A PRACTICAL ELECTRONICS Publication

SOUND DESIGN

...a selection of musical projects for home constructors

MINISONIC 2

...and special sound effects

SOUND DESIGN

SECOND EDITION

Contents...

A PRACTICAL ELECTRONICS Publication...

With electronics playing such an important role in music today, P.E. have pleasure in presenting the pick of just some of its most popular musical projects in this 96-page book.

With one exception these articles are as originally published in P.E. save for the incorporation of certain designer approved amendments or corrections.

The Minisonic 2 is a new, as yet unpublished, development of the original Minisonic design.

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MINISONIC 2

By G.D. SHAW

A full specification keyboard synthesiser with the facility of instant push button patching

- Temperature compensated oscillators with vernier tuning
- Two envelope shapers for extra effects
- Stereo output with panning controls
- External inputs, such as voice or instrument, can be processed
- V.C.F. can provide a vast range of sounds
- Integral three octave keyboard
- 23 push button patching system

Although the original Minisonic was designed with an eye on the younger generations of amateur constructors, it soon became fairly obvious that the instrument was capable of a far wider range of application than was ever initially envisaged. Thus, with the co-operation of a group of musicians, it was decided to submit the original Minisonic to a process of intensive sophistication and re-development, where necessary, in order to see whether it could be turned into a professional quality instrument. The result is an electronic music synthesiser which it is believed will stand direct comparison with commercially available instruments costing upwards of £1,000. Such a result could not have been achieved without the co-operation of Synthesiser Music Services Ltd., to whom the author offers his grateful thanks.

INSTRUMENT DESCRIPTION

The Minisonic II is a three octave keyboard synthesiser featuring push-button patching of audio signal and control voltage options. Two temperature compensated log law v.c.o.s of greatly improved design offer, in addition, facilities for cross modulation and phase locking up to the ninth harmonic. These latter features considerably extend the range of sounds of which the instrument is capable and it is particularly easy to create close imitations of a number of reed or woodwind instruments.

Two envelope shapers and voltage controlled amplifiers are incorporated with the envelope from channel 1, being made available in its normal and inverted form as a modulation source. The envelope state from each device is monitored by a light emitting diode which varies in brightness with the envelope level.

The keyboard controller is similar in form to that which appeared in the original Minisonic but offers a number of improvements. Like the original it is possible to adjust the register and span of the keyboard but, in the case of the Minisonic II, provision has been made for balancing the gain of the operational amplifiers in the controller so that interaction between tune and span controls can be eliminated. Additionally, an equal temperament span is available at the touch of a button. The hold circuit, or pitch memory, of the controller has been provided with a variable "Portamento" control so that glide effects of up to one second duration may be achieved.

In addition an isolator has been incorporated to reduce drain on the hold capacitor thereby ensuring that an unvarying pitch may be maintained for periods up to 40 seconds.

A voltage controlled low-pass filter having a passband variable in the range 3Hz–15kHz is incorporated.

RING MODULATOR

Another sound treatment device is the ring modulator which may be used as a frequency doubler or multiplier giving sum and difference frequencies each with a gain of about 25dB. Provision is made to route external signals into the ring modulator.

The noise generator is essentially the same circuit which appeared in the original Minisonic except that it now features an up-to-date version of the Z1J, appearing in axial form as the Z5J noise diode.

The output stages comprise, in each stage, a two input summing mixer and a 0·25 watt power amplifier which is arranged for headphone driving. A headphone jack is provided for this purpose although the output from this jack socket may also be directly linked to a pair of 8 ohm external speakers if required. The output from each mixer stage is routed to separate jack sockets on the rear panel of the instrument. Drive to the output stages is taken from the voltage controlled amplifiers via panning controls so that there is provision for a subjective spatial positioning of the signal elements in a stereo image.

KEYBOARD AND SWITCHES

The keyboard and contact assemblies are of the type supplied by Kimber Allen Ltd. The keyboard frame, which is an exceptionally rigid aluminium extrusion, forms a bridge between two end cheeks manufactured from solid afrormosia/teak. Additional stiffening and rigidity are provided by the front and base panels which are of 14 s.w.g. aluminium and screwed to aluminium angles in turn screwed into the end cheeks. The flat rear panel carries the circuit boards of the synthesiser and is hinged to the top edge of the front panel. Thus the whole of the electronics of the instrument are easily available for servicing and/or setting up.

Constructors who have built the original Minisonic may be interested to learn that although a number of design changes have taken place it should be possible to convert existing instruments to the improved form with a minimum of difficulty.

COMPONENT NUMBERING

For the Minisonic II component numbering will follow a logical sequence. Where there are two circuits of one type the components will be numbered, for example, R23/1 or R23/2 and so on. For the benefit of those constructors wishing to convert an original Minisonic, the alphabetical coding for circuit board pins has been retained but, since there are a number of extra pins required, these will be given the same coding as the nearest adjacent pin followed by the suffix /X, i.e. P/X, JJ/X and so on.

Apart from the push button patching system which embodies a series of additional components, a number of additional active and passive electronic components are required. These components, together with other component value changes, will be itemised at the beginning of each section again for the benefit of original Minisonic constructors.

It is strongly recommended that construction should follow a pattern in which individual circuits are built and tested as separate units. Experienced constructors should not find much difficulty if they press on ahead and save all the setting-up to one marathon end-of-project session. However, the not so experienced constructor would be well advised to heed this recommendation and thereby save a great many possible problems.

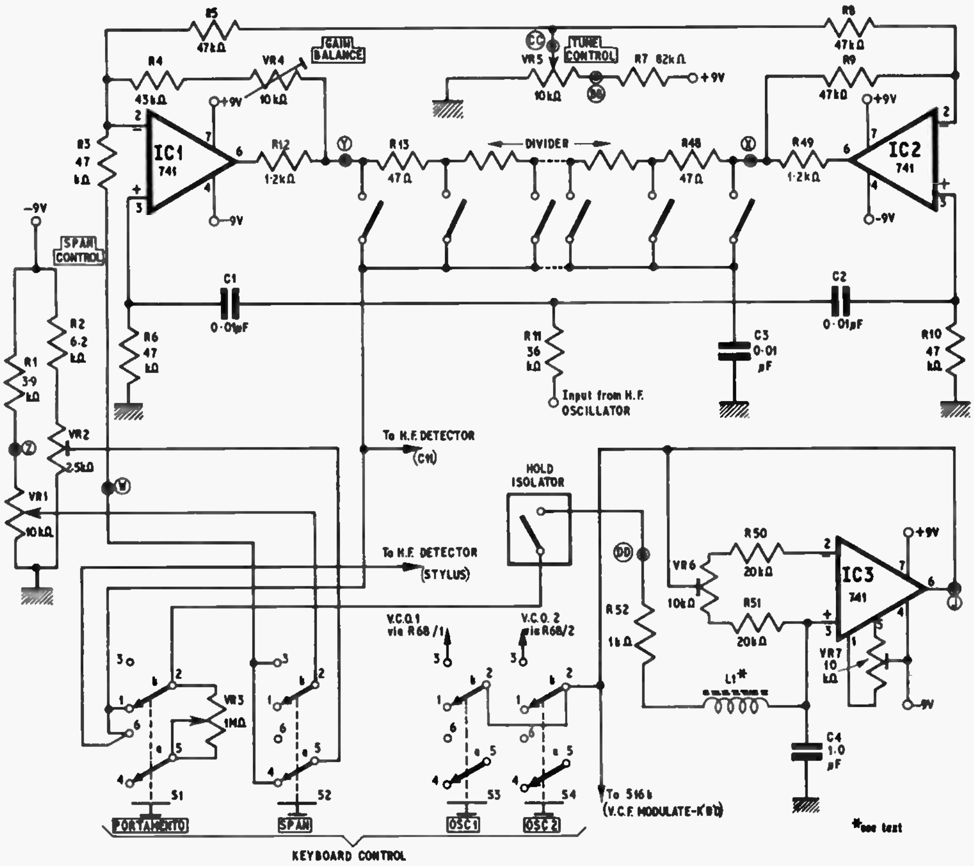


Fig. 1. The keyboard controller and hold circuit

Testing of individual circuits is, of course, rather time consuming since it frequently requires components external to the circuit board to be connected temporarily. It is well worth doing though and a great morale booster when, as construction proceeds, it is known that all one's previous work is functioning correctly. It is quite in order to employ batteries to power the circuit boards for testing. PP9s are an ideal size and should see through all the testing procedures quite adequately.

It is a good idea to obtain some fairly substantial reservoir capacitors of, say, 1,000μF, for connecting across the batteries particularly as the circuit board begins to fill up. This will help to prevail against the increasing resistance or "stiffness" of the battery as voltage tends to fall with the increased drain.

TEST EQUIPMENT

Most of the routine testing and setting up may be accomplished with a good quality high resistance voltmeter. Whilst such meters are perfectly suitable for measuring fairly high level voltages and, in many instances currents down to a few microamps, it should be borne in mind that where low level voltages are being measured in the presence of a substantial resistance value, there is a danger that the meter can load the circuit thereby introducing an error into the reading.

COMPONENTS . . .

KEYBOARD CONTROLLER AND HOLD

Resistors

R1	3·9kΩ	R8–10	47kΩ (3 off)
R2	6·2kΩ	R11	36kΩ
R3	47kΩ	R12	1·2kΩ
R4	43kΩ	R13–48	47Ω (36 off)
R5–6	47kΩ (2 off)	R49	1·2kΩ
R7	82kΩ	R50–51	20kΩ
All ±5% ¼W carbon		R52	1kΩ

Capacitors

C1–3 0·01μF ceramic (3 off)
C4 1μF metal polyester

Integrated Circuit

IC1–3 741 (3 off) or 741 (2 off) FETMOPA (1 off)

Inductor

L1 100 turn of 34 s.w.g. on ring core CRO71–12A/A (ITT)

Potentiometers

VR1	10kΩ linear
VR2	2·5kΩ preset
VR3	1MΩ linear
VR4	10kΩ preset
VR5	10kΩ linear, 10 turn wire wound
VR6	10kΩ preset 20 turn
VR7	10kΩ preset 20 turn

It is recommended that where low levels are to be measured a d.c. coupled oscilloscope or active voltmeter having an input resistance of at least 1 megohm should be used. The oscilloscope is, of course, an excellent test instrument particularly if it is d.c. coupled and those constructors having access to one will find that testing and setting up is greatly simplified.

What follows now is a detailed description of circuit function.

KEYBOARD CONTROLLER

The keyboard controller as illustrated in Fig. 1 is a means of providing a range of control voltages which, when applied to the input of a voltage controlled oscillator, cause it to oscillate over a range of pitches normally associated with the equal temperament chromatic scale or, alternatively, over a range of pitches quite outside what might be termed normal musical acceptance.

Components additional to those in the basic Minisonic are as follows: R2, VR2, VR3, R4, VR4, R6, R10, R11, R12, R49, R52, C1, C2, C3. The hold isolator is an additional active circuit entity and will be dealt with in a following section.

IC1 and IC2 are inverting operational amplifiers whose outputs are linked by a chain of resistors, the junctions between which are connected to the keyboard contacts. R7 and VR5 form a voltage divider between the positive rail and ground such that the swing of the potentiometer covers a range of about 4·7 volts. The wiper of VR5 is linked to both i.c.s so that the outputs of these devices will track, in unison, the setting of VR5. R1 and VR1 form a second voltage divider connected between the negative rail and ground with the wiper linked to IC1 only. Thus VR1 is able to provide an offset to IC1 which is variable over 4·5 volts.

TUNE AND SPAN

The purpose of VR5, the "Tune" control, is to set the upper frequency limit of the three octave keyboard. With a control voltage law of 500mV/octave VR1, the "Span" control, is adjusted to provide an offset of 1·5V to IC1. Since the divider resistors R13–R48 are of equal value, equal voltage increments will appear across each resistor and the "logarithmic" oscillator will be programmed to produce an equal temperament scale. If it is assumed that the setting of the "Span" control is now fixed then VR5 may be adjusted over a wide range without varying the equal temperament capability of the oscillators.

This only holds good, of course, if the gains of IC1 and IC2 are identical. If there is any variation in gain then any change in the setting of VR5 will change the span of the keyboard and cause the oscillators to go out of tune. The arrangement of R4 and VR4 is introduced, therefore, to enable a close matching between the gains of IC1 and IC2. Ideally, VR4 should be a 15 or 20 turn cermet preset.

The purpose of R2 and VR2 is to provide a preset or fixed span facility which may be alternated with the variable span by means of S2a/b. This facility offers the advantage for "live" performance that the variable span position may be adjusted to provide a microtone or macrotone span between consecutive notes and then switched in as required to provide a particular effect.

DIVIDER RESISTORS

The values of the divider resistors R13–R48 are not critical although they should ideally be around 47Ω to 56Ω. It is important, however, that whatever value of divider resistor is selected, the total value of resistance per octave is, as far as possible, identical. Unless very high precision resistors are chosen for the divider (a needless expense) then it pays to check each resistor individually, since there will always be slight variation in the ohmic value within the tolerance range of the devices.

If it is found, for example, that there are three resistors which are, say, 1 ohm below the rated value, then these should be allocated the same relative position in each of the three octaves. This means that, providing the same procedure is followed for all other resistor positions, perfect octaves will sound when the span control has been correctly set. Although there may be very slight frequency errors between consecutive notes, these tend, in practice, to be very difficult to detect.

The Minisonic II has a unique method of envelope shaper triggering which enables single contact operation from the keyboard. Detailed operation of the system will be described later but, in brief, it necessitates the superimposition of a very high frequency signal of low amplitude onto the divider resistor chain.

R6, R10, R11, C1 and C2 are additional components which serve to route the high frequency signal into the keyboard controller without it being affected by operation of the "Tune" and "Span" controls.

THE HOLD OR ANALOGUE MEMORY

The sustain function in any musical instrument is of considerable importance in that it imparts a flowing characteristic to the music. In the keyboard controller the closure of a contact instantly programs the oscillator to its specified pitch or frequency and just as instantly removes the programming voltage when the contact is broken. Without some means of sustaining the programming voltage, the resultant "music" would be a series of tone bursts with any sustained tones remaining at a constant loudness only so long as the key contacts are closed.

The hold circuit provides the means whereby the last programmed keyboard voltage is retained, thereby maintaining the oscillators at a constant pitch, either until a new voltage is programmed in or until such time as the envelope shaper has completed its decay cycle.

The hold circuit comprises the components around IC3, an operational amplifier in which the output signal is divided by means of R50, VR6 and R51 in order to provide balanced levels of positive and negative feedback. When the balancing is carefully done the circuit is theoretically capable of providing an infinite impedance to incoming signals. In practice, however, it is more usual to calculate the input impedance on the basis of the parallel value of the feedback resistors times the open loop gain of the amplifier. Thus, in the circuit under consideration, the input impedance is of the order of 2,500 megohms.

The hold capacitor C4 is ideally a low leakage type. A charge applied to C4 is reflected at the output of IC3 with any drift at the output due to a combination of capacitor leakage and minor thermal effects within the i.c.

BALANCING THE CIRCUIT

It is possible to balance the circuit such that the output drift is better than 1mV/second but to do so requires considerable patience and care. There are a least two methods of balancing the circuit. In the original Minisonic the wiper of VR6 was temporarily disconnected from the output of IC3 and linked to ground with VR6 having first been adjusted so that its wiper was in the electrical centre. A 10 megohm resistor was then temporarily linked between the output and pin 2 of IC3 thereby turning IC3 into a high gain amplifier.

With power applied VR7 was then adjusted so that the output of IC3 was maintained at precisely zero volts. Having thus set the offset of IC3 the circuit was then returned to the form shown in Fig. 1 and the final balancing carried out on completion of the instrument using subjective aural assessment rather than test instruments.

Although the latter method of balancing proved to be very successful it was also rather cumbersome and involved too many manipulations around the circuit. Another method is to carry out the balancing with the circuit as it stands. In this case both VR6 and VR7 require to be set to their electrical centre and a voltmeter, set to the 25V range, coupled between the output of IC3 and ground. With power applied the output of IC3 will tend to drift fairly rapidly either positive or negative depending on the polarity of the offset.

A short piece of wire temporarily connected to the input pin "DD" is then touched on the 0V rail thereby restoring zero output to IC3 and VR7 adjusted, touching the 0V rail as necessary and readjusting, until the output drift has been substantially reduced in rate. At this point the voltmeter can be switched down to a lower range and the exercise repeated until such time as the minimum rate of drift has been achieved by adjustment of VR7. Final adjustment and balancing is then carried out as previously by means of VR6 and aural assessment.

HOLD ISOLATION

Isolation of the front end of the hold circuit is achieved by two means. Firstly, a degree of a.c. isolation is maintained by means of R52 and the high frequency choke, L1. Originally R52 was specified as 20 kilohm with the alternative of replacing this with a ferrite ring carrying approximately 30 turns of 34 s.w.g. enamelled copper wire. On its own the 20 kilohm resistor proved to induce an unacceptable degree of glide when oscillator frequencies were changed by means of the keyboard and, in the prototype Mark II, 30 turns of enamelled copper wire proved to be insufficient to entirely isolate the r.m.s. component of the a.c. signal applied to the divider.

This latter effect is characterised by a slight fall in oscillator pitch each time a key is released. Although this is generally discernible only when any degree of decay is employed it is nevertheless an undesirable characteristic.

Improved a.c. isolation can be achieved by utilising a ring-core type CRO71-12A/A from ITT which carries 100 turns of 34 s.w.g. wire. Additionally a series resistor of the lowest possible value may be added to entirely eliminate the pitch change effect. 1 kilohm has proved, in practice, to be an effective value but introduces a 1 millisecond time constant into the charging rate of the hold capacitor. It is therefore recommended that the value of R52 should not exceed 3 kilohm otherwise the beginnings of a glide effect will become apparent.

REED RELAY

A second means of isolation to the hold circuit is achieved by means of a reed relay introduced between the input of the hold and the "Portamento" controls. The necessity for this feature in the Mark II is based on the fact that the greater length of hold input lead required due to the expanded layout of the instrument introduced a loading effect on the hold capacitor which greatly accelerated the rate of drift. This effect occurred whether or not the input lead was included in, or separate from, the wiring harness of the instrument.

The hold isolator is triggered from the h.f. detector and, since this latter circuit is dependent for its operation on connection to the envelope shapers, details of circuit function will be described in detail in the appropriate section.

PORTAMENTO

In the Mark II a "Portamento" control has been included in the scheme in order to provide a variable degree of glide effect on pitch change programmed from the keyboard. Thus the input to the hold circuit may be routed direct or via a 1 megohm potentiometer with the switching accomplished by means of S1a/b. With the potentiometer set to maximum a time constant of 1 second is introduced into the charging of the hold capacitor and thus glide effects of greater than one second duration can be achieved.

ALTERNATIVE HOLD CIRCUIT

An alternative form of hold circuit which is being adopted for professional users of the Minisonic is to replace the 741 operational amplifier with a pin compatible device having f.e.t. inputs. Although rather more expensive to construct the circuit, shown theoretically in Fig. 2, offers the great advantage that drift is reduced to around 1mV/minute and that, apart from setting the output offset of the device to zero, there is no prolonged balancing procedure required.

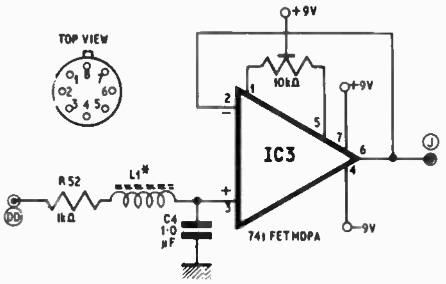


Fig. 2. Alternative form of hold circuit

The alternative circuit may be built on the main circuit board with a minimum of modifications in the way of wire links to bridge out the provision for the feedback balance preset. Constructors should note that both the hold circuits establish a d.c. output potential when the instrument is switched on for the first time and that these offsets can be of either positive or negative potential. When switching on therefore, allow a few moments for the circuits to stabilise then depress one of the lower keys on the keyboard to set the hold circuit output to a voltage within the range of, and compatible with, the v.c.o.s.

V.C.O.s—CONTROL NODES

The v.c.o.s in the Minisonic Mark II have been the subject of a number of radical design changes and we will deal firstly with the control node. In the original Minisonic the option of temperature stabilisation was offered utilising a so-called transistor oven arrangement which had proved to be very successful in the P.E. Sound Synthesiser.

The oven, which provided stabilisation for both v.c.o.s and for the v.c.f., was constructed around a transistor array, the ML3046P by Microsystems International. In May of 1975 however Microsystems ceased production and it was necessary to utilise the RCA equivalent array, the CA3046. Unfortunately the characteristics of the CA3046 proved to be slightly different from those of the ML3046P the most serious

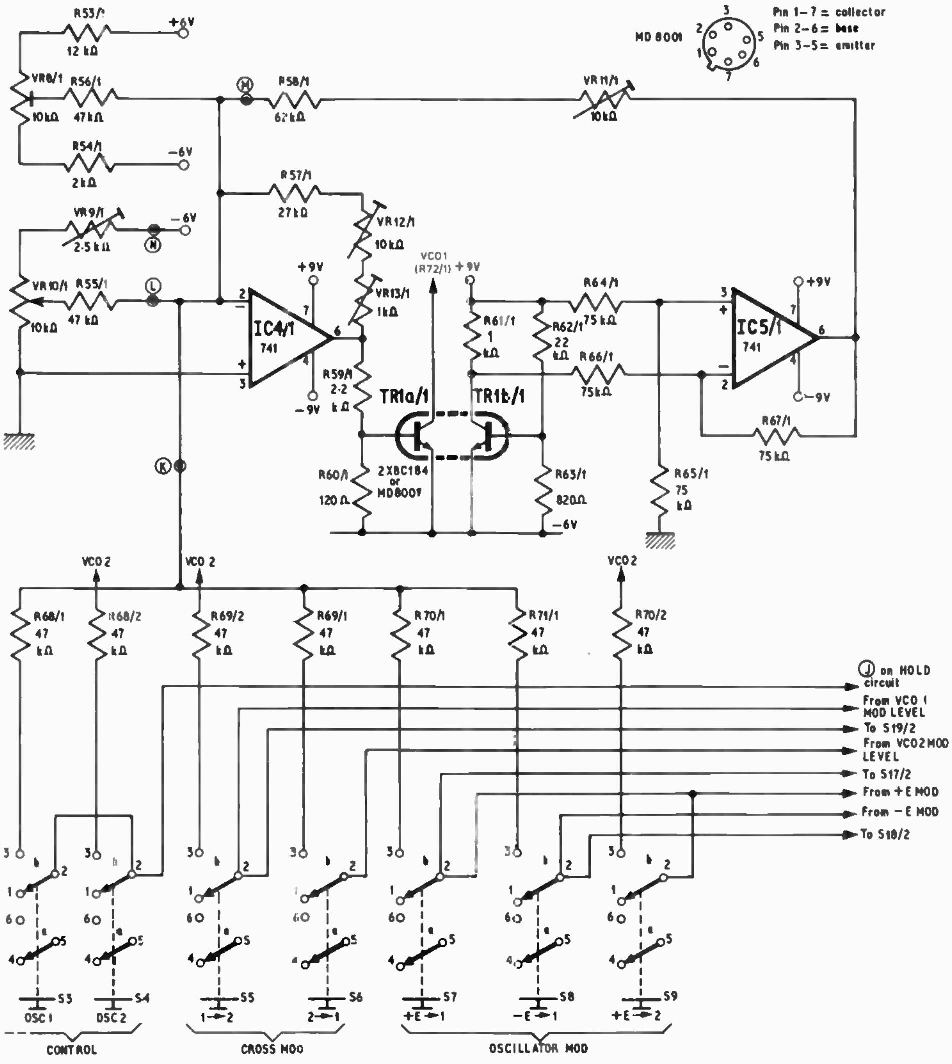


Fig. 3. Temperature compensated v.c.o. control node

COMPONENTS . .

VCO— CONTROL NODE (2 REQUIRED)

Resistors

R53	12kΩ
R54	2kΩ
R55	47kΩ
R56	47kΩ
R57	27kΩ
R58	62kΩ
R59	2.2kΩ
R60	120Ω
R61	1kΩ
R62	22kΩ
R63	820Ω
R64	75kΩ
R65	75kΩ
R66	75kΩ
R67	75kΩ
R68	47kΩ
R69	47kΩ
R70	47kΩ
R71	47kΩ

All ±5% ¼W carbon

Potentiometers

VR8	10kΩ preset
VR9	2·5kΩ preset
VR10	10kΩ lin. 10 turn W/W
VR11	10kΩ preset 20 turn
VR12	10kΩ preset 20 turn
VR13	1kΩ preset 20 turn

Semiconductors

IC4	741
IC5	741
TR1a/b	MD8001

difference being that the transistors would provide a true exponential function over rather less than one decade, approximately three octaves. In practice the degree of error introduced was about 0·25 per cent at the extreme of a three octave span and it was decided that this was rather too much to tolerate for really serious musical purposes.

The alternative form of temperature stabilisation adopted is shown in Fig. 3 and, although very simple in concept, it has been found, in practice, to work extremely well. Basically the idea is founded on the assumption that two transistors having identical gains, fixed bias characteristics and mounted in the same environment will behave in a predictably similar way in response to changes in temperature. It is known, for example, that the V_{be} of a transistor will tend to fall with an increase in temperature. If the bias is fixed however then an increase in temperature will cause an increase in current through the device.

If our two transistors are arranged such that one is employed as the controlled current generator for the v.c.o. and the other as a controller and the current in the controller is monitored with the resultant signal led back to the control node of the c.c.g. then, since both transistors behave in an identical manner, the feedback from the controller can be made to vary the bias on the c.c.g. in inverse proportion to changes in temperature. This is, essentially, the function of the circuit shown in Fig. 3.

CIRCUIT ACTION

IC4 is the control node which provides drive to the c.c.g., TR1a. TR1b is the controller which is biased by R62/1 and R63/1 such that it is passing about 0·5mA–0·75mA. The differential voltage across R61/1 is monitored by IC5 which has unity gain, thus the voltage output of IC5 will be in the range +0·5 to +0·75V. This value is not too critical since variations here can be compensated for by adjustment of VR8 in the control node.

The output signal of IC5 is led back to the control node IC4 via R58/1 and VR11/1 the total resistance value being adjusted such that the attenuation factor in relation to the feedback resistor value in IC4 is approximately 0·5. In other words the overall response of the system between the differential voltage across R61/1 and the output of IC4 is less than unity. The transistors TR1a and TR1b were, in the prototype Mark II, a pair of selected BC184s which were bonded face to face by means of a thin smear of Araldite.

To provide a better thermal contact and, at the same time, a degree of thermal inertia, three turns of tinned copper wire were bound tightly around the plastic bodies of the transistors and secured by twisting the ends tightly together. The use of a dual transistor such as the MD8001 undoubtedly improves the control characteristic but at a slight additional expense.

TEN TURN POTS

In the control node itself VR8 sets the bias on IC4 and fixes the lower frequency limit of the v.c.o. VR10/1 is the manual frequency control and it is recommended that this be a ten-turn precision pot. VR9/1 adjusts the potential across VR10/1 such that it is precisely 5 volts. Since we will be setting up the control node such that the response or law is 500mV/octave this means that each turn of the potentiometer gives an octave change in frequency of the oscillator. It is recommended that each control node be constructed and set up individually before adding the oscillator components to the p.c.b.

During construction the following points should be observed. R58/1 should not be connected until the control node operating points have been established. VR12/1 and VR13/3 should be set to the mid point of their electrical travel before mounting.

It is recommended that these devices be 15–20 turn cermet presets. Although more expensive than the standard open types the greater precision in setting up and the ease with which this operation may be accomplished adequately compensates for the increase in cost.

When connecting VR10/1 temporarily to set up the control node it should be linked directly with its "high" end to the −9V rail and adjusted so that its wiper is initially at the earthy end of its travel.

SETTING UP

When assembly of a control node has been completed, except for the connection of R58/1, power may be applied and VR8 adjusted so that its wiper is around −0·5V potential.

Connect a microammeter between the collector of TR1a and the 0V rail with its positive terminal to the 0V rail. No reading should be discernible at this stage. Advance VR10 until the microammeter begins to indicate a flow of current through TR1a and adjust the pot until such time as the current reading is at a rational level, say 1 or 2 microamps. At this stage monitor the voltage on the wiper of the pot and make a note of the voltage/current relationship. Advance the pot so that its wiper voltage increases by 0·5 volts and again note the voltage/current relationship.

Repeat this procedure a further four times or until there is a table of at least six relationships. The aim is to achieve a doubling of the current through TR1a for each successive 0·5V increase in potential at the wiper of VR10 and the purpose of making a number of measurements initially is to accurately establish the error relationship in the control node.

ASSESSMENT

If, for example, there is less than a doubling of current between each successive voltage reading then the gain of the control node is too low. Conversely if the current more than doubles for each 0·5V increment in voltage then the gain is too great and, in either case, adjustment has to be made to the feedback resistance on IC4. If, for example, the gain is too great and for a 5·0V input the current is 16μA with an increase to 40μA at a voltage of 5·5 then the voltage should be set at 5·5 and VR12 adjusted so that the error is halved i.e. the current would now be 36μA.

Taking this as the datum level VR10 is now adjusted back in a series of steps of 0·5V until a new set of readings has been established. VR12 should be adjusted again so as to halve the error and the procedure repeated until the error in current doubling is very small.

Final adjustments may then be made using VR13. When the required current doubling relationship has been achieved VR10 should be set so that its wiper is at −6 volts and VR8 adjusted so that the current through TR1a is 20μA.

Note that it may not be possible to quite get to 20μA at this stage since the positive voltage swing of VR8 is limited to compensate for the bias applied to the control node from IC5. If this is the case set the current through TR1a as close to 20μA as may be possible. A slightly moistened finger now placed on top of TR1a/b should result in an increase in the current reading.

FEEDBACK CONNECTION

R58 may now be connected having first adjusted VR11 so that it offers zero resistance. The effect of connecting R58 will be to apply a negative bias to TR1a thereby reducing the current. VR8 should therefore be readjusted until the current is 20μA as before. Again place a slightly moistened finger on TR1a/b and observe the effect on the current through TR1a. In this case since the gain factor is greater than unity the current through TR1a should decrease thereby implying too great a correction.

Allow the transistors to stabilise at room temperature and adjust VR11 to increase the resistance in the feedback loop and try the finger test again. This procedure should be repeated until there is no change in the current through TR1a or, at worst, only a very small change.

Note that the finger test is a very severe one and that, furthermore, if the finger is not applied squarely over both transistors in the pair then it is possible to impart a bias to either one or the other which, due to the thermal lag between the active junctions, may introduce an error into the procedure. The exercise of care and some patience is therefore required. Where a dual transistor is employed there is, of course, no problem of thermal lag and the procedure is rather more straightforward.

If, when R58 is connected for the first time and the finger test carried out, the current through TR1a increases check first of all that the finger is being applied squarely to both transistors. If this is so monitor the output of IC5 during the period that the finger is applied to the transistors. As stated earlier the output of IC5 should normally be in the range 0·5V–0·75V positive and should become more positive as TR1b is warmed up. If this is also the case then the feedback from IC5 is insufficient and R58 should be reduced to the next lower preferred value, say 56 kilohms.

When temperature balance has been achieved check that the wiper of VR10 remains at −6·0 volts and finally adjust VR8 so that the current through TR1a is 40μA. No further adjustments need be made to the control nodes until the instrument is fully completed at which time the oscillators can be tracked aurally at mid range by means of small adjustments to VR13 and set to similar minimum operating frequencies by means of VR8.

COMPONENTS . . .

VCO—INTEGRATOR/COMPARATOR
(2 REQUIRED)

Resistors

R72	1kΩ
R73	10kΩ
R74	20kΩ
R75	15kΩ
R76	470kΩ
R77	2kΩ
R78	47kΩ
R79	6·8k

All ±5% ¼W carbon

Capacitors

C5	0·047μF polyester
C6–7	22μF tantalum 25V (2 off)
C8	0·1μF ceramic (VCO1 only)

Potentiometers

VR14–15	10kΩ linear
VR16	10kΩ linear (VCO1 only)

Semiconductors

IC6	741
IC7	LM318N
TR2	2N5459
D1	1N914 (VCO1 only)

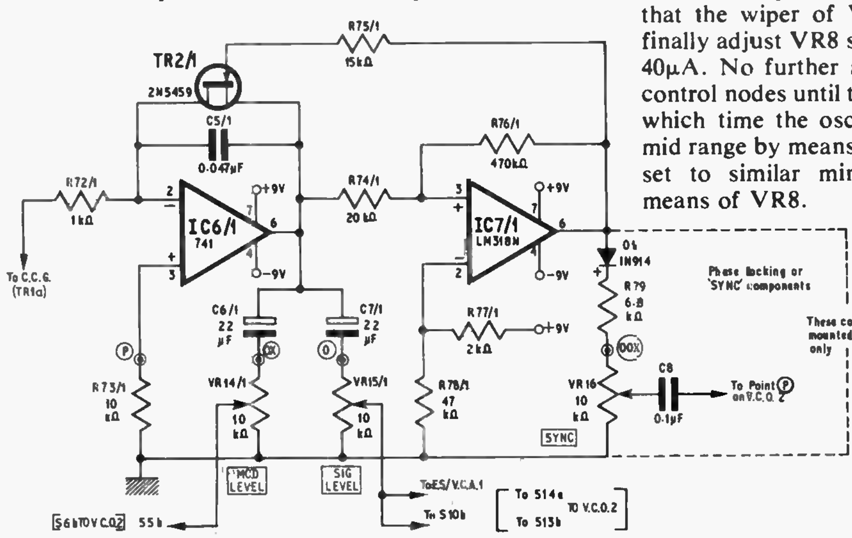


Fig. 4. Circuit of v.c.o.

OSCILLATOR SECTION

The v.c.o. illustrated in Fig. 4 comprises an integrator built around IC6, a comparator around IC7 and a reset switch TR2. In common with the control node the oscillator section introduces a number of fairly radical departures from the original design particularly around the comparator.

The original Minisonic v.c.o. was designed as a simple, low cost circuit and, as such, exhibited a number of features which are undesirable in an oscillator to be used primarily as a musical sound source. Basically the range of the original tended to be restricted by relatively slow switching rate of the comparator which, in that circuit, was a 741 operational amplifier.

Secondly the method of resetting the integrator was rather crude in that it depended on a fairly substantial current pulse being applied to the inverting input direct from the positive rail. Whilst effective, the arrangement introduced a peaky reset pulse on to the integrator output waveform which although too fast to be audible tended to introduce further problems if any attempt was made to synchronise the oscillators.

In the present design the latter problems have been overcome by replacing the 741 acting as comparator with the extremely fast LM318N, a device which has a specified slewing rate—in the feed forward mode—of 70V/microsecond i.e. it is some 140 times faster than the 741 it replaces.

Secondly the integrator reset is accomplished by a fast f.e.t. strapped across the integrating capacitor. In this design the width of the reset pulse is governed principally by the period required to discharge the integrating capacitor rather than the speed of the devices initiating the reset.

OPERATION

The operation of the circuit is as follows: the c.c.g. in the control node draws a constant current from the inverting input of IC6 thereby causing it to ramp in a positive direction. IC7 is held hard negative at this stage by the combined effect of the divider R77–R78 and positive feedback via R76 all of which establishes a threshold level. When the positive going ramp from IC6 exceeds the threshold level IC7 switches rapidly positive turning on TR2, discharging C5 and causing the output of IC6 to return to its minimum level again at which point the cycle repeats.

In order to achieve the fastest possible discharge of C5 it is necessary, in this circuit, for a fairly substantial potential difference across TR2 and thus the threshold level set by R77–R78 is around 6·3V. In practice this means that the integrator ramps about a point approximately +5V. It is therefore important that care is taken to correctly observe the polarities of the coupling capacitors C6 and C7.

With the component values given this oscillator produces an exceptionally clean waveform of nominally 1V peak-to-peak up to a frequency of about 50kHz–60kHz. If driven above this frequency the output waveform tends to take on a triangular shape as the rate of integration approaches the rate of reset.

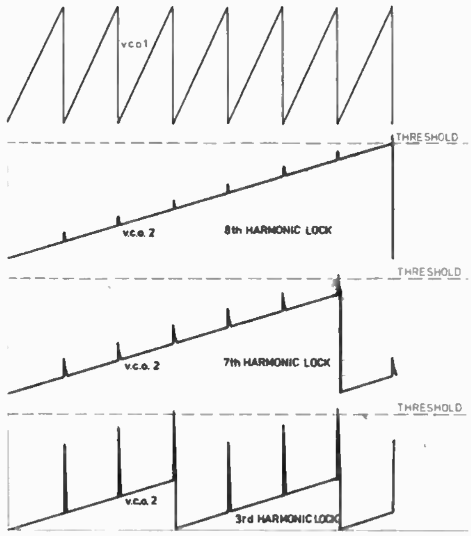


Fig. 5. Oscillograms showing locking at different harmonics

Over the entire audio range however the output waveform is, for all practical purposes, a perfect sawtooth.

OSCILLATOR SYNCHRONISATION

In electronic organ circles there is a continuing dialogue as to the relative merits of phase locked or free phase operation. In the synthesiser it is generally accepted that the oscillators work in the free phase mode, indeed, some oscillator designs go to great lengths to avoid any possibility of phase lock occurring.

The success of free phase operation depends, in the synthesiser, very largely on the accuracy with which the matching of oscillator performance has been carried out. In the Minisonic Mark II it is demonstrably possible to match the oscillators within a few hertz over a range of three to four octaves. Above this the beats between the oscillators can become sufficiently fast as to create discords.

Some thought was thus given to the feasibility of providing a phase locking facility which could be used particularly for exercises involving the ring modulator where discordant beats in the oscillators also give rise to sum and difference discords thereby greatly accentuating the effect.

SYNC COMPONENTS

Referring to Fig. 4, phase locking is achieved by sampling the output of the comparator through a gating diode D1 and divider R79–VR16. The resultant positive going spike of maximum 4·75V is then applied to the non-inverting integrator input of the other v.c.o. The effect is to superimpose the spike on the output waveform of the other integrator as shown in Fig. 5 with the result that when the combined value of the ramp and spike exceeds the threshold level the second comparator will switch and restart the cycle.

Several interesting and useful effects are possible. Suppose, for example, that it is required to synchronise the oscillators at the musical interval of a fifth. The drill is to set VCO1, the oscillator supplying the sync pulse, to run at the lower frequency and to tune VCO2 to achieve a fifth. VR16 is then advanced—only a small movement will be needed—until phase lock is achieved. This point is clearly audible and characterised by a fixed beat difference between the oscillators. When now programmed from the keyboard both oscillators will stay in tight lock over their entire range.

Another method is to set VCO1 to run at about eight or nine times the frequency of VCO2 and again to apply the lock signal. Here again tight phase lock is achieved but, with the wide frequency difference between the oscillators, it is possible to apply a greater level of synchronising signal and achieve lock again at a lower harmonic.

At the lower harmonic the amplitude of the ramp portion of VCO2 integrator output waveform has been decreased since the synchronising signal is of greater level. A further increase in sync level reduces the harmonic ratio still further with a proportional reduction in ramp level until, at the extreme, phase locking in unison is achieved. Fig. 5 shows schematically how the locking at different harmonics appears on the oscilloscope.

ADVANTAGES OF PHASE LOCKING

It is perhaps superfluous to state that this facility greatly extends the versatility of the system. Phase locking in conjunction with the ring modulator can create a range of remarkably rich chords which, in turn, give the impression that the instrument has more than its two oscillators. Additionally, the ability to achieve lock at a high harmonic ratio means that, with the ring modulator, it is possible to build up a series of high order harmonics for the purpose of reinforcing "brassy" type sounds.

CROSS MODULATION

Another feature of the Minisonic Mark II is one taken from the P.E. Sound Synthesiser, that of enabling the oscillators to cross modulate one another.

The integrator output is sampled via C6 and level control VR14 and is switchable into the control node of the other oscillator. Since the output of the integrator is nominally 1V then each oscillator is capable of modulating the other over a range of two octaves. Although this is well below the modulation range of the P.E. Sound Synthesiser it is nevertheless sufficient to allow the formation of a whole series of harmonically rich waveforms.

On its own this facility enables the creation of a substantial range of sounds which are closely imitative of conventional acoustic instruments. In conjunction with the phase locking facility the range is still further extended.

In Fig. 4 VR14/1 provides the variable signal level for cross modulation—and for oscillator modulation of the filter—while VR15/1 controls the audio signal amplitude.

H.F. OSCILLATOR

As was explained previously, the method of triggering the envelope shapers in the Minisonic comprises the application of a high frequency signal to the keyboard controller and thus to the divider resistors. The d.c. level and a.c. signal are extracted from the divider chain and presented to circuits which can detect, on the one hand, the d.c. level only and, on the other, the a.c. signal.

This particular method was indicated when the instrument was to have been operated by means of a stylus and presents the advantage that a single make contact only is required in the keyboard assembly.

The circuit of the h.f. oscillator is shown in Fig. 6 and is unchanged from the original P.E. Minisonic.

VR17 sets both the frequency and the amplitude of the h.f. oscillator output. As the slider of VR17 approaches the R80 end of its travel a greater level of positive feedback will be applied to IC8 thereby reducing frequency and increasing the amplitude of the output signal. The converse applies as VR17 is moved in the opposite direction. With the output of IC8 unloaded the waveform is effectively square gradually reducing to a more trapezoidal shape and ultimately to a triangular waveform as frequency is increased.

With the values given the oscillator will operate over the range 2kHz–250kHz with the optimal frequency for envelope shaper triggering being around 60kHz.

Setting of the oscillator frequency should be finalised during the instrument lining up stages since there is some room for adjustment around the optimal frequency particularly when the oscillator is loaded by the input circuitry to the keyboard controller and the subsequent loading of the divider by the hold capacitor.

These factors will affect the amplitude of the signal reaching the h.f. detector. Too low an amplitude and the detector will not trigger satisfactorily whilst too much and the detector will remain permanently on.

H.F. DETECTOR

The circuit of the h.f. detector is shown in Fig. 7. A 709 operational amplifier is chosen in order to obtain the benefit of the higher gain bandwidth of this device.

The 709 is connected as a follower having a gain of around 660 and is decoupled from the keyswitch bus by means of C11. C10 provides additional decoupling for the bus thereby ensuring that hum signals do not cause spurious triggering of the detector and thus the envelope shapers.

The output of IC9 provides drive to TR3 which, in turn is coupled to the envelope shapers through D3 and also to the hold isolator mentioned earlier through R88. Under quiescent conditions the output of IC9 is nominally zero volts and TR3 is off. An a.c. signal of sufficient level on the keyswitch bus will cause IC9 to follow with each positive excursion

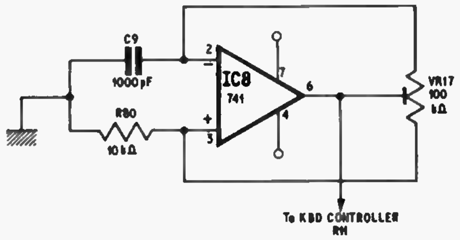


Fig. 6. Circuit of h.f. oscillator

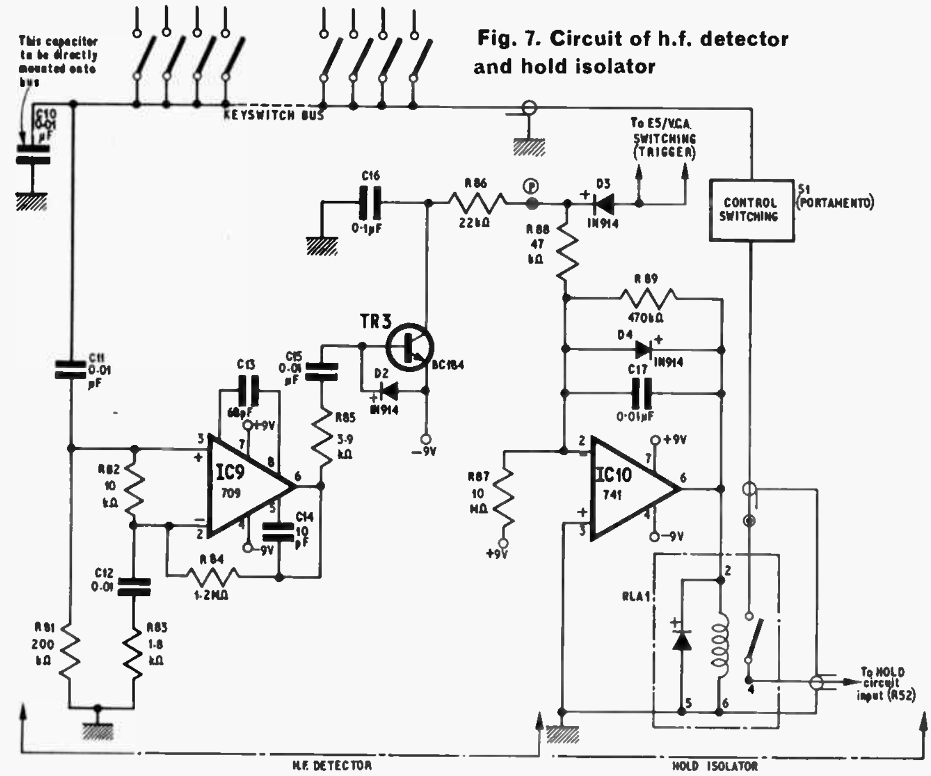


Fig. 7. Circuit of h.f. detector and hold isolator

COMPONENTS . .

HF OSCILLATOR /DETECTOR

Resistors
R80 10kΩ
R81 200kΩ
R82 10kΩ
R83 1·8kΩ
R84 1·2MΩ
R85 3·9kΩ
R86 22kΩ
All ±5% ¼W carbon

Capacitors
C9 1000pF
C10–12 0·01 ceramic
C13 68pF
C14 10pF silver mica
C15 0·01μF ceramic
C16 0·1μF ceramic

Potentiometers
VR17 100kΩ preset

Semiconductors
IC8 741
IC9 709
TR3 BC184
D2 1N914

HOLD ISOLATOR

Resistors
R87 10MΩ
R88 47kΩ
R89 470kΩ
All ±5% ¼W carbon

Capacitors
C17 0·01μF ceramic

Relay
RLA1 Type A d.i.l. (R.S.)

Semiconductors
D3–4 1N914
IC10 741

of IC9 output turning TR3 on and causing the collector to go to about −8·5 volts.

During the negative going excursion of IC9 TR3 is off but the time constant of its collector capacitor is sufficient to maintain the negative level on the collector until such time as a further positive going signal is received from IC9. C16 thus acts in a smoothing capacity so that the collector of TR3 moves unipotentially in the presence of a triggering signal from IC9.

Correct functioning of the circuit may be checked by monitoring point "P" (Fig. 7) when the h.f. detector is connected to the envelope shapers. In these circumstances the bias which is holding the envelope shapers off is reflected as a potential of between +0·3 and +0·9V at "P". When the h.f. detector is triggered this voltage falls to around −2·2V.

THE HOLD ISOLATOR

The hold isolator circuit is a completely new addition to the original Minisonic. In the prototype keyboard instrument it was found that the relatively long length of wire between the keyswitch bus and the hold capacitor was sufficient to load the capacitor and thus considerably increase the rate of drift of the hold circuit. Whilst this was of little importance during what might be termed normal playing of the keyboard it became much more significant when the very long decays of which this instrument is capable were being employed.

Essentially the circuit comprises an inverter having a gain of ten which drives a reed relay interposed between the keyswitch bus and the hold capacitor. Under quiescent conditions the inverter (IC10) receives positive inputs from R88 and R87 which combine to try and drive the output negative. A full negative excursion of IC10 is, however, prevented by D4 which limits the negative going output to around −0·6V.

When the h.f. detector triggers and point "P" goes to −2·2V IC10 will be forced into positive saturation thereby activating the relay coil and closing the switch. C17 in the feedback loop of IC10 provides additional smoothing so that any fluctuation of level at point "P" will not be reflected in the hold isolator circuit in terms of contact bounce of the relay.

The relay itself incorporates a diode to protect the driving circuitry against the back e.m.f. generated when the driving signal is removed.

COMPONENTS . .

ENVELOPE SHAPER
(2 REQUIRED)

Resistors

R90	75kΩ
R91	3·9kΩ
R92	560Ω
R93	20kΩ (ES1 only)
R94–95	1kΩ (2 off)
R96*	10kΩ
R97	10kΩ
R98	8·2kΩ→18kΩ (see text)
R99	910Ω
R100	680Ω
R101	680Ω
R102	560Ω
R103	39kΩ
R104	3·9kΩ
R105	6·8kΩ (ES1 only)
R106–7	6·8kΩ

All ±5% ¼W carbon

Potentiometers

VR18	1MΩ log.
VR19	250kΩ log.
VR20	20kΩ–22kΩ preset
VR21	25kΩ log.

