV2	10k reverse log pot PCB mounting	
RV3	1k lin pot PCB mounting	
RV4	10k log pot PCB mounting	
RV6	47k lin pot (with central 'click') PCB mounting	
RV7	47k log pot PCB mounting	
RV8	22k min horiz preset	
RV9,10	10k min horiz preset	2 off
RV11	47k min horiz preset	
RV12	2k2 min horiz preset	

Capacitors

C1,2,9,14,18,19, 21,30,32	4u7 15V tantalum	9 off
C3	27nF polycarb	
C4,20	470pF ceramic	2 off
C5-8,33-36	4n7 polycarb	8 off
C10-13,26-29	1n5 polycarb	8 off
C15,23,25,40-43, 51-64,76-83	470nF 15V tantalum	30 off
C16	220nF 25V tantalum	
C17,37	220nF polycarb	2 off
C22	1n0 ceramic	
C24	220pF ceramic	
C31	47nF polycarb	
C38,66-69,71-74	10uF 15V tantalum	9 off
C39	22nF polycarb	
C44,46	1000uF 25V electrolytic	2 off
C45,47,49	47nF ceramic	3 off
C48a,[illegible]b,48c	4700uF 16V electrolytic	3 off
C50	100uF 10V electrolytic	

Semiconductors

TR1,3	BC182L	2 off
TR2	BC212L	
D1	4V7 zener	
D2,4,5,15	1N4148	4 off
D3,16	Red LED	2 off
D6	Green LED	
D7-10	1N5402	4 off
D11-14	1N4002	4 off
IC1,14	LM311	2 off
IC2,3	CA3080	2 off
IC4	74LS04	
IC5,11,35,36	74LS138	4 off (see below)
IC6	74LS86	
IC7	74LS27	
IC8,16	74LS393	2 off
IC9	74LS74	
IC10	LF398	
IC12	74LS193	
IC13,33	DAC76	2 off
IC15,18,19	74LS08	3 off
IC17	AM2502	
IC20	74LS244	
IC21-25,28-31	RC4558	9 off
IC26,32	TL081	2 off
IC27	1458	
IC34	74LS374	
IC37-68	2114L	32 off (see below)
IC69-72,74	7805	5 off (see below)
IC73	74LS32	
IC75	7812	
IC76	7912	

For ¾ memory omit IC61-68,72
For ½ memory omit IC53-68,71,72,36
For ¼ memory omit IC45-68,70,71,72,36

Miscellaneous

JK1-4	¼" mono jack socket, switched	4 off	
S1,2	4P2W latching push switch	2 off	
S3	SPST mini toggle switch		
S4-15	12 x 4 P2W interdependant switch bank		
S16	DP mains rocker switch		
T1	Transformer 15-0-15V, 8.5V		
	T0220 heatsink	7 off	omit 1 for each ¼ memory not fitted
	8 pin DIL socket	17 off	
	14 pin DIL socket	10 off	
	16 pin DIL socket	6 off	omit 1 for ½ or ¼ memory
	18 pin DIL socket	34 off	omit 8 for each ¼ memory not fitted
	20 pin DIL socket	2 off	
	3 PCBs — main, switch and PSU		
	Knobs	7 off	
	Case and hardware		
	Mains lead, plug and grommet		
	10-way ribbon cable — 8" length		
	5-way ribbon cable — 8" length		

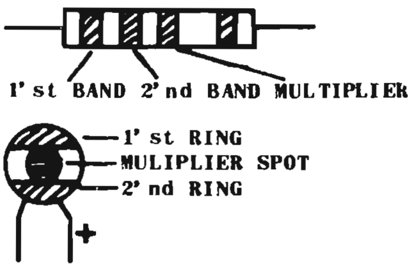
missing [illegible]

General notes on kit construction

Identifying parts

Before assembling any part of the kit, familiarize yourself with the components. Clear yourself plenty of space in a spare room and sort out the parts according to the check list but do not mix up the packs. Read all the parts and put them in order.

