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The ETI Guide to Making Music

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Published by Argus Specialist Publications Ltd
1 Golden Square
London W1R 3AB
Tel: 01-437 0626

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The ETI Guide To Making Music is published as a sister magazine to Electronics Today International — the leading UK magazine for electronics enthusiasts. If you have any comments on this magazine or wish to contribute articles or projects to ETI please write to the Editor at: ETI, 1 Golden Square, London W1R 3AB.

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CHORUS UNIT

Kick up an audio treat with this trio of matching effects units. First up is the chorus unit, offering maximum versatility in the minimum space

Chorus is an attempt to emulate the sound of another instrument or voice playing or singing along in perfect unison with the original. In actual fact if this was properly achieved — with the original and copy exactly simultaneous and identical — the audible effect would merely be an increase in volume. The chorus effect works by slightly delaying a portion of the audio signal, slowly varying this delay, and mixing the result with the original.

The two signals then interact, sometimes adding together, sometimes cancelling out. Our ears are fooled into hearing a second instrument or voice.

The Circuit

The block diagram is shown in Fig. 1. The heart of the chorus unit is a Bucket Brigade Device (BBD), in this case a TDA1536 which has 1536 stages or 'buckets'. The device works by pouring water from one bucket to the next, thus delaying its passage to the end of the bucket line.

Well, actually it isn't water that travels but charge. The buckets are capacitors and transistors do the passing, with the delay time dependant on the clock frequency applied to the BBD line.

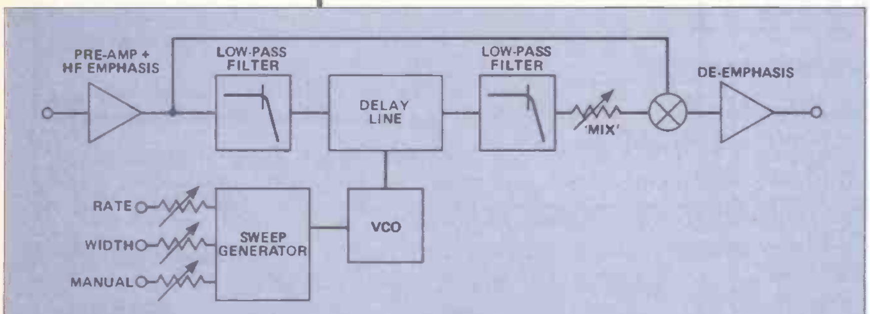


Fig. 1 Block diagram of the chorus

For chorus, a delay range of seven to 20 milliseconds is about right and this gives a minimum clock frequency of 38.4kHz and a maximum of 109.7kHz. The TDA1097 is only specified to 100kHz, but no problems were encountered with the prototype.

A BBD line is essentially a sampling device, and the clock signal will inevitably find its way to the output, albeit at a much lower amplitude than the audio signal. The clock frequency itself never falls below 38.4kHz so it will not be audible. The problem is that as the harmonic or noise frequency components of the input signal approach half the clock frequency, the lower sideband of the clock frequency will become audible.

For example, noise components in the range 10kHz to 40kHz will mix with the clock signal to produce difference signals in the range 1.6kHz to 28.4kHz. Trying to filter out these difference signals is obviously impractical, since doing so would also get rid of most of the audio signal.

The solution is a low-pass filter immediately before the input to the BBD line, with a cut-off frequency of around 6kHz. In this way, difference signals can only be produced above 32kHz, and a similar low-pass filter on the output of the BBD line effectively gets rid of those.

A cut-off frequency of 6kHz may seem a bit brutal but the chorus effect ceases to be audible above this, and the original signal will be unaffected anyway.

The filters in this design have an actual cut-off of 6.2kHz and a slope of −20dB/octave. In addition, pre-emphasis on the input and de-emphasis on the output endow this unit with a good noise performance.

The sweep rate of the delay time can be varied from once every ten seconds to ten times a second, and the width of the sweep is continuously variable from full sweeps between the limits of 7ms and 20ms delay and no sweep at all (a constant delay period).

EFFECTS

HOW IT WORKS

The circuit diagram is shown in Fig. 2. IC2a is a high-impedance input buffer, R7 and C5 provide high frequency pre-emphasis to the input. Part of this signal goes to Mix control RV1 and part goes through IC2b and a low-pass filter built around IC3a. IC4, the BBD line, requires two DC bias voltages and these are provided by the divider chain R12, R13 and R14. Two anti-phase clock signals are supplied by IC5 used as a simple voltage controlled oscillator. IC4 has increased attenuation since its supply is well below the normal 12V.

IC3b is the other low pass filter and Q1 switches the signal through to the next stage if the effect is selected. IC3c compensates for the attenuation of IC4. RV1 mixes the effect with the undelayed signal. It can be seen that at one end of the Mix potentiometer is the un-delayed signal and at the other is the delayed signal. The position of the Mix control determines the proportion of each that appears at the output.

The 1n0 capacitor (C21) across the feedback resistor of IC2c will reduce the high-frequency response of this stage but remember that high-frequencies were boosted at the input stage, so the overall response is fairly flat.

IC6 is the sweep-generator. IC6a is an integrator, and IC6b is connected as a Schmitt. If the voltage on IC6 pin 7 is of a sufficiently high level, pin 1 will also be high. This will cause pin 7 to ramp downwards at a rate determined by R32, C23, C24 and the Rate control. When the voltage is slow enough, it will cause IC6b to switch, sending its output low. This will cause IC6 pin 7 to ramp upwards, and the cycle repeats itself.

At the 7ms end of the delay-time range, the rate of change is high but inaudible. At the other end, the rate of change is much slower. This is thanks to bipolar transistors having non-linear switch-on characteristics at low base currents, and helps us achieve an effect like that in Fig. 4. RV3 and RV4 are used to adjust the shape of the waveform.

When the width control is fully clockwise, it can be seen that the signal present at IC3d will appear at the input of the VCO, and therefore the delay-time will sweep over the entire range. As the width control is turned counterclockwise, the input to the VCO is derived more and more from IC2d, whose output is set by the Manual control, RV5. Thus the Width control provides the option of a fully swept delay-time, a fully manual delay-time, or anything in between.

Further versatility is provided by a *Mix* control which allows the delayed and undelayed signals to be mixed in any proportions. This makes it possible, for example, to use the full amount of sweep but still achieve a very subtle effect.

Power to the unit is provided by a PP3 size battery (preferably alkaline) and a socket allows connection to an external supply. The unit is switched on by inserting a mono jack plug into the input socket, and *Effect* or *Bypass* mode is selected by the built-in footswitch or by a remote switch connected to the REM socket. An LED indicates when the unit is in *Effect* mode.

Construction

Commence assembly by installing the wire link, the four jack sockets (see *Buylines*) and if desired sockets for ICs 4, 5 and 6. Sockets cannot be used for ICs 2 and 3 or they will interfere with the potentiometers when the unit is assembled.

Solder into place the resistors, capacitors, and presets, taking care that the capacitors near the connector end of the board are mounted flat so as to clear the potentiometers. Next fit the diodes, transistors, and ICs 1, 2 and 3. Cut to length three pieces of ordinary insulated connecting wire, and solder them between the points shown on the PCB overlay, then fit the two battery-guide pillars and the battery connector. Connect the four potentiometers, R28 and the LED to the PCB using insulated wire. Lastly, fit ICs 4, 5 and 6. The board can now be tested.

An oscilloscope is almost essential if the chorus unit is to be accurately set up, so if you don't own one you will have to borrow or otherwise acquire one before proceeding further.

Connect the oscilloscope input to IC5 pin 2 and

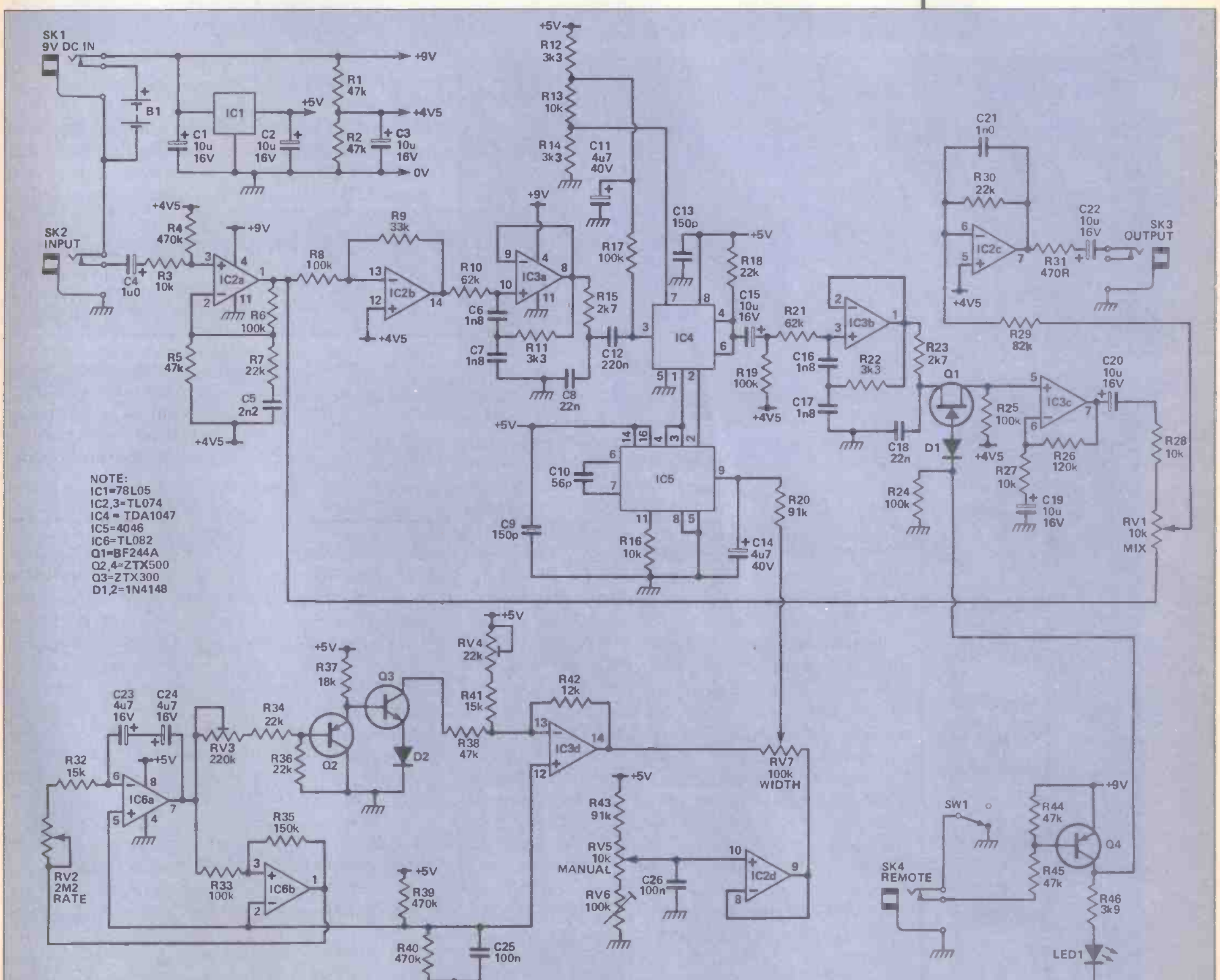


Fig. 2 Circuit diagram of the chorus unit

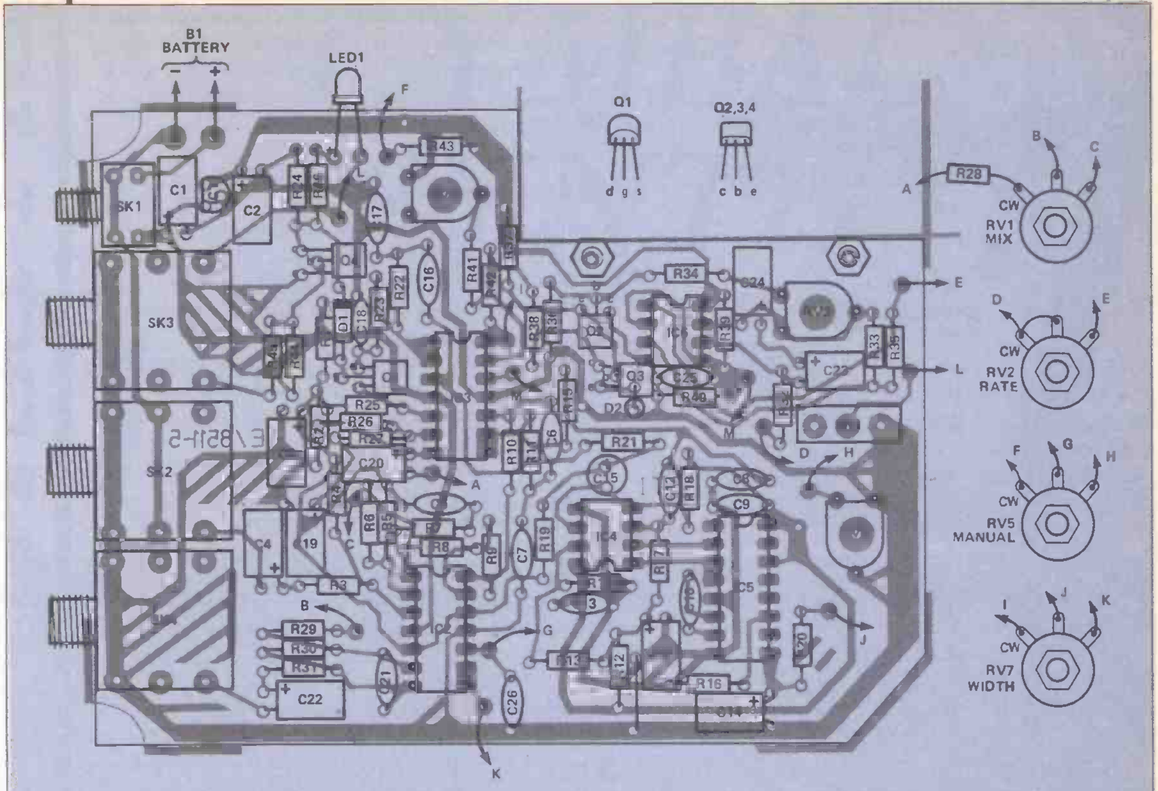


Fig. 3 Component overlay of the chorus unit

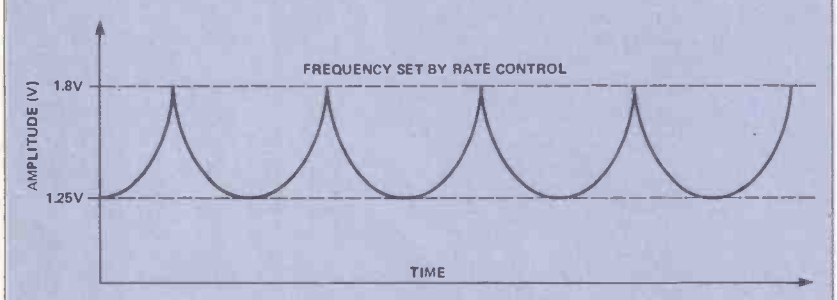


Fig. 4 Sweep generator output at IC3 pin 14

EFFECTS

check that a square wave signal is present with an amplitude of about 5V peak-to-peak. Turn the *Width* and *Manual* controls fully anti-clockwise and adjust RV6 until the frequency of the square wave is about 38.4 kHz. Turn the *Manual* control fully clockwise and check that the frequency rises to about 109kHz. If either of these frequencies are outside the range of adjustment of RV6, try altering the value of R43.

Move the scope input to IC6 pin 7 and check that a triangular waveform is present with an amplitude of about 2V peak-to-peak. Rotate the *Rate* control (RV2) and check that the frequency varies from about 0.1Hz when it is fully anti-clockwise to 10Hz when it is fully clockwise. Move oscilloscope probes to IC3 pin 14 and check that the waveform present is similar to that shown in Fig. 4. Make any necessary adjustments, using RV3 to set the amplitude and RV4 to alter the offset.

Return the oscilloscope connections to IC5 pin 2 and set the *Width* control fully clockwise and the *Rate* control fully anti-clockwise. The frequency of the signal should be slowly changing between the previously-set limits of 38.4kHz and 109kHz. Carry out any fine tuning required using RV3 and RV4 and the setting up is complete.

If you are unable to find the recommended box choose one that is slightly larger than our prototype so as avoid cramping the components.

Refer to Fig. 5 and drill the necessary holes as accurately as possible. If you have any doubts about your skills in this direction, try drilling the holes a little smaller than is required in the first instance, then offer up the PCB to check that they coincide and make any necessary adjustments with a small file before enlarging the holes to their final diameter.

When drilling is complete, rub down the outside of the box with glass paper to deburr the holes and prepare the surface. Clean the box thoroughly and then prime and paint it allowing suitable drying times. Loosely assemble the potentiometers and knobs as a guide and use rub-down lettering to apply the legends. Remove the fittings, lightly buff the surface to remove any fingerprints, then apply a coat of clear varnish and leave to dry.

Glue a piece of foam rubber inside the box to prevent the battery rattling around. Mount the switch in position through the appropriate hole in the case but do not tighten the fixing nut. Mount the potentiometers through the top panel of the case. Place fibre washers onto each of the sockets on the PCB, then offer the board up to the case, socket end first. Loosely assemble the socket securing nuts from the outside to stop the board slipping back through. Jiggle the switch and the PCB until the switch pins appear through the holes in the board, then solder them to the pads and tighten the switch into position. Complete the construction by adding the switch cap, the knobs and the base plate and tightening the securing nuts on the sockets. Don't forget to install a battery!

Test the chorus unit by applying an input signal of about 1kHz at a few hundred millivolts to SK2. Inserting the jack plug should turn the unit on. Use the scope to check the output closely resemble the input when the unit is off, and that the waveform is altered with the unit on.

Having got the pretend stuff out of the way, you can now plug in a real instrument, hook the unit up to your amplifier and get chorused away!

PARTS LIST

RESISTORS (all ¼W 5%)

R1,2,5,38,44,45	47k
R3,13,16,27,28	10k
R4,39,40	470k
R6,8,17,19, 24,25,33	100k
R7,18,30,34,36	22k
R9	33k
R10,21	62k
R11,12,14,22	3k3
R15,23	2k7
R20,43	91k
R26	120k
R29	82k
R31	470R
R32,41	15k
R35	150k
R37	18k
R42	12k
R46	3k9
RV1,5	10k linear potentiometer
RV2	2M2 linear potentiometer
RV3	220k horiz skelton preset
RV4	22k horiz skeleton preset
RV6	100k horiz skeleton preset
RV7	100k linear potentiometer

CAPACITORS

C1-3,15,19,20,22	10µ 16V radial electrolytic
C4	1µ0 63V radial electrolytic
C5	2n2 polyester
C6,7,16,17	1n8 polyester
C8,18	22n polyester
C9,13	150p polystyrene
C10	56p polystyrene
C11,14	4µ7 40V radial electrolytic
C12	220n polyester
C21	1no polystyrene
C23,24	4µ7 16v tantalum
C25,26	100n polyester

SEMICONDUCTORS

ICI	78L05
IC2,3	TL074
IC4	TDA1097
IC5	4046
IC6	TL082
W1	BF244A
Q2,4	ZTZ500
Q3	ZTX300
D1,2	1N4148
LED1	miniature red LED with mounting bezel

MISCELLANEOUS

SK1	3.5mm miniature jack socket, PC mounting, with switch
SK2	¼in stereo jacket socket, PC mounting
SK3	¼in mono jack socket, PC mounting, with switch
SK4	¼in mono jack socket, PC mounting
SW1	SPDT alternate action push switch, panel mounting

PCB. Case. Knobs, 4 off. Battery connector, ¼in (20mm) high pillars, 2 off and screws or bolts to fit. IC sockets if desired, 2 off 8 pin and one off 16 pin DIL. Thin foam rubber. 9V battery, PP3 or similar.

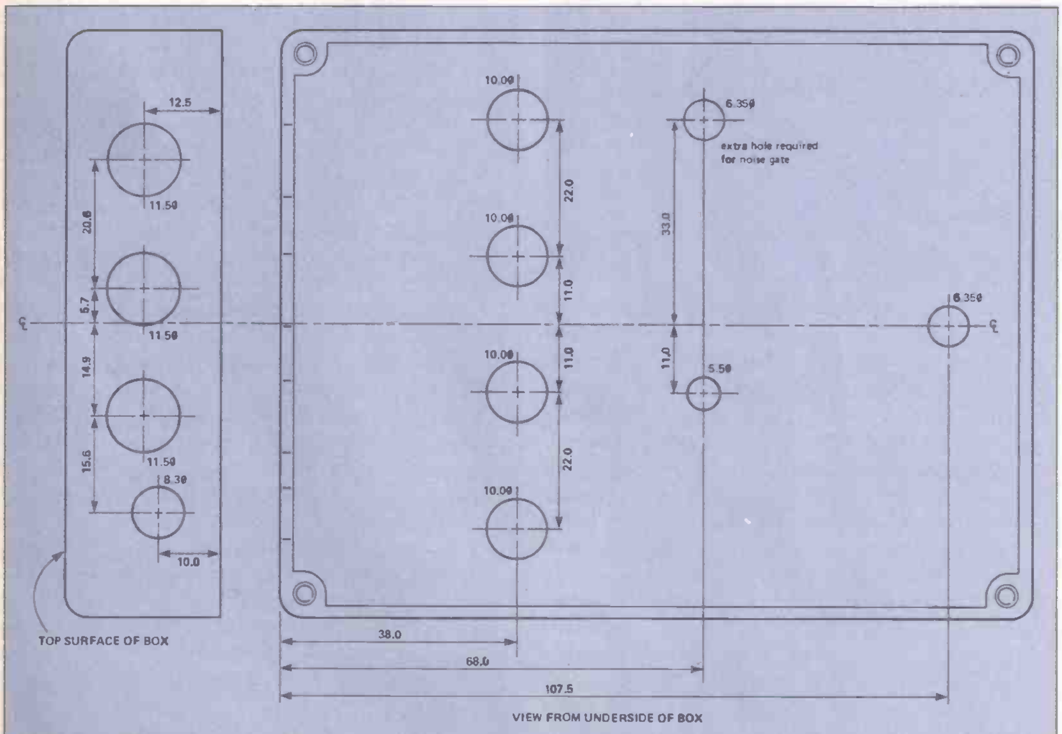


Fig. 5 Drilling diagram for the case

BUYLINES

The ¼in sockets in the prototype were PCB mounting sockets made by Cliff – unfortunately tricky to get hold of in small quantities. However, Cliff's panel mounting sockets are available from Electrovalue (Tel: (0784) 33603). These have solder tags with eyelets easily cut to suit the PCB holes. Many available sockets will fit, but check the spacing of the tags against the PCB. The potentiometers used were also from Electrovalue, from their P20 range.

Electromail (Tel: (0536) 204555) stock a suitable switch (part 339-241) and 15mm button for it (339-279). The box is from STC (Tel: (0279) 626777).

All other components should be available from any supplier except perhaps IC4 – TDA1022 – available from Electromail.

FLANGER UNIT

This flanger pedal sweeps as it beats as it filters

EFFECTS

Flanging is closely related to the chorus effect. The delay time in the chorus effect is in the range 7 to 20ms. That of a flanger (similarly varied by a sweep generator) is shorter, in the range 1 to 13ms.

The other difference is the addition of a regeneration control by means of which part of the delayed signal is sent back round to be re-delayed. This would be a bit like reverberation if the delay-time was not so short. In fact, it is regeneration that gives the flanger its characteristic metallic sound.

Charge of The Bucket Brigade

The device at the heart of the flanger is again a Bucket Brigade Delay-line (BBD) but a different one, a TDA1022 IC which has 512 stages or buckets, and requires clock frequencies in the range 19.69kHz to 356kHz. Again there is a low-pass filter both before and after the BBD to deal with harmonics and noise.

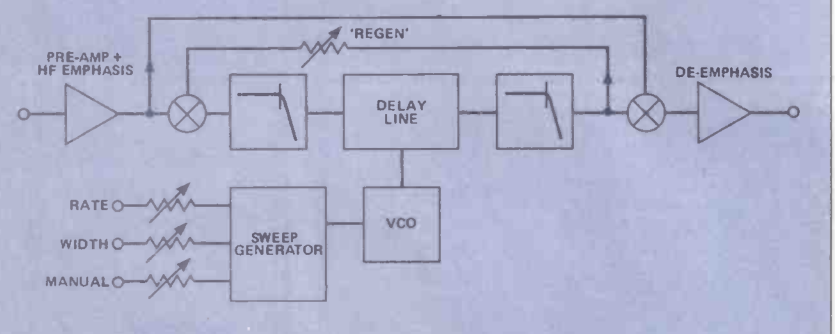


Fig. 1 Block diagram of the flanger

The bulk of the energy in most music is contained in the low frequencies so a little bit of high-frequency pre-emphasis is added to the flanger's input signal. This gently lifts lifts the top-end of the input signal above the noise generated by the BBD line, under which it may otherwise get lost. De-emphasis has a further benefit in that it also filters out a little more of the noise introduced by the BBD line.

The delay time of the flanger is, of course, completely adjustable within the range 1 to 13ms. Rate, width and manual controls allow a wide range of effects to be obtained.

With the width control at a maximum (fully clockwise), the delay time will sweep from maximum to minimum at a speed determined by the position of the rate control. As the width control is turned anti-clockwise, so the width of the sweep is reduced until it reaches zero sweep. At this point the delay time is set by the manual control and the rate control will have no effect.

The regeneration control lets the user determine how much of the signal will go back into the BBD line to be re-delayed. Fully anti-clockwise, the flanger will sound something like a chorus. With the regeneration control at a maximum (fully clockwise), the flanger produces an intense, metallic sound.

In between lies the classic phlanged sound, ideal for adding depth to picked electric and 12-string guitar passages or for giving a new dimension to metal guitar in line with a fuzz (or hyper-fuzz!) pedal.

Construction

Begin by fitting the four jack-sockets (see *Buylines*) and the wire link to the PCB. Sockets can be used for IC4, IC5 and IC6 but not for IC2 or IC3 since height is restricted at the end of the PCB when the unit is assembled in its case and IC Sockets would raise the chips to high.

Next fit resistors, capacitors, and presets. Note that most of the capacitors at the socket end of the board are mounted flat, again because of restricted height.

Continue assembly by soldering lengths of insulated connecting wire between the points shown on the component overlay diagram and connect the four potentiometers and the LED. Next, wire-up the battery connector and fix the two battery guide pillars.

The switch may be fitted temporarily for testing the PCB but will have to be removed before the board can be assembled into the case. Alternatively, use a piece of insulated wire.

Testing

The PCB can now be tested. An oscilloscope will be necessary for accurate setting-up of the flanger.

With a battery connected, press the switch and check the LED lights. Connect the scope to IC5 pin 2 and check that a square wave signal is present, with an amplitude of about 5V peak-to-peak. With the width and manual controls fully anti-clockwise, adjust PR3 until the frequency of the square wave is about 20kHz. Turn the manual control fully clockwise and check that the frequency rises to about 250kHz. If either of these frequencies is outside the range of adjustment of PR3, try altering the value of R45. A frequency counter could be used for accurate setting of the limits 19.69kHz and 256kHz.

Next, connect the scope to IC6 pin7 and check that a triangular waveform is present with an amplitude of about 2V peak-to-peak. With the rate control fully clockwise this frequency should be about 10Hz and fully anti-clockwise it should be about 0.1Hz.

Connect the scope to IC3 pin 14 and check for a waveform similar to that shown in Fig. 2. PR1 adjusts the shape of this waveform and PR2 its offset.

Connect the scope to IC5 pin 2. Turn the width control fully clockwise and the rate control fully anti-clockwise. The frequency of the signal should be slowly changing between the previously-set limits of 19.69kHz and 256kHz. Use PR1 and PR2 to make any final adjustment. A frequency-counter cannot be used for this measurement, because the frequency is constantly changing.

HOW IT WORKS

The functions of IC2a, IC4 and IC3b are exactly as described in the chorus article. The next stage is an amplifier based on IC3c, with enough gain to compensate for the attenuation of the delayline. The output from IC3c goes to the output stage, IC2c, and also back to IC2b via the regeneration control.

The RC network (R17 and C11) across the feedback resistor (R16) of IC2c will reduce the high-frequency response of this stage but remember that the signal received HF pre-emphasis at the input stage, so the overall response is flat. This de-emphasis on the output stage also serves to get rid of more of the noise generated by the BBD line.

The sweep generator operates in the same way as for the chorus pedal.

The modified ramp signal is applied to one end of the width control, and at the other is a DC voltage, set by the position of RV4, the manual control. With the width control fully clockwise, the VCO gets its control voltage entirely from the sweep generator, so the delay-time is swept across its range. Turn the width control fully anti-clockwise, and the VCO frequency is set solely by the manual control. Thus the width control provides the option of a fully swept delay-time, a fully manual delay-time, or anything in between.

The effect is selected by the footswitch, SW1 or by a remote switch connected to the REM (remote) socket.

All the op-amps are supplied with +9V, except IC6, which gets +5V. The BBD line and its clock generator are also supplied with +5V from IC1, a 78L05 voltage regulator. Some parts of the circuit require half the battery voltage, and others require +2.5V. These voltages are provided by potential dividers.

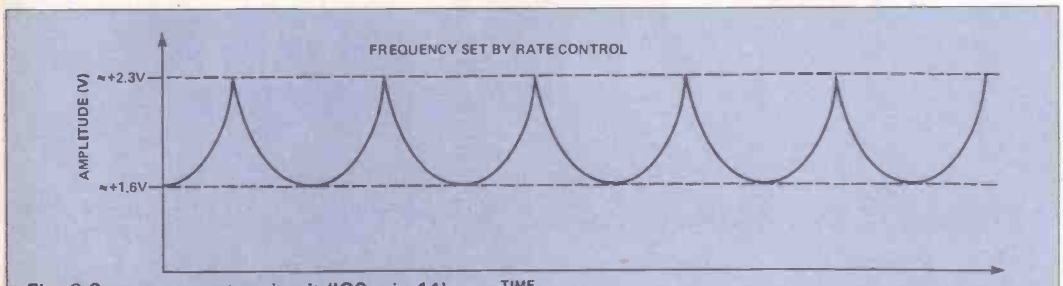


Fig. 2 Sweep generator circuit (IC3, pin 14)

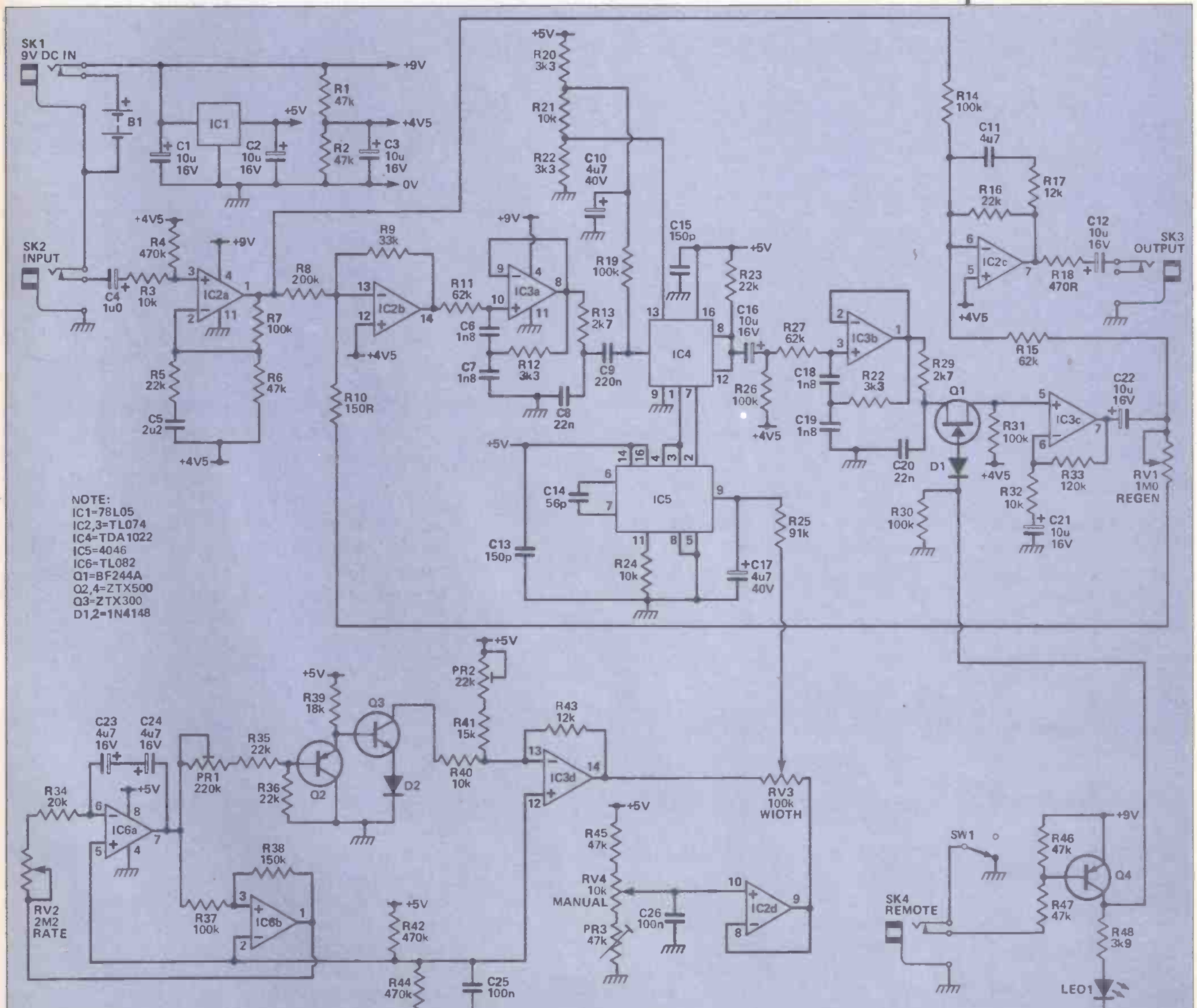


Fig. 3 Circuit diagram of the flanger

The Case

The same case is used as for the chorus unit. Drilling is identical (Fig. 4 in the Chorus Unit article) and the box is put together in the same way to produce a matching effects line — the noise gate is also of the same external design.

Again test the unit with a 1kHz signal at a few mV and watch the output on a scope. checking input and output are identical when the switch turns the LED off. When the LED is on. the pedal should be operating and its effect should be observable — and of course audible!

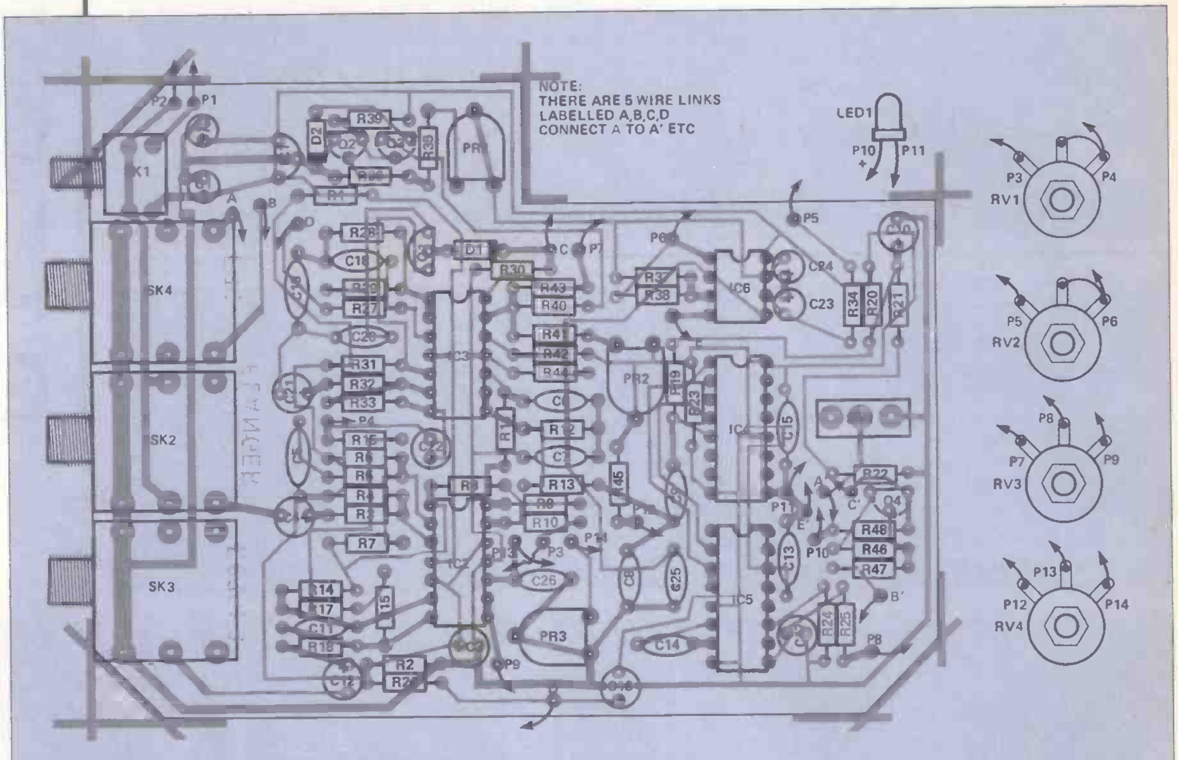


Fig. 4 Component overlay for the flanger

PARTS LIST

RESISTORS (all ¼W 5%)	
R1,2,6,45-47	47k
R3,21,24,32,40	10k
R4,42,44	470k
R5,16,23,35,36	22k
R7,14,19,26,30,31,37	100k
R8	200k
R9	33k
R10,38	150k
R11,15,27	62k
R12,20,22,28	3k3
R13,29	2k7
R17,43	12k
R18	470R
R24	91k
R33	120k
R34	20k
R39	18k
R41	15k
R48	3k9
RV1	1M0 linear pot.
RV2	2M2 linear pot.
RV3	100k linear pot.
RV4	10k linear pot.
PR1	220k horiz. skelton preset
PR2	22k horiz. skelton preset
PR3	47k horiz. skelton preset

CAPACITORS	
C1-3,12,16,21,22	10µ 16V radial elect.
C4	1µ0 63V radial electrolytic
C5	2n2 polystyrene
C6,7,18,19	1n8 polystyrene
C8,20	22n polyester
C9	220n polyester
C10,17	4µ7 40V radial elect.
C11	4n7 polystyrene
C13,15	150p polystyrene
C14	56p polystyrene
C25,26	100n polyester
C23,24	4µ7 16V tantalum

SEMICONDUCTORS	
IC1	78L05
IC2,3	TL074
IC4	TDA1022
IC5	4046
IC6	TL082
Q1	BF244A
Q2,4	ZTX500
Q3	ZTX300
D1,2	1N4148
LED1	miniature red LED with mounting bezel

MISCELLANEOUS	
SK1	3.5mm miniature jack-socket, PC mounting, with switch
SK3	¼in stereo jack sockets, PC mounting
SK2,4	¼in stereo jack socket, PC mounting, with switch
SW1	SPDT alternate action push-button switch, panel mounting

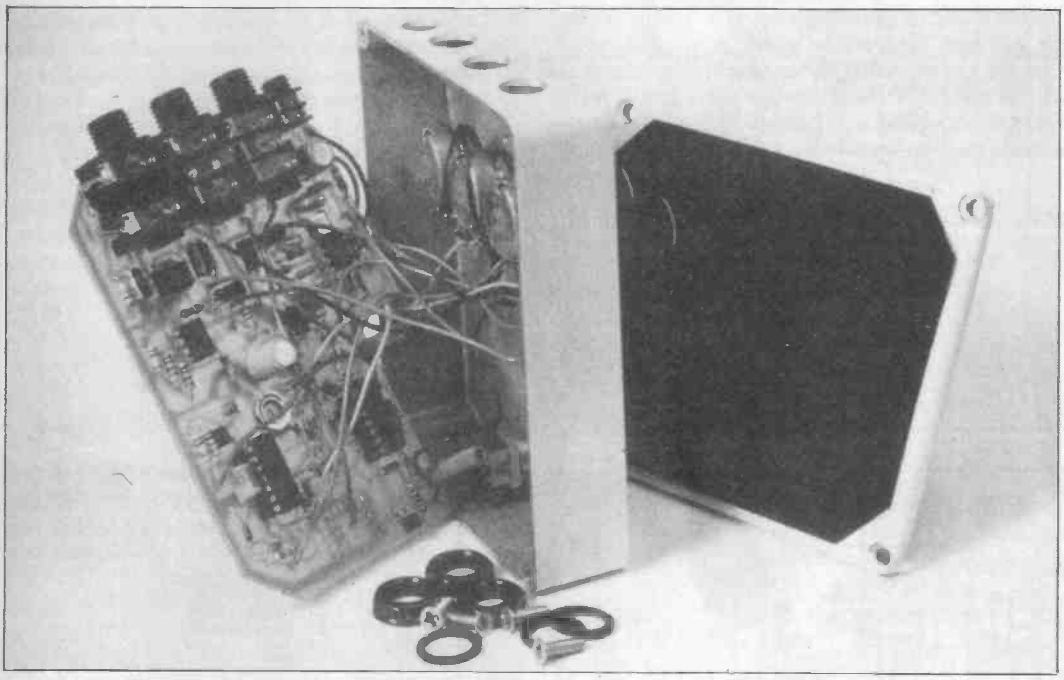
PCB. Case. Knobs, Battery connector. Two 20mm pillars, with screws to fit. IC sockets. 9V battery, PP3 or similar. Connecting wire. Thin foam rubber. Rubber sheet for base.

For *Buylines* see Chorus Unit.

NOISE GATE UNIT

EFFECTS

The last of our matching pedals is an enveloped noise gate to sort the notes from the noise



Every musician knows the problems caused by noisy leads and effect-units: whenever you stop playing, the snaps, crackles and pops are still there. This state of affairs is acceptable when practising but is a major headache when recording or playing live. One solution is a noise-gate, the electronic equivalent of pulling the jack-plugs out every time you stop playing. Needless to say, the noise-gate does it so unobtrusively that you'd never know it was there, which is the whole idea!

Important parameters of a good-noise-gate are:
Threshold: the input signal level required to open the gate, adjustable from −35dBm down to −65dBm approximately. Normally it will be set just above the noise-floor, so that when playing begins, the increase in signal-level is sufficient to open the gate.
Response time: this is the time taken for the noise-gate to begin opening once the threshold has been crossed — ideally instantaneous, in practice less than a millisecond.
Attack-time: the time from fully closed to fully open. Most noise-gates open instantly, which is what is usually required. This design will do so if you want it to, but can also be adjusted to take up to 100ms to open.
Hold-time: the period for which the noise-gate remains fully open after playing has stopped. It is adjustable between 100ms and 2s.
Decay-time: the time taken for the noise gate to close after the Hold-time has elapsed. This really sets this design apart from others — it will reach the fully closed state within 100ms if you want but it can be set to take as long as two seconds, causing any noise to go away unobtrusively rather than abruptly.

As well as being triggered by the incoming signal, the noise-gate may also be opened by another signal connected to the EXT. KEY socket, by a logic level on the REM socket, by a switch contact (also on the REM socket), or by the built-in footswitch. Whichever triggering method is used, the attack, hold, and decay controls still function. Because the envelope shape is

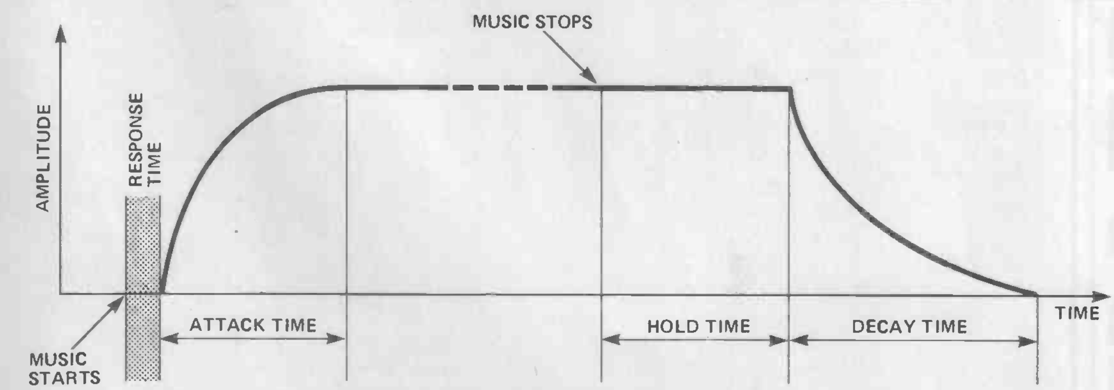
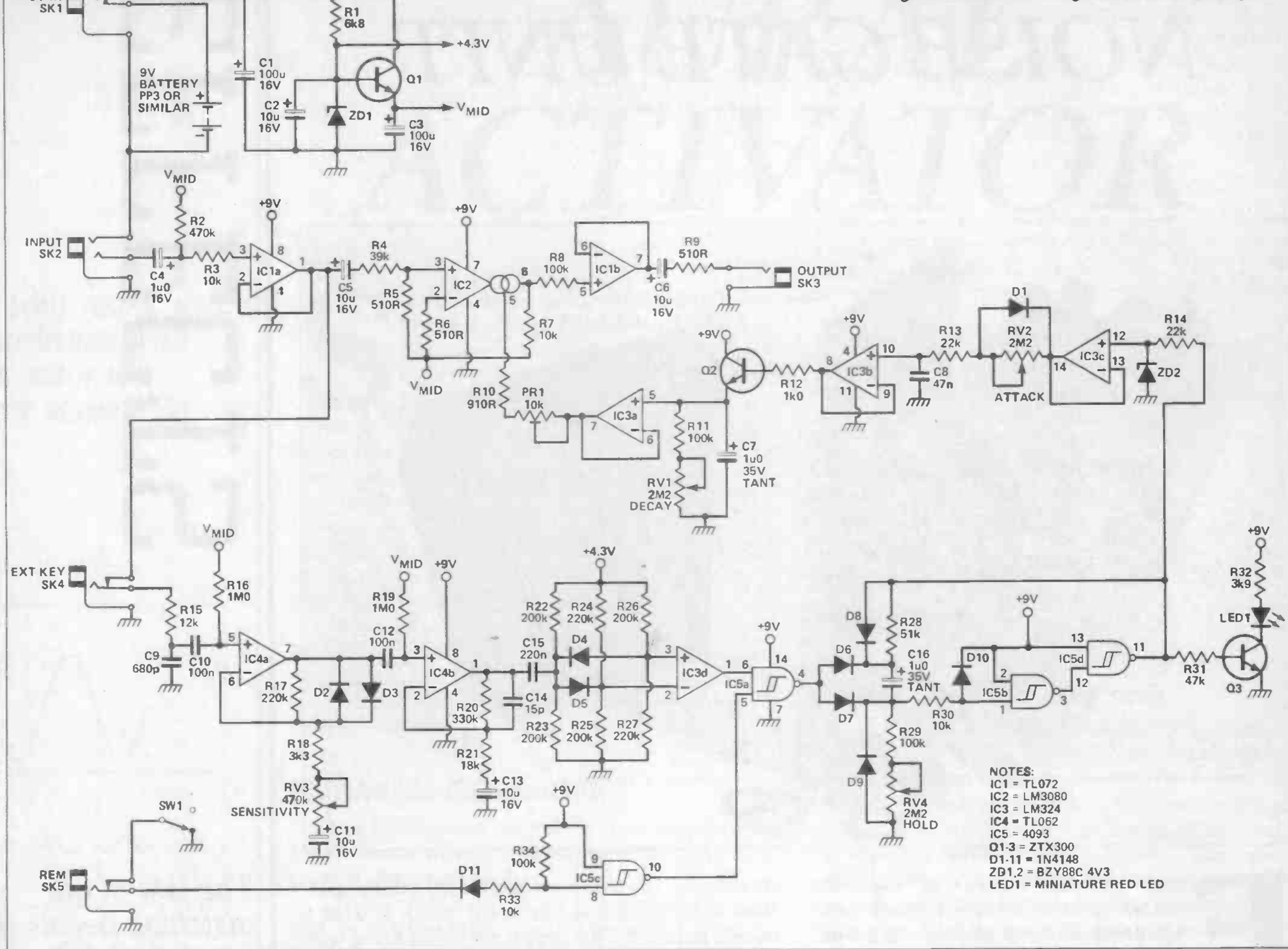


Fig. 1 The response envelope produced by the noise gate

Fig. 2 The circuit diagram of the noise gate



PARTS LIST

RESISTORS (all ¼W 5% unless otherwise stated)	
R1	6k8
R2	470k
R3,7,30,33	10k
R4	39k
R5,6,9	510R
R8,11,29,34	100k
R10	910R
R12	1k0
R13,14	22k
R15	12k
R16,19	1M0
R17,24,27	220k 2%
R18	3k3 2%
R20	330k
R21	18k
R22,23, 25,26	200k 2%
R28	51k
R31	47k
R32	3k9
RV1,2,4	2M2 logarithmic
RV3	470k logarithmic
PR1	10k horizontal skeleton preset
CAPACITORS	
C1,3	100µ 16V radial electrolytic
C2,5,6,11,13	10µ 16V radial electrolytic
C4	1µ0 16V radial electrolytic
C7,16	1µ0 35V tantalum bead
C8	47n multi-layer
C9	680p polystyrene
C10,12	100n multi-layer
C14	15p polystyrene
C15	220n multi-layer
SEMICONDUCTORS	
IC1	TL072
IC2	LM30809
IC3	LM324
IC4	TL062
IC5	4093
Q1-3	ZTX300
D1-11	1N4148
ZD1,2	BZY88C 4V3
LED1	miniature red LED with mounting bezel
MISCELLANEOUS	
SK1	3.5mm miniature jack socket, PC mounting, with switch
SK2	¼in stereo jack socket, PC mounting, with switch
SK3	¼in mono jack socket, PC mounting
SK4	3.5mm miniature jack socket, panel mounting, with switch
SK5	¼in mono jack socket, PC mounting, with switch
SW1	SPDT alternative action push switch, panel mounting

PCB. Case. Knobs. Battery connector. ¾in mounting pillars. 14-pin DIL IC sockets. Foam rubber. 9V battery (PP3).
For *Buylines* see Chorus Unit.

HOW IT WORKS

The circuit is shown in Fig. 2. Again IC2 is a transconductance amplifier with IC1 as unity gain buffers. The threshold detector consists of IC3d and the two halves of IC4. R15 and C9 form a low-pass filter which removes RF noise followed by amplifier stage IC4a whose gain is set by the sensitivity control. IC4b ensures that sufficient level is available to reach the threshold of the comparator.

The window comparator is based around IC4d which pushes pin 2 of IC3d higher than pin 3 via D5, or pulls pin 3 lower than pin 2 via D4. Provided the gate is not in the bypass mode, pin 5 of NAND Schmitt trigger IC5a will be logic high and the negative going pulses from the output of IC3d will produce positive going pulses on pin 4 of the Schmitt.

D6 and D7 will then conduct and hold the two ends of C16 at the same potential, preventing it from charging. IC5b and IC5d both have one input connected to the positive supply and will thus act as Schmitt inverters. Pin 1 of IC5b will be held high via R30 causing its output to stay low, and this low appearing on pin 12 of IC5d will force pin 11 high.

When the pulses at the output of IC5a cease, D6 and D7 will no longer conduct and C16 charges via D8 from the logic high on IC5d's output, the rate determined by RV4. The voltage across R29 and RV4 will fall and pin 1 of IC5b will be pulled low via R30. At a point determined by the Schmitt, IC5b will change state, going high and switching IC5d low. No further charging of the capacitor can take place and the circuit will remain in this state until a further train of pulses is received from IC3d and IC5a.

If bypass mode is selected either by SW1 or a logic signal into SK5, IC5a pin 5 will be held low via the Schmitt inverter IC5c. This will cause IC5a pin 4 to remain high, whereupon D6 and D7 will conduct, IC5b pin 1 will be held high via R30 causing pin 3 to go low, and the resulting low on pin 12 of IC5d will cause pin 11 to remain high. This pin will stay high as long as the unit is in bypass.

This high level drives the GATE OPEN LED via Q3 and R31, R32 and provides a voltage into pin 12 of IC3c. This voltage is held down to 4.3V by ZD2 and R14. IC3c is a unity gain buffer stage which can charge C8 via R13 and RV2. The time taken to charge C8 is the attack time and is adjusted by RV2. The voltage on C8 is buffered by IC3d and drives Q2 which charges C7. This voltage corresponds to the decay portion of the envelope and the discharge is adjusted by RV1. IC3a is another unity gain buffer which couples the composite envelope shape voltage to the gain-determining pin of IC2. PR1 adjusts the overall gain of the audio path back to unity.

The complete circuit operates as follows. When the input signal exceeds the threshold, pulses at pin 1 of IC3d send IC5d pin 11 high with no apparent delay and this in turn produces 4.3V at pin 14 or IC3c. IC3b pin 8 will also rise to 4.3V but will do so exponentially because of the action of C8, R13 and RV2. C7 is much larger than C8 but it will charge at the same rate because of Q2. As the voltage on this capacitor rises, so will the current flowing into pin 5 of IC2 and so the gain will increase.

When the input falls below the threshold, pulses on IC3d pin will cease and pin 11 of IC5d will go low after a period determined by RV4. The output of IC3c will also go low and C8 will discharge through R13 and D1. C7 will also discharge but at a rate determined by the setting of RV1. This falling voltage will reduce the current into pin 5 of IC2 and hence the gain of the audio path will fall.

Most of the circuit operates directly from the 0V and +9V supply, but some parts of it require a centre tap to provide something approaching dual-rail operation. This intermediate voltage is provided by ZD1 and Q1.