Capacitors

C18	10μF tantalum
C19	1μF tantalum
C20	0·1μF tantalum
C20a/b	680pF ceramic/ 10μF tantalum

Indicator
LP1 miniature red l.e.d.

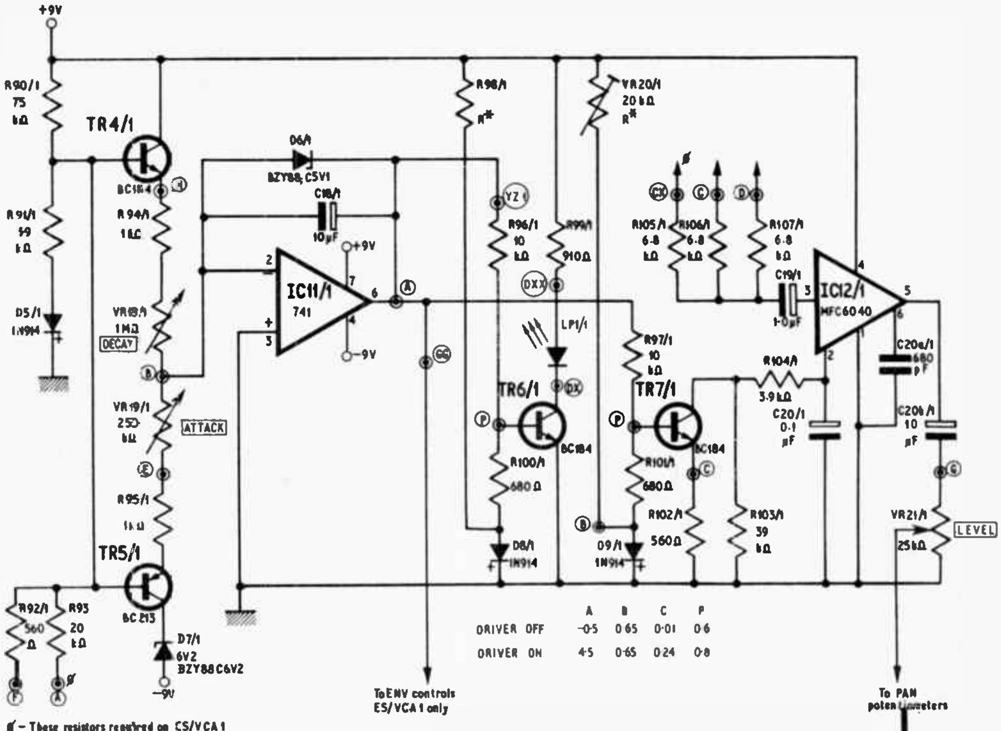


Fig. 8. Envelope shaper and voltage controlled amplifier. R* should be adjusted for 0·6V at P.

Semiconductors

D5	1N914
D6	5·1V 400mW Zener
D7	6·2V 400mW Zener
D8	1N914*
D9	1N914

TR4	BC184	TR6	BC184*	IC11	741
TR5	BC213	TR7	BC184	IC12	MFC6040

Note: Items marked * are for proportional l.e.d. driving. If l.e.d. is to be driven from IC11 direct, these may be omitted and R99 changed to 750Ω

ENVELOPE SHAPERS AND V.C.A.s

The theoretical circuit of the envelope shaper and voltage controlled amplifier is shown in Fig. 8. This circuit incorporates a number of modifications over the original Minisonic. Firstly the circuit around the integrator IC11 is very much simplified in that three diodes and one capacitor have been removed without significantly changing the operating characteristics of the circuit.

Secondly R*, originally a fixed value resistor, has been changed to a preset in order to ease the problem of biasing up the v.c.a. driving transistor.

Finally an l.e.d. has been incorporated into the circuit in order to provide a front panel indication of envelope state. Drive to the l.e.d. is through TR6 which is biased to a point just below conduction. when the envelope is in its most negative state i.e. −0·6V.

Additional components required for a Minisonic conversion are R96; R98; R99; LP1; TR6; R100 and D8 and the l.e.d. A simpler form of conversion is to drive the l.e.d. directly from the output of IC11. In this case the series resistor, R99 in the circuit given, should be 750 ohms.

It should be noted that if the latter form of conversion is employed there will be a propagation delay between the time that the envelope shaper starts its excursion and that at which its envelope level is sufficient to illuminate the l.e.d.

Additionally it is important that, if the direct drive system is employed, the l.e.d. chosen should be of the low current type.

E.S. OPERATION

The operation of the envelope shaper is as follows: IC11 is a linear integrator whose output voltage excursion is bounded in both negative and positive directions by the Zener diode D6. With the output of IC11 tending to go negative D6 acts as a normal diode such that the integrator output is held to −0·6V. As the integrator moves positively the output voltage is bounded to the limit set by the Zener voltage of D6. Thus the output voltage excursions of the integrator range between −0·5V and +4·5V.

In the quiescent condition R90; R91 and D5 set the bias on TR4 and TR5 such that TR5 is off and TR4 is on. Current reaching the inverting input of IC11 via TR4, R94 and VR18 charges C18 and thus, with the aid of D6, holds the integrator output at −0·5V.

When a negative going trigger signal is applied via R92 or R93 TR4 turns off and TR5 turns on. The charge on the integrating capacitor C18 thus leaks away via VR19, R95 and TR5 and the integrator output ramps in a positive direction until it reaches the bounded value set by D6. When the trigger signal is removed TR4 and TR5 revert to their original states and the output of the integrator ramps negatively until it reaches −0·5V. The rate at which ramping takes place, in either direction, is controlled by the value of the resistance in series with these latter transistors. Thus VR19 controls the rate of attack of the envelope and VR18 the rate of decay. Both these potentiometers should be of logarithmic law ideally since this type offers much better control when small changes in attack and decay rates are required close to the minimum values possible.

DIVIDER NETWORK

The integrator output is linked to a divider network R97–R101 to the base of TR7 which, with the output of the integrator at 0·5V, is held at the point of conduction by means of a current supplied from the positive rail by means of VR20. The table in Fig. 8 gives the "on" and "off" d.c. conditions around TR7 and VR20 should be adjusted so that these conditions are met as closely as possible.

The setting of VR20 is quite critical since too much conduction in TR7 in the quiescent envelope state will mean that the gain/attenuation range of the v.c.a. is restricted with the possible result that signal breakthrough will be present. Too little conduction on the other hand will result in a propagation delay between the onset of an envelope and the appearance of the audio signal at the output of the v.c.a. Thus fast attack times would not be possible in this latter condition since the produced sound would be of very short duration and severely attenuated.

E.S. TRIGGERING

Triggering of the envelope shapers can be accomplished in a number of ways, Fig. 9 indicating the arrangement of switching for the trigger signals. Envelope shaper VCA2 is permanently linked to the output of the h.f. detector and thus this circuit will always be operational whenever keys are depressed on the keyboard.

Envelope shaper VCA1 can also be triggered from the keyboard when S21 is closed. Additionally this latter circuit may also be triggered via S22 and S23 the "Hold" and "Trigger" buttons respectively.

With S21 open, closure of S22—a latching switch—will ensure that the envelope shaper is in the "on" condition and that signals to its related v.c.a. will be audible whatever is happening to the keyboard.

S23 is a momentary action switch. With S21 open, closure of S23 will create an envelope and make audible any input signal to VCA1 in addition to signals from VCA2 which are programmed through from the keyboard. Methods of use for these trigger switches will be described in a later section of this article.

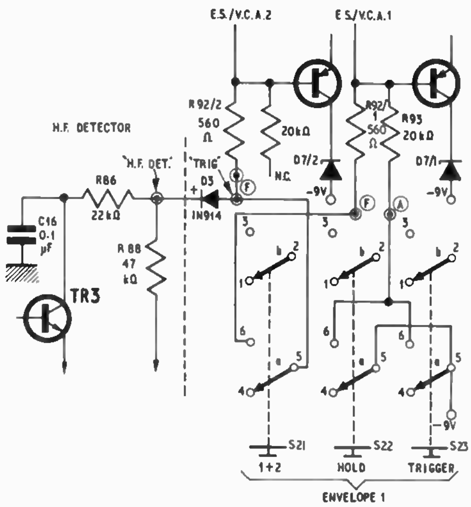


Fig. 9. Arrangement of switching for the E.S. trigger signals

Note that operation of either S22 or S23 will not affect the hold isolator since D3 provides a gating action in this respect.

ELECTRONIC ATTENUATOR

The v.c.a., or to give it its proper title, electronic attenuator, is a purpose designed i.c. by Motorola.

The specification of the device is to provide an attenuation of 77dB and a gain of 13dB relative to the input signal which should not exceed 500mV r.m.s., when the current sink from the control input (pin 2) is varied from minimum to maximum respectively. In the Minisonic II the relatively low operating voltages result in a reduction of the overall attenuation/gain range to about 54dB which is sufficient for most practical purposes.

The current sink from pin 2 of IC12 is, in the off condition, restricted by the series combination of R103 and R104. As TR7 turns on it progressively short circuits R103 with the result that the current sink increases proportionately to a maximum which is limited by R104. It should be mentioned, of course, that the linear envelope of IC11 is converted into a negative exponential characteristic by TR7.

Although this is not ideal for an audio signal envelope, experience has shown that it is extremely difficult to differentiate subjectively between a negative exponential and a positive exponential, or square law, envelope which is considered to give the best effect.

SIGNAL ROUTING

Having given consideration to the v.c.o.s and envelope shaper v.c.a.s which are the mainstays of the audio signal system in the Minisonic we will now look at the methods of signal routing between them.

Any method of switched patching arrangement between signal sources and treatments is bound to be somewhat restrictive when compared with a "free-for-all" patchcord system which enables virtually any module combination to be made. In the Minisonic II therefore very careful thought has gone

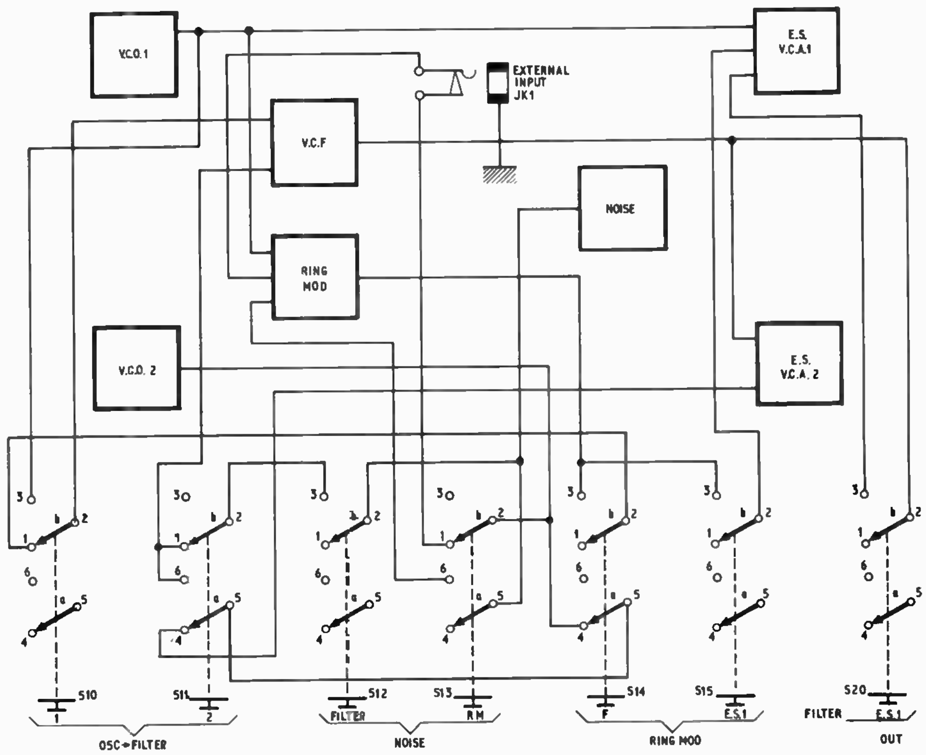


Fig. 10. Arrangement of signal switching

into the switching options available with the view of combining maximum flexibility with the simplest possible arrangement of switches. Fig. 10 illustrates the system finally adopted.

The first thing to remember is that all signals reaching the output stages of the Minisonic have, first of all, to pass through the v.c.a.s in which they are modified by the envelope characteristics. This applies equally to signals which originate in the instrument or those which are supplied from external sources through the input breakjack.

In the case of VCO1 the output signal is routed directly to envelope shaper/VCA1 and also to the ring modulator. An indirect link is provided by means of S10 which enables VCO1 signal to be routed into the voltage controlled filter (v.c.f.).

From VCO2 the output signal can only reach its related envelope shaper/v.c.a. by an indirect route via S11. With this latter switch in one position VCO2 signal goes direct to its v.c.a. whilst, in the other, the signal is routed into the voltage controlled filter. VCO2 signal is also routed to the ring modulator via S13 and the input breakjack.

OVERRIDE

Thus the application of an external signal into the breakjack will override the routing and, similarly, closure of S13 will override VCO2 signal into the ring modulator and substitute that of the noise generator.

Both the noise generator and ring modulator output signals can be routed into the filter but only through S10 and S11. The oscillators thus take precedence as far as filtering is concerned and operation of either S10 or S11 will respectively override the ring modulator and noise generator.

Finally, the filter output is linked directly to envelope shaper/VCA2 and can, in addition, be linked to envelope shaper/VCA1 through S20. The ring modulator output has two output routing options, to the filter through S14 and/or to envelope shaper/VCA1 through S15.

Some consideration of the foregoing in combination with Fig. 10 will show that it is possible to ring the changes in terms of output signal combinations and over quite a wide range. Furthermore, since each switch provides an "either/or" arrangement the system as a whole presents a flexible and rapid means of changing from one signal form to another.

VOLTAGE CONTROLLED FILTER

The v.c.f. is perhaps the most useful of the signal processing circuits in any synthesiser. Its principal function is to vary the passband of signals passing through the circuit thereby proportionally varying the harmonic content of complex signals.

Whilst it is possible to apply a fixed passband characteristic to the filter it can also impart a tracking capability such that a succession of musical tones from anywhere in the audio range will have the same, or closely similar, harmonic content. Furthermore it is also possible to vary the passband during the audible period of a signal such as to impart an amusing characteristic to the sound. Each of these methods of operation finds an important niche within the tone colour spectrum of the synthesiser.

The v.c.f. is a two part circuit comprising a control node similar in form to that used in the v.c.o. and a ladder network in which the filtering action takes place.

CONTROL NODE

Fig. 11 illustrates the control node. Although quite similar to the control node of the v.c.o.s there are nevertheless some important differences. The first is that temperature compensation has been abandoned since it has been found that stability in this respect is not of prime importance. Secondly the number of modulation sources has been greatly increased and, since these can be operated in any combination, it is necessary to provide some form of current limitation so that the filter does not overload the reference voltage supplies.

For Minisonic conversions the additional components required are: D10; R112; VR24; and R116–R118.

IC13 is an inverting amplifier which provides drive to a constant current generator TR8. The gain of the amplifier is prescribed by adjustment of VR24 and VR25 which give fine and coarse control respectively. D10 provides a limit to the output of IC13 and therefore sets a limit to the current through TR8. The effectiveness of D10 in this respect depends to a certain extent on the gain of TR8 and also on the absolute value of the Zener voltage.

SETTING-UP

The setting-up follows similar lines to that of the v.c.o.s in that the gain controls VR24 and VR25 are adjusted so that each 500mV increment of control voltage applied by VR23 will result in a doubling of the current through TR8. With a control voltage of −6V at the wiper of VR23 the bias control should be adjusted so that the current through TR8 is a maximum of 10μA.

Advance the frequency control VR23 so that its wiper voltage is at −9V. The output of IC13 should now be the sum of the wiper voltages at VR23 and VR22 times the gain constant of IC13 which is around 0·69. In other words the output of IC13 will be in the region of +6·0V ±0·25V.

Under these conditions the current through TR8 will be around 2·5mA but bear in mind that due to the self-heating in the transistor this current will gradually increase until the heating effect of the current is balanced by the heat loss from the transistor body. It is prudent therefore to make the current measurement early in the proceedings.

A second control voltage potentiometer should now be connected temporarily as for VR23 and its wiper linked into the inverting input of IC13 through a 47 kilohm resistor. With the wiper initially at the earthy end and a wary eye on the milliammeter monitoring current through TR8 advance the wiper until either the current through TR8 reaches 10–12 milliamps or, until the output of IC13 limits at +7·5 volts.

COMPONENTS . . .

VOLTAGE CONTROLLED FILTER

Resistors

R108–109	6·8kΩ (2 off)	R121	390Ω
R110–111	47kΩ (2 off)	R122	1kΩ
R112	27kΩ	R123	390Ω
R113	2·2kΩ	R124	1kΩ
R114	120Ω	R125	15kΩ
R115–118	47kΩ (4 off)	R126	20kΩ
R119–120	2·2kΩ (2 off)	R127–128	470kΩ (2 off)

All ±5% ¼W carbon

Capacitors

C21–22	10μF tantalum (2 off)
C23–26	0·047μF polyester (4 off)
C27–28	10μF tantalum (2 off)

Potentiometers

VR22	10kΩ preset	VR26	10kΩ preset
VR23	10kΩ linear	VR27	10kΩ log. or lin.
VR24	1kΩ preset (20 turn)	VR28	1kΩ linear
VR25	10kΩ preset (20 turn)		

Semiconductors

D10	7·5V 400mW Zener
D11–D22	1N914 (12 off)
TR8–TR10	BC184 (3 off)
IC13	741
IC14	741

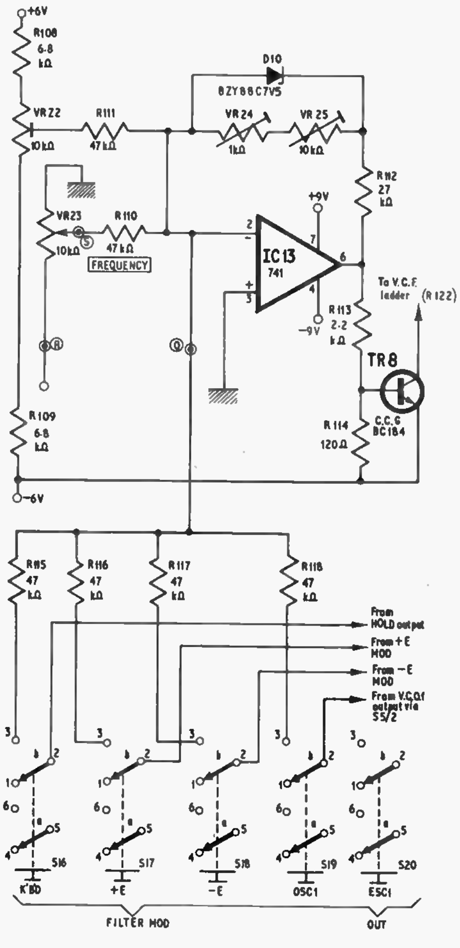


Fig. 11. V.c.f. control node

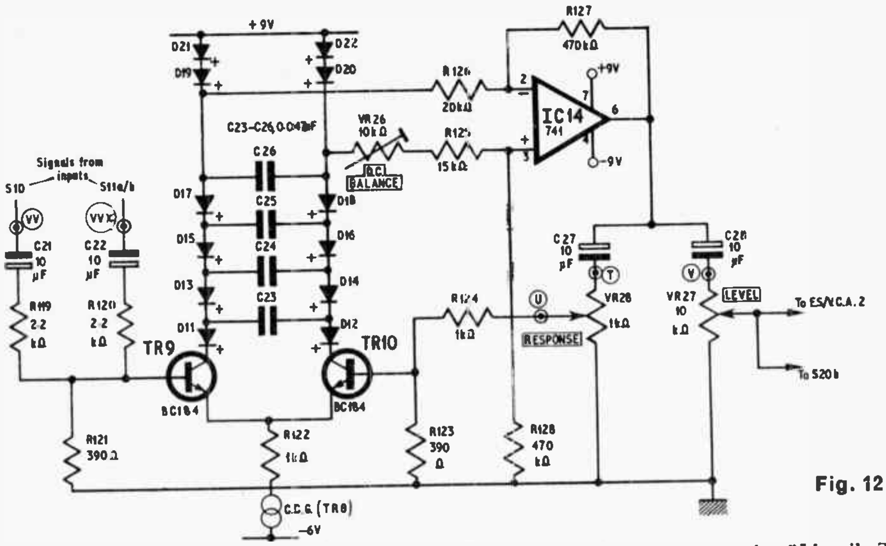


Fig. 12 Low-pass filter ladder network

INTERPRETING THE MEASUREMENTS

If both these measurements are noted to occur at the same time then all is well and further advances to the second control voltage potentiometer will not increase the current through TR8. If, however, the current reaches its maximum figure before IC13 limits then the value of D10 will have to be reduced to, say, 6·8V.

In the event that the 6·8V limitation prevents the maximum current of 10–12mA being achieved then there are two courses of action possible. If the current discrepancy is small, i.e. less than 0·5mA then it can be ignored. Where the discrepancy is around 1mA or more a resistor can be placed in series with D10 and adjusted in value as necessary until, with IC13 limiting, the current through TR8 is close to the specified value.

Having made the above adjustments any further increases in control voltage will have no effect on the current generator and the upper limit of operation has been set. It now remains to increase the minimum current so as to fix the lower operating point.

For this adjustment the temporary control potentiometer should be removed and the wiper of VR23 returned to the earthy end. VR22 should now be adjusted so that the minimum current through TR8 is around 35μA to 40μA.

LOW PASS FILTER

The theoretical circuit of the low-pass filter ladder network is shown in Fig. 12.

The ladder itself comprises a series of diodes in combination with capacitors. The diodes present an impedance which varies inversely as the current through them. In other words at low levels of current the impedance is high and vice versa. Thus, as the current through the ladder is varied, the effective resistance in combination with each capacitor will also vary and thereby change the turn-over frequency of each separate combination.

The ladder terminated in transistors TR9 and TR10 which are effectively biased on by referring their bases to the 0V rail. Thus any current drawn through the network by means of the c.c.g. will pass, without restriction, through these transistors.

An audio signal applied to the base of TR9 will, however, create a proportional variation in the current through the transistor and thus also a voltage variation at each diode junction in the ladder. This applies virtually to every value of current drawn by the c.c.g. so that, for a given level of a.c. signal, the smaller the current through the network the smaller will be the proportional variation induced by the signal. Thus the concept of variable impedance is, in fact, due to a combined effect of diode, transistor and current generator.

SEVERAL DECADE RANGE

The circuit has a range which extends over several decades and, as shown, the −6dB passband is, at maximum, from nominally 3Hz to 15kHz.

Four filter stages are cascaded in the ladder network and since each stage has a theoretical roll-off of 6dB/octave the maximum roll-off of the filter should be 24dB/octave. Efficiency in this respect however can only be achieved if every precaution is taken to prevent loading of the network at the point of entry and the point of extraction of the audio signal. In the interests of simplicity and economy buffer stages have not been included in this circuit but, even so, the roll-off possible is around 12dB–15dB/octave.

The output from the ladder network is amplified differentially by IC14 with VR26 being employed to minimise any d.c. imbalance caused by variations in diode characteristics. Such variations can be minimised at the outset by individually selecting the diodes such that each pair presents identical forward voltage characteristics. Even so it will be found that there is a degree of d.c. offset appearing at the output of IC14 when the ladder current is increased from minimum to maximum. VR26 should therefore be employed to centre the offset about zero volts such that when the filter is operating around midrange the d.c. output of IC14 is zero.

OUTPUT

The output from IC14 is capacitor coupled into two potentiometers. VR27 is simply the output level control while VR28 is the feedback or "Q" control. VR28 may also be referred to as the "Response" control. With VR28 wiper at zero volts the base of TR10 is referred closely to the 0V rail and thus TR9 and TR10 behave essentially as a differential pair. The output of IC14 is therefore nominally in phase with the input signal at the base of TR9.

As VR28 is advanced from zero a proportion of the output signal appears at the base of TR10 thereby tending to induce a signal in the collector circuit which is oppositely phased with the signal already present due to the effect of the input signal on TR9. The result is that the output signal will become significantly attenuated except at the frequency whose period is equal to the adjusted time constant of the network.

At this critical frequency the output of the filter will peak up with the bandwidth of the signal depending largely on the degree of feedback applied.

Further application of feedback will cause the filter to oscillate. The frequency of oscillation is proportional to the current through the ladder network and the oscillation, which is of sine form, will be superimposed on the filter output signal. With the current limits quoted for the revised control node design the filter oscillates from around 2kHz to greater than 30kHz.

FILTER CHECK-OUT

Checking filter performance can most effectively be carried out after assembly of the complete instrument. For this purpose the audio output of VCO2 should be patched into the v.c.f. by means of S11 with the monitoring signal for headphones or external power amplifier being taken from channel 2. Initially the v.c.f. frequency and "Response" controls should be at zero and VCO2 set to run at a middle range audio frequency.

Advance the v.c.f. frequency control to maximum. As this is done the v.c.o. signal should become audible in the monitored channel rising from a fairly bland sound to the full harsh bite of the sawtooth waveform as the frequency control of the v.c.f. approaches its maximum setting.

Repeat this procedure with the "Response" control at both extremes. With "Response" at minimum the overall level of the sound should be somewhat greater than when it is at maximum but there will be less subjective change in the harmonic content of the resultant sound.

The next procedure is to check out the effect of automatically programming the v.c.f. signal. Advance ES1 attack and decay controls to approximately one third of their rotation. Connect by closure of S17, the output of the control envelope inverter into the control node of the v.c.f. Set the "control + envelope" level about halfway.

Closure of a keyboard contact will now result in a slow rise in audibility of the sound together with a distinct change in harmonic content as the sound becomes louder.

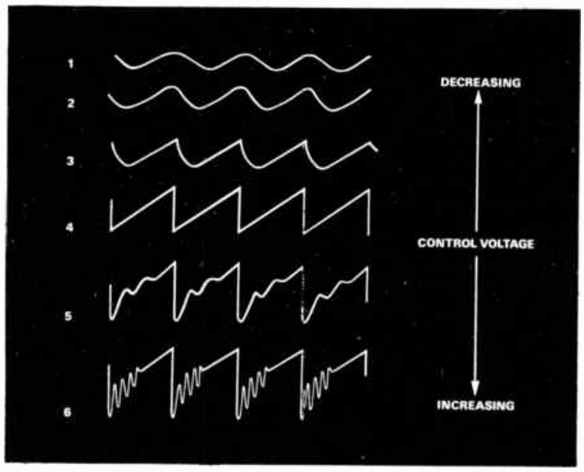


Fig 13. Effect of v.c.f. on a sawtooth voltage

Try various settings of the attack, decay and envelope level controls to achieve a typical synthesiser "Waa-Waa" effect.

USES OF THE V.C.F.

There are three principal ways in which the filter may be used as a sound treatment, of which two have been examined during the check-out procedure. Before going into these in any detail however let us look for a moment at what exactly it is that the filter does to the sawtooth waveform.

Fig. 13 illustrates a number of waveforms with the filter control voltage at different levels. In stage one the control voltage is very low i.e. with the frequency control just off the minimum end stop. If the sawtooth signal is around 1kHz say, the effect of the filter is to remove virtually all the upper harmonics leaving the fundamental which is almost of sine form.

Stage two and three illustrate the situation which occurs when the control voltage is increased successively; in each case the output waveform is assuming more of the sawtooth characteristic albeit still severely rolled off.

In stage four the control voltage is such as to allow the filter to admit the whole of the sawtooth without any roll-off.

RESPONSE CONTROL

The degree of roll-off of the filter is affected very largely by the amount of feedback admitted to the ladder network by means of the "Response" control. With "Response" at minimum the roll-off is much less accentuated and, indeed, the signal level from the filter is significantly greater than when the "Response" is at maximum.

Thus, with the "Response" at minimum the filter can act very much in the same way as a tone control i.e. passing all those frequencies lying below that set by the control voltage and rolling-off all those which lie above the set value at around 6dB per octave.

Increasing the feedback above a critical point will induce the filter to commence self oscillation.

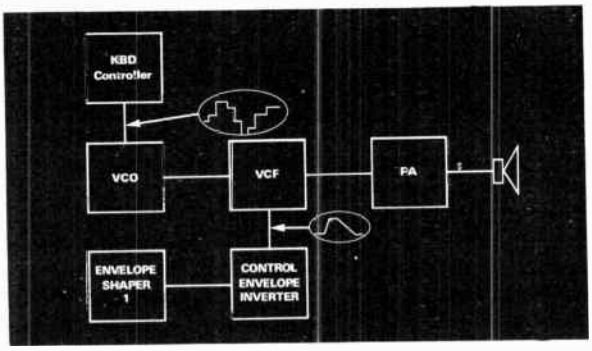


Fig. 14. Method of patching to achieve a v.c.a. or Waa-Waa

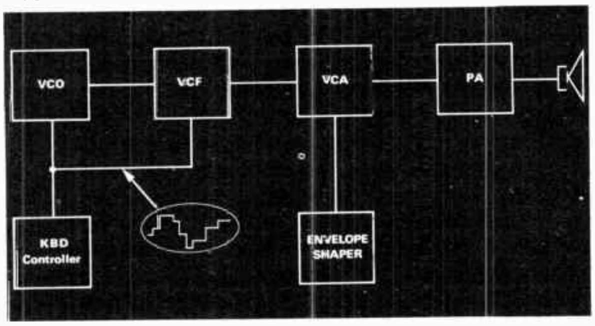


Fig. 15. Patch for v.c.f. to track the frequency of the v.c.o.s

Similarly when operating at high Q the filter will also begin to oscillate when the control voltage is advanced beyond a point where the input signal is wholly accepted. This situation is illustrated in stages five and six of Fig. 13, the frequency of oscillation being proportional to the increase in ladder current.

What applies, in general terms, to the changes occurring in a sawtooth waveform also applies to other waveforms which are rich in harmonics. In the case of a sine wave input however the effect of the filter is simply to cause a variable degree of attenuation to the signal in a manner dependent on the input frequency, control voltage and "Response" control settings.

USING THE FILTER AS A V.C.A.

Fig. 14 illustrates schematically the method of patching to enable the filter to act as an automatic Waa-Waa or as a voltage controlled amplifier.

In this case the positive output of the control envelope inverter is patched into the control input of the filter by closure of S17. The "Attack" control of ES/VCA1 should be at its minimum setting and the decay control at maximum.

Set the inverter level control about midway with the attack and decay controls of ES1 set about one third of their full rotation.

Close a key contact momentarily and when the resultant sound has decayed away—say in four or five seconds—adjust the frequency control of the filter so that the v.c.o. signal is just barely audible.

The keyboard may now be played in the normal way during which time the attack, decay and control envelope controls may be adjusted to achieve the desired effect. Note that the greater the level of the control envelope the harsher will be output signal when the envelope is at its peak.

An inverted Waa-Waa effect can be achieved by setting the filter frequency control to maximum and using the negative going envelope to programme the filter by closure of S18.

TRACKING THE V.C.O.s

With the arrangement of patching as shown in Fig. 15 the filter may be used to track the frequency of the v.c.o.s. This is because the control input of the filter is directly linked to the output of the hold circuit and thus variations in this level will adjust the passband of the filter.

This method of operation is particularly useful if the instrument is being used in an imitative sense or if the constructor wishes to achieve a softer, harmonically reduced output signal. With this mode, the keyboard should be played at the same time adjusting the filter frequency and "Response" controls until the desired sound is achieved.

It will be found that a number of acoustic instruments can be effectively imitated using this method. For example, wind instruments such as the horn and trombone, string instruments such as the violin and cello and a clarinet tone have all been successfully synthesised with the Minisonic.

THE FILTER AS A TONE CONTROL

In the previous method of operation the passband of the filter was continuously being adjusted as the keyboard was being played such that the proportion of harmonic roll-off was effectively constant regardless of the frequency of the input signal.

If switches S16–S19 are all opened, the v.c.f. passband is now entirely dependent on the setting of the manual frequency control. With this at maximum the filter will pass frequencies up to 15kHz (−6dB) more, in fact, than the Minisonic would normally produce in a strictly musical sense.

With the frequency control near its minimum setting the −6dB passband is only 3Hz and thus the greater part of any filtered musical signal from the v.c.o.s would not reach the power amplifier stages.

The filter is now acting as a treble cut system with the degree of cut obtainable being varied by the "Response" control. With this at minimum the roll-off is about 6dB per octave and at maximum about 15dB per octave.

THE RING MODULATOR

The Minisonic ring modulator is an improved version of the circuit which originally appeared in the *P.E. Sound Synthesiser* (August 1973). The essential features of the circuit is shown in Fig. 16.

The ring modulator produces a unique output waveform which comprises, at the same instant, the sum and difference between any two applied input frequencies. This function is carried out in a purpose-built integrated circuit, the SG3402. Both the "N" or "T" packages of this device are suitable.

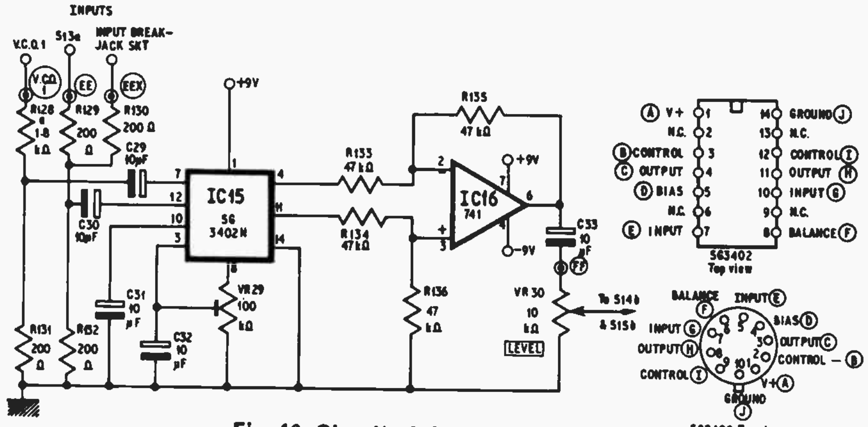


Fig. 16. Circuit of ring modulator

COMPONENTS . . .

RING MODULATOR

Resistors
R128a 1·8kΩ
R129–132 200Ω
R133–136 47kΩ
All ±5% ¼W carbon

Capacitors
C29–33 10µF tantalum

Potentiometers
VR29 100kΩ preset
VR30 10kΩ log.

Semiconductors
IC15 SG3402 "N" or "T"
IC16 741

With one of the input frequencies fixed, variation in the other will ring the changes in the output frequencies as shown in Table 1.

Referring to Fig. 16, R128 and R131 form an input attenuator on the so-called carrier input (pin 7) such that, when driven from a v.c.o., the input signal level at C29 will be about 40mV to 80mV.

Similarly R129 and R132 attenuate the modulator or control input so that, when driven by a v.c.o., the input at C30 is about 200mV to 500mV. This procedure results in an output signal of about 1·5 volts at pin 4 and the same signal in antiphase at pin 11. The antiphase signals are amplified differentially by C16 to give a peak output signal of three volts which is then applied via C33 to the level control VR30.

SETTING UP THE RING MODULATOR

Setting up the ring modulator is very simple. With the circuit completed, link the modulator input to the 0V rail and connect the output to a suitable power amplifier. Apply a signal of about 1kHz to the carrier input (normally connected direct to VCO1) and adjust VR29 until the output signal reduces to the lowest possible level. This should, with a correctly wired circuit, be 50dB or more below the peak signal level. At this point the ring modulator is correctly balanced with minimum carrier breakthrough.

MINOR CHANGES

The Minisonic II ring modulator features two minor changes from the original circuits. The first is provision of a second "modulator" input via R130 which leads to the external input socket. Secondly, the attenuating resistors on the output of IC16 have been removed to give a greater signal level. These were R9 and R10 in Fig. 3.7 of the original Minisonic articles.

TESTING THE RING MODULATOR

Patch the ring modulator output into envelope shaper VCA1 by closure of S15. VCO1 is connected directly to the ring modulator and similarly VCO2 is connected into the ring modulator through S13 and the external input breakjack (see Fig. 10).

With both v.c.o. level controls at about three-quarters setting, set their frequencies manually such that there is a low beat frequency discernible from the ring modulator. Listen carefully for this because the output of the ring modulator is being mixed with the output of VCO1 in VCA1. Turn the level control on VCO2 to zero. At this point the beat frequencies should entirely disappear.

Route the output of the ring modulator into the v.c.f. by closure of S14. The v.c.f. frequency control should be at maximum and the response control at minimum. The level control of VCA2 should also be at maximum.

With these settings the output signal of VCO1 should not be discernible in channel 2. If it is detectable then there is some signal breakthrough in the ring modulator and VR29 should be re-adjusted to minimise this effect.

Make sure that the panning controls (Fig. 18) are hard left and hard right respectively for this test otherwise the breakthrough detected could prove to be panned signal.

USING IT

In a musical sense the ring modulator can be used to produce rich chord structures. For example, with both oscillators tuned apart by the interval of a fifth, i.e. the frequency of one oscillator is 1·5 times the frequency of the other, the output of the ring modulator will be, in the case of the sum frequency 2·5 times, and in the case of the difference frequency 0·5 times the frequency of the oscillator producing the lowest pitch.

Table 1: OUTPUTS FROM THE RING MODULATOR

	Frequency						
Carrier	700	600	500	400	300	200	100
Modulator	400	400	400	400	400	400	400
Sum	1100	1000	900	800	700	600	500
Difference	300	200	100	0	100	200	300

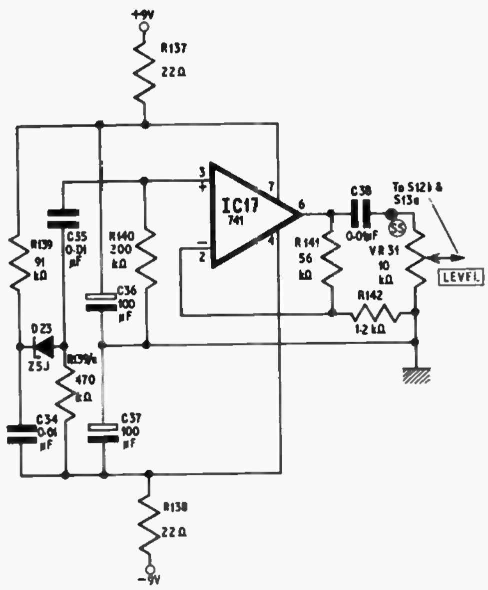


Fig. 17. Circuit of noise generator

If the output of this latter oscillator is taken as being the fundamental, then the output of the ring modulator may be said to comprise the sub-octave and twelfth with respect to the fundamental.

If this signal is now mixed with the outputs of the v.c.o.s originating the signals then the end result is a musically concordant four note chord.

Similar effects may be obtained when the oscillators are in unison, an octave apart or, as was explained in a previous section, tuned wide apart in phase lock. In all cases the richness of the resultant sound quite belies the simplicity of the system producing it.

OTHER RING MODULATOR EFFECTS

Apart from its musical possibilities the ring modulator may be extensively used in the production of sound effects. For example, with white noise patched into its uncommitted input (closure of S13) and with VCO1 running at low frequency—say around 10Hz—the reset point of the sawtooth will be differentiated by the ring modulator input decoupling capacitor such that the output of the ring modulator will comprise a series of staccato cracks akin to machine-gun fire. Filtering of the output signal can ring the changes quite widely over this one, very simple sound.

Dalek type voices can be produced by patching a crystal microphone or a pre-amplified magnetic microphone signal into the external input socket. A range of effects may be achieved by varying the frequency of the v.c.o. between about 20Hz and 1kHz bearing in mind that the greater part of the resultant audio signal will be derived from this oscillator.

A previously taped signal may also be connected into the external input socket and ring modulated with musically programmed v.c.o.s in order to produce some really weird and exotic sound forms.

COMPONENTS . . .

NOISE GENERATOR

Resistors

R137–138	22Ω (2 off)	R140	200kΩ
R139	91kΩ	R141	56kΩ
R139a	470kΩ	R142	1·2kΩ

All ±5% ¼W carbon

Capacitors

C34–35	0·01μF ceramic (2 off)
C36–37	100μF elect. 25V (2 off)
C38	0·01μF ceramic

Potentiometers

VR31 10kΩ log.

Semiconductors

D23	Z5J (Z1J)
IC17	741

WHITE NOISE GENERATOR

The noise generator is a very simple circuit built around the highly successful Z1J (now Z5J) noise diode manufactured by Semitron Ltd. The circuit is shown in Fig. 17 and comprises the noise diode D23 whose output drives a high gain amplifier IC17. Output from the amplifier is decoupled by C38 and led to its level control VR31. The maximum output from this circuit should be around 2V peak-to-peak but variations may occur as a result of differences in the characteristics of the noise diode. If the output is significantly below the specified maximum some reduction in the value of R139/a may be tried with the view of increasing the current throughput of the diode.

If this does not appear to have any effect then the gain of IC17 may be increased by reducing the value of R142. In circumstances where the gain may have to be substantially increased there is a danger that drift in IC17 may impart a d.c. offset at the output which results in clipping of the noise signal. In these circumstances R140 should be reduced sufficiently to overcome the effect.

Under normal circumstances the noise generator should not require any special setting up procedures.

OUTPUT STAGES

The principal change in the output stages of the Minisonic relates to the inclusion of panning controls by means of which the respective outputs from the voltage controlled amplifiers may be mixed to create a stereo image. Additionally, the gain of the mixers has been increased to times 2 to fulfil the requirement for a stronger signal at the output.

The circuit of the output stages appears in Fig. 18. VR32a/b–1 and VR32a/b–2 are the panning controls. Although linear potentiometers are employed a log-antilog characteristic is imparted by the addition of resistors R143 and R144 strapped between the high end of each track and its respective wiper. The signal transition between one channel and the other is therefore accomplished very smoothly.

COMPONENTS

OUTPUT STAGES
(2 REQUIRED)

Resistors

R143–144	22kΩ
R145–146	47kΩ
R147	100kΩ
R148	1·5kΩ
R149	910Ω
R150	1kΩ
R151	4·7kΩ
R152	10kΩ

All ±5% $\frac{1}{4}$W carbon

Capacitors

C39	4·7μF tantalum
C40	3,300pF polystyrene
C41	470μF elect. 25V
C42	0·005μF ceramic

Potentiometers

VR32a/b	100kΩ lin. ganged
VR33a/b	10kΩ log. ganged (1 only required)

Semiconductors

IC18	741
IC19	MFC4000B

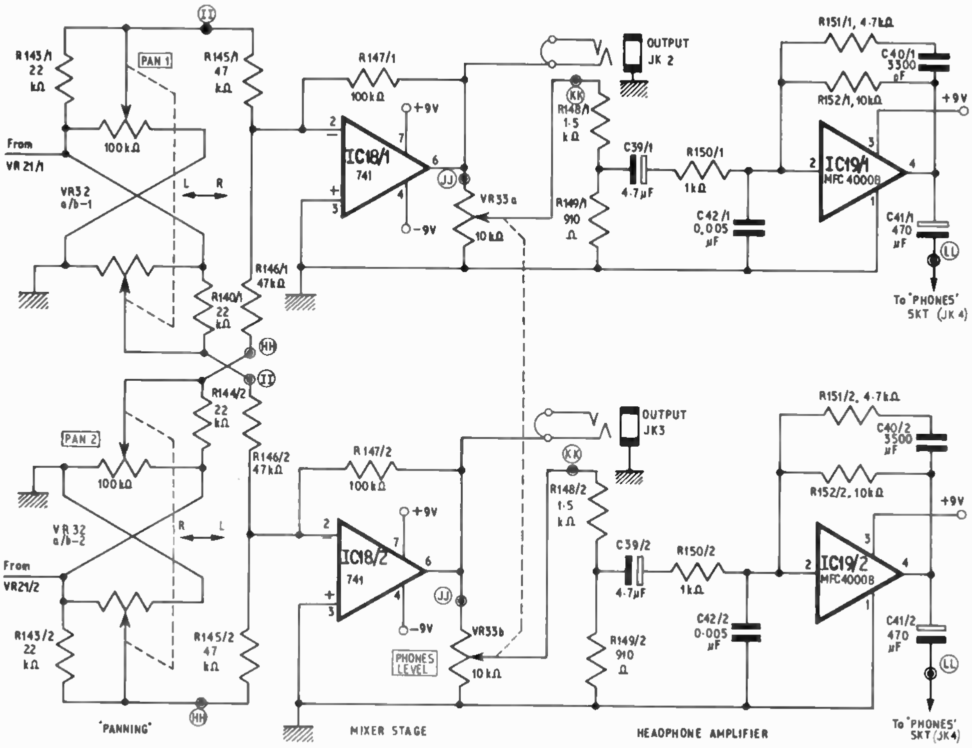


Fig. 18. Circuit of output stages

IC18/1–2 are the mixer stages, each having two inputs. The outputs from the mixer stages drive into VR33a/b which provides level control to both output sockets and to the headphone amplifiers IC19/1 and 2. The headphone amplifiers themselves are based on the MFC4000B, a 250mW power stage by Motorola.

These units are quite suitable for driving direct into a pair of low impedance headphones (around 8 ohms) although it may be that a suitable listening level can only be obtained at the expense of reducing the setting of VR33 to a point where the signal at the output sockets is too small, i.e. substantially less than one volt. If this is the case the gain of the headphone amplifier may be reduced by increasing the value of R150. Doubling the value of this resistor will effectively halve the gain and thereby allow an increase in the input signal for comfortable listening.

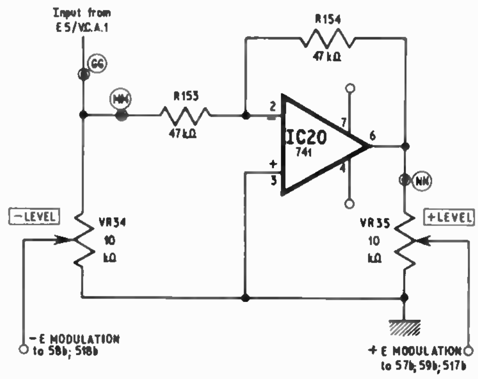


Fig. 19. Circuit of the control envelope inverter and modulation signal routing

CONTROL ENVELOPE INVERTER

In the original Minisonic the control envelope inverter provided a means whereby a negative going envelope could be used to modulate other voltage controlled circuits in the instrument. Although the positive going envelope was also available, it was uncontrolled and thus limited in its usefulness. In the Mark II instrument the same basic circuit is used with the addition of a level control at the input.

The circuit of the control envelope inverter is shown in Fig. 19. IC20 is a unity gain inverter driven by a positive going signal derived from Envelope Shaper 1. The positive envelope together with the negative envelope derived from the output of IC20 are both routed into the modulation control switching via VR34 and VR35 respectively.

COMPONENTS . . .

CONTROL ENVELOPE INVERTER

Resistors

R153–154	47kΩ (2 off)

All ±5% $\frac{1}{4}$W carbon

Potentiometers

VR34–35	10kΩ lin. (2 off)

Integrated Circuits

IC20	741

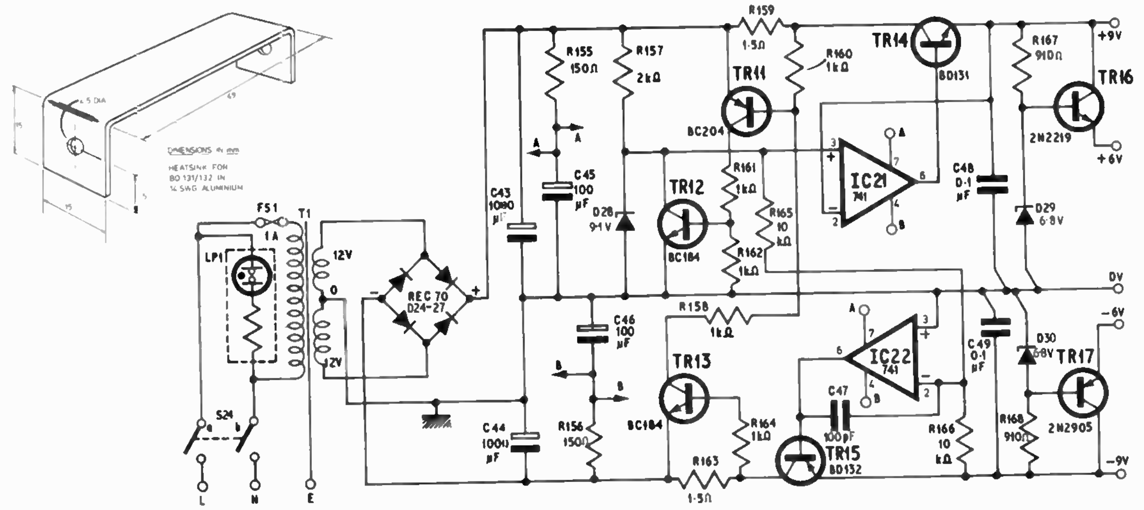


Fig. 20. P.s.u. and voltage reference supply

It should be noted that there is an apparent paradox in the naming of the control envelopes. The −E modulation is, in fact, the positive going envelope and vice versa. This nomenclature was adopted because it was felt more important to name the control envelopes in terms of their function rather than on their polarity.