Resistor Colour code



Colour	1st & 2nd band/ring	Multiplying factor
Brown	1	10
Red	2	100
Orange	3	1000
Yellow	4	10000
Green	5	100000
Blue	6	
Mauve	7	
Grey	8	
White	9	
Black	0	1
Gold		1/10

example: 390K = orange white yellow (39 × 10000 = 390000 ohms)
Note: K = kilohm (1000 ohms) M = megohms (1000000 ohms)

rather than writing 5.6K for 5600 ohms usually 5K6 is written as it avoids the risk of missing the dot and reading 56K

Capacitors: these may be marked as pico farads (P or pF or nothing at all) nano farad (n or nF) or micro farads (µF or µ or mF)

$1pF = 10^{-12}$ farad $= 10^{-6}\mu F = 10^{-3}n$

$1nF = 10^{-9}$ farad $= 10^{-3}\mu F = 0.001\mu F = 1000pF$

$1\mu F = 10^{-6}$ farad $= 1000n$

Some capacitors are colour coded. The bead tantalum types are coded in uF the striped polycarbonate types are coded in pF

Soldering

Soldering is really very easy but strangely if a constructor runs into problems with his kit it is the soldering which is nearly always at fault. To solder well it is best to understand the principles of soldering.
Solder is an alloy (of tin and lead) which is electrically conductive and has a fairly low melting point (about 190°C) making it suitable for joining together pieces of metal (except aluminium) both mechanically and electrically without causing heat damage. Solder supplied for electronic construction is in the form of wire which has small cavities containing 'flux' which is a chemical which when hot and freshly melted will remove from the pieces being joined the oxide layer and other dirt which otherwise would prevent the solder wetting the surfaces - if the solder does not wet the surfaces the joint will be unsound both mechanically and electrically - you then have a 'dry' joint.
To solder well the iron must be clean - wipe the tip on a damp sponge whenever it has a lot of black rubbish (burnt flux) on it. WARNING some beginners books talk of filing the bit or rubbing it on abrasive paper - good modern irons have iron plated bits which will be ruined by this procedure. To solder a component into a circuit board, insert its leads and bend them slightly to hold the component in place. We do not recommend flatenning the leads onto the board (except for the very thin leads of polystyrene capacitors - the ones which look like rolled up Sellotape) because doing that makes removal much more difficult if you later find you have fitted it into the wrong space. Next trim the leads to about 2mm from the board. Now comes the easy bit which so often goes wrong.

Some constructors cover the iron with melted solder and attempt to transfer this to the joint. Apart from the risk of solder spreading across the tracks, solder carried on the iron no longer has active flux to clean the joint which is necessary for the solder to be able to wet the components.

The way to make solder work for you is to touch the solder onto leads and tracks which are hot enough to melt the solder and release active flux onto the joint, so place the tip of the soldering iron onto the circuit board track touching the component lead at the same time, wait about a second then place the solder where the track, lead and soldering iron tip meet. The solder will then flow around the joint. Make sure the lead is surrounded by solder then remove the solder and iron. Now look at the joint and you will see that you DO *NOT* HAVE A BLOB OF SOLDER around the lead - you will see solder smoothly tapering off in thickness as it spreads away from the joint.

Soldering wires to component tags

On some components to which wires are attached there are holes in their tags. To solder a wire to these wrap the wire (insulation stripped off the last ¼" first) through the hole so as to hold the wire in place then heat tag and lead simultaneously and apply solder to where the lead, tag and iron meet and do not remove the iron and solder until the joint is smoothly covered which may take a few seconds with thick tags or thick wire.