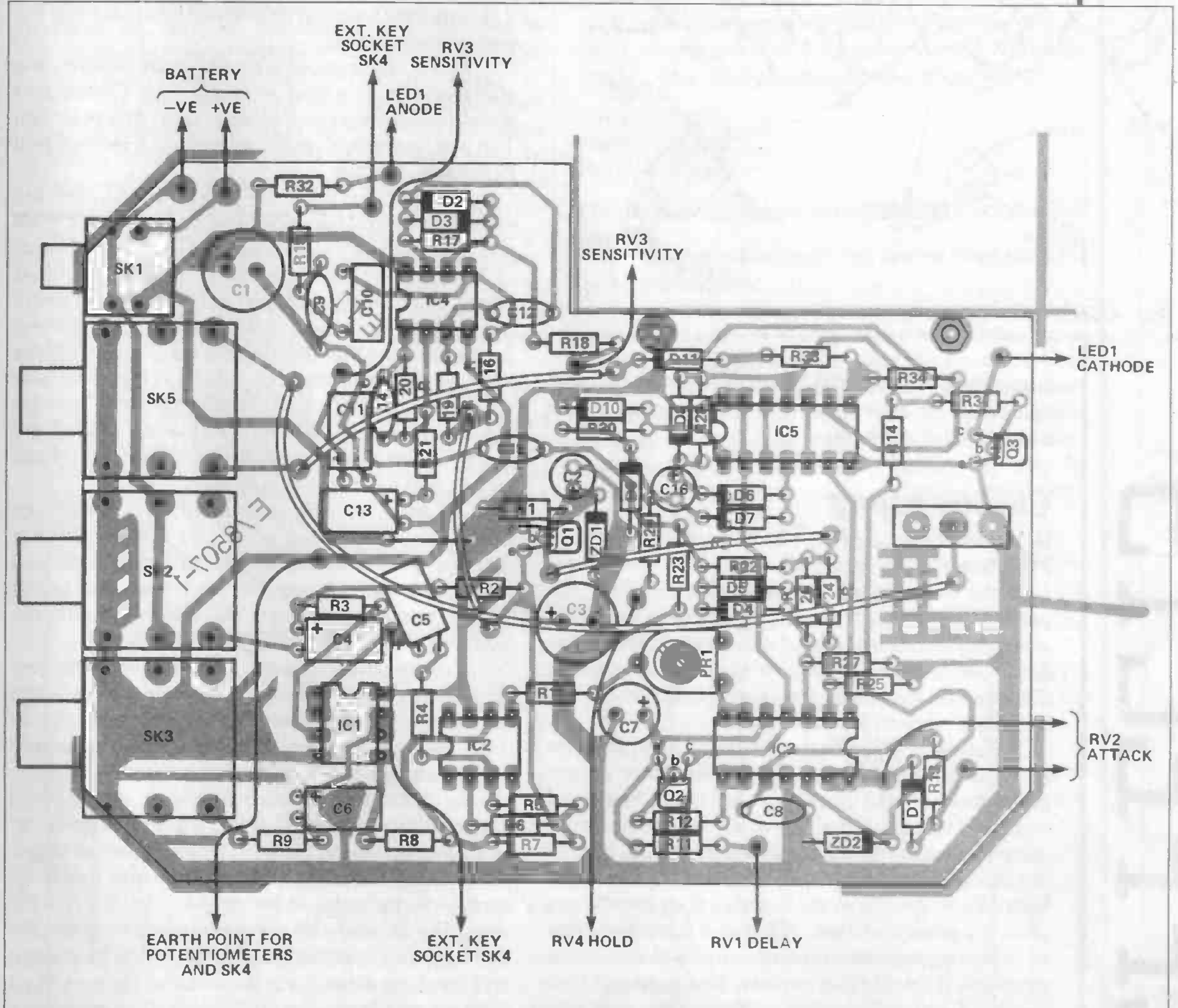
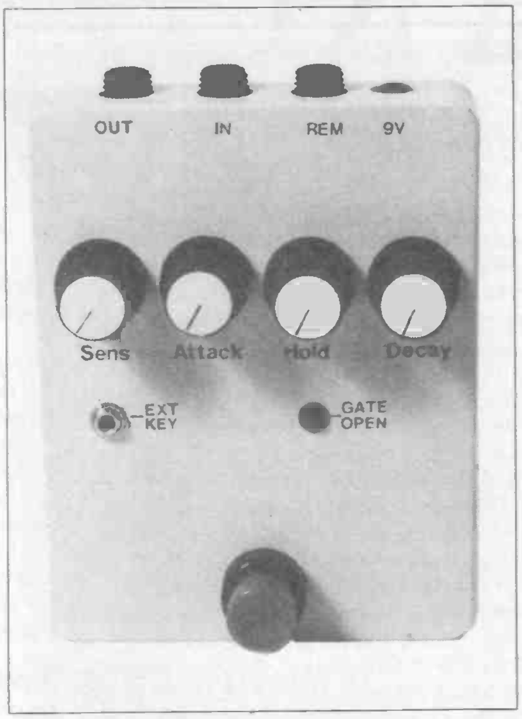


Fig. 3 The component overlay for the noise gate



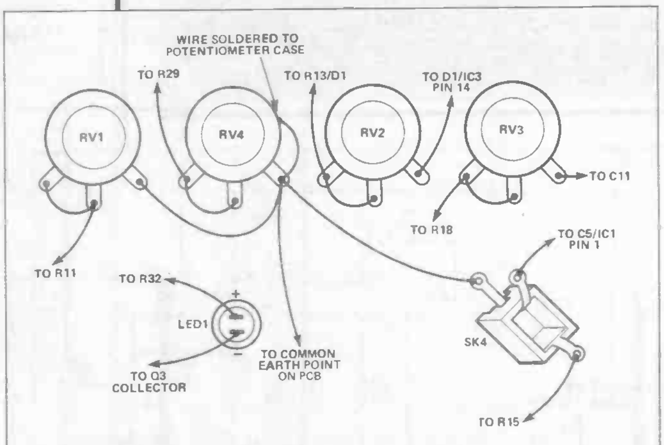


Fig. 4 Connection details for the front panel

completely adjustable and the unit can be controlled by a variety of inputs, it can be used as an envelope shaper in its own right.

Construction

Before soldering anything into place, check that your PCB (see Fig. 3) has a hole under PR1 and, if not, carefully drill a ¼in hole there. This will allow the preset to be adjusted from the underside of the board when the unit is assembled into its case. When the bare board is ready, commence assembly by installing the wire link, the four jack sockets (see *Buylines)* and if desired, sockets for ICs 3 and 5.

Continue assembly by soldering into place the resistors and capacitors, making sure that all the capacitors near the connector end of the board are mounted flat so as to make room for the potentiometers when the board is installed in its case. Next fit the diodes, transistors and ICs 1,2 and 4 which must be soldered directly to the board or they too will not clear the potentiometers. ICs 3 and 5 are well clear of the potentiometer positions and will not cause problems if fitted using sockets. Cut to length four pieces of ordinary insulated connecting wire and solder them between the points shown on the PCB overlay, then fit the two battery guide pillars and the PCB is complete.

The case is drilled and prepared as for the chorus unit (see Fig. 4 of that article — note the additional hole).

It is important to use the recommended potentiometers, switch and EXT. KEY socket or difficulty may be encountered in getting everything to fit within the space available. Mount the LED, the socket and the potentiometers through their respective holes in the front panel and connect them up to the PCB, taking care not to use greater lengths of wiring than is necessary. Solder the battery connector leads to the board and place one fibre washer on each of the three larger jack sockets. Mount the switch in the front panel but do not tighten it up.

Offer the PCB up to the case, guiding the jack sockets into their holes and aligning the switch pins with the pads provided. A little bit of force may be necessary but any serious opposition should be investigated lest anything be damaged. When the PCB has settled into place, solder the switch pins onto their pads, tighten the switch mounting from the front panel and secure the large jack sockets with the nuts provided. Construction is then complete.

Setting Up And Use

Connect up a 9V battery, switch on and apply a signal of about 2V peak-to-peak to the input. The LED should light up. Monitor the output with an oscilloscope or an AC millivoltmeter and adjust PR1 until the output level is of the same amplitude as the input level. This is the only adjustment necessary and if all is well the base can be screwed into place and the unit is ready for use.

In use, the noise-gate should come between any effects and the amplifier or mixing desk. Connection should be by a screened cable as short as is practical. The unit is switched on by connecting a (mono) jack to the input socket.

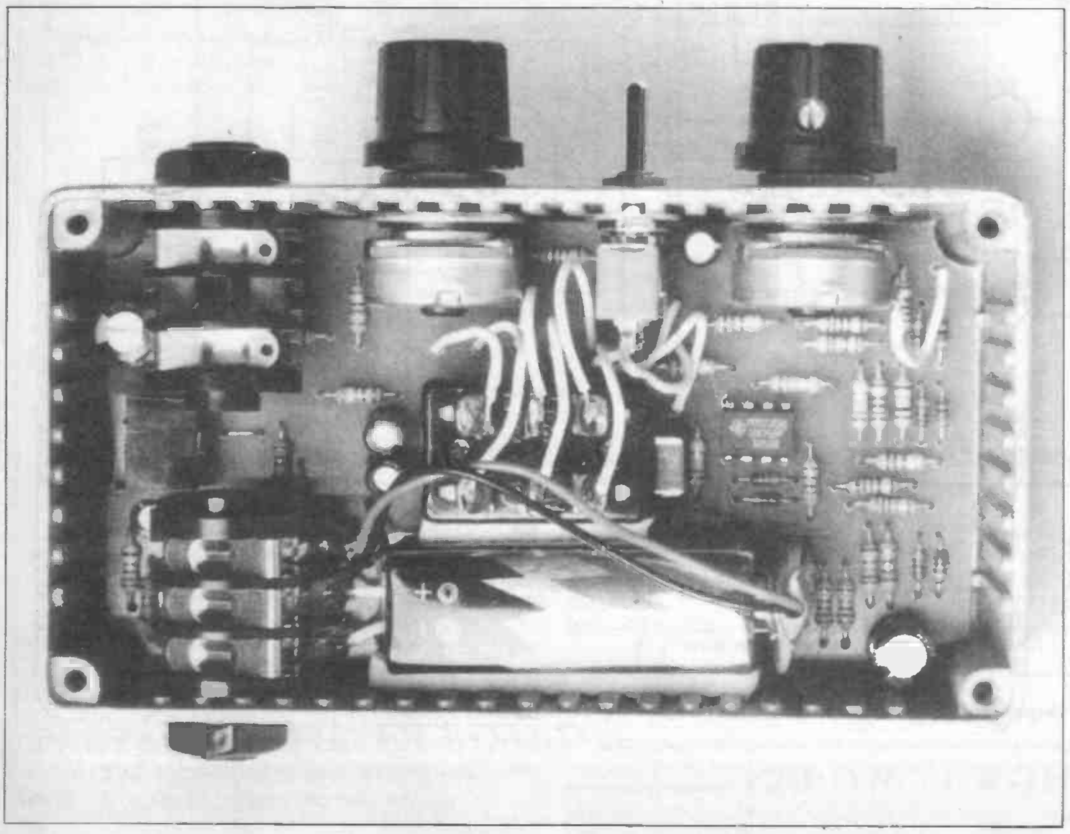
When setting the noise-gate up initially, turn the sensitivity control fully clockwise and the attack, hold and decay fully anticlockwise. The LED should be off (if it isn't, press the footswitch). If using any other effects, switch them on to produce all the noise you're trying to get rid of and rotate the threshold control anticlockwise (thus lowering the threshold) until the LED lights (at this stage you should be able to hear the noise getting through to your amplifier). Turn the threshold control slightly clockwise, raising the threshold just above the noise-floor. The LED should go off, and the noise should stop.

As you play your instrument, the gate should open, and should close when you stop. Remember that the other controls are still at a minimum and should now be set to suit. Normally the attack will be left at a minimum, giving a short rise-time, with the hold and decay at about a second or so.

Pressing the footswitch will open the noise-gate regardless of input level, this is very useful when tuning-up. A remote footswitch can be connected to the REM socket, disconnecting the unit's own switch.

The noise-gate can also be used as an envelope shaper with the attack-hold-decay cycle being triggered in a number of ways. An audio signal can be connected via the EXT. KEY socket and will trigger the envelope shaper but still allow the threshold control to be used. Alternatively, the EXT. KEY should be shorted with a miniature jack plug and the unit triggered from the REM socket either by making and breaking a mechanical contact or by applying a logic signal. Closing the REM contacts or applying a 0V level will close the gate while opening the contacts or applying a +5 to +15V signal will open it.

HYPER-FUZZ



Not another fuzz box! Well, not just another fuzz-box. This is the Hyper-fuzz

Ever since the 60s, the fuzzbox has been the most used guitar effect of them all, thickening and warming single note solo runs and blasting the listeners with the powerful crash of classic power chords (do I hear *Smoke on the Water?*).

The distortion pedal or fuzzbox makes the sound of a musical instrument more 'interesting' by adding harmonics to it — usually achieved by using a non-linear amplifier of some kind to clip or round-off the peaks and troughs of the audio signal. The resultant distortion consists of a wide range of harmonics at multiples of the input frequency.

The Hyper-fuzz is a new type of distortion effect. The input/output characteristic is shown in Fig 1. Each half-cycle of the audio signal is 'folded over' three times before being clipped. This gives rise to a narrow band of harmonics, the frequency of which is dependent on the amplitude of the input signal. When used with a guitar, a filter-sweep effect is produced as each note dies away.

In addition, the circuit can produce conventional clipping distortion and an intermediate effect, selected by a three position toggle switch, SW1. These extra characteristics are also shown in Fig. 1.

The circuit board is mounted in a small diecast box fitted with a foot switch (SW2) which is used to switch between the effect and a 'straight through' signal. A metal box was chosen in preference to a ready-made foot switch case because of its lower cost and better screening properties.

The other controls on the unit are *depth* (RV1) which varies the severity of the distortion, and *level* (RV2) which is used to match the distorted and straight-through signals in volume.

Power for the effect comes from a PP3 battery or from an external 9V supply, the current consumption being only about 2mA. The internal battery is only connected when a jack is plugged into the input socket, so there is no need for an on/off switch.

Construction

If the recommended case is used, it should be drilled as accurately as possible (Fig. 3). The positioning of the holes is fairly critical and constructing the PCB first will demonstrate how far each pot will protrude from its drilled hole.

The component overlay is shown in Fig. 4. Assembly should cause few problems if you start with the simplest components and work up. Everything should sit as close to the board as possible and leads should be cropped short.

When the two pots are mounted, their spindle centres should be about 15mm above the top of the PCB.

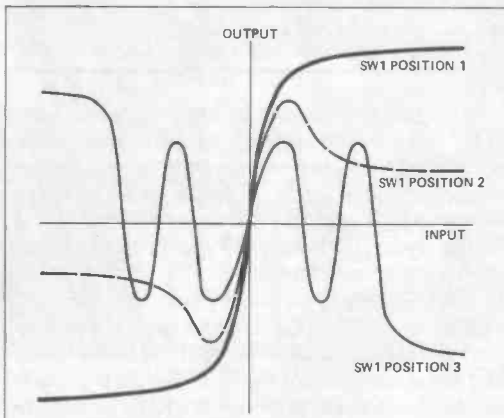


Fig. 1 The three selectable input/output characteristics of the Hyper-Fuzz

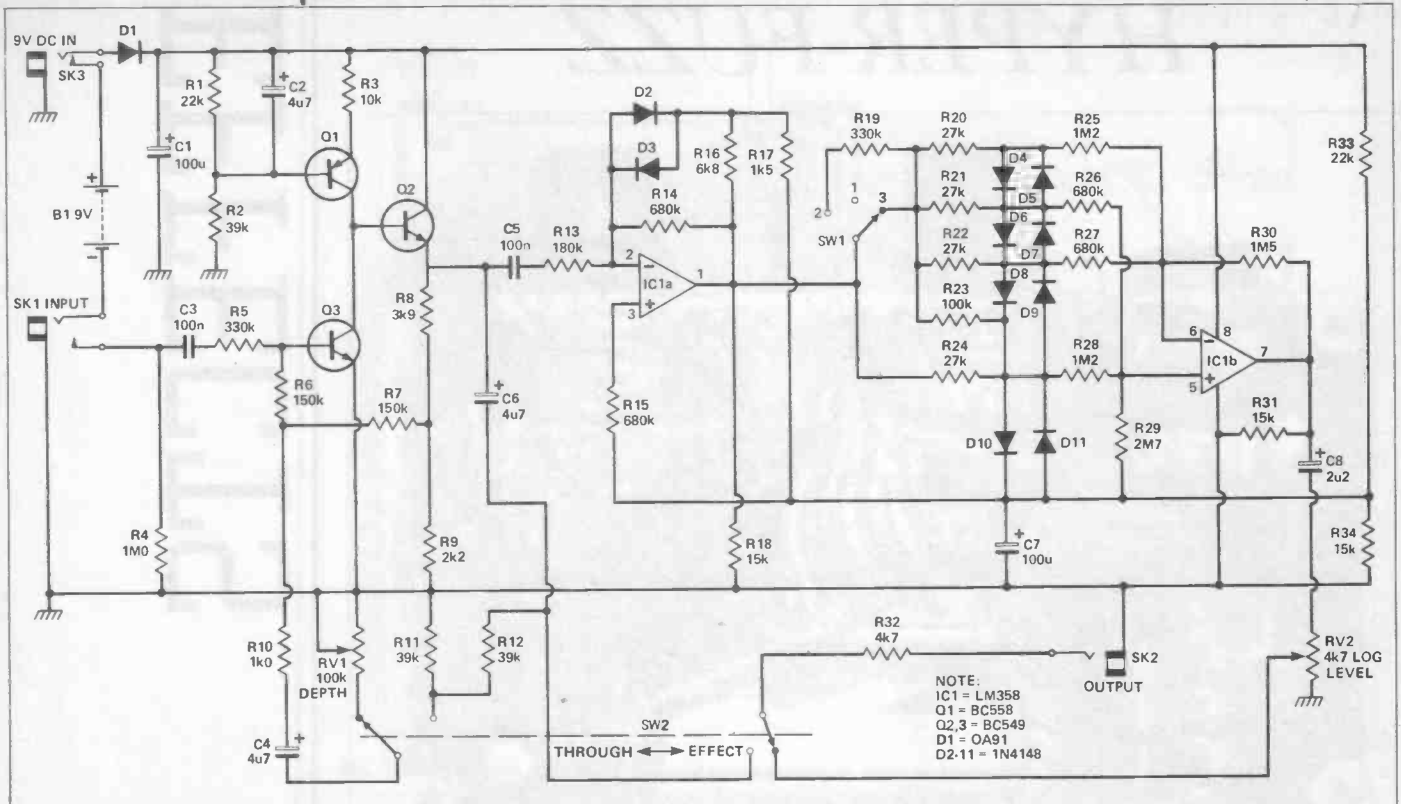


Fig. 2 The circuit diagram of the Hyper-Fuzz

HOW IT WORKS

The input signal is fed to a pre-amplifier comprising Q3 which provides the gain, a constant current source (Q1) and an emitter-follower (Q2) to buffer the output. With SW2 in the 'Through' position, the gain of the pre-amp is set at about one by negative feedback through R10 and R6. Its output goes to the output socket (SK2) via C6.

With SW2 switched to 'Effect', RV1 is used to vary the amount of feedback and hence the gain of the pre-amp.

The distortion-generating part of the circuit uses an LM358 dual op-amp, which was chosen because of its low current consumption and wide output voltage range. R33/34 and C7 provide a stable 3.5V mid-rail for the op-amps. The signal from the pre-amp is further amplified by IC1a. The op-amp is prevented from clipping by D2 and D3, which limit its output to about 6V peak to peak.

With SW1 in position 3, the output of IC1a will drive the four pairs of diodes, D4-11, to produce four waveforms clipped at ± 0.5, 1,1.5 and 2V. These are then fed to alternate inputs of a difference amplifier (IC1b). So as each diode begins to conduct, the gain of the circuit reverses polarity.

With SW1 in position 1, only the lower pair of diodes is driven so the circuit produces 'ordinary' fuzz. For the intermediate effect, R19 is used to attenuate the signal reaching the upper diodes. R23 is necessary to match the three effects in volume.

If the jack sockets have break contacts fitted, the tags for these should either be cut off or bent under the socket. The sockets should then be attached to the board by 15mm lengths of flexible wire. The toggle switch and RV2 case can then be connected to the board along with the battery clip and six 45mm pieces of wire for the foot switch.

The PCB spacer is a piece of 4mm thick perspex, wood or paxolin 22 × 26mm, with a ½in central hole. To make assembly easier the spacer can be lightly glued to the track side of the PCB. Before the unit is assembled an insulating grommet should be fitted to the hole for the power connector.

The PCB assembly can then be inserted into the drilled case. To do this, the output jack and the two pots should first be located in their respective holes (the pots may need to be bent back a little).

Once the pots are pushed through, the board should fit neatly inside the case and the other socket and toggle switch can then be fitted. The foot switch should be fitted through the holes in the board, spacer and case so that it holds them together, and its leads soldered as shown on the component overlay.

If you intend to drill the base plate to take screw-mounting feet, make sure that the screws will not interfere with the jack sockets. With care they can be positioned so that the feet slightly overlap the retaining screws, thus preventing them getting lost. The battery can be cushioned using strips of draught excluder stuck inside the case and on the side of the switch, and held in place by a piece of foam rubber glued to the base plate.

Operation

The unit should be set up in the same way as a standard distortion unit. Adjust RV1 and SW1 until the desired effect is heard then adjust RV2 so that there is little change in volume when the foot switch is pressed. Because of the severity of the distortion, the full Hyper-fuzz effect (Setting 3) works best with simple 'pure' signals in solo work. Playing chords produces harsh ring-modulator-like effects which are interesting but not exactly musical. Although the unit was designed for use with electric guitar and bass, it can also be used for experimenting with the sound of keyboard instruments, drum synthesisers and even vocals.

BUYLINES

All the components are available easily from any supplier. The case used in the prototype (Type 5004, 120×65×40mm) is available from Maplin as part LH71N.

The PCB is available from our PCB Service. See the back of this magazine for details.

PARTS LIST

RESISTORS (all ¼W 5%)	
R1,33	22k
R2,22,12	39k
R3	10k
R4	1M0
R5,19	330k
R6,7	150k
R8	3k9
R9	2k2
R10	1k0
R13	180k
R14,15,26,27	680k
R16	6k8
R17	1k5
R18,31,34	15k
R20,21,22,24	27k
R23	100k
R25,28,30	1M5
R29	2M7
R32	4k7
RV1	100k lin PCB mounting
RV2	4k7 log PCB mounting
CAPACITORS	
C1,7	100µ 16V radial electrolytic
C2,4,6	4µ7 63V radial electrolytic
C3,5	100n polyester
C8	2µ2 63V radial electrolytic
SEMICONDUCTORS	
IC1	LM358
Q1	BC558
Q2,3	BC549
D1	OA91
D2-11	1N4148
MISCELLANEOUS	
SK1	¼in stereo socket
SK2	¼in mono socket
SK3	2.1mm power socket (PCB mounting)
SW1	SPDT centre-off toggle
SW2	DPDT foot switch

PCB. Case. Battery clip. Knobs. Feet. PCB spacer. ⅜in grommet. Foam rubber. M10 washers.

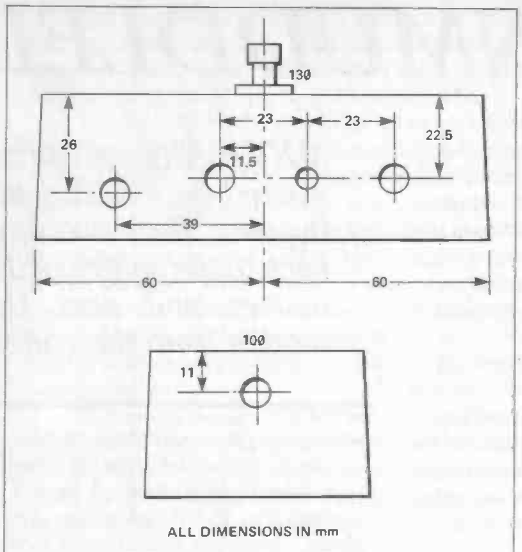


Fig. 3 Drilling the case for the Hyper-Fuzz

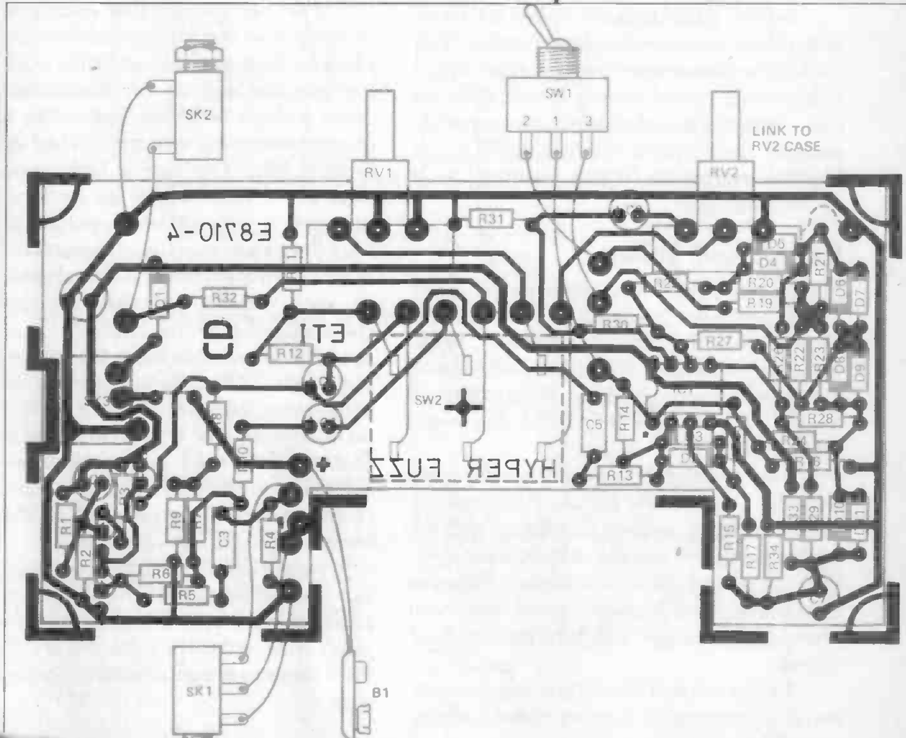


Fig. 4 The component overlay

UNMUDDLED MIDI

MIDI facilities are often wasted by mistake and misuse. Here we look back to the source of the matter and sort the modes from the codes

Back in the days of monophonic analogue synths — the Wasps and the Cats of old — linking could be achieved using just two control signals — a gate voltage which indicated *note on* and *note off* and a control voltage which varied with the pitch of the note.

The development of polyphonic synthesisers made the gate/CV system (as it is known) rather cumbersome very quickly — a separate pair of control wires is needed for each voice. More than one keyboard player ended up taking small telephone exchange systems on the road to recreate their studio sounds.

Refinements such as touch-sensitivity and pitchbending created too much information for the gate/CV method to handle.

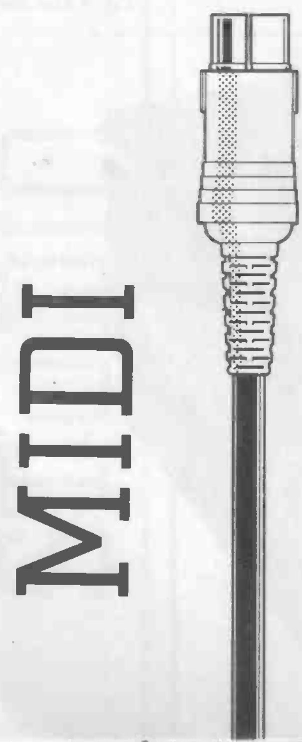
So the music manufacturers set about designing a new communication method that could cope with new technology. After an uncomfortable period during which different manufacturers launched totally incompatible systems, the industry settled with MIDI, the Musical Instrument Digital Interface, as a general standard.

East West Talks

MIDI came out of discussions in 1981 and 1982 between Sequential Circuits and Oberheim in the US, and Yamaha, Korg, Kawai and Roland in Japan. Its existance was first announced publicly in Robert Moog's column in the October 1982 *Keyboard* magazine.

MIDI instructions are capable of far more than the old gate/CV signals. Each *note on* and *note off* instruction is transmitted with the pitch information and the velocity with which the key was pressed or released. Pressure sensitive keyboards can also send after touch data — the force with which the keys are held down.

Keyboards that are not touch or pressure sensitive transmit a dummy value halfway between the two possible extremes.



Each block of information is headed by a channel code numbered from 1 to 16 (actually 0 to 15 but conventions are funny things). Receiving instruments can be set to react only to certain channels so that a single serial stream of data can trigger up to 16 different voices, keyboards or whatever.

It is this seriality that overcomes the multiple links and wiring made necessary by polyphonic keyboards and their ever more sophisticated features. All information is sent down a single MIDI link, not unlike RS232 computer links but with a much higher transmission rate. The rate is high enough to support 16 sets of note instructions to be transmitted without any appreciable delay from first to last, even when accompanied by other information that MIDI can handle such as pitch bending, modulation, program changes and real time clocks.

The serial link is made through a standard 5-pin 180° DIN connector, although some manufacturers have put XLR sockets on equipment that is likely to spend most of its time on the road. XLR sockets are more rugged than DIN (and more expensive of course) and it is easy enough to make up leads to connect between them.

Note that DIN leads from your hi-fi won't run your MIDI system - most such leads are cross coupled with the left pins of one DIN plug connected to the right pins of the other. MIDI requires straight correlation between the pins.

Gonna Make You A Star

If your master keyboard is using the different MIDI channels to instruct more than one extra synth, its MIDI output needs to be split so that each synth receives the information.

The most obvious way to achieve this is to use the output as a bus, with each keyboard attached in a 'star' formation. The problem here is that the current supplied by the master keyboard (about 5mA) is split between the slave synths and could well fail to drive any of them properly. The arrangement will usually drive two peripherals without problems but any more than that and it all goes horribly wrong.

Hence the existance of three MIDI sockets: IN, OUT and THRU. The simplest arrangement is shown in Fig. 1, where two synths are cross coupled so that playing either synth will trigger both.

It is the THRU socket that avoids the need for 'star' connections. The THRU socket produces a buffered replica of whatever is received at its IN socket, so that when a system like that in Fig. 2 is constructed, each slave synth receives the complete output from the master keyboard at full power — powered by the previous synth in the chain. Rather sensibly, this setup is called a chain connection.

There is another type of star connection where the master instrument — more likely a computer or sequencer here — is equipped with a number of MIDI IN and OUT sockets, each separately powered. In this case of course there is no problem running a star system like that shown in Fig. 3. You could add chains onto each star output and run an enormously complex system of peripherals all based on your one master instrument. The system is limited only by the size of the garage in which it all has to sit.

A quick note to point out is that although Fig. 2 shows nothing but synths in the chain they can of course be drum machines, effects units — anything with MIDI facilities — all controlled by a MIDI guitar if you so wish.

Hardware

If the hardware bores you stiff, hang around because we'll be getting to the software soon.

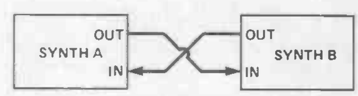


Fig. 1 Cross coupled MIDI setup

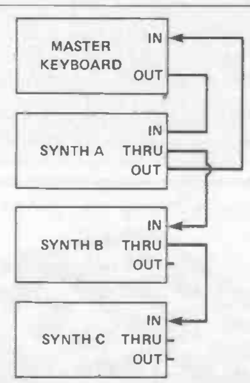


Fig. 2 The MIDI chain connection

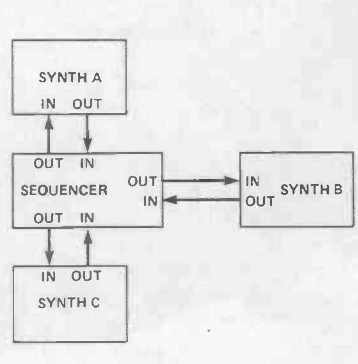
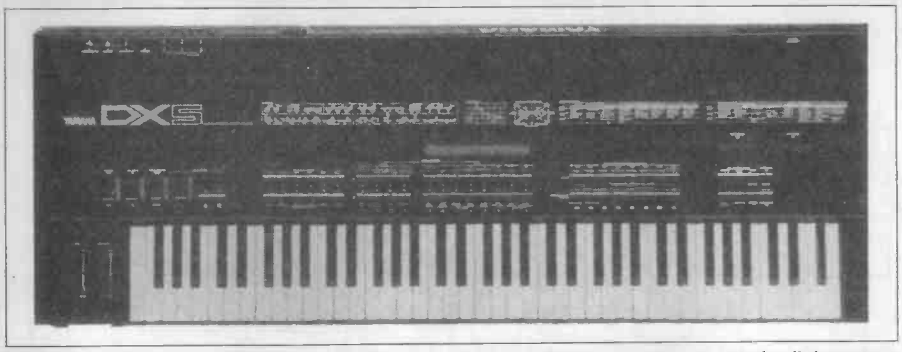
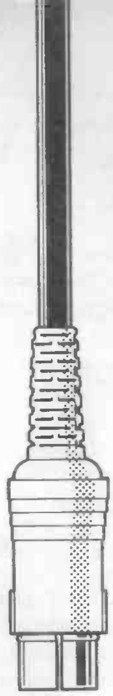


Fig. 3 A multi-socket star connection

MIDI



Yamaha's revolutionary DX range of FM synthesisers were instrumental in standardising many MIDI controllers



The MIDI interface uses the same sort of serial interface chips as RS232 systems (known as Universal Asynchronous Receiver/Transmitters — UARTs — or alternatively Asynchronous Communication Interface Adapters — ACIAs — but we shall stick to calling them UARTs).

The highest transmission rate of RS232 is 19200 bits per second but MIDI operates at the even higher rate of 31250 bits per second — or one bit every 32μs. UART chips can handle this higher bit rate without difficulty. The choice of MIDI baud rate can make the hardware implementation particularly easy because UARTs operate from a clock running at 16 times the baud rate — 500kHz in this case. This can often be derived from the microprocessor clock by a simple divider.

MIDI uses a fixed format of 8-bit words with one stop bit and no parity. Each byte transmitted over the interface is in the form of ten successive bits, eight of which are the data. When the transmitter is idle, the UART outputs and inputs are in 'mark' state with a continuous '1' or high output.

The first bit of any transmitted byte is the 'start' which signals the beginning of the data proper. Its falling edge is used by the receiver UART as a reference to time the reception of the subsequent bits. After the start bit, eight data bits follow with the least significant bit first.

After the last data bit a '1' stop bit follows. If another byte follows immediately this stop bit will be followed by the start bit of the next byte, otherwise the interface will return to the mark state, waiting for the next start bit.

Contemporary UARTs

There are a number of UARTs which can be used, depending on the application. Readily available types include the 6402, particularly suitable for MIDI circuits not using microprocessors, the 6502-series 6551, the 6800-series 6850 and the 8080/Z80-compatible 8251. All of these four devices will both transmit and receive serial data, so only one is needed for each MIDI IN/OUT combination. There will usually be just one of these per instrument. Microcomputer chips often have UARTs built into them and an external device is not needed.

Figure 4 is a simplified diagram of typical MIDI hardware. It is not essential that the transmitting and receiving equipment is microprocessor-based but this is almost always the case in practice so the UARTS are shown connected to microprocessor buses.

The serial output from the transmitter UART passes via a buffer to the MIDI OUT socket, which is connected to the MIDI IN socket of the receiver. To avoid hum loops and prevent digital noise getting into audio circuits, an opto-isolator is used at the receiving end between the MIDI IN socket and the serial input of the receiver UART.

Grounded By The Light

Another important purpose of the opto-isolator is to block voltage differentials between grounds on different instruments. These differentials are hardly likely to zap the musician using the system but could certainly damage those delicate ICs in the microprocessor circuits — particularly if there is a computer as part of the setup.

Also, the cable is screened but the screen is only grounded at the transmitter end, again to avoid hum loop and digital noise problems. Often the output of the receiver opto-isolator will also be connected to a buffer driving the MIDI THRU socket, which then transmits a copy of the MIDI IN data.

Figure 5 shows the serial data buffers, the interface cable and the opto-isolator circuit in more detail. The interface cable is ordinary screened twin, with 180° 5-pin DIN plugs at each end. Pins 1 and 3 are not connected. The screen is connected as usual to pin 2 and the cores of the cable are connected to pins 4 and 5.

The screen is grounded in the transmitter but left open circuit inside the receiver, while pins 4 and 5 are connected in the transmitter to the driver circuit and in the receiver to the opto-isolator circuit. There is no connection between the opto-isolator input circuits and any other circuits in the receiver.

R1 and R2 ensure that dodgy leads or incorrect connections cannot damage or short circuit the drive or power supply. R3 limits the current through the opto-isolator LED even if an incorrect signal voltage is connected directly to pins 4 and 5 and inverse parallel diode D1 protects it against reverse applied voltages. This arrangement is reasonably idiot proof — although incorrect connections will usually stop it working, they are unlikely to blow anything up.

The diagram shows R3 connected between pin 5 of the MIDI IN socket and the opto-isolator LED cathode but it could equally well be between pin 4 and the LED anode, should this be more convenient from a layout point of vew. What matters is that the two are in series.

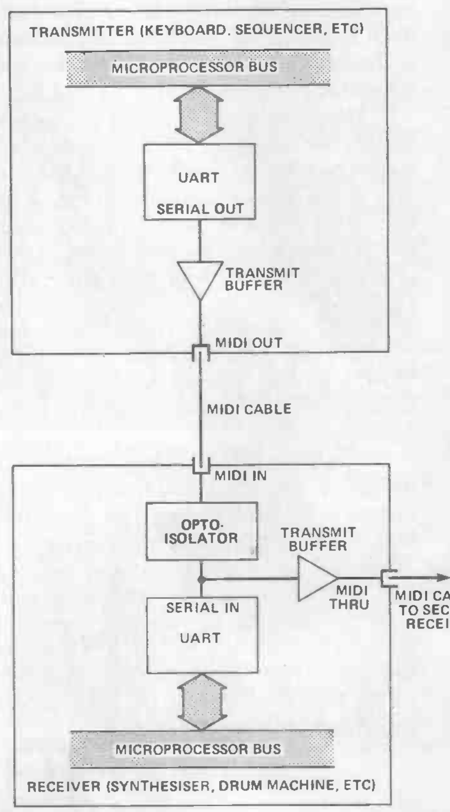


Fig. 4 Block diagram of a typical MIDI system

Buffer To Buffer

When the transmitter is idle, or a 'one' bit is being sent, the UART output is in the high state. This puts the output of the transmit buffer in the high state too, so no current flows in the interface.

When the UART transmits a 'zero' bit, its serial output goes low and so does the output of the transmit buffer. As a result, a current of about 5mA flows through the loop including R1, the interface cable, the opto-isolator input, R3, and R2. This turns on the opto-isolator output transistor, pulling its collector low so that the low output from the transmitter UART is passed on to the receiver UART serial input.

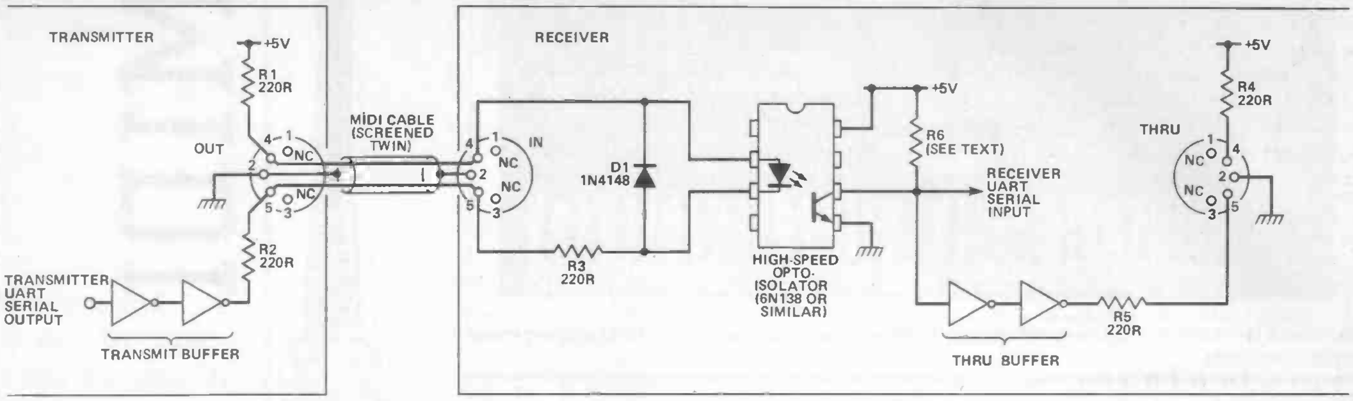
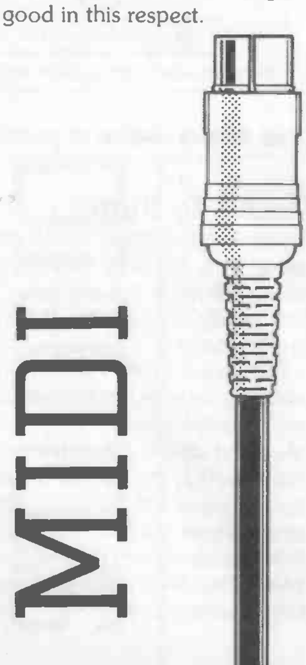


Fig. 5 The major circuit elements of a MIDI transmitter and receiver

When the transmitter UART output returns to the high state, the interface loop current stops, the opto-isolator output turns off, R6 pulls it high and applies a high to the receiver UART input. The output from the opto-isolator will also be connected to another transmit buffer if the receiver has a MIDI THRU socket. This is basically the same as the transmit buffer in the transmitter, with R4 and R5 as equivalents to R1 and R2.

The transmit buffers are needed because UARTs are invariably MOS devices which cannot be relied on to sink the 5mA loop current in the low output state. In Fig. 3 the buffers are drawn as pairs of inverters but in practice almost any logic gate, buffer, or non-inverting combination of gates and buffers capable of sinking the loop current in the low state will do the job. 4000 series CMOS does not have the drive capability but 74LS or 74HC logic does. Discrete transistors can also be used in place of the second inverter to sink the loop current.

One potential problem with MIDI is that the turn on time of the loop current and opto-isolator is generally shorter than the turn off time. Anything that helps speed up the turn-off process can only help, so totem pole outputs are actually a little better than open collector outputs or discrete transistors because they actively pull pin 5 of the interface socket high rather than relying entirely on the loop current to do so. 74HC logic is particularly good in this respect.



MIDI THRU And THRU

There is of course a delay in producing the receiver's MIDI THRU signal from the transmitting UART. There is also distortion or skew. The loop current and opto-isolator generally turn on faster than they turn off, so that the propagation delays of low to high transitions are longer than those of high to low.

This means that in our chain setup of Fig. 2, the data received by keyboards at the end of the chain will have been cumulatively delayed and skewed and the longer the chain the worse the skew will get.

Figure 6 shows an example of what might happen to a byte transmitted by the mater keyboard. The point at which a UART samples the input data is timed from the leading edge (high-to-low) transition of the start bit, to the nearest sixteenth of a bit period (if clocked at 16 times the bit rate). Each bit is sampled at what should be its mid point.

The effect of the difference in propagation delays is to delay the low-to-high transitions relative to the high-to-low transitions and shorten the high pulses. When this relative delay, or skew, becomes nearly half a bit period (16μs) data errors will happen. In our example here each of the individual units would receive data quite happily from the transmitter and any combination of two of them would work as well, but the cumulative skew is enough to prevent the last unit from receiving the data reliably.

There is no specification as to how many units can be connected in series or what an acceptable skew is, though the IMUG MIDI specification does go as far as to say that higher speed opto-isolators should be used to avoid these problems with long chain lengths — more than three instruments. Fortunately it is not difficult to keep skew down to a reasonable level if the opto-isolator circuit is designed properly.

Although MIDI THRU sockets cannot be split in parallel to several synths (for the same reason that a single OUT socket will not run a star system), the problem can be relieved using a THRU box. These take the signal from the transmitting OUT socket and produce a number of separately buffered duplicates. These can each be treated as normal THRU sockets so that a number of peripherals can receive what is effectively a second generation signal.

No maximum and minimum levels are specified for the 5mA MIDI loop current. Allowing for series resistor tolerances (assuming 5% components), variations in power supply voltage and possible driver gate and opto-isolator LED on-state voltage variations, it is reasonable to assume that the loop current will be somewhere between absolute worst case limits of 3.5mA and 6.0mA, provided of course that the transmitter and receiver are properly designed. In practice the current may be higher because not all transmitter circuits include both series resistors. For example, there are some designs in which pin 4 of the MIDI OUT socket is connected directly to +5V, omitting the series resistor altogether.

A Little Ohm Work

Returning to Fig. 5 the value of R6 must obviously be decided. Its minimum value can be determined by assuming a combination of the lowest loop current and lowest current transfer ratio (CTR) of the opto-isolator. The 6N138 is a high speed Darlington opto-isolator often used for MIDI interfaces and has a minimum CTR of at least 300% at the kind of LED current we are talking about. This means that it will be able to sink at least 3× 3.5mA (or 10.5mA) through R6 in the on state. Provided R6 is 470R or greater, the output is guaranteed to pull low. To keep turn-off time down to a minimum, R6 should be as small as possible, so a value of 470R is a good choice for use with this opto-isolator. Typical skews are about 2μs.

The 6N136 can also be used and is a little faster, if used with care. Because it does not have a Darlington output stage, it has a lower CTR of 19% minimum, so a comparatively high 6k8 pull-up resistor should be used.

Vince Clarke with BBC Micro and MIDI interface

Its speed is sensitive to stray capacitance on the base pin (pin 7), especially between it and the collector pin. This is not a problem on PCB layouts but should be borne in mind for any circuit built on stripboard — the base and collector will be on adjacent strips unless the strip to pin 7 is cut close to the pin. Also, stray capacitance on the collector pin should be kept low so as not to compromise the rise time. The low-state input current of any logic circuits connected to the opto-isolator output have not been taken into account in the calculations of R6 above but should be if LS TTL is involved.

Saturation Point

The CNY17 is also sometimes specified for the job of opto-isolator. It has a minimum current transfer ratio of 40%, so a calculation similar to the one above would indicate that it is guaranteed to sink 1.4mA through R6 and that R6 should therefore be about 3k9. Unfortunately the CNY17 is not particularly fast and gives poor performance when used this way. One reason for this is that the transistor saturates in the 'on' state, and takes time to recover.

Provided saturation can be avoided, the turn on and turn off times of the CNY17 are about equal.

In theory it would be possible to avoid saturation by selecting R6 so that the output falls to say half a volt in the low state. This is not really a practical solution though, because the loop current cannot be relied on to be constant (for example, it could change by connecting to a different transmitter). Also the CTR is temperature dependent and slowly changes as the opto-isolator ages. This is not a good way to avoid saturation and should be avoided.

Another speed problem is that the collector/emitter voltage changes as the transistor switches on and off, which slows down the switching process due to Miller effect.

To get the best performance from the CNY17, it should be operated as a non-saturating current switch with a relative constant collector/emitter voltage. In practice this is not difficult and Fig. 7 shows how. The opto-isolator transistor switches current into the base of the external transistor Q1, which does the job of pulling the UART input low when the loop current is on. R7 is calculated so that it takes nearly all of the worst case minimum opto-isolator transistor current with Q1 base-emitter voltage across it, the balance flowing into Q1 base and turning it on. The minimum opto-isolator current is 1.4mA with the CNY17-1, or 2.2mA with the CNY17-2, which has a higher minimum CTR of 63%. Q1 can be almost any general purpose signal transistor — something like a BC107 or BC182 does admirably.

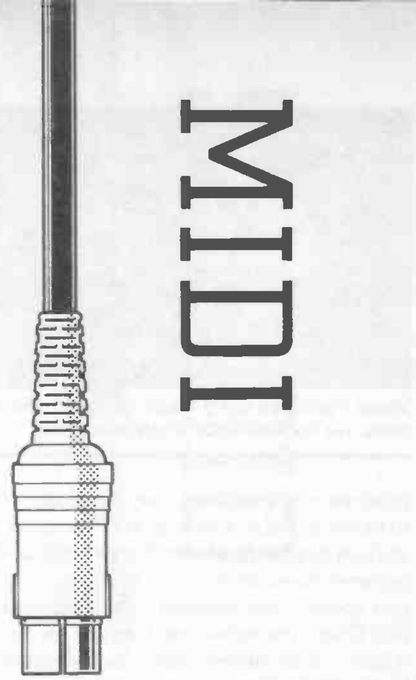
The Software

Enough about the hardware. If you are a user rather than a builder, you'll probably be more interested in the information needed to drive peripherals — the MIDI software. In fact, you may have skipped the last page or so and come straight here. If you did then have a glance back and make sure you understand the limitations that propagation delays impose on chain networks — otherwise you may run some accurate software which fails through hardware linking.

Another limitation should also be mentioned here, this time down to the individual manufacturers rather than the standard itself. Although MIDI sets down how every sound parameter is controlled, this doesn't mean that all MIDI keyboards are capable of executing or receiving all MIDI commands. For example, early MIDI equipment commonly is unable to recognise pitch bend information and will simply ignore it.

Instruction manuals should detail what an instrument can and cannot do. Sometimes it is possible for keyboards to act upon incoming MIDI instructions to do things it can not do alone — Yamaha DX keyboards without touch sensitivity may react to touch sensitivity data through its MIDI socket. It has the chips but not the keyboard.

Hence two instruments can only communicate at their lowest common level of implementation. This will invariably include all the basics but not necessarily the more subtle features.



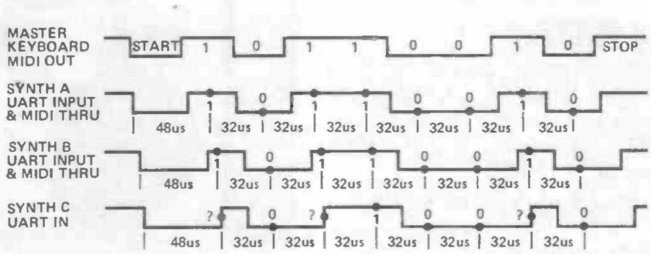


Fig. 6 Timing diagram of a single MIDI byte through a chain

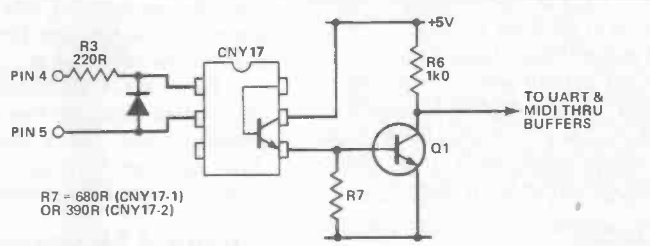


Fig. 7 CNY17 opto-isolator arranged as a non-saturated current switch for optimum performance

Modes

Don't confuse modes with codes. MIDI codes are the individual instructions for *note on* and so forth. MIDI modes describe the way these codes will be received — which channels the receiver will monitor for instructions and so on.

There are four MIDI modes.

Mode 1, often called **OMNI ON/POLY** mode. With your receiving instrument in Mode 1, it will respond to MIDI information on all 16 channels (OMNI ON), performing polyphonically using its own internal voice assignment algorithm. This is the basic operating mode where one synth will drive another — most equipment will go to Mode 1 at switch-on by default.

Mode 2, often called **OMNI ON/MONO** mode. In mode 2 the receiving instrument still responds to all 16 channels (OMNI ON) but will assign everything it gets to a single voice. This is fine for a monophonic instrument but makes all but one of a polyphonic keyboard's voices completely redundant. It also gets easily confused trying to assign too much information to a single voice. Consequently it doesn't get much use in most MIDI systems.

Mode 3, often called **OMNI OFF/POLY** mode. Here the receiving instrument listens to a single MIDI channel and ignores everything else. Polyphonic instructions on that channel are assigned using the instrument's own internal assignment algorithm. The specific channel in question is called the basic channel or sometimes base channel and is usually set from the instrument's own controls.

Mode 4, also called **OMNI OFF, MONO** mode. This is the mode you need to grasp to make the best of most MIDI setups. In Mode 4 the receiving instrument listens to a defined number of channels and assigns one voice to each channel. Actually, if you define zero as the number of channels to be monitored, the

Vince again, blowing Casio's DH100 MIDI horn, yet another MIDI implementation

receiving instrument will normally assign itself to as many MIDI channels as it has voices. It starts at its specified basic channel and works up from there. With some instruments you can specify any channel combination you like. Check the instruction manuals. In Mode 4 then, a multi-timbral synth can be instructed to play eight different sounds at the same time — a wonderfully versatile system.

Sending Modes

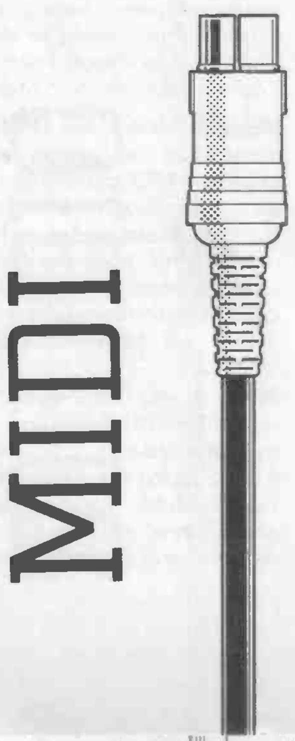
Things can get confusing if you start switching between modes regularly. Changing the receiving mode also alters the way a keyboard transmits MIDI information (just the MIDI OUT, not the MIDI THRU).

In modes 1 and 3, all information is transmitted on the basic channel — and mode 2 differs simply in that only monophonic messages will be sent. In Mode 4, different voice messages are sent on channels selected in the same way as for reception.

Note that not all instruments are capable of operating in all these modes. Some are assigned to fixed basic channel numbers the user has no control over. On power up, instruments usually default to OMNI on, POLY mode. It is also normal for an instrument to initially assign itself to basic channel 1 until told otherwise by its operator.

The Codes

Having looked at the operating modes we can now move on to the instructions. Two types of bytes are transmitted — status bytes and data bytes. Status bytes indicate the type of instruction coming up, whereas the data bytes contain the data itself.



The two types are immediately distinguishable by the most significant bit (bit 7) which is 1 in status bytes and 0 in data bytes. Sometimes the data uses more than seven bits and is transmitted as a pair of bytes (low byte first) — a total of 14 bits of data.

Information transmitted is sent in batches of bytes called messages. Each message starts with a status byte (with bit 7 set) indicating the type of data about to be transmitted, and then a number of data bytes containing the data itself.

However, there is one complication. To speed up transmission there is a feature called *running status*. Basically, this means a status byte only need be transmitted when it differs from the one at the start of the last message. If a receiver receives data bytes which are not

Casio's DZ-1 MIDI drum module

preceded by a status byte, it assumes the status is the same as last time and the data is more of the same.

For example, the transmitter might send information requiring that a chord of three notes be played. This is done by transmitting three *note on* messages, the status byte omitted for the second and third messages.

Channel Messages

There are two basic types of message — channel messages and system messages. Forget about system messages for now. Channel messages relate to instrument sounds or voices. They are so called because each message is sent to the right receiving instrument by being allocated to the correct channel, usually the receiver's basic channel (unless you're in Mode 4 — remember?).

Watch out for those channel numbers incidentally. In everyday use we call them MIDI channels 1 to 16 but down at coding level they are binary values 0 to 15. The binary number transmitted is always one less than the user's idea of the channel number - another trap for the unwary.