For example, the +E modulation causes an increase in frequency of the v.c.o.s and expands the passband of the v.c.f. while the −E modulation has the opposite effect.

COMPONENTS . . .

POWER SUPPLY UNIT

(±9V p.s.u. and ±6V reference unit)

Resistors

R155–156	150Ω (2 off)	R163	1·5Ω
R157	2kΩ	R164	1kΩ
R158	1kΩ	R165–166	10kΩ (2 off)
R159	1·5Ω	R167–168	910Ω (2 off)
R160–162	1kΩ (3 off)		

All ±5% ¼W carbon

Capacitors

C43–44	1,000μF elect. 16V (2 off)
C45–46	100μF elect. 25V (2 off)
C47	100pF polystyrene
C48–49	0·1μF ceramic (2 off)

Transformer

T1–0–12V/0–12V/6VA mains transformer

Semiconductors

IC21–22	741 (2 off)	TR14	BD131*
TR11	BC204 or BC213/BC214	TR15	BD132*
		TR16	2N2219
TR12–13	BC184	TR17	2N2905
D24–27	REC70 bridge rectifier (RS)		
D28	9·1V 400mW Zener		
D29–30	6·8V 400mW Zener		

* mount in heatsink

Switch

S24 2 pole c/o main push button (Jean Renaud)

P.S.U. AND REFERENCE VOLTAGES

Both the p.s.u. and the reference voltage section of the original Minisonic exhibit a number of amendments. In the case of the negative rail regulator the 741 operational amplifier takes its d.c. feedback from the emitter of TR15 rather than from its own output. Additionally a 100pF capacitor provides an a.c. feedback loop around IC22 in order to limit the tendency towards oscillation on the negative rail. The current sensing resistors R159 and R163 have been reduced from 6R2 to 1R5 in order to extend the limiting range of the circuit.

In the voltage reference section emitter follower transistors are included in order that the current demands on the 6V rails may be more adequately met. Paradoxically this latter modification reduces the quiescent current drain by more than 30 per cent thereby making it more than ever feasible to drive the instrument from batteries.

Although, as originally, a dual secondary 12V transformer is specified, it is quite possible to utilise a dual secondary 9V type such as the Douglas MT235 CS. This is due to the fact that the mean current drain of the complete instrument is around 40mA and thus the loading on the transformer is such as to leave a margin of two to three volts across the series-pass transistors TR14 and TR15.

CIRCUIT

The complete circuit of the power supply unit and voltage reference supply is shown in Fig. 20. A dual secondary transformer and conventional arrangement of bridge rectifier and smoothing are employed. It would, of course, be possible to employ a 24V centre-tapped transformer in this arrangement but in no circumstances should these voltages be exceeded otherwise there is a danger that the operational amplifiers may become damaged. Transformers offering a total secondary voltage lying in the range 18V to 24V may be employed providing that their specified power rating is at least 6VA.

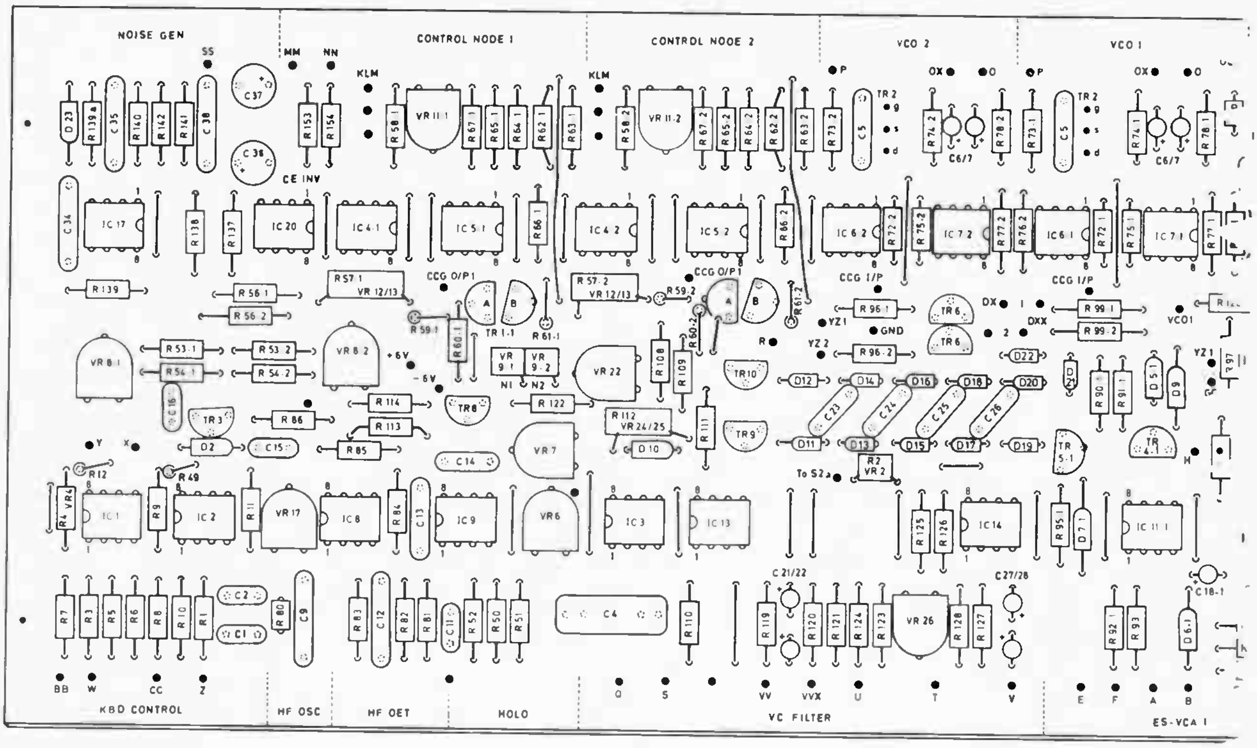
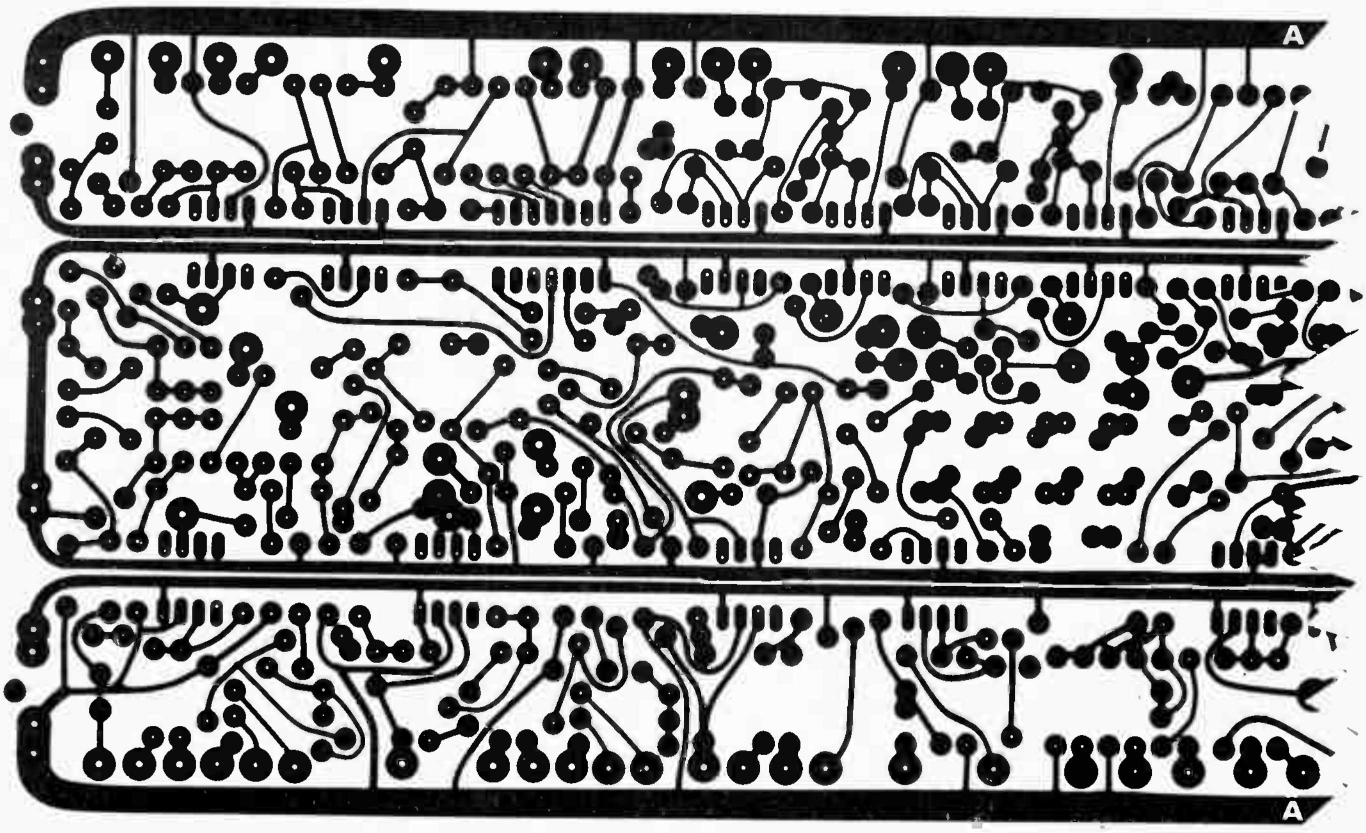
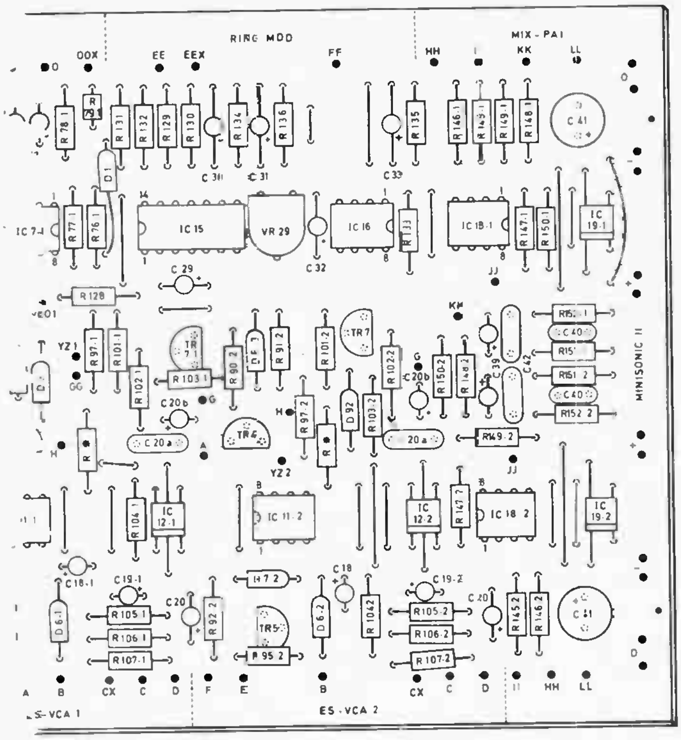


Fig. 21. Main circuit p.c.b. master (full size). Note that each section should be overlapped at "A". Also, all pad areas should be drilled for components





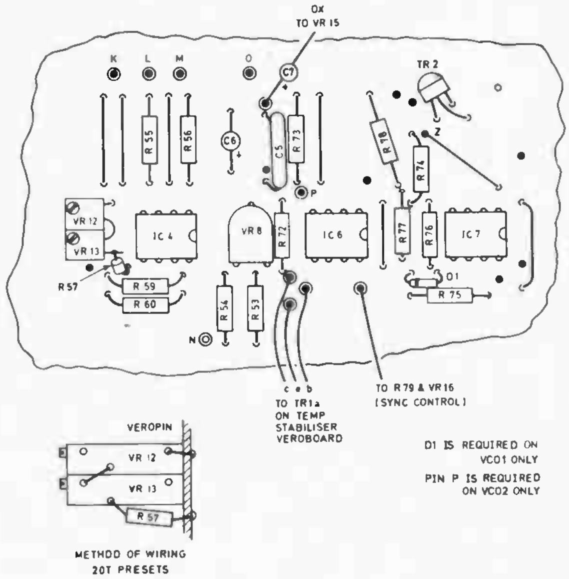
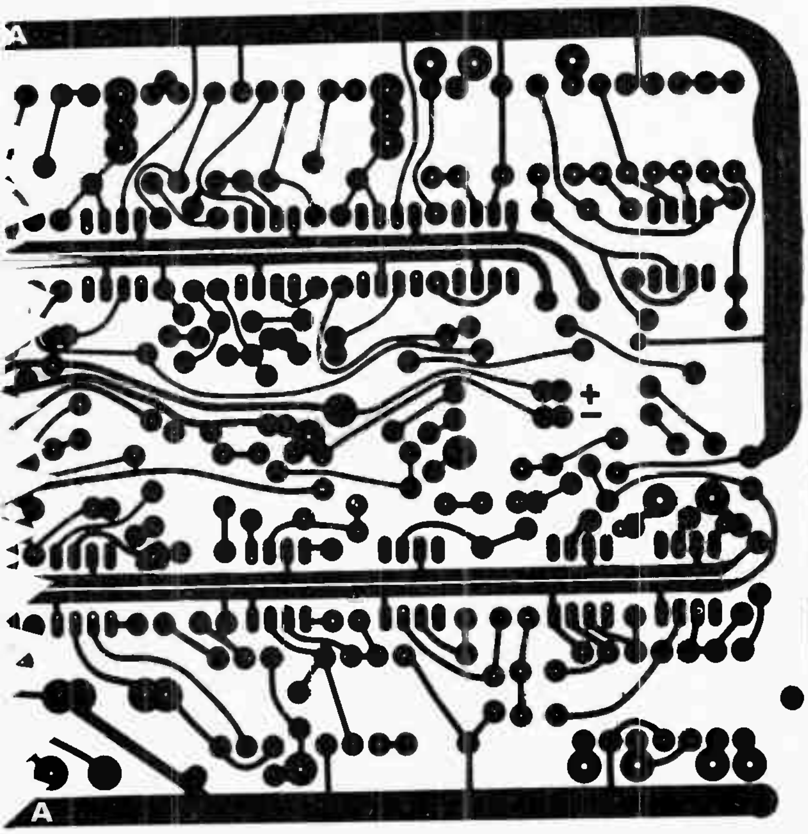


Fig. 23. Modifications required to the original Minisonic p.c.b.

Fig. 22. Component layout of main circuit p.c.b.



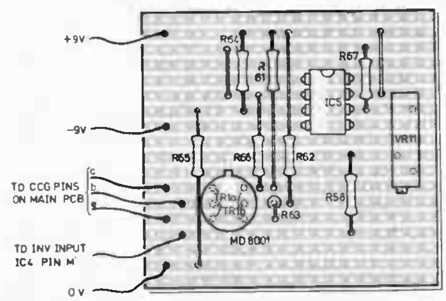


Fig. 24. Component layout for temperature stabilisation circuits for modified Minisonic

COMPONENTS . . .

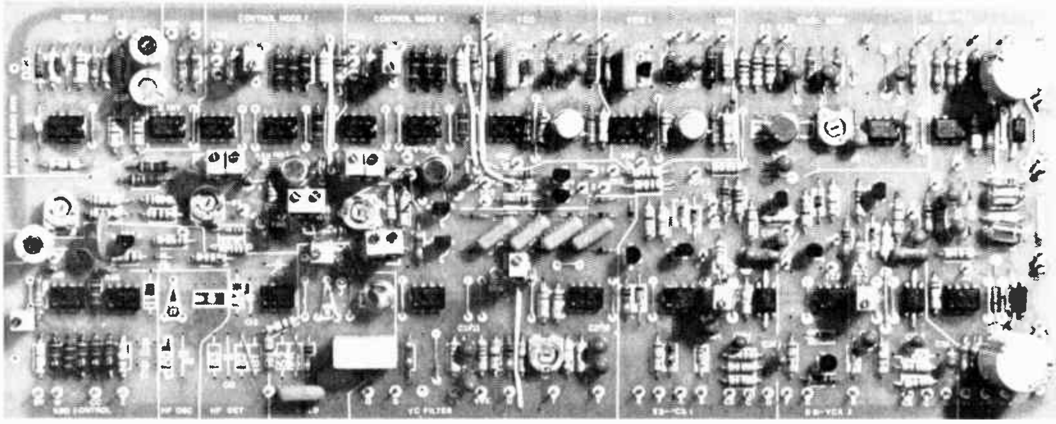
KEYBOARD

1 Pair of end cheeks
3 octave C-C keyboard
37 contact assemblies type GJ s.p.c.o.

(All items available from Kimber Allen)

MISCELLANEOUS

3 Panels (see figures), 23-2 pole changeover push button switches (Jean Renaud), 24 chrome push buttons, 21 aluminium knobs 18 mm dia., 3 aluminium knobs 22 mm dia., single switch mount, 10 way switch mount, 6 way switch mount, 10 turn analogue dials (2 off), 1-¼in. stereo socket (JK4), 2-¼in. mono sockets (JK2/JK3), 1-¼in. mono breakjack (JK1), miniature mains plug and socket (2 off), 4 metres of 0·4 sq. mm mains cable, 4BA × 6 mm screws, nuts and washers (16 off), 4BA × 12 mm self tapping screws (4 off), 4BA × 12·7 mm screws, nuts and washers (4 off), 6BA × 6mm countersunk screws, nuts and washers (4 off), rubber feet (4 off)



Assembled main p.c.b. except for h.f. oscillator and detector areas

The reference voltage for the power supply is derived from a Zener diode which provides the input signal to IC21 and IC22. IC21 operates as a follower and drives the positive series pass transistor TR14 with overall feedback being taken from the emitter of this device.

IC22 is connected as an inverter and drives the negative series pass transistor TR15 with a similar feedback arrangement as for the other rail. In this latter case, however, it is necessary to provide some a.c. feedback around IC22 in order to restrict the tendency towards oscillation on the negative rail. Current limiting is provided by means of the arrangement comprising TR11, TR12 and TR13. The operation of the current limiting circuit is as follows: In the case of the positive rail, normal current demands result in a minimal potential difference across R159 and thus the base of TR11 is almost at the same potential as its emitter. TR11 is therefore not conducting. As the current drain increases, however, the p.d. across R159 also increases to the point at which TR11 starts to conduct. In these circumstances TR12 is progressively turned on and reduces the reference voltage by shorting out the diode across it.

Similarly, in the negative rail TR13 provides current limit control. In this case, however, when TR13 begins to conduct it has to turn on TR11 before affecting the reference voltage in the way previously described. With the values given the circuit will pass currents of up to 200mA before limiting occurs.

REGULATION

Within the current limits quoted, short term regulation is carried out by means of the operational amplifiers IC21 and IC22. A sudden increase in the current demand on the positive rail, for example, will tend to cause the voltage to fall. This is reflected at IC21 in terms of reduced feedback and thus the output will drive positive in order to restore the situation. The reverse situation will occur when there is a reduction in the current demand.

The 6V reference rails are derived from R167, D25 and TR16 for the positive 6V rail and from R168, D26 and TR17 for the negative rail. The actual value of the reference voltages will depend on the absolute value of the Zener voltages and it is not necessary that they are maintained at exactly 6 volts. The main thing is that these rails should be as stable as possible and that the transistors TR16 and TR17 are able to meet exceptional current demands such as during periods when the v.c.f. is being driven hard.

If it is found that, in such latter circumstances, the reference rails show a drop in voltage—particularly the negative rail—then it will be necessary to make a token reduction in the appropriate Zener series resistor in order to provide more drive to the transistor.

PRINTED CIRCUIT BOARDS

It is appreciated that there will be a number of constructors who, having built the original Minisonic, will want to consider the possibility of converting their instruments into the Mark II version. Such an operation is quite feasible, indeed there are quite a number of Minisonic IIs in circulation in which the majority of the modified circuits are incorporated into the original circuit board.

Details of p.c.b. modifications are therefore included here but are limited to the major circuit amendments such as the oscillators. To contain the hold isolator and modifications for the original p.s.u. p.c.b. it is probably simpler to make up the board as in Figs. 25/6. Most of the other modifications require only the addition or deletion of the odd component and it is felt that most constructors having got this far will be perfectly able to carry out the necessary work without special instructions.

Fig. 21 shows the copper side of the main circuit board devised for the Mark II Minisonic, while the component layout for the same board is given in Fig. 22.

Fig. 23 gives the modifications required to the original Minisonic p.c.b. in order that it can carry the re-designed v.c.o.s. The layout of this particular board

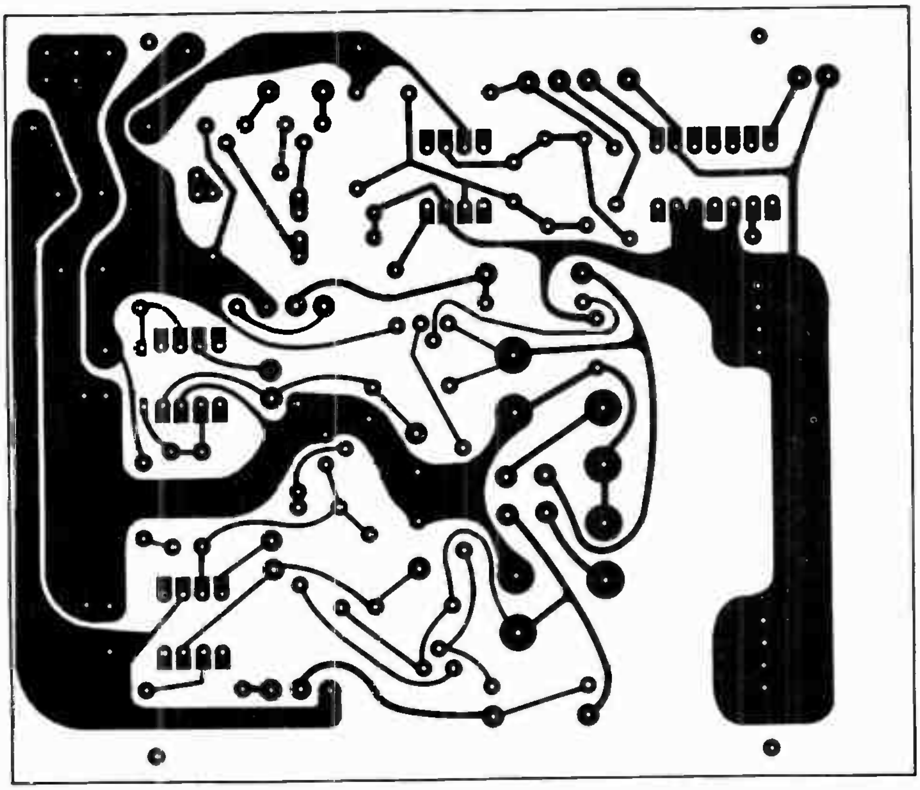


Fig. 25. Mark II p.s.u. p.c.b. incorporating reference supply and hold isolator

is not sufficiently compact to enable inclusion of the independent temperature stabilisation circuits for the new oscillators, so these are shown on a Veroboard layout in Fig. 24.

This small additional Veroboard can be mounted adjacent to the main p.c.b. in the finished instrument. In the prototype Minisonic II the additional board was divided and mounted directly over the comparator stage of the v.c.o.s on short pillars.

The revised power supply p.c.b. for the Minisonic II is shown in Fig. 25 for the copper side and in Fig. 26 for the component layout. This particular circuit board carries the power supply unit, the reference voltage supply and the additional circuit known as the hold isolator.

INSTRUMENT HOUSING

A general idea of the construction of the Minisonic housing is given by the accompanying photographs.

The recommended 3-octave keyboard for the Minisonic is supplied by Kimber-Allen Ltd. The exceptionally rigid aluminium extrusion comprising the baseframe of the keyboard acts as a bridge between the two end cheeks of the instrument which are detailed in Fig. 27. Further rigidity is imparted to the structure by means of the base panel which is permanently screwed to the aluminium brackets fixed to the end cheeks. The front panel of the instrument is screwed to the sloping aluminium brackets while the rear panel is hinged to the front panel and secured to the base by means of four self-tapping screws.

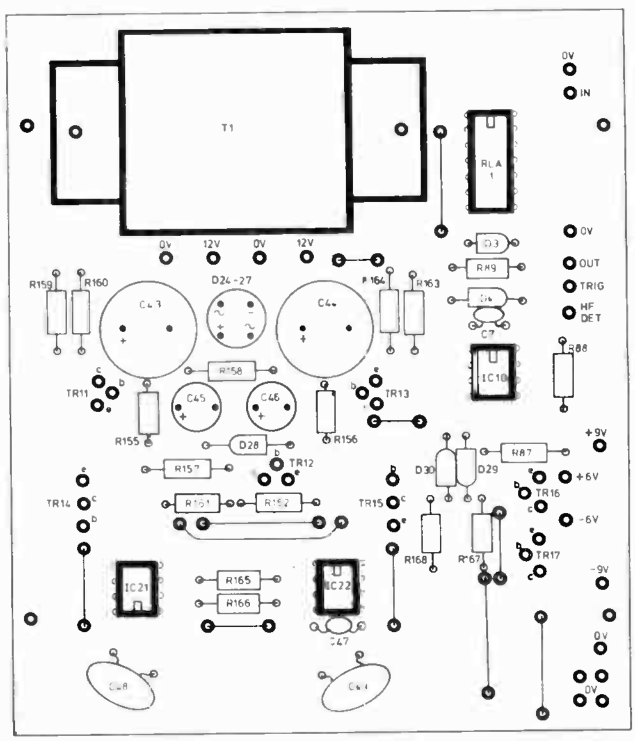


Fig. 26. P.s.u. p.c.b. component layout

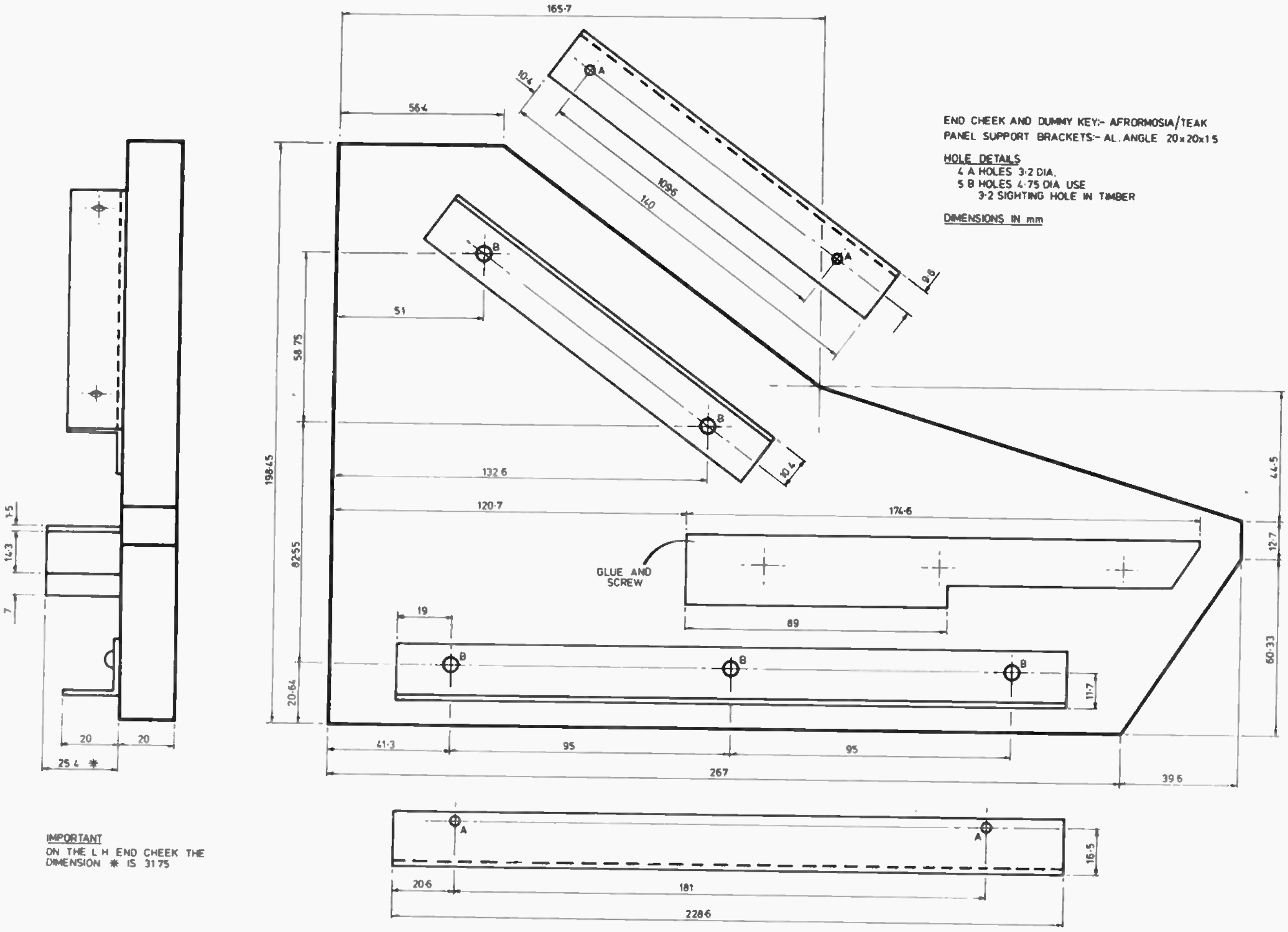


Fig. 27. Detail of right-hand end cheek for keyboard. The left-hand cheek is a mirror image of this

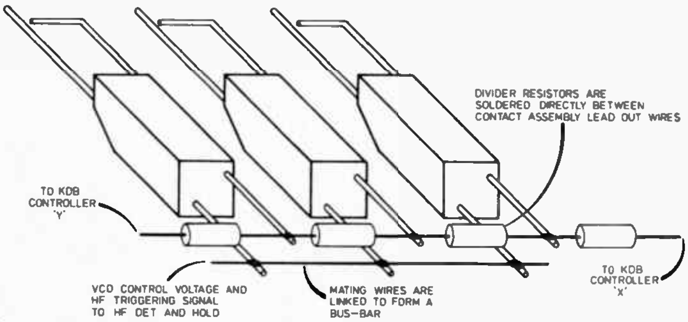


Fig. 28. The wiring of the single pole contact assemblies in a conventional keyboard

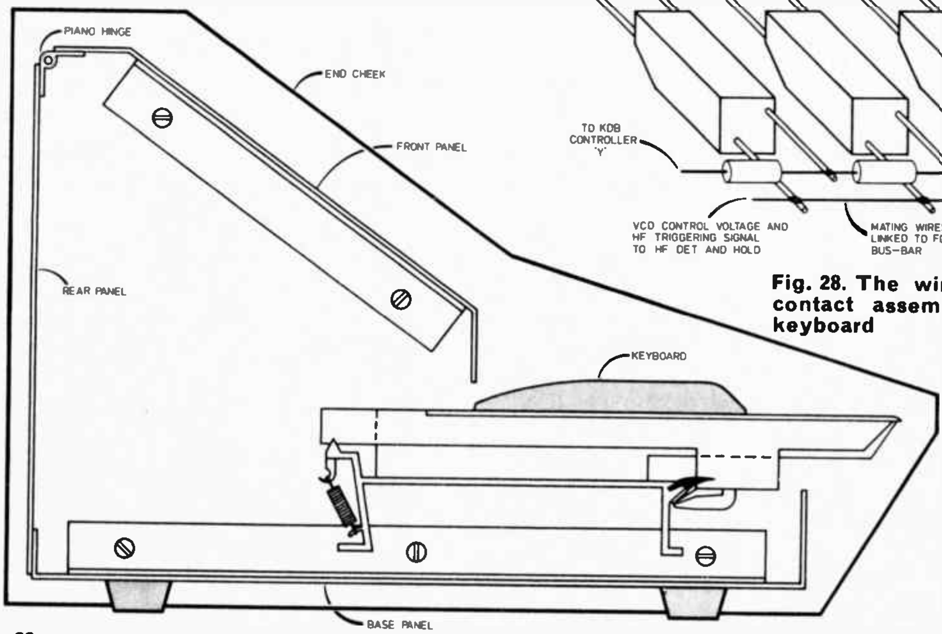
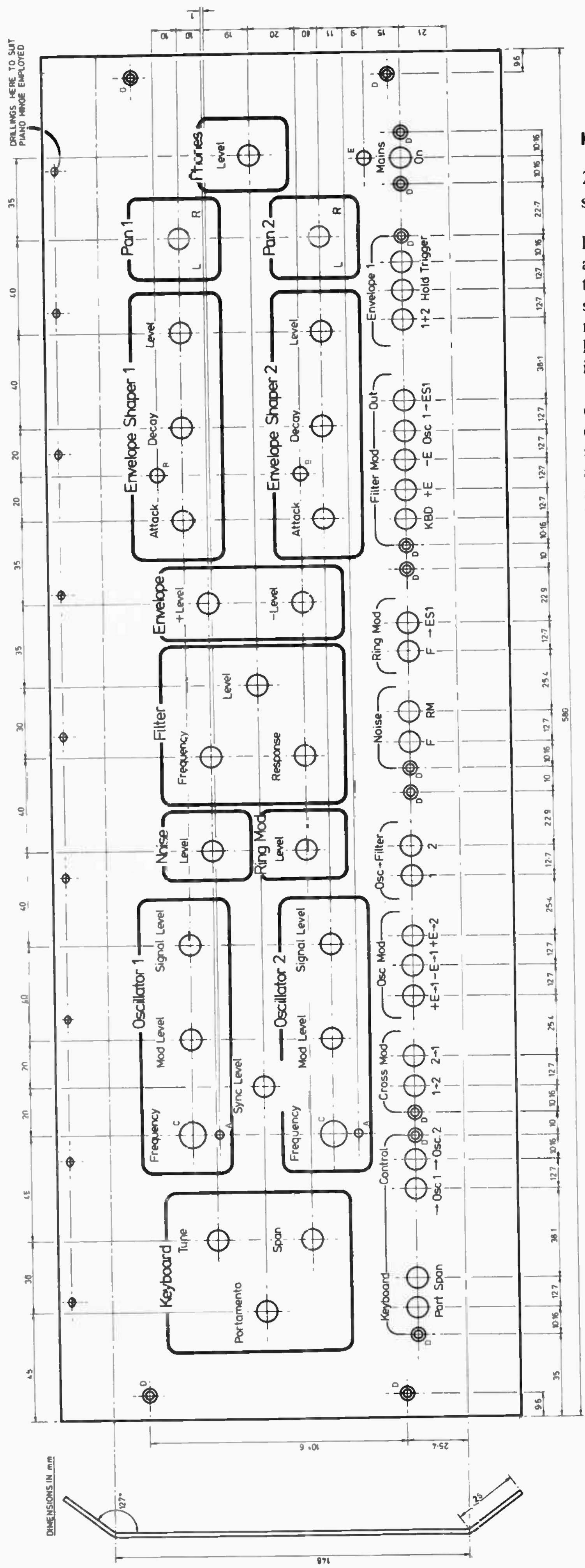


Fig. 29. Section through Minisonic housing with panels and keyboard in position



KEYBOARD PREPARATION

After making up the end cheeks as detailed in Fig. 27 the first stage in the assembly of the housing should be the preparation of the keyboard.

The keyboard baseframe is screwed to the 89mm long thickened portion of the dummy keys. To ensure a flush fit it will be necessary to remove that part of the felting which extends to the end of the upper surface of the keyboard baseframe. It will also be necessary to file away approx. $\frac{3}{32}$in (2·5mm) from the lower edge corners of the baseframe extrusion so that it clears the upper edge of the aluminium angles.

When this has been done the keyboard can be offered to each end cheek in turn and aligned with the dummy keys in preparation for drilling the securing screw holes. Two 3mm holes should be drilled approximately along the centre line of the dummy key and about 65mm apart. The holes should go through the baseframe and extend to a depth of about 15mm into the dummy key.

When both end cheeks have been similarly drilled the holes in the keyboard baseframe should be enlarged to accommodate No. $6\times\frac{3}{4}$" woodscrews which are used to secure the baseframe to the end cheeks.

During assembly of the keyboard to the end cheeks ensure that the baseframe closely abuts the inner face of the cheeks. If this is not done there will be unsightly gaps when the other panels are applied.

CONTACT ASSEMBLY

With the keyboard mounted between the end cheeks the assembly of the keyboard contacts can be started. Kimber-Allen single pole change-over contacts of the G.J. type are recommended and these should be mounted on a strip of 2mm × 50mm hardboard. The hardboard should be cut to extend approximately 12mm either side of the keyswitch actuators and initially placed loosely in position parallel to and about 15mm behind the line of the actuators. The contact assemblies are now placed on the hardboard strip such that their moving wire is centrally over the actuator and glued into position using Araldite Rapid.

When all assemblies have been positioned and set the whole hardboard strip may now be glued to the keyboard baseframe such that the moving wires are central to and extend beyond the actuators by approximately 5mm. The keyboard divider resistors are now fitted to the contact assemblies generally as shown in Fig. 28 with the mating wires connected as a bus-bar. No further wiring of the keyboard is required at this stage.

HOLE DETAILS
2 A HOLES 3·2DIA
2 B HOLES 5·5DIA
2 C HOLES 11·2DIA
14 D HOLES 3DIA AND CSK AT 90°
1 E HOLE 6DIA
ALL OTHER HOLES 9DIA (48 OFF)

Fig. 30. Front panel details including control legends

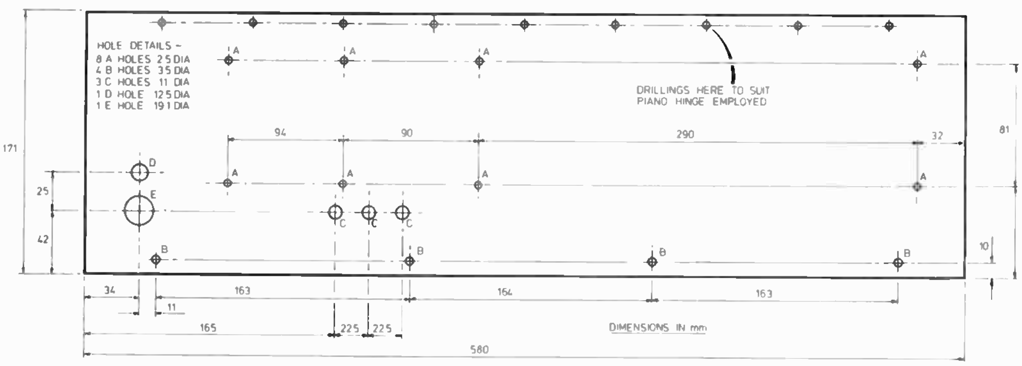


Fig. 31. Rear panel detail

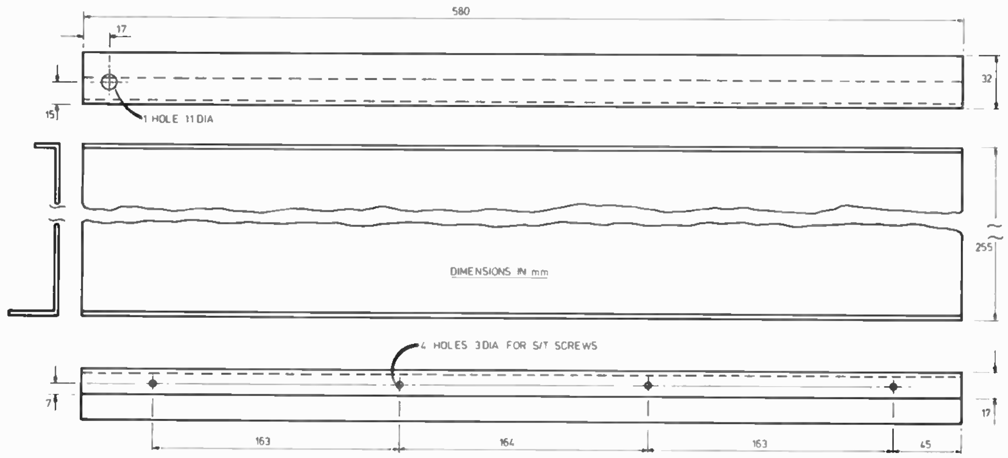


Fig. 32. Base panel detail

PANEL WORK

Bending and assembly details for all panels are shown in Fig. 29 while drilling and cutting details for front, rear and base panels are given in Figs. 30, 31 and 32 respectively. Having prepared and finished the panels as specified the assembly of components into the front and rear panels can be started.

In the case of the front panel all potentiometers and switches should be mounted in position and the switches wired as shown in the photograph. At the same time all those potentiometers having a connection to the 0V rail can be linked with a stout copper wire. The only one of these which is not absolutely straightforward is the pan-pot. and a wiring diagram for this device is shown in Fig. 33.

When component assembly and initial wiring on the panels has been completed the front and rear panels may be joined together by means of the piano hinge and the assembled—and tested—circuit boards mounted to the rear panel on 12mm standoff spacers. The main wiring may now be carried out with the panels opened out and laid flat.

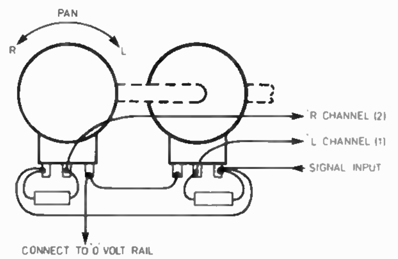


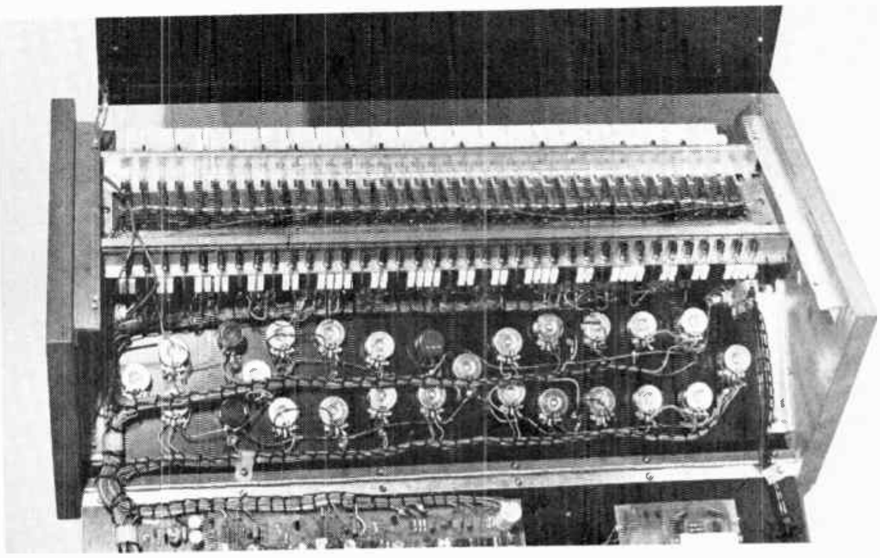
Fig. 33. Pan pot wiring

WIRING HARNESS

It is strongly recommended that constructors take the trouble to make up a properly formed wiring harness for the job of interconnecting all the various circuits and controls in the instrument. The extra effort is adequately rewarded in terms of neatness of the finished layout and in the ease of servicing when this requires to be carried out.

For ease of making up a wiring harness diagram should be drawn in full scale on to a sheet of squared paper. With the drawing then pinned out on a piece of

Prototype Minisonic shown inverted and opened up. Note the wiring harness layout.



board two parallel rows of ordinary nails should be laid along each main route of the wire runs. The channels so formed will retain the wire runs as the harness is assembled.

At each point where a wire has to exit the harness place another nail in line with the point to which the wire is to be routed and, finally, yet another nail at the termination point of each wire run. Additional nails may be required at various points to retain groups of wires exitting the main harness at different places.

When all wire leads have been laid down the next step is to lace them up into a tight harness. The half knots on each terminal nail should now be untied and the wires trimmed to length, stripped ready for soldering and, in the case of the screened wires, pre-tinned. On the screened wires the insulation and lap screening should be cleared back about 15mm from the tip of the signal wire.

LAYING THE HARNESS

The harness may now be lifted from its channels and laid on to the opened out panels carrying the circuit boards and controls ready for soldering up.

When the main harness has been soldered into position the panels are lifted on to the instrument and the front panel secured to its aluminium support brackets. The connections to the keyboard divider chain and busbar are made at this stage. Finally, with the instrument on its side—resting on one end cheek—the leads to the headphone socket are connected and the base panel secured into position.

Having connected the power supply and mains leads the instrument is now ready for final lining up and testing.

FINAL CHECK OUT

If the testing and setting up procedures outlined in the circuit description have been carefully followed then the final check out will be a straightforward verification of circuit performance. Where batteries have been used for the initial testing rather more care will be required since the operating voltages will undoubtedly be slightly different and reference voltages such as that set by VR9 in the v.c.o. control nodes will perhaps require some adjustment.

The principal points requiring attention are as follows:

1. Check that the voltage appearing across VR10 in the v.c.o. control nodes is precisely 5 volts.
2. Depress S22 (you are switching the ES/VCA's "on").
3. Check that the "law" on each oscillator is 500mV/octave. Note that this is a nominal figure only and that, providing that the law of the oscillators is precisely the same, some variation from the nominal value can be tolerated.

   The law verification should be carried out using the manual frequency controls (VR10). If the law does vary from the nominal value VR9 should be re-adjusted so that the voltage across VR10 is equal to ten-times the law voltage. This will ensure that each full turn of VR10 will shift the oscillator frequency by one octave.
4. Check that, for a given voltage on the wiper of VR10 in both oscillators, the frequencies are within a few Hertz of one another. This check should be made at a number of points over the full swing of VR10.
5. Connect a voltmeter set to 2·5V range across pins "X" and "Y" on the p.c.b. positive lead to pin "Y". Set VR5, the "Tune" control, to approximately mid-position and with S2 in the out, or open, position adjust VR2, the fixed "Span" preset so that the voltmeter reads precisely three-times the law voltage. If the law is set to 500mV/octave then the meter should read 1·5V.

The completed Minisonic 2 showing control panel and keyboard

With one oscillator only being monitored verify that the keyboard plays in an equal temperament scale. Adjust VR2 as necessary to achieve an equal temperament scale.

6. Set both oscillators to exactly the same frequency by means of the manual frequency controls. Press any one key on the keyboard and note whether both oscillators remain in unison. Repeat at various points over the span of the keyboard.

   If any marked departure from unison is noted this will more than likely be due to some variation in the values of R68/1 and 2. Either one of these resistors can therefore be removed from its switch and replaced with a 50K preset which should be adjusted until unison is achieved.

7. With channel 1 panning control hard anticlockwise and channel 2 panning control hard clockwise release the "Hold" switch (S22). Set the v.c.a. level controls to maximum and headphone level control to maximum. Check that there is no signal breakthrough in either channel with the oscillators running at a convenient mid-range frequency. If breakthrough is present in either channel adjust the appropriate VR20 to minimise the effect.

SIGNAL AND CONTROL ROUTINGS

This completes the most important aspects of final setting-up. The functions of the remaining circuits can be checked out with reference to the signal routing diagram (Fig. 10) and the details given in their respective descriptions.

Similarly the modulation routings should be individually checked with reference to the appropriate sections and figures.

Having built the Minisonic II it now remains for the constructor to gain an understanding of the instrument's various controls and of the combinations of signal and control routings which give rise to particular forms of sound.

MINISONIC 2 POSTSCRIPT

In Fig. 1 R1 and R7 are changed to 10kΩ and 8·2kΩ respectively. VR5 should be a 10kΩ linear carbon potentiometer. L1 may be omitted, particularly when dual contact keyswitches are employed.

Confusion has arisen as a result of Fig. 1 showing two connections from the single pole keyswitches. These are marked "To H.F. Detector (C11)" and "To H.F. Detector (Stylus)". On the p.c.b. these are, in fact, the same point. The purpose of showing the two connections was an aid to those constructors who were converting the original Minisonic to the Mark 2 version. It is suggested that the upper marking (H.F. Detector—C11) be substituted for the lower and the upper arrow deleted.

Some constructors have experienced difficulty in setting up the temperature compensation the main problem being related to the biasing of TR1/b (Fig. 3). It is suggested that R63 is soldered into the p.c.b. on long leads, with the circuit assembled, the output of IC5 is measured. This should be in the range 0·5V–0·75V. If the output is substantially less than 0·5V, R63 should be increased in the value—by easy stages—until the required output is achieved.

Testing the control node should be carried out in accordance with the details given. Note however that the ideal situation is when adjustment of VR11 will change the control characteristic, during the finger test, from a negative one (oscillating frequency increasing) to a positive (oscillator frequency decreasing).

In Fig. 8 R98, R100 and D8 can be omitted since l.e.d. indication of envelope state is adequate.

In Fig. 11 D10 should be shown spanning all three resistors in the feedback line.