Soldering wires to pins on PCB's

This calls for a different technique as it is not normal to make a mechanical joint before soldering. Holding a wire, a soldering iron and solder in place simultaneously requires 50% more hands than most people are born with so instead of attempting that, it is usual to pre-coat the pin and lead with solder and then join them with heat. This pre-coating is called 'tinning' - heat the bared wire with the iron and apply solder to where the wire and iron meet until the wire is coated - next repeat this for the pin. Hold the wire against the pin (pliers or tweezers help in some difficult to reach places), with the solder resting on the table, with the end bent up so it can be touched with the iron without burning the table melt a *SMALL* amount of solder onto the iron (this is the ONLY time loading the iron is permissible) and as fast as possible apply the iron to the joint and remove as soon as solder is seen to have joined the wire to the pin but do not yet remove your grip on the wire. As solder cools it passes through a semi molten stage (eutectic) when visually it may appear solid but is in fact very weak and if the joint is disturbed at this stage it will be permanently weakened and likely to fail later.

Cleaning of the circuit board

The importance of cleaning the track side of circuit boards cannot be overstressed. Unless all the resin has been cleaned from the joints it is very easy to miss spotting an unsoldered joint, a dry joint or shorted tracks.

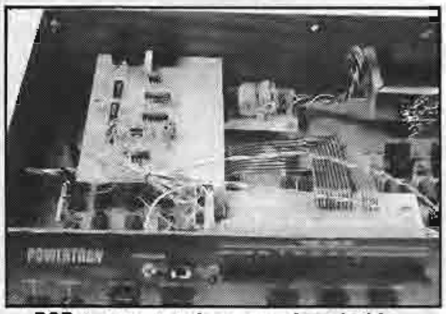
To remove the flux brush the soldered joints with a stiff brush (a ½" wide paint brush with its bristles cut down to ½" length is ideal) and a suitable solvent. Proprietary board cleaners are available - RS Components are one such supplier and this can be obtained from many component dealers, however, cheaper alternatives are also available. Acetone can be purchased readily as solvent/brush cleaner for polyester resins from Strand Glass Co. the fibre glass specialist. Snopake thinners are available from stationery suppliers (Snopake is the white correcting fluid used by typists). Nail polish remover also gives reasonable results and Boots own brand is good value.

When cleaning, keep the solvent off the component side of the board as it could remove the markings of the components and possibly damage them (particularly likely with polystyrene capacitors). Solvent could flow through any spare holes on to the component side of the board so these should be soldered over before cleaning. Check the board when dry under a bright light (in our laboratory we use a 6' fluorescent fitting mounted 22" above the test bench) look very closely (a magnifying glass helps) at each joint and also all regions where tracks are close to each other. Sometimes very thin whisps of solder fall across a pair of tracks and these have to be looked for very closely. Even with very careful soldering a surprising number of faults can be revealed by this examination.

BUILD IT

An Emulator for £10...

Well, not quite. But this mini-project does let you control the pitch of a sound frozen inside a Powertran DDL from any monosynth equipped with a trigger output, and there's a PCB available from E&MM. *Patrick Shipsey*



PCB on rear panel, as seen from inside.

Not being able to afford the sophistication of Powertran's MCS1 MIDI Controlled Sampler, I was once faced with the prospect of not being able to do very much with the sounds I could store in the memory of the same company's 'conventional' DDL. Sure, the built-in Freeze function was useful, but what fun I could have if only I could figure out a way of controlling the pitch of the frozen sound from a keyboard. Well, now I can. And if you're afflicted with the same problem and not averse to getting the soldering iron out from the cupboard under the stairs, you'll be able to as well.

Basically, the modifications I'm about to discuss make it possible for frozen sounds to track the pitch of an external instrument (probably a monosynth, though it could conceivably be something more exotic). They also include an external trigger input, which works from a zero to +5V trigger pulse, but could no doubt be converted to operate from an S-trigger. Both external trigger and oscillator can be switched in or out at any time, and with each of them out of the way, the DDL performs as usual.

The really good news is that making said modifications doesn't involve removing the Powertran's main PCB, as all wires are soldered either to PCB tracks on the top of the board or to IC legs bent out from under the protective body of their associated chip.

And there's more. The mod will work on the minimum amount of memory (4K), so it isn't even necessary to have all the memory fitted to your DDL before commencing work on this project (though this would still be an advantage). However, I'll be assuming throughout the piece that you have access to the original DDL circuit diagram – the technical description won't make a lot of sense if you haven't...