The channel number is contained in the least significant four bits of the status byte at the start of a channel message. Receivers must examine these four bits to decide whether or not a message is addressed to them, and ignore it if not.

Channel messages are further divided into channel *voice* messages (which actually play something) and channel *mode* messages (which alter the aforementioned MIDI modes).

Channel Voice Messages

Channel voice messages are obviously the more frequent. These allow one piece of MIDI equipment (typically a keyboard or sequencer) to make another piece of equipment (typically a synth) to play something. This includes turning notes on and off, but also gives control of pitch bend and modulation and many other aspects of the sounds produced as well.

Table 1 summarises the channel voice messages. The last four bits of the status byte form the channel number (shown as nnnn) and the most significant bit is always 1 to distinguish it from data bytes. This leaves bits 4, 5, and 6 to show the actual function of the status byte. Of the eight possible values only seven are used (Table 1). The last — Fn — is used for those system messages you're forgetting about for the moment.

Status messages 8n, 9n and An give information relating to individual keys on the keyboard. The first data byte transmitted in all three cases is the MIDI key number. The key numbers range from 0 to 127 so that in theory MIDI can handle keyboards of up to 128 notes, a span of well over ten octaves.

Middle C is defined to be key number 60 (or 3C in hex), so that key number 0 is the C five octaves below middle C, which at concert pitch has a fundamental frequency of about 8.2Hz. Needless to say it is not used too often! Key number 127 is the G five and a half octaves above middle C, with a frequency of 12.5kHz at concert pitch. That is not used often either.

Note On/Off

Status message 9n is the *note on* message generated when a key is pressed. The first byte transmitted is the key number. The second byte is the key velocity — how hard

Status Byte Binary	Hex	Description	Date Bytes
1000nnnn	8n	Note off	1. key number 2. release velocity
1001nnnn	9n	Note on	1. key number 2. Key velocity
1010nnnn	An	Polyphonic Key pressure	1. key number 2. key pressure
1011nnnn	Bn	control change (see tables 2-4)	1. control number 2. control value
1100nnnn	Cn	Program change	1. program number
1101nnnn	Dn	Channel pressure	1. channel pressure
1110nnnn	En	Pitch bender	1. pitch bend (LSB) 2. pitch bend (MSB)

Table 1 Channel voice messages

it was struck. Many cheaper keyboards are not velocity sensitive and these transmit a halfway value of 64 (40 hex). The exact relationship between volume and velocity is left to each manufacturer to decide.

The opposite of the 9n *note on* message is the 8n *note off* message, generated when a key is released. This is very similar to *note on* message in that the first data byte transmitted is the key number. The second data byte is the release velocity. Again the release velocity in the *note off* message is not precisely defined and keyboards without velocity sensing transmit the median value of 64 (40 hex). Every *note on* message must be followed eventually by a corresponding *note off* message.

There is another potential trap for the unwary with *note on* and *note off* messages. A *note on* message with a key velocity of zero is defined to be a *note off* event instead. In this case the receiver has to assume that the release velocity is median value of 64 (40 hex), since this cannot be given. This special case allows *note on* and *note off* information to be transmitted in the running status mode without re-issuing the status byte, which can sometimes speed things up a bit. Some keyboards will always transmit *note off* messages this way though, so beware!

Press For Action

The third kind of individual key related message is an An *polyphonic key pressure* or *after touch* message. Again the first data byte sent is the key number. The second is the pressure value. The specification says nothing about the relationship between pressure and pressure value sent but in practice a value of 0 to 1 means virtually none and a value of 127 means loads of it.

The polyphonic key pressure message is not often used because only the most expensive keyboards have pressure sensing for individual keys. A more common and closely related message is the Dn *channel pressure* message. This is the same except that the key number is not transmitted so the message contains only one data byte. The data represents the average force applied to all the keys.

Status message Cn is the program/patch change message which uses a single data byte to select the new patch on the receiving instrument (from 0 to 127). That's great if you're sequencing but awkward if you're just using one keyboard playing with another. The chances of wanting to change both synths simultaneous to patch 10, say, are pretty low. Happily it is usually possible to disable these messages either on the transmitting or receiving instrument.

Status message En is for the pitch bender. In this case the data transmitted is a 14-bit number, transmitted as two data bytes containing seven data bits each. The median value in this case is 8192, or 2000 in hexadecimal, which is transmitted as the three hexadecimal bytes, En 00 20. This value corresponds to a pitch bend of zero.

Again the MIDI specification is not specific about the relationship between the pitch bend value transmitted and the amount of pitch bend it corresponds to. Often instruments only have 7-bit resolution, with the least significant byte containing only dummy information (fourteen bits are always transmitted).

The two different kinds of key pressure message and the pitch bender message are unlike the *note on* and *note off* messages in that they are continuously variable values which must be repeatedly transmitted. The MIDI specification makes no mention of the rate they are transmitted at. This is up to the equipment designer. In practice, these messages are only transmitted when the pressure or pitch bend value changes so transmission rate depends on the resolution of the sensors, the rate they are scanned, and their use at the time.

Control Change

If you were getting worried that MIDI could only control seven parameters, then fear not. The Bn channel voice message deals with a good few more. It is called the control change message and contains two data bytes: first the control number and second the control value.

The 128 control numbers are by no means all allocated to specific controls, in fact the MIDI standard itself identifies only one — modulation on control number 1. Others have slowly become standardised (see Table 2) but most are undefined. This is not a bad thing. As more and more equipment becomes MIDI compatible, parameters arise that simply didn't exist before. The versatility of these undefined control numbers is one of MIDI's greatest assets.

Control numbers 0-63 are paired to give 14-bit resolution as shown in Table 3. Modulation then, has its MSB with control number 1, and its LSB with control number 33. Control values can be anything from 0 to 16383 and these controls are called continuous controllers — the digital equivalent of a potentiometer.

Controller numbers 64 to 95 are assigned to switches which can be either on or off. To turn a switch off, a value of 0 is sent and to turn it on, a value of 127 is sent. How a receiver interprets any other values (1-126) in between is not defined, but most equipment will ignore them.

Mode Messages

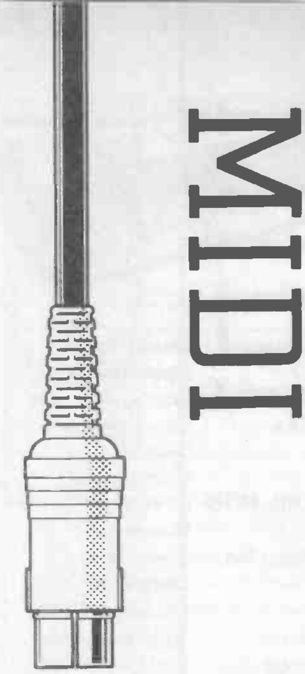
The six control numbers from 122 to 127 are used to select between the different MIDI modes with which you are of course totally familiar (see Table 4). Mode 1 for instance is selected by sending a message to controller 125 then another to controller 127 (OMNI ON, POLY).

These all have the additional function of turning all notes off — controller 123 being different in that it serves no other purpose. However, the MIDI specification says that in no case should these be used to turn off notes previously turned on by *note on* messages. These should be turned off by specific *note off* commands. Since there is no possibility of notes otherwise being left on, 'All notes off' commands may safely be ignored. Readers puzzled by this strange aspect of the specification may rest assured they are not alone!

Control number 122 can be used to disconnect an instrument's keyboard from its sound generating circuits so that the keyboard data goes only to the MIDI OUT socket and the sound generators are only controlled by the MIDI IN data received. This is an optional feature. Not all instruments are capable of being controlled in this way.

System Messages

Now we can think about system messages. These are unlike channel voice and mode messages. They are not related to channels and do not have channel numbers included



Number	Function
1	Modulation Wheel
2	Breath Controller
4	Foot Controller
5	Portamento Time
6	Data Entry Knob
7	Main Volume
64	Sustain Pedal
65	Portamento
66	Sostenuto
67	Soft Pedal
96	Data Entry Increment
97	Data Entry Decrement

Table 2 Common use of controllers

Number	Function
0	Continuous Controller 0 (MSB)
1	Continuous Controller 1 (MSB)
2-31	Continuous Controllers 2-31 (MSBs)
32	Continuous Controller 0 (LSB)
33	Continuous Controller 1 (LSB)
34-63	Continuous Controllers 2-31 (LSBs)
64-95	On/Off Switches (0=off, 127=on)
96-121	Undefined
122-127	Reserved for Channel Mode message

Table 3 Assignment of control numbers

Message	Status	Data
Local control off	Bn	122,0
Local control on	Bn	122,127
All notes off	Bn	123,0
Omni Mode off	Bn	124,0
Omni Mode on	Bn	125,0
Mono Mode on	Bn	126, No. channels
Poly Mode on	Bn	127,0

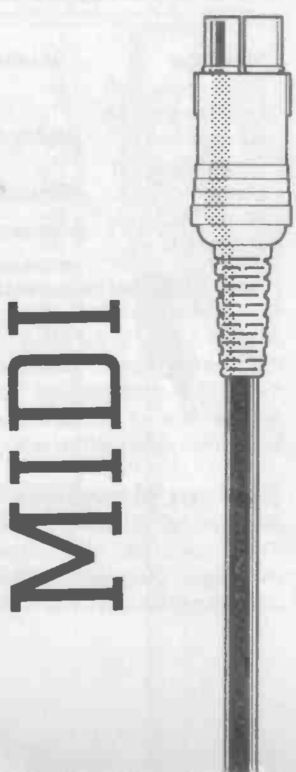
Table 4 Channel mode messages

	Status	Data	Function
Enter exclusive	F0	ID data data	Data specific to equipment (any number data bytes)
Common	F1	—	Undefined
	F2	LSB, MSB	Song Position Pointer
	F3	Song No.	Song Select
	F4	—	Undefined
	F5	—	Undefined
	F6	none	Tune Request
	F7	none	End of Exclusive
Real time	F8	none	Timing Clock
	F9	none	Undefined
	FA	none	Start
	FB	none	Continue
	FC	none	Stop
	FD	none	Undefined
	FE	none	Active Sensing
	FF	none	System Reset

Table 5 System messages

Manufacturer	Hex code	Decimal code
SCI	01	1
Big Briar	02	2
Octave	02	3
Moog	04	4
Passport Designs	05	5
Lexicon	06	6
Ensoniq	0F	15
Oberheim	10	16
Bontempi	20	32
Siel	21	33
Kawai	40	64
Roland	41	65
Korg	42	66
Yamaha	43	67
Casio	44	68

Table 6 System exclusive ID codes



in the status byte. Particularly important, the system messages include the codes that are used to synchronise sequencer timing so a synth and a drum machine can play along together without any voice messages being transmitted between them. Each sequencer is programmed separately and only the timing codes are needed to make them play together.

System messages are distinguished by status bytes with the four most significant bits set (Fn). They are divided into three types, *System Common, System Real Time,* and *System Exclusive.* Table 5 summarises the system messages.

System exclusive messages give the ultimate in MIDI versatility. In one respect they go against MIDI in that there is absolutely no standardisation between manufacturers, but in another respect it helped MIDI gain acceptance because manufacturers knew they could do as they pleased in this little niche of the MIDI system.

The status byte for systems exclusive is F0 and this is followed by the manufacturer's unique ID code (Table 6). After that there is no limit to the number of data bytes sent or received and little restraint on what manufacturers can achieve here. System exclusive messages are becoming more complicated and so more powerful all the time.

The message is terminated by an F7 'end of exclusive' status byte or any other status except real time codes F8 to FE. An instrument which does not recognise the ID code at the start of a data stream ignores the data until another status byte indicates the end of the exclusive data.

There are four defined system common messages. F3 status messages are *song select* messages which choose between songs numbered 0 to 127 and F6 *tune request* is used to start analogue synthesisers tuning to their automatic oscillator, if they have one.

F2 status messages are *Song Position Pointer messages.* These messages are closely related to the real time *timing clock, start, continue,* and *stop* codes, and allow the user to start or continue a sequence from a specified point in the song. The data given is a 14-bit value coded as two bytes, the least significant being sent first. It indicates a time within the song in multiples of sixteenth notes, from the beginning of the song. Few keyboard instruments actually implement the song position pointer feature.

Clocking In

There are six system *real time* status codes. These do not have associated data bytes and with the exception of system reset code EF can be transmitted at any time, even in the middle of other messages. Another little pitfall for the unwary! This aspect of the specification is to allow greater timing accuracy in their transmissions, though this is only really important in the case of the timing clock F8. 'Running status' is not changed by the system timing codes.

F8 timing clocks are sent at the rate of 24 clocks per quarter note or crochet and are used to synchronise sequencers and instruments with built-in sequencers — a synthesiser sequencer and a drum machine, for example. Timing clocks must be ignored until an FA *start* or FB *continue* code is received. This is important to note because some instruments continuously transmit timing clocks regardless of whether their sequencers are running or not. FC *stop* codes stop the sequencer running.

The *start* code starts the sequencer from the beginning very time, whereas the *continue* code starts it from where it was last stopped or, if the *song position pointer* feature described above is used, from where the pointer was set.

FE codes are for *active sensing*. This means some transmitters continuously produce this code when nothing else is being transmitted, just so the receiver knows the transmitter is still active and plugged in. The transmitter should transmit this code at least every 300ms (preferably faster, say every 100ms to 250ms) when there is no other activity on its output.

A receiver which does not receive any input for at least 300ms (or perhaps 400ms to be on the safe side) then knows that the transmitter is no longer active and can turn off its voices and return automatically to stand-alone operation. However, not all transmitters produce this code so receivers should not assume its presence until an FE code is actually received in the first place.

The remaining real time code is *system reset.* This is used to reset the system to the condition of just having been switched on. It should be used sparingly, if ever at all. Instruments must not send out this as part of their cold start routines because this could result in two instruments connected back to back forever resetting each other when swtiched on and never actually getting to the stage where they could be used!

In some cases it does not seem to make much sense for an instrument to completely reset itself just because a MIDI command has told it to, especially in the case of something like a computer used as a sequencer. Does the user really want it to completely re-boot the sequencer software and quite possibly lose its programmed sequence because its MIDI IN socket told it to? Perhaps the computer sequencer would no longer even be a sequencer without changing disks. Not very helpful! In this case a decidedly warm start would make more sense than a cold start. This is one aspect of the MIDI specification that should perhaps be given a somewhat loose interpretation!

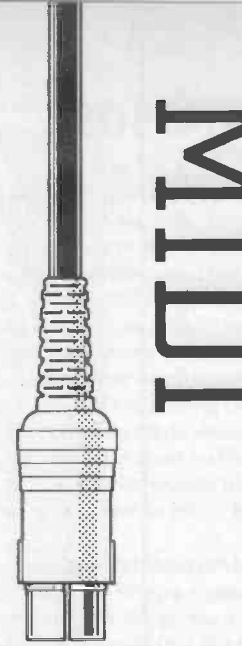
In Conclusion

Just a quick note to qualify the rather rash statement earlier that the MIDI baud rate is high enough to transmit 16 channels of channel messages without any audible delays. Apart from the skew problems of having 16 units chained together, it actually isn't that difficult to overload a MIDI link if you send enough modulated, pitch bent, polyphonic megachords down enough channels simultaneously.

Most MIDI setups will never reach these limits in normal use unless they are enormously expensive or uncommonly busy. If this is the case then the owner must either design his MIDI linking very carefully or wait for some sort of MIDI Turbo with faster propogation and more channels.

WHAT PRICE MIDI?

This comprehensive round-up reveals how much MIDI your money will buy



Although MIDI is a relatively inexpensive facility for manufacturers to add on to new equipment, when it comes to keyboards and budget synthesisers, the option is usually ignored.

Why should this be? Is there no demand for MIDI at this level? Are the prices too low for the big boys to bother making the effort? Or is MIDI simply no use on cheap keyboards?

No to all three. Manufacturers assume users of budget and home keyboards are amateurs who would have no use for MIDI and probably wouldn't understand how to put a system together anyway.

This is obviously unfair. Bands on the pub and club circuit and home studio enthusiasts don't make enough money to buy the latest DX wonder synth but that doesn't mean they are incapable of operating a useful MIDI set-up using cheap keyboards. Indeed it is in situations like this where MIDI can make its greatest impact.

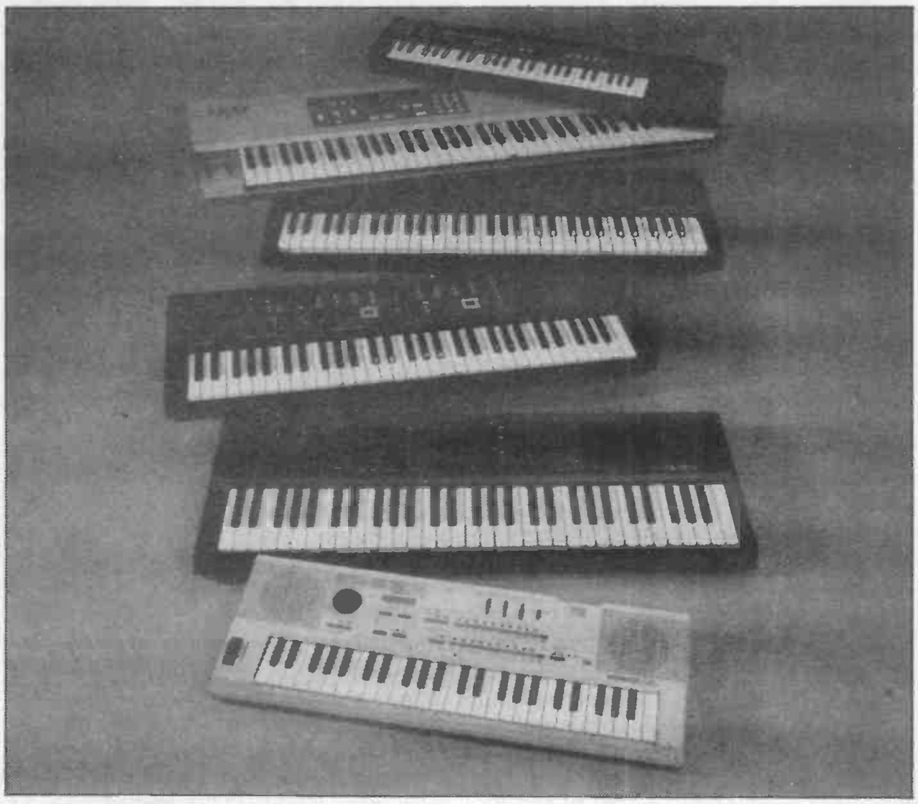
In recent years the usual distinctions between synthesisers and home keyboards have become dangerous yardsticks by which to judge an instrument — the borders between the two have become fuzzy to say the least. This round-up puts everything into the melting pot together, the only criteria being the inclusion of MIDI facilities and a price under £600.

Why £600? No reason, except perhaps to allow a few serious synths from the big boys onto the list. In any case, a price of £600 doesn't mean a whole lot — you should be able to get that down to about £525 with a little bit 'o luck. (Remember, RRPs are for people with money to burn. We're working on a budget and we'll fight to keep the coins in the coffers).

Each review is pretty much self-explanatory. A few general points though, mainly to save repetition later. First it is remarkable how many keyboards have no modulation wheel. Modulation can be as expressive as pitch bend and if one had to be jettisoned to keep the hardware down, I'd hang on to modulation.

There is also a marked lack of MIDI THRU sockets. These won't be too long in coming but this does hinder the creation of a complex MIDI system without the use of THRU boxes — just the sort of expenditure we're trying to avoid.

Enough preamble. Let's get our teeth out and our hands on. Lock up your Access card, there's some goodies coming up.



Akai AX73

The Akai AX73 (second from the top in the photo) is a serious synthesiser. The full size 73-note (6 octave) touch sensitive keyboard is housed in the most solid steel enclosure imaginable, with the front panel controls functionally designed using two sliders (volume and data value) and grey keypads.

A backlit 16-character LCD displays the names of selected sounds or the parameter under alteration if in edit mode. Pitch bend and modulation wheels are provided. The AK73 is 6-voice polyphonic.

There are no rhythms, autochords or accompaniment here. The 100 memories are all rewritable and can be saved to tape either individually or in bulk. The sounds supplied with the review keyboard varied from those described in the manual but most were good — brass sounds are especially effective and make excellent use of the touch sensitivity. Synth sounds are rather few and far between although the capability is certainly there to create some crackers. One novelty sound — the Boeing 747 — is astoundingly accurate and powerful, a lovely piece of programming.

Editing is reasonably simple using the list of parameters in the manual. Data entry is by slider and incrementation pads — parameter selection is made using the same bank and number buttons as for the voices. Complex results tend to be stumbled upon rather than designed — I created a very passable seagull while aiming for a toy piano! Full control is given over VCO, VCF, VCA and LFO. Performance parameters (pitch bend range, modulation depth, sensitivity and so on) are programmed into each sound. The only settings common to all voices are MIDI channel and program change information.

As a MIDI keyboard, the AX73 is exceptional. It transmits the full six octaves, sending and recognising velocity information, pitch bend, modulation and sustain. A MIDI split facility can be introduced which sends note information from the lower or upper section (selectable) on the MIDI channel six above the chosen basic channel. Its main omission is the inability to receive in MIDI mode 4 — no multitimbral operation of any kind is possible.

Nevertheless the AX73 is a highly professional instrument offering 100 fully programmable memories and a valuable master keyboard. Its drawbacks are mainly in the absence of any home keyboard facilities (rhythms or on-board sequencing) but this is obviously not the purpose for which it was designed. The value of its rugged construction and roadworthyness should not be underestimated.

Casio CZ-230S

Casio has gone to some pains to present this keyboard as a serious synthesiser. The main barriers to professional use are the fiddly mini-keyboard, the limits of 4-note polyphony and the inability to edit preset sounds or create your own (except using a computer — see below).

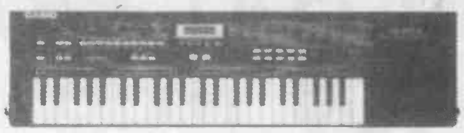
There are 100 presets to choose from, all created using Casio's phase distortion (PD) synthesis. In actual fact there is 8-note polyphony but only eight presets achieve it. All the others are doubled up for more complicated 4-voice sounds.

The quality of the presets is excellent — the only thing missing is a good string sound. Synth sounds are imaginatively designed and amusingly named (all 100 sounds are listed on the front panel). Many sounds have preprogrammed reverb and sustain — occasionally too long but generally very useful.

The 12 PCM percussion samples are good and the 20 preset rhythms are reasonable — the funk is excellent. There are no autochord or bass accompaniments. There are however 10 memories for your own rhythms and the manner of entering them credits you with a fair bit of musical knowledge. Each PCM sound is entered individually. The keyboard represents a complete bar and pressing middle C, say, puts the sound at the midpoint of the bar. It takes some time to master but can produce useful results. It does however limit rhythms to 12 quantised intervals per quarter bar.

Up to four monophonic accompaniments can be added to programmed rhythms but these have to be individually scored note by note — real time entry really would speed things up here. Nevertheless it is powerful stuff — the CZ-230S is fully multitimbral and each line can use a different sound but if all four accompaniments are used there is no polyphony left for the keyboard itself.

The 199 step song memory gives about seven minutes at an average tempo so it is



certainly a realistic tool — the song and rhythm melodies can also be dumped to tape. I wouldn't recommend live use but in a home studio under the control of a MIDI clock it can be a handy little function.

MIDI specs are excellent. Full mode 4 allows multitimbral sequencing on four separate and adjacent MIDI channels. It sends and receives pitch bend, portamento on/off, program changes, clocks and rhythm information. A 16-page manual is supplied with the keyboard describing control via systems exclusive. This gives access to the complete phase distortion synthesis parameter system and with a computer, some custom software and a few lessons on PD synthesis, the CZ-230S becomes totally programmable and immensely powerful.

Farfisa TK120

Boldly emblazoned on the front panel of the TK120 are the words 'Keysynth-Digital-PCM-MIDI'. This sounds most impressive and would be so if the keyboard was a little simpler to use.

The 5-octave keyboard is heavy and rather rigid. There is no pitch bend or modulation. The 16 preset orchestra sounds are quite respectable — a pair of fine organs, good vibes and mellow brass — all helped by the variable speed stereo chorus.

There are in addition five rather ordinary solo presets together with a clever 'MPS' feature which automatically uses the solo voice if a single key is pressed but the orchestra voice if a chord is played.

The 13 PCM sampled percussion sounds are good but limited by their number. The 32 preprogrammed rhythms are dull in the extreme though the autochords and bass are extremely good. There is also a facility for the solo voice to play an auto-melody (a facility which seems to select itself on occasion). These melodies are well programmed but with all systems go leave little room to actually play anything yourself. There are 16 memories for your own rhythm, bass, chords and auto-solo and some quite complicated patterns can be produced.

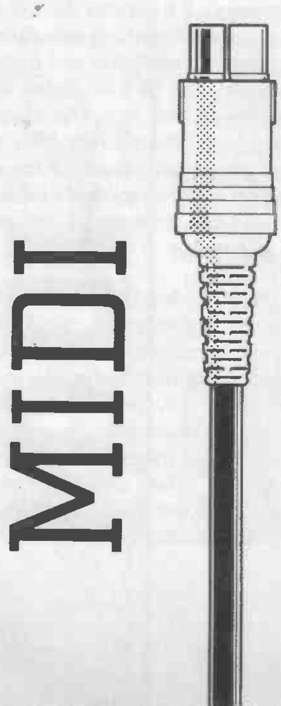
Creating voices is not so simple and the first step to success is to ditch the owner's manual. Suffice to say that the possibilities are limited and aiming in a specific direction is difficult unless you want an organ sound!

MIDI facilities are good. A THRU socket is provided. It transmits and receives as Omni off or on and is most flexible using a mixture of MIDI mode 3 (for the orchestra voice on the basic channel) and mode 4 (for the bass, chords and solo voices on separate channels).

Bearing the price in mind, the TK120 provides powerful auto-accompaniment, a good range of MIDI facilities and a full size 5-octave keyboard, but limited scope of preset voices and a small range of percussion.

Casio HT-700

The HT-700 is also licensed to Hohner as the KS49 and the boys in Japan have done themselves proud. The rather fiddly mini-keyboard is the main problem but the presets are excellent — 20 ROM and 20 RAM (which can be rewritten or replaced) plus 20 more on RAM card.



Several of the presets are rather muffled and need a bit of EQ to brighten them up. The output is also a bit noisy and needs a noise gate in a studio. Most voices are excellent. Brass is very aggressive, the strings are rich when chorused and the jazz organ is a great Emerson *America* Hammond sound. The synth-type sounds range from brilliant to phenomenal for this price bracket.

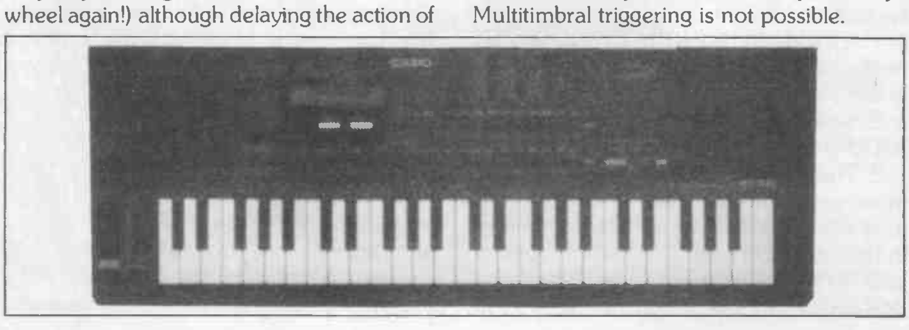
With the manual on your lap, programming your own sounds is simple and quick. The system is a simplified version of Casio's Phase Distortion. Parameters and data are entered using the entry wheel and new sounds can be saved to the internal voice bank.

All this can be done during performance but freezes the rhythm facilites. Editing is the only way of using modulation (no dedicated wheel again!) although delaying the action of the modulating LFO can be very effective.

The percussion is solid and, although the PCM sounds are not perfect sample quality, patterns are imaginative and the auto-accompaniment is often inspired. Rhythms can be written to replace the ones on the keyboard or for storing on RAM card.

Each accompaniment voice can be PD programmed and used with the rhythm off, effectively splitting the keyboard. The ranges are rather small like this and triggering from a full-sized keyboard has the added advantage of accessing an extra octave above the range of the HT700's 4-octave keyboard.

The HT700 operates in Omni On, Poly mode (MIDI mode 3) with the four channels above the basic channel being used for chord, bass and rhythm data respectively. Multitimbral triggering is not possible.



Kawai K1

The K1 is easily identified as a professional synthesiser rather than a home keyboard. The 61-key full size keyboard is fully touch sensitive and recognises after-touch as well. Both pitch bend and modulation wheels are provided.

The facilities offered and techniques involved in editing puts the K1 as a clear competitor to Yamaha's DX11. Like the DX11 the K1 is multi-timbral which enables sounds to be significantly thickened by layering one on top of another. However, many of the single sounds on the K1 are so rich and thick on their own that combinations are a bonus rather than a necessity.

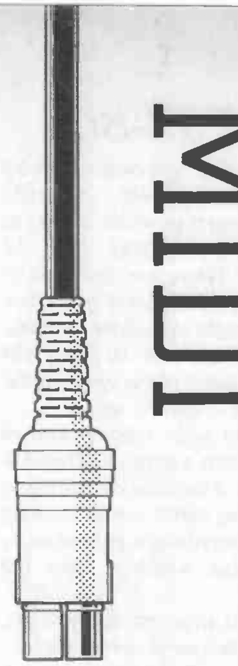
The secret is in the LA synthesis. Each single sound comprises 4 source waves — equivalent to DCOs in analogue synthesis or the operators in FM. In normal FM these are sine waves, in the DX11 there are various shaped waveforms available but in the K1 some 204 different VM (Variable Memory) sources are available. Each source is independently tuned, enveloped and modulated before being put with the other three sources to form a single voice.

The results created from four VM waveforms are admirable, with some great synth sounds and good clean brass not dissimilar to 4-operator FM creations. Kawai holds its trump card in the additional 52 PCM sampled waveforms that can be used alongside the VM shapes to create voices. These include a complete drumkit plus looped string bows, piano and guitar timbres and breathy pipes. The ethereal breathy quality PCM waves are without doubt the best utilised and the most effective. Four of these sources in combination produce an 'Ahhh' sound (that's how it is labelled on the K1!) worthy of other keyboards priced £1500.

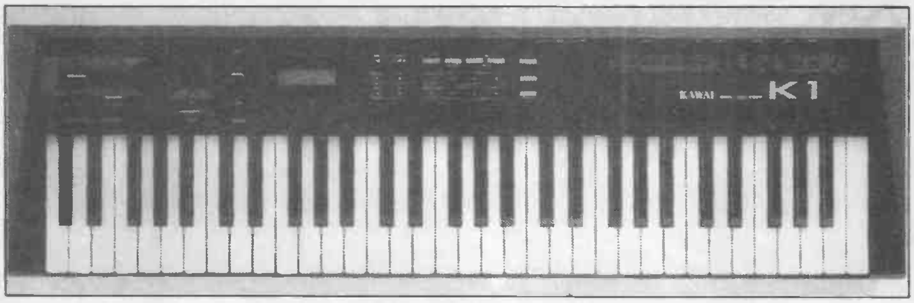
As if that wasn't enough, the multitimbral functions can be used to produce multi-patches of up to eight single voices, each individually adjusted for volume, delay and keyboard range. In addition you can specify if each of the eight single voices is to be triggered only by a fast key depression, only by a slow depression or by any playing. This enables one sound, say piano, to be produced by soft playing, and an entirely different one, say bells, by loud playing, as well as having up to eight sound splits and overlaps across the keyboard. This can produce enormous performance flexibility.

The keyboard is also totally multitimbral as far as MIDI is concerned — each of the eight sound splits of a multisound can be assigned to a different MIDI channel for transmission or reception (full MIDI Mode 4 in effect).

Once the synthesis method is grasped, editing is simple though slightly longwinded as some adjustments must be made by pushing the same button many times to cycle through the various parameters. The 64 single and 32 multi-memories can be



expanded with memory cards. Most single sounds are 8-voice polyphonic, although it is possible to have 16-voice polyphony using just two sources for a voice instead of four. Polyphony is obviously reduced with multi-patches though this depends how the keyboard splits have been arranged.



Kawai MK10

The first thing to commend the Kawai MK10 is the front panel. The 92 LEDs (yes, 92) look marvellous in action, just like the old days. Wildly extravagant of course — the whole lot could be replaced with a well thought-out LCD display — but Kawai are likely to sell a good few units on this point alone.

More important is the keyboard. The MK10 has a velocity sensitive 61-note range and a good keyboard feel as well, a trifle light and bouncy perhaps but very playable. The pitch bend wheel rocks nicely but there's no facility to adjust its range. No modulation wheel is provided.

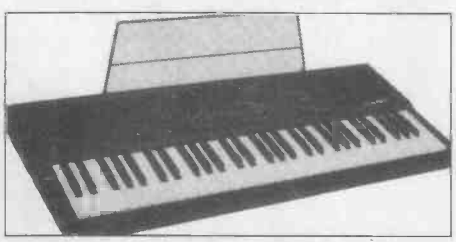
The sounds are arranged in two main sections, solo (monophonic) and orchestra (polyphonic). With both sections selected the solo voice plays the top note of chords and the orchestra voice will play the others. This arrangement works well in general but removes the ability to play bass lines with the solo voice.

There are other voices in there but they are allocated exclusively to the auto play accompaniment. This is a shame since some of these voices are better than the available presets — notably the electric basses and the violin. To get at these, the drums must be running and then the sounds will sequence anyway. All you can do is change the chord and play over the top.

The presets themselves are reasonably good if rather unimaginative. There are 18 solo presets, the best of which are the chimes, guitars and synths. From the 18 orchestra sounds, organ 1 is recommended (a good warm Hammond) and marimba or vibes are great for those *Tom Waits* covers. The strings are almost there but are too metallic and digital. I would have liked a few more interesting synth sounds but Kawai has chosen to follow the tradition of having home keyboard sounds resembling real instruments.

The keyboard can have an orchestra preset on each side of a 3-position split point. The solo voice still operates in the top part, bearing in mind that you have only 8-note polyphony in total.

Going on to MIDI the MK10 works well as a touch sensitive master keyboard but



doesn't send pitch bend information for orchestra voices. With the keyboard split in operation each part is sent on a different MIDI channel predefined by Kawai.

It supports Omni On or Off but is not fully multitimbral — it doesn't operate in MIDI Mono mode.

The PCM sampled drum sounds are excellent (28 of them in all). Apart from the 32 rhythms available, these can be played from the keyboard in 'hand percussion' mode which also makes them touch sensitive. There is no facility for writing your own patterns to memory.

While the auto play accompaniment voices are pretty good, the patterns are tedious and the fingering system for indicating chords is ludicrous — one finger gives major, one finger plus F# (regardless of the key) gives minor. You don't get say an Ab chord by pressing Ab, you have to press A and Db (Db is the key which indicates a flat key). The system is utterly hopeless if you have any idea what a chord is and will go a long way towards stopping those who haven't from ever learning.

The MK10 represents value for money thanks to its velocity sensitive keyboard, split facilities and drum sounds. Its drawbacks are zero programmability and a badly organised (though versatile) auto accompaniment section.

Yamaha PSR-80

Front panel controls on the PSR-80 are deceptively simple considering it provides access to 400 preset voices and 32 rhythms. These are squeezed in courtesy of a novel 'voice variator' giving five timbre variations (bright to mellow) and five attack variations to each of 16 basic FM presets. The only problem of this system is the difficulty in recalling a specific voice.

The presets are wide ranging and of impeccable calibre with a gorgeous 'brass & chimes' combination, a versatile electric piano and some interesting synth noises coming from the variator. Everything is enhanced by the on-board chorus which is light but effective.

There are eight monophonic presets, less useful without pitch bend or modulation. There is no programmability of any sound, nor is there any way to adjust the sound balance between orchestra and solo sounds.

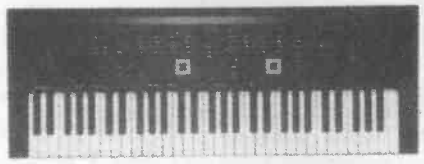
The PCM percussion samples are superb. The 33 sounds have six different snare types including two with brushes and an excellent gated snare. Sample lengths of up to a second allow a decent ring on open hi-hat and cymbals — the only criticism is that some samples are too quiet — brushwork and bongos especially.

The preprogrammed rhythms are well thought-out, the auto accompaniment is not (except baroque). A second variator sets the tempo and allows selection of three 'sizes' of accompaniment (small group to large group). There is a single memory for programming a new rhythm and accompaniment. This produces excellent results but since there is no facility to store more than one creation (other than a bulk data dump via MIDI) its usefulness is limited.

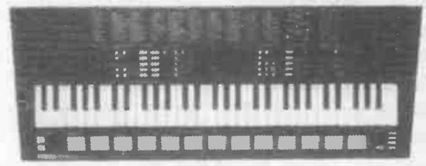
A chord sequencer of 2500 chords is included but again this is limited to a single creation.

The MIDI specs for the PSR-80 are fairly comprehensive although confusing to implement without any form of display for confirming instructions. Normal operating mode is Omni On, Poly (MIDI Mode 3) but a limited mode 4 can be achieved with orchestra, solo, bass, chords and rhythm each receiving (or transmitting) on separate channels. Local off and program change off are available and panel settings or sequencer/custom accompaniment data can be sent via a systems exclusive bulk data dump. There is however no MIDI THRU socket and no recognition of pitch bend, modulation or key velocity.

The helpful design and high quality FM presets of the PSR-80 are spoiled by the lack of adequate storage for custom accompaniment and on-board sequencing. The MIDI facilities lend themselves well to external control although several basic control parameters are not recognised.



PSR-80



PSS680

MAKE & MODEL	PRICE	KEYBOARD	VOICES	EXT STORE	SPLIT	VOICE PROGRAM	RHYTHM	ACCOMPANIMENT	MIDI	OTHER
Akai AX73	£599	full size 73-note velocity sensitive	6-voice 100 RAM	tape	A	●	none	none	IN, OUT, THRU mode 3 plus 2-channel send	pitch bend modulation chorus LCD display
Bontempi ES6500	£299	full size 61-note	8-voice 8 preset		B		10 presets	autochords, bass, arpeggio	IN, OUT note on/off only	stereo chorus
Bontempi ES700	£399	full size 61-note	8-voice 10 preset		B		PCM 10 presets	autochords, bass, arpeggio	IN, OUT note on/off only	stereo chorus
Casio CT460	£299	full size 49-key	10-voice 30 preset + tone bank		B		PCM (49 sounds) 20 presets	autochord, bass	IN, OUT	real time sequencer 8 sound FX
Casio CT630	£349	full size 61-key	8-voice 60-preset		E		20 preset	autochord, bass	IN, OUT	
Casio CT640	£299	full size 61-key	10-voice 30 preset + tonebank		B		PCM (49 sounds) 20 presets	autochord, bass	IN, OUT	real time sequencer
Casio CT660	£349	full size 61-key	10-voice 30 presets + tonebank		B		PCM (49 sounds) 20 presets	autochord, bass	IN, OUT, THRU	stereo delay, panning 8 sound FX
Casio CZ230S	£345	mini-keyboard 49-key	4-voice 100 presets				PCM (12 sounds) 20 presets 10 RAM	10 RAM for 4-voice accompaniment	IN, OUT, THRU mode 2, full mode 4	pitch bend programmable via MIDI
Casio HT700	£299	mini-keyboard 49-key	8-voice 20 preset 20 RAM	RAM RAM card	B	●	PCM (15 sounds) 20 presets 10 RAM	autochord, bass 10 RAM	IN, OUT mode 3 plus 4-channel transmit	pitch bend light chorus
Casio HT3000	£429	full size 61-key	8-voice 30 preset 30 RAM	RAM card	E	●	PCM (15 sounds) 20 presets 10 RAM	autochord, bass 10 RAM	IN, OUT, THRU mode 3 plus 4-channel transmit	pitch bend modulation light chorus
Casio MT240	£149	mini-keyboard 49-key	10-voice 20 preset + tonebank		B		PCM (49 sounds) 20 presets	autochord, bass	IN, OUT	real time sequencer
Casio MT540	£199	mini-keyboard 49-key	10-voice 20 preset + tonebank		B		PCM (49 sounds) 20 presets	autochord, bass	IN, OUT	real time sequencer 8 sound FX
Casio MT600	£179	mini-keyboard 49-key	8-voice 40 preset		B		20 presets	autochord, bass	IN, OUT	chord memory pitch bend
Casio MT630	£349	full size 61-key	8-voice 60 presets		B		20 presets	autochord, bass	IN, OUT	chord memory
Elka EH105	£499	full size 61-note	8-voice 30 preset 8 RAM		B	●	PCM (17 sounds) 18 presets 4 RAM	autochord, bass	IN, OUT mode 3 plus 4-channel transmit/ receive	digital recorder
Farfisa FK58	£249	full size 61-key	8-voice 6 preset		B		8 presets	autochord, arpeggio	IN, OUT	RS232

Yamaha PSS680

The PSS680 is one of three keyboards that Yamaha has brought to the market under the banner of *Style Play* — so called because of the carefully preprogrammed rhythms and chords that apparently rise above the realm of auto-accompaniment to become 'professionally orchestrated styles'.

Both the PSS680 and its little brother the PSS480 do indeed have a wide variety of rhythms. The PSS680 has the major advantage of PCM drum sounds, which can also be triggered from eight percussion pads below the keyboard. There are 32 of these excellent samples (special commendation for the latin-style whistle) — a range which equals the top of Yamaha's home keyboard range, the PSR80 and 90.

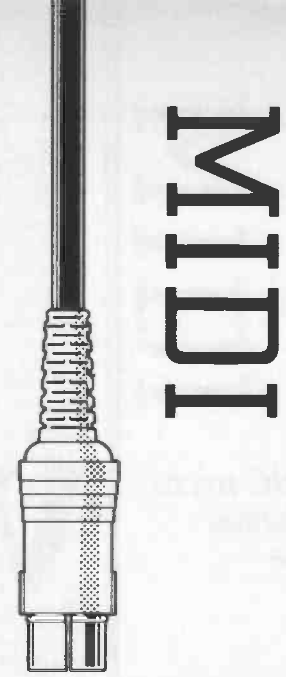
There are 100 preprogrammed rhythms each with its own accompaniment. Thinking up 100 rhythm names obviously proved tricky and titles such as 'Argentinian Folklore' fill out the ranks but the programmed accompaniments are spectacularly effective. Each makes use of 5-track multitimbral sequencing, an astonishing facility at a price under £250 and one which gives a powerful and complicated backing that justifies Yamaha's pride and promotion.

Even more impressive is its MIDI implementation which sends each of the five accompaniments out on a separate channel and similarly can receive data for five separate timbres over and above that selected for the main keyboard.

The most frustrating aspect of this powerful capability is the provision of only one memory for your own sequence rhythm and 5-part backing, with no way to dump to external storage.

Similarly the 100 preset FM sounds can all be edited to a limited extent but only five memories are available to store new sounds. The FM presets are clean (as you would expect) but slightly weedy — only the purer sounds such as flute, organs and whistling are really effective. Since the keyboard has multitimbral applications it would be wonderful if layering and keyboard splits had been made available but at this price it is difficult to complain.

However, it is possible to moan about the size of the keyboard on these home instruments which seem to have a few millimetres shaved off the keys of each new



model. Problems will definitely be encountered by anyone used to a normal keyboard. However its FM presets, PCM percussion and multitimbral facilities place it in a commanding position as a home keyboard with professional extras. The price is astoundingly low.

MAKE & MODEL	PRICE	KEYBOARD	VOICES	EXT STORE	SPLIT	VOICE PROGRAM	RHYTHM	ACCOMPANIMENT	MIDI	OTHER
Farfisa FK65	£299	full size 61-key	8-voice 8 preset		B		10 presets	autochord, bass	IN, OUT	RS232 stereo chorus
Farfisa FK70	£399	full size 61-key	7-voice 10 preset		B		PCM 12 presets	autochord, bass	IN, OUT	
Farfisa TK80	£499	full size 61-key	11-voice 40 preset		B		PCM 16 presets 1 RAM	1 RAM autochord, bass, solo	IN, OUT	
Farfisa TK120	£349	full size 61-note	16 preset 5 solo			•	PCM (13 sounds) 32 preset 16 RAM	16 RAM for bass, chords and autosolo	IN, OUT, THRU mode 3 plus 4 channel transmit	stereo chorus
GEM DSK5	£549	full size 61-note	7-voice 10 preset		C		PCM 12 presets	autochord, bass	IN, OUT mode 3 plus 4-channel transmit/ receive	
GEM DSK6	£549	full size 61-note	8-voice 16 presets		D		PCM (11 sounds) 14 presets	autochord, bass	IN, OUT	
Hohner KS49	£365	mini-keyboard 49-note	8-voice 20 preset 20 RAM	RAM card	B	•	PCM (15 sounds) 20 presets 20 RAM	autochord, bass 20 RAM	IN, OUT mode 3 plus 4-channel transmit	pitch bend light chorus
Hohner KS61	£450	full size 61-note	8-voice 30 preset 30 RAM	RAM card	E	•	PCM (15 sounds) 20 presets RAM	autochord, bass RAM	IN, OUT, THRU mode 3 plus 4-channel transmit	pitch bend modulation light chorus
Kawai K1	£595	full size 61-note velocity sensitive	16-voice 96 RAM card	RAM	F	•			IN, OUT and THRU modes 1, 3, 4	pitch bend modulation
Kawai MK10	£626	full size 61-note velocity sensitive	8-voice 36 presets		E		PCM (28 sounds) 32 presets	autochords, bass	IN, OUT and THRU mode 3 plus 5-channel transmit	pitch bend
Yamaha PSR36	£360	full size 61-note	8-voice 32 preset + edit		E		PCM (25 sounds) 80 presets	autochords, bass	IN, OUT, THRU	
Yamaha PSR80	£600	full size 61-note	8-voice 400 preset		B		PCM (33 sounds) 16 presets 1 RAM	autochord, bass 1 RAM	IN, OUT, THRU	stero chorus
Yamaha PSS480	£180	mini-keyboard 49-note	6-voice 100 preset 5 RAM		B B	• •	PCM 100 presets 1 RAM	5-part accompaniment	IN, OUT, THRU	reverb
Yamaha PSS680	£250	mini-keyboard 61-note	6-voice 100 preset 5 RAM		B	•	PCM (32 sounds) 100 presets 1 RAM	5-part accompaniment multitimbral	IN, OUT, THRU mode 4	pitch bend reverb
Yamaha SHS200	£200	mini-keyboard 49-note	49 presets		B		49 presets	autochord, bass	OUT	pitch bend

A - programmable split as MIDI control keyboard only
B - split between accompaniment and orchestra
C - multisplit with memory
D - bi-timbral voice layering
E - two orchestra voice plus accompaniment
F - full multitimbral layering or multisplit

MIDI FOR THE BBC MICRO

Equip your BBC micro with a dual channel MIDI interface

While MIDI interfaces for the Commodore 64 abound and the Atari ST micros have one built in, the BBC micro has been largely left out — a shame as this is an ideal micro for controlling MIDI instruments.

This project provides a dual channel MIDI interface for the BBC micro and it also serves as a useful diagnostic tool for building the MIDI Master Keyboard (in this mag).

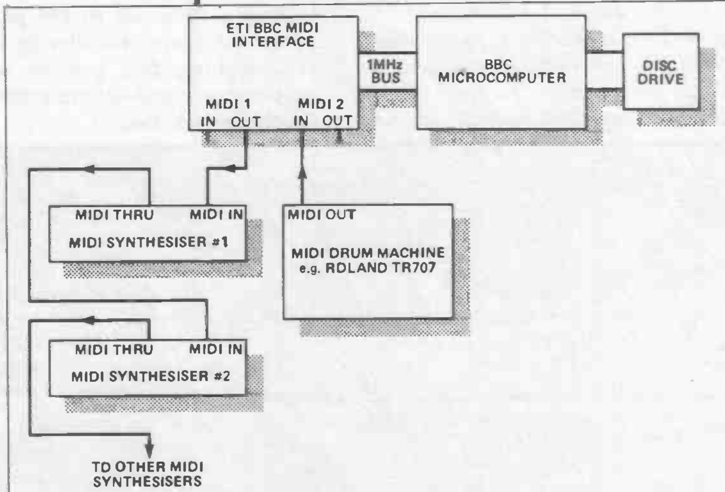


Fig. 1 Multi-track sequencer arrangement

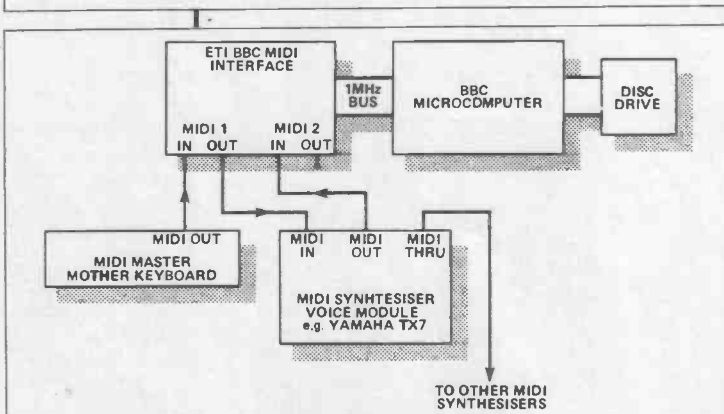


Fig. 2 Using the interface with a MIDI mother keyboard

MIDI

Operation

The interface has four DIN sockets, two for MIDI IN and two for MIDI OUT. The primary reason for the dual channel capability is to enable one MIDI channel to control a MIDI synthesiser whilst the other is connected to a MIDI drum machine such as the Roland TR707, as shown in Fig. 1.

This configuration can be used when the BBC micro operates as a MIDI sequencer, using the drum machine as an external synchronisation source.

Another useful configuration is shown in Fig. 2. Here a MIDI mother keyboard is connected to an external synthesiser module via the MIDI interface. Using suitable software both the mother keyboard and BBC micro can communicate with the synthesiser module. this is particularly useful for combining performance with voice dump and edit operations.

Construction

The interface is constructed on a compact sized double sided PCB. Figure 4 shows the component overlay. Mount all the resistors and terminals first, soldering the leads to both sides of the PCB where applicable. Next add the capacitors and diodes, leaving just the integrated circuits to be mounted.

The important word here is that the IC devices must be mounted directly onto the PCB without IC sockets, since many of the pins require soldering on both sides of the PCB. Sockets can however be used to mount the opto-isolators IC7 and IC8. Indeed this is suggested due to the relatively high cost of the devices.

There are two power supply alternatives. The first is to mount a 5V regulator on the PCB (IC10) along with associated components and power the whole unit from an unregulated DC power source. An ordinary mains battery eliminator giving out 9V at 300mA will be sufficient for the purpose. If you already have a regulated 5V source then the regulator circuitry components (IC10, C3, 4, R6 and LED1) can be simply omitted and bypassed.

Once the PCB has been assembled, check for shorts and unsoldered pads on both sides of the PCB. If everything is satisfactory then proceed with wiring up the 5-pin DIN MIDI sockets as shown in Fig. 3. The case used to contain the board is not crucial. Any suitable plastic case will serve.

Using The Midi Interface

The program presented in Listing 1 is a simple monitoring program. Connect a MIDI keyboard to the first MIDI IN socket of the interface and run the program. Any data transmitted by the MIDI keyboard will then appear on the screen. Notice that multi byte MIDI events are displayed on the screen if any keys are played, if a program change is made or if a pitch bend or modulation wheel is moved.

The operation of the program is as follows. The ACIA in the MIDI interface is programmed to interrupt the BBC micro every time it receives a byte of MIDI data through the first MIDI IN terminal. The BBC micro responds to the interrupt by placing the data in a buffer in memory. Whilst all this is happening the Basic program running checks to see if this buffer is empty. If not, it prints out the contents one by one until it is empty again.

Such a FIFO (first in first out) buffer is implemented because a simple Basic program to print out MIDI data as it arrives would not respond fast enough to the relatively high data rate of the MIDI messages.

Listing 2 is a program which allows the BBC's micro QWERTY keyboard to play any MIDI synthesiser that is set to receive on channel 1. The

PARTS LIST

RESISTORS (all ¼W 5%)	
R1-9	1k0
R10-15	220R
CAPACITORS	
C1, C2, C3	100n polyester
C4	1µ0 25V axial electrolytic
SEMICONDUCTORS	
IC1	74LS75
IC2	74LS74
IC3	74LS04
IC4	74LS30
IC5, 6	6850 ACIA
IC7, 8	6N138
IC9	74LS05
IC10	78M05 voltage regulator
D1, 2	1N4148
LED 1	Red LED
MISCELLANEOUS	
CON1	34-way IDC PCB mounting bus connector (male)
PL1	34-way IDC plug
SW1	2 pole, 2 way toggle switch (see text)
SK1-4	5 pin DIN sockets

PCB. Heatsink for IC10 (if used). Suitable power input socket. Plastic case. 34-way IDC plug. 34-way IDC ribbon cable. Nuts and bolts.

BUYLINES

None of the components used for the MIDI interface are difficult to obtain.

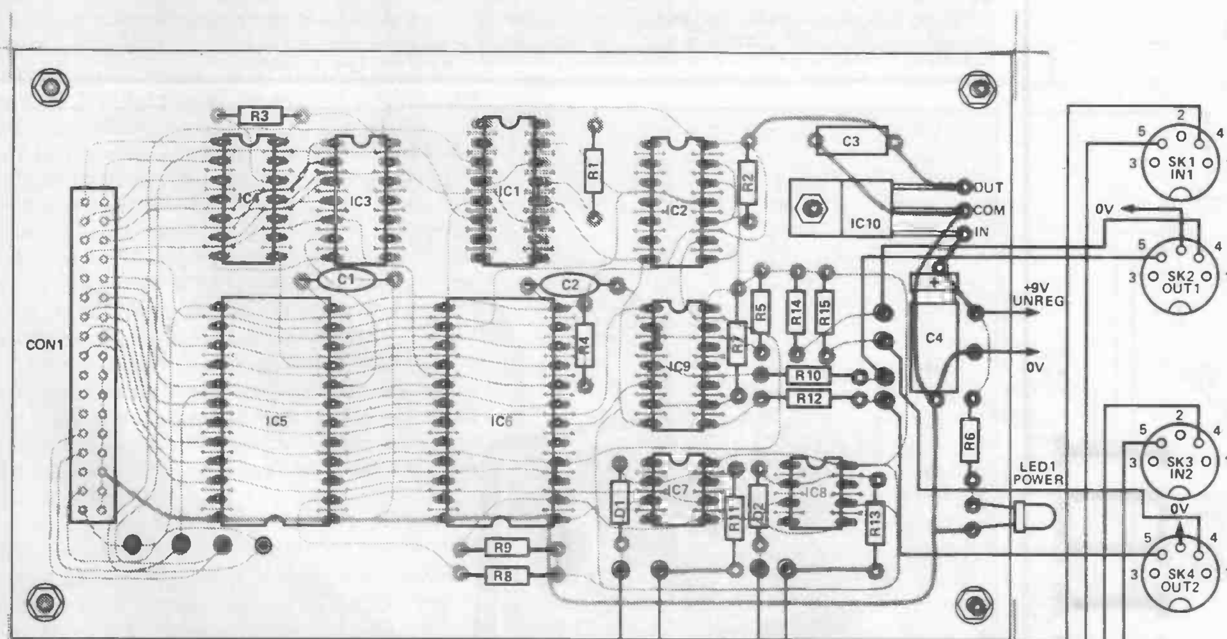
A Verobox type 103 was used for the prototype. This is available from Maplin (Tel: (0702) 552911) as are the 6850 ACIAs.