Note that the MFC6040 used in the v.c.a. is now replaced by the MCP3340 a device having similar characteristics but in an eight pin d.i.l. package.

The simplest way to accommodate them is on a small piece of Veroboard supported above the main p.c.b. by stiff wire links connected to the appropriate drilling for the MFC6040. ★

PE JOANNA

By A. J. BOOTHMAN B.Sc.

A new generation electronic piano with an exciting specification

- **High degree of piano tone simulation**
- **Fully touch sensitive keyboard**
- **Alternative voicing of honky-tonk and harpsichord**
- **Simple single control tuning**
- **Soft and sustain pedals**
- **Lightweight, attractive styling with amplifier incorporated in stool**

SPECIFICATION . . .

MUSICAL COMPASS
Five Octaves C to C 61 Notes

FREQUENCY COMPASS
Fundamental Frequency Range 60Hz to 2kHz approx
Master Oscillator 500kHz approx

NOMINAL OUTPUT LEVELS
External Amplifier 1V into 1MΩ
400mV into 10kΩ
100mV into 2kΩ
Headphones 50mW into 4Ω

TOUCH CHARACTERISTICS
Dynamic Range >27dB
Key Action <1oz

MAINS INPUT
240 Volts 50 Watts

SOUND ENVELOPE (nominal times)
Touch Range 2 to 40mS
Attack Period <1mS
Early Decay 500mS
Sustain Period 3 to 5S (varies over compass)

TREMOLO FREQUENCY (nominal)
Slow 5Hz
Fast 10Hz

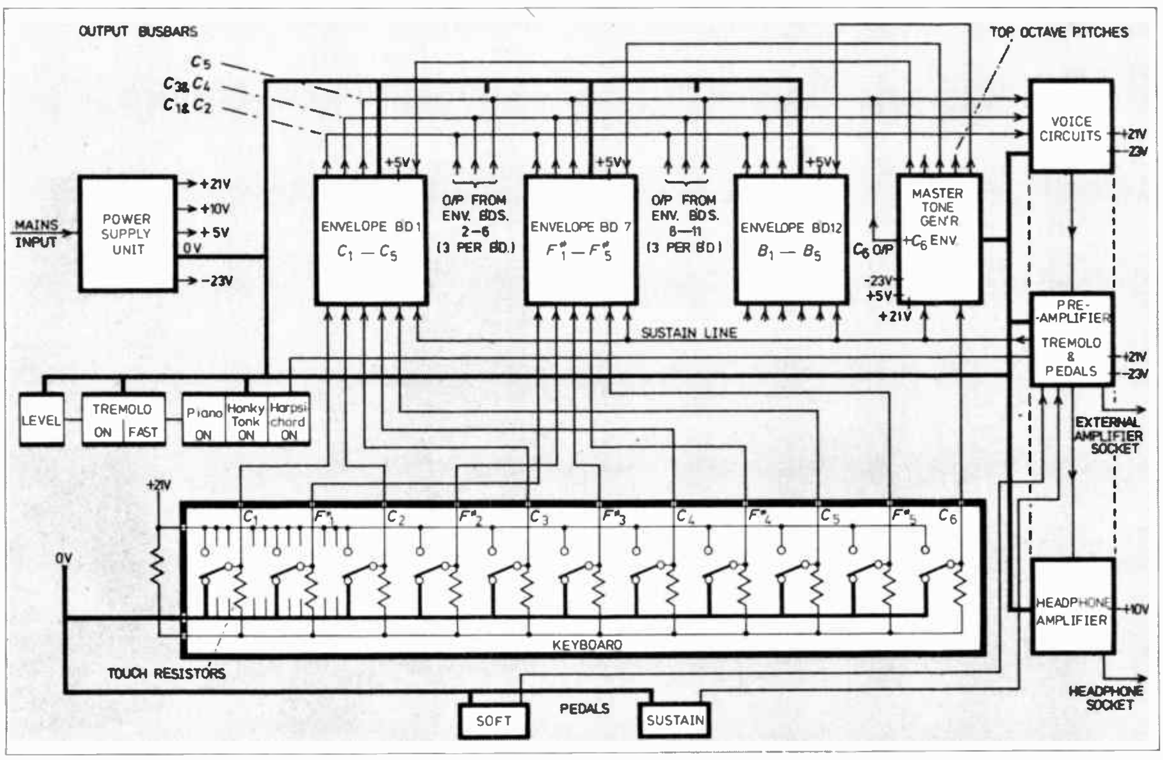
DIMENSIONS AND WEIGHT
Case 38in × 14in × 5in
Height of legs 24in
Weight 20lb

CONTROLS
Level Control/On-Off Switch
Piano Tab
Honky-Tonk Tab
Harpsichord Tab
Tremolo Slow Tab
Tremolo Fast Tab
Sustain Foot Pedal
Soft Foot Pedal

SOCKETS
Mains Supply Socket
Pedal Socket
External Amplifier Socket
Headphone Socket
Stool Amplifier

POWER
Output Power 20–50 Watts

Fig. 1. Block diagram of the "Joanna"



Since the first publication of an electronic piano design, which was in *Practical Electronics* over the period September 1972 to January 1973, the instrument has been the subject of a sharp increase in popularity. A number of commercial versions have become available on the market, whilst many organ manufacturers now include a piano stop on a large part of their range.

Many of the types on offer are not touch sensitive, which, although being an advantage when used in conjunction with an organ, is a distinct drawback when a full piano substitute is required. Other designs have questionable tone or voicing which can soon become unacceptable.

The P.E. Electronic Piano had a very small degree of touch sensitivity and a basic functional tone, and has now grown into the "Joanna" which combines a very wide range of touch with a good piano tone, whilst achieving a further decrease in size. In addition to its ideal adoption as the family instrument this therefore makes it very convenient to sit on top of an organ as an extra manual with or without touch sensitivity.

In establishing the new design the standard piano was investigated in more detail and the following facts and comments will be of interest to the reader.

PIANO HISTORY

According to past literature on the subject the first stage in the evolution of the piano was the clavichord, which used the impact of metal blades on horizontally stretched wires. The other notable predecessor was the harpsichord in which a mechanically coupled plectrum was used to pluck the strings. Instruments of the latter type can still be obtained, and in the "Joanna" a harpsichord tabswitch is provided.

The pianoforte, in which the wire is struck by a felt covered wooden hammer which is allowed to quickly fall back, has evolved into two basic types, namely the grand and the upright. The latter, which is of course the most common type, covers a wide range of tonal qualities, many of which have their own individual appeal. Apart from the rich full piano sound obtainable from a high quality instrument, the other most characteristic variation, which is usually associated with age, is the honky-tonk. The "Joanna" provides a honky-tonk tabswitch, and by using all three tabs in combination a wide range of tone is available to the performer

COMPASS AND SIZE

A normal piano covers 7¼ octaves, with some smaller versions having a limited availability. The stringing of the piano is necessarily complex and takes up a considerable amount of space, fold-over techniques being used to minimise the volume where possible. Limiting the compass to five octaves greatly assists in reducing the size of the electronic instrument to minute proportions when compared with the conventional instrument, and is convenient from the electronic component point of view in simple economic tone generation.

Bulk Component List

To take advantage of any concessions offered by retailers for bulk purchases the following quantity list is included. (Quantity in italics.)

Individual detailed component lists appear in the text.

Integrated Circuits

ZN7404	*2*
ZN7472	*1*
ZN7493	*12*
μA741 (8-pin d.i.l.)	*5*
MFC6040	*1*
MFC4000	*1*
μA7805	*1*
AY-1-0212	*1*

Transistors

ZTX108	*77*
ZTX501	*1*

Diodes

Cheap silicon diodes 20 volts *247*
(e.g. 1S44, OA200 types)

Capacitors

4·7μF Electrolytic 25Vmin	*61*
4·7μF Electrolytic 12Vmin	*61*

Resistors

56Ω	*61*	120kΩ	*122*	270kΩ	*12*
2·2kΩ	*12*	150kΩ	*50*	330kΩ	*12*
33kΩ	*61*	180kΩ	*24*		
47kΩ	*170*	220kΩ	*26*		

All ⅓ watt carbon 5% tolerance

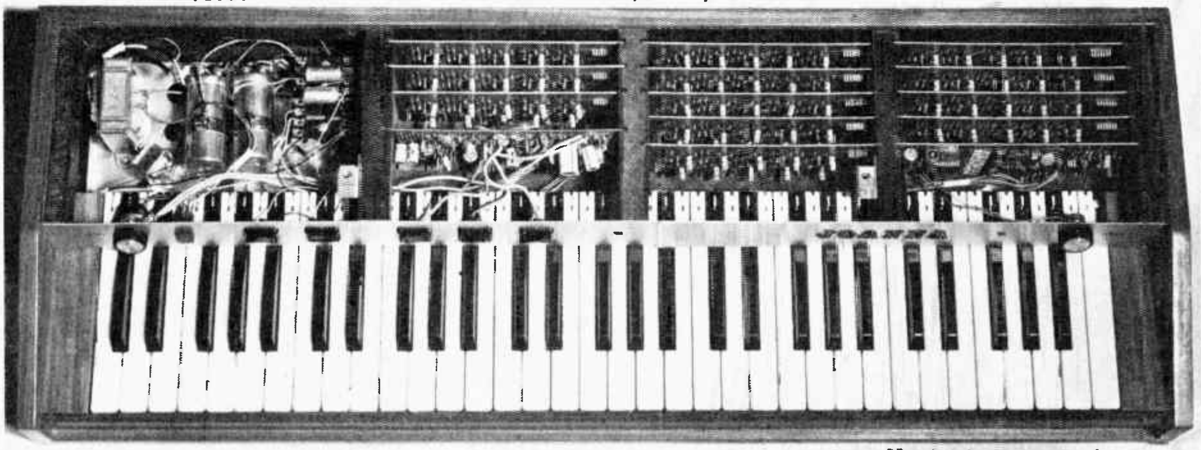
The above list includes all semiconductors, 80% of the resistors, and approximately 50% of the capacitors

The choice of compass is an individual matter, but the author now prefers to use C-C in the general home environment, whereas F-F has been used for group work. In order to give some compensation for the loss of bass notes it has been simple to arrange for a slight enhancement of the level of the bottom two octaves, and the result has been assessed to be a good compromise by a number of pianists with widely differing styles.

TOUCH ACTION

The keyboard dynamic range of a piano is in the range 50–60dB, but this level of touch sensitivity is not often available in the average instrument, and the 30dB or so offered by the "Joanna" should be pleasing to most pianists. The principle of touch action is the subject of much discussion but suffice it to say that the majority opinion is that the final hammer velocity is the only determining factor in the resulting sound. In the conventional piano the velocity defines the level of output and the tonal quality, and the means by which this velocity is achieved, e.g. a gradual build up of key speed, effects neither parameter.

In the "Joanna" the level of output is varied by the average speed of the key over the depression period,



Interior of piano showing location of sub-assemblies

and the tone is not altered, other than through the normal effect in the ear of the listener where perceived tone varies with level. The level control is used to limit the maximum volume achievable to suit the environment and is not operated during the execution of a musical score. The final aspect of touch is the key-weight which is considerably less than the normal 2oz. This gives a very fast action, but does of course feel different to a normal piano keyboard, and requires time to become adjusted.

ENVELOPE

Many resonance effects are present in a piano which result in a complex decay portion of the envelope, both in terms of level and harmonic content. A large influencing factor is the use of multiple stringing and the way in which sympathetic resonances can be initiated by the note being struck.

Simulation of these effects could be achieved by extensive electronic synthesis, but to produce a polyphonic instrument by such techniques is outside the economic scope of the large area of interested constructors who are also prepared to accept that a full detailed synthesis of a piano is not of the utmost importance, whilst an improvement on presently available designs is welcome. The "Joanna" offers a decay characteristic which varies in length across the compass, and is in the region of five to three seconds.

VOICING

Two tonal aspects are mentioned above where simulation is not attempted, i.e. variation of the harmonics content with key velocity and over the decay period. The harmonic content of the basic note differs over the range of the board, and the voicing circuits are designed to give a greater level of high harmonics at the lower end of the piano to match the conventional instrument. The voicing circuits allow the constructor some freedom of fine adjustment to suit his own taste, and are split into three groupings across the keyboard.

PEDAL ACTION

The soft pedal of a grand piano operates the hammer action mechanism to move it into a position whereby the number of strings hit on the multi string sections is reduced, whereas the upright system is to reduce the hammer travel and thus its terminal velocity. The "Joanna" by simultaneously reducing the output level from all notes is similar to the upright action. The sustain pedal raises all dampers, and is directly simulated in the electronic version.

ENVIRONMENTAL ACOUSTICS

As with all sound sources, the acoustics of the environment can have a very great effect on the overall result obtained from a piano. The first noticeable effect is the difference in tone experienced at varying distances from the sound source which occurs with both the conventional and electronic piano. The second is the way in which the size of the room influences the bass response of the instrument.

Reverberation can enhance the piano tone considerably as would be experienced by comparing the result on a high quality piano in the average living room, with the same instrument located in a concert hall. The "Joanna" can easily be fed into a spring line reverberation unit, which if used to a very small degree can give a pleasing sound in a small room. Headphones can be plugged into the instrument for silent practice, or to create your own sound stage without any influence from the room environment.

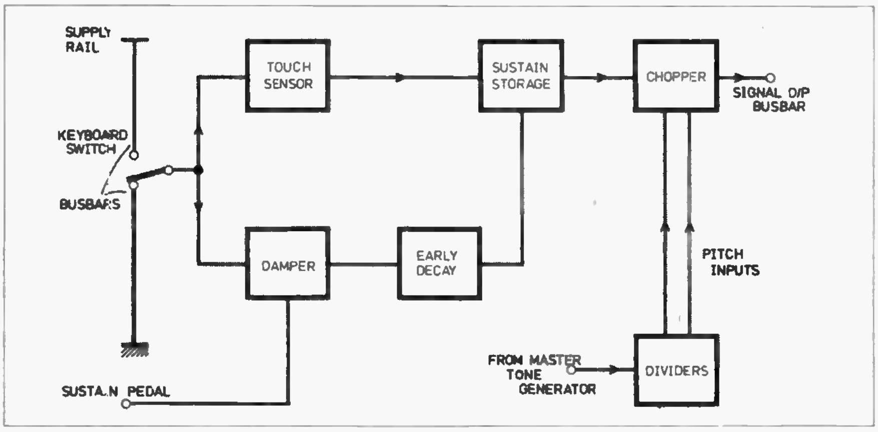


Fig. 2. Block diagram of envelope shapers. 61 are required

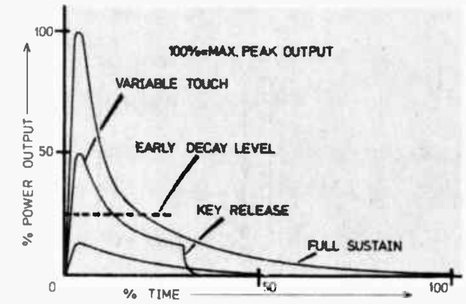


Fig. 3. Envelope Shapes

TUNING

The conventional piano requires tuning at least once a year, and is often left considerably longer. The use of the master tone generator in the "Joanna" ensures that the relative tuning of all notes is extremely accurate and not subject to variation. Compared with some two hundred tuning adjustments on the normal piano, the "Joanna" has only one.

COST

As stated earlier it is not beyond the reach of electronics to completely synthesise the action and sound of a piano, but the value of attempting such a project is extremely questionable. We are not going to see a concert pianist changing to an electronic piano for major performances and the project should be judged against the background of the average domestic upright, taking up considerable room space and having a present day cost of at least £400. For an outlay of approximately £120, depending on the amount of effort which the constructor wishes to apply, an attractive practical instrument can be provided for the use of all the family which will give many hours of pleasure.

OVERALL SYSTEM

The block diagram Fig. 1 shows the way in which the various sub-units are interconnected to make up the "Joanna" and from the diagram the overall principles of operation can be understood.

The keyswitch assembly, linked to the five octave keyboard, consists of 61 single-pole changeover switches, operating between ground potential when at rest and a rail voltage of approximately 20 volts when the key is depressed. Touch resistors are mounted on the keyswitch assembly as indicated.

The switch outputs are routed to 12 boards, each containing five envelope circuits, such that all octaves of one semitone go to one board. Three outputs are taken from the Envelope Boards, grouping the bottom two octaves, the middle two octaves and the top octave, and lead to the Preamplifier Board. This contains separate voice filters for each of the groupings mentioned, feeding to a preamplifier the gain of which is determined by a voltage controlled amplifier in the feedback loop. The latter has both tremolo and soft pedal control inputs. A direct output is taken from the preamplifier for use with an external power amplifier and a low power output stage is included to drive headphones when required. The Preamplifier Board also contains a sustain pedal circuit, the output of which is linked to the Envelope Boards.

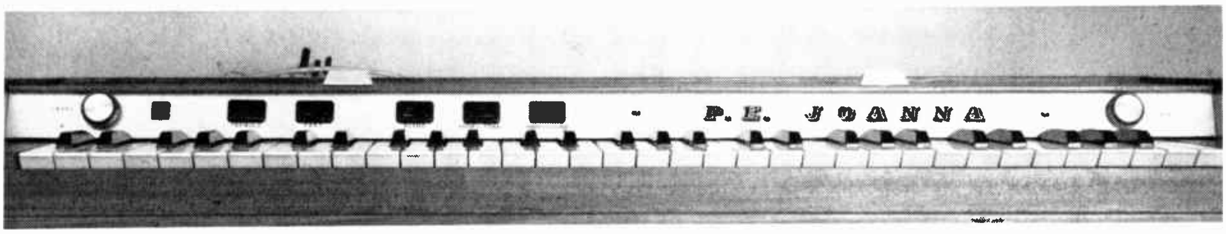
Generation of the 12 semitones in the top octave is carried out by one integrated circuit fed by a 500kHz input frequency. This is mounted on the Master Tone Board which also carries the envelope circuit for the top note of the piano. The dividers for the lower four octaves are included on the Envelope Boards.

The voice switching is performed between the voice circuits and the preamplifier. The use of a single tab gives the pure sound for that instrument, whilst more than one tab can be used simultaneously in order to provide tonal variations to suit the performer. The tremolo switching offers two speeds of tremolo as required.

PIANO ACTION

The fundamental piano action is obtained from the envelope shapers which are shown in block form in Fig. 2.

The time taken for the keyswitch to travel between the two busbars is converted by the touch sensor into a reducing voltage which is passed onto the sustain storage circuit as a fast leading edge to the envelope.



The early decay circuit emphasises the percussive nature of the sound, and the damper operates in the normal piano manner, where it can be overridden by the sustain pedal. The resulting envelope is shown in Fig. 3.

The final decay is obtained from the loading of the chopper on the sustain storage circuit. The chopper is fed from the divider circuits and feeds an output to the signal busbars.

INTEGRATED CIRCUITS

It can easily be envisaged that an electronic piano is an ideal equipment for total integration, and this will be achieved when a special i.c. is designed for the envelope circuit. At present it would seem that different circuits, in discrete form, are used by each manufacturer of commercial equipment, but eventually one configuration would be expected to dominate, and the design of a hex or quintuple envelope i.c. will emerge. The "Joanna" uses i.c.s for master tone generation, dividers, voice circuits, preamplifier, headphone amplifier, and power supply.

PHYSICAL CONSTRUCTION

The prototype was housed in a teak veneered cabinet, mounted on chromium plated tubular legs. The pedals are fixed to a similar tube which spans the supporting feet which are made of solid teak. Whilst this combination gives a very attractive finish, a cheaper cabinet could be produced using a mixture of ordinary chipboard and blockboard, covered in a heavy vinyl material, which would also be more suitable if portability is required.

All the controls, including the pitch adjustment, are mounted on the narrow strip immediately above the keyboard, in an easy playing position. For maximum safety the lid should be screwed down, particularly to protect venturesome children.

Referring to the photograph of the interior details, the Power Supply is built on its own separate chassis, whilst the cabinet is designed to provide natural support for all the printed boards. Plug in connectors have not been used, in order to keep the cost to the minimum. The socket panel, positioned under the instrument, provides external amplifier, and headphone outputs, mains and pedal inputs, and carries the mains fuse. The keyboard is mounted directly to the base of the cabinet. This has the particular advantage, when used as an extra manual in an organ, of reducing the height between the keyboard and the top surface of the organ. The base of the cabinet is designed to give very easy access to the keyswitch assembly, without removal of the keyboard.

The piano stool has been designed to accommodate a power amplifier and 12in heavy duty loudspeaker. A mains socket has also been provided to give a neater appearance to the necessary interconnecting leads between piano, stool, and wall socket.

KEYBOARD AND KEYSWITCH

The "Joanna" cabinet design is based on the use of the five octave Kimber-Allen Swedish keyboard type S.K.A., and the Clef Products keyswitch type CPS 1027, which is specially designed for use with this keyboard. The high quality Kimber-Allen unit has been adopted due to its relatively low weight and high mechanical rigidity which also simplifies cabinet construction.

BUILDING SEQUENCE

The recommended sequence for construction is to commence by building a simple jig as shown in Fig. 4. This allows accurate construction of the final cabinet to be carried out in parallel with the electronic work, thus avoiding the delay associated with high standard woodwork and the possibility of damaging the cabinet during the testing of the sub-assemblies.

If finances allow the purchase of the keyboard and keyswitch at the start of the project, they can be mounted in the jig immediately and used later in testing out the envelope boards, but an alternative is to make a set of five single pole two-way switches as shown in Fig. 5, and mount them on the front base-panel of the jig. This will allow the testing of one Envelope Board at a time.

The first electronic units to build are the Power Supply and Tone Generator Board, which can be placed in their appropriate positions in the jig. Any power amplifier can be used to test the outputs from the Tone Generator Board, and the Socket panel can be used to give a permanent connection to the amplifier, with a test lead connected to the output socket via a 47kΩ resistor and a 0·5μF capacitor. The next step will be to build the Envelope Boards, which after insertion into their allotted places in the jig can be tested using the experimental keys and the test lead described earlier.

The final board to construct will be the Preamplifier, which will then allow testing of the complete system to be carried out. At this point the tested system, including the Keyboard, can be transferred to the completed cabinet.

TEST/CONSTRUCTION JIG DETAILS

The base of the jig is split into two parts, the front part 4in wide, and the rear part $8\frac{3}{4}$in wide. This leaves a gap of $3\frac{1}{4}$in which is spanned by the keyboard. The two base panels are held together by timber end cheeks, screwing from underneath. Four p.c.b. supports are mounted on the rear base panel in the positions shown. Two of the supports require slots cutting on both sides.

The Keyboard and Power Supply are fixed in the positions indicated. The p.s.u. chassis carries all plugs and sockets, and will provide the anchor point for the test lead. Four standard legs can be screwed onto the jig which produces a very useful work bench for the period of the project.

EXPERIMENTAL KEYS

If it is decided to leave purchase of the keyboard until later in the programme, the experimental key system shown in Fig. 5 can be constructed.

The five spring strips should be biased to press against the top screws, which are linked together and returned to ground potential. The five lower screws are also linked and returned to the supply voltage. In order to test the touch action the touch resistors should be wired as shown in the diagram.

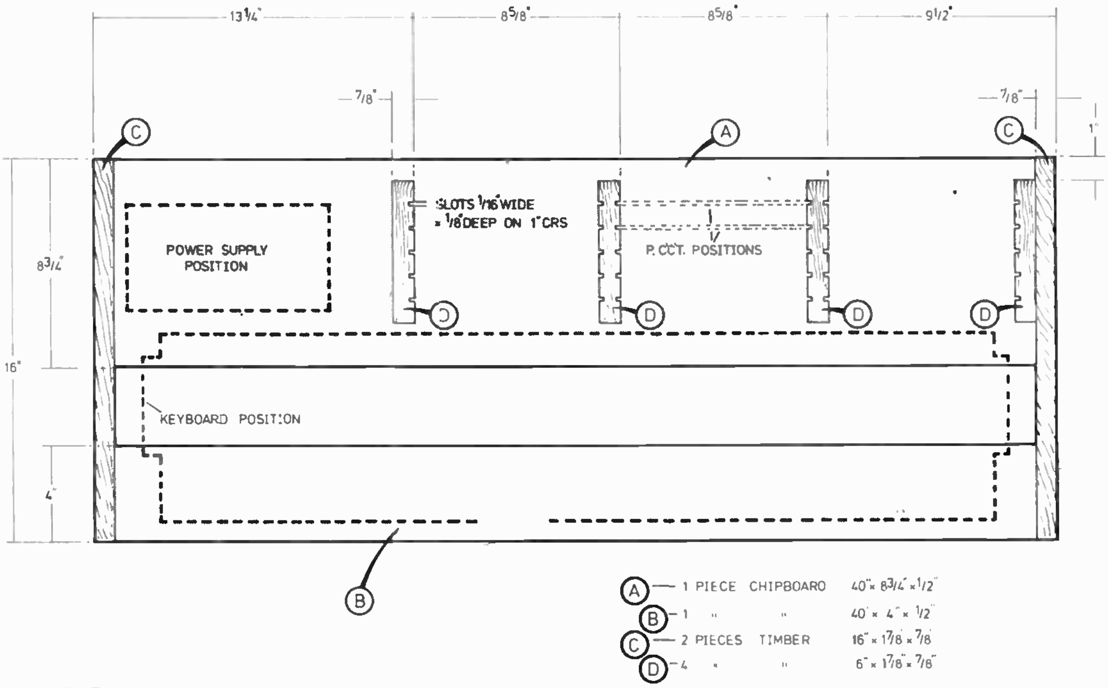


Fig. 4. Simple jig for testing sub-assemblies

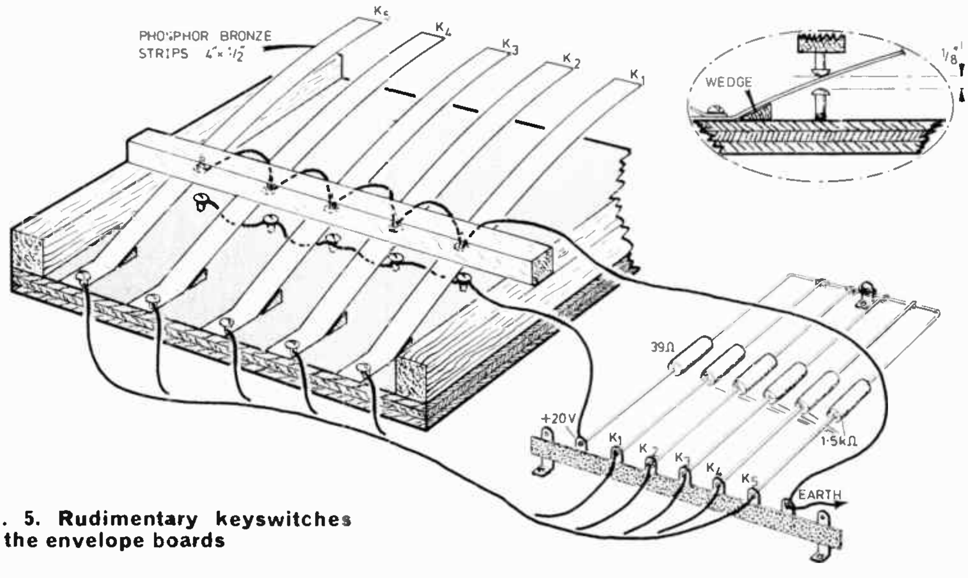


Fig. 5. Rudimentary keyswitches for the envelope boards

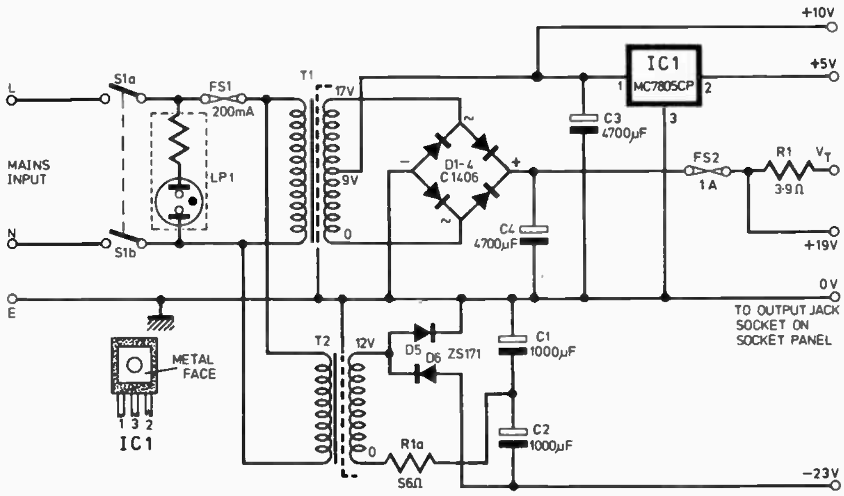


Fig. 6. Circuit of power supplies

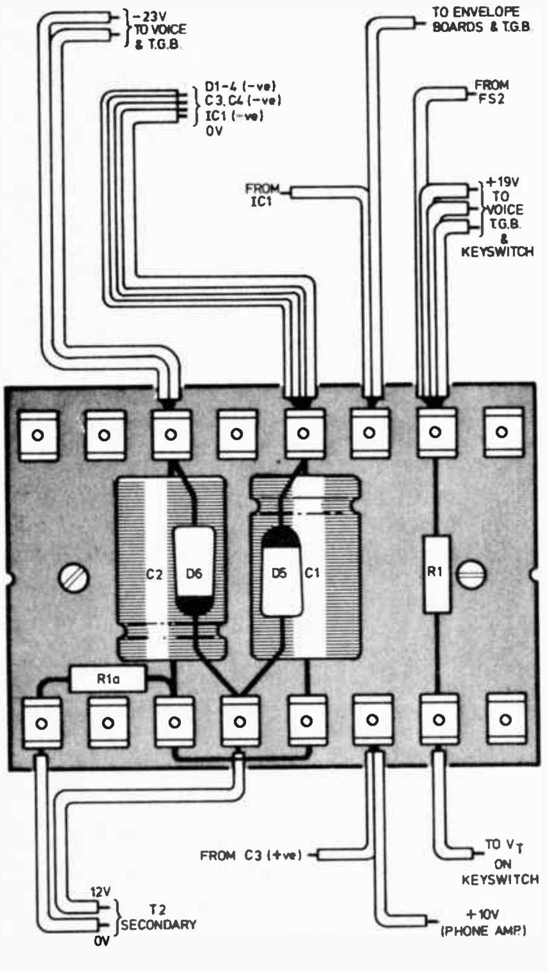


Fig. 7. Top and Bottom views of group board

COMPONENTS . . .

POWER SUPPLY UNIT
(Fig. 6)

Resistor
- R1 3·9Ω 2W
- R1a 56Ω ½W

Capacitors
- C1 1,000µF elect. 16V
- C2 1,000µF elect. 16V
- C3 4,700µF elect. 16V
- C4 4,700µF elect. 25V

Diodes
- D1–D4 50 2A bridge rectifier (Electrovalue)
- D5–6 ZS171

Integrated Circuit
- IC1 MC7805CP (Motorola)

Transformers
- T1 Mains transformer—Norman CT2 (Electrovalue) (see text)
- T2 Mains transformer—12V 100mA

Miscellaneous
FS1 500mA fuse, FS2 1 amp fuse, LP1 mains neon, S1 mains switch on level potentiometer, Aluminium sheet 9½in × 5½in, 8-way standard group board, SK1 3 pin miniature mains socket, SK2 Standard ¼in jack socket mono, SK3–SK4 Standard ¼in jack socket stereo.

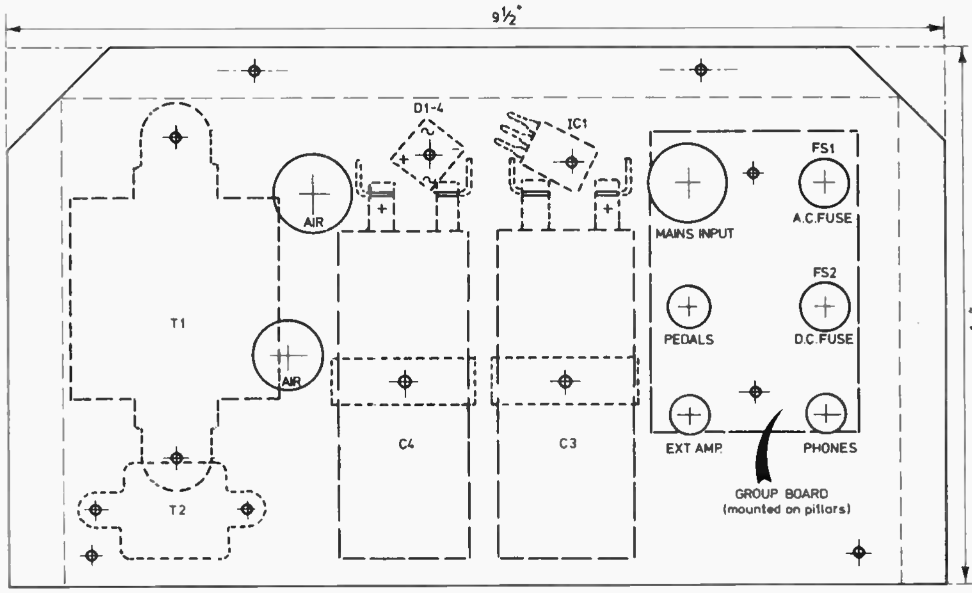


Fig. 8. Chassis drilling and bending detail for p.s.u.

POWER SUPPLY

The circuit of the power supplies is shown in Fig. 6 and is basically a simple unregulated system apart from the 5V logic supply which is provided by a three terminal integrated regulator.

Transformer T1 is rated at 17V, 2A, with a 9V tapping. Other suitable alternatives would be 9-0-9V or 10-0-10V, but in the latter case it would be necessary to increase the value of supply line resistors to compensate for the higher voltage. The potted bridge rectifier D1–D4 should have a p.i.v. rating of greater than 50V and a 2A capacity. Using the Norman CT2 transformer the input voltage to the 5V regulator is approximately 9V on load. The value at the high level output is 19V, with a 2V drop across R1 which is linked to the touch resistors on the key switch assembly.

The negative supply is obtained from a voltage doubler. The requirement here is for less than 50mA so that the current rating of diodes D5 and D6 can be very low, with a p.i.v. of greater than 70V. A number of manufacturers can supply miniature transformers suitable for T2; on the prototype an Eagle version was used which gave −22·5V on load.

SOCKET PANEL

All sockets for the Piano are integrated into the Power Supply chassis (see photograph) providing a convenient mounting point for the mains input, pedal input, external amplifier output, and headphone output sockets, and the mains input fuse. The zero level output of the Power Supply is used for connections to the other units in the Piano as will be shown later, but the earth input from the mains lead is connected to the output jack socket of the instrument, to avoid hum due to earth loops.

POWER SUPPLY ASSEMBLY

The plan in Fig. 8 and the photograph indicate the position of all the major components in the Power Supply Unit. A piece of aluminium sheet 9½in × 5½in is required to make the chassis which should first have two corners cut off as shown.

Holes ⅛in diameter should be drilled on the centres indicated. Holes for jack sockets are 7/16in diameter, fuseholders ½in and the Mains input socket ¾in diameter. The ¾in air vent holes should be punched to give maximum air flow to the transformer.

When all the holes have been completed the chassis should be folded along the dotted lines shown in Fig. 8. The long edge should be folded down for easy screwing from the bottom of the cabinet, whilst the two short edges should be folded upwards to give strengthening to the chassis.

The chassis is fixed to the cabinet using screws into the rear panel and centre base spar.

GROUP BOARD

The rectifiers and negative supply smoothing capacitors are mounted on a 16 terminal group board, the layout of which is shown in Fig. 7. The board is fixed to the chassis on pillars, consisting of threaded spacers 1½in long, in an inverted position. C1, C2 and D1–4 are soldered to the lugs whilst D5, D6 and FS2 are fixed to the back of the group board.

POWER SUPPLY CONNECTIONS AND FUNCTIONAL CHECK

In order to make the Power Supply a self- contained unit, all terminations are brought out to a tag strip running the full length of the supply. When all the interwiring has been completed the unit can be tested. Since both the internal resistance and the open circuit voltages of apparently similar transformers vary by a surprisingly high degree the unregulated voltage may have a fairly wide range.

The input to the 5V regulator should be in the region of 9–12V on a 600mA load but is not critical from the point of view of the regulator if it is in the range 8V to 15V. However, if for instance a 10-0-10V transformer is used, care should be taken not to exceed the voltage rating of C3.

The +19V and −23V are on load measurements and are likely to be exceeded by at least 20 per cent when measured under open circuit conditions.

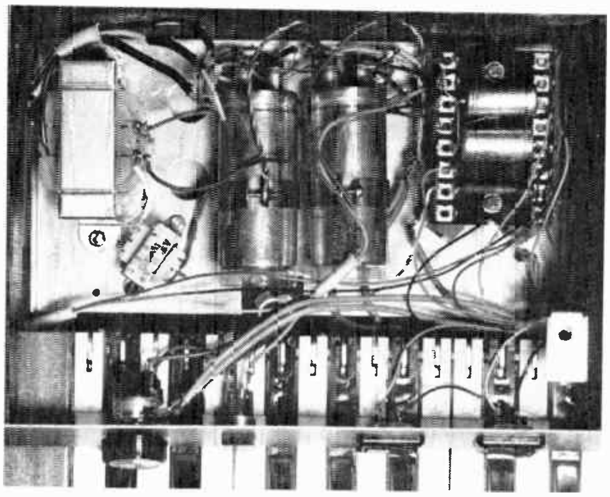
Again, if a transformer other than the Norman CT2 is used it would be advisable to use a higher voltage capacitor for C4.

Open circuit voltages measured on the prototype were +24V, −30V, and +12V at the input to the 5V regulator.

The transformer specified for T2 is commonly found with a 230V primary giving a high output. A resistor of 56 ohms between 0V and the junction of C1 and C2 will reduce the voltage across C1 and C2 on load (R1a in Fig. 6.)

TONE GENERATOR

Frequency generation in the Piano is dependent on the use of the General Instrument Microelectronics Master Tone Generator type AY-1-0212. This unit



Power Supply Unit note that input sockets and fuses are mounted under the group board—see Fig. 8

leads to the realisation of very accurate relative tuning across the keyboard, which is also extremely easy to adjust.

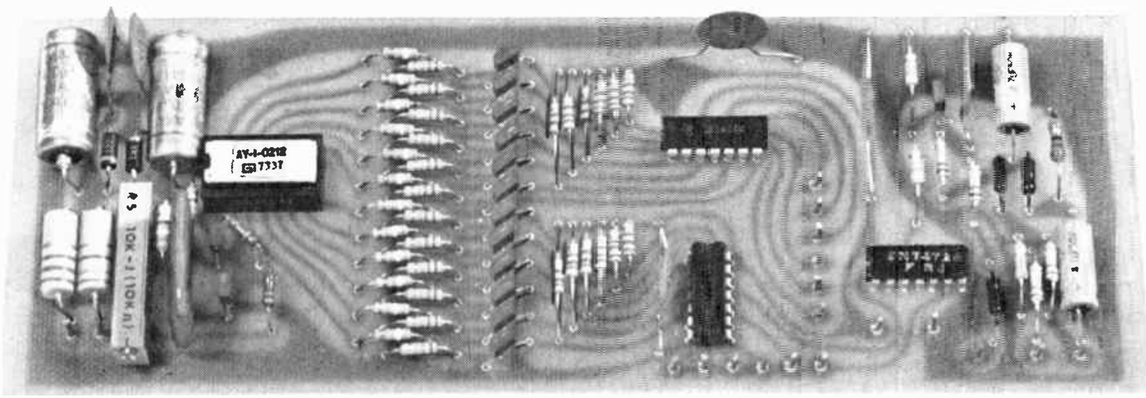
TUNING ACCURACY

Comparison of the top octave frequency values for the equal tempered scale and the chip indicate that maximum deviations occur in D, Eb, F and G, the worst representing 0·113 per cent of the absolute frequency.

Experiments have been reported to have shown that the average person can distinguish an 0·7 per cent deviation if two tones are sounded simultaneously with this error, and that an excellent musician might approach 0·03 per cent (probably by luck).

These tests were of course carried out under an unreal situation in that such a rigid comparison is not relevant to the intervals on a single piano keyboard. The author is convinced that the AY-1-0212 accuracy is irreproachable for this application, and the resulting chordal sounds from the keyboard are highly acceptable.

Photograph for M.T.G. and Top C Board



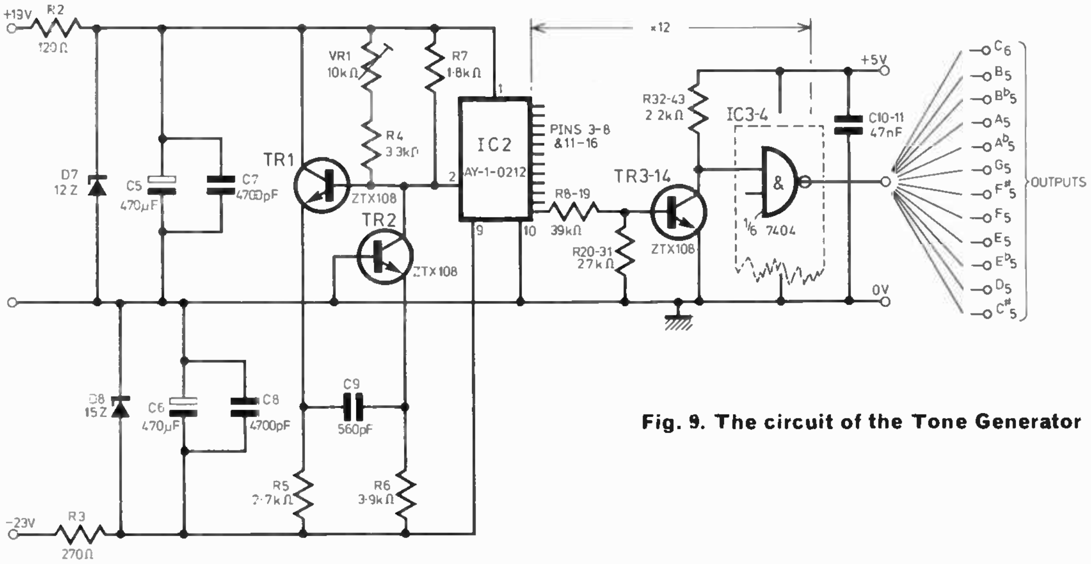


Fig. 9. The circuit of the Tone Generator

TONE GENERATOR CIRCUIT

The circuit of the Tone Generators is shown in Fig. 9. The raw supplies of 19V and −23V are regulated by Zener diodes to +12V and −15V respectively and supply a Bowes astable multivibrator driving the AY-1-0212. The Bowes astable is a non-saturating circuit easily capable of oscillating at the required frequency. The frequency of operation is determined by C9 and the combination of R7, VR1 and R4. For stability C9 should be mica or polystyrene, whilst R4 and R7 could be metal film.

The collector of TR2 swings between approximately 1V and 11V, which is the required input for the Master Tone Generator. The outputs of the AY-1-0212 are equivalent to a 3·5 kilohm resistor connected to a source alternately swinging between zero and 12V, and since the "Joanna" uses TTL dividers for economy it is necessary to have a buffer amplifier for each semitone of the top octave. TR3–TR14 perform this function, and each transistor is followed by a TTL gate in the hex inverter (7404) packages. The use of this additional logic buffer improves the input triggering reliability of the quad divider packages, and is well worth the small additional cost of the two i.c.s.

TOP C ENVELOPE CIRCUIT

For layout convenience the Tone Generator Board also contains the Envelope Shaper Circuit for top C. The circuit is shown in Fig. 11, with the single divider (7472) which provides pitch C5. The detailed operation of the Envelope circuit will be described later, and is not identical for all notes.

TONE GENERATOR BOARD ASSEMBLY

The Tone Generator and top C Envelope circuitry are combined on a single printed circuit board, the etching and drilling details of which are given in Fig. 10 with the component assembly details.

To assemble the board, the terminal pins should be fitted, followed by resistors, capacitors, diodes and transistors. VR1 should be mounted on the front panel. The printed circuit board has been planned to accept two popular versions of this component. IC3, 4 and 5, should be soldered into the board, and it is recommended that a socket be used in the IC2 position, ready to accept the Master Tone Generator.

If metal can transistors are used instead of the ZTX 108, great care should be taken to ensure that the cans do not touch each other.

PRECAUTIONS WITH THE M.T.G.

The AY-1-0212 is an integrated circuit using MTOS (metal-thick-oxide-silicon) p-channel enhancement mode field effect transistors. Whilst General Instruments have designed protection circuits into the device it is wise to take a number of precautions in its use.

The devices are normally supplied packed in a conductive foam, and should not be removed from this until the constructor is ready to insert the unit into the

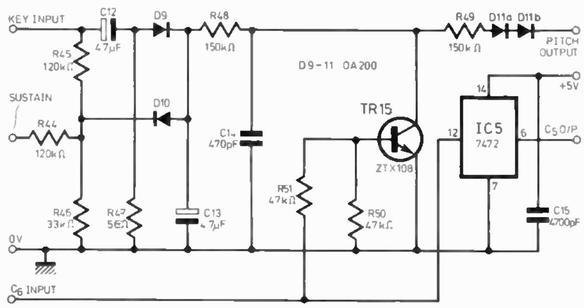


Fig. 11. Top C Envelope shaper

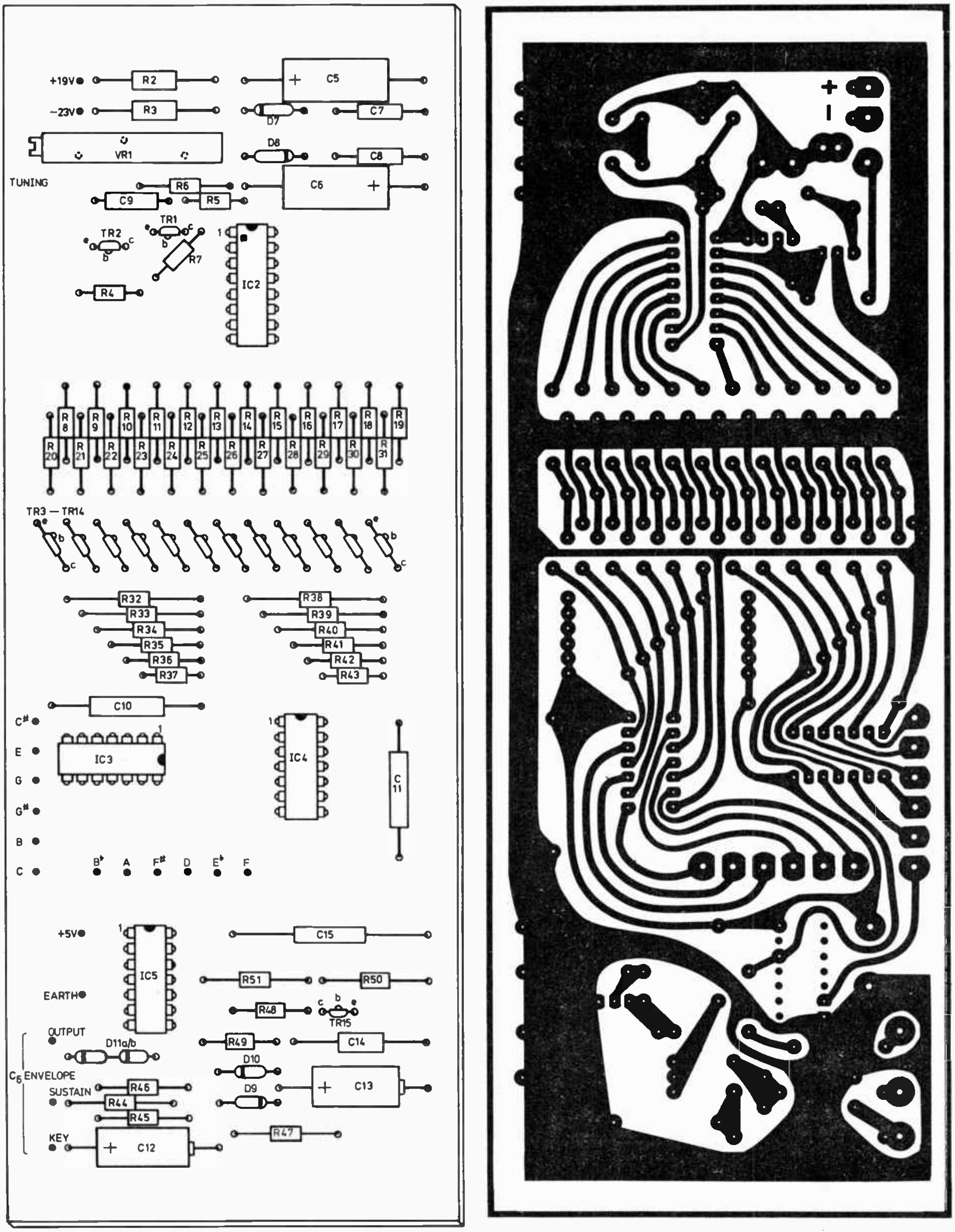


Fig. 10. Etching details and component layout for Master Tone Generator and Top C board

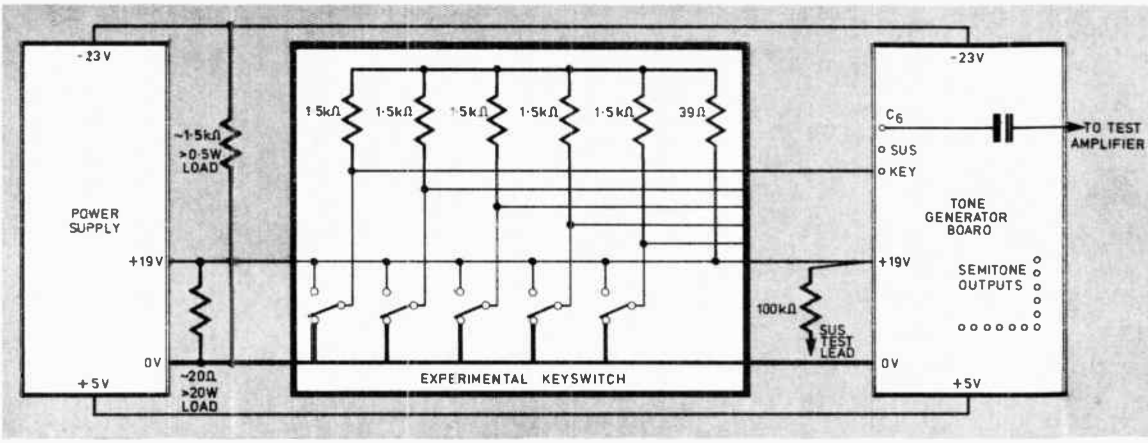


Fig. 12. Test set-up for the Tone Generator Board

socket. Handle the device carefully, having previously touched an earth lead, and if it is decided to risk avoiding the use of a socket then soldering should be carried out with a well earthed soldering iron.