As for prototype development, I've successfully fitted the modifications to both my own DDL and that of a friend,

Figure 1. Circuit diagram of trigger section.

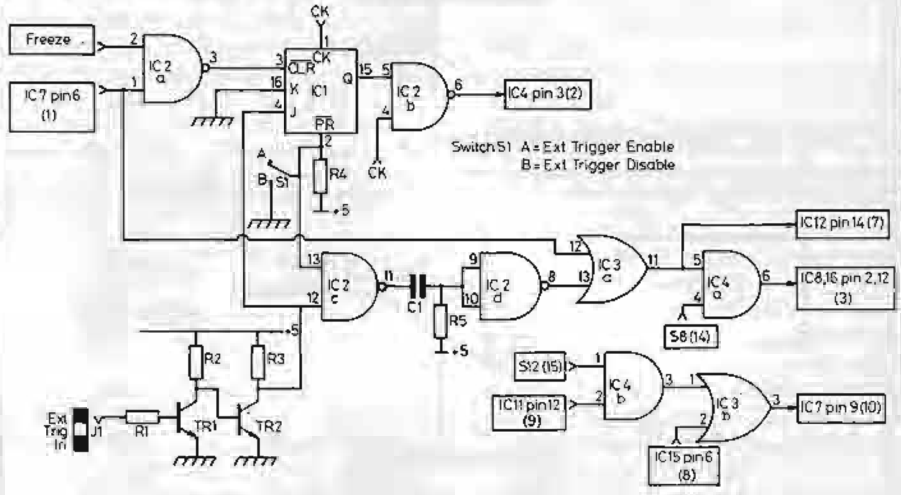
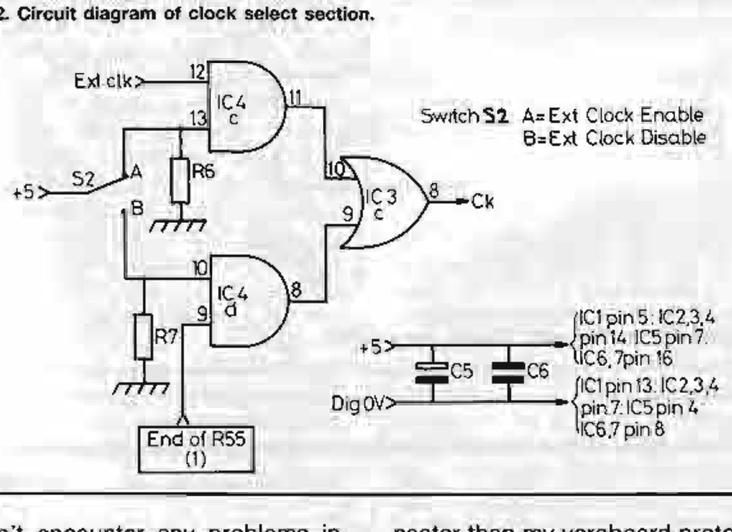


Figure 2. Circuit diagram of clock select section.



and didn't encounter any problems in doing so. A second friend (I'm just *so* popular) has managed to fit the frequency-tracking part of the circuit to a Boss DE200, too. Even E&MM's Technical Editor built one, and his worked first time.

Sound quality has been nothing short of remarkable, and I've had endless fun mucking about with the usual sampling clichés like human voice 'aahs' and slapped bass sounds. And all for well under a tenner, PCB and DDL excluded.

Building It

The PCB layout shown in Figure 6 should result in something a good bit neater than my veroboard prototype will ever be, and the component overlay isn't too densely packed, so there should be no soldering problems to speak of. You're best off using IC sockets for the two CMOS chips, IC6 and IC7, and I'd also suggest inserting the components in the following order: PCB wire links, IC sockets, resistors, capacitors, and finally the two transistors. Chances are your jack sockets will be of the solder lug variety, in which case you'll have to use short wire links to mount them on the board. And make sure they're fairly sturdy, as they're all that supports the PCB once it's been fitted inside the delay unit's case.