The 6N138 opto-isolators are available from Electromail (Tel: (0536) 204555) as catalogue number 302-126.

The author has a range of software on 5¼ in disk to make full use of the MIDI interface. Orders and requests for further details should be addressed to John Yau, 9 Harden Place, Edinburgh, EH1 1JD.

- Track dump software for the Roland TR707 drum machine. This enables the currently held patterns and track programs to be stored and read from floppy disk.
- Voice Dump/Editor software for the Yamaha DX/TX7. This software allows single voices or complete banks of voices to be stored and read from floppy disk. This software also enables DX voices to be edited from the BBC micro. All the parameters are displayed on the screen and can be modified using a simple slider control which plugs into the BBC micro's analogue user port. Wiring details for the slider control are supplied.
- Four track step time MIDI Sequencer for synthesisers arranged as in Fig. 1 with external synchronisation.

The programs cost £12, £18, and £16 respectively.



◉ = SOLDER TO BOTH SIDES OF THE BOARD

Fig. 3 The component overlay for the MIDI interface

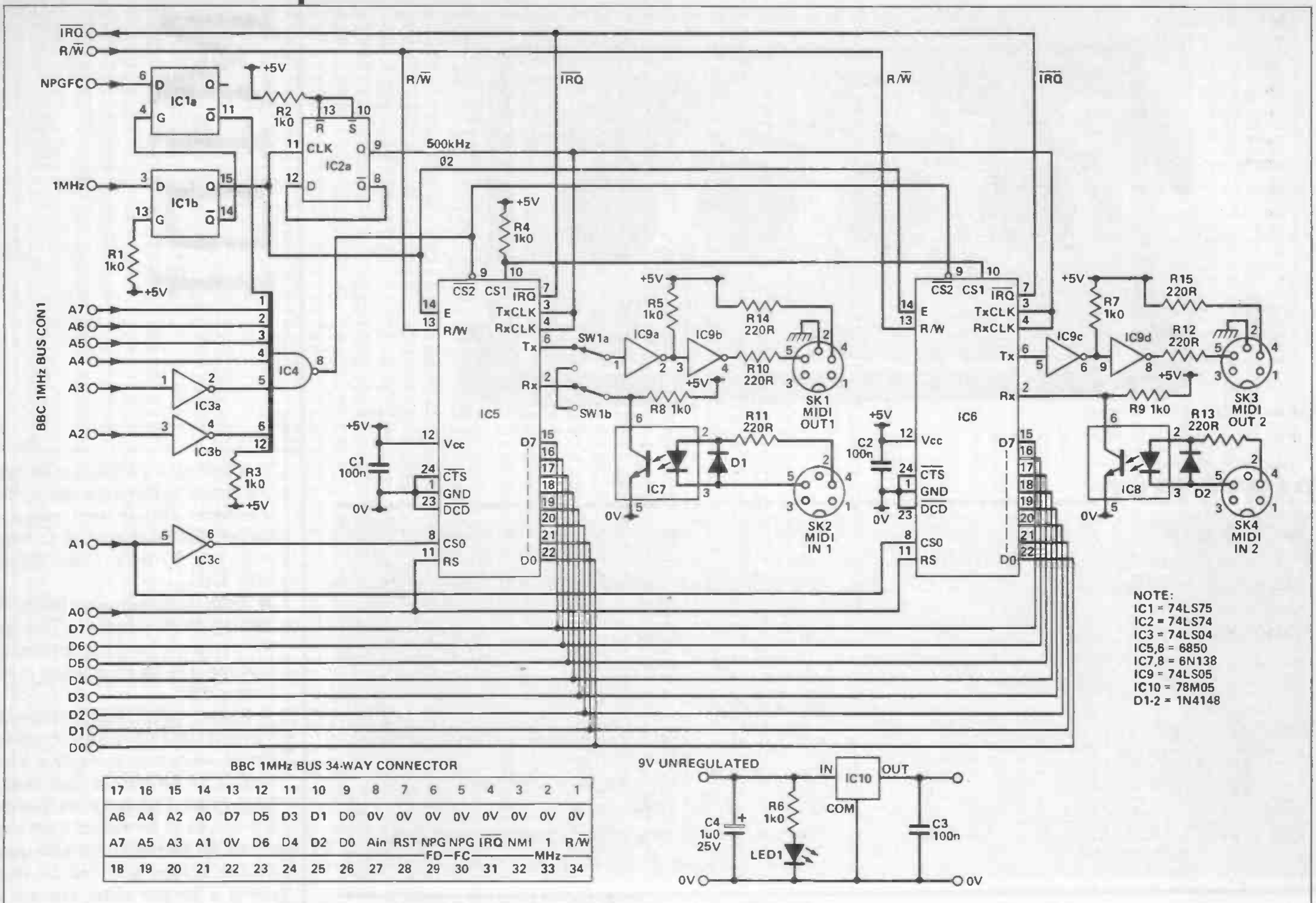
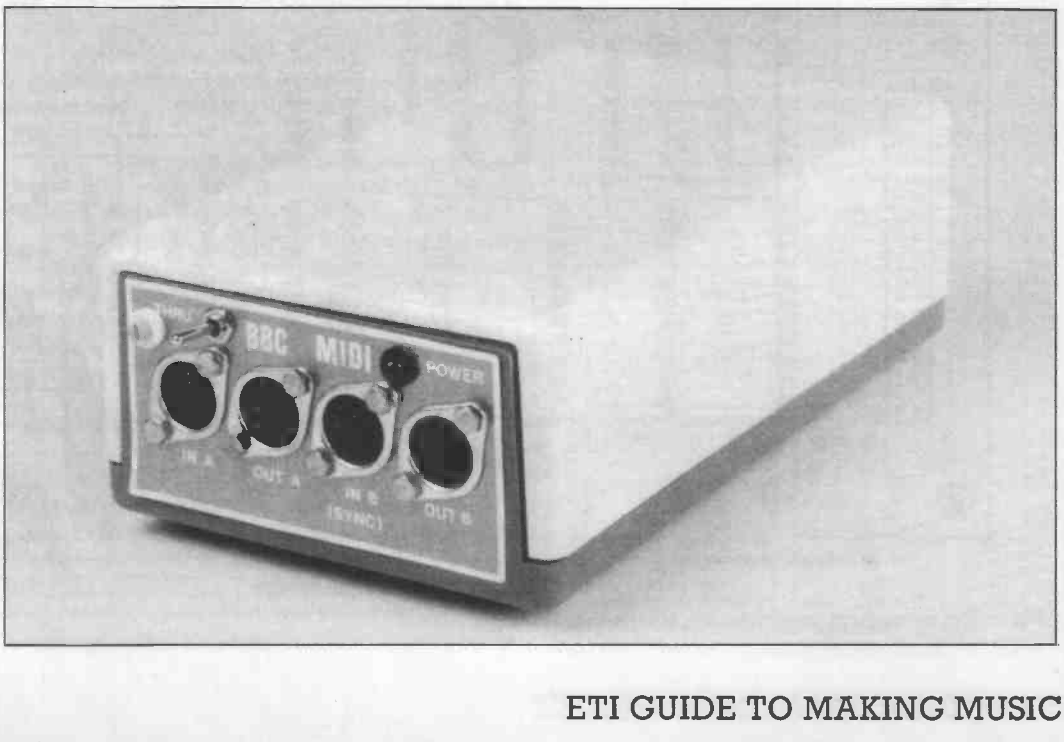


Fig. 4 The circuit diagram of the MIDI interface

HOW IT WORKS

Figure 3 shows the complete MIDI interface circuit. The circuitry is centered around IC5 and IC6, which are 6850 ACIAs (Asynchronous Communication Interface Adaptors). The two devices are interfaced to the BBC micro's 1MHz expansion bus using IC1, IC3 and IC4. IC1 is used to 'clean up' the NPGFC bus signal as recommended by Acorn in the applications notes for using the 1MHz bus. IC3 and IC4 provide the address decoding necessary to map the two ACIAs into memory locations & FCF0-1 and &FCF2-3 respectively in the BBC micro's address space. One of the bistables in IC2 is used to divide the BBC micro's 1MHz system clock down to the 500kHz required for the ACIAs. The remaining circuitry built around IC7, IC8 and IC9 make up the MIDI IN and MIDI OUT terminals. The open collector inverters act as current sinks, forming the MIDI OUT terminals. The MIDI IN terminals reach the ACIAs via opto-isolators in order to minimise the risk of earth loops occurring resulting from interconnection of MIDI equipment.



Basic INKEY keyword is used to detect whether a particular key is held down or not. If pressed, the procedure PROCnoteon is called. This transmits the three data bytes of a MIDI *note on* event. The program then waits until the key is released, when it transmits a *note off* event. This very simple program only permits monophonic playing — only one note at a time. However it serves to demonstrate *note on* and *note off* events, which are the most frequently used MIDI messages as far as synthesisers are concerned.

Applications

Applications for the MIDI interface depend on what MIDI equipment it is to be used in conjunction with. The most immediate application is to use the BBC micro's disc filing system to store synthesiser voice programs or drum machine track programs. More complex software to use the MIDI communication protocols found in the data manuals for the equipment concerned can also be written to make full use of the interface.

For synthesisers such as the Yamaha DX7 and its derivatives, it is possible to write a voice editor to program new voices from the BBC micro. When programming a new sound on the DX7 only one parameter can be accessed at any one time. With a voice editor program all the parameters can be seen on the screen at once and freely accessed.

Voice editing from the BBC micro, along with the advantages of its disk filing system, forms a powerful sound management system for the DX7.

A suite of more serious software to accompany this project is available from the author. This includes general MIDI utilities programs, voice and track dump software for the Yamaha TX/DX7 and Roland TR707 and a voice editor for the TX/DX7.

Thru Switch

The THRU toggle switch shown on the front panel of the prototype unit serves to link the MIDI IN 1 to the MIDI OUT 1 for use when the BBC micro is not running software which actively merges the MIDI IN 1 input stream. The switch simply links pin 6 of IC7 to pin 1 of IC9 while isolating them from their original connections to Tx and Rx of IC5. This arrangement allows data to pass unaltered from MIDI IN 1 to MIDI OUT 1, even when the BBC micro is switched off. The THRU option was added as an afterthought and the PCB does not include this feature but it can be easily included with the minimum of alterations if this feature is required.

```
   10 REM BBC micro MIDI interface
   20 REM Receiver Program by J.F.S.Yau
  100 MODE7
  110 PROCassem
  120 FOR I%=0 TO 1:VDU132:VDU157:VDU141:VDU131
 :PRINTSPC(6)"MIDI Receiver Program":NEXT
  130 VDU28,1,24,39,3
  140 CALL start%
  150 REPEAT
  160   IF ?outptr%=?inptr% GOTO 210
  170   data%=buff%?(?outptr%)
  180   IF (data% AND 128)<>0 THEN PRINT
  190   PRINT data%;
  200   ?outptr%=?outptr%+1
  210   UNTIL FALSE
  220 END
 1000 DEFPROCassem
 1010 DIM start% 100,buff% 256
 1020 @%=4:sysvec%=&70:irq2v%=&0204
 1030 inptr%=&72:outptr%=&73
 1040 ACIA_C%=&FCF0:ACIA_D%=&FCF1
 1050 FOR pass%=0 TO 2 STEP 2
 1060   P%=start%:[OPT pass%:SEI
 1070   LDA irq2v%:STA sysvec%
 1080   LDA irq2v%+1:STA sysvec%+1
 1090   LDA £acia% MOD 256:STA irq2v%
 1100   LDA £acia% DIV 256:STA irq2v%+1
 1110   LDA £&03:STA ACIA_C%:STA &FCF2
 1120   LDA £&95:STA ACIA_C%:STA &FCF3
 1130   LDA £0:STA inptr%:STA outptr%
 1140   CLI:RTS
 1150   .acia%
 1160   PHA:TYA:PHA:PHP
 1170   LDA ACIA_C%:AND £128:BEQ exit%
 1180   LDY inptr%:LDA ACIA_D%:STA buff%,Y
 1190   INC inptr%
 1200   .exit%
 1210   PLP:PLA:TAY:PLA:JMP (sysvec%):]
 1220   NEXT pass%
 1230 ENDPROC
```

Listing 1: The BBC Basic MIDI receiver program

```
   10 REM BBC micro MIDI interface
   20 REM key test program by J.F.S.Yau
  100 DIM N% 23
  110 *FX11
  120 PROCinit:OCT%=4:L%=0
  130 ACIA_C%=&FCF0:ACIA_D%=&FCF1
  140 ?ACIA_C%=&03:?ACIA_C%=&15
  150 D$=CHR$9+"Q2W3ER5T6Y7UI9OOP@^[\_"
  160 REPEAT
  170   PROCnoteoff(L%)
  180   REPEAT:J$=INKEY$(0):UNTIL J$<>""
  190   K%=INSTR(D$,J$)
  200   IF K%=0 GOTO 180
  210   L%=K%+OCT%*12:PROCnoteon(L%)
  220   PRINT'"Note ";L%;"played"
  230   IF INKEY(-N%?K%) THEN 230
  240   UNTIL FALSE
 1000 DEFPROCinit
 1010 FORI%=1TO23:READ A%:N%?I%=A%:NEXT
 1020 ENDPROC
 1030 DEFPROCwaitTx
 1040 REPEAT:UNTIL ?ACIA_C% AND &02:ENDPROC
 1050 DEFPROCnoteoff(NT%)
 1060 PROCwaitTx:?ACIA_D%=144:PROCwaitTx:?ACIA_
 D%=NT%
 1070 PROCwaitTx:?ACIA_D%=0:ENDPROC
 1080 DEFPROCnoteon(NT%)
 1090 PROCwaitTx:?ACIA_D%=144:PROCwaitTx:?ACIA_
 D%=NT%
 1100 PROCwaitTx:?ACIA_D%=100:ENDPROC
 1110 DATA 97,17,50,34,18,35,52,20,36,53,69,37
 1120 DATA 54,38,39,55,40,56,72,25,57,121,41
```

Listing 2: The program for playing MIDI instruments

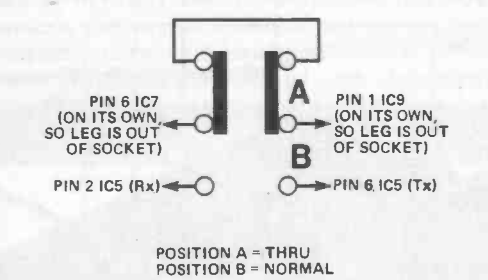
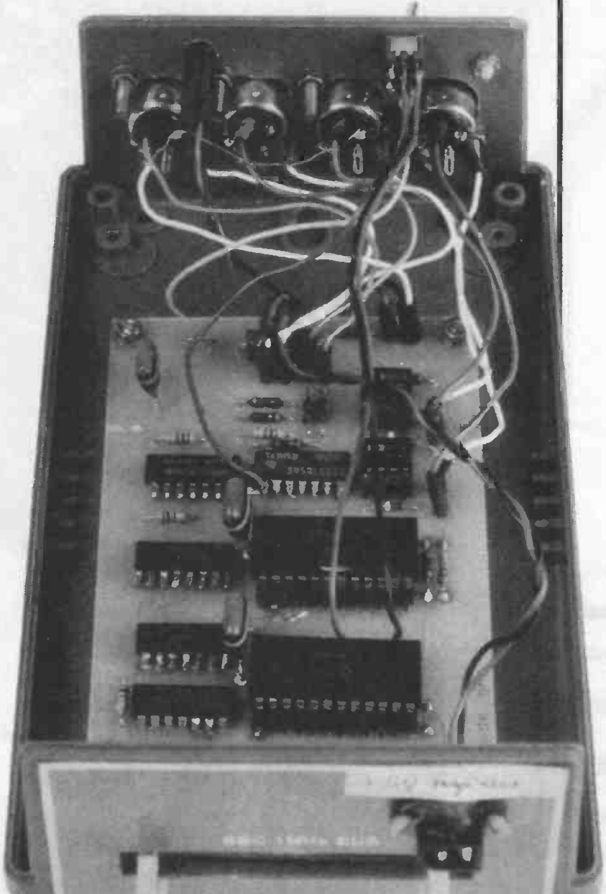


Fig. 5 Wiring the optional 'THRU' switch



MIDI MASTER KEYBOARD

This high specification master keyboard is a major music project

The Musical Instrument Digital Interface (MIDI) is now the universally accepted standard for communication between synthesisers, drum machines, music computers and other musical peripherals. Probably the most common MIDI application is to link synthesisers together in a way that permits a single keyboard to play all the other units attached to it via MIDI, as shown in Fig 1.

This project is a six octave velocity sensitive keyboard designed to be a central controller for any number of connected MIDI synthesisers.

MIDI information transmitted by the keyboard controller includes note on/off events, program changes, pedal hold on/off and also pitch bend and modulation information, more details of which will be given later.

A full description of the MIDI standard was given in *Unmuddled MIDI*. Using MIDI, a single synthesiser keyboard can independently select up to 16 separate slave instruments. In these systems where a master mother keyboard controls other instruments, the slave instruments may be just MIDI equipped sound modules and need not have keyboards at all. Moreover the mother keyboard itself does not need to have the capability of generating sound of its own.

More and more synthesisers are being made available in the form of a black box equipped with a MIDI interface without a keyboard, for incorporation into a master mother keyboard system. Examples of such products are Yamaha's TX81Z and the rack-mounting keyboardless versions of the Ensoniq Mirage and its successors.

The project presented here is a MIDI master keyboard that is suitable for controlling slave instruments in this way. The actual keyboard unit itself does not have its own sound generation hardware but is capable of full MIDI control of any external connected device.

Specifications

For maximum flexibility the 72-note keyboard is fully touch sensitive. A uniquie feature of the system is that the touch sensitivity of the keyboard can be varied over a range of 16 different preset settings. This means you have complete control of the dynamic range of the synthesiser module direct from the mother keyboard. For synthesisers that do not feature touch sensitivity, the velocity information transmitted by the mother keyboard is simply ignored.

Most MIDI synthesisers have performance controls which are accessible via MIDI. To accommodate this the mother keyboard system has two joystick

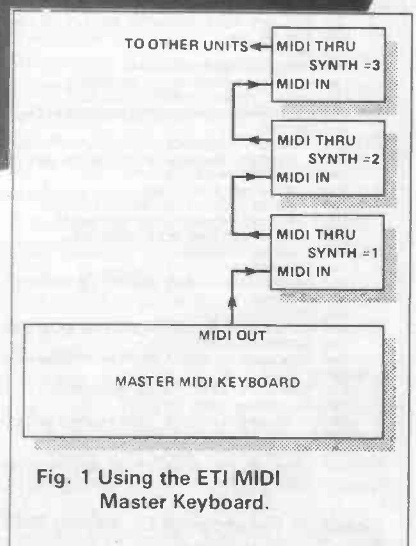


Fig. 1 Using the ETI MIDI Master Keyboard.

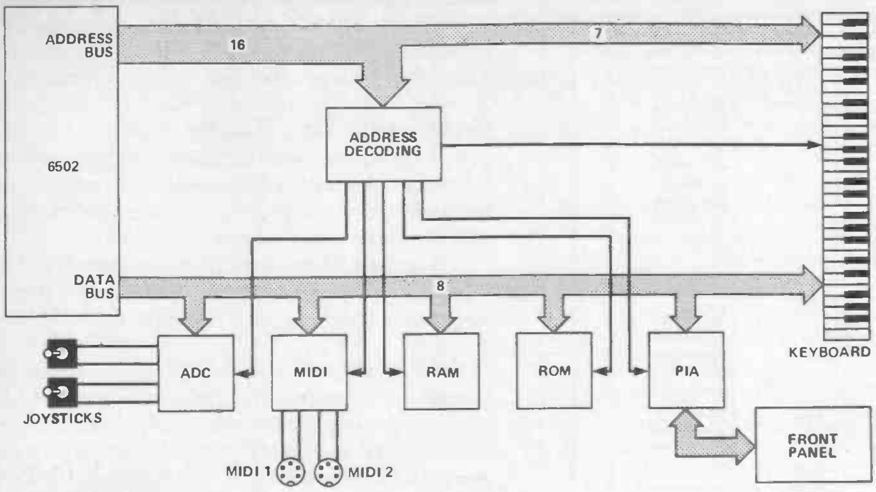


Fig. 2 The MIDI Master Keyboard

controls and three footswitch inputs. The joysticks enable the system to transmit MIDI information relating to pitch bend and modulation. The footswitches enable sustain pedal hold, portamento switch and program advance information to be sent.

MIDI information is transmitted in two output streams, called channel A and channel B. The default setting is that channel A transmits on MIDI channel number one and channel B transmits on channel number two. However, the output streams can be independently assigned to any of the possible 16 MIDI channels.

The main purpose of having two MIDI output streams is to facilitate a split point on the keyboard. A split point can be programmed to be anywhere on the keyboard. Any notes played to the left of (and including) the split point key are sent to the MIDI channel assigned to the channel A output stream while notes to the right of the split point are assigned to channel B.

Using this facility two separate synthesiser modules can both be played from the mother keyboard with one synthesiser assigned to either side of the split point.

The transpose feature in the master keyboard allows note information to be transposed before it is transmitted. So it is possible to play in a fixed key in relation to the physical keyboard while the actual notes played and transmitted through MIDI can be of any key signature.

This facility is especially useful when playing keyboards to accompany singers. The music can be instantly transposed at a touch of a button to suit the singer's range.

An additional feature incorporated into the mother keyboard system is the ability to independently transpose the keyboard at either side of a programmed split point. The main use of this feature is to transpose voices independently over octave ranges. Although one does not relish the thought of two synthesiser modules being played at different key signatures the possibility is there!

The MIDI Master Keyboard project enables the music enthusiast to acquire a high specification MIDI mother keyboard at low cost. Some time and patience is required for the construction, especially within the mechanical construction side. However, the level of finish is up to the individual. You may be content with the keyboard nailed onto a wooden base (the prototype remained in that state for quite a spell!). Alternatively, a professional style cabinet finish may be easily achieved.

Hardware Overview

There are a total of six printed circuit boards making up the guts of the hardware for the ETI MIDI Master Keyboard. This may seem a lot but some consolation can be drawn from the fact that three of them (the keyswitch PCBs) are identical. These three boards are mounted along the length of the keyboard and serve the dual purpose of holding the CMOS multiplexer circuitry and providing the mounting base for the keyswitch springs.

Of all the PCBs the only double sided one is the main CPU board. This board holds the 6502A processor circuitry and all its associated peripheral devices as well as the analogue parts of the pitch bend and modulation joystick circuitry. The fifth board (the front panel board) holds all LED display circuitry and also serves as a mounting base for the push button switches that form the data entry keypad. Last but not least is the power supply PCB, which provides the necessary power rails for all the boards.

The heart of the system lies within the CPU board as shown in Fig. 2. All the necessary functions for the operation of the MIDI keyboard are directed by the onboard 6502A processor running at 2MHz. Physical tasks to be performed by the processor include scanning the keyboard, scanning the data entry keypad and footswitches, reading the joystick

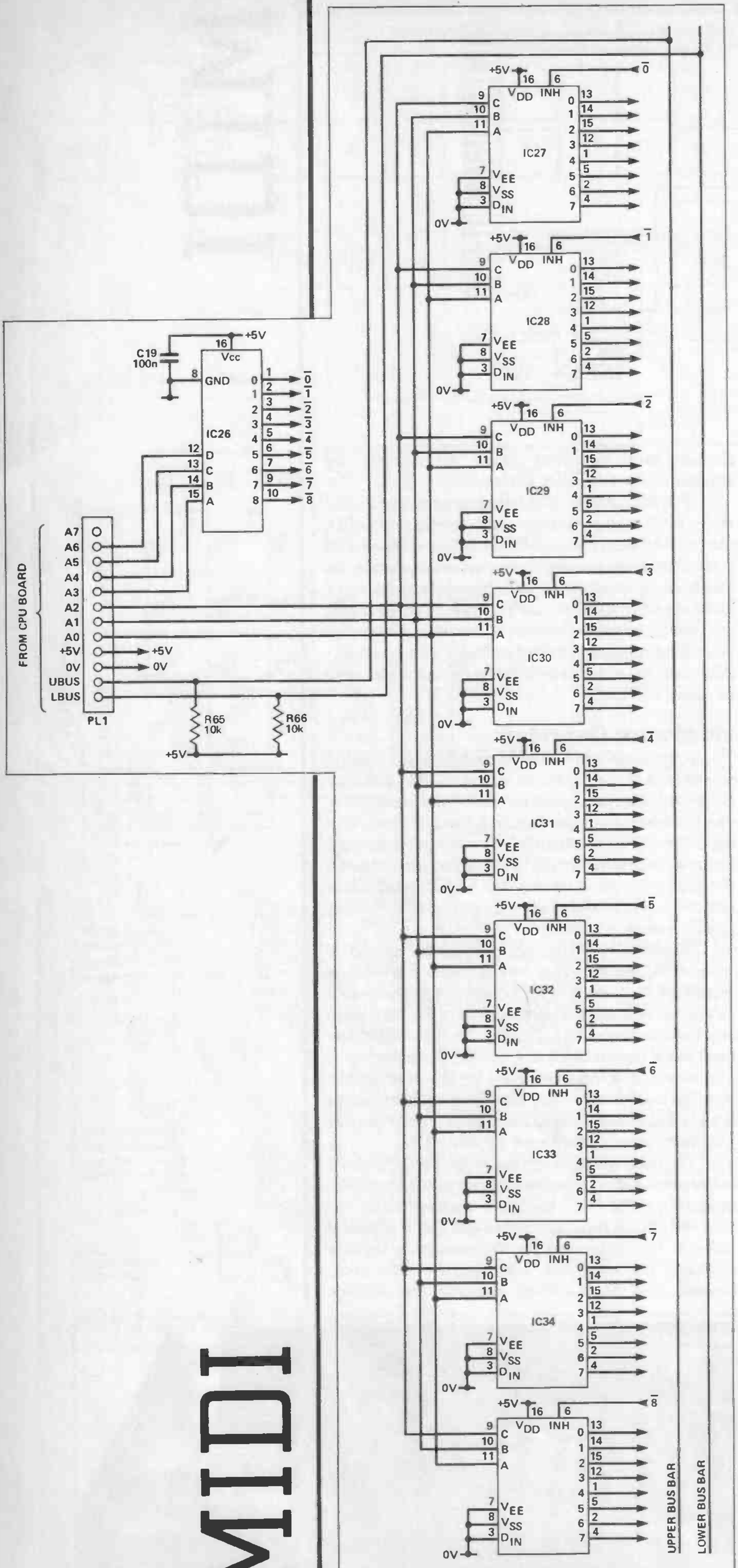


Fig. 3 The keyboard multiplexer circuit

positions, setting the current LED display and finally transmission of MIDI data. All these tasks require a substantial number of peripheral devices to be placed within the address space of the 6502A processor.

Scanning The Keyboard

Perhaps the most critical task performed by the 6502 is the scanning of the keyboard. Each note on the keyboard can be in any one of three possible states pressed, not pressed and in transition.

Detection of the transition state is required for velocity sensing and necessitates a two bus bar system as shown in the diagram of the keyboard mechanics in Fig. 5.

When a note is in neutral or unpressed state its contact spring touches the upper bar. Pressing a note causes the plunger to force the contact spring against the lower bus bar. When in the transition state the contact spring is touching neither the upper nor lower bus bar.

The software examines each key at precise time intervals of 2ms. Velocity sensing is achieved by timing how long a key takes to transverse from the upper to lower bus bar when pressed, by counting the number of discrete time intervals that the key spends in the transition state.

For the moment we will concentrate on the hardware aspect of the keyboard scanning with the construction of the keyboard and keyswitch PCBs.

Construction Of The Keyboard

The keyswitch PCBs should be made up and incorporated into the keyboard mechanics first. Thereafter the CPU board, the front panel board and the power supply can be assembled separately and wired into the keyboard unit to make up the complete system.

Keyswitch Mounting

Keyboard mechanics come in the form of 72 plastic notes which are mounted on a steel plate chassis. Each note pivots on a flange protrusion from the chassis and has a return spring at the back of the note, pulling it towards the chassis (Fig. 5).

Attached to the underside of each note is a nylon plunger tab which, when the note is pressed, causes the contact spring for that note to move from the upper bus bar and come into contact with the lower bus bar, as shown.

HOW IT WORKS: KEYBOARD

Figure 3 shows the circuit of the three keyswitch PCBs mounted along the length of the keyboard chassis.

Both the upper and lower bus bars are tied to +5V via pull up resistors R65 and R66. IC27-35 are 4051 CMOS 8 channel multiplexer devices.

Each device has its data input tied to 0V. The three bit address input presented at pins 9, 10, 11 routes the data input (0V) to one of the eight outputs. However, at any time only one of the CMOS multiplexer devices is selected to be active by IC26, a 4-to-16 decoder (of which only nine outputs are actually used).

With a 7-bit address (four bits for the 4-to-16 decoder, IC26, and three bits for the CMOS mulitplexers, IC27-35) a 0V signal can be routed to any one of the 72 keyswitch contact springs that are electrically connected to the CMOS multiplexer outputs.

To examine the state of a key, all that has to be done is to supply the 7-bit address of the key and then read the state of the upper and lower bus bars. If the key were idle the contact spring would be touching the upper bus bar, thereby pulling its potential to 0V (actually not quite 0V due to the resistance of the CMOS switch). Similarly, a key depression would ground the lower bus bar. In the transition state both bus bars would remain at 5V potential.

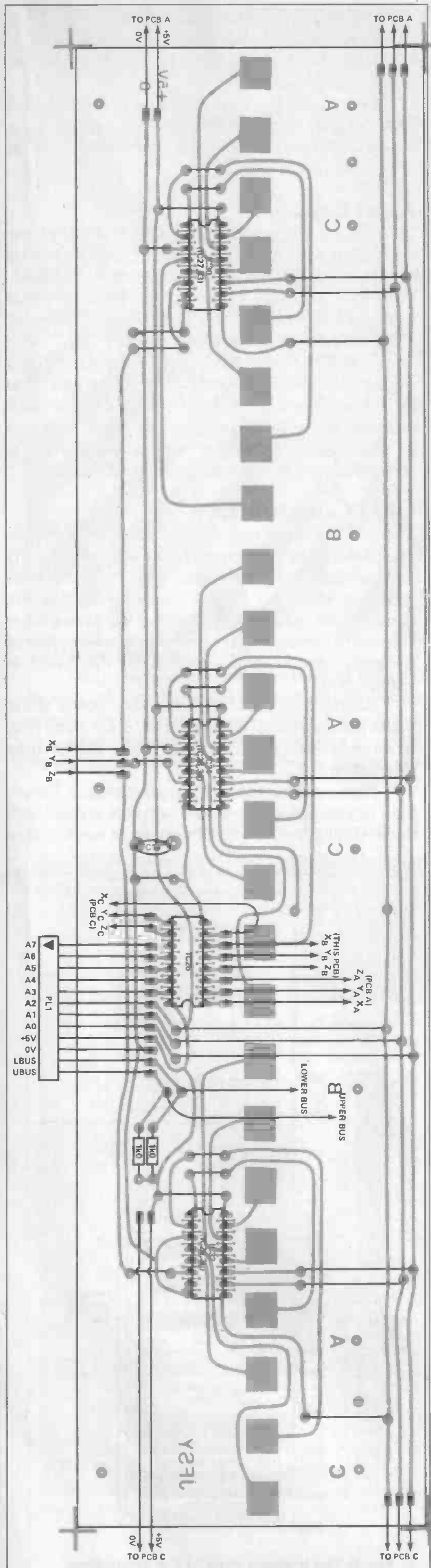


Fig. 4 The component overlay for the keyboard PCBs

Start with the assembly of the three keyswitch PCBs. Although all three boards are identical the component layout for the middle board (called board B and shown in Fig. 4) is slightly different because of the inclusion of IC26 (the 74LS42 decoder) and the bus bar pull up resistors R65 and R66.

All the jump links should be kept identical on all boards for simplicity's sake.

It is important to note that the contact springs are *not* to be soldered in at this point. It is better to mount them after the boards have been bolted onto the chassis of the keyboard, in order to ensure the best alignment.

A 12-way cable should be fitted to board B around IC26 and terminated with a female PCB multiway connector. Cable length should be about 50cm — long enough to reach the CPU board. The 12-way cable should be soldered directly to board B since there is no room for a multi-way connector when the board is mounted component side facing the chassis with only about 8mm clearance.

After all the components and links have been mounted, the next task is to mount the upper and lower bus bars. These are gold plated contact rods which run parallel to each other along the whole length of the keyboard. Ready drilled plastic mounting blocks were used. These have two holes spaced a short distance apart in which the contact rods are mounted and a larger single hole for mounting onto the PCB.

Use eight mounting blocks and insert the rods through all of them and space them roughly equally apart. On board A, drill holes marked by the letter A, drill the B holes on board B and C on board C. Adjust the mounting blocks so they are positioned above these holes and use suitably sized bolts to fasten them to the boards.

The three boards are now held together by the parallel bus bar assembly. At this stage, link the three PCB tracks which run parallel along the edge of the boards so that they continue throughout the span of the three boards. Also link the power rail terminations so that they span all three boards.

Wire the bus bars by soldering two wires from appropriate points on the bus bars to the take-off points on board B (lightly tin a small area on each bus bar before soldering the wires). Finally, wire the nine decoder outputs from IC26 to the appropriate terminals on the three PCBs as shown in the component overlay diagram.

Some keyboards are supplied already assembled and mounted on a steel chassis. Buyers of such keyboards should skip the next section.

To mount the keyswitch PCB assembly, drill holes in the keyboard chassis and use a combination of nuts, bolts, washers and nylon spacers to firmly fix

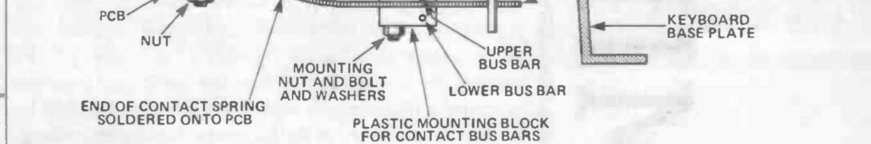
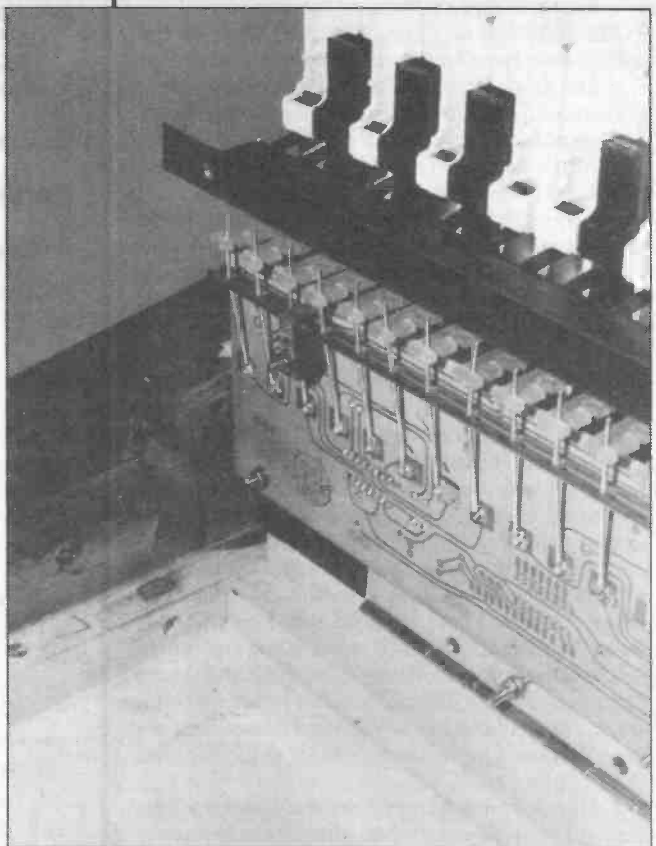


Fig. 5 The keyboard mechanics



the three PCBs in place. The stand-off height of the PCBs is important since it affects the keyswitch action. In the prototype, the stand-off height was about 8mm (see Fig. 5 for details of assembly) but a different height may be required depending on the dimensions of the keyboard mechanics.

It is essential that the stand-off height should be such that when a contact spring is soldered onto the PCB pad and threaded through the lower hole in the nylon plunger (having passed between the bus bars) it is in contact with the upper bus bar with slight tension when the key is neutral and in contact with the lower bus bar when pressed. In order to achieve the optimum position, it may be necessary to fine adjust the height by adding or removing washers.

When mounting the boards onto the chassis some notes on the keyboard will have to be temporarily removed in order to access the other side of the steel chassis to tighten up the bolts.

Before finally tightening up the bolts make sure that the boards are positioned squarely so that the bus bar edge lies perfectly parallel to the row of nylon plungers in the keyboard mechanics.

The final stage in the keyswitch assembly is to mount the contact springs themselves. When handling the contact springs only pick them up by the bulbous end, otherwise sweat or dirt from your hands can easily impair the spring's conductivity.

Take each spring and tin the bulbous end very lightly with solder. Act very quickly when doing this to avoid flux creeping along the length of the spring and making it stiff.

Tin the copper pad on the PCB and position the spring, by passing it between the bus bars and threading it through the nylon plunger. Find the best position to solder the spring within the copper pad by testing the key action. When the optimum position is found a mere touch with the soldering iron will be all that is necessary to fix the contact spring into place. Check the spring is squarely in position and that the key action is correct.

Repeat the procedure for the rest of the springs until all 72 notes are completed (see you tomorrow...). When complete, check the overall keyboard action for correct operation.

That completes the construction of the keyboard itself. Now we can move on to the major part of the MIDI Master Keyboard — the controlling CPU and its associated electronics.

Total Control

The 6502A CPU micro-processor looks after the entire keyboard. The majority of the circuitry is on two boards — the CPU board and the Front Panel board. The latter is largely a base for all the front panel switches and displays. The CPU board is the most complex board and is a double sided PCB.

The MIDI Master Keyboard is implemented in a simple 6502 memory map (Fig. 6). All the interfaces for the keyboard itself, the front panel, the MIDI interfaces and so on are mapped into specific locations in the memory map and the whole system is controlled by a monitor program in ROM.

CPU Construction

The first task is to insert and solder all the through-pins. Refer to the component layout diagram (Fig. 11) and use double sided veropins. Make sure that *both* sides are soldered. Making doubly sure that the through-pins are correctly soldered will save a lot of time and frustration later on. It only takes one side of a through-pin to be overlooked for the board to malfunction.

Solder in all the IC sockets next. Some of the tracks between the IC pins on the PCB are rather fine, so be careful not to accidentally bridge any of them with solder.

Next, solder in the multi-way connectors. Ensure their orientation is correct and keep the contact with the soldering iron to a minimum, so as not to soften

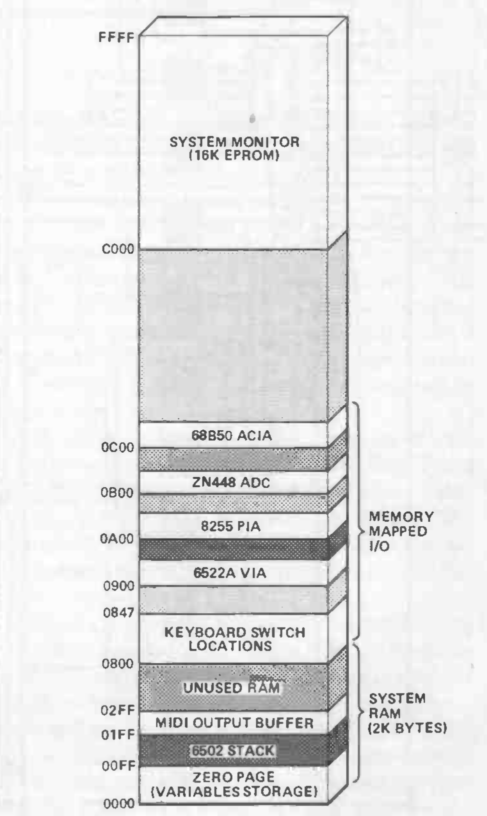
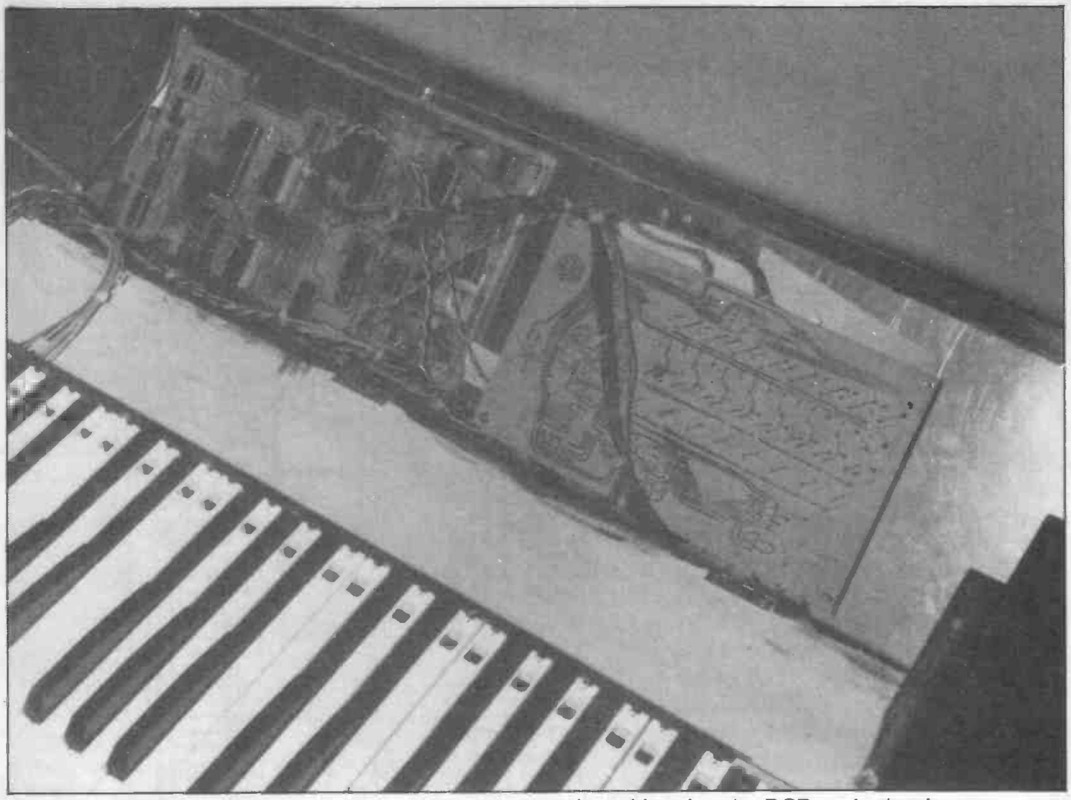


Fig. 6 The memory map of the controlling microprocessor



the plastic and bend the pins.

The other components can then be mounted, starting off with the resistors, followed by the capacitors, diode and presets. Make sure that the orientation is correct for the electrolytics and diode. Note that a few of the components require their leads to be soldered to the PCB on *both* sides.

Finally, insert all the integrated circuits and, as always, ensure correct orientation. Finish off by cleaning the board with flux remover and check thoroughly for any shorts which may be caused by stray blobs of solder.

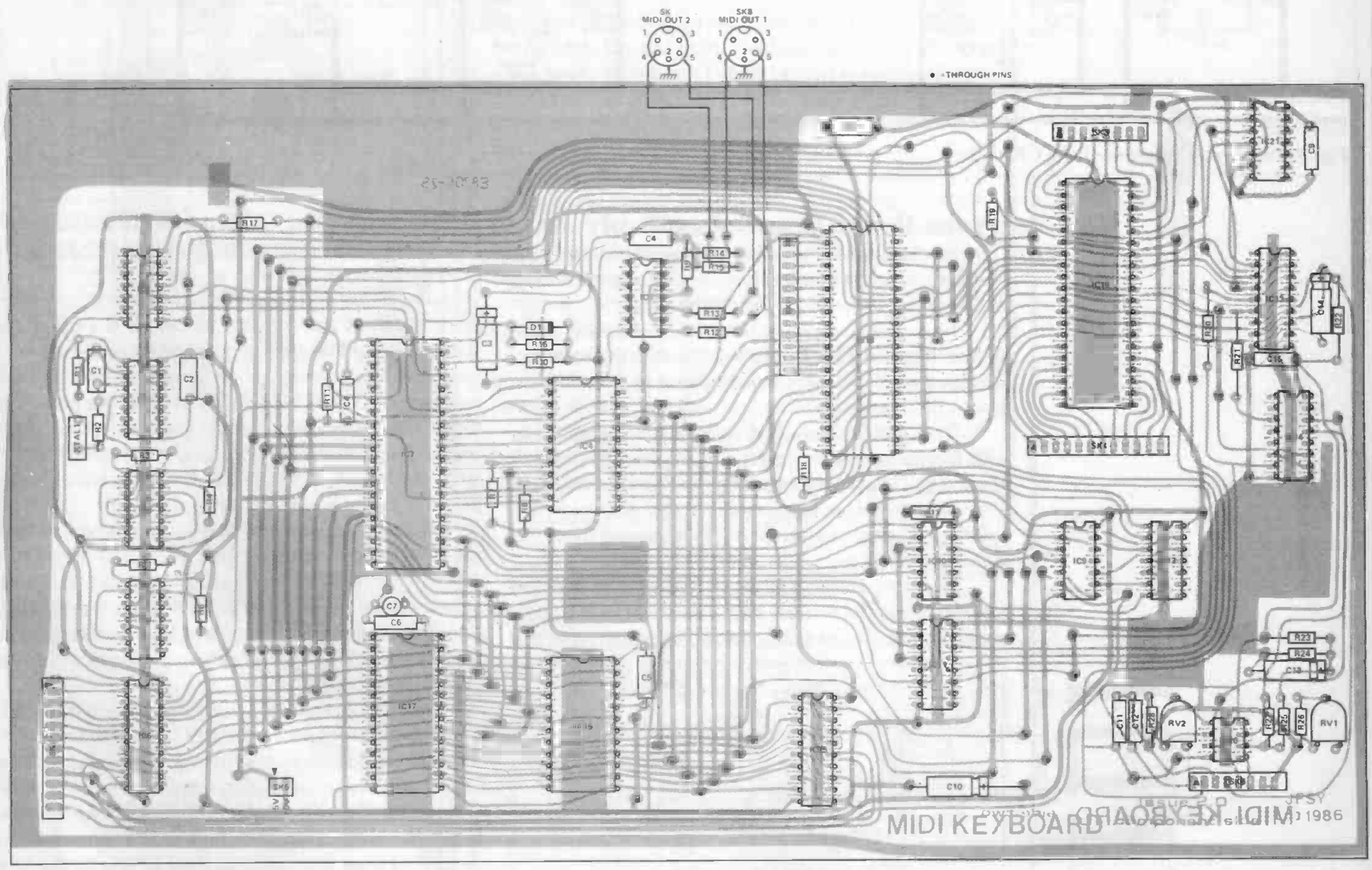


Fig. 7 The component overlay for the CPU board

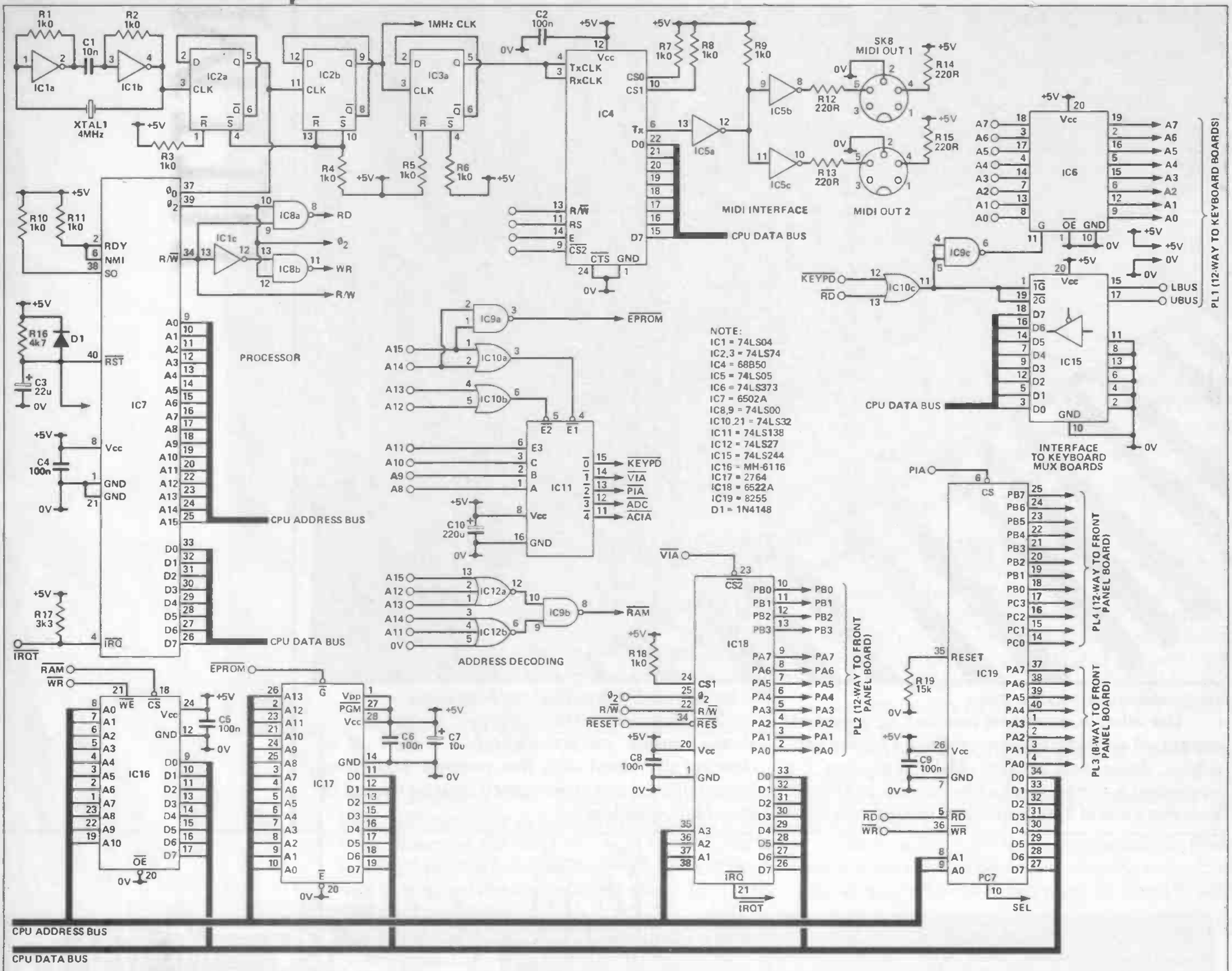


Fig. 8 The circuit diagram of the CPU board

Front Panel Board Assembly

The front panel PCB holds the display driver and display circuitry as well as functioning as a mounting base for the bank of push buttons.

First of all, solder in the links as shown in the layout diagram (Fig. 10). Follow this with the resistors, diodes and IC sockets, making sure the diodes are correctly orientated.

The dual 7-segment display plugs into a 24-pin DIL IC socket mounted on the PCB. Since the display device package only has 18 pins, be sure that it is plugged into the socket correctly. It should occupy the leftmost position when the PCB is viewed from the component side.

The next components to be added should be the push buttons. To achieve best alignment of the push button bank, the switches should be soldered to the PCB fully assembled, complete with the key cap tops. Solder the push buttons one by one, making sure the positioning is straight and square.

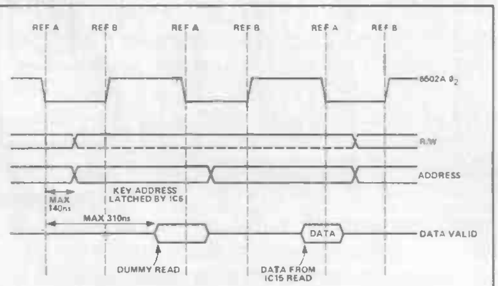


Fig. 9 Timing diagram for the keyboard scan

There are three wiring harnesses which span from the front panel board to the CPU PCB. Each harness should be soldered directly onto the relevant pins and terminated with a female multi-way PCB connector. As with the keyswitch PCBs, the wiring has to be directly soldered onto the front panel PCB (rather than using the multi-way connectors) due to lack of clearance when the board is mounted in position in the keyboard cabinet.

Follow the component layout diagrams of both the front panel board and the CPU board (Figs 7 and 10) to ensure correct connector orientation. Cable length should be such that the wiring reaches the CPU board easily from the front panel board when the two boards are in their final mounting positions within the keyboard cabinet.

After plugging in ICs, all that remains is the wiring of the two harnesses for the LEDs. Wiring should be direct to the board or via veropins. Be careful with the LEDs' orientation as its very easy to get one of them wrong (see Fig. 13). Use solid insulated wire for the anode connection to give the LEDs a more definite mounting base and position the LEDs with the final front panel positions in mind (see Fig. 11).

That completes the construction of the CPU and front panel boards. Break out the beer and have a rest. Then we'll move on to the power supply.

HOW IT WORKS: CPU

Figure 8 shows the circuit diagram for the CPU board. C3, R16 and D1 form a simple power up reset circuit that is used by the 6502A and also the 6522A VIA. IC1a and IC1b form a 4MHz crystal oscillator. IC2a divides the latter signal by two to form the 2MHz clock for the 6520A. Further division of the 2MHz clock signal is performed by flip flops IC2b and IC3a to obtain the 31.125kHz clock signal used by IC4, the 68B50 ACIA.

Address decoding for the CPU is performed by IC9, IC10, IC11 and IC12. The address decoding enables all the peripheral devices to be memory mapped within the 6502A's address space.