EXPERIMENTING WITH THE TONE GENERATOR BOARD

The units built so far allow a test to be carried out, using the experimental keys and jig described previously. A test arrangement is shown in Fig 12 in which one of the five experimental keyswitches is used to trigger the C6 envelope, and the resultant output can be checked using an external amplifier as described previously. The 12 semitones can also be checked and should give continuous square wave outputs.

When performing this experiment the power supply should first be loaded and the outputs measured to ensure that the voltages applied to the Tone Generator Board are correct, or damage to components on the board could result. For the touch mechanism to work correctly the four unused keyswitches must be in the grounded position, and the 39 ohm resistor used to give the correct touch voltage.

The 100 kilohm resistor soldered to the +19V terminal on the Tone Generator Board provides the correct level to check the Sustain pedal action on the single envelope. This part is only a crude test, and connection of the leads may introduce hum which should be ignored.

In order to achieve independent operation for each note in the Piano is it necessary to provide a complete envelope generation system linked to each key on the keyboard. Each envelope shaper consists of a Touch Sensitive circuit followed by a Decay circuit. The latter is also designed to mix in the required pitch and to simulate the sustain pedal and damper action of a conventional piano.

TOUCH SENSITIVITY

The touch characteristics are shown in Fig. 13, together with the circuitry used to achieve the effect. The keyswitch is normally at ground potential until a note is played, such that the voltage across capacitor CT is zero. On depression of a key, the switch leaves the ground busbar and starts to travel towards the rail (19 volts) busbar. This allows capacitor CT to charge through the resistors RT, such that the voltage on the positive plate of CT follows curve A, according to the time constant RTCT to the final touch level voltage on RT of approximately 17 volts.

When the key completes its travel a 19 volt pulse is applied to CT, which for a very short time raises the

COMPONENTS . . .

TONE GENERATOR BOARD
(Fig. 9 and Fig. 11)

Resistors

R2	120Ω 1 watt	R32–43	2·2kΩ
R3	270Ω 1 watt	R44–45	120kΩ
R4	3.3kΩ metal film	R46	33kΩ
R5	2·7kΩ	R47	56Ω
R6	3·9kΩ	R48	150kΩ
R7	1·8Ω metal film	R49	150kΩ
R8–19	39kΩ	R50–51	47kΩ
R20–31	27kΩ		

All $\frac{1}{3}$ watt, 5%, except where stated otherwise

Capacitors

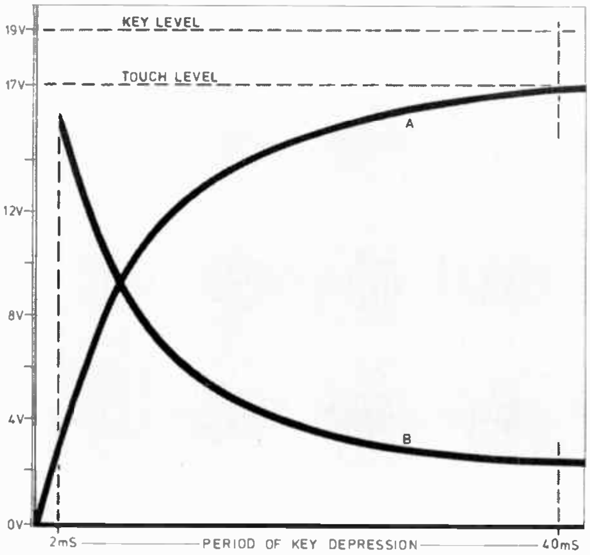
C5–6	470μF elect. 16V	C12	4·7μF elect. 25V
C7–8	47nF ceramic	C13	4·7μF elect. 16V
C9	560pF mica	C14	470pF
C10–11	47nF ceramic	C15	47nF

Potentiometer
VR1 10kΩ linear

Diodes
D7 12 volt Zener diode 1 watt
D8 15 volt Zener diode 1 watt
D9-11 a, b Silicon diodes

Transistors
TR1–14 ZTX108 TR15 ZTX108

Integrated Circuits
IC2 AY-1-0212 G.I.M. IC5 ZN7472
IC3–4 ZN7404



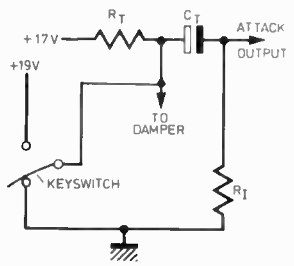
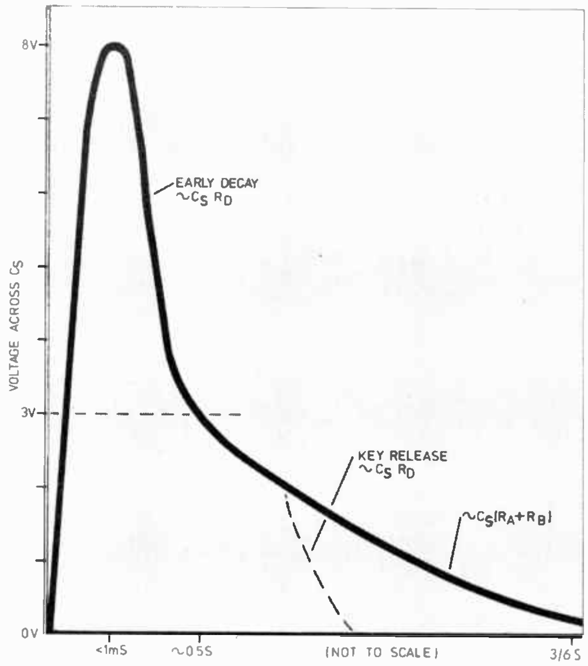


Fig. 13. Basic Touch circuitry and explanatory curves



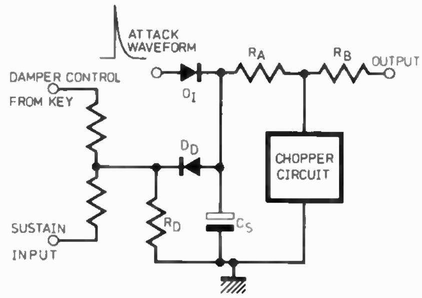


Fig. 14. Basic Decay circuit and curve

voltage at the junction of C_T and R_I by an amount equal to 19 volts minus the voltage across C_T at that time. This results in an output which follows curve B, over the range of normal key-travel times of between 40ms and 2ms, offering a variable attack voltage which is passed on to the Decay circuitry.

A closer investigation of the attack pulse shows that two pulses do in fact occur. The first pulse is very small, and occurs at the moment when the keyswitch leaves the ground busbar, and is kept to a minimum by the choice of a high ratio for $R_T : R_I$. Later components in the Decay circuit ensure that this pulse does not get through to the preamplifiers.

The values established for R_I are critical in obtaining maximum touch feel, and since they obviously take a fairly high current drain in the rest position, consumption has been minimised by the use of slightly higher values than optimum at the extreme ends of the keyboard.

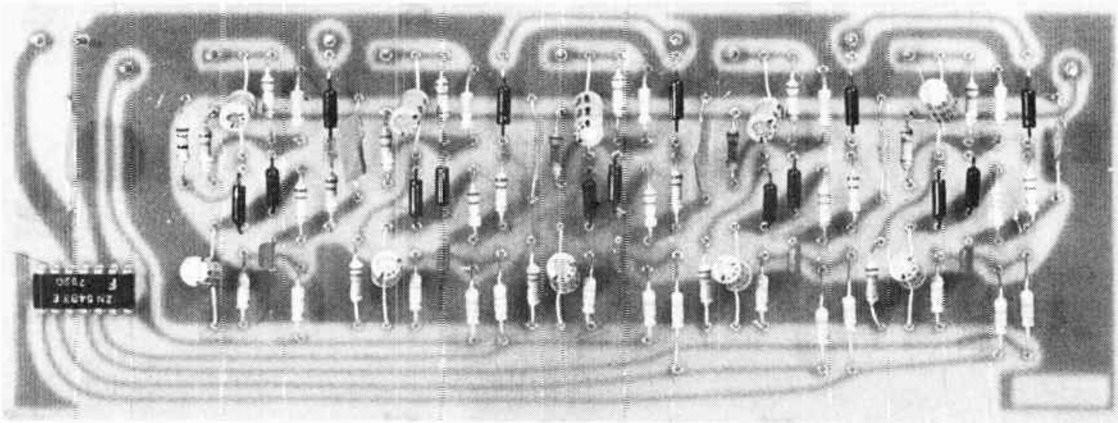
The attack trigger decays very quickly ($C_T R_I$) due to the necessarily low value of R_I. The attack level is proportional to the average speed of depression of the key over the full travel, which is a very similar situation to the final key velocity characteristic of a conventional piano since the latter is normally achieved by an even application of energy.

DECAY CHARACTERISTIC

The Decay circuitry is shown in Fig. 14 together with the resulting characteristics in the various modes of operation. The circuit consists of a capacitor C_S, which stores the energy passed to it from the attack pulse, a Damper and Early Decay circuit, and a chopper circuit via which the envelope is modulated to introduce the pitch.

At the moment when the keyswitch reaches the rail busbar, as described in the previous section, the rail voltage is applied to the Damper circuit and the attack pulse appears at the isolating diode D_I in Fig. 14. Damper diode D_D normally holds the voltage across capacitor C_S to nearly zero via the damper resistor R_D, but the application of the rail voltage lifts the voltage on the cathode of D_D to approximately three volts.

Thus as the attack pulse is applied to C_S through diode D_I the capacitor is allowed to charge to a voltage determined by the ratio of C_S to C_T, followed by a quick decay to a level of three volts plus the forward volt drop of diode D_D.



Complete Envelope Board assembly

This action is termed the "early decay", and whilst it is fast compared with the final decay action, it is long compared with the collapse of the attack pulse ($C_T R_I$), such that it is not influenced by the touch portion of the circuit which is isolated by diode D_I immediately after the attack voltage has appeared.

The early decay characteristic emphasises the percussive nature of the instrument.

Assuming the key remains depressed the voltage across C_S will continue to decay, but at the much slower rate defined by resistors R_A and R_B. It will be shown later that the chopper transistor works on a 1 : 3 mark space ratio, such that for three quarters of the period the decay time is determined by R_A, and for the remaining quarter of the period it is defined by the sum of R_A and R_B.

Different values of R_A and R_B are used for each octave to give a variation in decay time of from approximately 6 seconds to 3 seconds across the compass.

If the key is released before the voltage has fully decayed, the damper resistor R_D will determine the rate of final decay. This action will however be overridden if the sustain pedal is used since the voltage on the cathode of D_D is pulled up by the sustain voltage in a similar manner to the damper release action described above. Release of the sustain pedal brings back the C_S R_D decay as before.

CHOPPER ACTION

The complete Envelope circuit is shown in Fig. 15, which also gives the chopper circuitry. The transistor is driven by two waveforms taken from the quad divider. The two basic frequencies are the fundamental and the second harmonic, both in the form of square waves. The resulting waveform on the collector of the transistor is shown in Fig. 16, together with its harmonic spectrum.

This is a relatively easy waveform to handle in circuits which are inherently non-linear a more usual staircase type of waveform is completely unsuitable, producing a harmonic change over the period of the decay which is the inverse of the conventional piano tone.

The fast edges of this waveform can be dangerous in their tendency to produce "beehive" break-through—i.e. a continuous background of every note in the instrument. Capacitor C_N slows the leading

COMPONENTS . . .

ENVELOPE BOARD (12 REQUIRED)

(Fig. 17)

Resistors

R52–61	120kΩ	R75	180kΩ
R62–66	33kΩ	R76	150kΩ
R67–71	56Ω	R77	220kΩ
R72	330kΩ	R78	180kΩ
R73	270kΩ	R79–81	150kΩ
R74	220kΩ	R82–95	47kΩ

All $\frac{1}{3}$ watt 5% carbon

Capacitors

C16–20	4·7μF 25V	C28	1nF
C21–25	4·7μF 16V	C29–30	470pF
C26	3·3nF	C31	47nF
C27	2·2nF		

Diodes

D12–26 Cheap Silicon Diodes (e.g. 1S44, OA200 types) (20 off)

Transistors

TR16–20 ZTX108

Integrated Circuit

IC6 7493

Miscellaneous

Terminal Pins—12 off.

Glass fibre printed circuit boards, and complete component kits, are available for this project from Clef Products, 31 Mountfield Road, Bramhall, Stockport, Cheshire SK7 1LY.

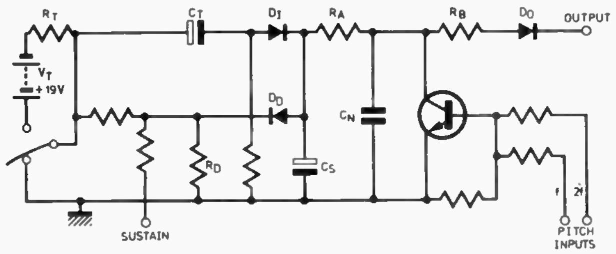


Fig. 15. Complete Envelope circuit

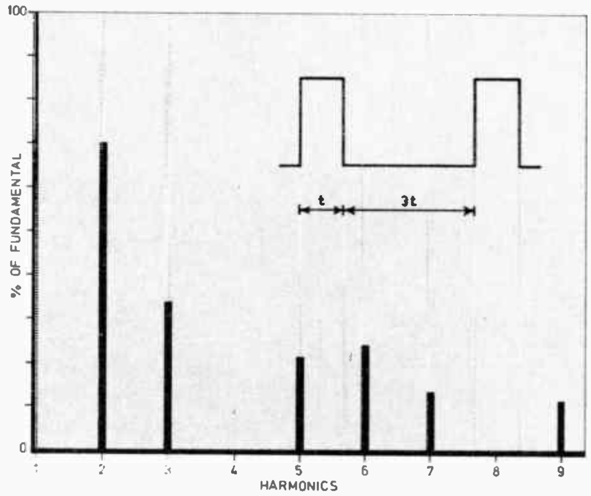


Fig. 16. Harmonic spectrum of basic waveform is shown on right

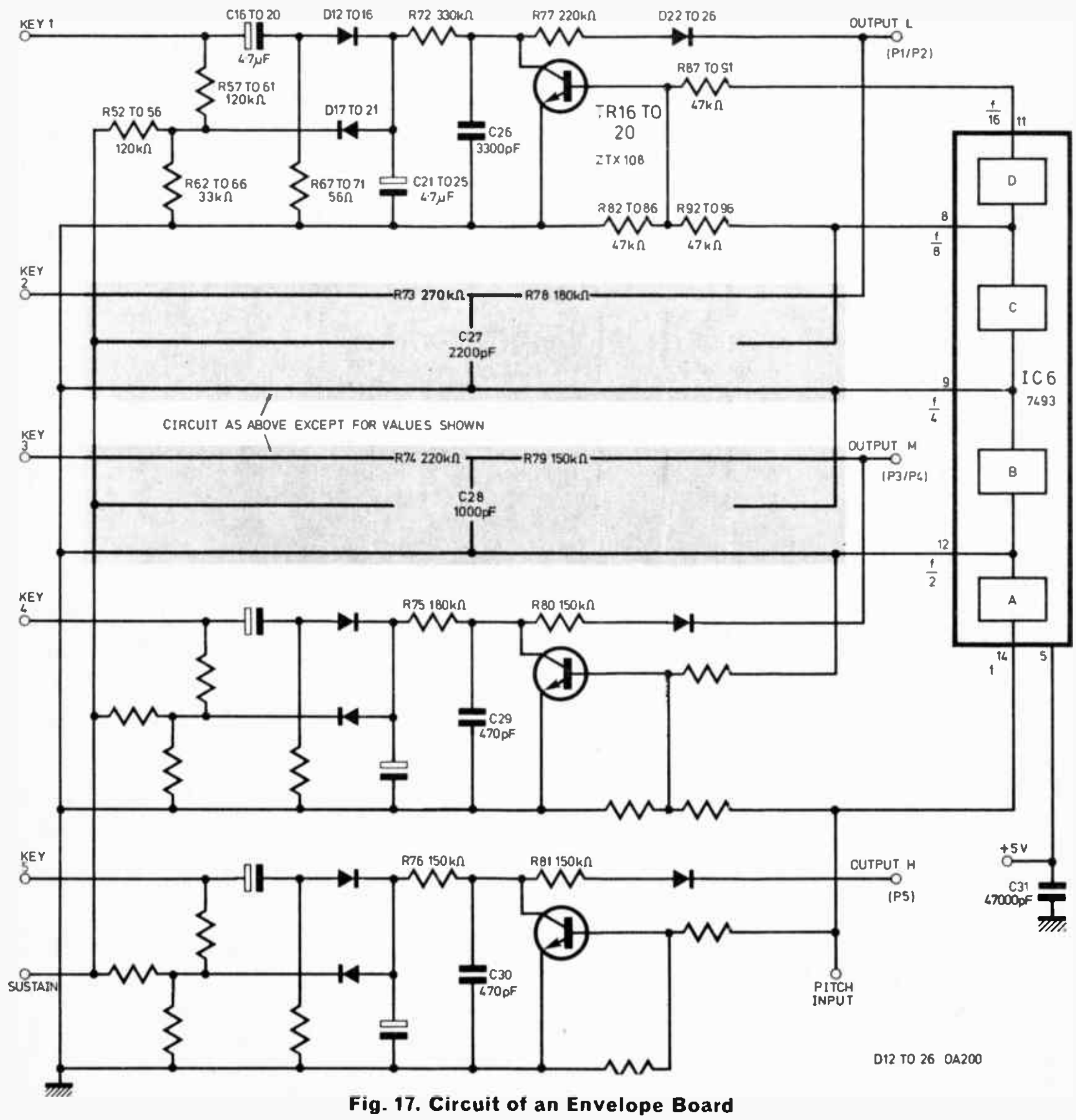
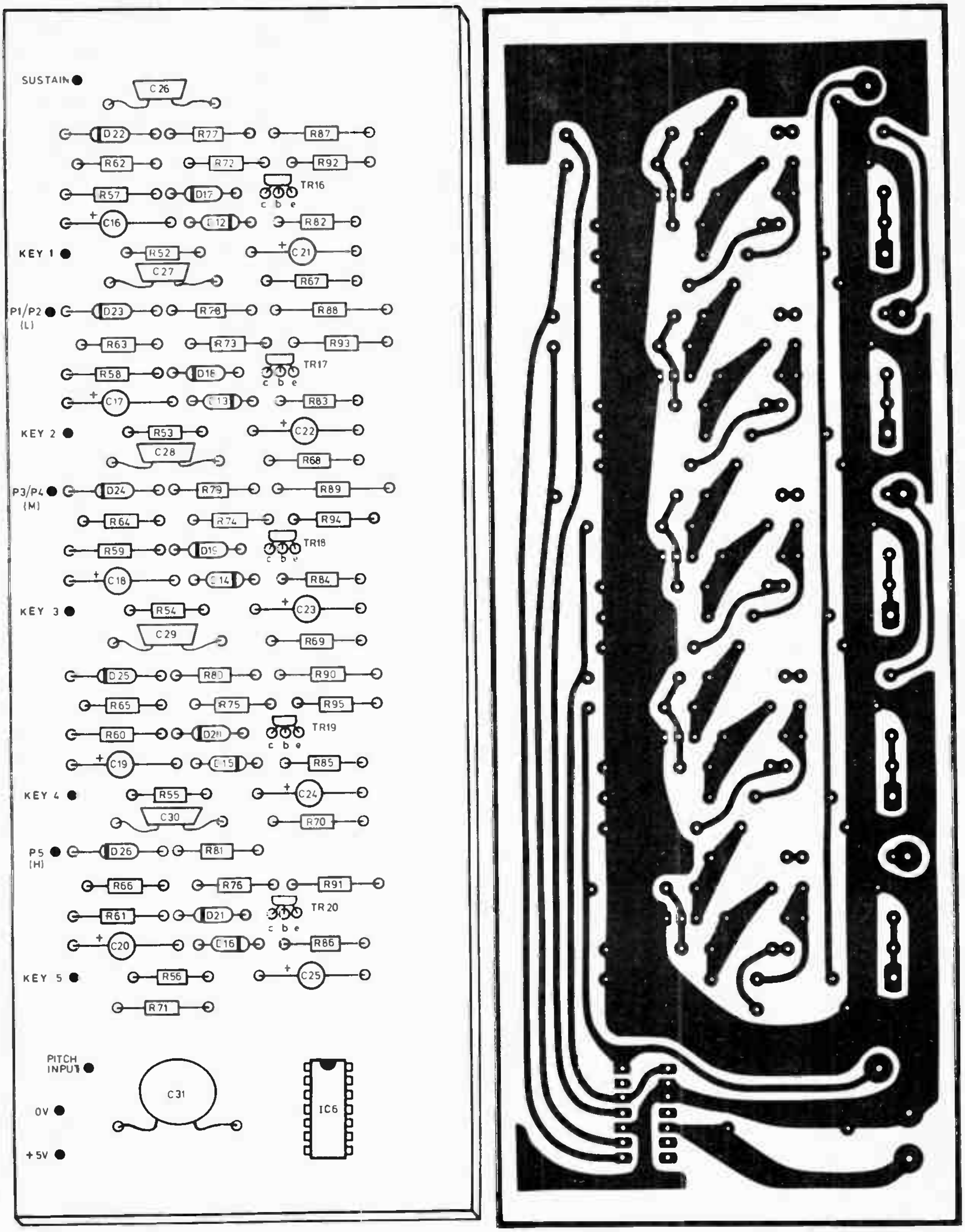


Fig. 17. Circuit of an Envelope Board



Note: D22 to D26 each comprise a pair of series connected diodes, a, b, (see Fig. 17)

Fig. 18. Component layout and etching details for one Envelope Board

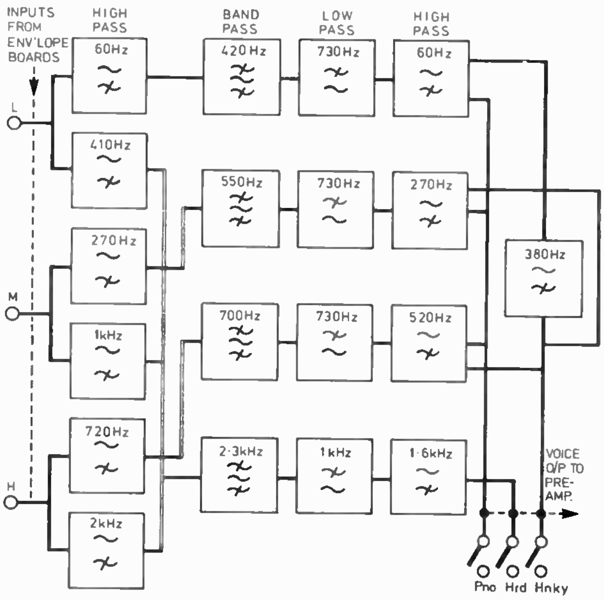


Fig. 19. Voicing filters for the Joanna

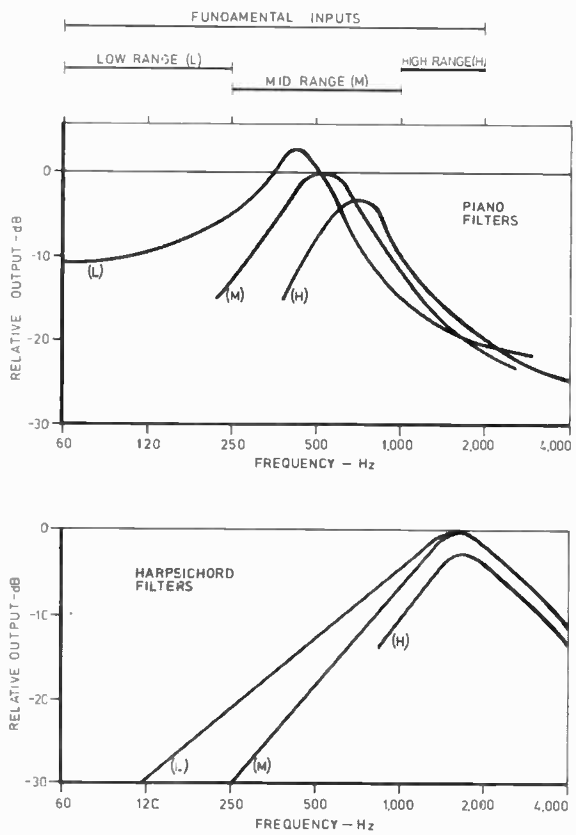


Fig. 20. Frequency responses for the Piano and Harpsichord filters

edge of the waveform to reduce this effect, and the output diode Do also acts as a noise reducing element.

Further beehive reduction is incorporated in the Voice circuits.

FREQUENCY DIVIDERS

The square waves to drive the chopper transistors are produced by a divide-by-sixteen counter, which has four outputs at half (A), one quarter (B), one eighth (C), and one sixteenth (D) of the input frequency.

The divider input is obtained from the gate outputs on the Tone Generator Board, described earlier, and is fed into the circuit shown in Fig. 17. The input frequency is also used to produce the top octave pitch waveforms which are simple 1 : 1 square waves.

PRACTICAL ENVELOPE CIRCUIT

The Envelope circuits are grouped as five per board, together with one quad-divider. Each of these combinations copes with all octaves of one semitone across the keyboard, leading to 12 identical Envelope Boards being required. The full circuitry for one board is shown in Fig. 17. The board contains five key inputs, one sustain input, and one pitch input. The outputs are grouped to cover the bottom two octaves, the middle two octaves, and the top octave, at separate output terminals. The board requires only one 5 volt supply, to power the divider.

ENVELOPE BOARD CONSTRUCTION

Each group of Envelope circuits is constructed on a printed circuit board 203 × 76mm, the etching and drilling details for which are given in Fig. 18 together with component details.

To assemble the board the terminal pins should be fitted, followed by resistors, capacitors, diodes, transistors and integrated circuit. It is important that both the transistors and the integrated circuit should be inserted with the correct orientation.

DIODES

The author has used diodes in the prototype which can be described loosely as manufacturer's rejects, of silicon planar type in DO-7 encapsulation.

To test the diodes a multimeter was used with 20kΩ/volt sensitivity (1kΩ range). Diodes were rejected where a movement of the needle was observed in the reverse polarity position, whilst the forward resistance was generally of the order of 12kΩ, although no specification was applied to this parameter.

Occasionally a diode selected in this way gave some trouble if used in the output position (breakthrough) but the success rate was very high, after the diodes had passed the test. The forward resistance is measured when the negative lead of the multimeter is connected to the anode, and the positive lead to the cathode of the diode.

TESTING THE ENVELOPE BOARDS

The Envelope Boards can be tested one at a time using the jig described earlier. Since the power supplies are unregulated (apart from the 5 volt logic supply) the warning is repeated to keep a check on the supply levels, particularly to the Tone Generator Board, whilst performing any partial check-out experiments.

VOICE FILTERS

There are two basic sets of Voice Filters, one set covers the Piano tone, whilst the other set produces the Harpsichord effect. The Honky-Tonk sound is produced by mixing the two voices together. The full set of filters is shown in schematic form in Fig. 19.

The bottom two octave outputs from the Envelope Board are connected to the low (L) input, the middle two octaves to the (M) input, and the top octave to the (H) input.

The voice circuits make wide use of passive low and high pass filters, but the main response is obtained from the use of adjustable band pass amplifiers based on the integrated circuit operational amplifier type 741.

FREQUENCY RESPONSE

The overall frequency response of the Piano and Harpsichord filters are shown in Fig. 20.

Referring to the Piano filter response curves it can be seen that the three filters are set up centred on different frequencies of approximately 420Hz, 550Hz, and 700Hz. This makes the bottom three octaves rich in at least the second harmonic, with a lower harmonic content for the top two octaves. The range of fundamental frequencies applied to each filter is indicated above the graph.

HARPSICHORD FILTERS

The Harpsichord filters have a very sharp bass cut response, with some variation between (L), (M) and (H) inputs. Consequently the second and higher harmonics are strongly emphasised, with the fundamental being well attenuated. A high frequency roll off is necessary in order to reduce the effect of beehive breakthrough.

From both Fig. 19 and Fig. 20 it can be seen that a high degree of low frequency cut is used on the mid and high range Piano inputs. This controls the attack "thump" in these registers to a realistic degree, and further high pass filtering on the low range input reduces the bass response when the Honky-Tonk voice is selected.

OUTPUT WAVEFORMS

Sample waveforms are shown in Fig. 21, for each of the C notes on the Piano voice setting, and middle C on Honky-Tonk and Harpsichord voice settings. On the Piano voice waveforms the variation in harmonic content across the compass is clearly demonstrated, with predictable modifications to the waveform for the other two voices.

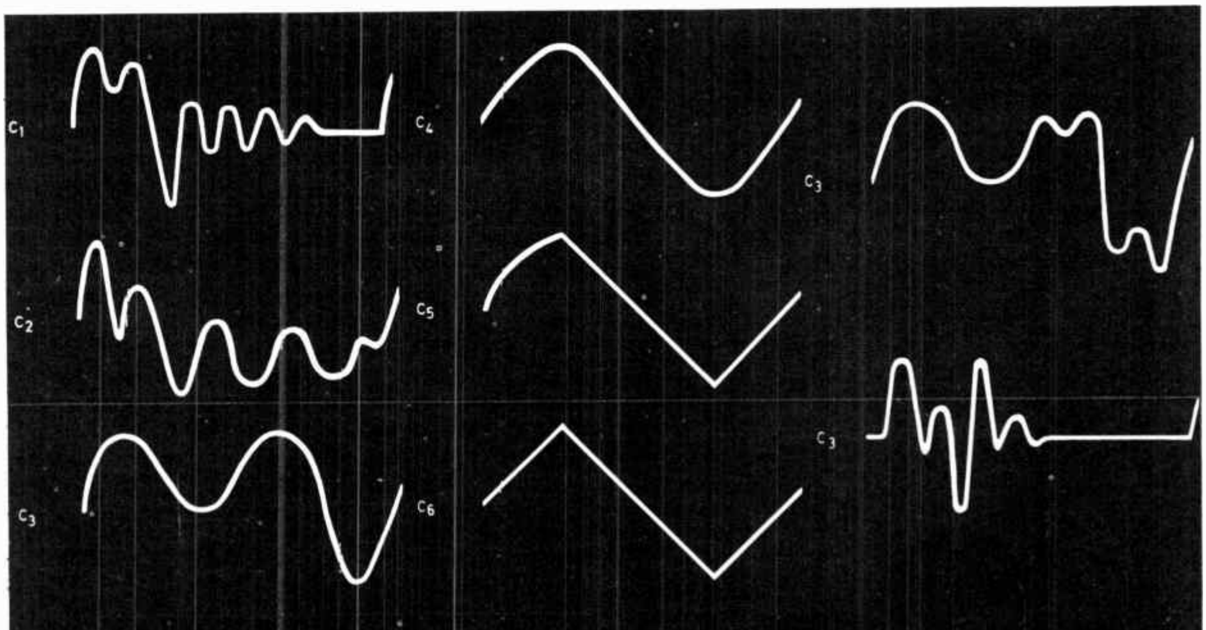


Fig. 21. Sample output waveforms for the three voice settings. C to C are for each of the Piano voice C notes, C upper right and C lower are for the Honky Tonk and Harpsichord voice settings

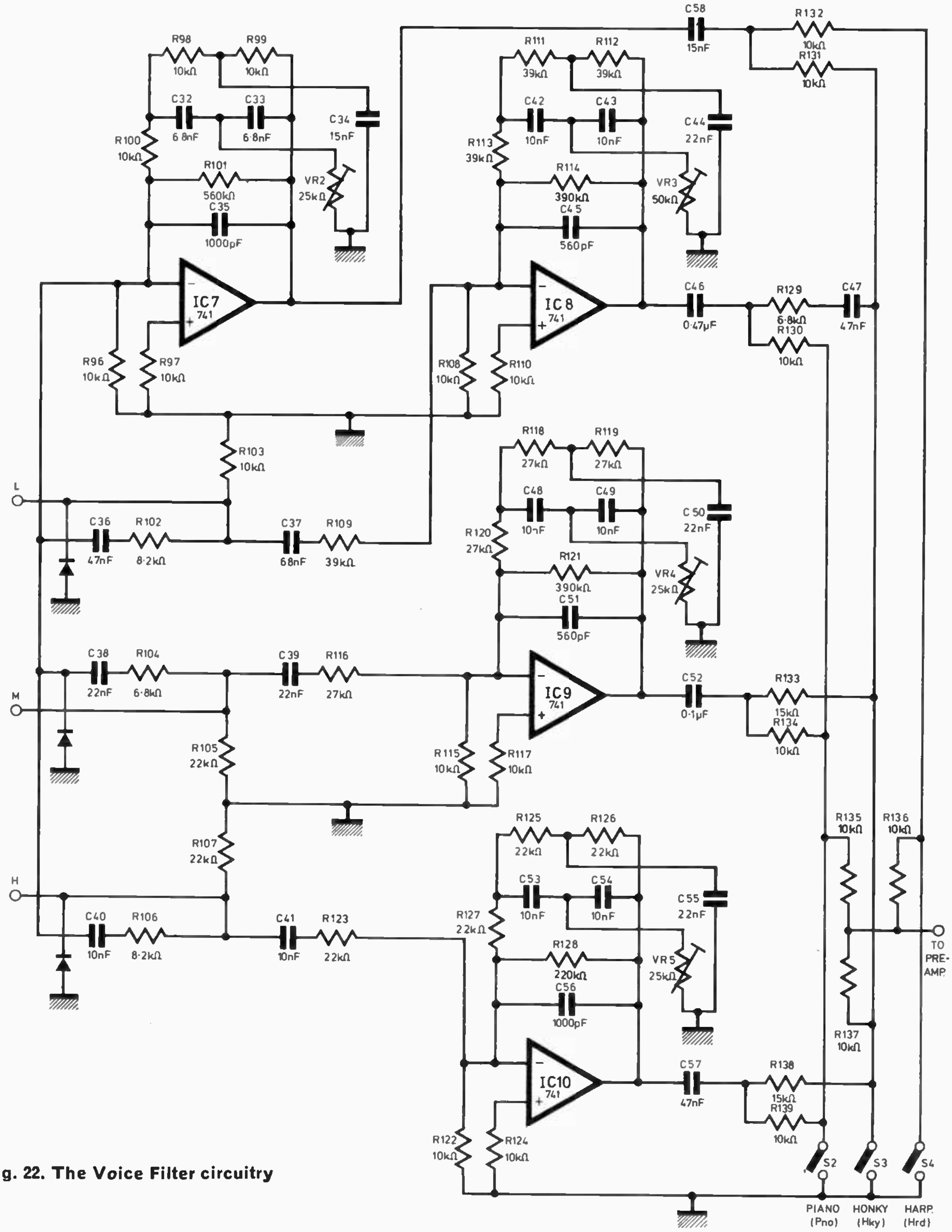


Fig. 22. The Voice Filter circuitry

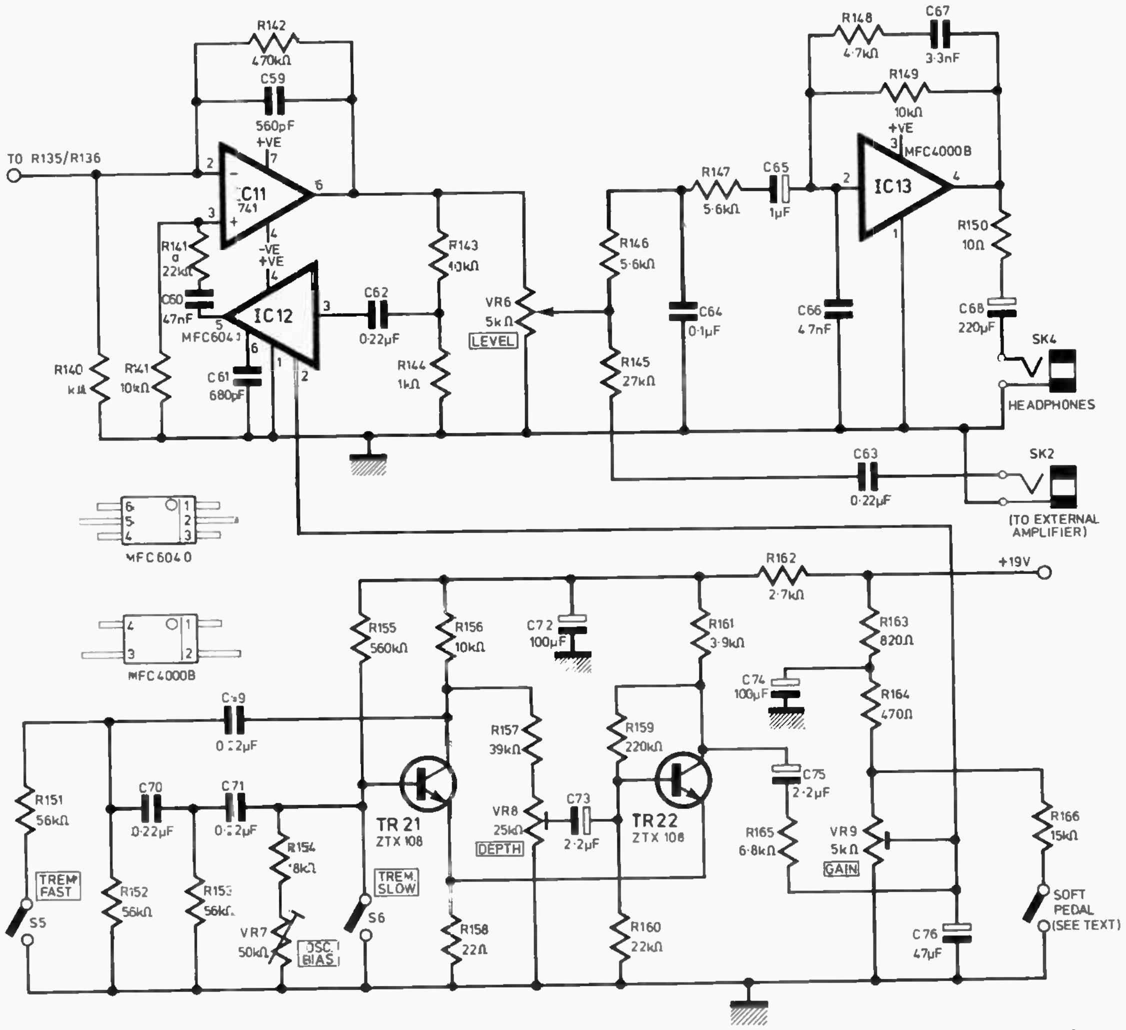


Fig. 23. Circuits for the Preamplifier, Headphone Amplifier, Tremolo Generator and Soft Pedal

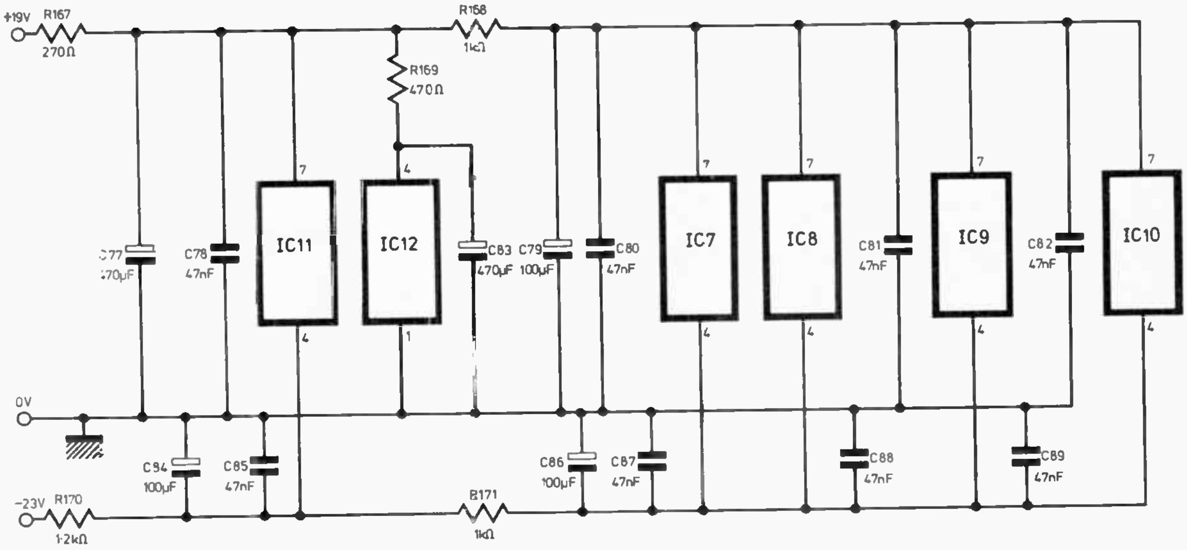


Fig. 24. Power supply line components for the Voice Circuits and Preamplifiers

VOICE FILTER CIRCUITS

The Voice Filter circuitry is shown in Fig. 22. The low, medium, and high inputs are controlled by I.C. 8, 9 and 10 respectively, whilst the main Harpsichord filter is built around IC7. The Q of these twin-T band-pass amplifiers can be adjusted by the associated preset potentiometers, and the voice selection is achieved by opening the correct voice switch which is normally shorting its signal to ground.

PREAMPLIFIER

The Voice Filter outputs are mixed into a 741 preamplifier, which has a voltage controlled amplifier type MFC6040 in a feedback loop. This i.c. can handle up to 500mV r.m.s. at pin 3 and therefore requires a voltage sharing resistor network on its input as shown in Fig. 23.

Since the amplifier is an invertor its output is fed into the non-inverting input of the 741 to give negative feedback.

The overall gain of the preamplifier is controlled by a d.c. input line to pin 2 of the MFC6040 the voltage of which is set by preset potentiometer VR9, and is further adjusted by the Soft Pedal switch and Tremolo Generator.

HEADPHONE AMPLIFIER

The circuit for the headphone amplifier is also shown in Fig. 23, and uses a low power (250mW) Class B integrated circuit type MFC4000B.

This amplifier will drive approximately 40mW into standard stereo headphones, which when connected in parallel have an impedance of approximately 4 ohms. The MFC4000B is powered by a 10 volt rail taken from the power supply unit on the input to the 5 volt regulator.

TREMOLO GENERATOR

A phase shift oscillator can be operated at two alternative frequencies to generate slow and fast

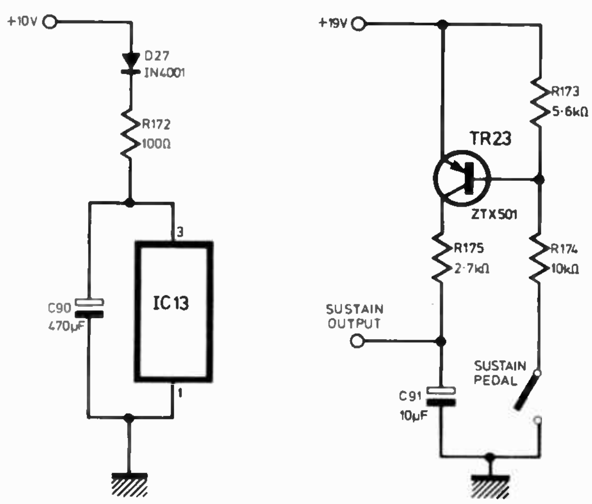


Fig. 25. Power supply components for Headphone Amplifier and the Sustain Pedal circuit

COMPONENTS . . .

VOICE/PREAMPLIFIER BOARD
(Fig. 22, Fig. 23, Fig. 24 and Fig. 25)

Resistors

R96–100	10kΩ	R128	220kΩ	R155	560kΩ
R101	560kΩ	R129	6·8kΩ	R156	10kΩ
R102	8·2kΩ	R130–132	10kΩ	R157	39kΩ
R103	10kΩ	R133	15kΩ	R158	22Ω
R104	3·8kΩ	R134	10kΩ	R159	220kΩ
R105	22kΩ	R135–137	10kΩ	R160	22kΩ
R106	8·2kΩ	R138	15kΩ	R161	3·9kΩ
R107	22kΩ	R139	10kΩ	R162	2·7kΩ
R108	10kΩ	R140	39kΩ	R163	820Ω
R109	39kΩ	R141	4·7kΩ	R164	470Ω
R110	10kΩ	R141a	22kΩ	R165	6·8kΩ
R111–113	39kΩ	R142	470kΩ	R166	15kΩ
R114	390kΩ	R143	10kΩ	R167	270Ω
R115	10kΩ	R144	1kΩ	R168	1kΩ
R116	27kΩ	R145	27kΩ	R169	470Ω
R117	10kΩ	R146–147	5·6kΩ	R170	1·2kΩ
R118–120	27kΩ	R148	4·7kΩ	R171	1kΩ
R121	390kΩ	R149	10kΩ	R172	100Ω
R122	10kΩ	R150	10Ω	R173	5·6kΩ
R123	22kΩ	R151–153	56kΩ	R174	10kΩ
R124	10kΩ	R154	18kΩ	R175	2·7kΩ
R125–127	22kΩ				

All ⅓ watt, 5% tolerance

Capacitors

C32–33	6·8nF	C62–63	0·22µF
C34	15nF	C64	0·1µF
C35	1,000pF	C65	1µF 16V elect.
C36	47nF	C66	4·7nF
C37	68nF	C67	3·3nF
C38–39	22nF	C68	220µF 10V
C40–43	10nF	C69–71	0·22µF
C44	22nF	C72	100µF 16V elect.
C45	560pF	C73	2·2µF 16 V elect.
C46	0·47µF	C74	100µF 16V elect.
C47	47nF	C75	2·2µF 16V elect.
C48–49	10nF	C76	47µF 16V elect.
C50	22nF	C77	470µF 16V elect.
C51	560pF	C78	47nF
C52	0·1µF	C79	100µF 16V elect.
C53–54	10nF	C80–82	47nF
C55	22nF	C83	470µF 16V elect.
C56	1,000pF	C84	100µF 16V elect.
C57	47nF	C85	47nF
C58	15nF	C86	100µF 16V elect.
C59	560pF	C87–89	47nF
C60	4·7nF	C90	470µF 10V elect.
C61	680pF	C91	10µF 16V elect.