With all the components on the PCB, the next task is to wire the whole thing into the DDL circuit. Figures 1 and 3 show boxed labels that refer to the points on the DDL circuit at which wire links should be inserted and are quite explicit in their instructions, as the inclusion of IC pin numbers will tell you. You'll also see numbers in all the boxes except the Freeze one, and these correspond to the numbered positions in Figures 5 and 7 for easy cross-referencing.

Making connections to IC legs shouldn't pose any problem, as it's simply a case of taking each IC out of its socket, bending the appropriate legs out from the body, returning the chip to the socket and soldering each bent-out leg (which is no longer connected to the circuit) to its corresponding wire. Take care not to hold your soldering iron against any leg for too long, or you'll damage the IC. There are a few connections which aren't to IC legs, *viz* the link between point 1 and one end of R55; the wires for points 12 and 13, which are +5V and digital 0V connections respectively and should be soldered to the tracks on top of the PCB; the wire between point 14 and S8, which can be inserted through a board connection hole (see Figure 7) and soldered from above; and the Freeze connection, which should be made according to the instructions in Figure 8 – note that S3 and JK4 are wired according to this Figure, not as shown in the original Powertran circuit diagram. One final point is to ensure that IC legs 3, 4, 5 and 6 should be linked together as shown in Figure 7. Any 5V and 0V lines you might need can be picked up from some convenient point on the main PCB.

The last job is to mount the PCB and the two switches inside the DDL case. The thing to do is mount the PCB on its two jack sockets towards the rear of the case, above the power supply board. In fact, Figure 9 should give you all the PCB mounting details you'll need.

Before you set out, bear in mind you'll have to remove both front and back panels in order to drill holes in them, not only because it makes life a hell of a lot easier, but also to avoid leaving any metal chips inside the case – they could short out tracks on the PCB. I've indicated that the switches should go on the front panel (it seems the most logical place for them to be), but you are of course at liberty to mount them wherever you wish. One disadvantage of putting them on the front panel is that they'll need fairly lengthy connecting wires to link them to the PCB, and you'll probably have to twist these together for neatness' sake.

Using It

If you want to use your modified DDL in 'Emulator' mode, you need two things: an external signal with a frequency between 190Hz and 1.5kHz and, ideally, a trigger signal as well. Assuming you're using a synthesiser of some description, set it up with just the one oscillator on and producing a sine or triangle waveform. If you have to use another waveshape, filter it so that it contains little more than a fundamental frequency component. Filter sweep should be set to zero, as should any VCO frequency modulation, at least when you're actually in the process of sampling a sound.

If no trigger or gate out facility is available on your particular synth, you'll have to use the VCA to effectively gate the clock signal on and off by setting its envelope to an on-or-off configuration.

Figure 3. Circuit diagram of external oscillator section.

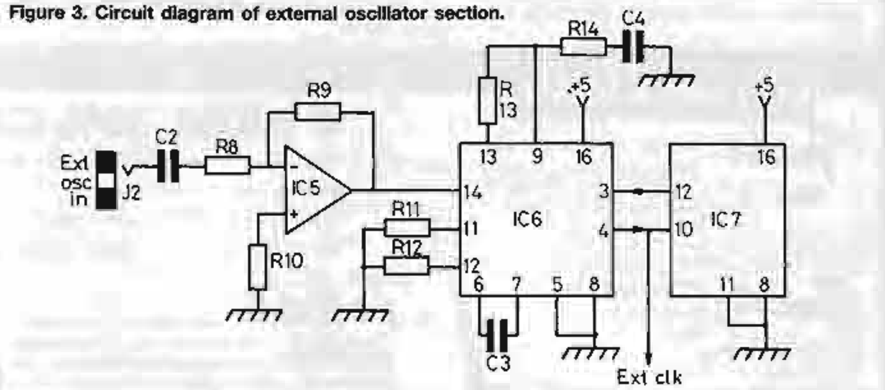


Figure 4. Operation of phase-locked loop as frequency multiplier.