The monitor program for the processor resides permanently in EPROM, the device used being a 16K 27128 EPROM (IC17). A 6116 2K CMOS RAM chip provides the required RAM workspace. Since the size of the monitor program is only about 2.5K of 6502 code, using a 16K EPROM may seem a bit of an overkill. However, the larger EPROM was chosen to give scope for expansion (such as storage of voice dumps for particular synthesisers) and also because of the relatively small price difference when compared to smaller devices.

6522A VIA Functions

IC18 is a 2MHz version of the 6522 Versatile Interface Adaptor (VIA), the 6522A. The device takes up 16 locations in the 6502's memory address space and these are accessed when using the device's dual 8-bit ports and on-board timer. The 6522A is configured to perform two functions.

Scanning of the data entry and function push button keypad is achieved by the two I/O ports. The 6522A's timer is responsible for generating the clock signal which interrupts the 6502A processor every 2ms. Virtually all the system software is interrupt driven, due to the demands of scanning the keyboard at the precise time intervals required for the key velocity sensing.

8255 PIA Functions

IC19 is an 8255 Peripheral Interface Adaptor (PIA) interfaced to the 6502A through four memory mapped locations. These addresses are the 8255's three 8-bit I/O ports and its control register.

The RD and WR signals for the device are derived from the 6502A's Phi-2 clock and R/W lines using IC1c, IC8a and IC8b. In use, the PIA is configured simply as three 8 bit output ports which are used to drive the LED displays of the MIDI keyboard.

In order to maximise speed and efficiency in the key scanning each key address is made to appear as a direct memory location in the 6502's address space (from 0800H to 0847H). The timing diagram for accessing the state of each key is shown in Fig. 9.

Two successive memory read cycles are required, the first being a dummy read that causes the address of the key to be latched at IC6, a 74LS373 octal latch. As there are only 72 keys to be addressed the most significant bit of the latched address will always be zero.

By the time the following read cycle occurs the two bit word describing the state of the key will be valid at the inputs of IC15, a 74LS244 tri-state octal latch whose output is connected to the processor data bus. It is necessary to latch the key address rather than to engage in a single read cycle because the CMOS multiplexer device outputs are unable to settle within the maximum 310ns access time of a single 6502A read cycle.

The MIDI Interface

Transmission of MIDI messages is achieved by IC4, a 68B50 Asynchronous Communications Interface Adaptor (ACIA). Note the device has to be a 'B' version in order to be compatible with the 2MHz 6502A. Only two memory mapped locations are required for full communications with the device.

When a single byte of MIDI data is to be transmitted it is simply written to the 68B50's transmit data register by the 6502 processor. The 6502 can then resume its other tasks whilst the ACIA has the job of converting the parallel data to a serial output of one start bit, eight data bits and one stop bit at a baud rate of 31.125K – the standard MIDI data configuration.

Inverters IC5a, b and c are of the open collector type and are used to form the current loop required for the serial link with an external MIDI device. Both MIDI OUTs are identical, the dual output may be useful in avoiding having a large number of MIDI devices being daisy chained from a single MIDI OUT. Using two MIDI OUTs in a star configuration greatly reduces the inherent delays in MIDI data propagating itself through the MIDI devices.

The receiver part of the ACIA is not used, since the keyboard does not make any use of any received MIDI data.

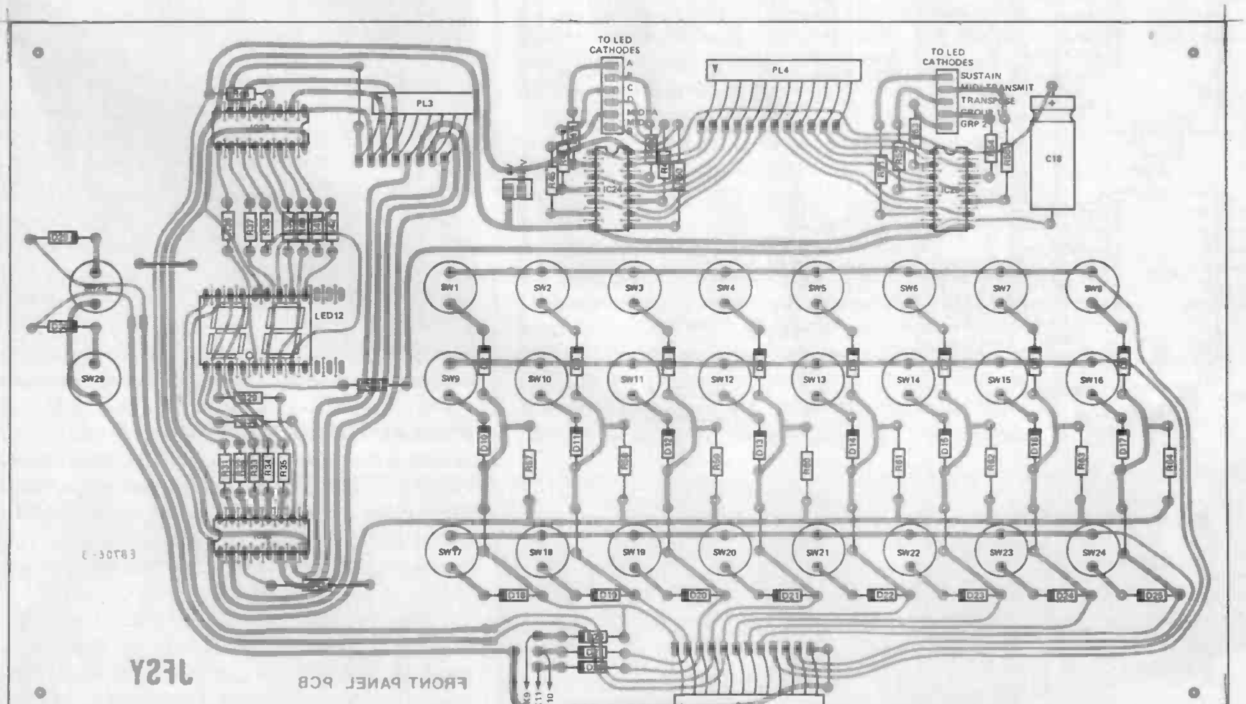


Fig. 10 The component overlay for the front panel board

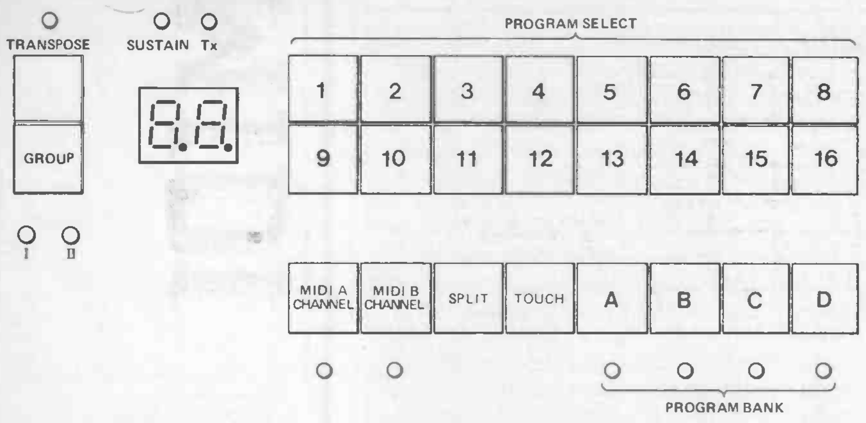


Fig. 11 The front panel layout

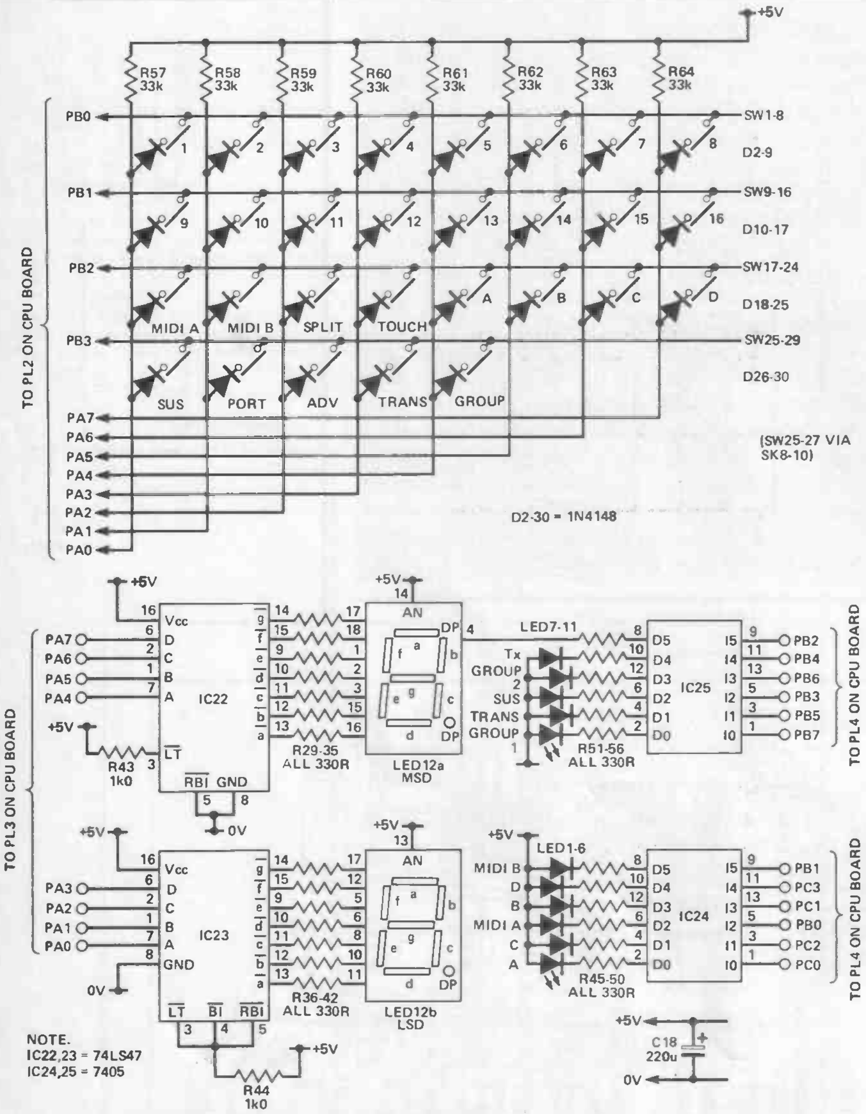
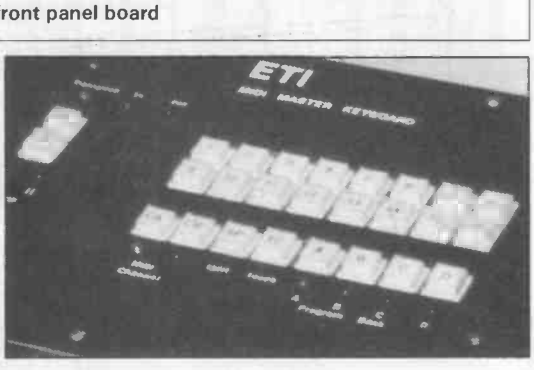


Fig. 12 The circuit diagram for the front panel board



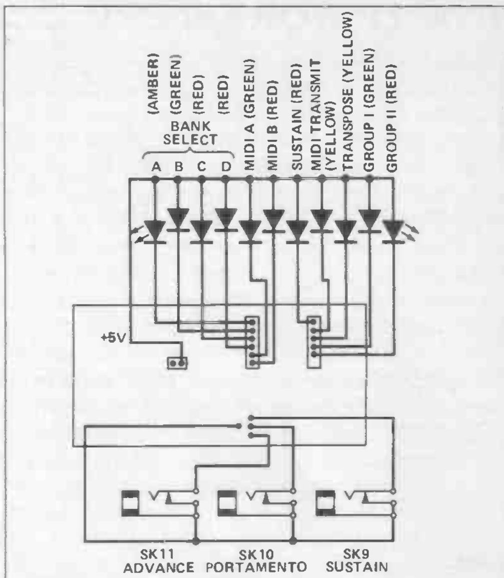


Fig. 13 Wiring the LED indicators and the footswitch sockets

HOW IT WORKS: FRONT PANEL AND PSU

Figure 12 shows the circuit of the front panel board.

Scanning of the push button keypad is achieved by the diode matrix circuit comprising D2-30, SW1-29 and R57-64. Port A of the 6522A VIA is configured to be an input port, and the low nibble of port B is configured for output. By reading the value of the data presented at port A, whilst one of the switch matrix rows is pulled low (strobed) by the relevant bit in port B, the state of all the switches in that row can be examined.

For example, if the binary number is 11111101, then it can be deduced that the numeric button '4' was being pressed.

Such circuits are commonly found in 'QWERTY' keyboards, but there are problems using this type of circuit for scanning the music keyboard as well. Full polyphonic scanning of the keyboard is awkward to achieve and the large number of diodes required would be undesirable.

The 8255 PIA on the CPU board drives the LED displays on the front panel. Port A data goes to the two 74LS47 seven segment display driver devices (IC22 and 23) on the front panel PCB, which in turn drives the dual digit display on the same board. Ports B and C drive the single LED displays via inverter buffers IC24 and 25, a pair of 7405s. To light up a LED, logic one has to appear at the relevant bit. The inverter then sinks the current required to illuminate the LED.

The power supply (Fig. 14) is constructed on a single PCB and supplies all the necessary power rails for the other boards.

The three regulator circuits are extremely standard and require little in the way of explanation.

Independent supply lines for the analogue joystick circuitry ensure maximum stability and noise immunity from the digital circuitry elsewhere.

Power Construction

Building the power supply PCB is straightforward. Ensure correct orientation of the bridge rectifier BR1 and all the electrolytic capacitors (especially the big ones!) as shown in Fig. 15. Mount C20 and C21 using the hole spacings on the PCB which are most suitable for the capacitors' sizes.

Solder regulators IC36 and IC38 directly onto the PCB and add a clip-on heatsink to IC36. IC37, the main 5V regulator, is to be mounted remotely to a large heatsink. In the prototype the aluminium lid of the keyboard cabinet was used for this but feel free to improvise.

Finish off the power supply PCB by adding veropins for the wire terminal points.

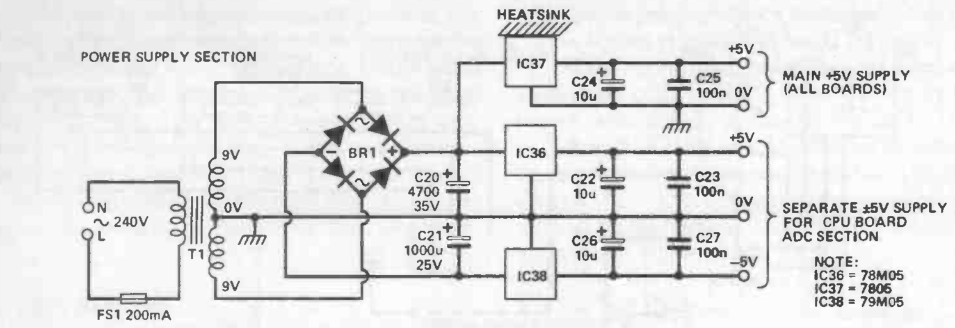


Fig. 14 The circuit diagram of the power supply board

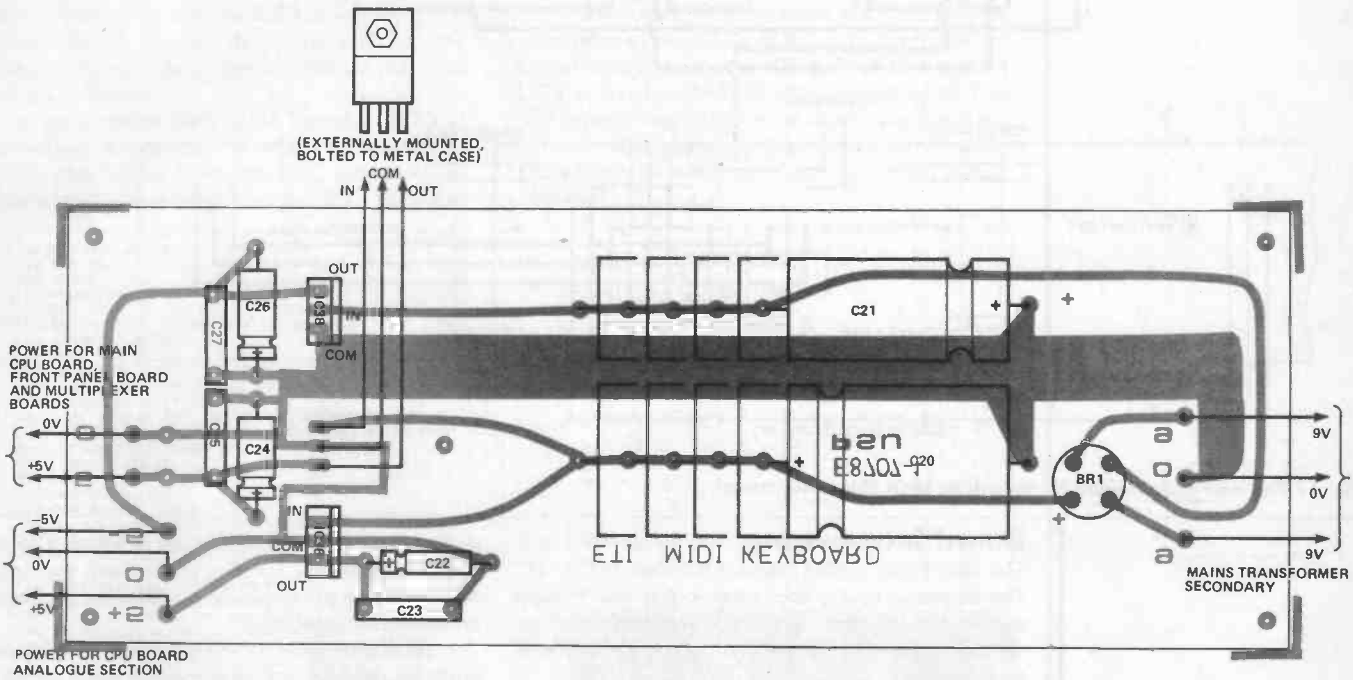
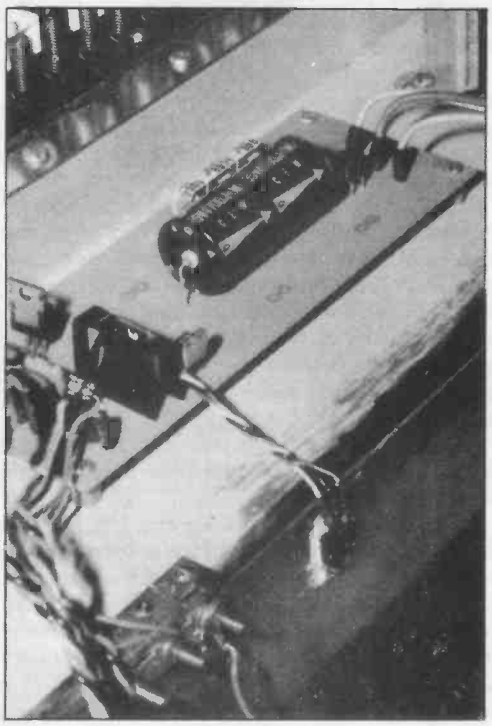


Fig. 15 The component overlay for the power supply board



The Cabinet

Figure 16 shows the details of the cabinet used in the prototype. This pattern can be followed religiously or you could design your own.

The power supply PCB and mains transformer are mounted on the plywood base, positioned close together.

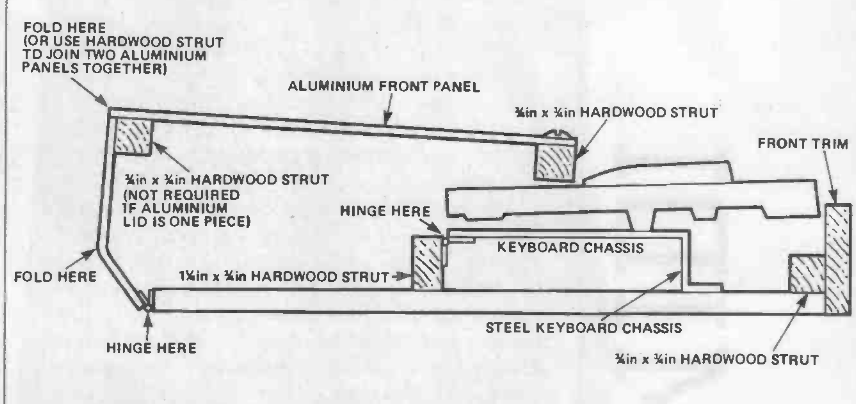


Fig. 16 Constructing the keyboard cabinet

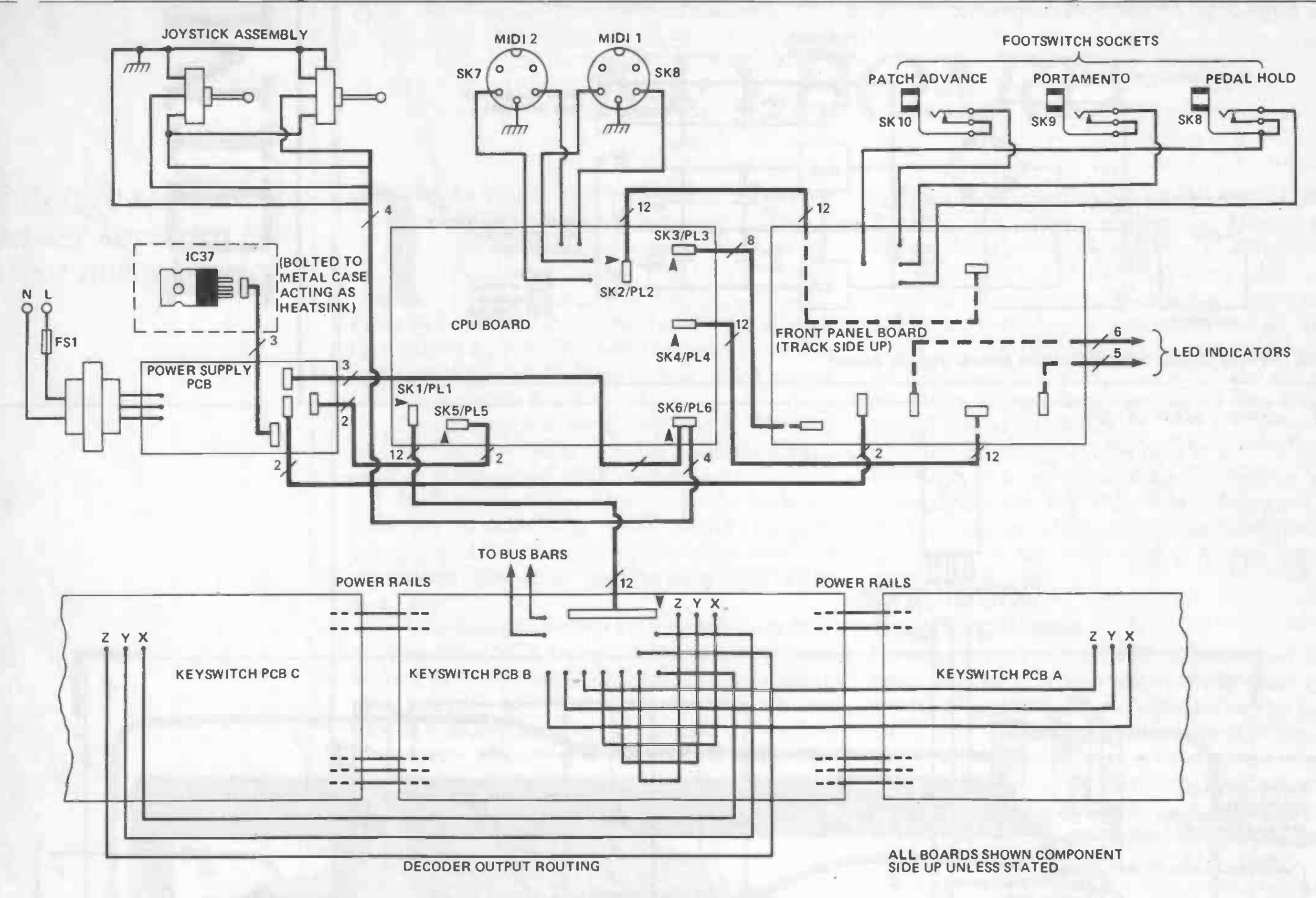


Fig. 17 The inter-board wiring for the whole MIDI Master Keyboard

MIDI

Board Inter-wiring

The inter-board wiring diagram is shown in Fig. 17. The diagram is by and large a topological one. Matters such as cable length and positioning will depend on the final board layout within the keyboard cabinet you have made.

The first task is to mount the power supply board and components within the cabinet. The power rails to all the other boards should be left to last. Do not connect the power rails until the power supply is checked and functioning correctly.

Great care should be taken to ensure there are no unnecessary mains voltage hazards when assembling the power supply components. All mains wiring should be kept as short as possible and should be sleeved for extra insulation.

Use more sleeving or a silicon rubber compound to insulate the mains transformer and mains power switch wire terminals. It is a good idea at this stage to earth the aluminium cabinet lid and the steel chassis of the keyboard.

Wire up the MIDI OUT DIN sockets to the CPU board and the footswitch sockets to the front panel PCB as shown in Fig. 17 before bolting them onto the aluminium lid. The board clearance should be such that the push button bank protrudes with the key caps flush with the lid.

Before finally bolting the front panel board into place, make sure the LEDs are all properly slotted into position.

Bolt the two joysticks onto the front panel and make up the wiring harness as shown in Fig. 18. Earth the two potentiometer cases as shown and use screened cable for the connections to the potentiometer wiper terminals.

Adjust the modulation joystick potentiometer so as to make the wiper voltage register 0V when the joystick lever is at its lowest point of travel. The potentiometer setting for the pitch bend joystick is adjusted later in conjunction with the CPU board powered and operating.

All that remains is to plug all the wiring harness multi-way plugs into their respective destinations — the connectors which jump between the front panel board and CPU board and the one which links the keyswitch PCBs to the CPU board. Prepare the wiring for the power supply rails but do not finally connect the PSU board until it has been verified that the power supply rails are functioning correctly.

Give the whole assembly a double check for correct inter-board wiring, paying particular attention to the power rail connections. Incorrectly connected power rails are catastrophic as far as the integrated circuits are concerned!

Getting the power supply working should be fairly straight forward as very little can go wrong provided all the components are correctly orientated on the PCB. Power on the mains and check the output voltages. If they are incorrect, switch off immediately and check everything over again. If the power supply regulators are operating properly, switch off and disconnect the mains supply before wiring the power rails to all the other boards.

Congratulations! You've finished it. But it doesn't work yet — you probably noticed that. Now we need to check out the software.

This is provided in the form of a 16K ROM (a 27128). Only around 2.75K of this is actually used, the rest is left for further expansion of the system. A hex dump of the used parts of the ROM is given here (Listing 1) and this can be laboriously entered by hand into an EPROM programmer. Alternatively ready-programmed EPROMs are available (see *Buylines*).

HOW IT WORKS: JOYSTICKS

The joystick interface (Fig. 19) has the task of enabling the 6502A processor to digitally read the positions of two joysticks. Both mechanical assemblies of the joysticks have 5k0 linear rotary potentiometers coupled to the joystick lever pivot points. The joystick assembly intended for the pitch bend has a self centering lever whilst the modulation joystick is of the ratchet type.

Since the full travel of the joystick assembly is only about 20% of the full potentiometer track there are added complications if we are to ensure that the maximum digitised value of 255 from IC13 (a ZN448 8-bit A/D converter).

The ZN448 is configured to use its own internal voltage reference of 2.5V. This voltage reference is also fed to the top end of the joystick potentiometer tracks. If the minimum joystick deviation corresponds to 0V at the wiper then at the other extreme position the output will be approximately 20% of the reference voltage – about 0.5V.

To utilise the full dynamic range of the A/D converter an input voltage of 2.5V is required when the joystick is at full travel. This is achieved by amplifying the wiper voltages before they are presented to the input of the A/D converter. IC20a and IC20b form two non-inverting voltage amplifiers with gains that can be fine adjusted by preset trimmers RV1 and RV2.

When the pitch bend joystick potentiometer on the prototype master keyboard was set to give a 0-0.5V range at the wiper it was found the neutral position did not correspond to the expected voltage of 0.25V, resulting in a slightly assymmetric deviation of voltages when moved from the centre to the extremes of travel. This is because the potentiometer track is not completely linear throughout the whole of its span. In order to minimise this non-linearity it was necessary to move the working part of the potentiometer track to roughly its middle region. In such a set-up the wiper voltage is in the range 1.0-1.5V when the joystick is at its lowest extreme of travel. The latter displacement voltage must be cancelled out so that the correct range of 0-2.5V appears at the A/D converter input. This is achieved by imposing a DC offset at pin 6 of IC20b, resulting in a voltage subtraction of approximately 1.2V at the output of the op-amp.

IC14 is a 4051 used as a two input analogue multiplexer. PC7 from IC19 selects one of the two joystick voltages to be presented to the input of the A/D converter (pin 6). The A/D converter appears to the 6502A processor simply as a single memory mapped location (0B00H). Writing to 0B00H initiates a conversion start and after the conversion time span has elapsed (approx. 10us) the data can be read from the same location.

For a given joystick position the data read from the A/D converter should always be exactly the same in the ideal case. This would allow the software to transmit the relevant MIDI information only when the joystick position has just changed. However, in reality the least significant bit can fluctuate in successive readings for certain joystick positions due to the limited resolution of the A/D converter. A solution to this problem is to collect successive sample readings and take the average, thereby greatly reducing any occasional perturbations in the data. The software fix was found to be quite effective, although it effectively resulted in a reduction in the rate at which the joystick positions were scanned.

Once the system hardware is all built and the programmed EPROM is installed in the CPU board (as IC17) the whole MIDI Master Keyboard can be tested and set up.

Testing

Power on the system and look to see if the LEDs light up correctly. On power-up, the double digit display remains blank while the lit LEDs correspond to those of the MIDI CHANNEL A, GROUP 1 and BANK A. All the other LEDs should be blank.

If the double digit display lights up when a program number is selected (by pushing a numeric keypad button) and if the MIDI TRANSMIT LED lights up whenever any note on the keyboard is held down then you can take comfort in knowing the system is most likely functioning properly. Final verification will depend on monitoring the MIDI OUT data and checking that all the keypad functions are working properly.

The best way to check if the hardware is transmitting the correct MIDI data is to monitor the MIDI OUT data stream with the BBC micro MIDI interface elsewhere in this magazine. With the MIDI monitor program running, any byte of transmitted MIDI data from the MIDI OUT of the keyboard is printed on the screen. It will then be easy to verify if the MIDI messages from the keyboard are correct.

If the basic level of operation cannot be attained it will be necessary to trace some signals around the CPU to try to locate the fault. The two essential diagnostic tools are an oscilloscope and a microcomputer MIDI interface to monitor any data coming out of the MIDI OUT connector.

After power-up check the clock signals to the 6502A (2MHz at pin 37 of IC7), 6522A (2MHz at pin 25 of IC18) and the 68B50 (500kHz at pin 4 of IC4) are present and correct. If the 6502A has correctly gone through its power-up reset sequence and is communicating with the 6522A properly, the VIA should be generating the periodic 2ms interrupt signal. This should be present at pin 21 of IC18 (6522A) and pin 4 of IC7 (6502A).