Diode
D27 1N4001

Transistors
TR21–22 ZTX108 TR23 ZTX501

Integrated Circuits
IC7–11 741 IC12 MFC6040 IC13 MFC4000B

Potentiometers

VR2	25kΩ horiz preset
VR3	50kΩ preset
VR4	5–25kΩ preset
VR6	5kΩ lin pot. with switch
VR7	50kΩ horiz preset
VR8	22kΩ horiz preset
VR9	5kΩ horiz preset

Miscellaneous
S2–6 Single pole on-off switch
Terminal pins (27 off)

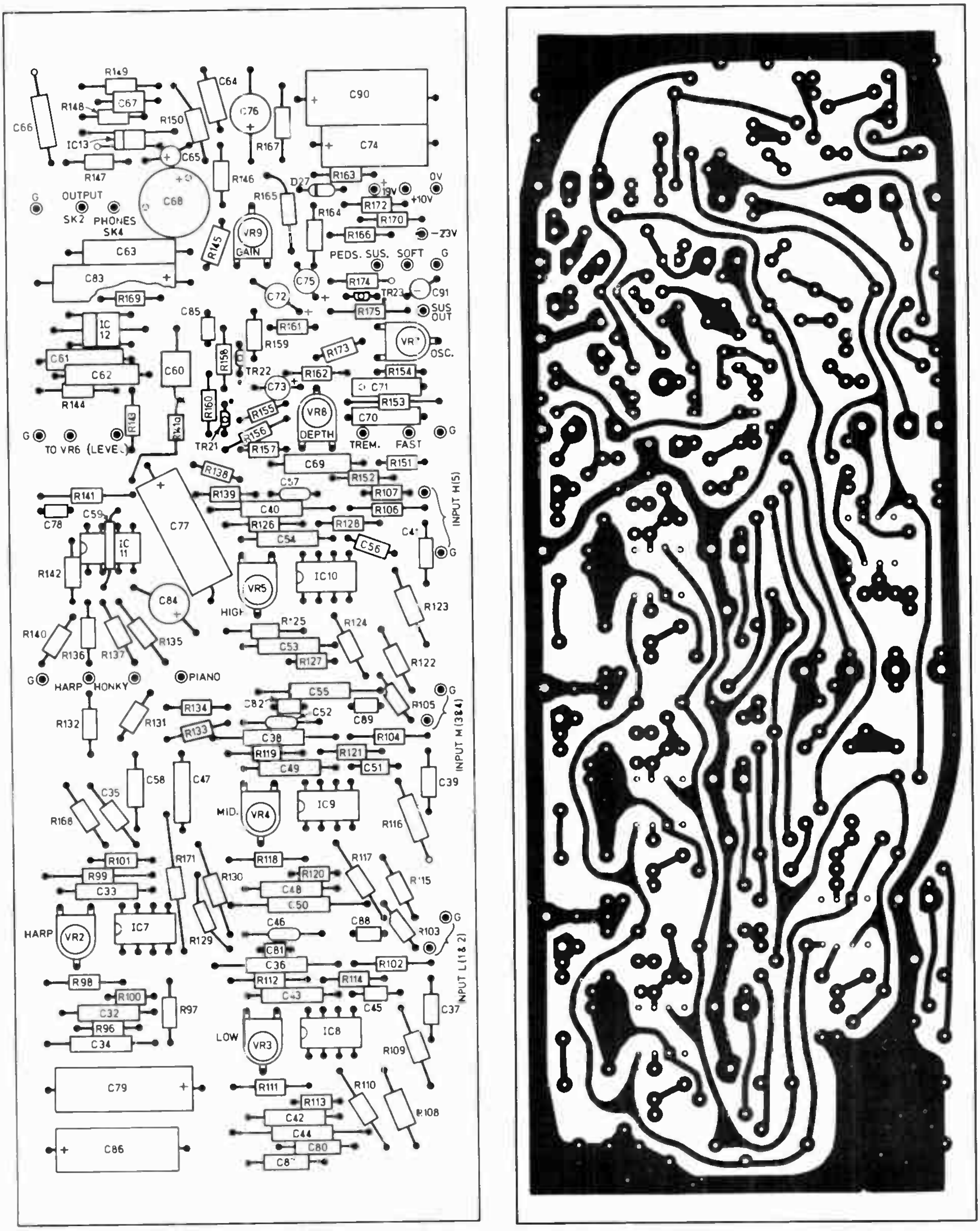


Fig. 26. Component mounting and p.c.b. etching details for the Voice Filters, Preamplifier, Headphone Amplifier, Tremolo, Soft and Sustain Pedal

tremolo effects, TR22 amplifies the sinewave output from the oscillator and modulates the d.c. control voltage to the MFC6040.

SUPPLY LINES AND SUSTAIN PEDAL CIRCUITS

In order to simplify Figs. 22 and 23, the power supply lines and components have been omitted and are shown in Figs. 24 and 25.

The Sustain Pedal circuit is also shown which uses a *pnp* transistor, normally in the off condition. When the pedal is depressed, the transistor is switched on, and the output rises from ground potential to approximately 9 volts (see Fig. 25).

VOICE/PREAMPLIFIER BOARD

The Voice Filters, Preamplifier, Headphone Amplifier, Tremolo circuitry, Soft and Sustain Pedal circuitry are all mounted on a single printed circuit board, the etching and drilling details for which are given in Fig. 26, with component mounting details.

The component density on this board is very high, and some extra care will be necessary in construction to ensure that the components used can be accommodated on the board.

Terminal pins and presets should first be assembled on the board, followed by resistors, small capacitors, transistors, and integrated circuits.

The larger capacitors should be assembled last, and will in some cases be positioned at a distance from the board surface to clear small components which will lie underneath them.

Care should be taken to ensure that the transistors and integrated circuits are inserted in the correct orientation, corresponding with the assembly details given in Fig. 26, and that the electrolytics are inserted with the correct polarity.

CUTTING LIST

CABINET	
Front Base Board	36¾in × 4in × ⅝in
Centre Base Spar	36¾in × 2¼in × ½in
Rear Panel	38in × 4½in × ⅝in
End Panels (2 off)	14in × 4½in × ⅝in
Front Keyboard Trim	36¾in × 1¼in × ⅝in
Front (fascia) Panel	36¾in × 1½in × 3/16in
Top Panel	36¾in × 7in × 3/16in
Bottom Cover Panel (Pegboard)	36¾in × 10¼in × ⅛in
P.c.b. Supports—shaped (4 off)	6⅞in × 3¼in × ⅞in
P.c.b. Bottom Retainers (4 off)	4in × ⅞in × ⅛in
Top Panel Support (p.s.u. end)	4¾in × 1in × ¼in
Front Panel Support (p.s.u. end)	2⅛in × ½in × ½in
End Cheeks (2 off)	7in × 1½in × ⅞in
Rear p.s.u. Support	11in × ½in × ½in
Felt Strip	36¾in × ½in × ⅛in
Magnetic Catches (2 off)	
LEGS	
Top Spar (2 off)	12½in × 1½in × ¾in
Bottom Spar (2 off)	14in × 1¾in × 1in
Legs (4 off)	Chrome plated tubing 24in × ¾in dia.
Bottom Tube	Chrome plated tubing 26in × ¾in dia.
PEDALS	
Rear Block	6½in × 2in × 1in
Foot Pedals (2 off)	5in × 1in × ⅝in
Base	7½in × 7in × ⅛in
Springs and hinges	
STOOL	
End Panels (2 off)	19in × 16in × ⅝in
Top/Bottom Panels (2 off)	20in × 14in × ¾in
Side Panels (2 off)	20in × 6½in × ½in

VOICE CIRCUIT ADJUSTMENT

The overall gain of the Preamplifier is first set by adjustment of VR9 to a level which does not distort the output when two heavy chords are played together at maximum weight, with the output level control VR6 set at maximum.

The tone/colour of the Piano sound is set by VR3, VR4 and VR5, for the lower two octaves, middle two octaves and top octaves respectively. These should be individually adjusted, with S3 and S4 closed, to suit the ear of the constructor.

VR2 controls the Harpsichord and should be adjusted with S2 and S3 closed, to the point of minimum background breakthrough.

TREMOLO ADJUSTMENT

With S6 open the Tremolo Generator may or may not oscillate immediately which can be determined by setting VR8 approximately at the centre of its travel. If oscillation does not occur, VR7 should be adjusted until oscillation commences. VR8 should then be adjusted until a good level of tremolo is obtained with minimum breakthrough sound in the speaker when a note is not being played.

S6 should be switched on and off to ensure that the generator starts reliably every time, and VR7 slightly readjusted until a consistent start is obtained. With S5 closed the fast tremolo should work. If the background tremolo breakthrough increases on this setting, VR8 should be adjusted.

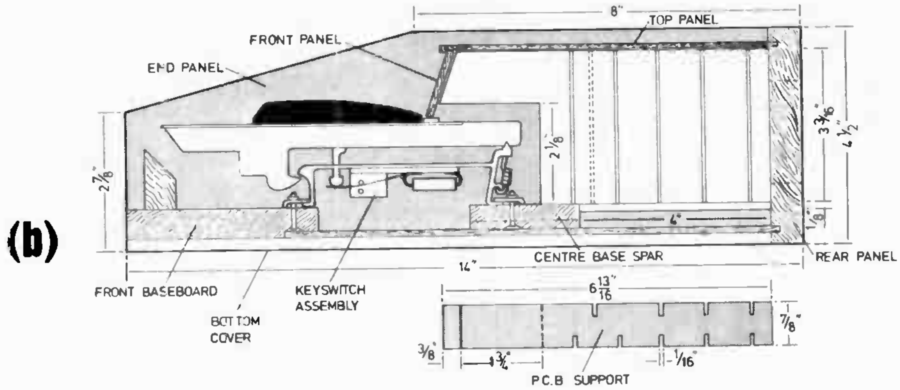
OVERALL TONE VARIATION

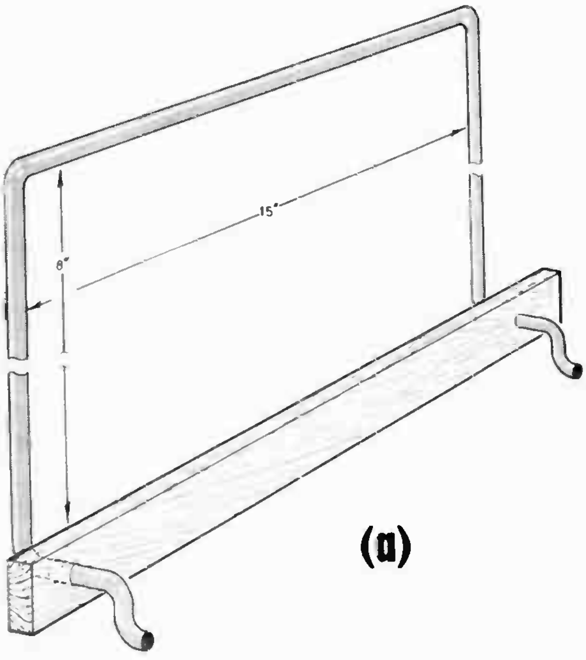
Some variation in overall tone may occur depending on the power amplifier and speaker combination used with the Piano. Further top cut may be obtained if required by increasing the value of C59.

PEDAL SWITCHES

The switches associated with the pedals have a very simple single pole on/off action, the mechanical arrangement of which will be described in the last part of the series. Pedal action occurs in both cases when the relevant switch is closed to ground, and can be simulated for test purposes by shorting straps on the board from the relevant pin to ground.

In the case of the Soft Pedal, the degree of attenuation is controlled by R166. The individual characteristics of each MFC6040 can effect the attenuated level, and it may in some cases be necessary to change the value of R166 to suit. A lower value should be used for increased attenuation.





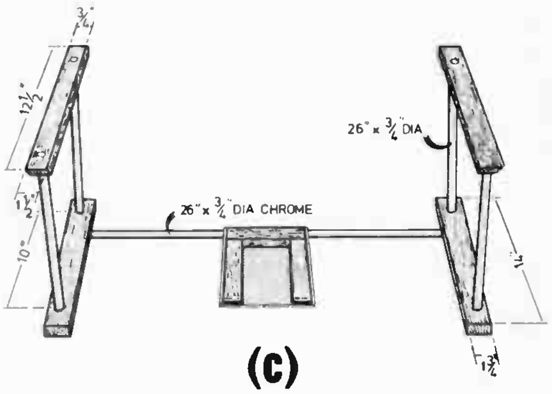
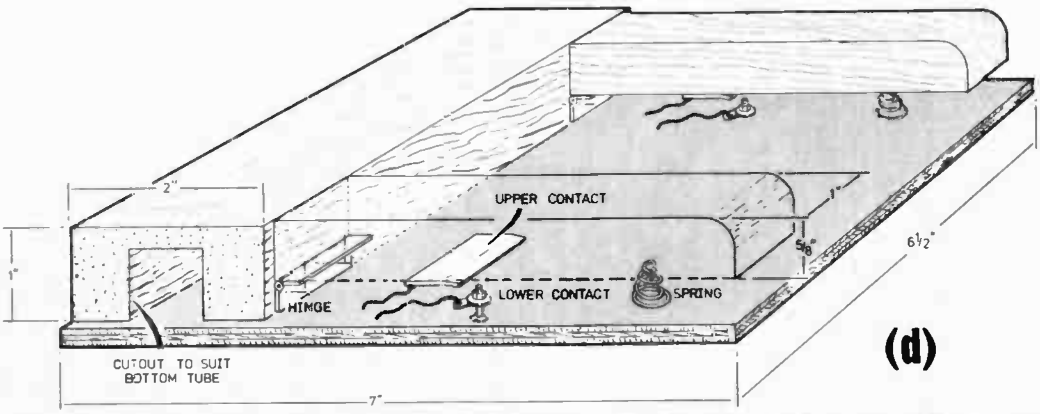
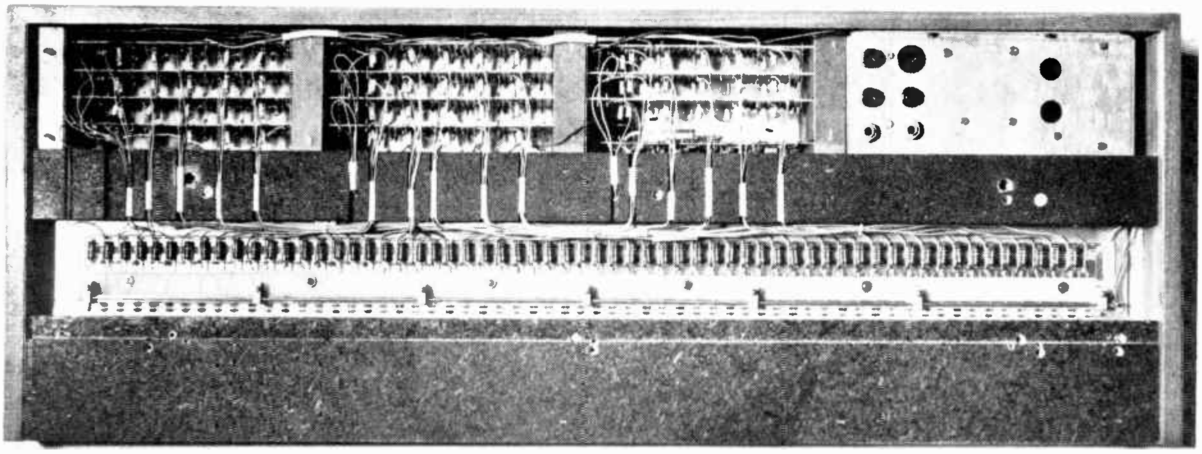


Fig. 27. (a) Music Stand (b) cross section of "Joanna" cabinet assembly; (c) leg assembly (d) soft and sustain pedal details





CABINET DESIGN

The basic philosophy behind the cabinet design is to allow maximum wiring to be carried out from the rear, simplifying the interconnections between the keyswitch and the p.c.b.s. This is a considerable improvement over the prototype arrangement in which wiring was carried out from the top, thus making it impossible to remove a single board with ease, and reduces the wiring complexity.

KEYBOARD POSITION

The keyboard spans a gap between the front baseboard and the centre base spar, allowing immediate access to the keyswitch, and a further 4in gap exists between the spar and the rear panel of the cabinet, revealing the terminal pin edge of all p.c.b.s (Fig. 27).

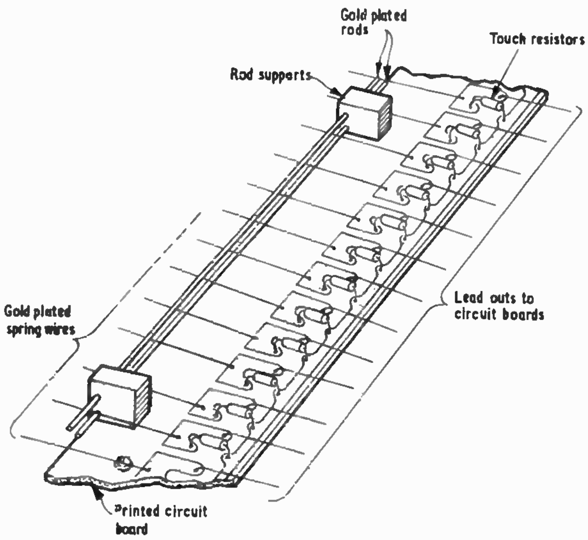


Fig. 28. Keyswitch assembly is screwed to the underside of the keyboard so that the actuators press against the spring wires

The two base members in combination with the rear panel provide the main rigidity in the cabinet, further strengthened by the p.c.b. supports and keyboard when fitted, and fix to the two shaped end panels. The p.c.b. supports have been designed to hold the p.c.b.s, provide fixing points for the front panel, and support to the top panel. It should be noticed that the p.c.b. slots are spaced differently on each support. The bottom retainer strips prevent the p.c.b.s falling to the bottom of the cabinet.

A bottom cover panel made of pegboard covers the wiring but allows ventilation of the cabinet.

Magnetic catches have been used on the prototype to retain the top panel.

ASSEMBLY OF THE CABINET

The front baseboard, centre base spar, rear and end panels should first be assembled using joints to match the skill of the constructor. The p.c.b. supports should then be fixed to the rear panel and centre base spar, and the p.c.b. retainers glued in position.

At this point the keyboard should be fixed into the cabinet inserting from the front. Large washers are used to trap the keyboard channel edging to both base bars ensuring that the keyboard is central by using the end cheeks as guides.

When the keyboard is fixed, the keyboard trims can be fitted, followed by the end cheeks which are screwed down to the keyboard channel after drilling suitable holes in the channel.

The top panel support, front panel support and front panel can now be fitted, followed by the rear p.s.u. support.

If the unit has been constructed accurately the top and bottom cover panels should easily fit into place, and the magnetic catches can then be fixed.

KEYSWITCH ASSEMBLY

The keyswitch assembly consists of a printed circuit board to which gold plated springy wires are soldered and seven plastic blocks are fixed to support two gold plated rods (Fig. 28).

Before assembling the kit of parts together, the p.c.b. should be placed on the underside of the keyboard, and used as a template for drilling the six $\frac{3}{32}$in holes required to take the self-tapping screws which fix the switch assembly to the frame. The front edge of the switch assembly should be placed approximately $\frac{1}{4}$in from the keyboard actuators, and should be centralised over its length with respect to the actuators. When holes have been drilled, and the p.c.b. removed from the frame, the keyswitch is assembled as follows.

1. Slide the seven rod supports onto one rod.
2. Insert the second rod through the remaining rod holes, and space the supports roughly opposite the holes which have been countersunk from the non-copper side of the p.c.b.
3. Fix the rod supports to the copper side of the p.c.b. using the countersunk screws, nuts and locking washers. The track is formed to indicate the correct positions.
4. Cut the gold plated wire into 61 lengths of $1\frac{3}{4}$in, and tin approximately $\frac{1}{4}$in of one end of each wire.
5. Pass each wire between the two rods and solder the tinned end to a main pad. About $\frac{1}{2}$in should be left to hang over the front edge of the p.c.b. The wire should be pressed flat to the p.c.b. close to the main pad using a matchstick, and the soldering iron applied to the wire and land for as short a time as possible.
6. The assembly can now be fixed to the underside of the keyboard using the self-tapping screws with two washers under each screw head.
7. The next operation is to adjust the bend of each wire to ensure that it seats satisfactorily against each rod in turn, whilst gently resting on the actuator in the non-depressed state. An attempt should be made to achieve a gentle curve in the wire commencing on the land side of the rod in order to prevent too much localised stress where the wire presses against the top rod.

The touch resistors are soldered to the pads provided, one end to the common touch rail, and the other to the land which also carries the gold-plated springy wire. Top and bottom octaves should be $1{\cdot}8\text{k}\Omega$, and the middle three octaves $1{\cdot}5\text{k}\Omega$.

The resistor leads should be bent back under the resistor and cut short before soldering to the board, which then allows the component to be fixed within the small space available.

INTERWIRING PROCEDURE

The p.c.b.s should be inserted into the slots with components facing forward and the top panel fitted. The cabinet should be turned upside down to reveal the back edge of the p.c.b.s (which have been effectively inserted upside down) and the keyswitch previously fitted as above. Wiring should be carried out as follows:

1. A group of three wires (suggest red, orange, yellow) should be soldered to the C_1, $C_1\#$ and D_1 pads on the keyswitch and threaded through a hole drilled through in the centre base spar. The hole is opposite the correct pins on the three Envelope Boards, and should be connected in the same order.
2. A group of five wires (red, orange, yellow, green, blue) should be soldered to the $E^\flat_1$, E_1, F_1, $E_1\#$ G_1 pads on the keyswitch and threaded through another drilled hole.
   This hole is opposite the pins in five Envelope Boards, and the wires should be soldered in the same order.
3. The operation should be repeated for $A_1^\flat$, A_1, $B^\flat_1$, B_1 (red, orange, yellow, green), passing through the $A^\flat_1$–B_1 hole in the centre base spar.
4. A further group of three wires (red, orange, yellow) should be soldered to the C_2, $C_2\#$ and D_2 pads, threaded through the hole marked C_2–D_2 and soldered to the relevant boards.
5. If this operation is repeated for 60 keyswitch pads (excluding C_6), each Envelope Board uses a single colour for five connections and is easily seen to be correct. The C_6 connection is made to the Tone Generator Board, which has been inserted upside down.

TONE GENERATOR BOARD TUNING LOCATION

The rotary tuning potentiometer VR1 should be mounted on the front panel. This replaces the multi-turn preset potentiometer shown on the component layout diagram for the Master Tone Generator board. Three leads should connect the potentiometer to the board, with the centre tag on the potentiometer connected to centre hole on board.

INTERWIRING BETWEEN BOARDS

The next operation is to link the Envelope Boards for the outputs, +5V, zero volt, and sustain rails. This is best achieved using bare single strand wire pre-cut to the required length.

The G connections associated with each input (L, M and H) on the Voice Board (Fig. 26) are redundant and should be ignored.

Since the pins are double-ended, one half protruding from each side of the board, links can be soldered between the pins which can be easily undone should it be necessary to remove a single board at a later date.

When all the boards are interlinked the interwiring of the three rear boards should be carried out as shown, leaving flying leads for connection to the +5V and zero volt connections on the Power Supply Unit.

At this point connections should be made to the two keyswitch rod ends. Black nearest the keyboard and red furthest away from the keyboard, and a third colour of wire connected to the common touch resistor track on the keyswitch. The three wires should be threaded through the relevant hole in the centre base spar, and left long enough for later connection to the power supply.

Still referring to the wiring diagram, connections should be made between the C Envelope Board and the Voice/Preamp Board—i.e. L, M and H outputs, and the common sustain connection between Voice/Preamp and the F Envelope Board.

Zero volt, +5V, and high output connections should then be made to the Tone Generator Board from the $A^{\flat}$ Board. The semitone outputs from the Tone Generator Board are wired to $A^{\flat}$, A, $B^{\flat}$, and B envelope signal input pins, whilst the remaining eight leads are passed through the centre base spar and distributed to the relevant Envelope Boards via the holes in the spar marked $E^{\flat}$–G and C–D signal input.

Colour coding of these wires is desirable, and this can be arranged in a group of three (C–D) and a group of the five ($E^{\flat}$–G).

WIRING FROM THE TOP

Before commencing top wiring the level control potentiometer, mains neon, and five control switches should be fixed in their respective positions on the front panel. Wiring can then be carried out in the following order:

1. Long leads (colour coded) should be soldered to the +19V (+V), and −23V (−V) pins on the Tone Generator Board, passed between the p.c.b. supports and keyswitch ready to connect to the Power Supply.
2. The four supply leads should be connected to the Voice/Preamp Board with similar coding for +19V and −23V as above.
3. All connections can now be made to the Power Supply Unit. The +10V supply (V/2) is obtained from the junction of C3 and IC1 in Fig. 6.

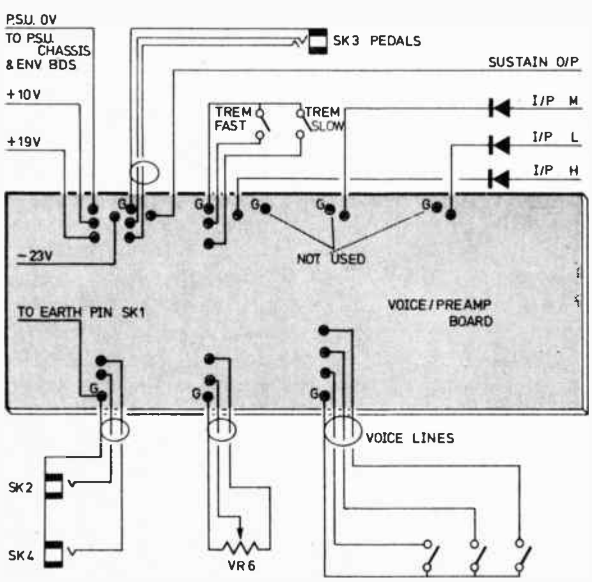


Fig. 29. Connection to Voice/Preamp Board including ground connections to minimise hum

4. Coded interconnections should be made between the Voice/Preamp Board and SK2 and 4 and SK3 on the p.s.u. chassis, preferably using 2-core coaxial lead in both cases.
5. Coded interconnections should be made to the tremolo switches and voice switches, using coaxial lead for the latter link.
6. A coaxial lead should be used to link the Voice/Preamp Board and the level potentiometer.
7. Mains switch and neon connections should be made to the Power Supply Unit.

The various links between the Voice/Preamp Board and switches and sockets are illustrated in greater detail in Fig. 29, along with the ground interconnections to avoid any hum due to earth loops.

LEG ASSEMBLY

Details of the legs are shown in Fig. 27. They are constructed from four (preferably hardwood) blocks shaped and dimensioned as indicated, interconnected with chromium plated $\frac{3}{4}$in diameter tubing. The legs are screwed to the Cabinet, into both front and centre base spars.

PEDALS

Pedal details are also given in Fig. 27. Again hardwood is used throughout.

This simple assembly, whilst fairly fragile, can stand up to sensible use, and is designed to give a neat appearance rather than for maximum strength. The rear block should be channelled or drilled to accept the bottom tube as a retainer.

The simple switches are made by gluing phosphor bronze strip under the pedals to contact with the ends of screws recessed into the base. Compression springs should be held in position by indentations in the base and pedals.

POWER AMPLIFIER

The electronic piano is a device which is generally improved by high peak power capability—one commercial manufacturer incorporated a 150W peak power rated system into his design. Completely satisfactory results have however been obtained from the system quoted above, and it is possible that a rating between 15 and 25W (sine wave) could be acceptable.

One amplifier parameter only seems to have greater than normal importance in this application and that is intermodulation distortion. This is the effect where the mixing of two notes, say at a third interval, produces a strong difference frequency signal which is detectable as a non-musically related note to the two intended notes. This is particularly noticeable in the higher registers of the keyboard where the difference frequency is a note far removed towards the lower end of the keyboard and is in the easily audible range.

It must be admitted that the author is somewhat critical on this point although it does not affect his enjoyment of the instrument, and many observers

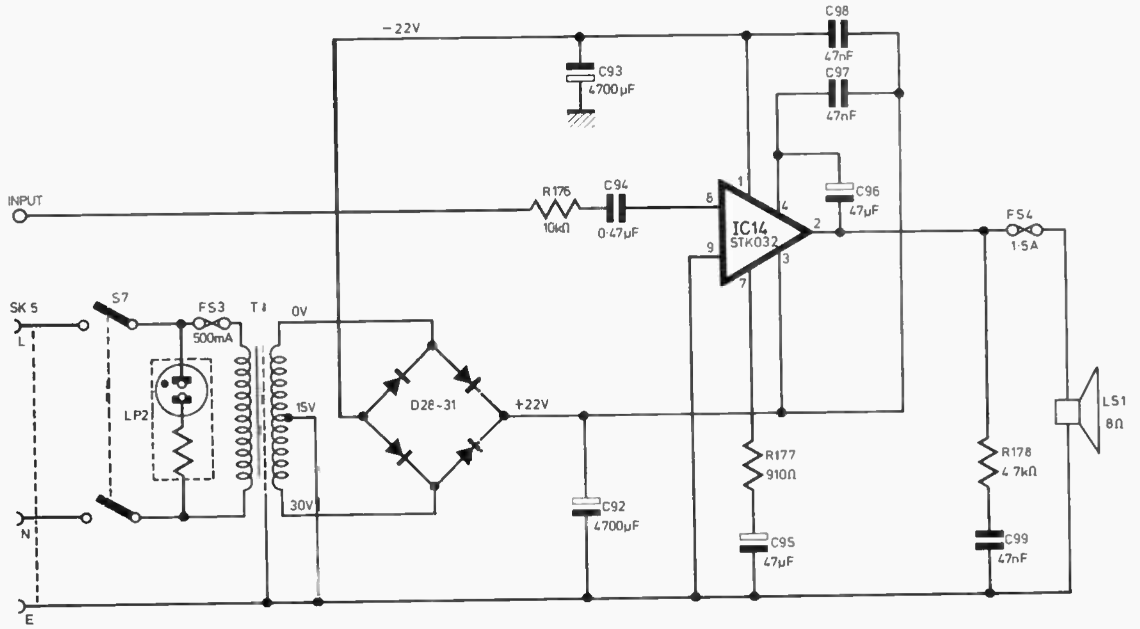


Fig. 30. Circuit of Amplifier

have judged the effect to be acceptable—it can be found on the majority of commercial organs.

Intermodulation can occur throughout any amplifier chain, and is minimised by maximising the linearity of the system. The final intermodulation culprit is the output stage of the Power Amplifier, and is related to the level of crossover distortion.

A number of amplifier designs provide adjustment of the output stage quiescent conditions to reduce crossover but the move to hybrid integrated modules removes this facility. Hybrid modules of course make for very easy construction and economy, and this must be weighed against the level of intermod which can be tolerated. The author's instrument continues to use such an amplifier and these notes are only included to make the situation clear, and not to deter the constructor from the use of these devices.

The circuit details of the amplifier and p.s.u. using a Sanyo STK032 are given in Fig. 30. The external components are mounted on a group board, and the completed unit fixed into the stool.

LOUDSPEAKERS

From the tone/voicing point of view the loudspeaker used with the "Joanna" is more important than the Power Amplifier. It is also important that the speaker has a relatively high power handling capacity, designed for use with electronic musical instruments where a rugged construction is vital.

If it is intended to use an existing hi-fi amplifier it is strongly recommended that the normal hi-fi speaker not be used since it is very likely that its capacity is insufficient to handle the peak power which can be and will be supplied by the amplifier when used in this application.

COMPONENTS . . .

AMPLIFIER
(Fig. 30)

Resistors
R176 10kΩ
R177 910Ω
R178 4·7Ω
5% ½W carbon

Capacitors
C92–C93 4,700µF elect. 25V (2 off)
C94 0·47µF
C95–C96 47µF elect. 25V (2 off)
C97–C99 47nF (3 off)

Semiconductors
IC14 STK032
D28–D31 Bridge Rectifier (50V r.m.s. 2A)

Miscellaneous
FS3 500mA fuse and panel fuseholder
FS4–1·5A fuse and panel fuseholder
LP2–Mains neon
S7–Mains switch
SK5 3 pin miniature mains socket
SK6 mono jack socket
T3–Transformer 30V centre tapped 2A
Heatsink (R.S. 149)
Groupboard
LS1–8 Ω>25W (preferably 50W) r.m.s.

Touch Resistors
R179–R214 1·5Ω 1W (36 off)
R215–R239 1·8Ω 1W (25 off)

Music Stand
Metal Frame 36in × ¼in dia. aluminium rod
Wooden crossbow 17in × ¾in × ¼in

Keyswitch
CPS 1027 (Clef Products)

Since the piano relies on very high peak power signals of short duration to produce an acceptable overall mean power, the ratio of peak to mean power required is much higher than in the average programme (music) played through any hi-fi system, and the danger is that you will increase playing volume without realising until you produce distortion. Distortion due to overloading the amplifier is not generally dangerous, but distortion due to overloading the speaker is fatal to that component.

The author has used 25W of amplifier (sine wave rating) and 50W (nominal) of heavy duty speaker over a period of five years with two designs of electronic piano, and whilst speakers of lower rating have been damaged in experiments, no deterioration has occurred in either a Fane POP50 or Goodmans 12P. This is not to say that other equivalent units are not available, but any speaker used should be designed for heavy duty work and its rating be treated conservatively.

SPEAKER VOICING

From the voicing point of view the design as presented is matched to the Goodmans 12P, mounted facing downwards in the stool. Some variation in tone quality occurs dependent on the floor covering as might be expected, but the range is acceptable. This speaker appears to apply a heavy top cut to the system, which could particularly affect the Harpsichord voice if the voice circuits were not generally bright.

Experiments have been tried with smaller 8in heavy duty units, where an improved treble response is present. In order to restore the mellow piano tone, capacitors were connected across R103, R105 and R107 with values of 47nF in each case. This restored the basic tone, finely adjusted by VR3, 4 and 5, but the body of sound in the bass associated with a 12in speaker was sadly missing.

STOOL

Cutting details for the stool used to house the Goodmans 12P are given and the introductory photo should prove adequate for constructional guidance. The enclosure volume is below that recommended by Goodmans, and the constructor may prefer to make a more conventional cabinet to house the speaker and amplifier.

FAULT FINDING

Generally in the event of a fault occurring in one note it can be easily located to one of the 61 similar circuits, and either experienced or inexperienced substitution of components attempted. Reference to the relevant theory section should assist in this.

Some possibilities and causes of malfunction are quoted—it is not intended to convey the impression that such faults should be expected to occur.

1. **Incorrect p.s.u. levels**—Any general malfunction should first be followed by a check on power supply voltages. There can be a fairly widespread on voltage levels—see pages 41, 42, but particularly a severe fall in levels can effect the operation of dividers and the master tone generator.

2. **"Frog"**—this could appear as a distinctive grumbling distortion of a note played on its own and may be present in all notes to some degree. Such an effect would be caused by a frequency modulation of the tone generation system at a mains frequency and indicates poor smoothing either in the p.s.u. (C1, 2, 4) or if only evident where a note is played it is likely to be a fault in either C5 or C6 on the tone generator board.

3. **Excessive break through**—If one or more notes break through predominantly into the background, output diodes D22 to D26 on the Envelope Boards should be suspected and changed for the note concerned. Some breakthrough will always be present and this is minimised by adjustment of VR3, 4 and 5, particularly the latter, on the Voice/Preamp Board. Maximum breakthrough will occur on the Harpsichord setting and is minimised by adjustment of VR2.

4. **Spurious sustain**—If a note sustains without the Sustain Pedal, the keyswitch should be checked for that note to ensure that it rests comfortably on the ground rod in the non playing condition. Sometimes this can be confused for breakthrough.

5. **Overload distortion.** As explained earlier, this may occur in the speaker if underrated or the Power Amplifier. Under exceptional circumstances it may occur in the preamplifier if the Power Amplifier is of very low sensitivity, VR9 (preset gain) should be adjusted to give zero distortion on the Piano stop for maximum level setting and heavy playing. Remember that two stops at once gives twice the power, and three stops gives three times the power. The level control on the front panel should be reduced when multi-stops are used.

6. **Mains hum**—If this is continuous it is likely to be due to interwiring—particularly with respect to ground connections.

7. **Intermodulation distortion**—This has been covered in detail above. Do not expect perfection in this—it is an effect normally present in a wide range of electronic musical instruments.

P.E. JOANNA POSTSCRIPT

Since the original publication of the "Joanna" project a number of minor changes have evolved which improve the overall result. Certain clarifications are also given where they may be of assistance.

SPECIFICATION *(page 34)*

Recommended power output for an accompanying amplifier is in the range 25–50 watt.

BLOCK DIAGRAM *(Fig. 1)*

The line +21V in Fig. 1 is normally referred to as +19V in the text and this level has proven fairly consistent with CT2 transformers.

This diagram also indicates the wiring of the keyswitch mounted touch resistors, the common touch busbar being supplied from the 19V rail (labelled 21V) via resistor R1 mounted on the p.s.u.

BUILDING SEQUENCE *(page 38)*

The building sequence need not include the jig procedure and thereby eliminate the need to interwire the circuit boards twice. However experience indicates that a good visual inspection is well worth carrying out after soldering the components onto the boards, particularly looking for the occasional lead which may not in fact have been soldered, or possibly for splashes of solder which have bridged two tracks.

POWER SUPPLIES *(Fig. 6)*

R1a has been added to ensure that the correct negative output voltage is obtained on load.

The 5 volt regulator IC1 is shown from the back face, i.e. the face which will contact the heatsink when mounted.

IC1 shows pins 1 and 2 reversed. Fuse FS1 should be 500mA. Alternatives for D5–D6 and IC1 are 1N4002 and µA7805 respectively.

POWER SUPPLY ASSEMBLY *(page 41)*

A simpler flat p.s.u. chassis can be made to span between the centre base spar and the rear panel shown in Fig. 27(b), with the output leads coming from the bottom of the cabinet. Dimensions of this chassis plate are 9½in × 5½in with three sides bent on ½in flanges for back fixing and side strengthening, giving a final size of 8½in × 5in.

MASTER TONE GENERATOR AND TOP C BOARD *(Fig. 10)*

The orientation of IC2 in important and is generally indicated by the square indentation shown in the top left-hand corner.

TOP C ENVELOPE SHAPER *(Fig. 11)*

Note the addition of diode D11b: this is important for both improved cut-off and beehive reduction mentioned later.

ENVELOPE BOARD *(Fig. 17)*

Similar diodes are added to D22–26 for the same reasons, and are series connected in D22–26 positions on the component layout of Fig. 18.

VOICE FILTER *(Fig. 22)*

A diode has been inserted in each input lead to the voice board in order to reduce any beehive break through to a negligible level (theoretically zero, but still somewhat dependent on tidy interwiring).

Chassis connected diodes at L, M, H inputs are now omitted.

VOICE/PREAMPLIFIER BOARD *(Fig. 23 and Fig. 26)*

R141a has been added in series with C60 as part of a noise reduction modification. R140, R141 and R166 are now 39kΩ, 4·7kΩ and 10kΩ respectively. C60 is 47nF.

TREMOLO ADJUSTMENT *(page 56)*

It is essential when setting the Tremolo that VR9 is set to a position where the gain is adjustable by small movements of the potentiometer in either direction. Both Tremolo and Soft Pedal operate by varying the control voltage on the MFC6040, the mid level of which is set up by VR9.

TONE GENERATOR BOARD TUNING LOCATION

It has been found a useful additional feature to have the tuning facility available on the front panel to quickly tune into other instruments which are off-pitch. A 10kΩ standard linear potentiometer is used. ★

GUITAR EFFECTS PEDAL

By R. GWINN

THERE are numerous guitar effects pedals available today, but there are still many areas of sound treatment in which it is possible for the amateur to produce something which is not just a copy of a commercial effect.

The Guitar Effects Pedal to be described makes use of voltage control techniques. There are two treatments, a voltage controlled amplifier and a voltage controlled filter; either of which can be selected by a switch. These are controlled by an oscillator which produces triangle, square and rising and falling ramps at controllable frequency and amplitude. The combination of four waveforms and two treatments gives eight basic effects, all of which can be considerably modified by adjustment of the controls.

WAVEFORM GENERATOR

The basic rising ramp wave is generated by IC1 and 2 (see Fig. 1). Integrator IC1 ramps upwards at a rate set by the speed control until it exceeds a limit set by comparator IC2. Then, a large reset current flows through D1 and R2 until the integrator is back to its starting point.

When the waveform switch is in the falling ramp position, IC3 acts as a unity gain inverter to give the required waveform.

In the square position, IC3 acts as a comparator. This gives a square wave of ±8V at the i.c. output, which is reduced by R14 to the same level as the other waveforms.

The triangle wave is shaped from the ramp wave by TR1. When out of saturation, this has a gain of −1. It is biased by VR2 so that for half the cycle it is saturated, when it has a gain of +1. The triangular wave at the collector of TR1 is amplified by IC3, VR3 is adjusted to offset the d.c. introduced by TR1 and its associated components.

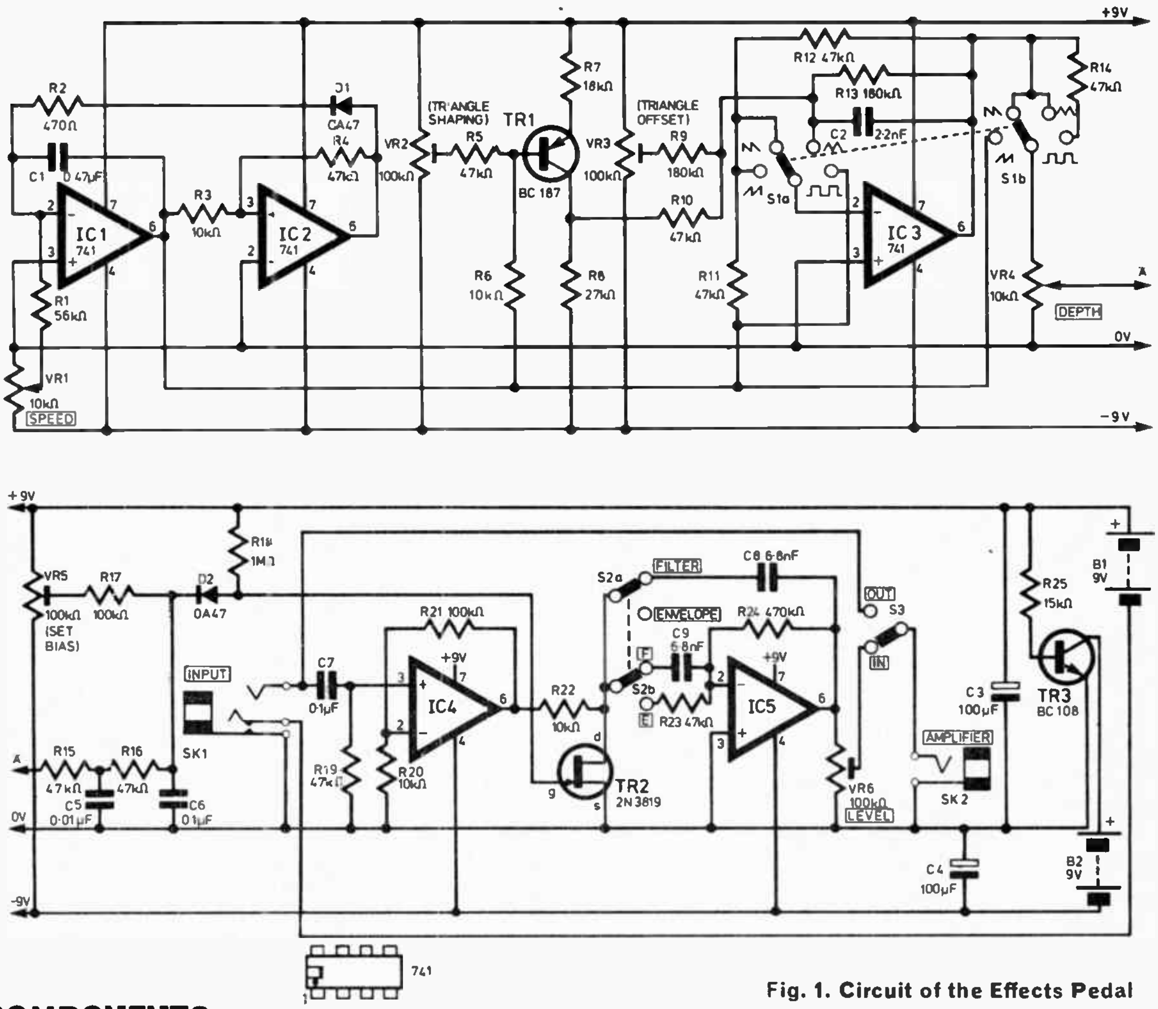


Fig. 1. Circuit of the Effects Pedal

COMPONENTS . . .

Resistors

R1	56kΩ	R14	47kΩ
R2	470Ω	R15	47kΩ
R3	10kΩ	R16	47kΩ
R4	47kΩ	R17	100kΩ
R5	47kΩ	R18	1MΩ
R6	10kΩ	R19	47kΩ
R7	18kΩ	R20	10kΩ
R8	27kΩ	R21	100kΩ
R9	180kΩ	R22	10kΩ
R10	47kΩ	R23	47kΩ
R11	47kΩ	R24	470kΩ
R12	47kΩ	R25	15kΩ
R13	180kΩ		

All ½ watt 10% carbon

Semiconductors

IC1–IC5	741 (5 off)
TR1	BC187
TR2	2N3819
TR3	BC108
D1–D2	OA47 (2 off)

Potentiometers

VR1	10kΩ log	VR4	10kΩ linear
VR2	100kΩ linear	VR5	100kΩ linear
VR3	100kΩ linear	VR6	100kΩ linear

Capacitors

C1	0·47µF
C2	2·2nF
C3–C4	100µF elect. 25V (2 off)
C5	0·01µF
C6	0·1µF
C7	0·1µF
C8–C9	6·8nF (2 off)

Miscellaneous

B1–B2 9V PP3 (2 off), S1—2 pole, 4 way switch, S2—2 pole, 2 way switch, S3—single pole changeover, SK1—jacket socket with make contacts, SK2—standard jack socket

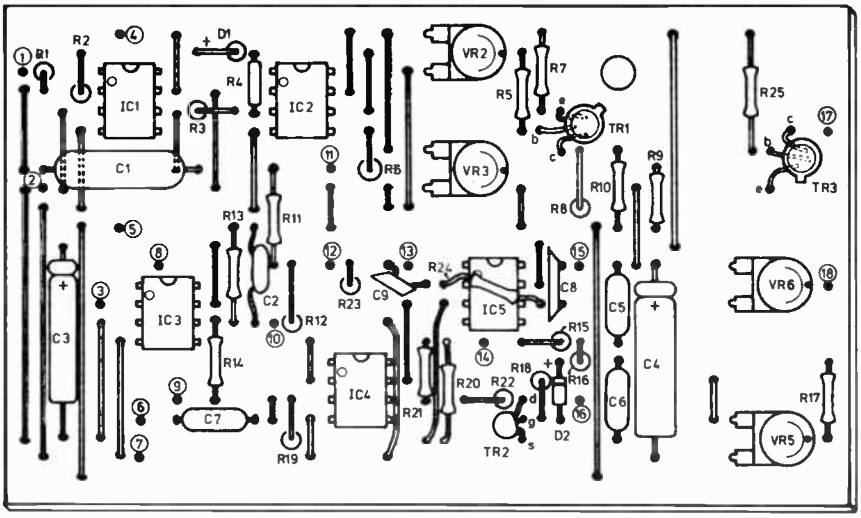


Fig. 2. Component layout and track cuts

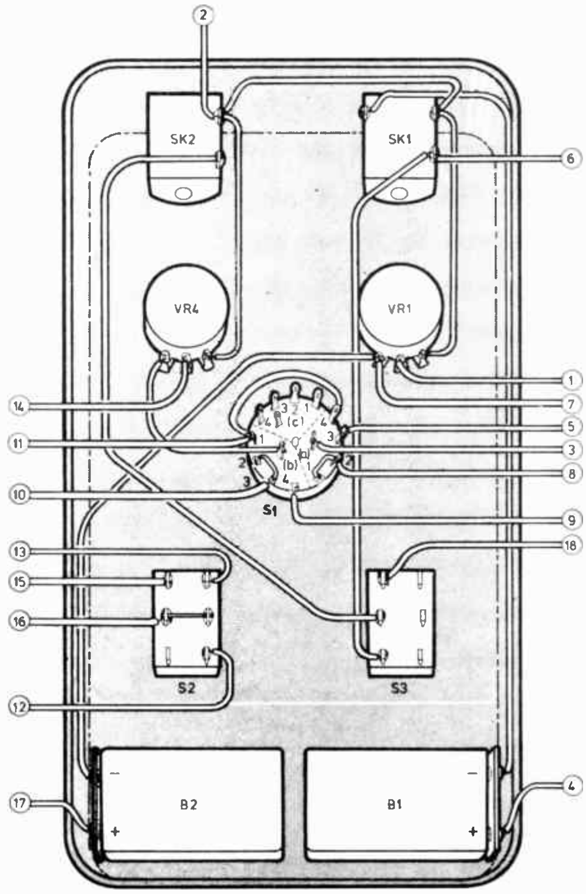


Fig. 3. Control panel wiring details

VOLTAGE CONTROLLED FILTER

When S2 is in the filter position, IC5 has multipath feedback with a minimum at a single frequency. The overall response is then bandpass peaking at that frequency, which can be changed by changing the voltage on the gate of the f.e.t.

VOLTAGE CONTROLLED AMPLIFIER

R22 and TR2 form an attenuator, and since the effective resistance of the f.e.t. can be varied by changing the gate voltage, the degree of attenuation can be changed. IC5 becomes an amplifier with a gain of 10 with S2 in the envelope position; this amplifies the previously attenuated signal.