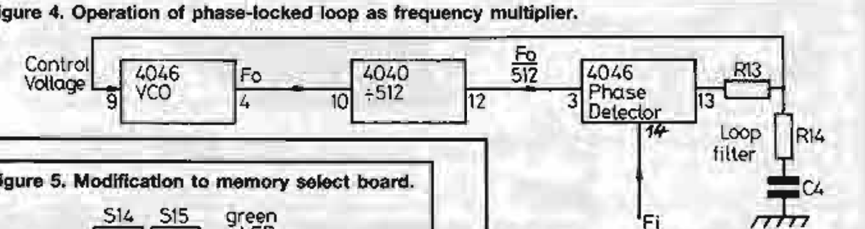
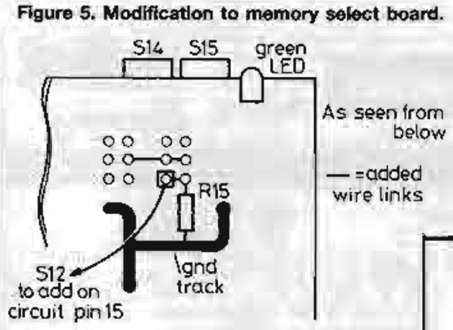


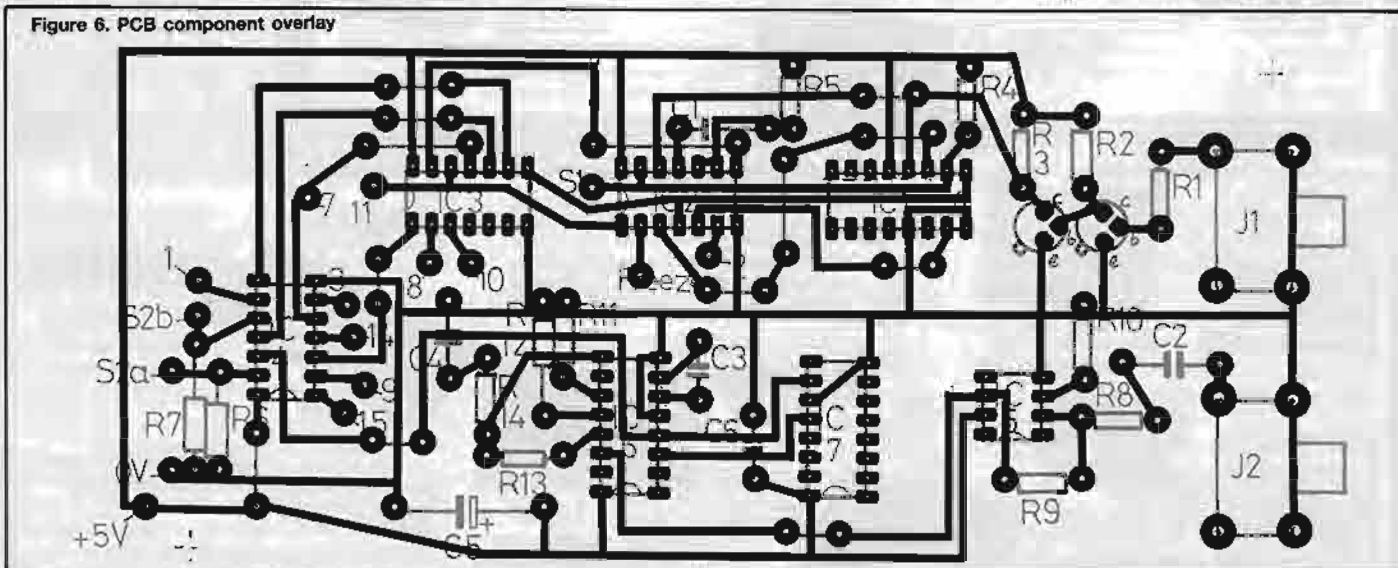
Figure 5. Modification to memory select board.