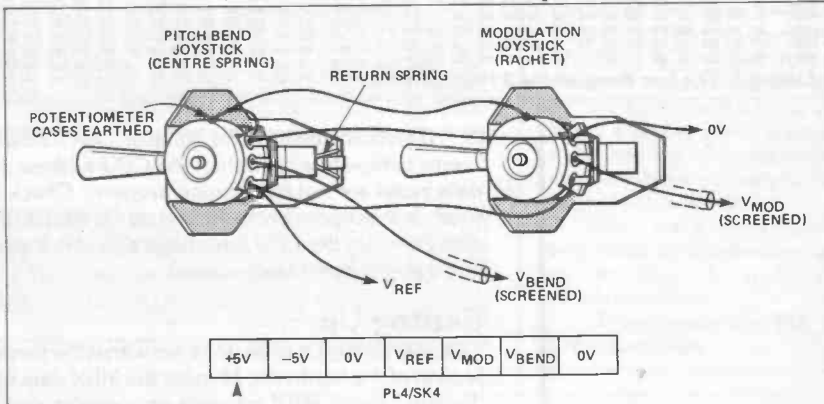


Fig. 18 Wiring the joystick assembly

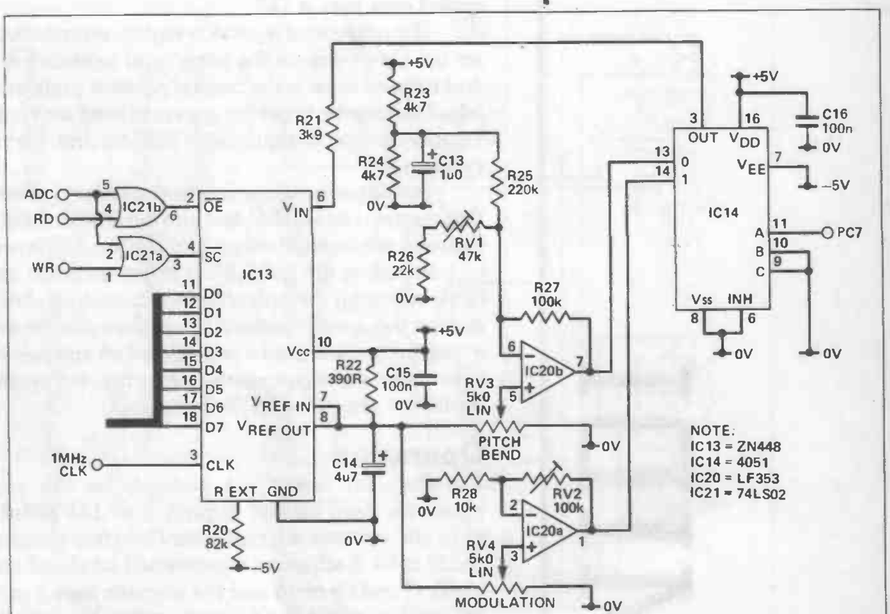


Fig. 19 The circuit diagram of the joystick circuitry

```
0000 78 D8 A2 FF 9A A9 03 8D 00 0C A9 15 8D 00 0C A9
0010 FF 8D 02 09 A9 00 8D 03 09 A9 40 8D 0B 09 A9 A0
0020 8D 04 09 A9 0F 8D 05 09 A9 80 8D 03 0A A9 FF 8D
0030 00 0A A9 81 8D 01 0A A9 01 8D 02 0A A2 00 A9 00
0040 8D 00 0B 95 0C E8 E0 48 D0 F9 A2 00 95 54 E8 E0
0050 48 D0 F9 A2 00 95 A2 E8 E0 1E D0 F9 85 01 85 00
0060 85 08 85 C6 85 9E 85 C8 85 C7 85 9F A9 04 85 C9
0070 AD 42 C2 85 0A AD 43 C2 85 0B A9 64 85 CD 85 CE
0080 A9 F0 85 05 A9 F1 85 06 A9 00 85 07 A9 80 85 9C
0090 85 9D A9 00 85 A0 85 D3 A9 00 85 D4 AD 00 08 85
00A0 D1 A9 80 85 A1 A9 C0 8D 0E 09 58 18 90 FD 48 8A
00B0 48 98 48 08 AD 04 09 A5 01 C5 00 F0 09 A8 B9 00
00C0 02 8D 01 0C E6 01 A2 00 BD 00 08 BD 00 08 85 03
00D0 B5 0C D0 10 A5 03 C9 03 F0 03 4C 91 C1 A9 01 95
00E0 0C 4C 91 C1 C9 01 D0 62 A5 03 C9 03 D0 0B 85 54
00F0 C9 3F B0 02 F6 54 4C 91 C1 C9 01 D0 09 A9 00 95
0100 0C 95 54 4C 91 C1 A9 02 95 0C A5 C3 F0 03 4C 91
0110 C1 A4 00 E4 07 90 08 A5 CD 85 CC A5 05 D0 06 A5
0120 CE 85 CC A5 06 29 9F 99 00 02 C8 E6 00 86 04 A5
0130 CC 38 E5 04 99 00 02 E6 00 B5 54 A8 B1 0A A4 00
0140 99 00 02 E6 00 A6 04 18 90 47 A5 03 C9 01 D0 39
0150 A9 00 95 0C 95 54 A4 00 E4 07 90 08 A5 CD 85 CC
0160 A5 05 D0 06 A5 CE 85 CC A5 06 29 9F 99 00 02 C8
0170 86 04 A5 CC 38 E5 04 99 00 02 C8 B5 54 99 00 02
0180 E6 00 E6 00 E6 00 18 90 08 C9 02 D0 04 A9 02 95
0190 0C E8 E0 48 F0 03 4C C8 C0 A5 01 C5 00 F0 09 A8
01A0 B9 00 02 8D 01 0C E6 01 A5 08 D0 06 20 04 C3 18
01B0 90 75 C9 01 D0 06 20 0C C3 18 90 6B C9 02 D0 06
01C0 20 14 C3 18 90 61 C9 03 D0 06 20 1C C3 18 90 57
01D0 C9 04 D0 06 20 43 C3 18 90 4D C9 05 D0 06 20 6C
01E0 C3 18 90 43 C9 06 D0 06 20 43 C4 18 90 39 C9 07
01F0 D0 06 20 76 C4 18 90 2F C9 08 D0 06 20 D7 C5 18
0200 90 25 C9 09 D0 06 20 71 C7 18 90 1B C9 0A D0 06
0210 20 C0 C7 18 90 11 C9 0B D0 06 20 6E C6 18 90 07
0220 C9 0C D0 03 20 EC C6 E6 08 A5 08 C9 0D D0 04 A9
0230 00 85 08 28 68 A8 68 AA 68 40 00 F0 40 F0 80 F0
0240 C0 F0 00 F1 40 F1 80 F1 C0 F1 00 F2 40 F2 80 F2
0250 C0 F2 00 F3 40 F3 80 F3 C0 F3 00 01 02 03 04 05
0260 06 07 08 09 10 11 12 13 14 15 16 17 18 19 20 21
0270 22 23 24 25 26 27 28 29 30 31 32 33 34 35 36 37
0280 38 39 40 41 42 43 44 45 46 47 48 49 50 51 52 53
0290 54 55 56 57 58 59 60 61 62 63 64 65 66 67 68 69
02A0 70 71 72 A5 9E D0 07 A5 9C 29 0F 18 90 0A A5 9D
02B0 29 0F 18 90 03 AA CA 8A 18 65 C8 AA A4 00 A5 9E
02C0 F0 07 A5 06 86 9D 18 90 04 A5 05 86 9C 29 CF 99
02D0 00 02 C8 8A 99 00 02 E6 00 E6 00 20 9B C3 60 A5
02E0 9E F0 05 A5 06 18 90 02 A5 05 25 D2 A4 00 99 00
02F0 02 C8 A5 CA 99 00 02 C8 A5 CB 99 00 02 E6 00 E6
0300 00 E6 00 60 A9 A2 85 C5 A9 FE D0 16 A9 AA 85 C5
0310 A9 FD D0 0E A9 B2 85 C5 A9 FB D0 06 A9 BA 85 C5
0320 A9 F7 8D 00 09 A0 00 AE 01 09 8A 4A AA B0 0A B1
0330 C5 D0 0A A9 02 91 C5 D0 04 A9 00 91 C5 C8 C0 08
0340 D0 E8 60 A9 00 85 C2 A2 00 B5 A2 C9 01 D0 08 E8
0350 8A 09 80 85 C2 D0 14 C9 02 D0 0B A9 01 95 A2 E8
0360 8A 85 C2 18 90 05 E8 E0 10 D0 DE 60 A9 00 85 C3
0370 85 BA 85 BB 85 BC A2 00 B5 B2 C9 01 D0 08 E8 8A
0380 09 80 85 C3 D0 14 C9 02 D0 0B A9 01 95 B2 E8 8A
0390 85 C3 18 90 05 E8 E0 0D D0 DE 60 A5 9E F0 09 A5
03A0 CE 85 CC A5 9D 18 90 06 A5 CD 85 CC A5 9C A8 29
03B0 80 F0 07 A9 FF 8D 00 0A D0 6B 98 29 40 D0 0D AD
03C0 01 0A 29 3F 09 80 8D 01 0A 18 90 0A AD 01 0A 29
03D0 3F 09 40 8D 01 0A 98 29 30 D0 0C AD 02 0A 29 F0
03E0 09 01 8D 02 0A D0 2E C9 10 D0 0C AD 02 0A 29 F0
03F0 09 02 8D 02 0A D0 1E C9 20 D0 0C AD 02 0A 29 F0
0400 09 04 8D 02 0A D0 0E C9 30 D0 0A AD 02 0A 29 F0
0410 09 08 8D 02 0A 98 29 70 85 C8 98 29 0F A8 C8 89
0420 5A C2 8D 00 0A A9 00 85 C7 A5 CC C9 64 D0 09 AD
0430 01 0A 29 DB 8D 01 0A 60 AD 01 0A 29 FB 09 20 8D
0440 01 0A 60 A9 00 85 C4 A2 00 B5 0C C9 02 D0 0C 8A
0450 85 04 38 A9 48 E5 04 85 C4 D0 05 E8 E0 48 D0 E9
0460 A5 C4 D0 09 AD 01 0A 29 EF 8D 01 0A 60 AD 01 0A
0470 09 10 8D 01 0A 60 A5 C3 D0 03 4C 8F C5 29 80 D0
0480 14 A5 C3 C9 04 D0 11 A5 C9 AA E8 BD 5A C2 8D 00
0490 0A A9 FF 85 C7 4C A6 C5 C9 03 D0 1F A9 00 85 09
04A0 A5 C4 F0 04 A9 FF 85 09 38 A9 48 E5 07 AA BD 5A
04B0 C2 8D 00 0A A9 FF 85 C7 4C A6 C5 C9 0C D0 25 A9
04C0 00 85 09 A5 C4 F0 04 A9 FF 85 09 A9 FF 85 C7 A5
04D0 9E D0 07 A5 CD 85 CC 18 90 04 A5 CE 85 CC 20 A7
04E0 C5 4C A6 C5 C9 01 D0 25 A5 05 29 0F AA E8 BD 5A
04F0 C2 8D 00 0A A9 FF 85 C7 A5 C4 D0 0E AD 01 0A 29
0500 FC 09 01 8D 01 0A A9 00 85 9E 18 90 39 C9 02 D0
0510 25 A5 06 29 0F AA E8 BD 5A C2 8D 00 0A A9 FF 85
0520 C7 A5 C4 D0 0E AD 01 0A 29 FC 09 02 8D 01 0A A9
0530 01 85 9E 18 90 70 C9 05 D0 0F A5 C8 29 CF 85 C8
0540 20 A3 C2 18 90 60 18 90 5D C9 06 D0 0E A5 C8 29
0550 CF 09 10 85 C8 20 A3 C2 18 90 4B C9 07 D0 0E A5
0560 C8 29 CF 09 20 85 C8 20 A3 C2 18 90 39 C9 08 D0
0570 0E A5 C8 29 CF 09 30 85 C8 20 A3 C2 18 90 27 C9
0580 0D D0 23 A5 C8 49 40 85 C8 20 A3 C2 18 90 17 A5
0590 C2 F0 0C 29 80 D0 0F A5 C2 20 B5 C2 18 90 07 A5
05A0 C7 F0 03 20 9B C3 60 AD 01 0A 29 FB 09 20 8D 01
05B0 0A A5 CC C9 64 90 0B 38 E9 64 AA BD 5A C2 8D 00
05C0 0A 60 38 A9 64 E5 CC AA BD 5A C2 8D 00 0A AD 01
05D0 0A 09 04 8D 01 0A 60 A5 C7 F0 27 A5 C3 29 7F C9
05E0 03 B0 22 A5 C2 F0 1B AA A8 29 80 D0 15 CA 8A 09
05F0 F0 A6 9E D0 05 85 05 18 90 02 85 06 B9 5A C2 8D
0600 00 0A 18 90 68 C9 04 D0 24 A5 C2 F0 60 AA A8 29
0610 80 D0 5A CA 86 C9 B9 5A C2 8D 00 0A 8A 0A AA BD
0620 3A C2 85 0A E8 BD 3A C2 85 0B 18 90 40 C9 03 D0
0630 19 A5 09 D0 38 A5 C4 F0 34 A8 B9 5A C2 8D 00 0A
0640 A9 48 38 E5 C4 85 07 18 90 23 C9 0C D0 1F A5 09
0650 D0 1B A5 C4 F0 17 18 69 44 85 CC A5 9E D0 07 A5
0660 CC 85 CD 18 90 04 A5 CC 85 CE 20 A7 C5 60 A5 CF
0670 C5 D1 90 10 38 E5 D1 C9 08 90 24 18 69 80 90 18
0680 A9 FF D0 14 A5 D1 38 E5 CF C9 08 90 12 85 04 A9
0690 80 38 E5 04 B0 02 A9 00 C5 A1 F0 22 18 90 08 A5
06A0 A1 C9 80 F0 19 A9 80 85 A1 4A AA 90 04 A9 01 D0
06B0 02 A9 00 85 CA 86 CB A9 EF 85 D2 20 DF C2 A5 D0
06C0 4A C9 02 90 11 C5 9F F0 22 85 CB 85 9F A9 01 85
06D0 CA D0 11 18 90 15 A5 9F F0 11 A9 00 85 9F 85 CB
06E0 A9 01 85 CA A9 BF 85 D2 20 DF C2 60 A9 F7 8D 00
06F0 09 AD 01 09 AA 29 01 C5 A0 F0 29 85 A0 A5 A0 D0
0700 0C AD 01 0A 29 F7 8D 01 0A A9 00 F0 0A AD 01 0A
0710 09 08 8D 01 0A A9 7F 85 CB A9 BF 85 D2 A9 40 85
0720 CA 20 DF C2 8A 29 02 C5 D3 F0 19 85 D3 A5 D3 D0
0730 04 A9 00 F0 02 A9 7F 85 CB A9 BF 85 D2 A9 41 85
0740 CA 20 DF C2 8A 29 04 D0 06 A9 00 85 D4 F0 21 A5
0750 D4 D0 1D A9 7F 85 D4 A5 9E F0 05 A5 9D 18 90 02
0760 A5 9C C9 80 B0 0A 29 0F AA E8 8A 29 0F 20 B8 C2
0770 60 A9 00 85 D5 85 D6 AD 02 0A 29 0F 8D 02 0A A0
0780 10 8D 00 0B A2 06 CA D0 FD AD 00 0B 18 65 D6 85
0790 D6 90 02 E6 D5 88 D0 E9 29 0F C9 08 90 06 A9 01
07A0 85 04 D0 04 A9 00 85 04 A5 D5 0A 0A 0A 0A A8 A5
07B0 D6 4A 4A 4A 4A 85 D9 98 05 D9 18 65 04 85 CF 60
07C0 A9 00 85 D8 85 D7 AD 02 0A 09 80 8D 02 0A A0 10
07D0 BD 00 0B A2 06 CA D0 FD AD 00 0B 18 65 D8 85 D8
07E0 90 02 E6 D7 88 D0 E9 29 0F C9 08 90 06 A9 01 85
07F0 04 D0 04 A9 00 85 04 A5 D7 0A 0A 0A 0A A8 A5 D8
0800 4A 4A 4A 4A 85 D9 98 05 D9 18 65 04 85 D0 60 00
3000 7F 40 2A 20 19 15 12 10 0E 0D 0C 0B 0A 09 08 08
3010 07 07 07 06 06 06 06 05 05 05 05 05 04 04 04 04
3020 04 04 04 04 03 03 03 03 03 03 03 03 03 03 03 03
3030 03 03 02 02 02 02 02 02 02 02 02 02 02 02 02 02
3040 7F 47 31 25 1E 19 16 13 11 0F 0E 0D 0C 0B 0A 0A
3050 09 09 08 08 07 07 07 07 06 06 06 06 05 05 05 05
3060 05 05 05 04 04 04 04 04 04 04 04 04 04 03 03 03
3070 03 03 03 03 03 03 03 03 03 03 03 03 03 03 03 02
3080 7F 5B 47 3A 31 2A 25 21 1E 1C 19 18 16 14 13 12
3090 11 10 0F 0F 0E 0E 0D 0C 0C 0C 0B 0B 0A 0A 0A 09
30A0 09 09 09 08 08 08 08 08 07 07 07 07 07 07 07 06
30B0 06 06 06 06 06 06 06 06 05 05 05 05 05 05 05 05
30C0 7F 5E 4B 3E 35 2E 29 25 21 1F 1C 1A 18 17 16 14
30D0 13 12 11 11 10 0F 0F 0E 0E 0D 0D 0C 0C 0B 0B 0B
30E0 0A 0A 0A 0A 09 09 09 09 08 08 08 08 08 08 07 07
30F0 07 07 07 07 07 06 06 06 06 06 06 06 06 06 06 06
3100 7F 62 4F 43 3A 33 2D 29 25 22 20 1E 1C 1A 18 17
3110 16 15 14 13 12 11 11 10 0F 0F 0E 0E 0E 0D 0D 0C
3120 0C 0C 0B 0B 0B 0A 0A 0A 0A 0A 09 09 09 09 09 08
3130 08 08 08 08 08 08 07 07 07 07 07 07 07 07 06 06
3140 7F 63 51 45 3C 35 2F 2B 27 24 21 1F 1D 1B 1A 18
3150 17 16 15 14 13 12 12 11 10 10 0F 0F 0E 0E 0E 0D
3160 0D 0C 0C 0C 0B 0B 0B 0B 0A 0A 0A 0A 0A 09 09 09
3170 09 09 08 08 08 08 08 08 08 07 07 07 07 07 07 07
3180 7F 66 55 49 40 38 33 2E 2A 27 24 22 20 1E 1C 1B
3190 19 18 17 16 15 14 14 13 12 12 11 10 10 0F 0F 0F
31A0 0E 0E 0D 0D 0D 0C 0C 0C 0C 0B 0B 0B 0B 0A 0A 0A
31B0 0A 0A 09 09 09 09 09 09 08 08 08 08 08 08 08 08
31C0 7F 6A 5B 4F 47 3F 3A 35 31 2D 2A 28 25 23 21 20
31D0 1E 1D 1C 1A 19 18 18 17 16 15 14 14 13 13 12 12
31E0 11 11 10 10 0F 0F 0F 0E 0E 0E 0E 0D 0D 0D 0C 0C
31F0 0C 0C 0C 0B 0B 0B 0B 0B 0A 0A 0A 0A 0A 0A 09 09
3200 7F 6C 5D 52 4A 43 3D 38 34 30 2D 2B 28 26 24 22
3210 21 1F 1E 1D 1C 1B 1A 19 18 17 16 16 15 14 14 13
3220 13 12 12 11 11 11 10 10 0F 0F 0F 0F 0E 0E 0E 0D
3230 0D 0D 0D 0C 0C 0C 0C 0C 0B 0B 0B 0B 0B 0B 0A 0A
3240 7F 6E 62 58 4F 49 43 3E 3A 36 33 30 2D 2B 29 27
3250 25 24 22 21 20 1F 1E 1D 1C 1B 1A 19 18 18 17 16
3260 16 15 15 14 14 13 13 13 12 12 11 11 11 10 10 10
3270 0F 0F 0F 0F 0E 0E 0E 0E 0E 0D 0D 0D 0D 0D 0C 0C
3280 7F 73 6A 62 5B 55 4F 4B 47 43 3F 3C 3A 37 35 33
3290 31 2F 2D 2C 2A 29 28 26 25 24 23 22 21 21 20 1F
32A0 1E 1E 1D 1C 1C 1B 1A 1A 19 19 18 18 18 17 17 16
32B0 16 16 15 15 14 14 14 14 13 13 13 12 12 12 12 11
32C0 7F 79 73 6E 6A 66 62 5E 5B 58 55 52 4F 4D 4B 49
32D0 47 45 43 41 40 3E 3C 3B 3A 38 37 36 35 34 33 32
32E0 31 30 2F 2E 2D 2D 2C 2B 2A 2A 29 28 28 27 26 26
32F0 25 25 24 24 23 23 22 22 21 21 21 20 20 1F 1F 1F
3300 10 10 10 10 10 10 10 10 10 10 10 10 10 10 10 10
3310 10 10 10 10 10 10 10 10 10 10 10 10 10 10 10 10
3320 10 10 10 10 10 10 10 10 10 10 10 10 10 10 10 10
3330 10 10 10 10 10 10 10 10 10 10 10 10 10 10 10 10
3340 20 20 20 20 20 20 20 20 20 20 20 20 20 20 20 20
3350 20 20 20 20 20 20 20 20 20 20 20 20 20 20 20 20
3360 20 20 20 20 20 20 20 20 20 20 20 20 20 20 20 20
3370 20 20 20 20 20 20 20 20 20 20 20 20 20 20 20 20
3380 40 40 40 40 40 40 40 40 40 40 40 40 40 40 40 40
3390 40 40 40 40 40 40 40 40 40 40 40 40 40 40 40 40
33A0 40 40 40 40 40 40 40 40 40 40 40 40 40 40 40 40
33B0 40 40 40 40 40 40 40 40 40 40 40 40 40 40 40 40
33C0 7F 7F 7F 7F 7F 7F 7F 7F 7F 7F 7F 7F 7F 7F 7F 7F
33D0 7F 7F 7F 7F 7F 7F 7F 7F 7F 7F 7F 7F 7F 7F 7F 7F
33E0 7F 7F 7F 7F 7F 7F 7F 7F 7F 7F 7F 7F 7F 7F 7F 7F
33F0 7F 7F 7F 7F 7F 7F 7F 7F 7F 7F 7F 7F 7F 7F 7F 7F
3FF0 FF FF FF FF FF FF FF FF FF FF FF FF 00 C0 AE C0
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Listing 1. The hex dump of the 27128 EPROM

If there is no sign of this signal or if the hardware seems completely dead then the CPU address and data buses are not functioning properly. Check for shorts in the copper tracks making up the address and data buses on the CPU board and also check that a through-pin hasn't been missed.

Setting Up

The only setting up to be done lies within the joystick section of the hardware. Monitor the MIDI data with the BBC micro MIDI interface or a similar device. Move the modulation joystick lever and adjust RV2 so that in the position of highest modulation, the transmitted data byte is 127.

The pitch bend joystick is slightly more tricky to set up. On power-up the pitch bend joystick is read and defined to be in the neutral position (data value 64). The target is to get the system to send zero in the lowest joystick position and 128 for the highest position.

Two adjustments are required in order to achieve this, the gain control RV1 and also the relative position between the potentiometer and the joystick lever.

Adjusting the gain (RV1) in combination with finely adjusting the potentiometer centering should achieve the correct results. An important point to note is that the hardware must be powered off and then on again after any adjustments to that the new neutral position is registered by the software.

Operation

The Program Select will probably be the most frequently used keypad request. The 128 possible program numbers are segregated into two groups (I and II) of 64. Each group is partitioned into four banks (A, B, C and D) of 16 and the numeric keys 1 to 16 are used to select the program within the selected bank. If your synthesiser has, say, only 32 stored voices which can be selected by MIDI then only banks

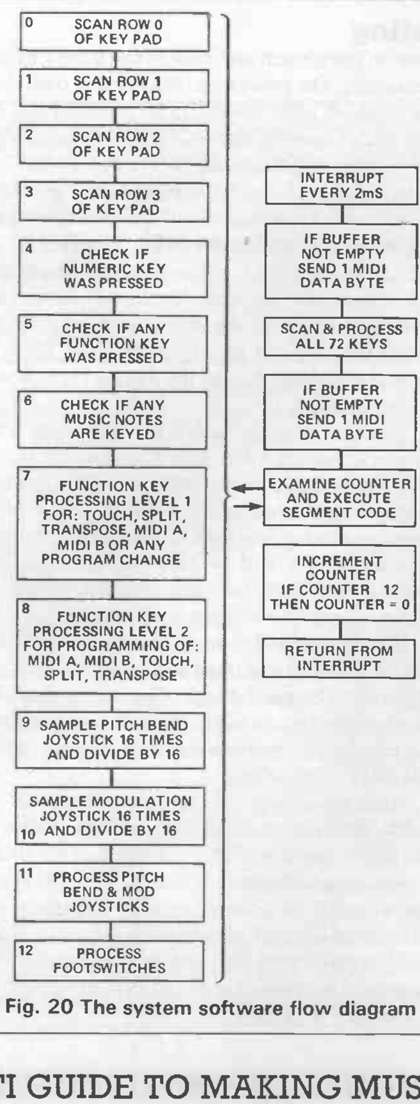


Fig. 20 The system software flow diagram

HOW IT WORKS: SOFTWARE

The most time-critical part of the system software is the part which does the keyboard scanning. Not only does it have to scan the keyboard polyphonically and detect key strikes and releases, it also has to determine key strike velocities. Such demands require complete scans of the keyboard to occur at precise time intervals. The smaller the time interval, the greater the velocity sensing resolution.

This is achieved by placing the key scanning code at the start of an interrupt routine which executes every 2ms. The periodic 2ms signal which iterrupts the 6502A processor is generated by the internal timer in IC18 (6522A VIA). Also high in priority is the section of code which empties the buffer containing data to be transmitted via MIDI. When the buffer is being emptied, two bytes of MIDI information can be transmitted every 2ms interrupt period. Such a data rate is sufficently fast to ensure delays in response are insignificant.

Since the key scan and MIDI buffer empty routines typically take up about half the available 2ms time slot there is some spare time allocation for other processing tasks. Figure 20 shows the structure of the software.

On power-up, once everything had been initialised, the main program divides into an endless loop so that thereafter the only real processor activity comes from the periodic interrupt routines.

The code pertaining to the house keeping activities is segmented into 13 parts. A software counter in maintained so that in successive interrupt routines the 13 sections of code are executed in turn.

Calculations show that in the extreme case when about a dozen notes are played at once (Sellafield musicians take note!) it is possible to exceed the 2ms time allocation. This however, is not at all catastrophic since all that happens is that a velocity count is missed, whch hardly affects the overall dynamics of the MIDI keyboard.

The current physical state of any of the 72 keys on the keyboard can be obtained by examining locations 0800H to 0847H which are memory mapped directly to the keyswitch hardware. However, for the software to determine when to send out MIDI note on/off events it must also know the state of all the keys from the previous scan. This is achieved by storing the states of the keys in 72 zero page RAM locations.

The contents of each of these RAM locations shows an indentifier code (0-3) for each of the four possible sequential states that a key can be in – 'transition going up' and 'transition going down'.

When a key is processed, its current state is compared with its last state (from the previous scan). Any necessary action is taken and the RAM location describing the previous stage is updated.

The software counts the number of 2ms scan periods that the keyswitch stays in the 'transition going down' state. So in addition to the key state RAM locations there are a further 72 zero page locations that serve as velocity counters for active keys.

The maximum count allowed is 63, so the time period that a key spends in transition is measured to be in the range 0 to 128ms, with a resolution of 2ms. In practice, most key strikes are well within half that time period range.

The velocity count for a key cannot be used directly since it corresponds to a time period, not a velocity. We need to derive a suitable velocity byte from the time period count. This is easily achieved by using the time period count to index a look-up tables giving 16 different touch response settings.

Note that the last four look-up tables are filled with constant values of 16, 32, 64 and 127 respectively. These non touch sensitive settings add versatility to the keyboard operation, since touch sensitivity is not always desired.

Transmission of MIDI data is accomplished by maintaining a first in first out (FIFO) buffer in RAM. The buffer size is 256 bytes and occupies page two of the processor's RAM workspace.

Two bytes of data from the queue are sent to the ACIA in every 2ms interrupt period until the buffer is empty. The buffer size is sufficently large to ensure it never overflows. A 256 byte size was also chosen because of ease of management with 8-bit pointers.

When a note on/off event is to be sent, an offset is added to the note number before transmission. On power-up, the offset defaults such that when middle C is pressed or released, the resultant note number sent is 60 – the correct MIDI code for middle C.

The keysplit faciltiy is implemented by maintaining a RAM location which holds the note number of the programmed split point. On transmission of a MIDI note event, the note number is compared to the split point. If it is less than the split point note number, the data is directed to the MIDI channel associated with channel A. Otherwise it is directed to channel B. On power-up, the split point defaults to 72 (the rightmost note) so that all output goes to channel A.

The positions of the two joysticks are read continually as part of the housekeeping software. In order to counteract least significant bit fluctuations in the readings (due to the finite resolution of the A/D converter), each joystick position reading is the result of averaging over a set of 16 successive samples.

The centering joystick lever can never be expected to spring back to precisely the same neutral position so it was necessary to implement a dead-band in software.

A and B will be used. All other sections will simply be a repeat of what's available in banks A and B.

Note that a MIDI program change is transmitted when any bank or group button is pressed as well as any numeric button.

Output Stream

The MIDI Master Keyboard's versatility is greatly enhanced by transmitting data through a programmable MIDI channel and by having a dual channel MIDI output stream. On power-up, all data is transmitted via the channel A output stream, on MIDI channel 1.

To change the MIDI channel the channel button must be held down and a numeric button 1-16 selected. The change can be verified by pressing the channel button on its own again so that the newly selected MIDI channel number is displayed.

If the channel B button is pressed, all data is transmitted on the MIDI channel that has been programmed into the B output stream (default on power-up is channel 2).

The only exception to this rule is when a split point has been programmed, in which case all notes to the left of the split point will be transmitted on the channel A output stream, whilst all notes to the right will be transmitted on channel B.

The dual output stream capability, although primarily intended for implementing the split point facility, can also be used to remotely change programs on MIDI digital effects units such as the Alesis Midiverbs and so on. The effects unit is simply set to receive on MIDI channel B, so that it responds to any program change requests made (when current output stream is set on channel B) from the function keypad.

Split Point

If the 'SPLIT' button is pressed and held the double digit display will show the currently selected split point. The number 1 corresponds to the leftmost key and 72 is the rightmost key. The power-up default is 72 so that all the notes are directed towards channel A.

To program a new split point the procedure is to simply press the selected note on the keyboard whilst holding down the SPLIT button. Once selected, release the SPLIT button and the new split will now be effective.

Transpose

When a split point is in effect, both parts of the keyboard are transposable independently. The currently selected output stream decides which part of the keyboard is to be affected when the transpose function is selected.

For example, if we wish to transpose the part of the keyboard to the left of the split point then the currently selected output stream has to be channel A. To implement the transpose facility, first of all press the TRANSPOSE button. The present setting is then displayed on the dual digit display in the form of an offset in semitones from middle C.

A number with the leftmost decimal point lit means that the offset is negative. A number on its own implies a positive offset.

In its non-transposed state the number displayed is 0. To program a new transpose setting, hold the TRANSPOSE button down and select a note on the keyboard. Pressing middle C will result in a zero offset and pressing any other note will result in the keyboard being transposed by the relative offset in semitones.

Note that once a non-zero transpose setting has been programmed the LED above the TRANSPOSE button will remain lit.

The present touch sensitivity is displayed when the TOUCH button is pressed and held down. To program a new touch setting, hold the TOUCH button down and press a number 1-16. Release the two buttons and verify the change by pressing TOUCH on its own again. You can also verify the change by playing the keyboard!

Footswitches

The footswitch sockets are intended for use with push-to-break switches which is consistent with most commercial equipment. Pushing the pedal hold switch should result in the hold LED lighting up as well as the appropriate MIDI messages being transmitted. Successive switching of the patch advance footswitch will result in successive program number changes within the currently selected bank of 16. The last footswitch switches portamento in and out via MIDI (not all synthesisers will make use of this).

PARTS LIST

RESISTORS (all ¼W 5%)	
R1-11, 18, 43, 44	1k0
R12-15	220R
R16, 23, 24	4k7
R17	3k3
R19	15k
R20	82k
R21	3k9
R22	390R
R25	220k
R26	22k
R27	100k
R28, 65, 66	10k
R29-42, 45-56	330R
R57-64	33k
RV1	47k horiz. preset
RV2	100k horiz. preset
RV3	5k0 lin. single axis joystick (centre sprung)
RV4	5k0 log. single axis joystick (ratchet)
CAPACITORS	
C1	10n polycarbonate
C2,4-6,8,9,11,12,15,16,17,19,23,25,27	100n polycarbonate
C3	22µ 16V electrolytic
C7,22,24,26	10µ 16V axial electrolytic
C10,18	220µ 16V axial electrolytic
C13	10µ 16V axial electrolytic
C14	4µ7 16v axial electrolytic
C20	4700µ 35V axial electrolytic
C21	1000µ 25V axial electrolytic
SEMICONDUCTORS	
IC1	74LS04
IC2, 3	74LS74
IC4	68B50
IC5	74LS05
IC6	74LS373
IC7	6502A
IC8, 9	74LS00
IC10, 21	74LS32
IC11	74LS138
IC12	74LS27
IC13	ZN448
IC14	4051
IC15	74LS244
IC16	HM-6116
IC17	2764
IC18	6522
IC19	8255
IC20	LF353
IC21	74LS02
IC22, 23	74LS47
IC24, 25	7405
IC26	74LS42
IC27-35	4051
IC36	78M05
IC37	7805
IC38	79M05
BR1	W01 bridge rectifier
D1	IN4148
D2-30	1N4148
LED1	Amber LED
LED2, 5-7, 9	Red LED
LED3, 4, 10	Green LED
LED8, 11	Yellow LED
LED12	Double digit 7 segment display (common cathode)
MISCELLANEOUS	
FS1	200mA fuse and holder
PL1, 2, 4	12-way female PCB connector
PL3	8-way female PCB connector
SK1, 2, 4	12-way male PCB connector
SK3, 6	8-way male PCB connector
SK5	2-way male PCB connector
SK7, 8	5-pin DIN chassis socket
SK9-11	¼in jack socket
SW1-24, 28, 29	Push button
SW25-28	Foot switch
T1	9-0-9V mains transformer
XTAL1	4MHz crystal

PCBs. IC sockets. PCB through pins. Clip-on heatsink for IC36. Connecting wire. Case. 72-note keyboard. 72 spring contacts. 2 bus bars and mounting blocks. 12-way ribbon cable.

BUYLINES

The joysticks used in the prototype were from SLM (modes) Ltd., Chilton Road, Prestbury GL52 5JQ. Tel: (0242) 525488.

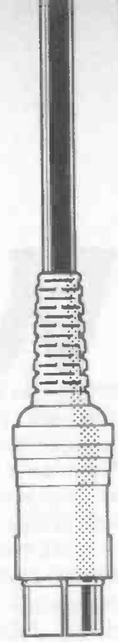
Just about any keyboard can be used for this project. Maplin (Tel: (0702) 552911) supplies two models of 49 and 61 notes which can be adapted in pairs. Alternatively Watkins Electric Music (WEM) does two octave joinable keyboards. WEM is on 01-761 6568.

The PCBs are available from the PCB service.

A pre-programmed EPROM of the software to drive the keyboard is available from the author for £16, supplied with the source code in BBC Basic/assembler. Orders and enquiries should be sent to John Yau, 9 Harden Place, Edinburgh EH1 1JD.

CASIO DG-20

Is it a guitar? Is it a keyboard? Is it both? Is it worth 250 quid?



In case you thought the Japanese had left the guitarist outside in the cold while keyboards players enjoy the warm glow of a MIDI fire. Casio has proven that MIDI can be applied usefully and relatively cheaply to other instruments too.

The DG-20 is Casio's cheapest MIDI guitar offering. At £249 it is no more expensive than many of the cheaper MIDI keyboards but, in the right hands, enormously more expressive.

The DG-20 *is* really a cheap Casio keyboard. It just happens to come in a guitar shaped case and is played with pick and frets rather than a line of undersized keys but otherwise all the familiar signs are there. It has a rhythm box with all the usual predictable patterns — rock, waltz, bossa nova and so forth. It has its own built-in amp (2W) and speaker with both line out and headphone sockets.

These features alone make it an unusual guitar. You can say goodbye to the practice amp with this one. It contains its own batteries so it is genuinely portable and once you've got over the rhythms provided, it does form an excellent trainer to practise playing in time.

However, the DG-20 is not recommended for learning guitar. You can tell immediately from the Darth Vader styling that this is not an altogether serious instrument. It is very plastic and despite a remarkably solid (and adjustable!) neck and head and a quite surprisingly fast action, it takes a lot of getting used to and many techniques have to be 'unlearnt' to play it.

One definite advantage for the beginner is that you don't have to tune the DG-20. The tension on the strings has no effect on the note. The string vibrations serve only as a trigger to start the entirely electronic sounds. The string tensions *can* be adjusted but simply to suit your taste.

The pitch is provided by pressure on the frets. Each fret is actually a rubber strip (that doesn't help the natural feel either!) under which are six switches, one for each string. Pressing a fret signals to the electronics which string is being fingered where on the fretboard to sound the corresponding note.

This creates its own problems. The fretboard must be fingered between frets, not right up behind the fret in the normal way, otherwise there is a possibility of pressing the string down over the next fret and triggering that switch.

Even more importantly you cannot bend notes and (silly, this) no tremelo bar is provided either.

However, hammer-ons and pull-offs work well although they can occasionally cause a re-trigger of the note.

The DG-20 is not touch sensitive and this seriously affects its credibility as a 'real' guitar but for slow arpeggio strumming and legato themes it works wonders, far more expressive than any keyboard (for a guitarist, anyway!) and more versatile than any other guitar because of the range of sounds available.

There are 20 sounds available built-in. These are selected with ten 'tone' and a 'select' membrane buttons on the top edge of the body. The positioning of these buttons makes it impossible to play the DG-20 with a short strap high on your chest.

The guitar sound-alikes are reasonable (except the 'distortion' sound which is not really distorted enough) and several other very non-guitar sounds are provided too (trumpet, flute and organ!).

The sound selected actually affects the way the instrument works. Sounds like the acoustic guitar have a short sustain which cannot be altered. Once started, the note cannot be stopped by deadening the strings. Sounds such as the organ sustain forever, if you let them. They are stopped by releasing the frets or with a mute button positioned on the scratchplate just below where the pickups should be.

The top panel also contains buttons for transpose up and down, reverb and 'solo' which turns the DG-20 monophonic — can be useful for butterfingered fast solo runs. The rhythm controls are mostly here too. Twelve patterns are available made up from PCM sounds of two toms, a cymbal and snare, again chosen from six buttons with an annoying select button. Tempo is selected with up and down buttons.

The scratchplate has separate volume controls for the main sound and rhythm. There's a rhythm start/stop button, a synchronisation button which also doubles as a rhythm fill-in trigger and four touch pads for playing along on the rhythm sounds (difficult when both hands are tied up with the guitar itself!). It's unlikely much serious use will ever be made of any of the rhythm section.

At the blunt end of the body is another plate with the output sockets, tuning control, 9V AC inlet, foot switch input for rhythm on/off and the all-important MIDI out.

The DG-20 can transmit MIDI data in MIDI modes 3 (omni off, poly) and 4 (omni off, mono). In mode 4 each string transmits on a different channel so you could connect the DG-20 up to six synths and play different sounds with each string. There can't be much call for that (it's also damn difficult to play like that) and the only sensible purpose would be to use six identical synths using the same multi-timbral voices for an incredibly rich sound.

MIDI mode 3 is much more useful and allows you to use the DG-20 as a genuine guitar synth. As well as the *note on/off* data, the DG-20 transmits program changes when the top plate buttons are pressed (although the different sustain performance associated with the guitar's internal program changes can cause quite some confusion here) and clock data for the rhythm start/stop and tempo (possibly a serious use for the rhythm, at last).

Used with a decent MIDI synth or expander the DG-20 opens up a whole new world for guitarists. It isn't a truly professional instrument. It takes quite some getting used to and requires a few compromises of playing style. However, it *is* a guitar, it's versatile and it's fun — for this reasonable price an exceptionally clever instrument.

MIDI HARMONISER

Teach your synth to sing in perfect harmony with this MIDI project

Way back in the pre-MIDI era, harmonisers processed audio signals to produce an output that followed the pitch of the input but was higher in frequency by some fixed musical interval. In practice there were problems with this approach, producing a small but noticable delay between new notes and the harmoniser's reaction. This type of harmoniser was also unforgiving of any less than perfect performance, particularly when used with guitars.

MIDI processors such as this unit process the data stream itself rather than being simply an audio processor instructed via MIDI. This way notes are clearly defined in digital form and the signal is processed at the MIDI baud rate of 31250 — 'instant' as far as the human ear is concerned.

The harmoniser works by identifying the note value bytes of *note on* and *note off* messages (see *Unmuddled MIDI* for the complete description of the messages). The *note value* gives the pitch of the note and is incremented by the desired interval — anything from 0 to 15 semitones.

Watch The Clock

There is a potential flaw in this system caused by clock messages. If your MIDI equipment generates clock messages and sandwiches them inside a *note on* message, the output of the harmoniser will be horribly confused. You could disable your clock but this might affect built-in sequencing action in your equipment. Most equipment will not cause these problems (none of mine does) but check your manual. It shouldn't be too hard to add a circuit that intercepts timing codes and passes them through unprocessed.

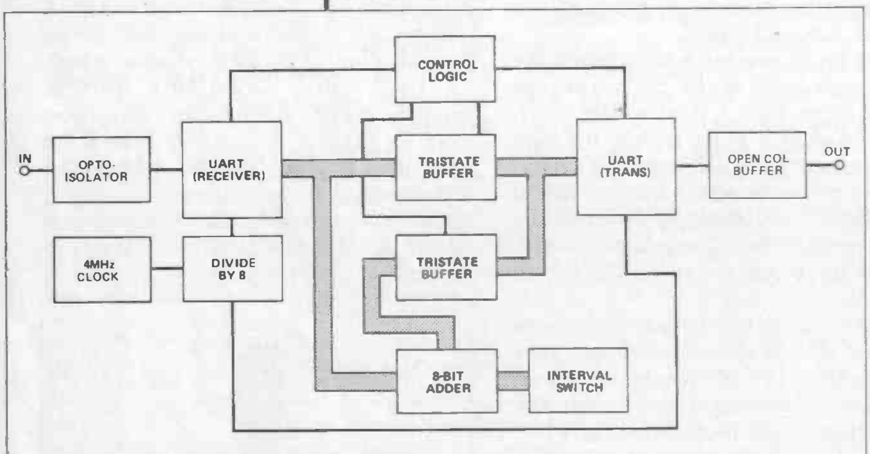
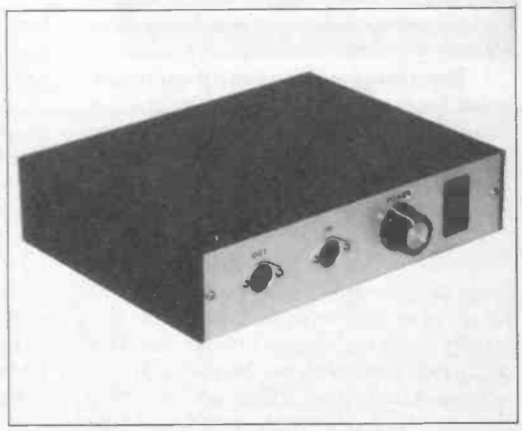


Fig. 1 The block diagram of the harmoniser



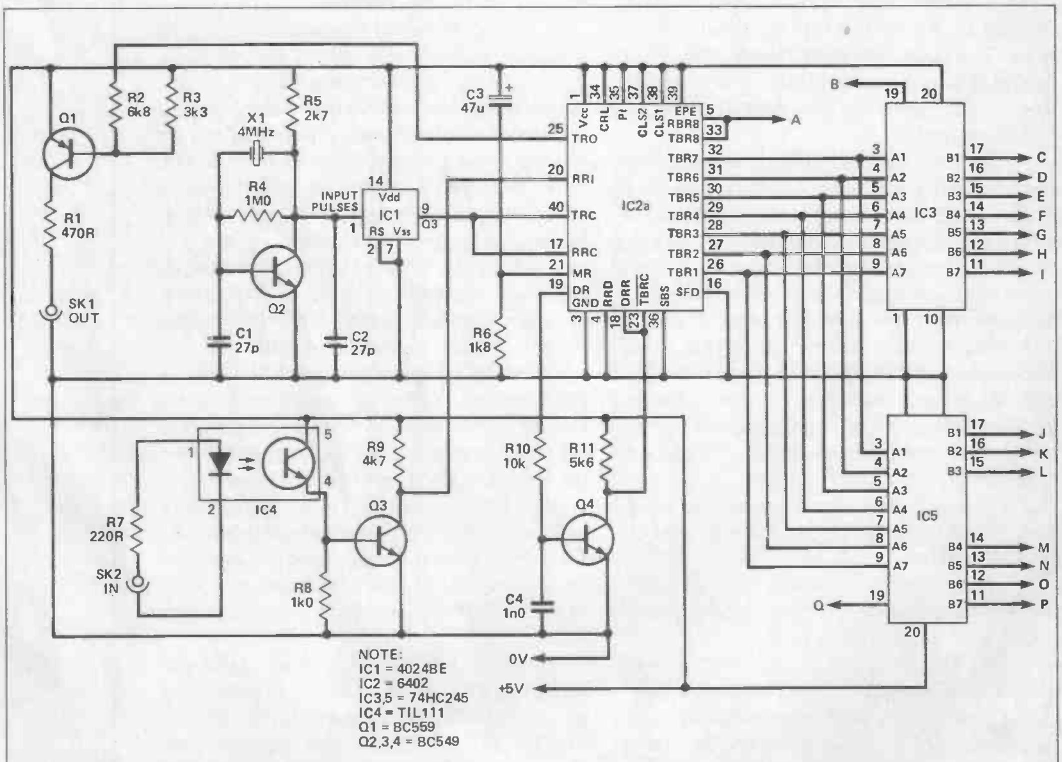


Fig. 2 Circuit diagram of the MIDI harmoniser

MIDI

HOW IT WORKS

Figure 1 shows the block diagram of the MIDI harmoniser. The input signal is coupled to the receiver section of a UART via an opto-isolator.

Normally the decoded bytes of parallel data are fed by way of a tristate octal buffer to the transmitter section of the UART and out in serial form to the output via an open collector buffer driver.

The control logic detects fresh bytes of data and monitors the most significant nibble to spot *note on* and *note off* messages. That byte is retransmitted as normal but the next byte (the note value) is taken instead to an 8-bit adder. A hex switch sets the musical interval by which the adder increments the note value, and the modified value is coupled to the UART transmitter via a second octal tristate buffer.

The circuit diagram is shown in Figs. 2 and 3. The UART is IC2, an industry standard 6402. Happily UARTs are equally at home controlled by microprocessor or by switches, LEDs and simple logic.

Of particular importance is the fact that a UART can be set to the right word format by connecting some of its inputs to the appropriate logic levels, and it does not need to be programmed via its data bus. In this case pin 36 is taken low, while pins 35, 37, 38, and 39 are taken high. This gives the appropriate word format of one start bit, eight data bits, one stop bit and no parity. Pin 34 must be taken high to load this control word into IC2 and it is perfectly satisfactory to take this pin permanently high.

An initial reset pulse is needed for IC2 and this is provided by C3 and R6. Q2 forms the basis of the clock oscillator and IC1 is the divide-by-eight circuit that produces the 500kHz signal required by IC2. IC1 is actually a seven stage binary divider but in this circuit only three stages are utilised. IC2 has separate receiver and transmitter clock inputs but split baud rate operation is obviously not needed in this case and both clock inputs are driven from the same output of IC1. Q1 is used in the open collector output stage and this is a basic common emitter switching stage. R1 limits the output current to 5mA. The 'data received' output of IC2 goes high when a byte of data has been received and is present on the data bus. The 'data received reset' input must be set low in order to reset this flag to the low state. Q4 operates as an inverter which connects between these two terminals so that the 'data received' terminal is almost immediately reset each time a fresh byte of data is received. This produces a negative pulse at the collector of Q4 each time a byte of data is received and this signal is used to trigger the transmitter section of IC2 so that bytes of data are retransmitted. C4 lengthens the trigger pulses slightly in order to ensure that they are sufficiently long to trigger the transmitter section section of IC2 reliably.

IC4 is the opto-isolator and on its input side it only has protection resistor R7. On the output side the transistor in IC4 is connected to what is effectively an emitter follower stage driving the discrete common emitter amplifier based on Q3. Using the internal transistor of IC4 in the emitter follower mode gives a relatively high switching speed, while the external transistor compensates for any lack of efficiency in IC4 and gives a nicely 'squared' output signal.

IC3 and IC5 are the octal tristate buffers. These are actually transceivers but in this circuit they are always in the 'receive' mode and they function as basic tristate buffers. Actually only seven buffers in each device are used since the most significant bit is coupled straight through from the receiver to the transmitter section of the circuit. As MIDI data values only utilize the seven least significant bits there is no need to process the most significant bit. IC9 is a three to eight line decoder and it monitors the most significant nibble of received bytes. One of its 'enable' inputs is brought into action in order to provide the fourth input. The outputs of IC9 are normally low but one of them goes high when a suitable 4-bit input code is present. Output 0 goes high when a *note off* header byte is received and output 1 goes high when a *note on* header byte received. D1, D2, and R15 form a simple OR gate which combines these two outputs into a single output that goes high when either type of message is present on the data bus. If you have one of the few instruments which have polyphonic aftertouch it might be worthwhile wiring a diode from output 2 (pin 13) to R15. The circuit should then process the note values in polyphonic aftertouch messages in the same way that it processes *note on* and *note off* types.

IC10 is a negative edge triggered monostable. It is therefore triggered when a new byte is received after a *note on* or *note off* message. Its Q and $\overline{Q}$ outputs each control one of the octal tristate buffers, and consequently only one of these drives the transmitter section of the UART at any one time. When the monostable is triggered it switches from the direct route through IC3 to the indirect route via IC5 to IC7. IC6 and IC7 are CMOS 4-bit full adders which are cascaded so that they function here as an 8-bit adder. They combine the value from the receiver section of the UART with the binary value set on SW1. This could be four separate switches, but in the interest of convenience a hex switch is probably a better choice. With zero shift selected the unit is effectively bypassed, and simply retransmits all MIDI bytes without modifying them in any way. Note that the pulse duration of IC10 is kept short so the processing is not maintained long enough to affect subsequent bytes.

The circuit takes 5V from a 9V battery and monolithic voltage regulator (IC8). The UART consumes only 1mA.

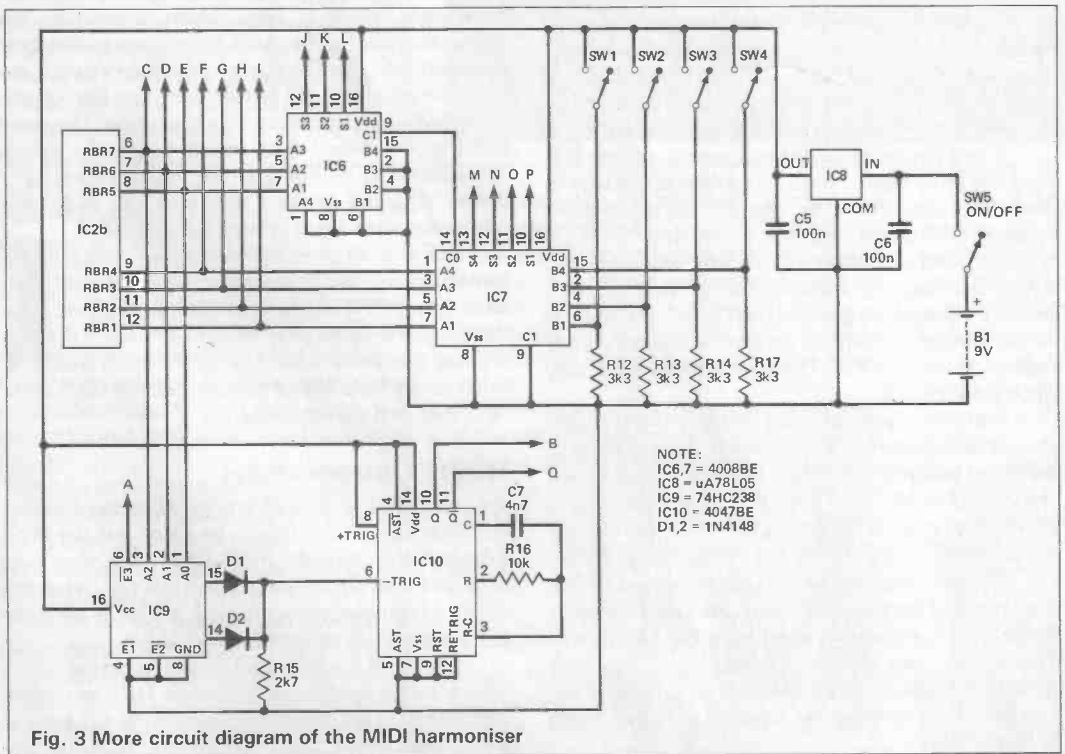


Fig. 3 More circuit diagram of the MIDI harmoniser

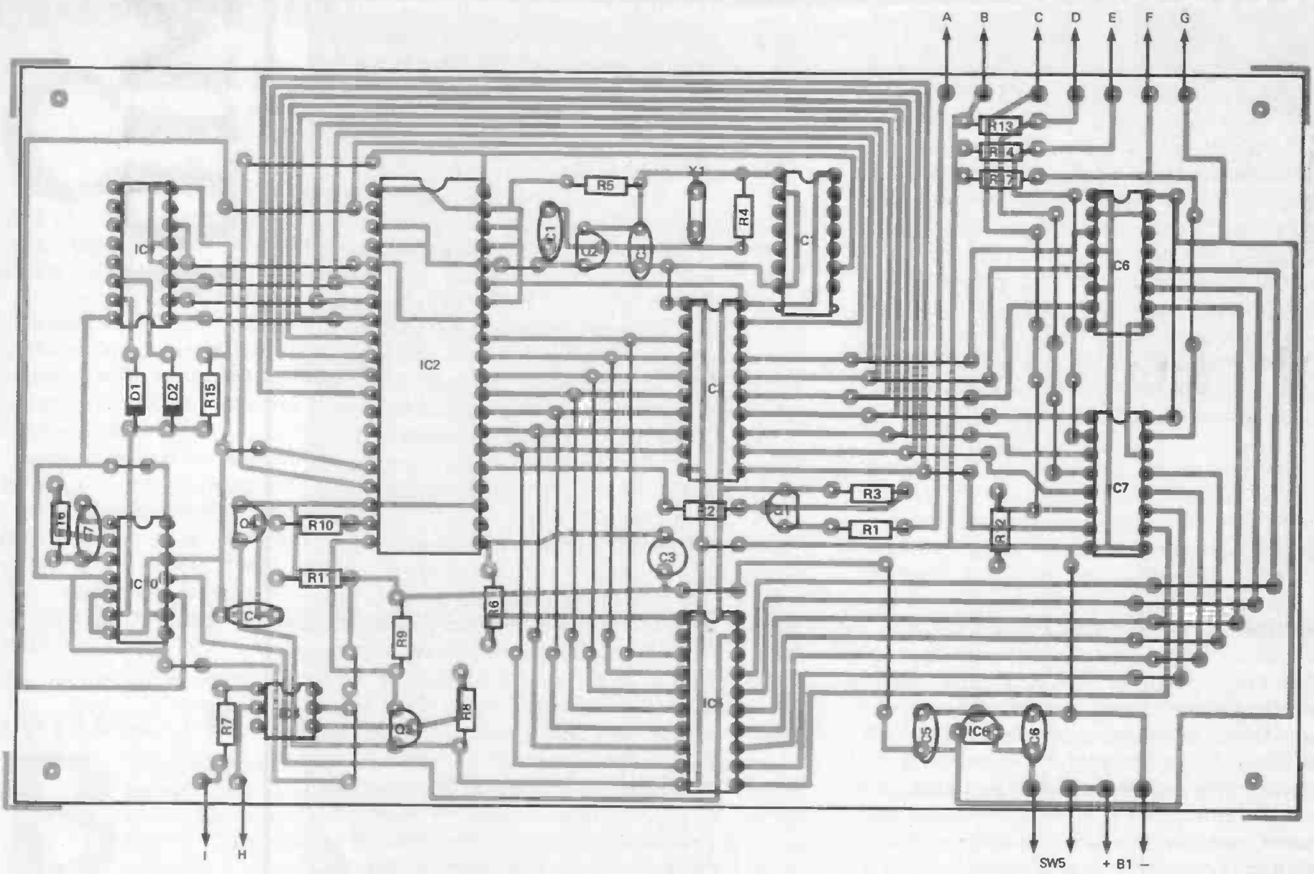


Fig. 4 The component overlay for the MIDI harmoniser

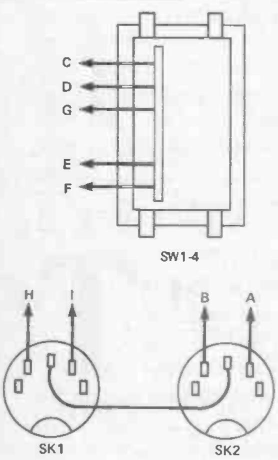


Fig. 5 Socket and switch wiring diagram

MIDI

Construction

Figure 4 shows the component overlay of the harmoniser. It uses a single-sided PCB but quite a profusion of links (22 swg wire). A double-sided board would have been more expensive and the links would only be replaced by PCB pins anyway. Make sure the links are quite taut before soldering in, particularly where several run side by side.

With the exception of IC4 the integrated circuits are all CMOS types. Sockets and standard anti-static handling precautions should be observed. Be especially careful with the UART which is not a particularly cheap component and note that IC6 has the opposite orientation to the other devices.