In both the v.c.a. and the v.c.f. the f.e.t. is being used as a voltage controlled resistor. The effective resistance between the drain and source depends on the amount of negative bias on the gate. As the amount required varies from transistor to transistor, preset VR5 is included.

The control voltage from VR4 is also fed to the gate via a low-pass filter R15, R16, C5 and C6. This removes the sharp edges from the signal and so reduces the breakthrough of the control into the output.

BATTERY SWITCHING

There are two batteries to be switched on by the insertion of a jack plug to SK1. It is possible to get sockets which have a single make connection, which is used to turn on the positive supply. This turns on TR3, which then turns on the negative rail. The leakage through TR3 when it is off is negligible.

CONSTRUCTION

Most of the components are mounted on a piece of Veroboard 67mm × 112mm (Fig. 2). These are rather tightly packed as there is a lot to be fitted on. The board is screwed into a plastic bracket to hold it in place.

The unit can be housed in any convenient case, which should be earthed to prevent hum. This could be done by soldering onto the back of a pot.

The batteries are prevented from moving with a sheet of foam rubber.

SETTING UP

Turn all presets to mid-positions. While monitoring the waveform at the output of IC3, with the waveform switch set to "triangle", adjust VR2 for the best triangle wave shape. A scope is useful for this. Now set VR3 for 0V d.c. at IC3 output.

Set S2 to "filter". With the depth control at maximum, adjust VR5 for the best sound—a smooth change in filter frequency without it breaking into oscillation.

Finally set VR6 so that the volume of the treated signal is the same as in the straight through position.

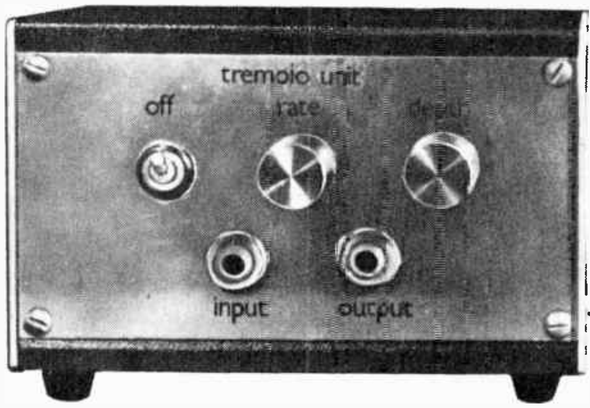
PLAYING TECHNIQUE

All the effects are repetitive, so it is best used on sustained chords or single notes. Apart from that, there are no set rules to stick to.

It will be noticed that rising and falling ramps have opposite effects on the two treatments; this is so that subjectively more interesting changes can be made simply by switching effects with one's feet. Thus a rising ramp selected on the switch will produce a decaying sound on the v.c.a.

A fast decaying ramp on the v.c.a. produces a sound like a mandolin; the same control into the filter gives a bubbling, which slows down into a repeated "Waa-Waa". A very slow triangle into the filter can be applied to any playing including fast runs.

The unit can of course be used to treat any instrument, with due attention to the matching of signal levels. ★



TREMOLO UNIT

By A. RUSSELL

Many electric guitar players will have noted the high cost of commercially available sound effects units. The Tremolo Unit described here was designed around cheap, easily available components. It is simple to build and economical with battery power and it will provide a potent tremulant effect for a guitar input with controls available for both tremolo rate and depth of sound produced.

HOW IT WORKS

In the circuit diagram of Fig. 1, the multivibrator circuit comprising TR1, TR2, switches at a rate made variable by VR1, between 1Hz and 10Hz.

As the collector of TR2 rises and falls between 0V and 8V the capacitor C3 will charge at a rate determined by the CR product of R5 and C3. As the voltage of C3 rises exponentially there comes a point when TR3 switches on. If a guitar is connected to JK1 the output to JK2 which is normally developed across VR2 and TR3 is suddenly very much reduced when TR3 does conduct. With the transistor switched off the guitar signal passes through the unit unchanged. As TR3 is being switched at a regular rate the output level will vary in depth to produce a tremolo effect.

With VR2 a variable resistor the depth of effect can be altered but there is a point when multivibrator breakthrough is slightly apparent as a ticking noise. While this is not objectionable the unit can be switched off when the guitar is not being played, although if used in a group the ticking would not normally be noticeable above the other instruments.

Increasing the value of C3 may damp this a little, but there will be a maximum above which the tremolo effect will not be satisfactory.

CONSTRUCTION

The majority of components are assembled on a 2in × 2½in piece of Veroboard as in Fig. 2. Also shown are the connections of this to the control panel.

A piece of ½in plastics angle was Araldited to the board and drilled for screw mounting to the case. For ease of operation S1 can be replaced by a footswitch connected by way of a socket at the front panel.

TESTING

When the unit is completed the wiring should be checked ensuring that the electrolytic capacitors in the multivibrator are the right way round. Should the polarity of these be reversed the multivibrator will probably operate but at the wrong frequency.

Connect the unit to the amplifier and guitar and switch on. Check the operation of the rate and depth controls. If all is satisfactory the case panels can be assembled so completing the construction.

Some loss of signal should be expected when the tremolo unit is connected and if the gain of the amplifier is not sufficient to compensate for this a preamplifier may be necessary. If so, it should be connected between the unit and the amplifier. ★

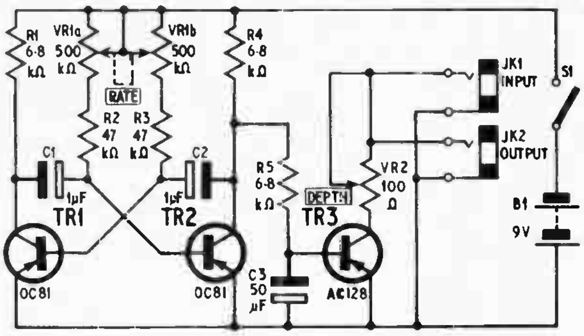
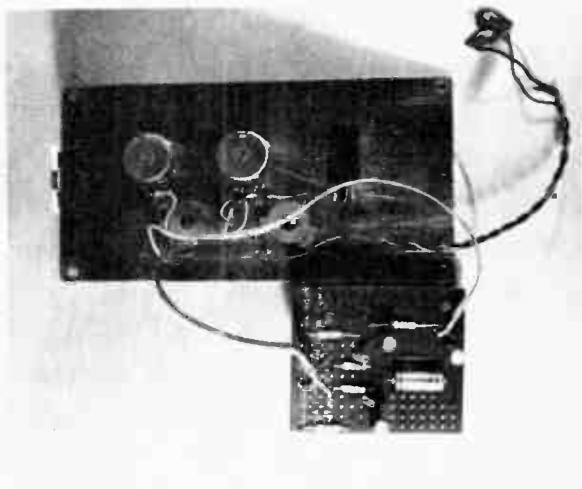


Fig. 1. Circuit diagram of Tremolo Unit



COMPONENTS . . .

Resistors
R1 6·8Ω
R2, R3 47kΩ (2 off)
R4, R5 6·8kΩ (2 off)
All ½ watt, 10% carbon

Capacitors
C1, C2 1μF elect. 12V (2 off)
C3 50μF elect. 12V

Transistors
TR1, TR2 OC81 (2 off)
TR3 AC128

Potentiometers
VR1 500kΩ dual gang carbon linear
VR2 100Ω carbon linear

Switch
S1 on/off toggle

Miscellaneous
SK1, SK2 Standard jack sockets (2 off)
BY1—PP3 9V
Battery connectors
Veroboard 0·15 matrix 2in × 2½in
Plastic angle (see text)
Instrument case 6½in × 4in × 4in
Control knobs (2 off)

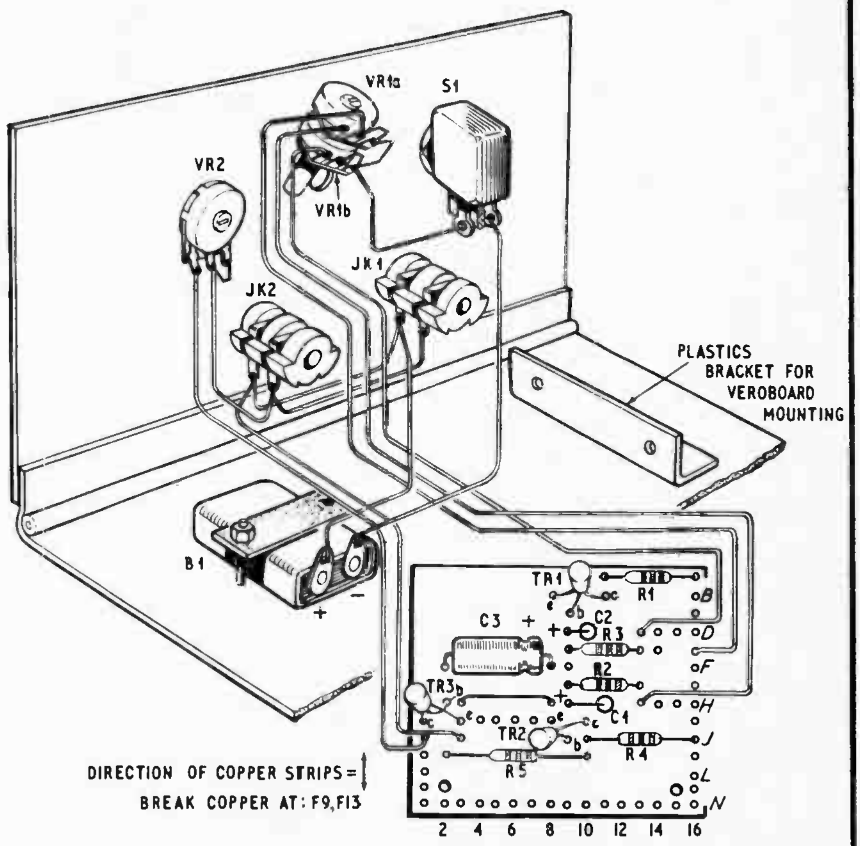
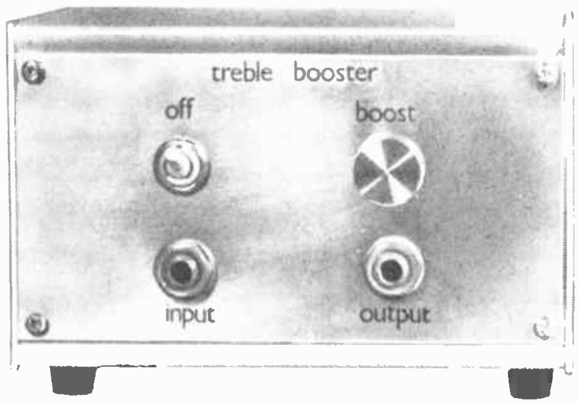


Fig. 2. Component board layout and interwiring details



TREBLE BOOSTER

By A. RUSSELL

A SIMPLE but effective unit for heightening the higher frequency harmonics of electric guitar sounds is a Treble Booster.

The actual sounds produced are akin to those made by the early "Rock 'n' Roll" guitarists, particularly when played near to the instrument bridge.

In addition to use as an effects unit the Booster will act as a straightforward pre-amplifier if required.

HOW IT WORKS

Basically the circuit (Fig. 1) consists of a simple pre-amplifier, using a low noise, high gain transistor.

In shunt with the input is an inductor L1. The impedance of this is less to low frequency signals and so the bass notes tend to be shunted to earth leaving the higher frequencies to be amplified and passed to the output socket JK2.

The "Boost" control VR1 is a 5 kilohm potentiometer in series with L1 and it controls the amount of bass cut applied to the incoming signal. When the wiper is rotated for maximum resistance there is almost no bass loss and all frequencies are amplified equally.

CONSTRUCTION

The prototype was built on a piece of 0·15in matrix Veroboard 1½in × 2½in as in Fig. 2. No breaks are required in the copper strips.

The primary of a small transistor output transformer is used for L1. The secondary winding and centre tap is not used and the leads from these should be cut short.

Control panel and Veroboard interwiring is straightforward (Fig. 2) and should present no difficulties. To prevent hum pick-up the input and output leads should be screened.

TESTING

With the unit completed, the wiring should be given a final check. With the battery connected you should find that the circuit will work first time since it is so simple.

A point to watch is the siting of the Booster. If it is placed near to the mains transformer of the amplifier, hum will be picked up by L1 so this should be avoided. ★

COMPONENTS . . .

Resistors
R1 1·5MΩ
R2 10kΩ
¼W 10% carbon

Capacitors
C1, C2 10μF elect. 25V (2 off)

Potentiometer
VR1 5kΩ lin. carbon

Inductor
L1 Eagle LT700 miniature output transformer

Transistor
TR1 BC169C

Miscellaneous
JK1, JK2 Standard jack sockets (2 off)
S1 On/off toggle switch, B1-PP3 9V battery, Battery connectors, Control knob, Veroboard 1½in × 2½in 0·15in matrix 2½in length of ½in × ½in plastics angle.

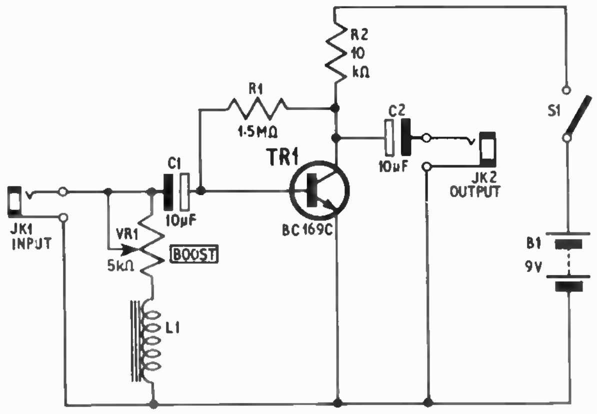


Fig. 1. Circuit diagram of Treble Booster

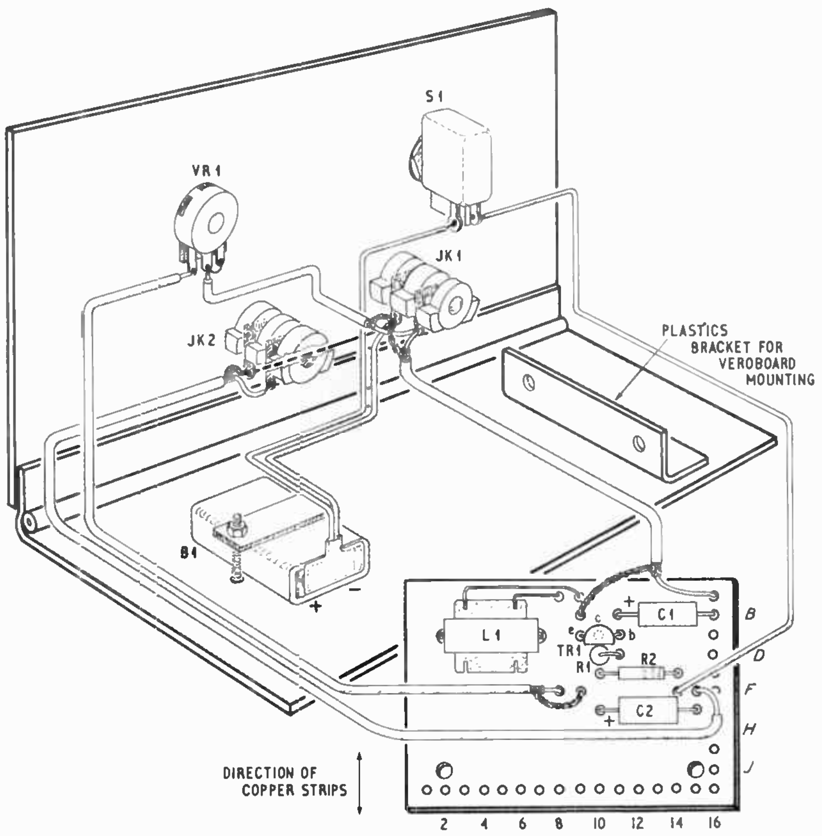
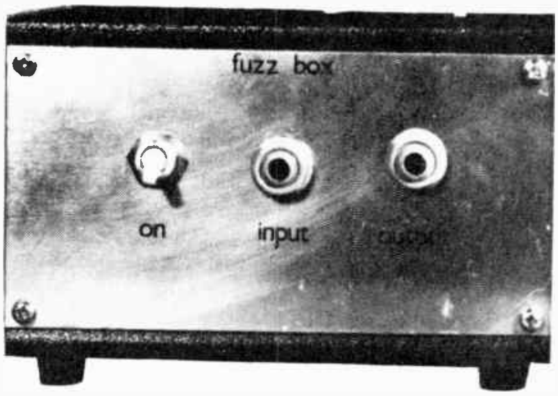


Fig. 2. Veroboard component assembly details and control panel interwiring



FUZZ BOX

By A. RUSSELL

THE effect of fuzz is to provide change in the tone produced of a guitar or other sound source so adding colour or interest to a particular musical statement.

The particular unit to be described uses cheap and readily available components and compares very favourably, both in cost and performance, with its commercial counterpart.

FUZZ PRODUCTION

In the Fuzz Unit circuit of Fig. 1 the pre-amplifier TR1 magnifies the incoming signal via the socket JK1 and this is passed in turn to a Schmitt trigger consisting of TR2 and TR3.

The action of this circuit is to amplify and square up the signal thus adding distortion to give the characteristic fuzz sound.

To protect the base/emitter junction of TR2 from reverse bias breakdown a diode, D1, is connected.

TONAL VARIATIONS

To introduce some variation in tone a low pass filter C3 is connected to the negative line from C4. The function of this is to shunt some of the higher frequency components of the square wave and so the tone of the output depends upon its value.

In the prototype a 0·22μF capacitor was used to provide a fairly mellow tone. If the value of this is decreased to 0·1μF or lower, the tone becomes harsher. Obviously, the choice here will depend on personal requirement.

CONSTRUCTION

Small circuit components are mounted and wired on a 3¾in × 2½in piece of Veroboard as shown in Fig. 2.

It should be noted that input and output leads from the control panel sockets are screened so as to prevent hum pick-up which might cause unwanted triggering of the Schmitt circuit.

THE UNIT IN USE

When using the unit, it should be borne in mind that while the guitar volume control will not affect the level of fuzz produced, if it is set too low the Schmitt will not trigger and there will be no output to the amplifier at JK2 at all.

It is possible to make a lot of unpleasant noise with a fuzz unit. This can be avoided by never playing "fuzzed" chords or over indulging in the effect in musical passages where fuzz just does not fit in. ★

COMPONENTS . . .

Resistors

R1	1·5MΩ	R6	1·5kΩ
R2	10kΩ	R7	6·8kΩ
R3	220kΩ	R8	1·5kΩ
R4	2·7kΩ	R9	15kΩ
R5	5·6Ω		

All 10%, ¼ watt carbon

Capacitors

C1 5μF elect. 25V
C2 0·1μF polyester
C3* See text
C4 5μF elect. 25V

Transistors

TR1 BC169C
TR2/TR3 2N2926 (G) (2 off)

Diodes

D1 DD000

Miscellaneous

JK1, JK2 Standard jack sockets (2 off)
S1 On/off toggle switch. Control knobs
Veroboard 3¾ × 2½in 0·15in matrix, B1-PP3
9V battery. Battery connectors

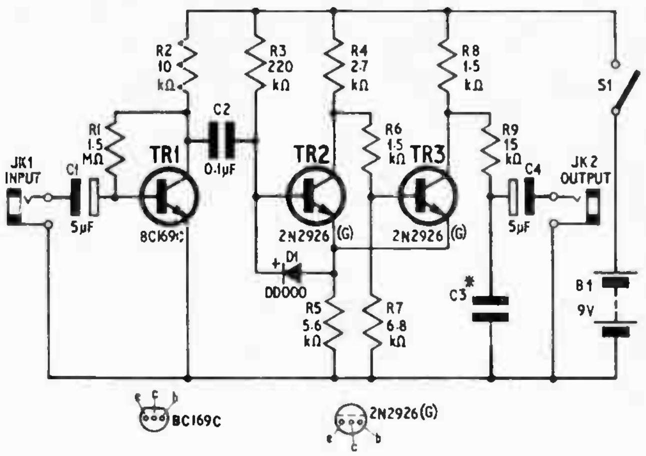


Fig. 1. Circuit diagram of Fuzz Box

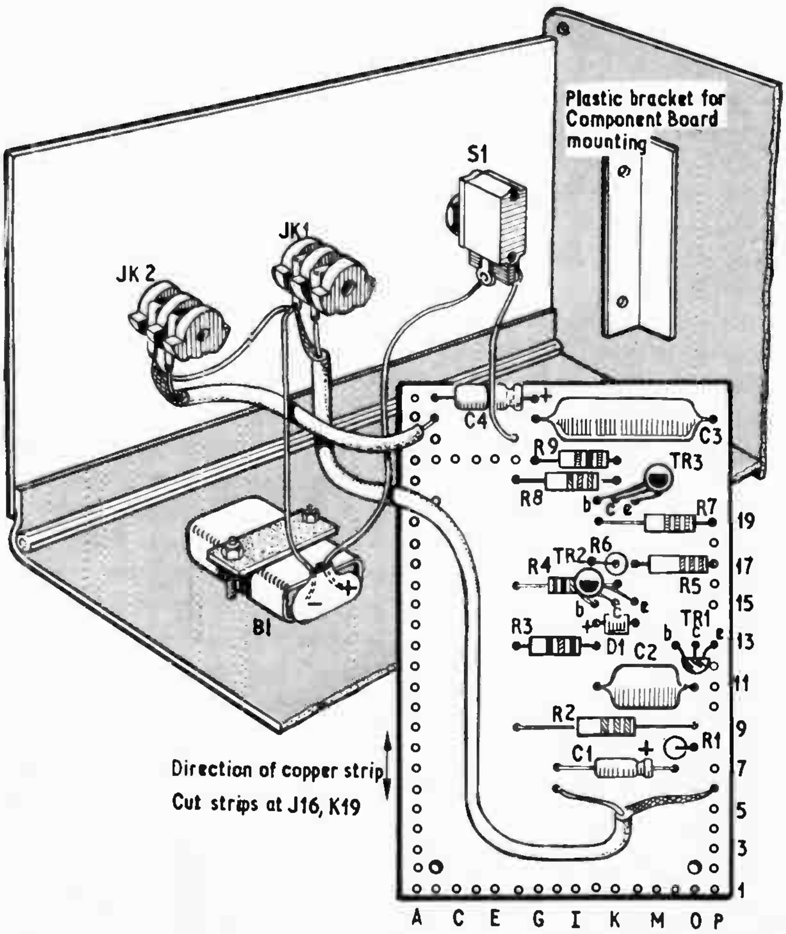
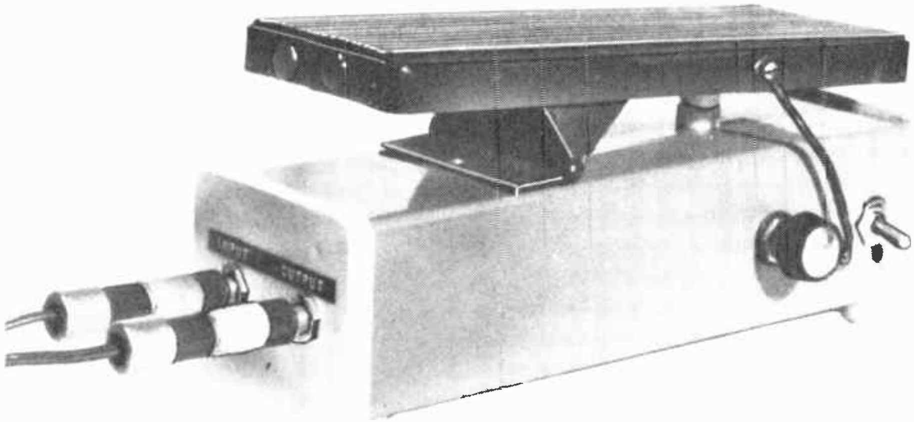


Fig. 2. Veroboard component assembly details and control panel interwiring

WAA-WAA



PEDAL UNIT

By B.H.BAILY

HARDLY a half-hour programme of "pop" music passes without the sound of the now-popular Waa-Waa effect. This extraordinary sound may lead the listener to believe that a fairly complicated circuit must be used.

Do not be deceived! The model described can be produced for a modest outlay and takes only an hour or two to build.

PRINCIPLE

The secret of the Waa-Waa lies in the use of a selective amplifier; that is to say, an amplifier which applies boost to a selected band of frequencies within the audio range, while amplifying the remaining frequencies to a lesser degree. The position of the boosted band, relative to the rest of the band, can be shifted up and down in frequency by operation of a foot pedal.

CIRCUIT DESCRIPTION

The circuit (see Fig. 1) uses only one transistor, type 2N2926, of green spot (high gain) classification. This is connected into a circuit, which, despite its unusual appearance at first glance, is basically a phase-shift oscillator, except that feedback is restricted to a value which is just insufficient to maintain self-oscillation.

COMPONENTS . . .

Resistors
R1, R2, R4, R5 47kΩ (4 off) R6 1MΩ
R3 220kΩ (see text) R7, R8, R9 56Ω (3 off)
All 10% ½W carbon

Potentiometers
VR1 2kΩ linear pre-set VR2 100kΩ log.

Capacitors
C1, C2 0·1μF plastic (2 off) C4, C5 3,300pF (2 off)
C3 0·047μF plastic C6 0·01μF plastic
All 160V polyester

Transistor
TR1 2N2926 (green spot)

Battery
BY1 9V (PP3 or equivalent)

Switches
S1 Single pole on/off toggle
S2 Single pole, press on, release off push-button

Sockets
JK1, JK2 Standard two-terminal jack sockets (2 off)

Miscellaneous
Four-way group board. p.v.c. covered wire. Wood. Wood screws. Rubber household adhesive, plastic trim beading, ribbed rubber sheeting, 64mm hinge. Plastics box, outside measurements 254mm × 64mm × 76mm, from D.E.W. Ltd., 254 Ringwood Road, Ferndown, Dorset

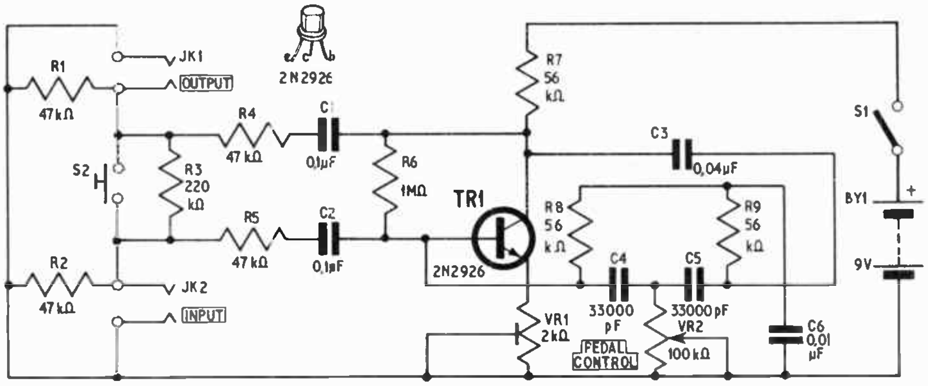


Fig. 1. Circuit diagram of the Waa-Waa pedal. R3 is adjusted on test to give minimum change in overall volume when S2 is operated

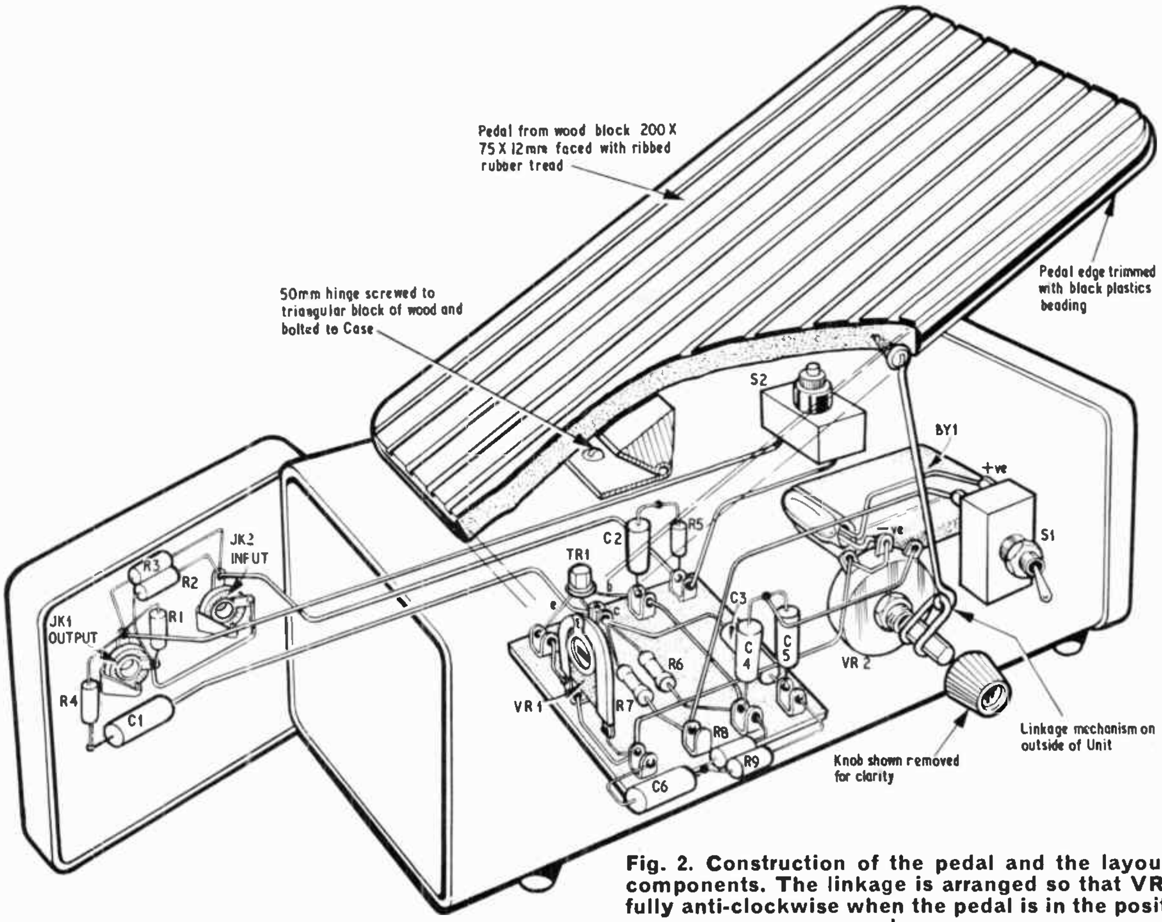
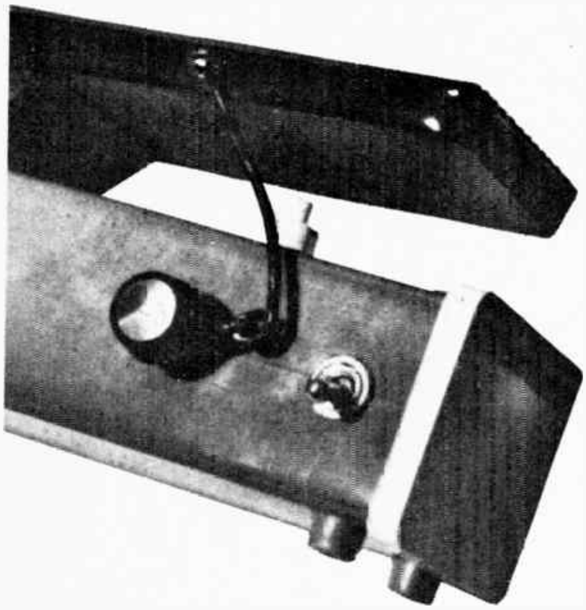


Fig. 2. Construction of the pedal and the layout of components. The linkage is arranged so that VR2 is fully anti-clockwise when the pedal is in the position shown



View of the pedal linkage. To avoid stress, the cut-out button (S2) should be positioned carefully so that it operates just before the pedal stops against the top of the case

When a signal is applied to the transistor base, the circuit behaves as a selective amplifier, and affords higher gain to all harmonics lying within a certain defined band than to those outside this band. The selective band lies between limits which are spaced on either side of the natural resonance of the circuit.

This natural frequency may be varied by changing the resistance of VR2, which is connected between the junction of C4/C5. Using the capacitor values shown, the value of this component should be variable between zero and about 50 kilohm. However, it was found necessary to use a 100 kilohm log-law potentiometer in this position, since the simple mechanical linkage allows only partial rotation of the potentiometer shaft. Hence, with the chosen component, it was found possible to get a maximum value of about 50 kilohm while having to rotate the shaft less than half its normal travel, from the fully anticlockwise position. Minimum resistance raises the boosted frequency band, whilst increasing resistance lowers the band.

BUFFER CIRCUITS

Since the input and output connections are made to the oscillatory circuit in rather a direct manner, it was found necessary to build in buffer circuits. These, while "matching" the input impedance to the more common 50 kilohm, allow for some variation in input and output matching with a minimum of variation in the performance of the circuit. The buffer resistor network is composed of R1, R2, R3, R4, and R5.

The emitter resistor VR1 is a preset potentiometer, which allows the sensitivity of the circuit to be adjusted. This control allows the feedback to be adjusted to the required near-oscillation point for optimum results.

Battery consumption is of the order of 100 microamps, which ensures many months of normal use from the PP3 battery.

CONSTRUCTION

The circuit of the prototype Waa-Waa unit was built on a four-way two-row group board. The 400V capacitors were used since space was not at a premium, but lower voltage types could be used instead to conserve space. However, avoid using the very low voltage disc-type (below 50V) capacitors in this circuit, because these often have a high leakage current and are unsuitable in the critical phase-shift circuit.

The components group board is mounted inside a case upon which is fitted the pedal. In the prototype a proprietary plastics box 154mm × 64mm × 76mm was used—see Fig. 2. However, a suitably strong case could be made from aluminium or wood, if preferred. Contact adhesive is used to fix the group board to the case.

THE PEDAL

The pedal was made from a piece of 13mm × 76mm × 203mm wood, pivoted by a hinge mounted on a short length of 25mm triangular cross-section strip.

The method of assembly should be first to screw the hinge to the triangular strip, and then screw the other half of the hinge to the box or base. Next, the pedal can be pinned and glued to the strip from above. The pedal is then ready to receive its trim. P.V.C. trim was used, and a small piece of ribbed rubber sheeting was obtained from a garage service department to give the pedal a professional and non-slip top finish. The details of the pedal construction are clearly shown in Fig. 2.

LINKAGE

The linkage from the pedal to the shaft of VR2 was fashioned from two short lengths of 10 s.w.g. galvanised fencing wire. One length of wire was formed into a crank by wrapping it around a sawn-off length of potentiometer shaft in a vice. It was then removed and pushed over VR2 shaft, and pinched on tightly with pliers. Fitting a small control knob prevented the wire coming off, while the half-flat section on the shaft prevented rotational slippage.

The other length of wire was bent to form a small loop at each end. One loop was secured under the head of a wood screw driven into the side of the pedal, and the other loop passed over the crank end, which was then doubled back to secure it.

Positions for the control VR2, the pedal pivot, and the link screw, as well as the finished linkage length, must be found by experiment since they are fairly critical. Final adjustments can be made after completion by slightly bending the crank and link to ensure that the "up" position of the pedal exactly corresponds to the fully anticlockwise position of VR2.

SETTING-UP

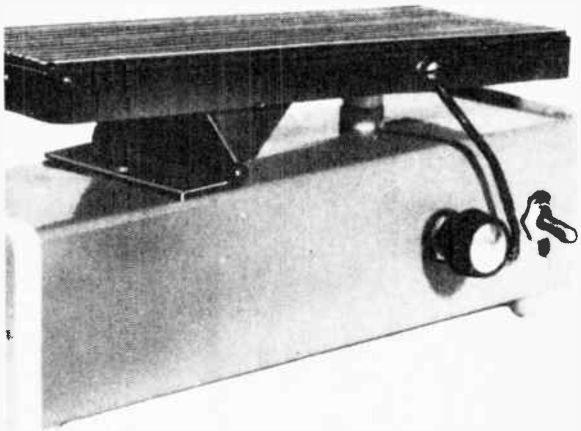
To set the position of VR1, connect the unit to the instrument and amplifier with which it will normally be used. The amplifier must be connected to the output jack, and the guitar or organ to the input jack. Connect unit to battery and switch on.

Adjust preset potentiometer VR1 to minimum resistance, and a howl should be heard from the loudspeaker of the amplifier. Back VR1 off slowly, until the howl just ceases, and rock the pedal slowly up and down over its full range. If the howl recurs at any position, turn VR1 back a fraction more. You should hear a slight *Waa-Waa* sound imposed on the background hiss, but no howl.

Ideally, VR1 should be mounted in a fairly accessible position, since it is just possible that it may require slight re-adjustment if the unit is used with other equipment. Should the Waa-Waa effect lack "life" on an instrument, it may be necessary to advance VR1 setting closer to the point of oscillation to obtain the right effect.

USE OF UNIT

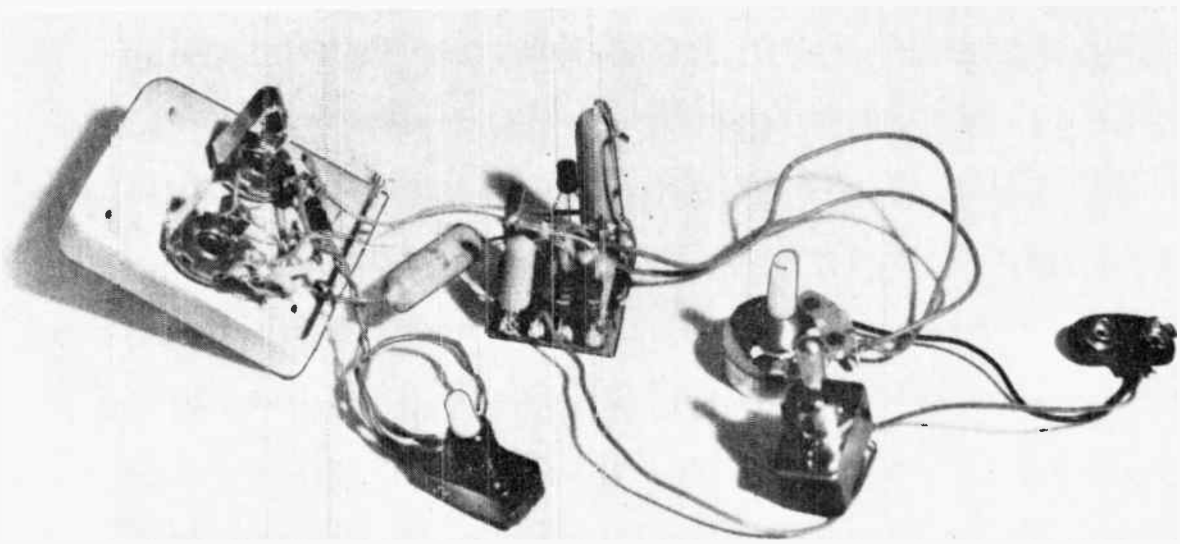
The unit may be used with any electronic musical instrument which gives an output rich in harmonics, e.g. guitar (not bass), organ, harmonica (with microphone), etc.



Details of the foot pedal hinge mounted on the triangular wooden block

The push-button S2 under the pedal allows the operator to cut out the effect completely if he desires, without having to reach down and disconnect the unit. The switch short-circuits the input direct to the output when the pedal is pushed fully down. The full range of frequencies is then passed to the output, with virtually no modification. The value of R3, nominally 220 kilohm may require to be selected carefully to ensure minimum change in overall volume when the switch is operated. ★

Completed circuit ready for insertion into the plastic case



PHASING UNIT

By D.BENFIELD

THERE is a continual demand for sound effects units which can be put together easily and cheaply by the home constructor. Indeed, it is this last feature, namely cost, which gives such devices an advantage over equivalent commercial units which are usually prohibitively expensive. To this end, the Phase Box about to be described has been made as simple as possible, consistent with an acceptable performance.

PRINCIPLE OF OPERATION

The essence of musical phase effect is to play a piece of music through two channels, with a slight time delay on one input.

To produce the required effect, the input signal is passed through a variable time delay network; the output from this is then mixed with the original signal which formed the input as shown in the block diagram of Fig. 1.

At a certain frequency, depending on the delay introduced, complete cancellation occurs during the mixing process and, by varying this frequency, the well known phasing effect is produced.

THE DELAY NETWORK

The basic circuit used to produce the required phase shift is shown in Fig. 2.

If equal amplitude sinewave inputs, of frequency f, are applied to "A" and "B", but with the input to "B" inverted with respect to "A", then the output will also be of the same amplitude, but will have a phase lag of $2\tan^{-1}(2\pi fRC)$ degrees.

Note that when $2\pi fRC = 1$, then the phase lag is 90 degrees. Thus, assuming we have suitable anti-phase driving signals at our disposal, we can cascade two such networks and, at a frequency given by $f = 1/2\pi RC$ we would get a total phase lag of 180 degrees.

Such antiphase signals are easily obtained by using a single transistor stage with equal emitter and collector resistors. Omitting biasing arrangements, we have the circuit of Fig. 3 for a single stage of our two-stage phase delay network.

The required phasing effect is obtained by varying the frequency at which cancellation occurs in the mixer stage. This is most easily altered by making R a variable component, which means a dual-gang potentiometer since two stages are being used.

COMPONENTS . . .

Resistors
R1 100kΩ
R2 47kΩ
R3 1·5kΩ
R4 1·5kΩ
R5 220Ω
R6 1kΩ
R7 1kΩ
R8 220Ω
R9 470Ω
R10 15kΩ
R11 100kΩ
R12 2·7kΩ
All resistors are ⅛W 10% carbon

Potentiometers
VR1 5kΩ log pot
VR2a, b 10kΩ + 10kΩ logarithmic dual-gang
VR3 10kΩ linear carbon preset

Capacitors
C1 6·4μF, 25V elect.
C2 6·4μF, 25V elect.
C3 6·4μF, 25V elect.
C4 6·4μF, 25V elect.
C5 0·1μF, mylar
C6 0·1μF, mylar

Transistors
TR1–TR4 BC168 (4 off)

Switch
S1 Double pole on/off toggle

Miscellaneous
JK1, JK2 Standard jack sockets (2 off)
B1 PP3 9V, battery connectors, 0·1in Vero-board Instrument case 6½in × 4in × 4in

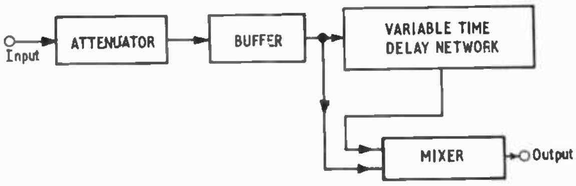


Fig. 1. Block diagram showing principle of operation

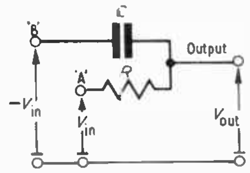


Fig. 2. The basic phase shift network employed

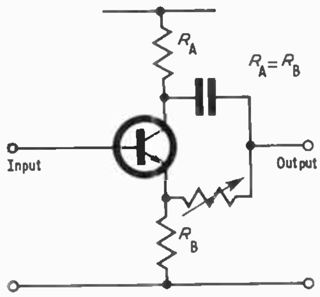


Fig. 3. Circuit for producing antiphase signals

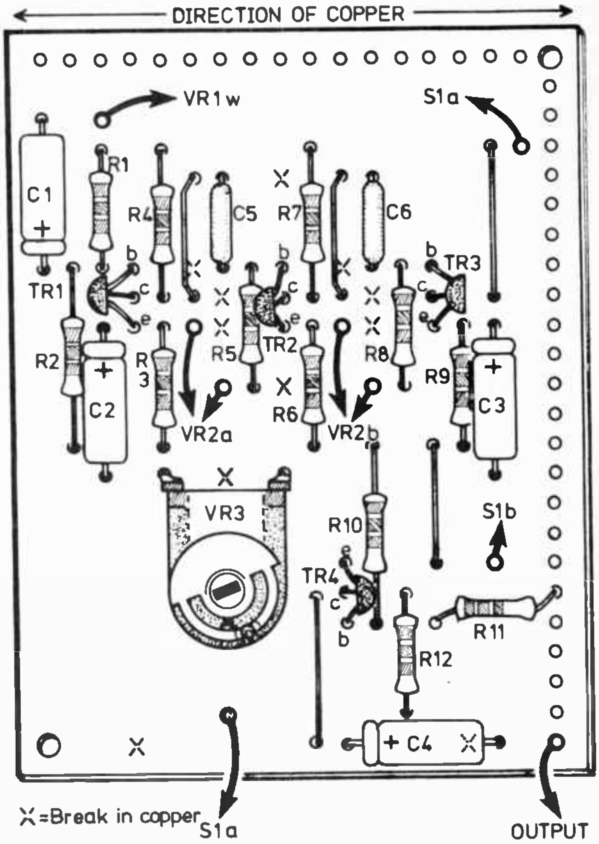


Fig. 5. Component layout and wiring on 0·1 in. matrix Veroboard

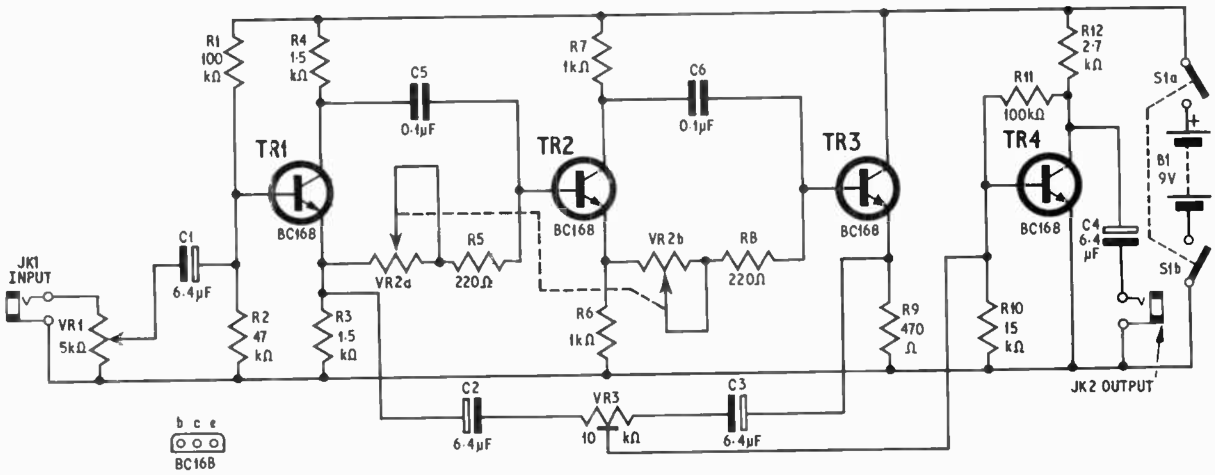


Fig. 4. Circuit diagram of Phasing Unit

FULL CIRCUIT

The phase delay is introduced by the circuitry around TR1 and TR2 as in Fig. 4. TR3 acts as an emitter follower so that a reasonably high impedance is presented to TR2 collector circuit, in order to isolate this stage from the low input resistance of the mixer.

The operating conditions for TR1 are set up by R1 and R2, while direct coupling through VR2a and R5 to TR2 base enables further bias chains and coupling capacitors to be eliminated; similar remarks apply to TR3 stage. This does mean, however, that there is a risk of unpleasant "plops" being produced should a slider of VR2 become momentarily disconnected. To overcome this, the unused ends of VR2a, b tracks should be connected to their respective sliders, and so maintaining a bias path for TR2 or TR3.

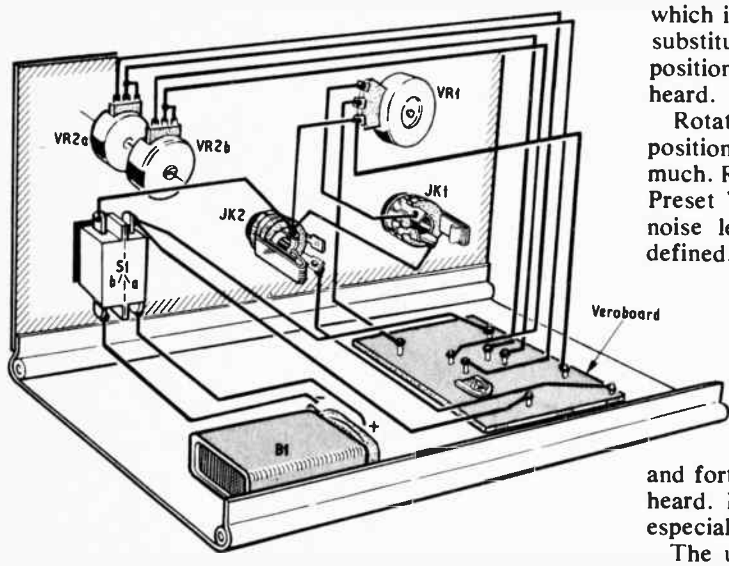


Fig. 6. Complete interwiring detail

Potentiometer VR1 provides a means of controlling the output level from the unit, by attenuating the input signal. This method is to be preferred here since it enables high level sources to be used without overloading the unit, which could occur if the level control were placed at the output.