However, be warned that without any form of trigger signal, the DDL won't return to the start of a sample each time a new key is pressed. Thus a trigger really is a necessity, especially if you're going to be using samples with a decaying amplitude envelope, like percussion sounds, for instance.

If you do have a trigger at your

Figure 6. PCB component overlay



disposal, you can set the signal output from the synth to be permanently on, using an infinite sustain or hold facility if the synth permits. What makes this possible is the fact that the DDL stops as soon as it's been clocked to the end of its memory, which means that it'll automatically play the whole of the sample and then stop (rather than continuously looping as it would ordinarily do in Freeze mode), awaiting a trigger signal to tell it to start all over again.

As for sampling, there are a couple of possible approaches you can use depending on how many free hands you have, though in either case, you'll find a footswitch for the Freeze function is a real boon.

The first option is to set the trigger repeating regularly, using either a drum machine or a synthesiser's envelope repeat as the trigger source; the former's audibility makes it preferable. Don't set the trigger to occur too frequently, as this'll limit the amount of DDL memory you can use for the sample itself.

The sampling rate is set by the key you hold down on the synth. Try using a rate related to the keys in the second octave above Middle C (ie. 512-1024Hz), as this will keep aliasing down to a minimum. If you need a longer sampling time, you *can* use keys in a lower octave, but you'll have to switch the Powertran over to 4kHz bandwidth, again to minimise aliasing.

Make the sound you wish to sample just after the trigger beat occurs, hit the Freeze button, and your sample should then be held in memory awaiting playback...

The second option – the one that's dependent on your having a free hand – is to trigger the DDL manually by striking the appropriate synthesiser key just prior to making the sound to be sampled, and then hitting Freeze as before. You can of course sample sounds with the external oscillator switched out, something that would be particularly desirable if you only wanted percussive sounds triggered by a drum machine, say.

And the applications of the circuit aren't limited to sampling, either. For example, applying a drum machine to the external trigger input on its own can give you repeat echoes that are exactly in time with the beat – no more awkward fiddling with the Delay Time control! Come to think of it, varying the number of trigger pulses per measure as you go will give you effects changes (from long echoes to reverb, say) within the one pattern, without having to lay a finger on the DDL.

Alternatively, using the external oscillator on its own will free you from the single triangle wave option on the Powertran's own LFO. Is there no end to this? ■

As mentioned above, a PCB for the above circuit complying to the layout illustrated is available direct from E&MM at £3.95 including VAT, postage and packing. Please make cheques/POs payable to Music Maker Publications, and allow 28 days for delivery.

Figure 7. Location of wiring points.