If you decide to fit IC4 in a socket a 6-pin DIL type will be required, which are relatively difficult to obtain but are offered by one or two component retailers. With care it is possible to trim down an 8-pin type to 6-way operation using a small hacksaw and a file. Most inexpensive opto-isolators (TIL111, CN27, MC72) will work properly in this circuit, and there is no advantage in using an expensive high efficiency type such as a CNY17. Darlington types are totally unsuitable though.

From the electrical viewpoint any 4MHz crystal should be suitable for X1. However, if it is to fit on to the board properly it must be a miniature wired-ended (HC-18/U or HC-49/U) type. At this stage only fit pins to the board at the points where connections to the off-board components will eventually be made.

The switches and sockets are mounted on the front panel of the recommended case (see *Buylines*). It's quite straightforward apart from the hex switch. The easiest and cheapest option is to use four miniature toggle or slider switches for S1 to S4, but you then have to enter the required note increment in binary form.

I used a thumbwheel type hex switch as detailed in *Buylines*. This is a push-fit into a 16.5×31mm rectangular cutout. This can be cut using a fretsaw, miniature round file and so on. It must be made quite accurately if the switch is to be a really good fit and it is advisable to make a slightly under-size cutout and then carefully file it out.

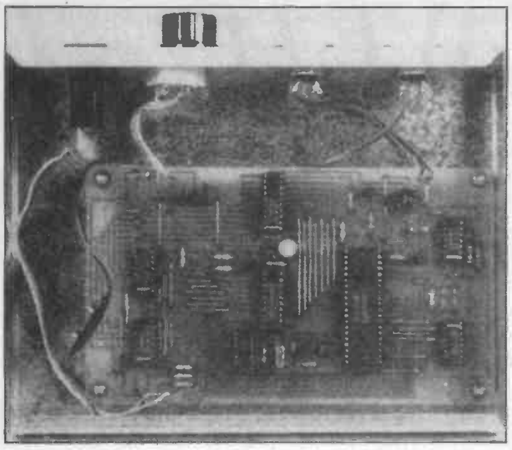
The printed circuit board is mounted on stand-offs on the base panel of the case and the point-to-point style wiring is then added. Details of the connections to the hex switch and two sockets are provided in Fig. 4. The contacts of the hex switch are a little unusual in that it has what is more like a piece of printed circuit board than the usual tags. There are holes in the board that enable physically strong connections to be made to the tracks without any undue difficulty or, as I did, you can make the connections via 1mm printed circuit pins.

The current consumption of the circuit is about 16 milliamps — too high for economic operation with an ordinary PP3 size battery but a high power type should give a reasonably long battery life. If the unit is going to receive a fair amount of use it would be better to opt for a higher capacity battery such as six HP7 cells in a plastic holder.

Right Connections

Provided SK1 and SK2 are wired in the manner shown in Fig. 5 the unit can be wired into the MIDI system using standard MIDI DIN leads. Any twin screened lead should suffice and it is not necessary to use a high quality type unless the leads are to be more than a few metres long.

The most fundamental way of using the unit is with a single synthesiser (or other MIDI equipped instrument) in the manner outlined in Fig. 6. Here the harmoniser takes the MIDI output signal of the



instrument, processes it, and feeds it back into the instrument again. Simply feeding the MIDI input of an instrument with its own MIDI output signal usually has no effect, and with the harmoniser set for zero shift this situation is created.

The situation is different if a shift is produced, and the instrument will then respond properly to the note messages received on its MIDI input as they appear to be coming from some form of external controller rather than being 'home-grown'. All the MIDI keyboard instruments I have encountered will respond simultaneously to information received via the keyboard and by way of the MIDI input. It would still be as well to check the MIDI specification sheet for an 'exclusion clause' before building the harmoniser with a view to using it in this way.

This arrangement can be quite effective, and would normally be used with a pitch shift of a fifth or an octave (the hex switch set at 7 or C respectively). It is a good way of obtaining thicker sounds, especially with synthesisers that provide only single VCO type sounds. You can use a shift of a seventh if that is what you like!

Remember that you will be playing two notes each time you press a key. This means that with (say) an eight note polyphonic instrument you should play no more than four notes at a time.

The unit could also be used should you be in need of key transposition. Setting your keyboard to 'local off' will leave it operating as a dummy keyboard transmitting MIDI data from the MIDI OUT socket, and separately as sound generating circuits driven from incoming data at the MIDI IN socket. The harmoniser will then act as a transposer driving the music circuits at an interval above that played on the keyboard. Of course you should see if your instruments have a transpose facility that can provide the same effect (most will) before going to the trouble and expense of building the MIDI harmoniser!

On The High Cs

One final point to keep in mind is that the unit will not function properly if the input note value plus the shift amount gives an output note value of more than 127. This is unlikely to be a problem in practice since the top note on most MIDI keyboards is simply not high enough to over-stretch the harmoniser. What is a more likely problem is that the note values from the harmoniser could sometimes be outside the compass of the instrument the unit is driving. This is more likely to occur with a sampler, as these generally have much more restricted compasses than normal synthesisers.

The result of any out-of-range notes is not likely to be too dire anyway. Most instruments simply ignore note outside their range or play the note in the nearest octave they can accommodate.

PARTS LIST

RESISTORS (all ¼W 5%)	
R1	470R
R2	6k8
R3,12,13,14,17	3k3
R4	1M0
R5,15	2k7
R6	1k8
R7	220R
R8	1k0
R9	4k7
R10,16	10k
R11	5k6
CAPACITORS	
C1,2	27p ceramic plate
C3	47µ 16V radial elect
C4	1n0 polyester
C5,6	100n ceramic
C7	4n7 polyester
SEMICONDUCTORS	
IC1	4024BE
IC2	6402
IC3,5	74HC245
IC4	TIL111 or similar (eg Maplin WL35Q)
IC6,7	4008BE
IC8	µA78L05
IC9	74HC238
IC10	4047BE
Q1	BC559
Q2,3,4	BC549
D1,2	1N4148
MISCELLANEOUS	
B1	9V (6×HP7 size cells in holder)
SK1,2	5-way 180° chassis mounting DIN sockets
SW1-4	hex thumbwheel switch (see Buylines)
SW5	rotary on/off switch
X1	4MHz HC-18/U

PCB. Case. Battery connector. Knob. IC sockets. Nuts and bolts. PCB pins.

BUYLINES

The case used for the prototype measured 215×166×51mm.

Most hex switches (SW1) are intended for on-board mounting so are unsuitable for this project. A thumbwheel switch is best – these are obtained as single switches plus optional end cheeks. These are available from Electromail (Tel: (0536) 204555). Switches are parts 337-093, end cheeks (pair) part 338-406.

The PCB is available from the PCB service.

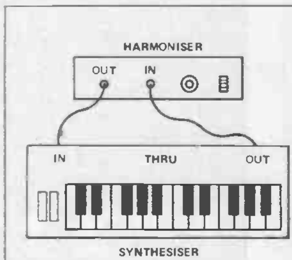


Fig. 6 Simple use of the harmoniser

TAPE IT OR LEAVE IT

What can the struggling artist hope to achieve in the sanctity of his own bedroom? — quite a lot by the look of it.

Let's take a typical musician. He's learnt his instrument to an acceptable level and he's thinking creatively. He has lots of musical ideas swimming about in his head and somehow wants to extract them without the need for surgery.

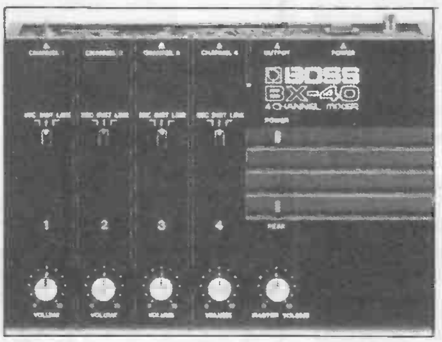
In short, he wants his music on tape so that he and his friends can actually make out what is he's been humming all this time.

When you find yourself in this position, the thought of the contents of a modern professional studio is enough to give you a severe pain in the wallet. How can you put together a worthwhile but affordable recording studio in the home?

Where To Begin!

Firstly you must decide where you are going to plant the seeds of your musical career. There's no single ideal place to build a studio, it all depends on where there's space, not too much noise and where Mum/girlfriend/wife will let you! The living room is out of the question. It has to be somewhere the equipment can be left set up and untouched. Get the idea? Good. Let's get some equipment together.

Initially your best bet is to use whatever you have around you, so you won't spend a fortune on the latest high tech wizardry only to find you're not as interested in the whole thing as you thought or (perish the thought) what sounded quite good and original in your head just comes out sounding like *Smoke on Water*.



The budget Boss BX-40 mixer

The first thing we need is a tape recorder. Anything with some form of microphone or line level inputs will do — microphones for vocals and guitar/wind instruments and line level for keyboards. If you want a combination of instruments (keyboard, guitar, whatever) you'll need some way of mixing the various levels together. One solution is to make a simple junction box to mix several inputs into one or two outputs with the volume levels altered at source.

For greater flexibility you can put together a simple mixing desk giving you a degree of control over the sound you put on tape. Nothing too extravagant is needed at this stage. Just faders, tone control and a means of panning a sound across a stereo image (if applicable). The mixer should have two line-level outputs but how many inputs will depend largely on your own needs. If you're going to produce a Tangerine Dream keyboard album then you'll just need say half a dozen line-level inputs. If on the other hand your music is guitar-based with vocals then you will need to build in some provision for mic (low level) inputs.

Another useful addition at this stage is a drum machine — whatever music you like, you'll find a degree of rhythm behind it. Your keyboard may have a built-in rhythm section and this can be more than adequate. This won't help guitarists and spending £100-£200 on a beat box of some kind is money well spent. Getting a rhythm unit that is MIDI equipped will save an upgrade later.

Being a musician, you'll probably have a collection of effects pedals under the bed (you mean you *don't*!). These will do fine for recording for the moment. A little audio trickery used sparingly can do wonders to the overall sound of a track. If you're trying to decide which to buy. it is difficult to advise since everyone has his own priority. In my experience a reverb unit of some description is useful and if you play a very dynamic instrument like a bass guitar, a compressor/limiter is in order. Any others (and there are quite a few) are by no means essential.

If you haven't built in provision for effects in your mixer, you will have to daisy-chain them together between instrument and the mixer or between the mixer and the recorder depending on whether the effect is for the overall mix or just one element of it.

Taking The Mic

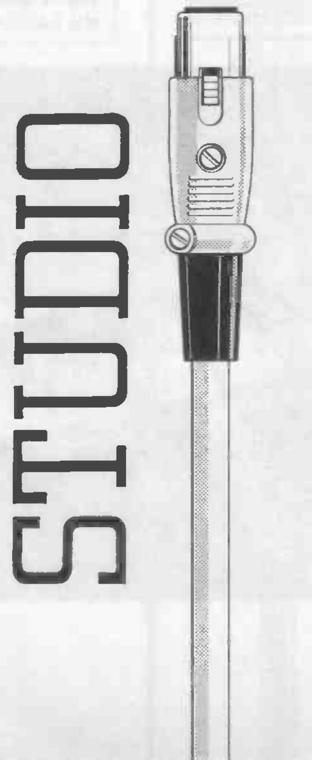
Not every sound goes conveniently down a phono lead. Some sounds need a little help from a microphone. Mics can cost anything from £10 to £1000 and recording salesmen will tell you all kinds of horror stories about what happens to you and your recordings if you don't buy the top model. In reality £50 is all that's needed. In fact one of the best mikes in this price range, the Tandy PZM, costs just over half that.

When picking a mike, choose with your ears. You may well find one model costing a few pounds more than another will actually sound worse for your specific application. Only your ears can tell you that.

Let's Go Multi-track

Having established that recording is something you want to explore further, the next step is purchasing a spare pair of hands in the shape of a multi-track recorder. With this you can record four separated tracks of music one at a time, giving the illusion of several musicians playing at once.

Four track recorders come in all shapes and sizes but the most popular format of recent years has been the 'Portastudio'. This is a 4-track cassette machine with a built-in mixer. They cost upwards of £200 and you do get what you pay for so it's a good idea to buy the best machine you can afford.



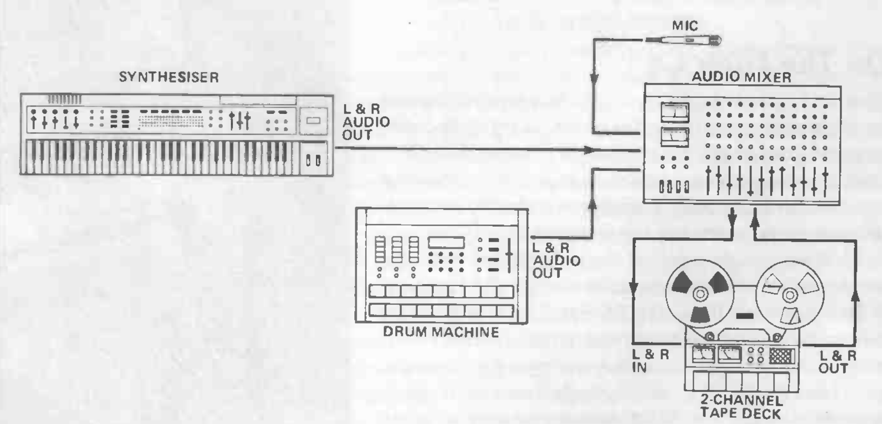


Fig. 1 A basic studio set-up

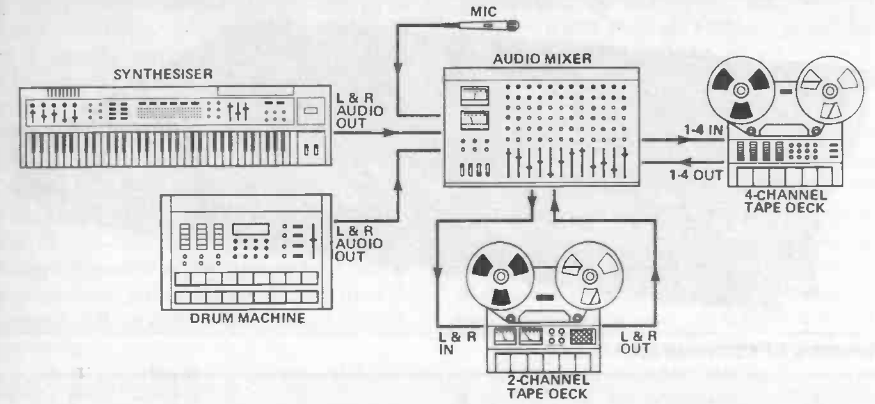
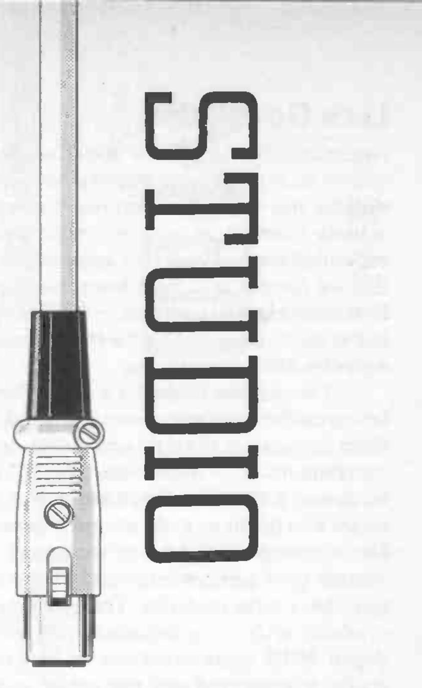


Fig. 2 Four-track recording arrangement



Mixing facilities on these machines are limited by nature of their size so a more comprehensive, stand-alone mixer is required to get down to business. For this level of recording you will need a respectable equalisation (EQ) section (bass, middle and treble) and a means of routing signals into the desk to the right track on the tape.

Additionally, you need a auxiliary 'send and return' facility on each input to allow you to send the signal to an effects unit and then back into the mixer. This way you can control the amount of effect each track or instrument gets. The mixer should also have the facility to mix the outputs of the recorder down to stereo so you can dump your recording in a format that everyone can use. This promotes your original stereo recorder to the position of mastering machine.

Your arsenal of effects should now be building up although you may now be noticing that some of those pedals you bought or built are sounding a bit noisy these days. Now's the time to either to open them up and put in some quieter circuitry or to upgrade the units altogether for something less noisy and more flexible.

Roland, Akai and Alesis all make excellent effect units costing between £100 and £200. The other good reason for upgrading is the mass of interconnecting wires on the floor which always manage to tangle themselves up into a rats' next. All the units from these companies stack on a convenient shelf or made-to-measure rack to make life tidy and your studio look better (bound to score you brownie points with Mum/girlfriend/wife).

Of course having an impressive effects collection means messing about with plugging and unplugging leads so another excellent acquisition is a patchbay which in effect (no pun intended) puts all your connections in one place.

These are very easy to make. All that is required is a multitude of jack sockets and somewhere to mount them, plus of course a collection of short jack-to-jack leads for patching the front panel.

Now you've amassed quite a few miles of audio and mains cable this is a good time to sort out where they're going to go. The easiest solution is to lose it under the floorboards as no one will trip over it bringing your pride and joy crashing to the ground. However under the carpet may well be just as good if the cables aren't too thick and can be spread out.

Try to keep mains and audio cables separate to avoid picking up mains hum. Also ensure the audio leads are of good quality and well screened. The sound can only be as good as your weakest link which in all too many cases is the cabling.

Now your system is ready for action you can concentrate on what sounds you have at your disposal. These can easily be improved with additional instruments as the 4-track gives the musical power of eight hands. A synthesiser is the best choice as it is capable of generating a vast spectrum of sounds at the touch of a button. In case there are a few guitarists out there shouting 'What about us', don't forget you can now buy guitars which can control synths thanks to MIDI.


The Cheetah MD8 drum machine


Akai's MIDI controlled MB76 mixer bay and PEQ6 EQ


Casio's PG380 digital guitar

The Yamaha FB-01 FM rack synth

The Yamaha REX50 multieffects unit

Let's Go MIDI

Assuming that you have some electronic sound sources at your disposal we should explore the ways in which the number of sounds running at any one time can be expanded beyond your four audio tracks. For this we need a little help from the Musical Instrument Digital Interface — MIDI. For the home studio environment the most important aspect is MIDI sequencing.

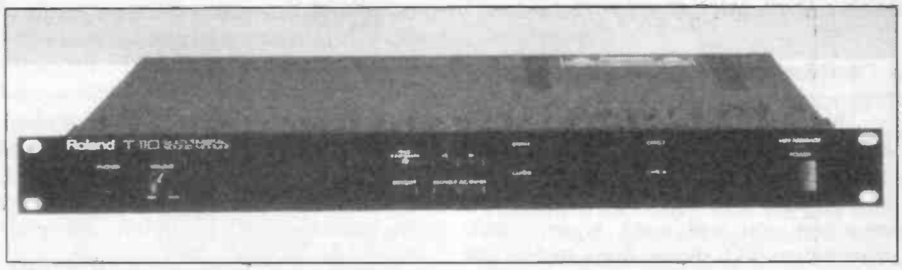
A sequencer performs a simple task. It listens out for numbers, stores them and spits them out again in the right order. Not any old numbers mind — MIDI data. Play a MIDI-equipped instrument (anything from a keyboard to a guitar or even a wind instrument like a saxophone) and a sequencer can 'record' your performance and play it back just like a tape recorder. That's where the similarity ends — a sequencer just records digital MIDI representations of the notes you're playing and not the actual sounds themselves. This allows a great deal of flexibility not enjoyed by mere tape owners.

The MIDI specification allows 16 independent channels of data so a decent sequencer should allow at least 16 'tracks' for music to go on. Some computer software-based sequencers have as many as 64 tracks. This is no better for getting more sounds out of your machines as you're still limited to 16 there however it does prove useful if you wish to act the indecisive producer and have half a dozen alternatives for each musical part. This feature alone makes a sequencer worth having as it is the most cost-effective means of upgrading the number of sounds you can *use and alter* at the same time.

Sequence data can be easily edited. Voices for parts can be changed as many times as you like without any loss of sound quality and without having to play the piece correctly again. Complex segments need only be entered once as they can be copied and looped as many times as required. You also score in the fidelity stakes as your song can be composed, tweaked and re-tweaked until you're totally happy before anything goes down to tape.

All this does require some additional

Yamaha's SPX90 multi-effects unit



Roland's T110 multi-timbral rack synth

musical hardware in the shape of extra synths but don't faint yet. You don't need to go out and buy another 15 to compliment the one you have already. All you need is one or two that are 'multi-timbral'. The most cost and space effective way to do this is with synth 'expanders'.

An expander is all the guts of a professional synth without the keyboard. All those currently available are multi-timbral and at least £100 cheaper than their keyboard counterparts (in many cases much more). A couple of these combined with your existing keyboard and drums will fill up the MIDI channels without any trouble at all.

You can also use samplers to add 'live' or 'acoustic' sounds to a sequence. For example you might want a piano sound — a sample avoids trying to fit a Steinway in your studio (and the hassle of miking the thing up).

MIDI can be a useful extra pair of hands when it comes to effects as well thanks to several MIDI-equipped multi-effect units available. By recording patch change numbers into the sequencer on a spare track, you can change the effect you're using far faster that you could manually.

If your 4-track is peering over your shoulder and reading this, it may well be feeling its days are numbered. Nothing could be further from the truth as you still require live sounds to overdub. All sequenced parts can be recorded onto one track (or two to retain stereo separation) and acoustic sounds can go on the remaining tracks.

The MIDI sync facility enables even greater power. A timecode on one of your recording tracks can keep your sequencer or drum machine in time with acoustically recorded tracks so that all the electronic parts can remain first generation right up to mixdown.

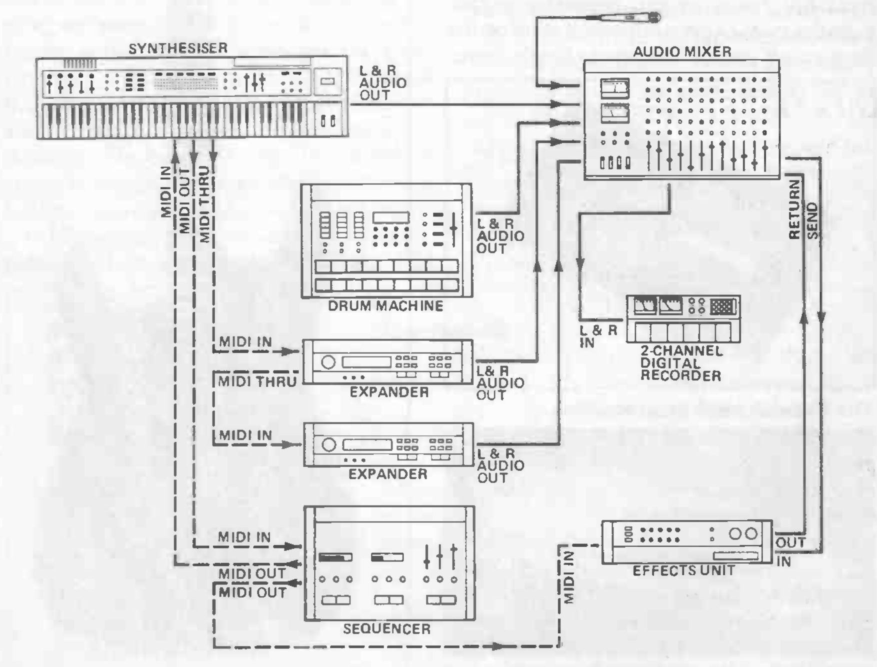
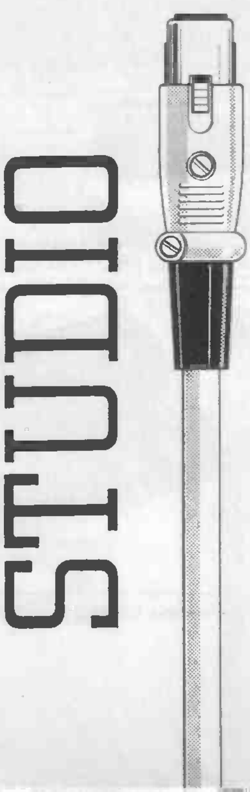


Fig. 3 A four-track set-up with MIDI control



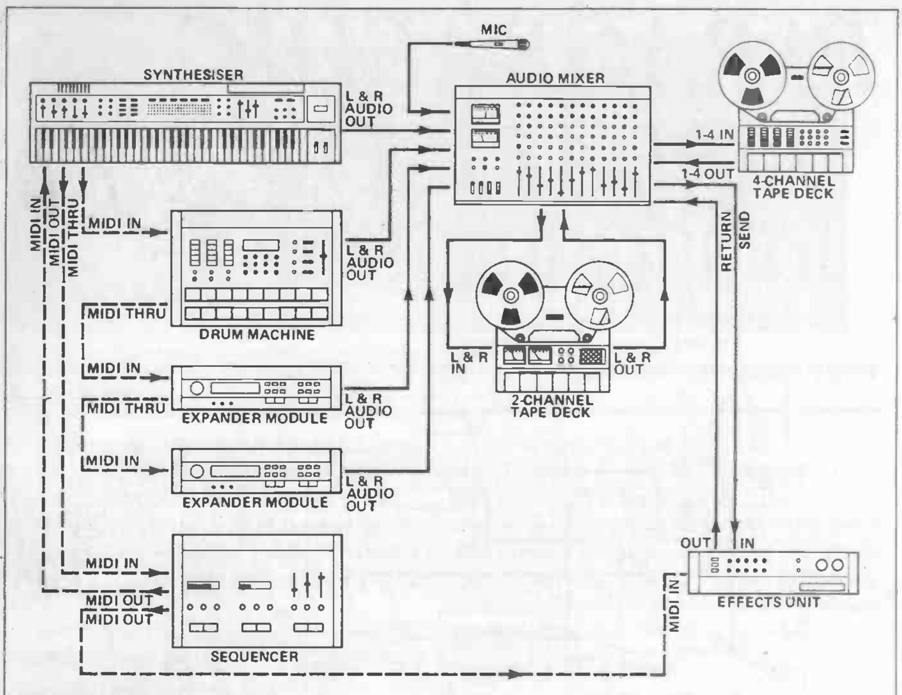
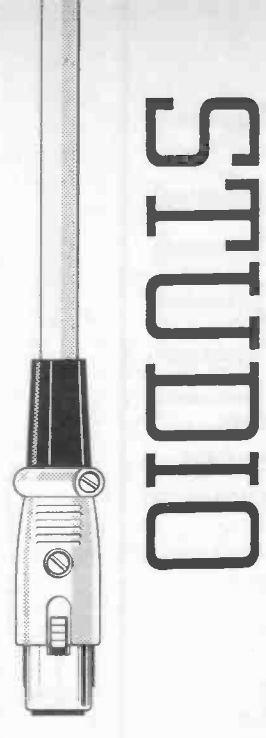


Fig. 4 A digital recording system



All this paints home recording as a bed of roses but even roses have thorns. MIDI generates its own problems, many of which require (unless you're a real masochist) dedicated MIDI processors. All this additional hardware does of course mean some extra bits and pieces on the audio side, most notably a bigger mixer to accommodate your additional sound sources.

By this stage you will be getting very fussy about sound quality and the little noises in the background of your earlier recordings (like the fire engine going by in the distance) will drive you insane if they dare to crop up while you're recording. Now is the time to look seriously at sound proofing. This can be cheap or expensive depending on how well you chose your studio location back at the start. It may be a question of putting up thicker curtains at the windows or of lining the entire room with soundproofing panels. As before, judge with your ears.

Let's Get Serious

You have now spent a sizeable chunk of time mastering all the equipment and you may well want greater flexibility and better sound quality from your studio. You have two paths to follow.

The first and simplest is to upgrade to eight or (if budget allows) 16-track tape facilities. This allows you many little luxuries previously denied like putting individual drums on different tracks so they can be processed separately or indeed putting the effects themselves onto their own tracks for greater sonic scope. Of course, luxuries cost. In this case a recorder and mixer to accommodate eight tracks will set you back at least £2000.

The other avenue is to go full circle back to two tracks but now with the ultimate in sound quality and the minimum between the sound source and the final stereo mix — a digital recording medium such as PCM or the more recently released DAT format, which introduce little or no sound degradation at all.

Cutting out the middle recording process does mean a lot more work doing the mixdown but help is available in the shape of mixer automation.

Again, thanks to MIDI you can obtain systems which allow you to set faders and EQ settings, store them in memory and recall them at will. With this facility you can dedicate a track of your sequencer to mixer control codes to alter mixer settings whenever required within a song and get them right every time. Furthermore these mixer settings can be edited with the same precision as the music itself.

The DMP-7 mixer from Yamaha allows you to do all this and patch in multi-effects built in to the unit, all digitally achieved so you won't even get any noise from the mixers' circuitry.

Coming Up Next

What have the next ten years in store for the home recordist? Eventually we'll all have 'recorders' that are a cross between a big sampler, a sequencer and a DMP-7. This will allow you to record, signal process, mix and output everything digitally.

We have already seen something similar with the Synclavier Direct to Disk system and it's only a matter of time before these sort of systems come down within the price range of people other than Sting and Paul Hardcastle.

We've been through the complete progression of a studio and you should have picked up a few pointers that will save you money along the way. Most importantly you should go for the biggest mixer and the best recorder you can afford right from the moment you decide to take up recording as a serious hobby, or even career.

When going for musical instruments go for MIDI. The standard will be with us for a while yet. Even if it is superceded (as it no doubt will be) there's so much equipment out there sporting the standard, that new equipment, support and repairs will be around for years to come. Most importantly when in doubt choose with your ears (or even with someone else's ears) because at the end of the day they are the final judges.

Casio's VZ-10M rack synth

PARAGRAPHIC EQUALISER

A novel equaliser which is neither just graphic nor parametric — it's both

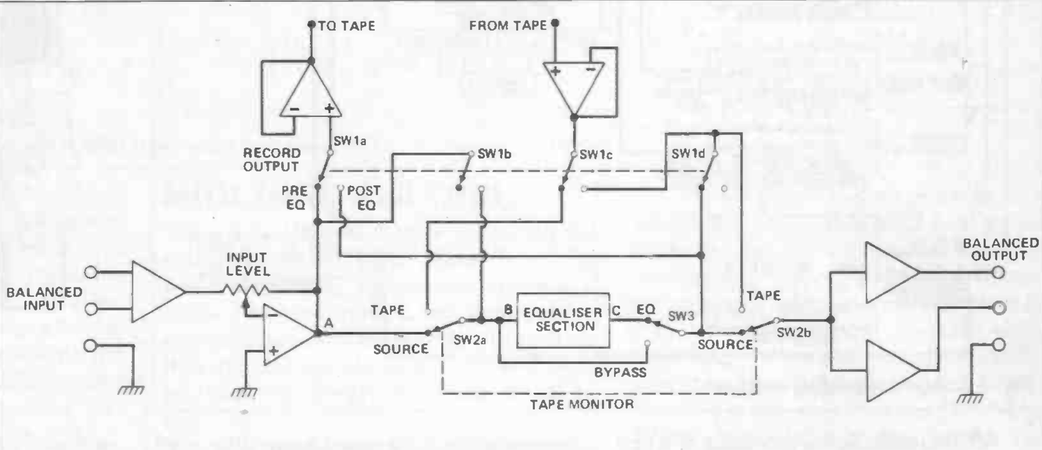


Fig. 1 Block diagram of the complete paragraphic equaliser

Graphic equalisers are a well-known sight in most recording studios and nestling amongst the PA mixers at gigs.

Graphic equalisers consist of a number of small slider potentiometers arranged so that the positions of their control knobs give an indication of the frequency response setting of the unit.

Another type of equaliser, often found as part of the input channels of large studio mixing consoles, is the *parametric*. This is an equaliser in which the three main parameters — amplitude, frequency and Q — are continuously variable.

The Paragraphic Equaliser is a combination of both equaliser types. Although it resembles a graphic equaliser, each slider is accompanied by two rotary controls that allow the frequency and 'Q' of the particular band — the width and tightness of the filter — to be adjusted. As a consequence, an almost infinite number of frequency response variations can be obtained making the unit far more versatile than either type of equaliser.

The circuitry of the equaliser is quite elaborate and its performance is well up to professional studio standards (vastly superior to a fair percentage of the esoteric hi-fi equipment that gets drooled over in certain circles).

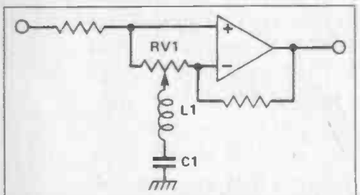


Fig. 2 An example of the type of filter element used in many graphic equalisers

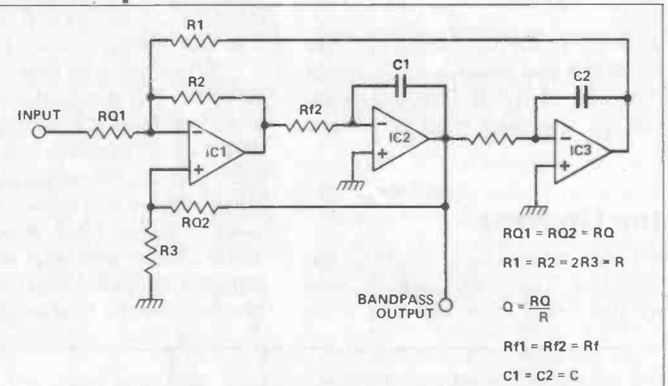


Fig. 3 A state variable filter of the type used in the equaliser

A block diagram of the parametric equaliser is shown in Fig. 1 and it will be seen that the input and output stages are electronically balanced to simplify connection to professional equipment. Provision is made for a tape output and return and the output is capable of being selected from before or after the equalisation stage so that either a flat or an equalised signal can be recorded.

For use with equipment without balanced inputs/outputs a matching transformer or DI box can be used.

The input level control is arranged so that in its central position the unit is operating at unity gain with 10dB of gain or attenuation available at the limits of the control.

Principles

The usual method of obtaining band pass and stop characteristics is shown in Fig. 2. An LC filter is used to shunt the input or feedback signal of a differential amplifier. This arrangement works extremely well but does not allow the centre frequency or Q to be easily adjusted, an essential requirement if you want total freedom over the response variations that can be achieved.

In the parametric equaliser active circuitry is used as the response shaping element in the form of *State Variable filters*. One of these is shown in Fig. 3 and consists of two matching integrators and a summing stage. The output of IC2 has a bandpass characteristic with unity gain at the resonant frequency (f_o) which is decided by the input resistors and integration capacitors of IC2 and IC3 by

$$f_o = \frac{1}{2\pi R_f C}$$

The bandpass Q may be independently adjusted by the input and feedback resistors RQ1 and RQ2, the value being

$$Q = \frac{RQ}{R}$$

The range of each Q control is from 0.5 to 5.5 which in practice has been found to be ideal.

The method of obtaining lift and cut is shown in Fig. 4. The main signal path is through the two inverting stages, IC1 and IC2. The output of IC1 drives the state variable filters and it can be seen that in the cut position of any of the control potentiometers, the associated filter is placed in the negative feedback loop of IC1. In the lift position R3, the input resistor of IC2 is bypassed by the output of the bandpass filter. This control system is extremely symmetrical so that the lift and cut response curves are virtually identical. As the outputs of the filters are added to the main signal path at summing points there is no inter-action between individual controls when several filter stages are used.

HOW IT WORKS

The Paragraphic input stage circuitry is shown in Fig. 5. The RF rejection filter formed by R1, R2, and C1 and C2 has its −3dB point at 88.4kHz and the network around IC1a gives a balanced input with unity gain. Under normal circumstances the input may be DC coupled as it will usually be driven by a balancing transformer or an AC coupled output stage but if there is any danger of DC voltages reaching the input a 10μ capacitor should be placed in series with both R1 and R2. If the unit is to be used exclusively with unbalanced equipment, the input stage may be modified by omitting R1, R3 and C1 and changing R4 to 1k0, R5 to 100k and R6 to 10R.

Following the balanced input is a gain adjustment stage IC1b which allows the overall gain of the system to be changed from unity by plus or minus 10dB. This should be sufficient to cope with most requirements but the swing can be increased to 20dB by reducing the value of R7 and R8 to 1k1.

It will be noted that the track and wiper of RV1 are not isolated from the DC conditions of IC1 and noise may be generated every time the control is adjusted. In practice this is unlikely to cause problems because once the system gain has been set, it will normally remain untouched while the equaliser is in use.

The output of IC1b is AC coupled by C3 and C4 so that clicks are not generated by the switches that follow. The two capacitors are connected in parallel because it has been shown the use of normal electrolytic capacitors in the signal path of high quality audio equipment can cause significant degradation of the signal, sufficient to be quite audible in most cases. The use of special non-polarised electrolytics cures most of the problems but they can cause the high frequency end of the audio spectrum to sound slightly rough.

This effect is cured by the addition of a small value bypass capacitor which employs polycarbonate or polypropylene in its construction.

Both tape input and output signals are buffered from the main signal path so that the operation and performance of the equaliser cannot be affected by external equipment. These buffer stages are shown in Fig. 6.

The main signal path summing stages are shown in Fig. 7 and hardly need an explanation, other than to note that as the overall signal phase is non-inverting, the section may be bypassed by a simple, single pole switch as shown in Fig. 1.

The ten state variable filters are identical expect for the values of the integration capacitors. The filter circuit is shown in Fig. 8, and Table 1 lists the capacitor values and frequency control calibration points.

Both the frequency and Q adjustments may be re-calculated to give different ranges to those suggested. It is not recommended that Q values above 10 are used, but the number of bands may be increased or decreased to suit individual requirements. However the suggested configuration would seem to offer the best compromise between over simplification and operational or constructional over-complexity.

The amount of available lift and cut is controlled by R36 (Fig. 8). The value shown gives a maximum of 10.4dB. A different value may be substituted, the resulting amplitude extremes being given by

$$A(dB) = 20\log\left[\frac{1}{R7}(10 + R7)\right]$$

The equaliser uses the balanced output stage shown in Fig. 9.

The equaliser contains ten stages with octave spacing between them, the nominal operating frequencies being 31.5Hz, 63Hz, 125Hz, 250Hz, 500Hz, 1kHz, 2kHz, 4kHz, 8kHz and 16kHz.

The frequency adjustment range of each band was rather difficult to decide. It would have been nice to have given each band a range of two octaves making, for example, the 1kHz control sweep from 500kHz to 2kHz. Although this is possible, it results in a situation where the 1kHz position of the control is not central, which could cause operational problems. The potentiometer law necessary to give a completely linear frequency sweep is so obscure (a kind of reverse semi-logarithmic) that a compromise was arrived at.

This uses linear controls and the circuit values are arranged so the central position gives the required frequency and the range is from approximately three-quarters of an octave below that frequency to one octave above.

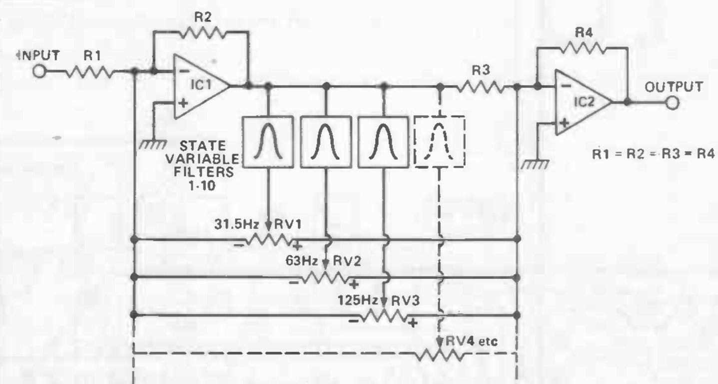


Fig. 4 The amplitude control system

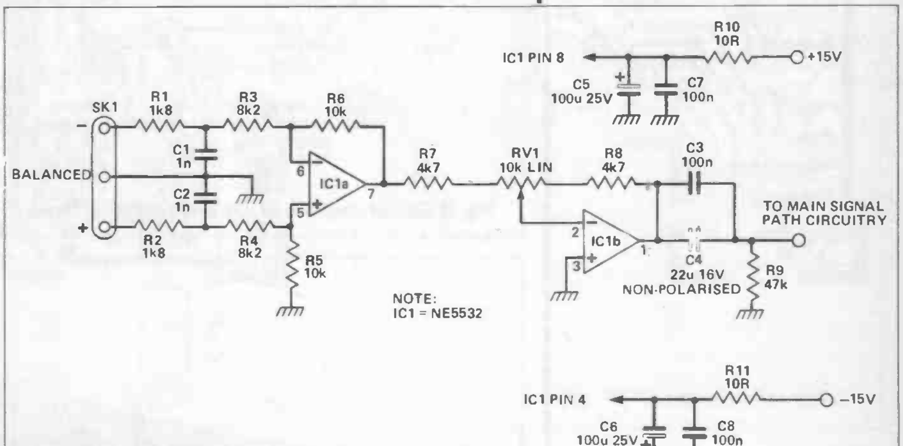


Fig. 5 Circuit diagram of the Paragraphic input

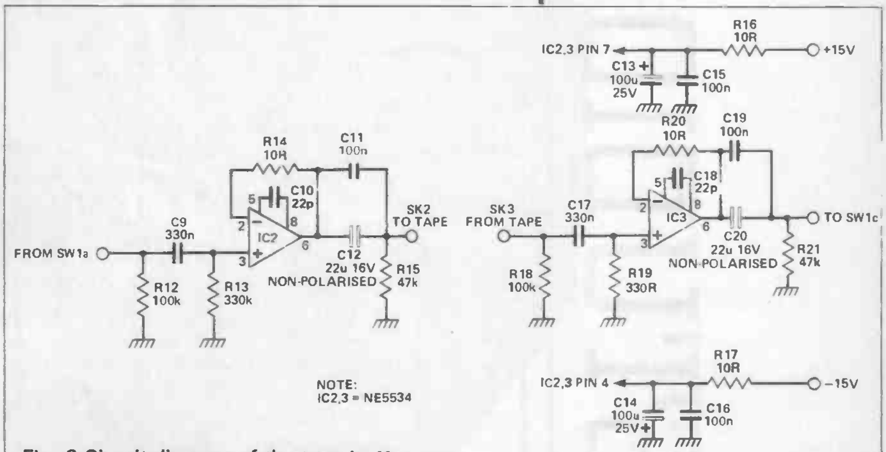


Fig. 6 Circuit diagram of the tape buffers

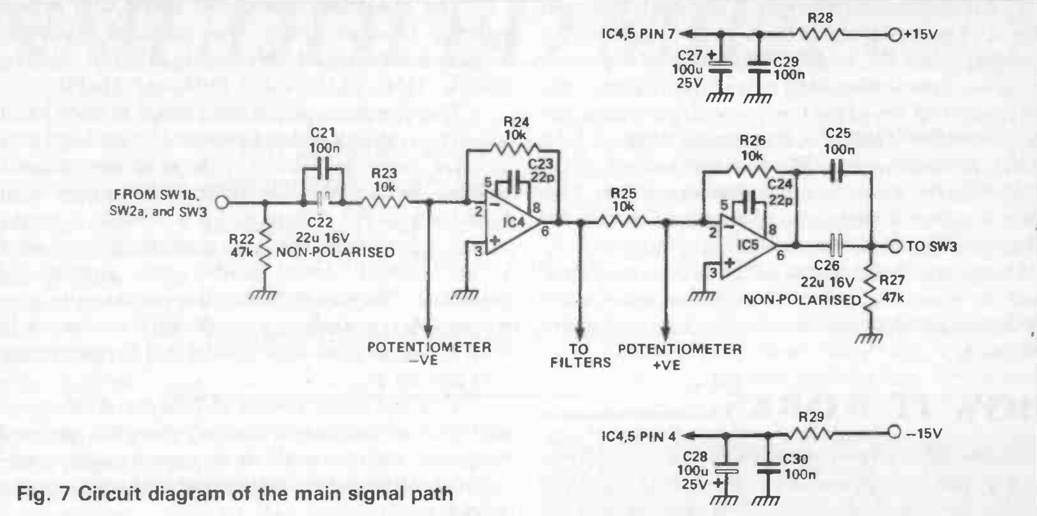


Fig. 7 Circuit diagram of the main signal path

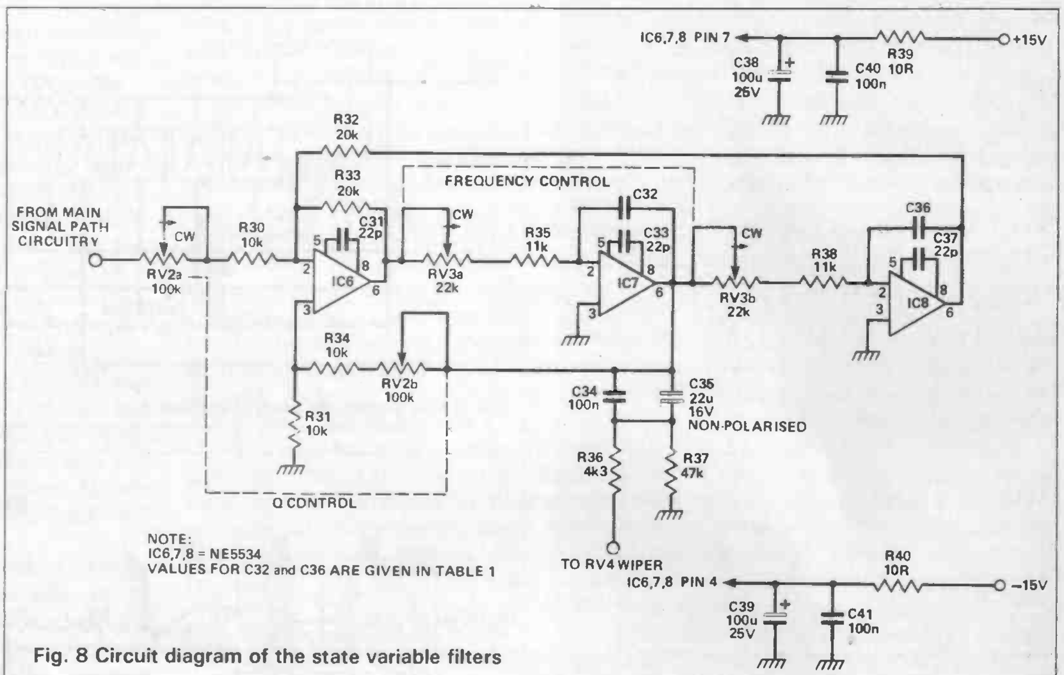


Fig. 8 Circuit diagram of the state variable filters

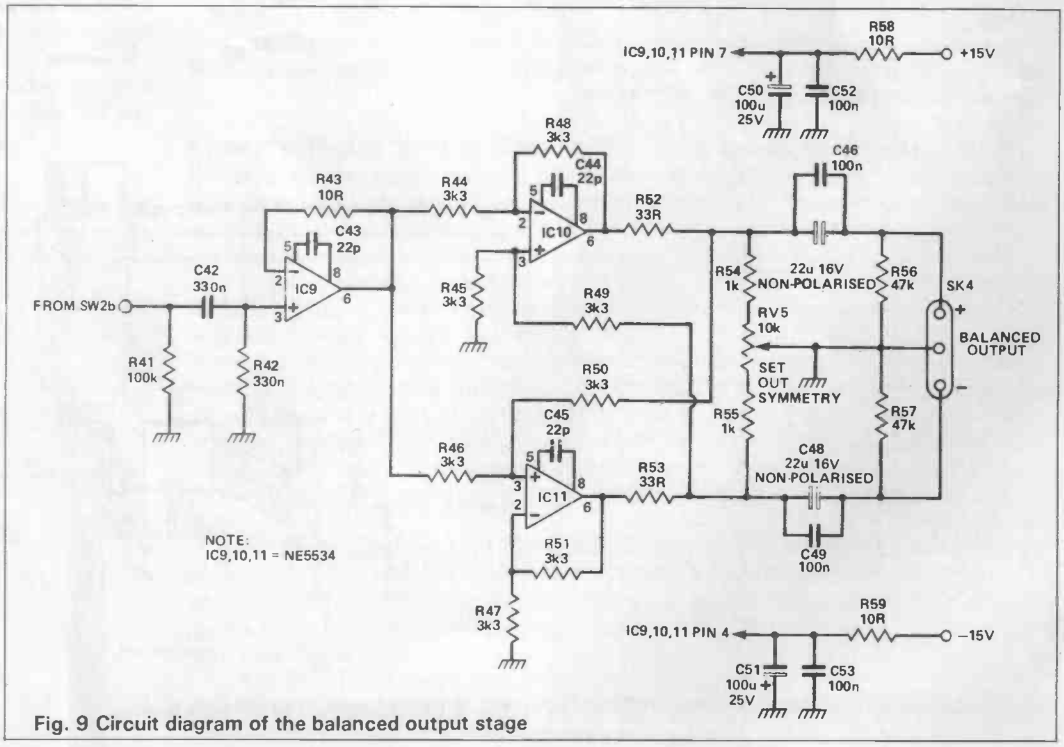


Fig. 9 Circuit diagram of the balanced output stage

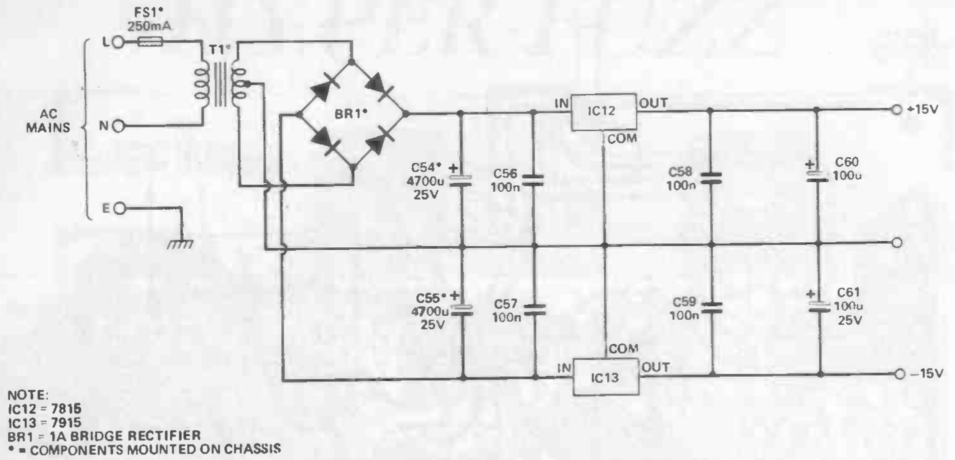


Fig. 10 Circuit diagram of the stabilised power supply

CAPACITOR VALUES	RESULTANT FREQUENCY (Hz) AT: A	B	C	D	E	CAPACITOR VALUES	RESULTANT FREQUENCY (Hz) AT: A	B	C	D	E
220n//10n//390p	20	25	31.5	42	63	3n6//3n6	670	800	1k	1k3	2k
100n//15n//220p	42	50	63	84	125	3n6	1k3	1k6	2k	2k7	4k
47n//10n//330p//220p	84	100	125	170	250	1n8	2k7	3k2	4k	5k4	8k
22n//6n8	170	200	250	335	500	680p//220p	5k4	6k4	8k	10k7	16k
10n//2n2//2n2	335	400	500	670	1k	220p//220p//10p	10k7	12k8	16k	21k4	32k

Table 1. Values for C32 and C36, the frequency-determining capacitors on the filter board. Close-tolerance polystyrene or polycarbonate types should be used throughout

PARTS LIST: FILTER BOARD

RESISTORS (all ¼W 1% metal film)

R30,31,34	10k
R32,33	20k
R35,38	11k
R36	4k3
R37	47k
R39,40	10R
RV2	100k lin dual gang rotary pot
RV3	22k lin dual gang rotary pot
RV4	10k lin slider pot

CAPACITORS

C31,33,37	22p polystyrene
C32,36	see Table 1
C34	100n polycarbonate
C35	22µ 16V non-polarised radial electolytic
C38,39	100µ 25V radial electrolytic
C40,41	100n polyester

SEMICONDUCTORS

IC6,7,8	NE5534

MISCELLANEOUS

PCB. IC sockets. Terminal pins for C32 and C36.
(These are the parts for one filter board only).

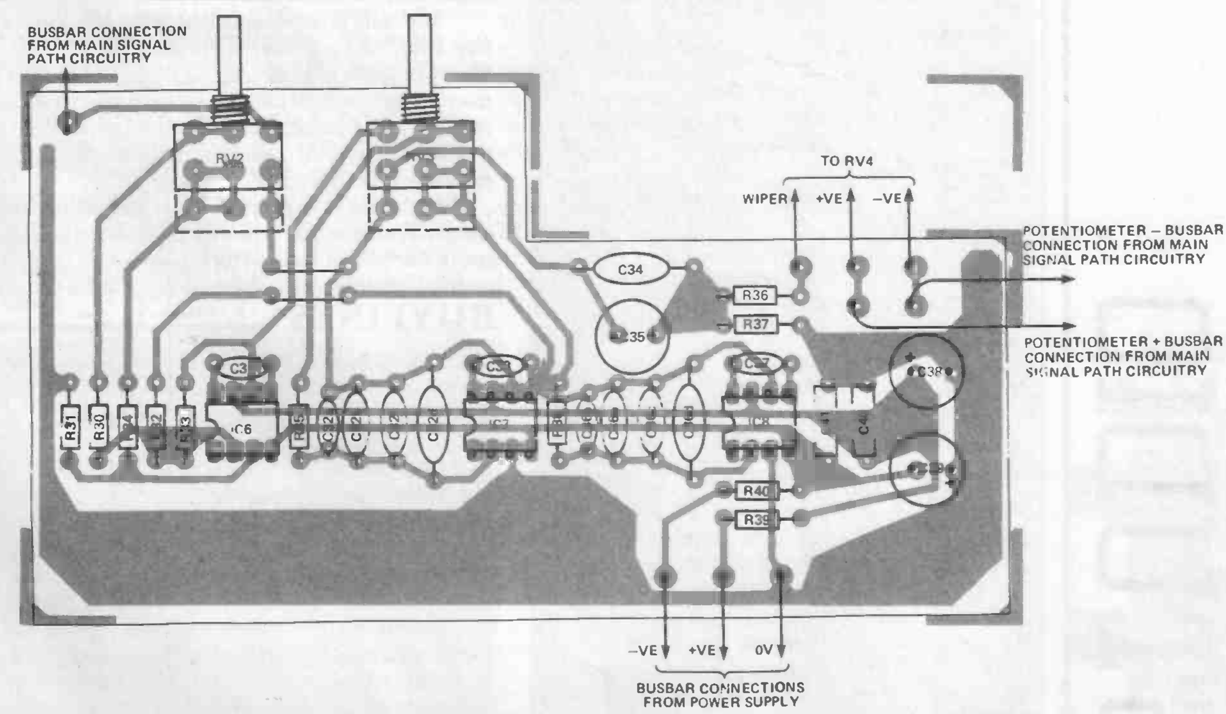


Fig. 11 Component overlay of the filter PCB. You will need one filter board for each channel of the equaliser

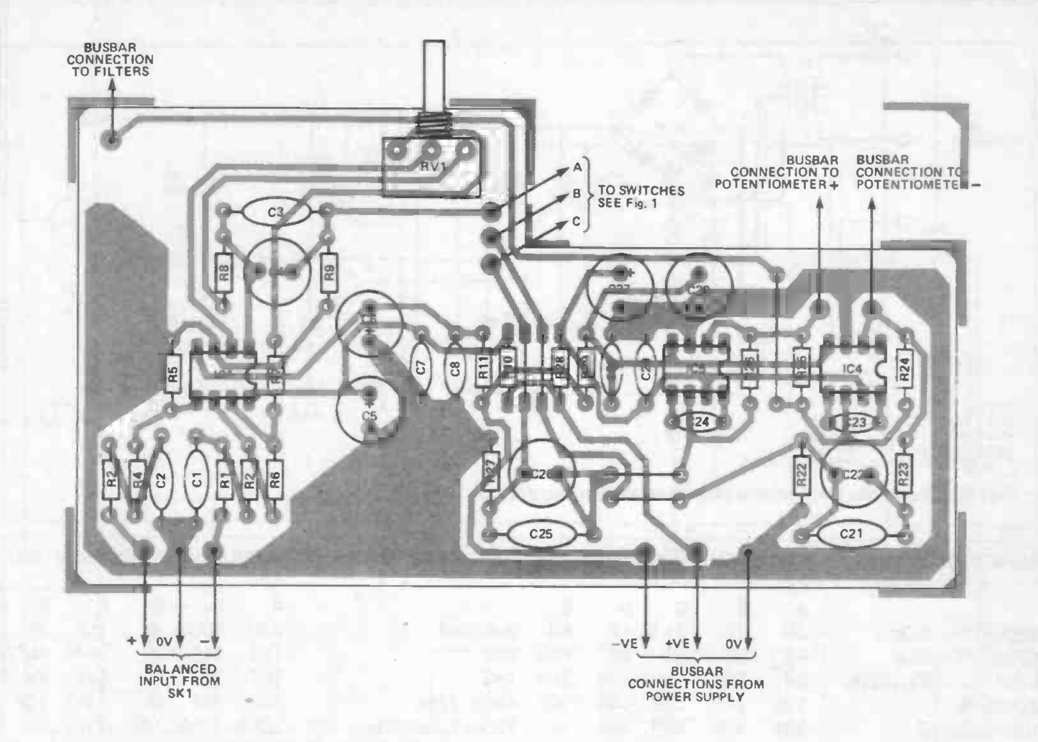


Fig. 12 Component overlay of the input stage and main signal path PCB

PARTS LIST: MAIN BOARDS

RESISTORS (all ¼W 1%)	
R1,2	1k8
R3,4	8k2
R5,6,23-26	10k
R7,8	4k7
R9,15,21,22,27,56,57	47k
R10,11,14,16,17,20,28, 29,43,58,59	10R
R44-51	3k3
R52,53	33k
R54,55	1k0
RV1	10k 1in rot pot
RV5	10k moulded preset
CAPACITORS	
C1,2	1n0 polycarbonate
C3,11,19,21,25,46,49	100n polycarbonate
C4,12,20,22,26,47,48	22µ 16V non-polarised radial electrolytic
C5,6,13,14,27,28,50, 51,60,61	100µ25V radial electrolytic
C7,8,15,16,29,30,52, 53,56-59	100n plyester
C10,18,23,24,43-45	22p polystyene
C54,55	4700µ 25V can electrolytic
SEMICONDUCTORS	
IC1	NE5532
IC2-5,9-11	NE5534
IC12	7815
IC13	7915
BR1	1A bridge rectifier
MISCELLANEOUS	
FS1	250mA anti-surge fuse and holder
SK1,4	XLR 3-pin connector (see text)
SK2,3	phono socket
SW1	4-pole 2-way toggle switch
SW2	2-pole 2-way toggle switch
SW3	single-pole 2-way toggle switch
T1	15-0-15 25VA mains transformer

PCB. IC sockets. Case. Knobs. Wire. Nuts and bolts.

Construction

The paragraphic equaliser is built on a number of boards (Figs. 11-13). One each of the input and main signal path board plus the output, tape buffer and power board are used and one filter board for each channel.

Each filter stage is built onto a separate circuit board attached to the front panel by the frequency and Q adjustment potentiometers. The board layout is shown in Fig. 11. The cutout area allows different types of slide fader to be used and ensures the rotary controls can be in line with the fader. Remember to purchase sliders which can be mounted by screws from the front and use a dummy front panel if you don't want the screw heads to show.

The only components that differ between one filter board and another are the integrator capacitors. Plenty of space has been left for these and instead of attempting to mount the various capacitor types and sizes in the normal way, small terminal pins should be pressed through the capacitor mounting holes, and the components soldered to these.

Once the boards have been assembled, they should be attached to the front panel making sure they are in the correct order. The busses which carry the various common connections should be fed through

BUYLINES

Radial non-polarised electrolytics are not readily available to the amateur but axial 50V types are sold by Maplin, Circuit and Electrovalue and should fit into the space if stood on end.

The polystyrene or polycarbonate capacitors used for C32 and C36 should ideally be 1% tolerance types but if you use 5% types instead you should omit some of the smallest capacitors from the parallel combinations listed in Table 1. There is little point in using either the 330p or the 220p in parallel with 5% tolerance 47n and 10n capacitors, for example, because the tolerance on the larger capacitors considerably exceeds the value of the smaller ones.

Maplin stock a range of 1% tolerance polystyrene capacitor which covers some of the values needed and it is perfectly permissible to use 5% small value capacitors in parallel with 1% tolerance large values. Watford, Rapid, Cricklewood and Technomatic are among those who stock both the NE5532 and the NE5534.

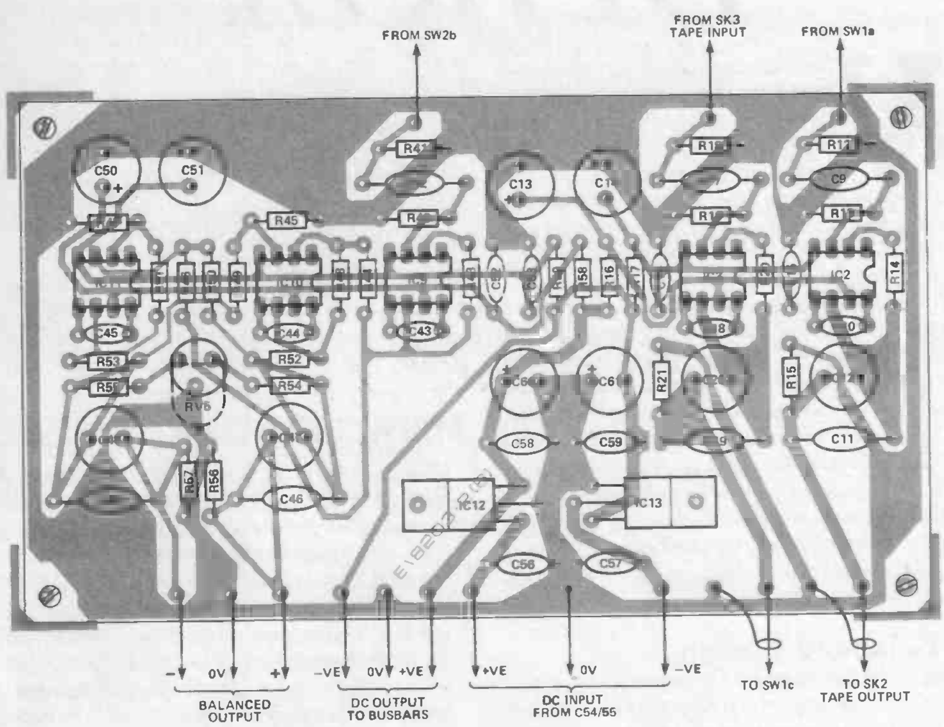


Fig. 13 Component overlay of the tape buffer, balanced output and PSU board

the circled holes and continued to the main signal path board (Fig. 12). A suitable gauge of tinned copper wire should be used for the busses and this may be insulated with short lengths of sleeving if it is felt that there is any danger of short circuits occurring.

The output stage and tape buffer amplifiers are on a separate circuit board, together with the power supply stabilisers (Fig. 13). This board may be mounted at any convenient point within the cabinet but should be kept as far away as possible from the mains transformer and any mains wiring. Connections between the circuit boards and function switches should prove quite straightforward, using Fig. 1 as a reference. Due to the low impedance of the switched connections, unscreened wire can be used throughout.

If the recommended balanced inputs and outputs are employed, it is suggested that professional XLR 3-pin connectors are used. These can be obtained at a reasonable price from a number of sources and will remain reliable for many years — unlike some of their lesser brethren. There is a permanent confusion, even in the professional world, over the correct wiring of these connectors but the generally accepted standard is all signal inputs via XLR 3-way chassis mounting sockets (female) and all signal outputs via XLR 3-way chassis mounting plugs (male). The wiring to both plugs and sockets is

Pin 1 — Earth
Pin 2 — Signal +
Pin 3 — Signal −

For unbalanced inputs or outputs, connect pin 3 to pin 1.

Unbalanced versions may also be fitted with DIN or phono sockets.