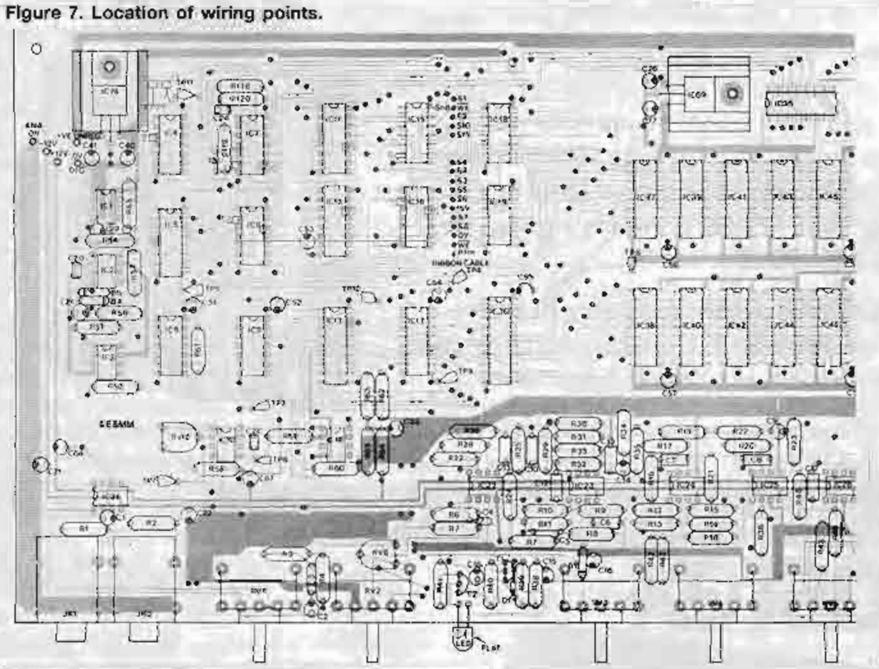


Figure 8. Location of Freeze connection.

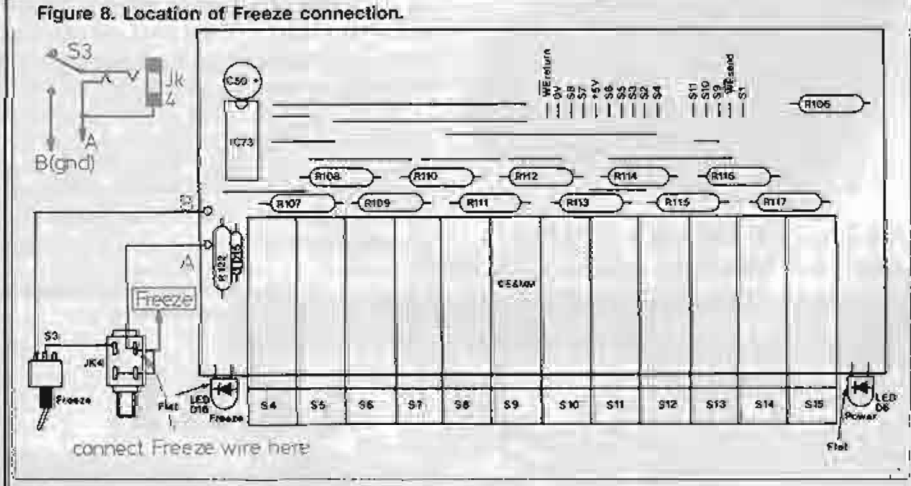
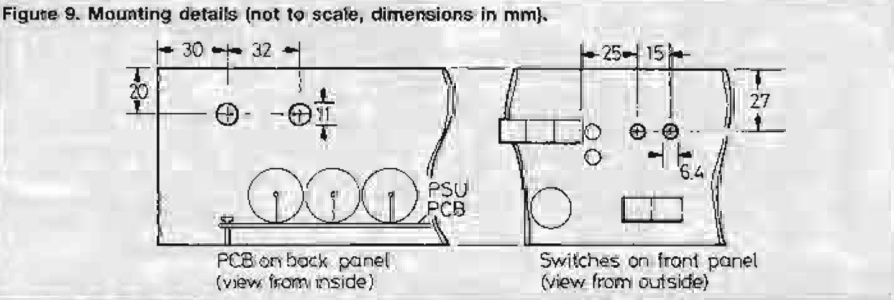


Figure 9. Mounting details (not to scale, dimensions in mm).



Parts List

Resistors

R1	10K
R2, 3, 4, 6, 7, 15	1K2
R5	2K4
R8, 10	33K
R9, 12, 13	470K
R11	6K8
R14	43K

Capacitors

C1	270pF polystyrene
C2	220nF carbonate
C3	47pF polystyrene
C4, 6	100nF carbonate
C5	47μF 16V axial electrolytic

Semiconductors

TR1, 2	BC108
IC1	74LS76
IC2	74LS00
IC3	74LS32
IC4	74LS08
IC5	UA741
IC6	4046BE
IC7	4040BE

Miscellaneous

J1, 2	mono jack socket
S1, 2	sub-min toggle (single-pole, changeover)

Eight-pin DIL socket, 1 off
14-pin DIL socket, 3 off
16-pin DIL socket, 3 off