For safety reasons, the metal cabinet must be connected to earth via the mains lead. If the signal earth is connected to the cabinet in any way, a hum loop will probably be formed whenever the equaliser is used with other equipment having common mains and signal earths. The best approach to this problem is to experiment once the unit is working correctly. As an interim measure, make sure that the signal earth is floating at this stage.

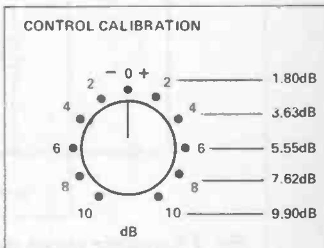


Fig. 15 Calibration of the input level control

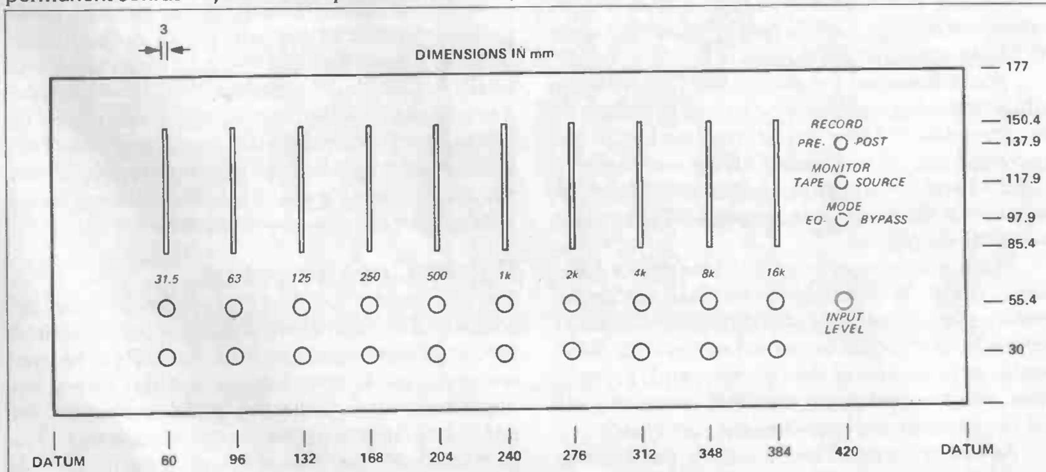


Fig. 14 Suggested front panel layout for the Pararaphic based on a 4U (7in) height 19in racking case

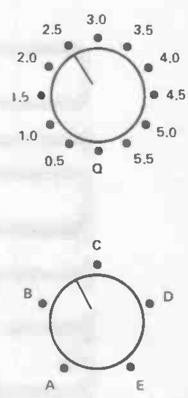


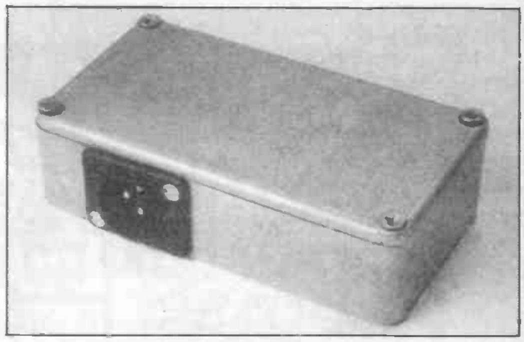
Fig. 16 Calibration of the Q and frequency controls.

DI BOX

Give your performance a shot in the arm with a simple direct injection box

Whatever your instrument, when you move out of the bedroom and into the studio or onto stage you'll need to connect it to a mixer or PA system. The easy way to do this is to mike up your backline amp but that not only ties up an expensive microphone but also introduces all the possibilites of distortion, feedback, cabinet resonances, microphony and the like.

Of course, many guitarists revel in this 'personalisation' of their sound but for the rest of us a DI box is the answer. This is a unit which takes the incoming signal from the instrument (or from the microphone in the case of an acoustic instrument) and splits it to produce two outputs, one of which is fed to the main PA while the other is taken to a nearby amplifier and speaker controlled by the musician. A DI box (as these units are generally known) has a high impedance input, a low impedance balanced output for the mixer and either a high or low impedance output for the stage amplifier.



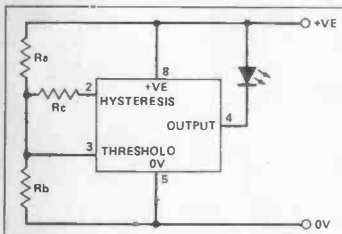


Fig. 1 Low voltage warning circuit using the Intersil ICL8211

HOW IT WORKS

IC1a works as a buffer with a selectable gain, R4 and C2 being added only if voltage gain is required. The offset voltage on the inputs is minimised by having a similar DC resistance on each input. The output of this buffer drives the in-phase output and also unity gain inverter IC1b which in turn drives the other output. Both outputs have their impedance set by series resistors and are DC blocked by electrolytic capacitors. The capacitors are polarised by load resistors R11 and R12.

The low battery detector is based on a purpose designed IC which contains a very low current band gap voltage reference, a comparator and a current limited output drive circuit. For this reason, the LED needs no current limiting resistor.

Well Balanced Design

The design to be described here is based around a very low noise dual op-amp which allows it to be used at low signal levels without significant noise problems. The gain is normally set at unity but a voltage gain of 2 can be achieved by making a few component changes. The unit operates from a single 9V battery and powers up automatically when a jack plug is inserted into the input socket. In addition, a low-voltage detector is included which lights an LED when the battery needs replacing.

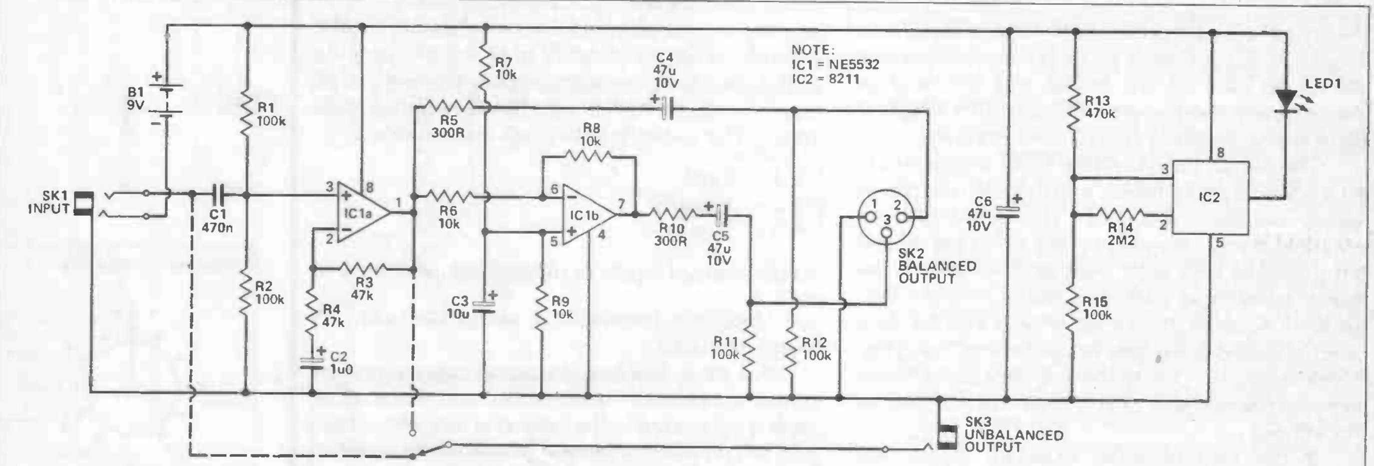


Fig. 2 Complete circuit diagram of the direct injection box

The unbalanced output can be taken from the output of the first op-amp which gives a reasonably low impedance drive or, if preferred, can be connected directly to the input. The mixer output is balanced and can either be used directly with high impedance balanced inputs or set at 600R by adding two 300R resistances.

The normal type of balanced line to use for audio work is 600R. To be completely correct, the source resistance for each signal connection should be 300R and each one should be terminated with a 300R resistance to ground at the receiving end. In many cases, a high impedance is used at the receiving end and the sending end impedance is just 'low'.

As long as the signal level is suitable, this unit may be used as a proper 600R driver. The 5532 op-amp specified has very low noise so very low level signals may be used without severe penalty.

If the box is operated with 300R output resistors into a terminated 600R line, the output signal will be potted down by 2:1. To compensate for this, voltage gain is provided by the addition of optional components R4 and C2. Equally, if the input signal is of a very low level, adding these components will boost it to a level above that of the interference picked up on the line. This is particularly useful to prevent buzz from phase controlled lamps being audible on mic circuits. For the purpose of driving a balanced line, a voltage gain of times two is required.

Power Consumption

The one drawback of the excellent NE5532 dual op-amp is that its current consumption is quoted as 8mA typical, 16mA maximum. If the DI box is to be used with reasonably large signals and not into a low impedance load, it may be preferable to use the LM358 op-amp to cut the power consumption. The gain bandwidth product of this device is only 1MHz so there is little scope for providing voltage gain without the risk of degradation of sound quality.

Another alternative would be the TL072. This is a BIFET device so it would be possible to use a high input impedance if necessary — 1M0 for example. The maximum current consumption of this is 5mA total so the battery life should be reasonable.

If the application requires substantial voltage drive into a 600R load, the DI box may be constructed with 25V rated electrolytic capacitors and powered from two 9V batteries in series. There is no room in the case of the prototype unit for another battery, so a large sized case would have to be used. The low battery warning would have to be recalculated to work at a different voltage as well of course, to give warning before the unit stopped working correctly, rather than afterwards.

Mention of the power consumption brings us neatly on to the battery voltage detector. This uses an Intersil IC, the ICL8211. This handy chip draws a quiescent current of about 25μA and provides a current limited LED drive which switches on when the voltage on its threshold input falls below 0.15V (Fig. 1).

The threshold voltage for the LED to switch off is given by the formula

$$V = 1.15 \times \frac{Ra+Rb}{Rb} \text{ volts}$$

Hysteresis is added by Rc (R14 in the final circuit). This does not affect the switch off voltage but the switch on voltage is lowered. This voltage is calculated from the formula

$$V = \left(\frac{Ra \times Rc}{Ra+Rc} + Rb\right) \times \frac{1.15}{Rb}$$

The component values specified in the circuit diagram give nominal switching voltages of 6.55V (off) and 5.60V (on). If this end of life voltage is too low, it may be raised by reducing RI5.

Construction

Just about any metal box can be used for the DI box. Avoid ones with PCB slots on the inside as these make fitting the jack sockets more difficult. A 110×60×30mm diecast box was used for the prototype and the arrangement of the components in this case is shown in Fig. 4.

The input jack is a stereo socket even though the signal is mono. This enables the extra contact to be used to connect the battery only when the input is plugged in. The negative wire from the battery clip sould be connected to the middle connection of the socket.

The unbalanced output socket can be connected either to the output of the first stage (a pad is provided on the PCB for this) or directly to the input.

Thin co-ax cable should be used for the input connection and three wires twisted together used for the balanced output. The convention is pin 3 for the out-of-phase signal (upper pad), pin 2 the in-phase signal (lower pad) and pin 1 ground.

If a regulated power supply is available it should be used for initial testing. Set the power supply to about 4V and connect up. If the LED does not light, then reverse its connections and try again. Once the LED works, increase the voltage until the LED goes off then reduce it until the LED switches on again. Measure the voltage and check that it is about 6.5V. Individual units may vary due to component tolerance, but if the voltage is not acceptable the value of R13 or R15 should be changed.

Now apply 9V either from a battery or from the power supply and use a voltmeter to check the op-amp output pins are at about 4.5V and that the outputs themselves are at 0V. If they are not, the most likely fault is a reversed electrolytic capacitor. Finally, connect up a signal source and a suitable amplifier and check that everything works correctly and that the sound is all it ought to be.

BUYLINES

Most of the components are widely available from the usual mail-order suppliers. Maplin (Tel: (0702) 552911) stock the 8211 and the diecast box used for the prototype was obtained from Cirkit (Tel: (0705) 669021). We used a low-cost plastic XLR connector for SK2 but the standard metal type would be just as suitable. The PCB is available from our PCB service.

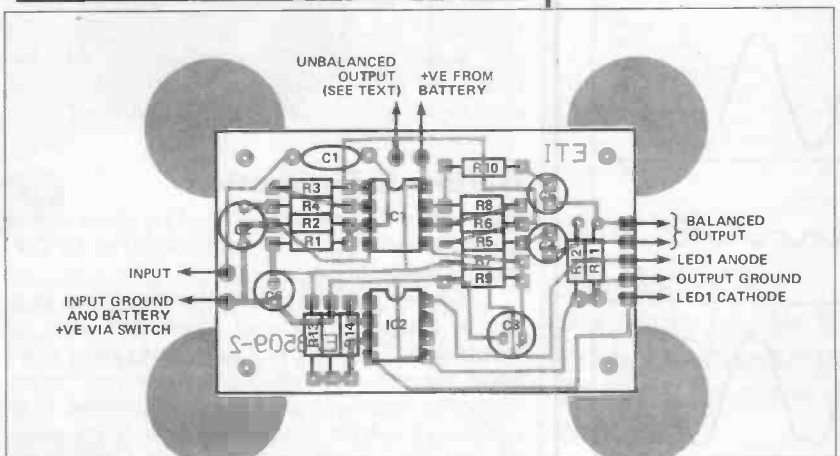


Fig. 3 Component overlay for the DI box PCB

PARTS LIST

RESISTORS (all ¼W 5% unless otherwise stated)	
R1,2,11,12,15	100k
R3	47k
R4	47k (see text)
R5,10	300R 1%
R6,8	10k 1%
R7,9	10k
R13	470k
R14	2M2 (see text)
CAPACITORS	
C1	470n polyester
C2	1µ0 tantalum (see text)
C3	10µ 10V radial electrolytic
C4,5,6	47µ 10V radial electrolytic
SEMICONDUCTORS	
IC1	NE5532
IC2	8211
MISCELLANEOUS	
B1	9V battery
SK1	¼in stereo jack socket
SK2	XLR 3-pole chassis plug
SK3	¼in mono jack socket

PCB. Case. Panel-mounting bush for LED1. PP3-type battery connector. Nuts and bolts to mount SK2 and the PCB.

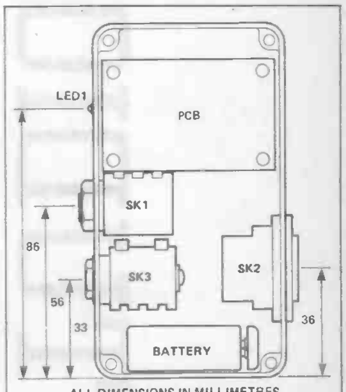


Fig. 4 Layout of the major components within the case

AURAL ACTIVATOR

Excite your music and your eardrums with this Activator for studio or stage

Since the idea of aural enhancement was first promoted by Aphex, much mystique has grown to shroud the technique — not surprisingly because the promoters knew they were on to a good thing. In fact, this technique which 'miraculously' cleans up dodgy recordings and adds sparkle to good ones is astonishingly simple.

The brain seems to rely on high order harmonics for much of our perception of detail in complex sound structures. These harmonics, being of low amplitude, are the first to be lost in recording due to noise and poor high frequency response.

If there was a way of restoring these low level, high order harmonics then much of the original clarity and detail of the recorded sounds would be recaptured. Improbable? Right. So let's cheat, and — surprise, surprise — we have the technology!

Harmonic Generation

Imagine that the sine wave of Fig. 1a is mixed with a small amount of third order harmonic of Fig. 1b. The result is shown in Fig. 1c.

Fourier symmetry (as we physicists like to say) tells us that we could bend a pure sine wave to resemble the waveform of Fig. 1c, using some non-linear network — in effect creating a third order harmonic component added to the fundamental. Fear not about problems with complex sounds, since any waveform can in principle be reduced by Fourier analysis to component sine waves.

We can bend a sine wave quite easily. Simply clipping it as in Fig. 1d will generate harmonics — the heavier the clipping, the more harmonics are produced (the more it approximates a square wave).

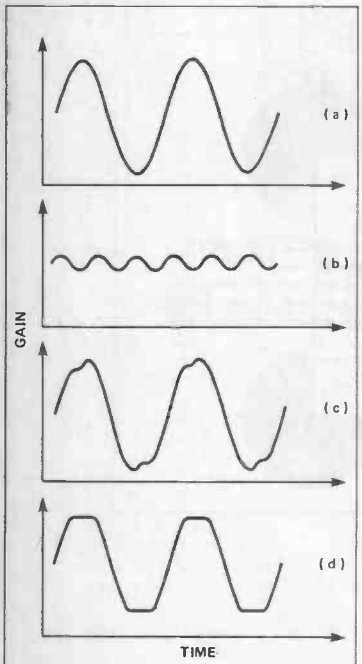
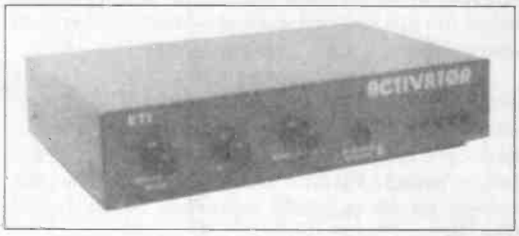


Fig. 1 Harmonic generation



The amount of clipping and consequent harmonic generation are very much dependent on signal level. In the Activator, a sophisticated system is used so that the 'bending' is like that of Fig. 1c and is independent of signal amplitude.

Doing it With Frequency

But wait a minute. What we are talking about is severe distortion, isn't it? And distortion is the last thing we want in quality audio.

Well, no and yes — in that order. The difference between aural enhancement and mere distortion is one of degree and frequency. Clipping a sine wave is rather severe and generates very large amounts of harmonics. Also the effect we seek only works at high frequencies. Thus the signal must be high pass filtered so that only frequencies above a few kilohertz are 'bent'. If you apply harmonic generation to the whole spectrum, the result does just sound like distortion. Also, the amount of harmonic generation must be kept low. Only then does the whole effect come to life.

When recording individual instruments or voices the Activator will create a sense of presence in a way that old fashioned presence controls never could. The frequencies which are 'activated' also happen to be those which carry most stereo information — so a

HOW IT WORKS

Figure 2 is the complete diagram of the stereo circuit.

Balanced and line level inputs are debalanced by IC1c and d while unbalanced inputs are buffered by IC2c and d. The resulting signals are mixed by IC2b and a. From this point the signal is split and is fed directly to the output drivers of IC8 and the harmonics-generating side chain circuitry.

IC3a and b are dual operational transconductance amplifiers (OTAs). IC4 and IC5 form a voltage controlled high pass filter with a 12dB per octave roll off below the centre frequency, which can be swept between 2kHz and 8kHz by the *Tune* control, RV2.

The *Selectivity* control, RV3, enables the negative feedback from the bandpass output at IC4 and IC5 pin 8 to be reduced, thus peaking up the response by about 12dB around the centre frequency.

The high pass filter output at IC3 pins 1 and 7 passes to the non-linear harmonic generation circuitry. In this design the harmonics are generated by abusing a 571 compander chip, IC6. It is provided with time constant capacitors C11 and C23 which are ten times too small. This method provides symmetrical 'bending' of the waveform which is fairly independent of signal level and is superior to other methods employing diode networks or the non-linear characteristics of OTAs.

The output of IC6 is inverted. Mixing it back with the filter output in the correct proportions (set by R25 and R27 and by R65 and R67) through IC7 cancels out the fundamental and leaves the harmonics which have been generated. It is these which are added to the original unprocessed signal via the *Process Level* pot RV1.

SW1a, R80, C25, Q1 and Q2 provide silent switching to prevent clicks when the effect is switched in and out, which it frequently will be in order to make 'before and after' comparisons. The other half of SW1 switches the bottom LED of the bargraph (LED6) from red (bypass) to green (process).

The rest of the circuitry concerns the display. The object of the exercise is to make the bargraph respond only to the level of the generated harmonics, rather than to the overall signal level.

To achieve this an analogue divider is built around IC9a and OTA IC10. The left and right harmonics signals emerging from Q1 and Q2 are summed by IC9a but the gain of IC9a is controlled by IC10, in turn controlled by the level of the input signal. In other words as the input signal level increases the gain applied to the harmonics signal is reduced.

The input signal is rectified by IC1a, D3, C26 and R96 and level shifted by IC1b in order to produce the control voltage which sets the gain of the OTA. The output of IC9a is amplified and rectified by IC9b, D4, C30 and R90. The resulting DC control voltage is fed to the bar driver chip, IC11.

stereo unit like the Activator enhances the stereo effect, too.

It's also worth mentioning the beneficial effects the Activator will have on muddy sounding cassette tapes. If there is no treble in the recording, then boosting the treble band with a graphic equaliser will achieve nothing but the amplification of tape hiss.

Aural enhancement, however, does not require treble to be present since it uses upper mid-frequencies to synthesise new high frequencies. This basic difference sets aural enhancement apart from any simple form of equalisation.

Dolby B was invented when the quality of cassette tape was very poor. The noise reduction it

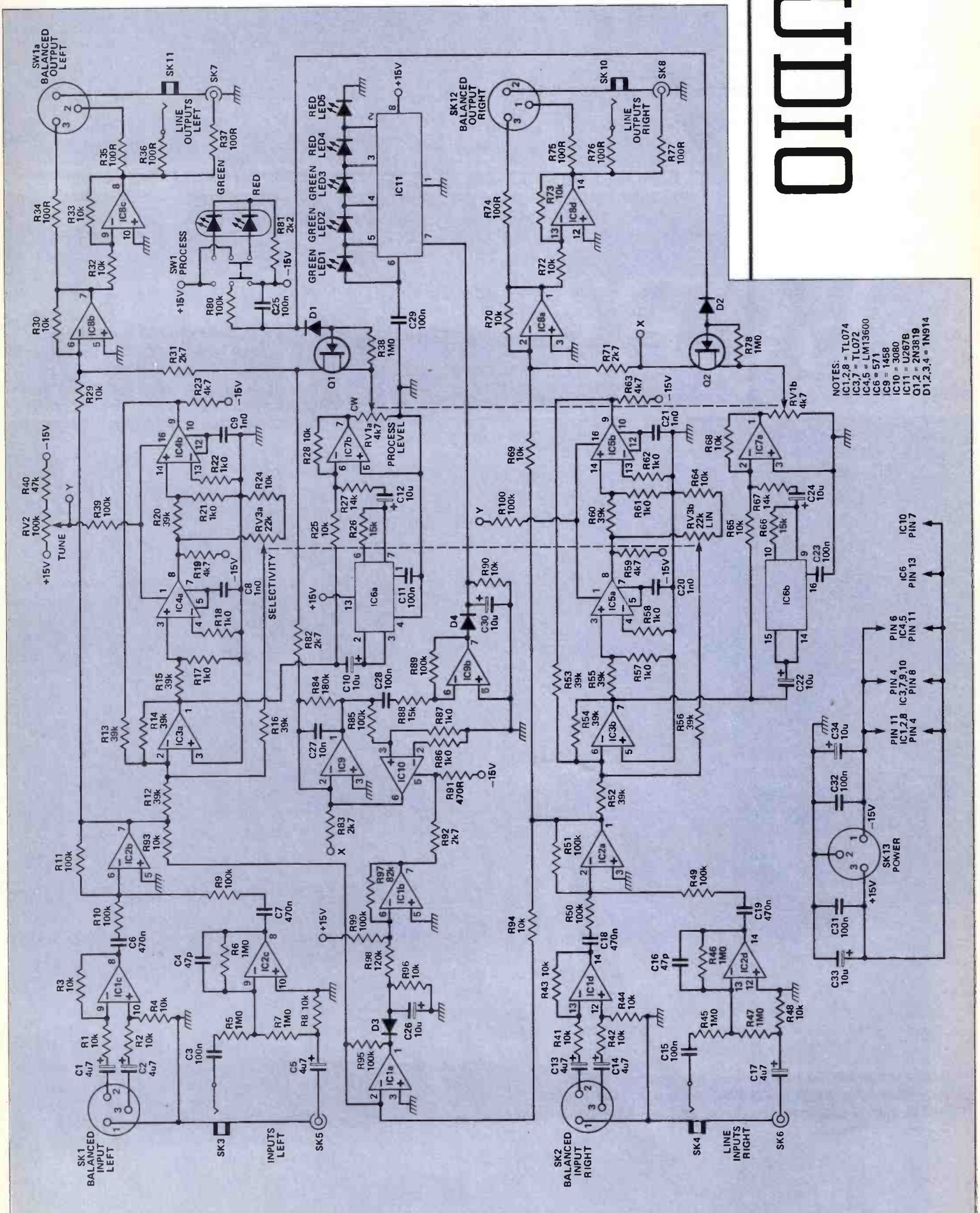


Fig. 2 Complete circuit diagram of the Activator

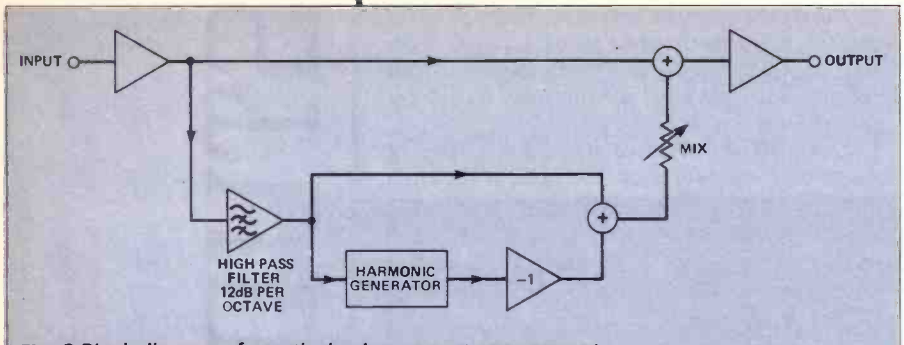


Fig. 3 Block diagram of practical enhancement arrangement

afforded made listening acceptable. The vast improvement in cassette and equipment quality has made Dolby (arguably) all but redundant, simply killing the top treble end of recordings. In fact, many people record tapes with the Dolby on to act as a treble boost and then play them back with the Dolby switched off. If you play tapes back through the Activator you can leave the Dolby on, taking advantage of the noise reduction without sacrificing prized high frequencies.

Active Design

A practical system is shown in block diagram form in Fig. 3. Notice that after the harmonic generator the resulting signal is mixed in antiphase with the signal

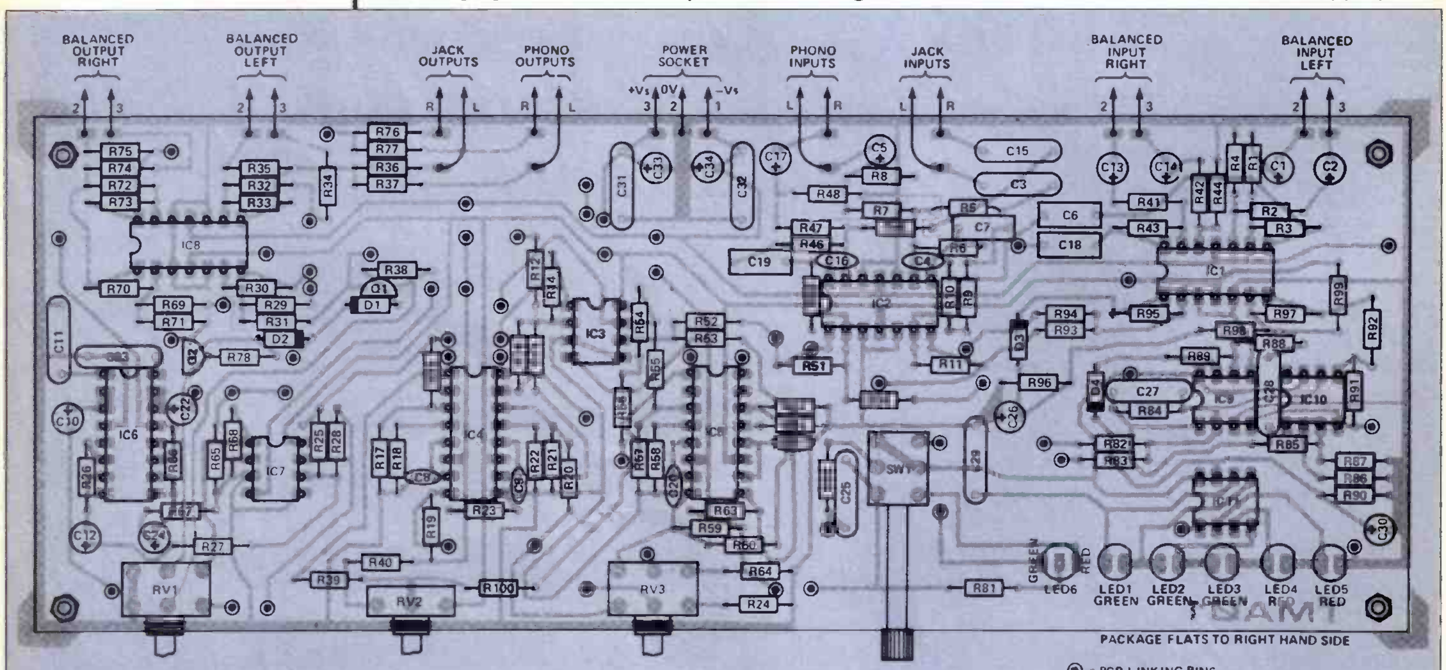


Fig. 4 The component overlay of the Activator

emerging from the filter so as to cancel out the fundamental, leaving only the newly synthesised harmonics to be added in the desired proportion to the output. There is very little change in overall signal amplitude when the harmonics are added. In the Activator, there is also a very slight reduction in high frequency level as the *Process* control is advanced, which keeps the subjective volume constant.

The high pass filter used in the Activator is voltage controlled with a 12dB per octave roll-off below a centre frequency — variable between 2kHz and 8kHz. It also has variable resonance or *Selectivity*. It is possible to tune and emphasise particular frequencies within the signal to be processed.

The Activator is designed so that whatever size signal you put in, the same size signal is output. Harmonic enrichment maintains the correct proportions. The relative level of different order harmonics is determined by a time constant rather than by signal amplitude, avoiding the use of separate 'Drive' and 'Mix' controls as on most existing designs. These interact and their combined effect is largely to make the display function correctly.

The Activator display is designed to indicate the proportion of harmonics added to the original signal, independent of overall signal level. The display will not respond very much to signals peaking much below −10dBm but, since −10dBm is the standard recording level, no problems are envisaged. The unit is equally happy with 0dBm signals.

The Ins And Outs

Line level, balanced and unbalanced inputs and outputs are standards in this type of unit. Phono sockets have been added to the Activator with the domestic hi-fi owner in mind. The unit will handle signals well in excess of +10dBm before clipping, unlike some of its more prestigious and expensive cousins.

Construction

Few problems should be encountered in using the double-sided PCB. The most important thing is to ensure that all the track-linking pins are soldered on both sides of the board! Enough said. Assemble components in order of height, ensuring correct orientation of diodes, LEDs and, where appropriate, capacitors. It is a good idea to bench-test the completed board prior to bolting it into the case and wiring it to the sockets (Fig. 5).

Use

With the Activator processing a signal (left-hand LED green) adjust the *Process Level* control so that the display peaks at about half full scale. The ultimate decision must be based on your listening judgement, but remember that excessive high frequency is very easy to get used to and you only find out about it when you come back to your recordings after a break. Suffice to say that if the display is constantly in the red, you are overdoing it!

The Activator will impart an up-market, up-front quality to live sounds. In recording almost anything will benefit — dull guitars, lifeless pianos, drums and,

PARTS LIST

RESISTORS (all 1% metal oxide)	
R1-4,8,24,25,28-30, 32,41-44,48,64,65, 68-70,72,73,90,96	10k
R5-7,38,45-47,78	1M0
R9-11,33,39, 49-51,80,85, 89,93,94, 95,99,100	100k
R12-16,20, 52-56,60	39k
R17,18,21, 22,57,58,61, 62,86,87	1k0
R19,23,59, 63	4k7
R27,67	14k
R26,66,68	15k
R31,71,82, 83,92	2k7
R34-37,74-77	100R
R40	47k
R81	2k2
R84	180k
R91	470R
R97	82k
R98	120k
RV1	4k7 stereo log pot
RV2	100k lin pot
RV3	22k stereo lin pot
CAPACITORS	
C1,2,5,13,14,17	4µ7 40V radial electrolytic
C3,11,15,23,25,28, 29,31,32	100n polyester
C4,16	47p ceramic
C6,7,18,19	470n polycarbonate
C8,9,20,21	1n0 ceramic
C10,12,22,24,26,30, 33,34	10µ 40V radial electrolytic
C27	10n polyester
SEMICONDUCTORS	
IC1,2,8	TL072
IC3,7	TL072
IC4,5	LM13600N
IC6	NE571
IC9	1458
IC10	CA3080E
IC11	U267B
Q1,2	2N3819
D1-4	1N914
LED1-3	Green LED
LED4,5	Red LED
LED6	Tricolour LED
MISCELLANEOUS	
SK1,2	female chassis XLR
SK3,4,9,10	mono in jack socket
SK5-8	phono socket
SK11,12	male chassis XLR
SK13	3-pin DIN socket
SW1	DPDT pushbutton switch.

PCB. Knobs. Case. PCB mounting pillars. PCB through connectors. Power supply. Nuts and bolts.

in particular, vocals can be given a breathy and intimate quality.

The unit will also be found indispensible for cassette duplication, helping overcome the inevitable loss of quality.

BUYLINES

A complete kit of parts including the fully finished steel case and associated hardware is available from Time Machine Sound Engineering for £87 including VAT, postage and packing. The double sided, legended PCB is available separately at £11.50 and the case at £18. The ready built power supply in a plug costs £24.00. A constructed activator can be supplied for £130. All prices include VAT, postage and packing. Contact: Time Machine Sound Engineering, Abbotsford, Deer Park Avenue, Teignmouth, Devon TQ14 9LJ. Tel: (06267) 23539.

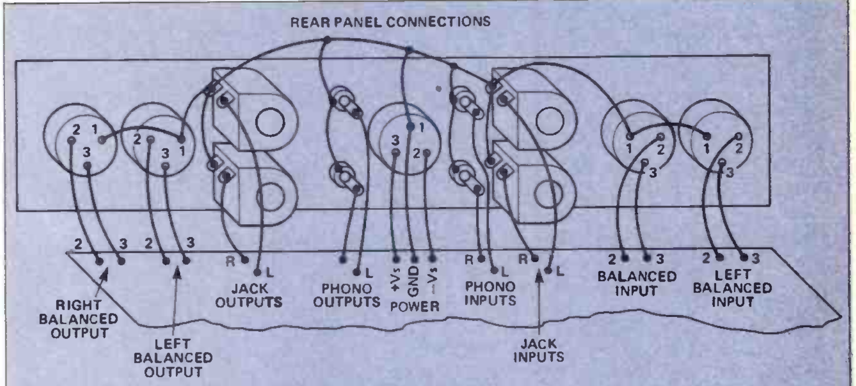
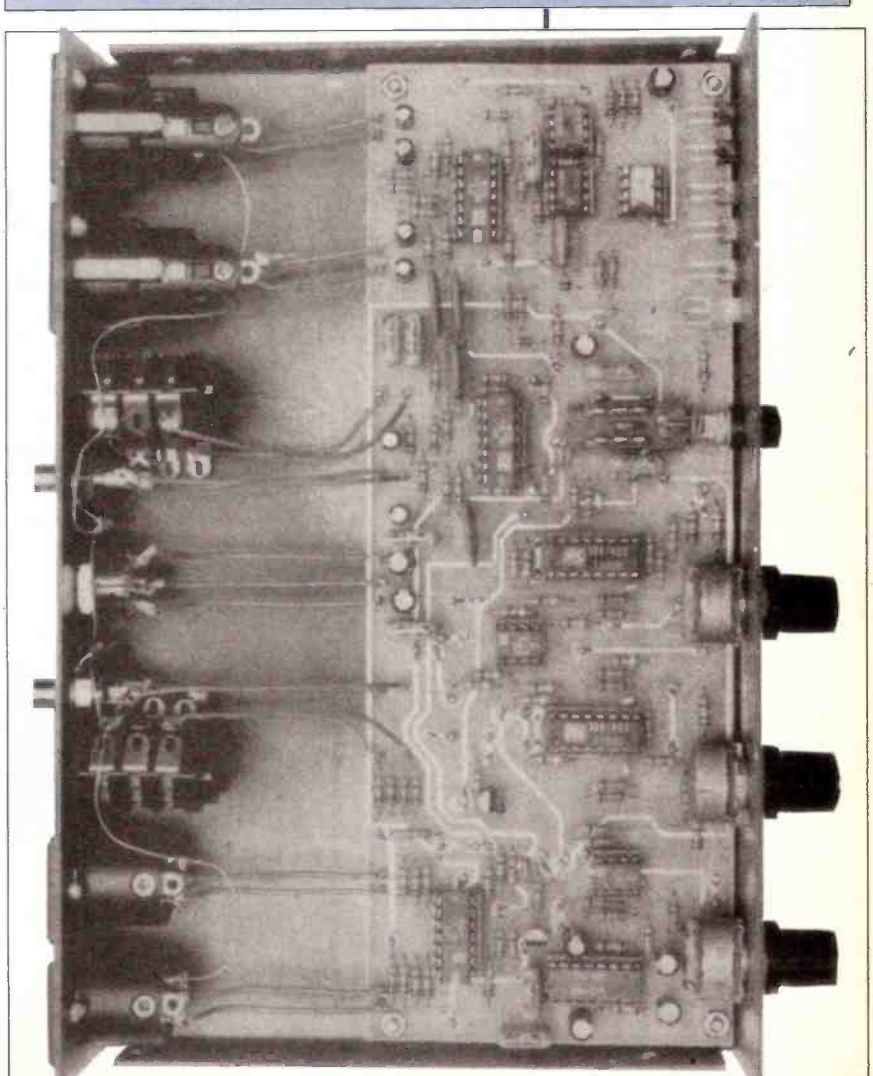


Fig. 5 Rear-panel connections

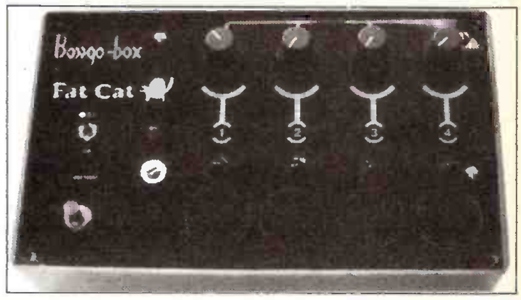


BONGO BOX

This budget beat box brings bongs to your ballads

The Bongo Box contains four resonant drum synthesiser circuits triggered by four finger pads on the front panel. The drums can also be triggered automatically from a computer by connection to its user port.

The pitch of each drum is fixed by the capacitor values used in the circuit but a feedback control on each channel allows the player to vary the amount of 'bong' in the sound.



The Bongo Box is powered from an internal PP3 battery and so is completely self-contained. Each drum output is mixed to a common volume control and taken to a standard ¼in jack socket for connection to an amplifier.

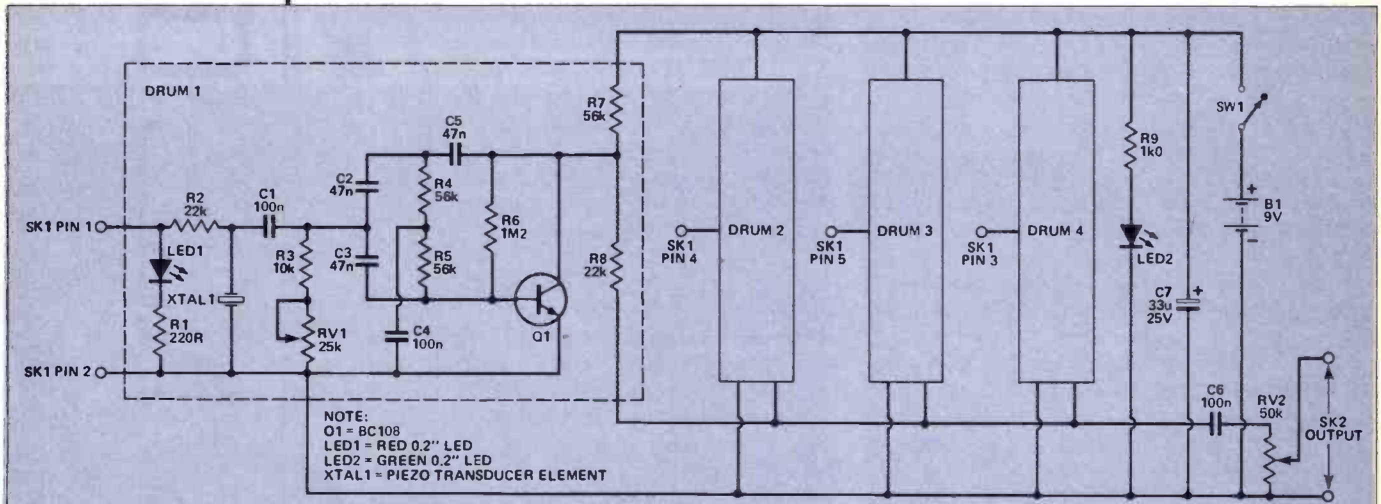


Fig. 1 Circuit diagram of the Bongo Box. Only one channel is shown in full

Construction

Construction of the Bongo Box is quite simple. Apart from the volume control, on/off switch, power-on LED, trigger plates and battery, all the drum circuits, controls and LEDs are contained on the PCB. Start by soldering in the resistors followed by the capacitors. Note the different values of C2-C5 for each drum. Solder in the four transistors but leave out the LEDs for now. Fit and solder in the pots.

Prepare a short length of 5-way ribbon cable (you will have to strip this down from a piece with six or more ways) and solder to the four outputs and ground on the PCB. Take the four piezo transducers and carefully solder a connecting wire to the centre and the outer rim of each. Scrape the surface gently first, to assist in soldering.

Drill the control panel and apply lettering as desired. Fit the five LED clips, the on/off switch and the green LED. Solder the 1k0 resistor to the anode of the LED. Drill the holes in the case and fit the 5-pin DIN socket and ¼in jack socket. Glue or bolt the PP3 clip to the bottom of the case, allowing plenty of room for the PCB.

Fit the four red LEDs into their clips and the four stand-off mounts to the panel. Gently lower the PCB

PARTS LIST

RESISTORS (all ¼W, 5%)	
R1, 201, 301, 401	220R
R2, 8, 202, 208, 302, 308, 402, 408	22k
R3, 203, 303, 403	10k
R4, 5, 7, 204, 205, 207, 304, 305, 307, 404, 405, 407	56k
R6, 206, 306, 406	1M2
R9	1k0
RV1, 201, 301, 401	25k linear potentiometer
RV2	50k logarithmic potentiometer
CAPACITORS (all 63V, min metal film except where stated)	
C1, 4, 6, 201, 301, 401	100n
C2, 3, 5, 202, 203, 204, 205	47n
C7	33μ 25V miniature radial electrolytic
C302, 303, 402, 403	10n
C304, 305	33n
C404, 405	15n
SEMICONDUCTORS	
Q1, 201, 301, 401	BC108 or similar general-purpose NPN
LED1, 201, 301, 401	red 0.2in LED and panel-mounting clip
LED2	green 0.2in LED and panel-mounting clip
XTAL1, 201, 301, 401	piezo transducer element
MISCELLANEOUS	
SK1	5-pin 180° DIN socket, chassis-mounting
SK2	¼in mono jack socket

PCB. Case (210mm×130mm×60mm). Knobs. PP3 battery and clip. PCB pillars. Foam pads for transducers. Connecting leads. Nuts and bolts.

Note that components for the second, third and fourth drum circuits are numbered 200, 300 and 400 respectively. The circuits are identical to the first circuit except for capacitor values.

HOW IT WORKS

Nothing terribly complicated here and not an IC in sight! XTAL 1 is a piezo-electric transducer. Its output along with the computer trigger pulse is buffered by C1.

The rest of each drum circuit consists of a twin-t oscillator circuit built around Q1, Q201 and so on. A free-running oscillator of this type would produce a continuous sine wave output. What is required here is a dampened oscillation which dies down after a few cycles. The main resonant frequency of the circuit is set by the values of C2, C3, C4, C5, R4 and R5. RV1 is adjusted so that the circuit is just shocked into oscillation by a positive-going pulse on the junction of C2 and C3.

This control gives a measure of 'resonance' adjustment to each drum, the pitch being largely unaffected by adjustments to it.

The dampened sine-wave oscillations from each drum are mixed together by the resistors R8, R208 and so on. RV2 is a passive volume control, the drum outputs being of sufficient level to drive an audio amplifier without further amplification. C6 is a DC blocking capacitor and C7 helps smooth the battery supply.

onto the mounts, positioning the legs of the LEDs carefully as you go. Solder the LEDs in position.

Solder the outputs and ground to the DIN socket and then solder a short piece of screened cable to the output socket and PCB. Solder the green LED's anode resistor and its cathode leg in position.

Slip the piezo plate connections through the appropriate holes and secure the plates to the front panel. This can be done with glue or Blutack. Make sure that the soldered connection points on the piezo plates are not touching the metal panel and then solder the wire connections to the PCB.

Finally, solder the two battery connections to the switch and PCB, fit the knobs and that's it!

Testing

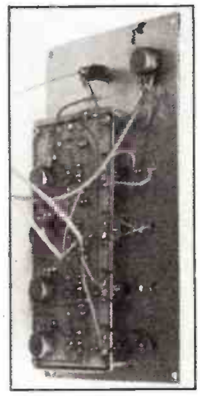
Insert a PP3 battery and switch on. The green LED should light. If it doesn't, switch off and check your soldering, battery polarity and PCB.

Turn the volume control and all four resonance controls fully anti-clockwise and connect the Bongo Box up to an audio amplifier. Turn the volume control up to halfway and tap any of the four pads. You should hear a loud click. Slowly turning the resonance control of that pad clockwise the click should turn into a satisfying 'bong' growing more sustained, until eventually a point is reached at which the circuit produces a continuous note (continuous instead of damped oscillation). Back the control off slightly until the note dies away as required. Do the same for all four drums.

Computer control

The DIN socket is available for operating the Bongo Box from a computer user port — the prototype made use of the Commodore 64. A lead is required to take the four triggers plus ground to the micro's user port connector. The connector should be tested before use by attaching the −terminal of a PP3 to the ground connection and testing each of the triggers with the +terminal. You should see each of the LEDs light up and, if the audio amp is still connected, each drum should sound.

To trigger the user port connections some sort of POKE command is usually used — check your computer and manual to see exactly what can be achieved.



BUYLINES

Just about all of the parts of the Bongo Box are available from the usual advertisers. The piezo-transducer elements are available from Maplin, along with a black foam-rubber self-adhesive pad to fit over the top of them, as in the photograph. The case and battery clip used in the prototype were from Electromail (Tel: (0536) 204555). A complete kit of parts for the Bongo Box is available from the same address and costs £29.99 inclusive. The PCB is also available separately from out PCB Service.

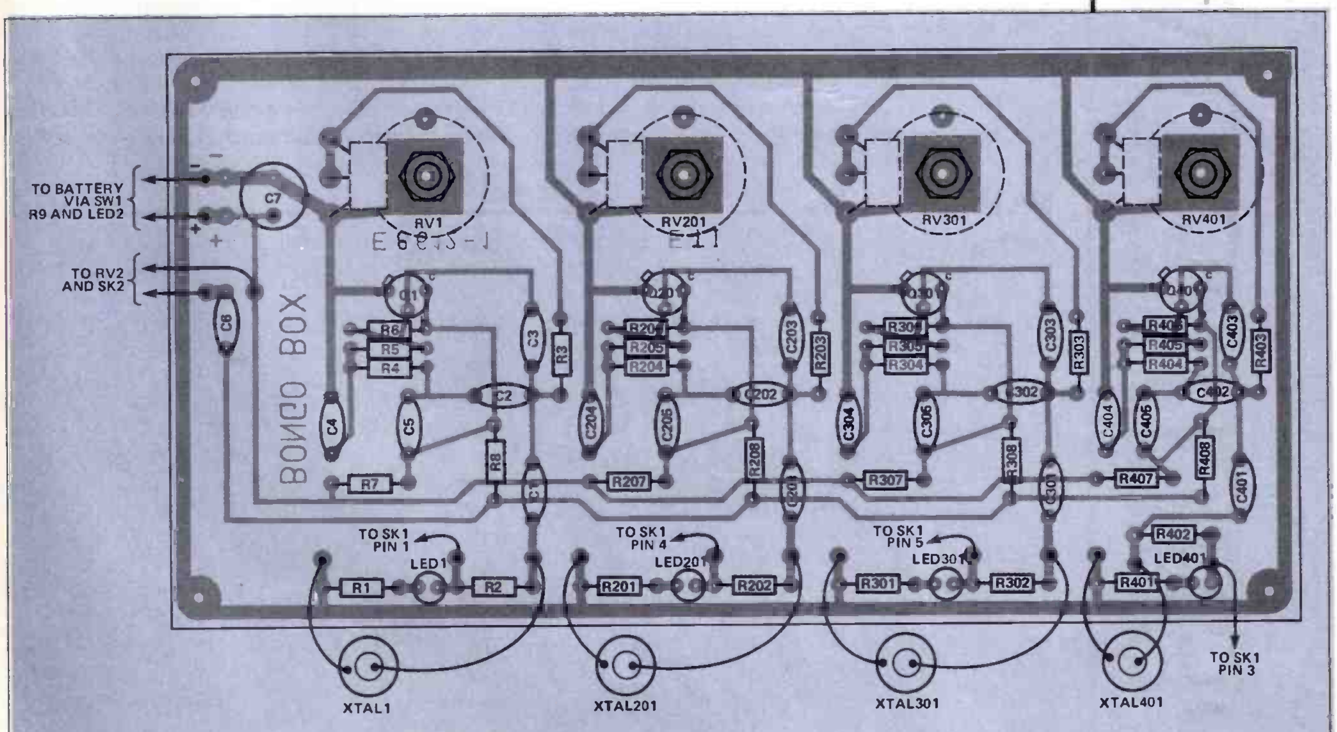


Fig. 2 Component overlay for the Bongo Box PCB

CYMBAL SYNTH

A simple but effective way to beat your own drum

In these days of digital this and sampled that, simple instruments to build are few and far between. However, this cymbal synth is both easy to make and versatile and effective in use.

It is only a two chip affair but is sufficiently adaptable to provide a useful adjunct to any drum-kit.

With a tunable noise output and a variable decay rate, the circuit could even be built into several units to form a more-or-less complete drum and cymbal kit — although in all fairness you'd be severely restricted as far as bass effects and 'touch sensitivity' go.

The unit does allow pure white noise to be fed to the envelope generator, which means it can be used for gun-shots and similar effects. A degree of touch sensitivity is incorporated.

Construction

The prototype was built for live stage work, where it has proved very useful and rugged. If toughness is not a major criterion, the circuit could be built into any suitably sized and resonant container. The home or studio recordist could make good use of it as a crash cymbal or snare-type drum.

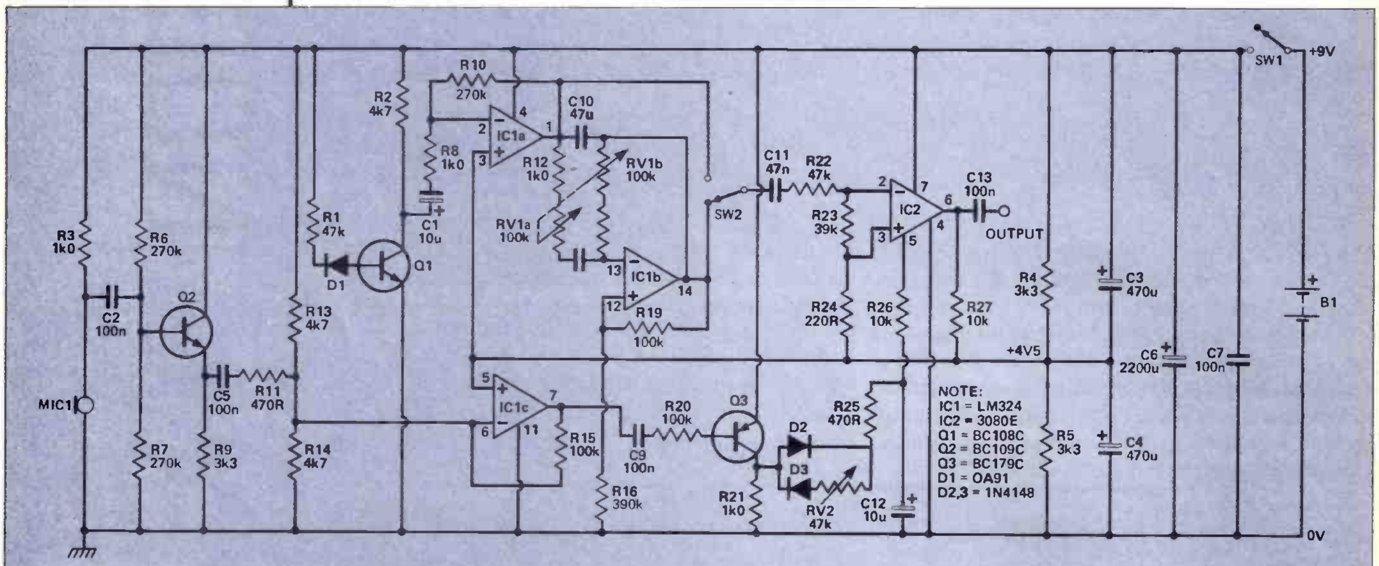


Fig. 1 Circuit diagram of the cymbal synth

The construction of the unit is straightforward. The assembly should follow the usual format of passive components first, followed by the semi-conductors and integrated circuits. The use of IC sockets is recommended to prevent damage to the chips by overheating and to ease the removal of chips should this become necessary.

HOW IT WORKS

The circuit associated with Q1 is the white noise generator, amplifying the noise produced in the diode D1 as a result of reverse leakage current. A germanium diode is used as the leakage current is higher than that of silicon for a given voltage, producing a higher noise signal level. This amplified signal is coupled by capacitor C1 to a further amplifying stage built around the op-amp, IC1a.

IC1b is a constant bandwidth bandpass filter. The filter centre frequency is set by the dual-ganged potentiometer, RV1. This is a classic second order bandpass filter of the multiple-feedback tuned type which turns the white noise into variable frequency range noise.

SW2 is to choose either the filtered noise or the unfiltered white noise. The filtered noise is (inaccurately) described as pink.

The voltage controlled amplifier (VCA) is built around the 3080E transconductance op-amp, IC2. Output current is a function of the control current fed to pin 5 of the package and the difference in voltage between the two input pins. The output current of the device is converted into a signal voltage by R27 and the signal is capacitively coupled to the output by C13.

The signal from the microphone is first amplified by Q2 and then fed to a pulse amplifier built around IC1c. This section inverts the signal and amplifies it to give a negative going pulse at its output whenever a sound is picked up. The duration for which the pulse remains negative depends to some extent on the volume of the input signal to the mic. This gives some sensitivity to the impact of a beat.

The negative going pulse is fed to Q3 which, with C12 and its associated circuit, forms a simple envelope generator.

When the pulse is received by Q3, the transistor turns on. C12 charges up rapidly through the transistor, D2 and R25, giving the fast attack which is characteristic of a drum. The transistor then turns off and C12 discharges through R25, RV2, D3 and R21. This gives a variable decay, considerably slower than the attack. The voltage is converted into a current by R26 and fed to pin 56 of IC2, the VCA.

C6 provides necessary supply capacity to eliminate any power thump which may find its way into the circuit when the drum is struck. C7 provides high frequency supply decoupling. A false signal earth is supplied in the form of a decoupled 4.5V rail. This rail is formed by C3,C4,R4 and R5, and eliminates the need for a two battery split rail supply.

There is a spare op-amp on IC1 available, should any adventurous constructor feel the need to expand the unit in some way!

The microphone was fitted to the circuit board near IC1 in the prototype with double-sided sticky fixing pads.

The prototype unit was housed inside an 8in Tambour. The Tambour has a removable drum skin which allows easy changing of the battery. They can be bought from any good music shop. A base was fitted to the Tambour with white modellers' 'Plasticard'. The whole unit was then sprayed with enamel paint and labelled with rub down letters.

PARTS LIST

RESISTORS (all ¼W, 5%)	
R1,22	47k
R2,13,14	4k7
R3,8,12,17,21	1k0
R4,5,9	3k3
R6,7,10	270k
R11,25	470R
R15,18,19,20	100k
R16	390k
R23	39k
R24	220R
R26,27	10k
RV1	100k dual gang log pot
RV2	47k lin pot
CAPACITORS	
C1,12	10µ 25V axial mounting electrolytic
C2,5,7,9,13	100n metallised polyester
C3,4	470µ 10V PCB mounting electrolytic
C6	2200µ 16V axial electrolytic
C8	10n metallised polyester
C10,11	47n metallised polyester
SEMICONDUCTORS	
D1	OA91
D2,D3	1N4148
Q1	BC108C
Q2	BC109C
Q3	BC179C
IC1	LM324
IC2	LM3080E
MISCELLANEOUS	
B1	9V battery
MIC1	Condenser Mic insert
SW1	On/off switch
SW2	SPST switch

PCB. IC sockets. ¼in jack socket. Knobs. Wire. Nuts and bolts.

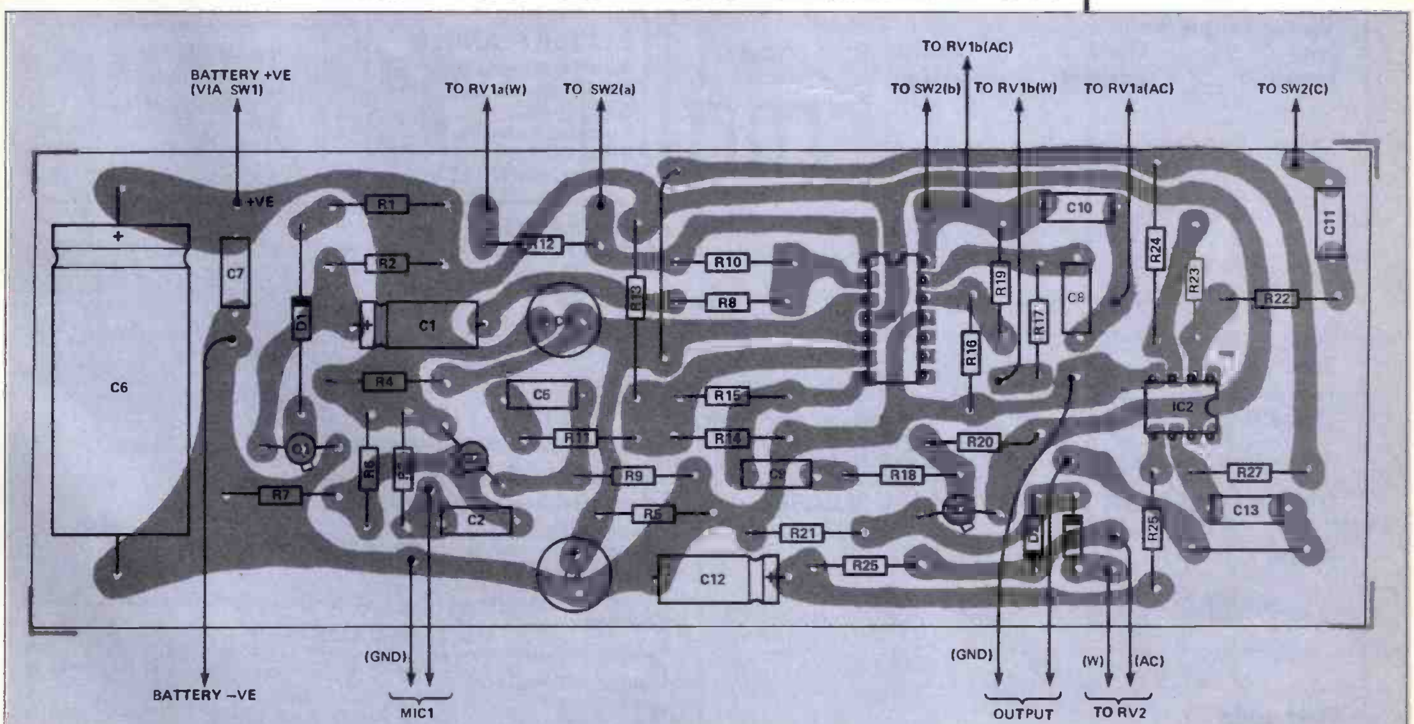


Fig. 2 Component overlay of the cymbal synth

PCBs

The printed circuits boards for the projects in this magazine are available made from the ETI PCB Service. All boards are supplied drilled and tinned ready for assembly. Please use the coupon below, or a photocopy, to order your boards.

Project	Code	Price
Chorus	E8511-5	£5.50
Flanger	E8701-3	£5.50
Noise Gate	E8507-1	£5.50
Hyper-Fuzz	E8710-4	£4.00
BBC Micro MIDI Interface	E8704-1	£8.80
MIDI Master Keyboard Keyboards (3)	E8705-3	£37.90
MIDI Master Keyboard Front Panel	E8706-3	£15.80
MIDI Master Keyboard CPU	E8706-2	£32.20
MIDI Harmoniser	E8710-9	£8.80
Paragraphic Equaliser Filter Board	E8503-6	£4.75
Paragraphic Equaliser I/P & PSU Boards	E8503-5	£8.80
DI Box	E8509-2	£3.25
Bongo Box	E8612-1	£6.62
Cymbal Synth	E8511-3	£6.62

Photocopies of the PCB foil patterns for all the projects in this magazine are available on receipt of a large (A4 sized) stamped addressed envelope to ETI Music Foil Patterns, 1 Golden Square, London W1R 3AB.

Please address your PCB order to:
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Hemel Hempstead
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Cheques should be payable to ASP Ltd.
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MUSIC BY NUMBERS

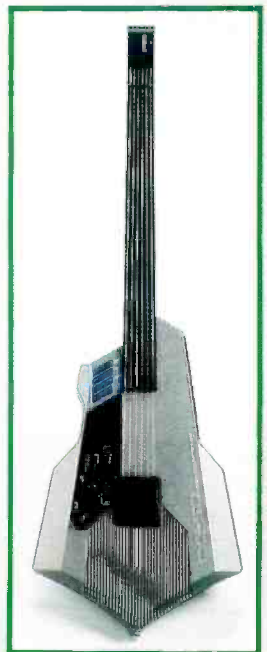
We wouldn't want to finish this magazine with anything less than a bang, so here's your chance to win one of the leading musical instruments reviewed earlier in the issue, absolutely free!

There are three prizes up for grabs. You could win any one of them — just complete the simple musical puzzle below, by matching the cryptic clues to the disordered digits.

CASIO HT-700

An 8-voice keyboard with 40 sensational preset voices plus complete programmability for creating your own patches in the 20 internal memories or on the optional RAM card.
The 20 rhythms use PCM samples and you can store your own patterns, together with a bass line and a chord accompaniment. The different parts are transmitted on separate MIDI channels. Enormous flexibility makes the HT-700 a giant among mini keyboards.

WORTH £299



KAWAI MK20

Following in Kawai's tradition of fine organs and pedigree synthesisers, the MK20 is a single keyboard with a wide range of sounds and a host of facilities to delight novices and accomplished musicians alike. The keyboard is fully touch sensitive across its five octave range and can be split to play two orchestra sounds plus a solo voice, each transmitting on separate MIDI channels. There are 18 orchestra sounds and 18 solo voices plus 32 rhythms using PCM samples of excellent quality with write, store, sequence and RAM card facilities. The MK20 offers powerful performance ability in a compact personal keyboard.

WORTH £777

CASIO DG-20

Casio's innovative guitar synthesiser. All the fun of the MIDI fair for guitarists. 20 sounds and a 2W amp and speaker built-in along with 12 rhythm patterns from PCM encoded toms, cymbal and snare. Line out and extension speaker sockets are provided along with the ubiquitous MIDI OUT connector which allows the DG-20 to control any MIDI equipped sound source in either MIDI mode 3 or 4. Much more than a guitar and much more than a keyboard.

WORTH £249

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A Seasoned Vivaldi
B Also Sprach Zarathustra
C Tulsa ETA
D Bowie's years
E Tom's motorway
F Classic arsenal overture
G Rolf's boys, toys and horses
H Glenn's Penn number
I Sheena's office hours
J Queen's vision

HOW TO ENTER

Listed here are 12 numbers. Ten of these are mentioned in the titles of songs and pieces of music cryptically hinted at in the ten clues. All you have to do is match the right numbers to the title clues.
Take a shiny new postcard (or use the back of a sealed envelope if you can't afford a postcard) and write on it the letters A to J. Now write next to each letter the number from the list which you think appears in the title of the song described in the clue. Clearly write your name and address on the card, state which prize you want and send the card to:

ETI Making Music Competition,
1 Golden Square, London W1R 3AB.

All entries must be in by 21st October 1988. The winners will be notified by post and all the results published in a future issue of ETI. The competition is open to all readers of The ETI Guide To Making Music except employees of Argus Specialist Publications, Casio, Kawai, and their families. One entry per person only. The judges' decision is final and no correspondence concerning this competition will be entered into.