The mixer circuit uses a single transistor, TR4. Bias conditions are maintained by the potentiometer formed by R10 and R11. By this means the collector potential is set at approximately five volts, regardless of supply voltage variations.

The inputs—direct signal and phase delayed signal—are applied to the two capacitors C2 and C3. Compensation for any difference in signal level can be provided by making either or both input resistors variable.

CONSTRUCTION

The original was assembled on a small piece of 0·1in matrix copper clad Veroboard, as this permits a neat and compact layout—see Fig. 5.

The board can now be mounted in a convenient case using 6 B.A. nuts and bolts, making sure the solder pins are well clear of the chassis by using nuts or washers as spacers.

The final wiring to input, output, VR1, VR2 and on-off switch can now be carried out; Fig. 6 shows the chassis mounted component connections.

The unit is conveniently powered by a small 9 volt battery, but it will work from any supply between 6 and 20 volts. Current drain is approximately five milliamps from a 9 volt supply.

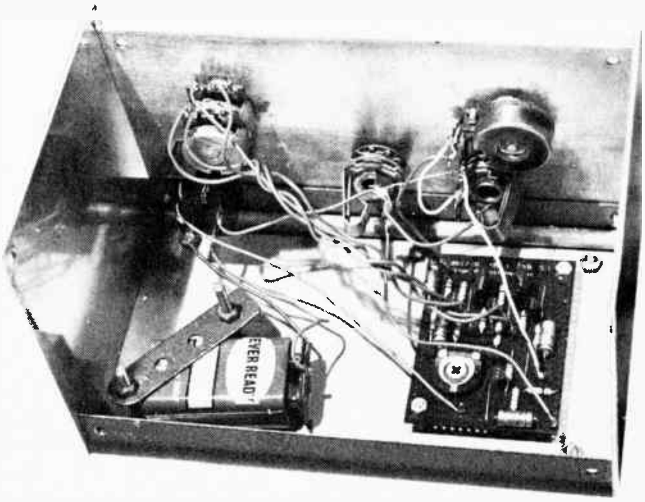
SETTING UP

Initially set VR3 to mid-position and both panel controls to their extreme counter clockwise positions. Switch on and check that current drain is about 5mA, using a suitable meter in series with the supply.

The input should now be connected to a source of "white noise". Failing this however, an f.m. receiver which is not tuned to any station makes a very good substitute. Set the level control to a convenient position whereupon the characteristic hiss should be heard.

Rotate the "phase" control VR2 to its extreme position; some sort of phasing may be heard but not much. Rotate this control back about $\frac{1}{8}$ of a revolution. Preset VR3 should now be adjusted for a minimum noise level, the correct position being fairly well defined. Upon rotating the phase control VR2 back and forth, the familiar phasing effect should now be heard. If this is not the case, check all connections, especially the links on the circuit board.

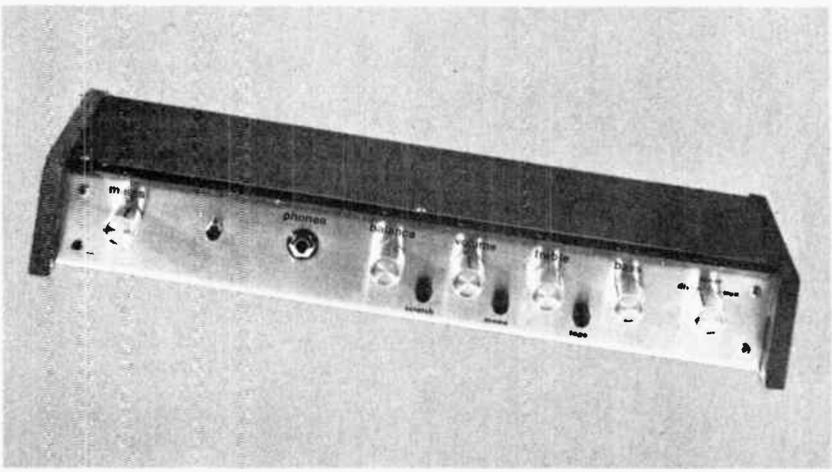
The unit may now be tried out with music input from, say, a tape recorder. The final degree of phasing heard depends largely on the content of the music used; Pop records provide a suitable starting point with their varied frequency content, and on some of these the effect can be quite startling. ★



PE ORION Hi-fi Stereo Amplifier

By D.S. GIBBS & I.M. SHAW C. Eng., M.I.E.E.

- Output: 20 + 20W into 8 ohms
- Distortion less than 0·1%
- Excellent transient response
- Will drive electrostatic loudspeakers
- Will accept inputs from radio, tape and disc
- Compact low profile case



SPECIFICATION . . .

Continuous Output Power

Load	Both channels driven	One channel driven
4 ohms	31 + 31 watts r.m.s.	44 watts r.m.s.
8 ohms	23 + 23 watts r.m.s.	30 watts r.m.s.
15 ohms	17 + 17 watts r.m.s.	19 watts r.m.s.

Measured at 1kHz

Toneburst Output Power

Load	Both channels driven	One channel driven
4 ohms	52 + 52 watts r.m.s.	57 watts r.m.s.
8 ohms	36 + 36 watts r.m.s.	42 watts r.m.s.
15 ohms	21 + 21 watts r.m.s.	22 watts r.m.s.

Measured with a 1kHz tone burst of 8 cycles on and 512 cycles off

Distortion

15 ohm load—Less than 0·1 per cent at any power level up to 15 watts between 100Hz and 10kHz. Less than 0·02 per cent below 1 watt output.

8 ohm load—Less than 0·12 per cent at any power level up to 20 watts between 100Hz and 10kHz. Less than 0·02 per cent below 1 watt output.

4 ohm load—Less than 0·5 per cent at any power level up to 30 watts between 100Hz and 10kHz. Less than 0·05 per cent below 1 watt output.

Frequency Response

Tuner and Aux. inputs { —1dB at 28Hz and 15kHz; —3dB at 17Hz and 30kHz

Tape input { —1dB at 25Hz and 30kHz; —3dB at 14Hz and 60kHz

Disc input—Within 1dB of the RIAA curve between 20Hz and 20kHz

Tone Control

Bass ±12dB at 100Hz, ±18dB at 30Hz
Treble ±12dB at 10kHz, ±16dB at 20kHz

Scratch Filter

—3dB at 5kHz. Slope 12dB per octave

Inputs

Disc —3·5mV at 47kΩ RIAA equalised
Tuner—100mV at 100kΩ Flat response
Aux. 100mV at 100kΩ Flat response
Tape 100mV at 100kΩ Flat response

Tape Output

100mV at 4·7kΩ Tape A/B facility

Signal to Noise Ratios

Unweighted figures measured with a band-width of 20kHz
Weighted figures follow CCIR C curve.
Volume control at max.

Tuner, Aux.	Unweighted —68dB,	Weighted —72dB
Disc	Unweighted —62dB,	Weighted —76dB
Tape	Unweighted —76dB,	Weighted —82dB

(Figures are relative to an output of 20 watts into 8 ohms)

MAIN AMPLIFIER ONLY (volume control at min.) { Unweighted—96dB; Weighted —100dB

Balance Control

Full rotation cuts off either channel

Dynamic Range

Disc input at 1kHz = 32dB (i.e. input of 150mV)

Interchannel Crosstalk

—50db

Stability

Unconditionally stable. Will drive electrostatic loudspeakers

Output Impedance

Less than 0·1 ohms

Dimensions

356mm × 152mm × 51mm

THIS amplifier has been designed on a value for money basis to give the highest standard of performance compatible with a small case and a low price components budget. The result is a circuit with an output of over 20 + 20 watts into 8 ohm loads at less than 0·12 per cent distortion, in a case measuring only 356mm × 152mm × 51mm.

This has been made possible by the use of a toroidal mains transformer, by the small size of the latest electrolytic capacitors and by the use of the case of the amplifier as a heatsink.

PRE-AMPLIFIER

The circuit of the pre-amplifier is shown in Fig. 1. Here TR1 and TR2 form a complementary feedback pair. This arrangement has excellent bias stability due to the d.c. feedback through R12. Both transistors are low noise types and TR1 is run at a bias current of only 150μA to minimise noise.

The equalisation components are connected in the feedback loop. R14, R15, C9 and C10 provide equalisation for magnetic pickups to within ± 1dB of the R.I.A.A. curve, between 20Hz and 20kHz, whilst R13 and C8 give a flat frequency response for the tuner and auxiliary inputs. No special equalisation has been provided for ceramic pickups as these seem to be falling out of favour nowadays, but ceramic pickups can be used with the magnetic equalisation by connecting passive matching networks of the type shown in Fig. 2 inside the record player plinth.

The frequency response of the disc, tuner, and auxiliary inputs is shown graphically in Fig. 3.

Emitter follower TR3 provides a high input impedance for the tape input and also enables a tape A/B facility to be provided. This is of particular value with tape recorders having separate recording and playback circuits as it enables one to make a direct comparison between the signal source and the recording. For example, if one wishes to make a recording of a radio programme the input selector is set to "Tuner" and the signal passes through the input stages and out to the recorder. The signal is recorded on tape and the tape playback signal appears at the emitter of TR3. By operating S2 one can then make a direct comparison between the input and the output of the tape recorder.

TONE CONTROLS

The tone control circuit is of the Baxendall type and uses an integrated circuit operational amplifier. The very high gain and large output voltage swing of this i.c. are advantageous in obtaining very low distortion and a good dynamic range, whilst the signal level in this stage is sufficiently high to make the noise negligible.

The characteristics of the bass and treble controls are shown in Fig. 4. With the tone controls flat the circuit has an overall gain of 2 and gives an output of 200mV.

The scratch filter is a second order type and gives an initial slope of 12dB per octave from its 3dB point at 5kHz. The response of the filter and its effect on the treble control is shown in Fig. 5.

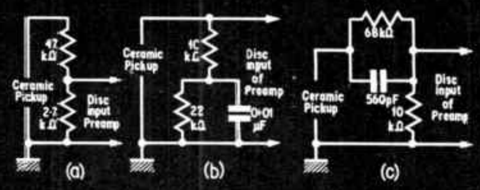


Fig. 2. (a) Circuit giving approximate matching for most types of ceramic pick up (b) circuit for matching the Decca Derem ceramic pick up (c) circuit for matching the Sonotone 9TAHC ceramic pick up

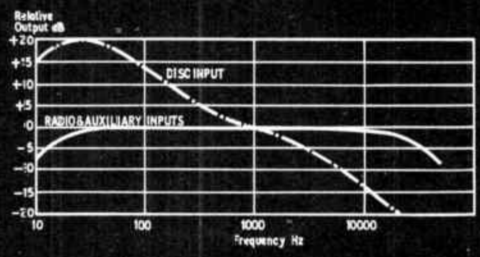


Fig. 3. Frequency response of disc, tuner and aux inputs

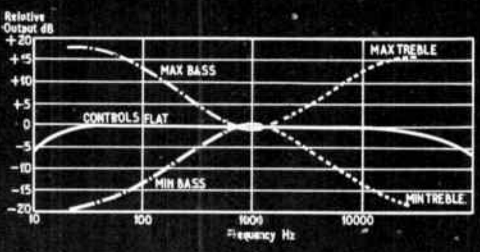


Fig. 4. Frequency response of the tone controls

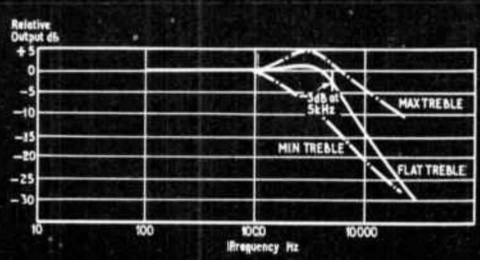
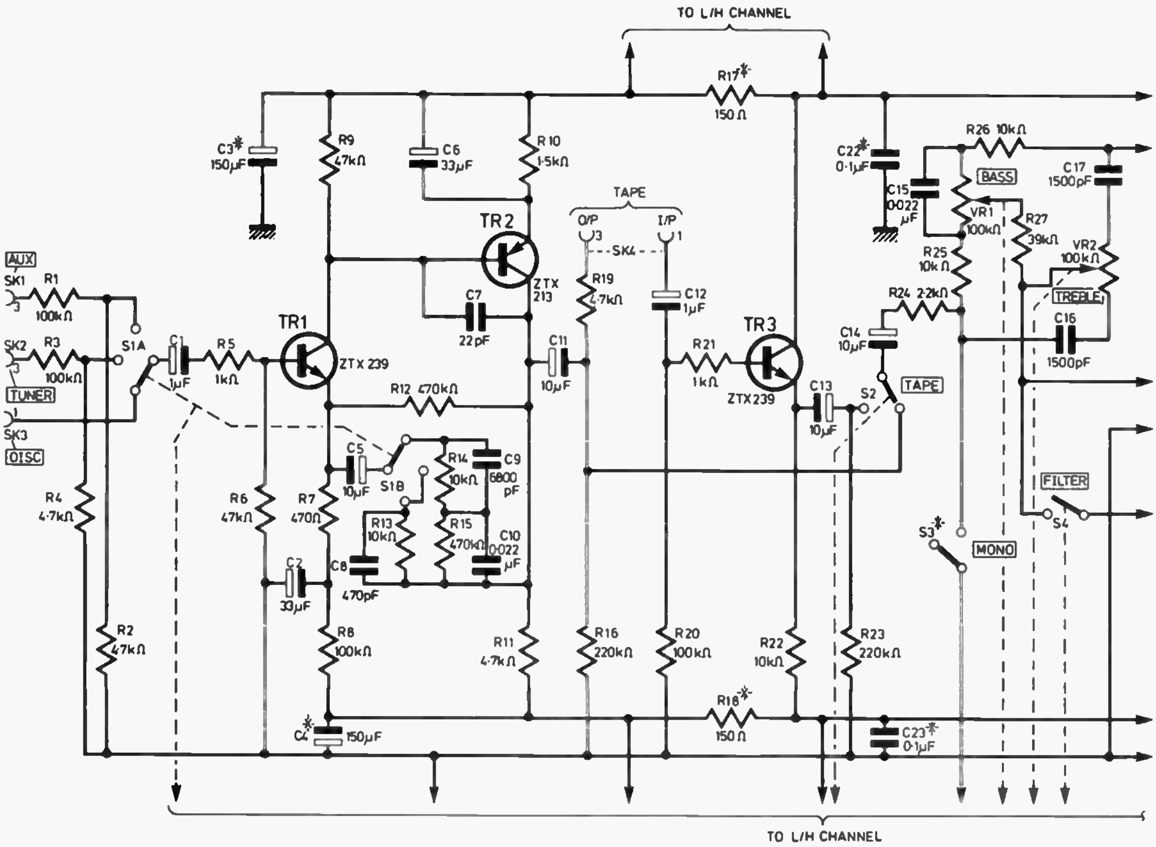


Fig. 5. Frequency response of the scratch filter



MAIN AMPLIFIER

The main amplifier has a number of interesting features.

A long tail pair has been used at the input which increases the loop gain and reduces distortion, but more important it provides an accurate ground reference for the output. The d.c. potential on the output terminal will normally be less than 50mV and this will ensure that any d.c. current through the loudspeaker is of negligible proportions. However, if an output transformer, or a loudspeaker containing a matching transformer (such as the Quad electrostatic) is used the d.c. resistance is then very much lower than the speaker impedance and it is advisable to connect a resistor of about 0·5 ohms in series with the output of the amplifier.

This will reduce the output power by a few per cent but will minimise the d.c. current. In practice the resistance of the speaker leads will often be sufficient to provide the 0·5 ohm required.

The d.c. coupled output ensures that the speaker damping is maintained right down to d.c. giving a clean solid bass response. Normally, when a speaker coupling capacitor is used, the reactance of this capacitor starts to become appreciable at low frequencies just when the most damping is required.

The d.c. coupled output is made possible by the use of balanced positive and negative supply rails, and these also assist in obtaining clean symmetrical limiting under all load conditions.

The constant current source TR8 helps in obtaining low crossover distortion by providing a rapid transition of drive current between the two output transistors. Bias transistor TR7 operates in the "amplified diode" mode and is thermally coupled to the output transistors by being clamped to the heatsink. This gives a great improvement in bias stability as any increase in heatsink temperature is compensated for by a reduction in bias.

OUTPUT TRANSISTORS

The output transistors TR9, TR10 on the circuit diagram are shown as single transistors for simplicity, but they are in fact monolithic Darlington pairs with a minimum current gain of 750 at 3 amps. These transistors have proven themselves to be electrically very robust and a 2 amp fuse in the positive rail is adequate to protect them against short circuits on the output.

Note that the fuse should be connected in the positive rail and not the negative rail. When the positive supply is removed the whole main amplifier is

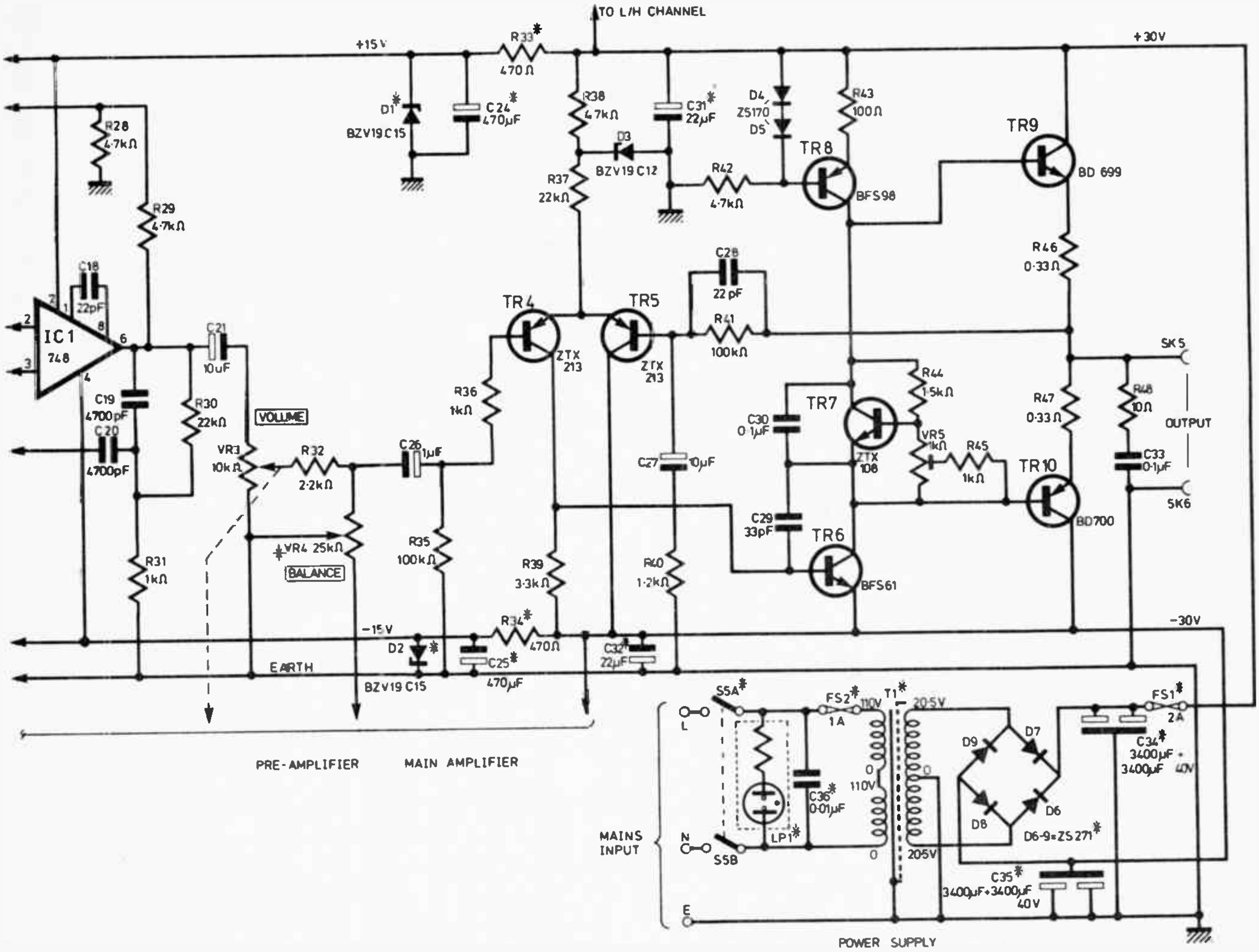


Fig. 1. Circuit of right-hand channel pre-amplifier, main amplifier and common power supply. Left hand channel is identical

turned off because the bias is removed from the constant current source TR8, and also from the input stage TR4 and TR5. This in turn turns off TR6.

No headphone listening facility has been incorporated in the main circuit diagram but a suitable attenuator network is given in Fig. 6. This circuit is connected across the output of the amplifier.

POWER SUPPLIES

When one is designing a small, low cost amplifier the first refinement that one has to do without is the use of a stabilised power supply. This introduces a number of problems, but there are compensating advantages.

The problems arise because of the lack of stabilisation. To get 20 watts output into an 8 ohm speaker we need a power supply which will deliver 45 volts on load. But when the amplifier is not giving any output and there is only a light load on the power supply, its voltage can easily rise to 60V. If we allow for mains voltage variations, then under the worst conditions the supply voltage could reach nearly 70V.

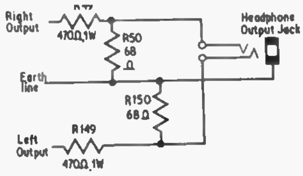


Fig. 6. Headphone attenuator circuit if required. Note that resistors have been omitted from Components List

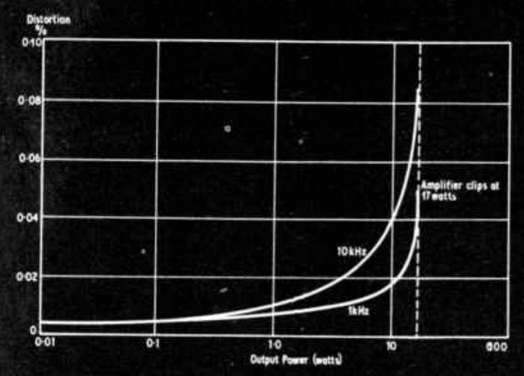


Fig. 7. Distortion against output power for 15 ohms measured at 1 and 10kHz

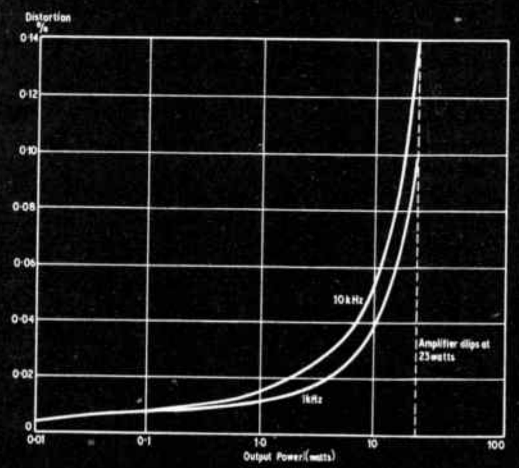


Fig. 8. Distortion against output power for 8 ohms measured at 1 and 10kHz

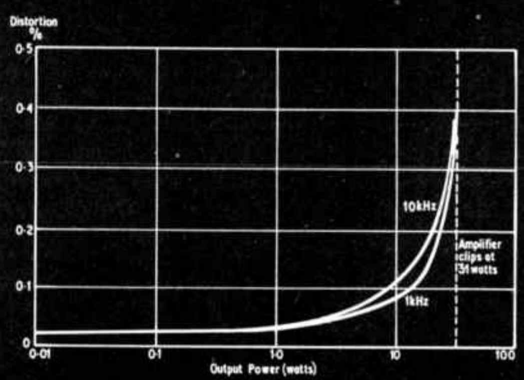


Fig. 9. Distortion against output power for 4 ohms measured at 1 and 10kHz

With a stabilised power supply output transistors rated at 50V would have been satisfactory, but with our unregulated supply we need output transistors which will stand at least 70V. Not only this but all the electrolytic capacitors need to be conservatively rated as well.

The unregulated power supply does however have one very great advantage. A musical signal has a low average power level with occasional peaks of high power. For a short period (until the power supply voltage drops) an amplifier with an unregulated supply can deliver a power output very much greater than its continuous rating. This amplifier will deliver 23 watts per channel continuously into 8 ohms, but on a musical signal it is almost as good as a 35 watt amplifier.

MAINS TRANSFORMER

A new type mains transformer has become available from R. S. Components. This has considerable advantages over the transformer originally specified. Fig. 19 and Fig. 20 give modification details to suit the new transformer.

Lastly it might be as well to clear up exactly what we mean by continuous power. With 15 ohm loads the power dissipation is sufficiently low for the amplifier to be run at full sinewave power continuously. With 8 ohm loads it will also safely run continuously provided that it is placed in a well ventilated position where air can circulate freely around it, but the back of the amplifier tends to become rather hot after about 30 minutes of full sinewave power. With 4 ohm loads the amplifier should not be run at full sinewave power for more than about 10 minutes at a time, or the temperature of the output transistors may become excessive.

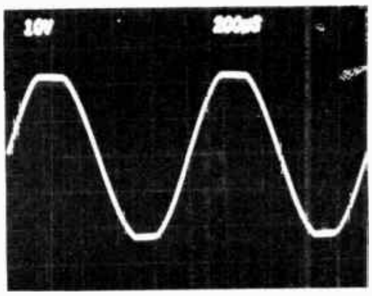
One does not normally listen to sinewaves of course and with a normal music or speech input the amplifier can be run continuously at full volume without any reservations.

DISTORTION

At full output the distortion introduced by the other components in the hi-fi system will be much greater than that of any reasonable amplifier. Moving coil loudspeakers can generate up to 10 per cent distortion and even electrostatic types can give 0·5 per cent. A good modern f.m. tuner can generate 0·5 per cent, a tape recorder about 2 per cent, and a gramophone pickup can reach as much as 20 per cent on the inner grooves. Compared to these figures the performance of all but the most mediocre amplifiers is adequate at full output.

There lies the snag. All the signal sources may have considerable distortion at full output, but the distortion falls rapidly at lower levels. This is not necessarily the case with an amplifier. If crossover distortion is present the distortion may be only 0·1 per cent at full output but may easily rise to 1 per cent or more at low levels. Crossover distortion is particularly unpleasant because it generates high order harmonics, which are discordant and easily perceived.

For crossover distortion to be negligible it should be less than 0·1 per cent at all power levels. Low order



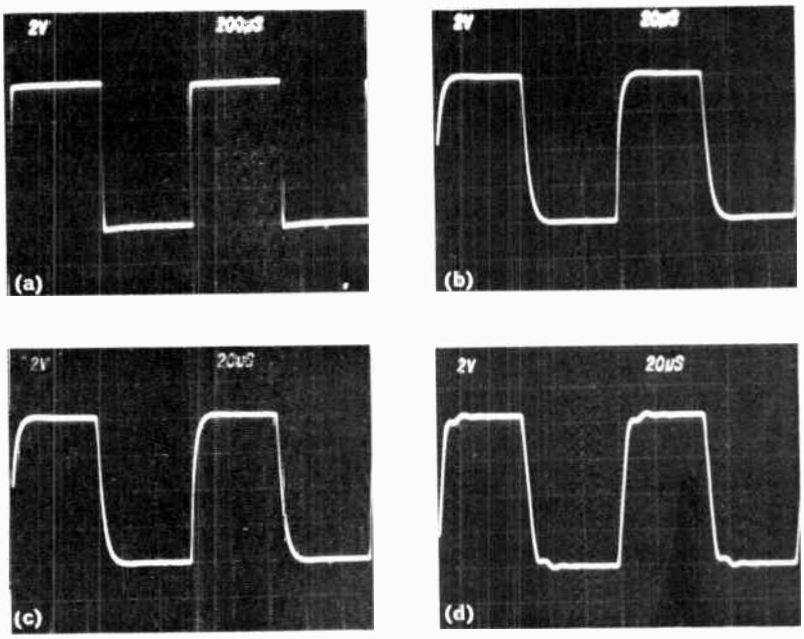


Fig. 10. Output waveform of the amplifier when slightly overdriven with a 1kHz sinewave showing the clean symmetrical limiting with freedom from latch-up

Fig. 11. (a) 1kHz square wave response with 8Ω resistive load; (b) 10kHz square wave response with 8Ω resistive load; (c) 10kHz square wave response with load of 8Ω and 0·1µF; (d) 10kHz square wave response with a load of 8Ω and 2µF

harmonic distortion is less objectionable and up to 0·5 per cent can be tolerated. So we can say that our amplifier should have a distortion specification of no worse than 0·5 per cent at full output, and below 0·1 per cent at all power levels below 1 watt.

The use of a constant current source in this amplifier has helped us to achieve a very low level of crossover distortion—typically about 0·01 per cent at 1 watt—and with 8 or 15 ohm loads the harmonic distortion is below about 0·1 per cent at all power levels up to full output. With 4 ohm loads the performance does not reach quite the same standard, but it is still below 0·1 per cent at 1 watt and 0·5 per cent at full output.

The distortion against output power at 1 and 10kz for 15, 8 and 4 ohms is given in Figs. 7, 8 and 9. The output waveform of the amplifier when slightly overdriven with a 1kHz sinewave is shown in Fig. 10.

FREQUENCY RESPONSE

Many constructors are firmly convinced that a very extended frequency response is a good thing. This is a complete fallacy because—

1. Human hearing extends from about 20Hz to 20kHz at the best. There is some evidence that transients containing harmonic components above 20kHz can be distinguished but there is certainly nothing to be gained by extending the response past 40–50kHz.

2. There are very few loudspeakers with any useful response below 30Hz or above 20kHz.

3. There are no radio signals with any audio above 15kHz.

4. There are no records or cassettes with any audio above 20kHz.

In fact the only audio signal available which might have anything above 20kHz would be a very high quality tape recording of a live performance.

A very extended frequency response can be a very bad thing. If the low frequency response is very extended then low frequency noise from turntable rumble, warped or off-centre records, or tape recordings can get through the system and cause the speaker cone to flutter violently. If the h.f. response is very extended then h.f. noise and multiplex sideband components can intermodulate with each other and with h.f. audio signals to produce audible distortion and noise. So what is the ideal response? Probably something like 20Hz to 50kHz ±3dB.

This amplifier has been designed so that the frequency response falls rapidly below 10Hz, and so a separate rumble filter is not necessary. The h.f. response of the tape input extends to 50kHz but the radio input has been restricted to 30kHz to attenuate multiplex and carrier components. With the scratch filter switched in all inputs are 3dB down at 5kHz.

TRANSIENT RESPONSE

For good reproduction it is essential that the amplifier should have a good transient response with as little ringing as possible, even when fed into a highly reactive load such as an electrostatic loudspeaker.

In this amplifier a particular effort has been made to achieve a good transient response and Fig. 11 shows the performance of the amplifier under various load conditions. Note that with a 2µF capacitive load the ringing is of very low amplitude and soon dies away. If a 0·5 ohm resistor is connected in series with a 2µF load (as we recommended for the Quad speaker), then even this small amount of ringing is completely eradicated.

COMPONENTS . . .

Resistors

R1, R101	100kΩ	R25, R125	10kΩ
R2, R102	4·7kΩ	R26, R126	10kΩ
R3, R103	100kΩ	R27, R127	39kΩ
R4, R104	4·7kΩ	R28, R128	4·7Ω
R5, R105	1kΩ	R29, R129	4·7kΩ
R6, R106	47kΩ	R30, R130	22kΩ
R7, R107	470Ω	R31, R131	1kΩ
R8, R108	100kΩ	R32, R132	2·2kΩ
R9, R109	47kΩ	R33*	470Ω 1W
R10, R110	1·5kΩ	R34*	470Ω 1W
R11, R111	4·7kΩ	R35, R135	100kΩ
R12, R112	470kΩ	R36, R136	1kΩ
R13, R113	10kΩ	R37, R137	22kΩ
R14, R114	10kΩ	R38, R138	4·7kΩ
R15, R115	470kΩ	R39, R139	3·3kΩ
R16, R116	220kΩ	R40, R140	1·2kΩ
R17*	150Ω	R41, R141	100kΩ
R18*	150Ω	R42, R142	4·7kΩ
R19, R119	4·7kΩ	R43, R143	100Ω
R20, R120	100kΩ	R44, R144	1·5kΩ
R21, R121	1kΩ	R45, R145	1kΩ
R22, R122	10kΩ	R46, R146	0·33Ω 2·5W
R23, R123	220kΩ	R47, R147	0·33Ω 2·5W
R24, R124	2.2kΩ	R48, R148	10Ω 1W

All ½W 5% carbon film unless otherwise rated

Note: R17*, R18*, R33* and R34* are common to both channels; similarly for all other components asterisked

Potentiometers

VR1, VR101	100kΩ twin gang linear law (RS)
VR2, VR102	100kΩ twin gang linear law (RS)
VR3, VR103	10kΩ twin gang log law (RS)
VR4*	25kΩ single gang linear law (RS)
VR5, VR105	1kΩ skeleton preset (RS)

Capacitors

C1, C101	1μF 35V tantalum
C2, C102	33μF 16V elect.
C3*	150μF 16V elect.
C4*	150μF 16V elect.
C5, C105	10μF 25V tantalum
C6, C106	33μF 16V elect.
C7, C107	22pF 160V polystyrene
C8, C108	470pF 160V polystyrene
C9, C109	6,800pF 400V polyester
C10, C110	·022μF 160V polyester
C11, C111	10μF 25V tantalum
C12, C112	1μF 35V tantalum
C13, C113	10μF 25V tantalum
C14, C114	10μF 25V tantalum
C15, C115	·022μF 160V polyester
C16, C116	1,500pF 160V polystyrene
C17, C117	1,500pF 160V polystyrene
C18, C118	22pF 160V polystyrene
C19, C119	4,700pF 400V polyester
C20, C120	4,700pF 400V polyester
C21, C121	10μF 25V tantalum
C22*	0·1μF 30V disc
C23*	0·1μF 30V disc
C24*	470μF 25V elect.
C25*	470μF 25V elect.
C26, C126	1μF 35V tantalum
C27, C127	10μF 25V elect.
C28, C128	22pF 160V polystyrene
C29, C129	33pF 160V polystyrene
C30, C130	0·1μF 250V polyester
C31, C131	22μF 63V elect.
C32, C132	22μF 63V elect.
C33, C133	0·1μF 250V polyester
C34*	3,400 + 3,400μF 40V elect.
C35*	3,400 + 3,400μF 40V elect.
C36*	0·01μF 750V disc

Semiconductors

	Recommended type	*Alternatives*
TR1, TR101	ZTX239 Ferranti	ZTX384, ZTX109C
TR2, TR102	ZTX213 Ferranti	ZTX214, ZTX530, ZTX502
TR3, TR103	ZTX239 Ferranti	ZTX384, ZTX109C
TR4, TR104	ZTX213 Ferranti	ZTX214, ZTX502, ZTX531
TR5, TR105	ZTX213 Ferranti	ZTX214, ZTX502, ZTX531
TR6, TR106	BFS61 Ferranti	ZTX451
TR7, TR107	ZTX108 Ferranti	ZTX238, ZTX239, ZTX109
TR8, TR108	BFS98 Ferranti	ZTX551
TR9, TR109	BD699 Motorola	BD699A
TR110, TR110	BD700 Motorola	BD700A
D1*, D2*	BZV19 C15 Ferranti	
D3, D103	BZV19 C12 Ferranti	
D4, D104	ZS170 Ferranti	Any ZS170/270 series diode
D5, D105	ZS170 Ferranti	Any ZS170/270 series diode
D6*, D7*, D8*, D9*	ZS271 Ferranti	ZS272 ZS274, ZS276 or ZS278
IC1, IC101	UA748CV Signetics	Texas SN72748P

Switches

S1A, S1B, S101A, S101B 4-pole 3-way rotary (Lorlin)

S2, S102	2-pole changeover pushbutton (RS)
S3*	2-pole changeover pushbutton (RS)
S4, S104	2-pole changeover pushbutton (RS)
S5	D.p.s.t. rotary mains switch (RS)

Miscellaneous

T1 Gardners mains transformer type SL8, 20·5-0-20·5 volts *or* R. S. Components 50VA toroid (20V) stock no. 207431 (Doram Ltd)

LP1 Neon panel lamp with internal resistor

Case—H.M. Electronics type GB1

Stereo jack socket, 4 DIN 5-way sockets, four 4mm sockets, five control knobs, three push-button switch buttons (RS), two Eagle 20mm fuseholders, 1A fuse, 2A fuse, four Lektrokit spring clips type LK2791 (1·5in), five way tag-strip, screws, spacers, grommets, aluminium angle, connecting wire.

QUADRAPHONICS

No special provision has been made for quadraphonics in the prototype, mainly because of lack of space, but it is a simple matter for the enthusiast to adapt the circuit for use with a quadraphonic decoder.

All that is necessary is to break the connection between the preamplifier and the main amplifier and replace it with a switch and a DIN input socket, as shown in Fig. 12. The two front channels can then be fed back into the Orion main amplifiers whilst the back channels are fed to another amplifier. An additional pair of Orion main amplifiers would be ideal for this purpose.

COMPONENTS

An effort has been made to use standard, easily obtainable components as far as possible, most of which can be obtained through firms advertising in *Practical Electronics*.

Components which require special attention are the 0·33 ohm wirewound resistors. These should be low inductance types such as the 0·33 ohm, 2·5W from R.S. Components (Doram), the Welwyn W21 3W or C.G.S. C3A 3W. No other types should be used.

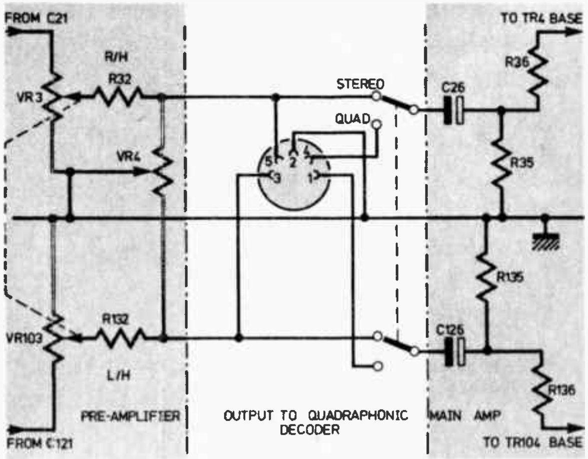
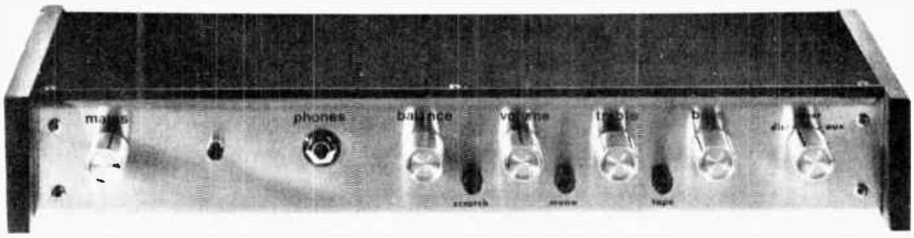


Fig. 12. Connections for quadraphonic decoder. DIN socket connections are as follows:—
1. **Left channel front input from decoder**
2. **Earth**
3. **Left channel output to decoder**
4. **Right channel front input from decoder**
5. **Right channel output to decoder**



CONSTRUCTION

It is best to start by assembling the printed circuit board. The layout of this is shown in Fig. 13. The three push-button switches should be mounted first, followed by all the resistors and capacitors. Take care with the polarity of the electrolytic and tantalum bead types. Some tantalum capacitors are now marked with the polarity, but the colour coded type are not. The polarity of these can be deduced from Fig. 14.

All the semiconductors can now be soldered in place. Take care to connect the E-Line transistors the right way round.

The four fixing screws for the output transistors must be nylon, and these screws also hold the heatsink to the board the heatsink is a 153mm length of heavy gauge 25mm × 25mm × 3mm aluminium angle, Fig. 16. Mica washers should be used under all the output transistors and a thin smear of silicone grease or heatsink compound is beneficial in reducing the thermal resistance.

The bias transistors TR7 and TR107 are held down on to the heatsink by 6BA screws and a thin strip of metal to give good thermal contact. Care should be taken not to overtighten these screws or damage to the transistors may result. The screws should be just sufficiently tight to hold the bias transistors firmly against the heatsink, and can be locked with a blob of paint if desired. A small smear of silicone grease under the bias transistor helps in achieving good thermal contact.

Finally, all the connecting wires required should be soldered into the board, and the board checked carefully for mistakes and for solder "bridges" between adjacent tracks.

DRILLING

The main chassis should be drilled as shown in Fig. 15 and all the holes carefully deburred. Take special care with the three holes for the push-button switches as these need to be accurately placed and will remain visible after assembly. Errors or careless drilling will spoil the appearance of the finished amplifier.

A slight modification is necessary for the four DIN input sockets to be fitted to the back of the case. The left hand side of both the back panel and the back of the cover are removed and the hole created is filled with a 51mm wide strip of aluminium sheet. This strip is clamped between the heatsink and the back of the case of the amplifier (Fig. 17).

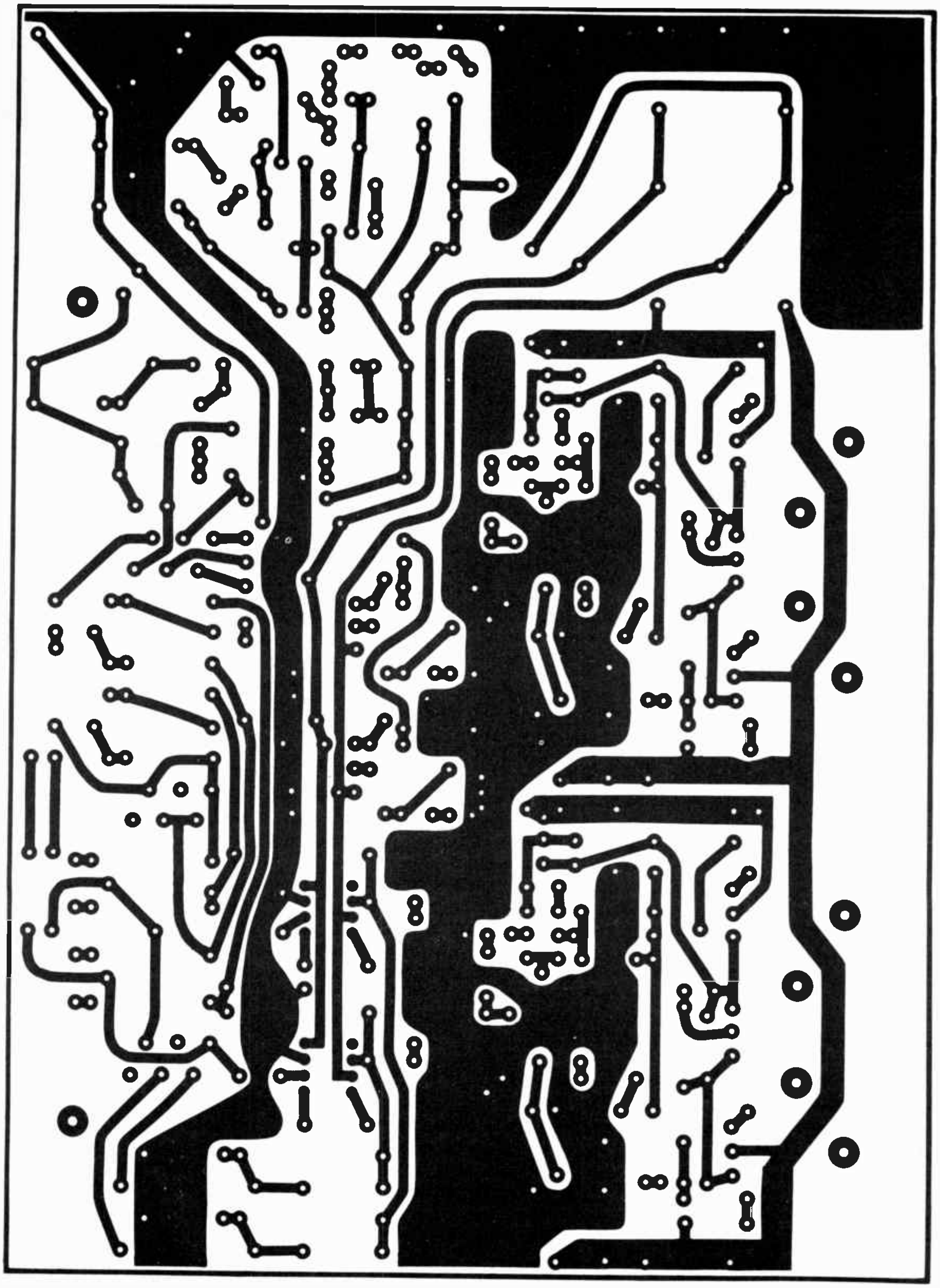


Fig. 13. Same size layout of printed circuit board

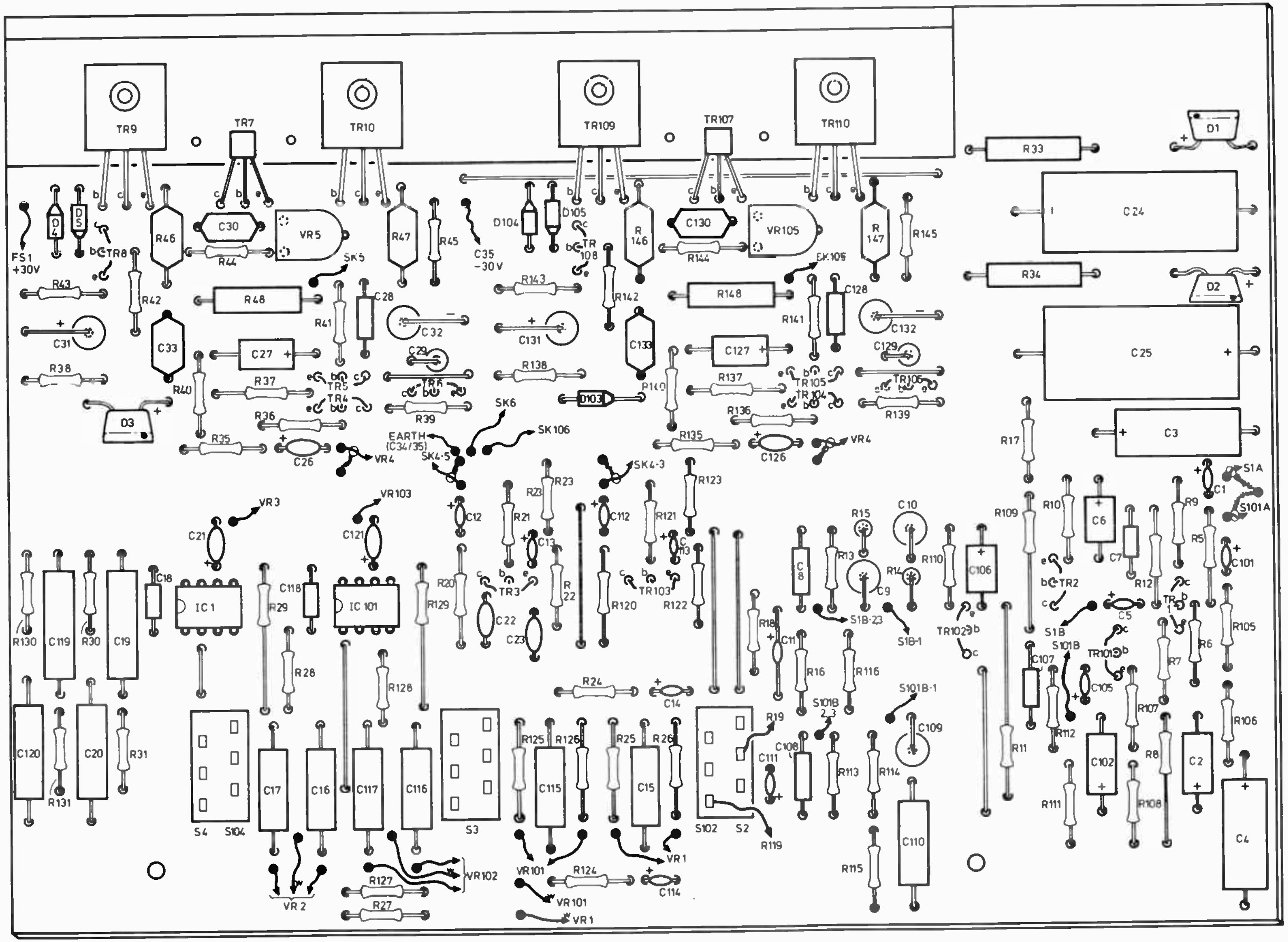


Fig. 14. Component layout of printed circuit board

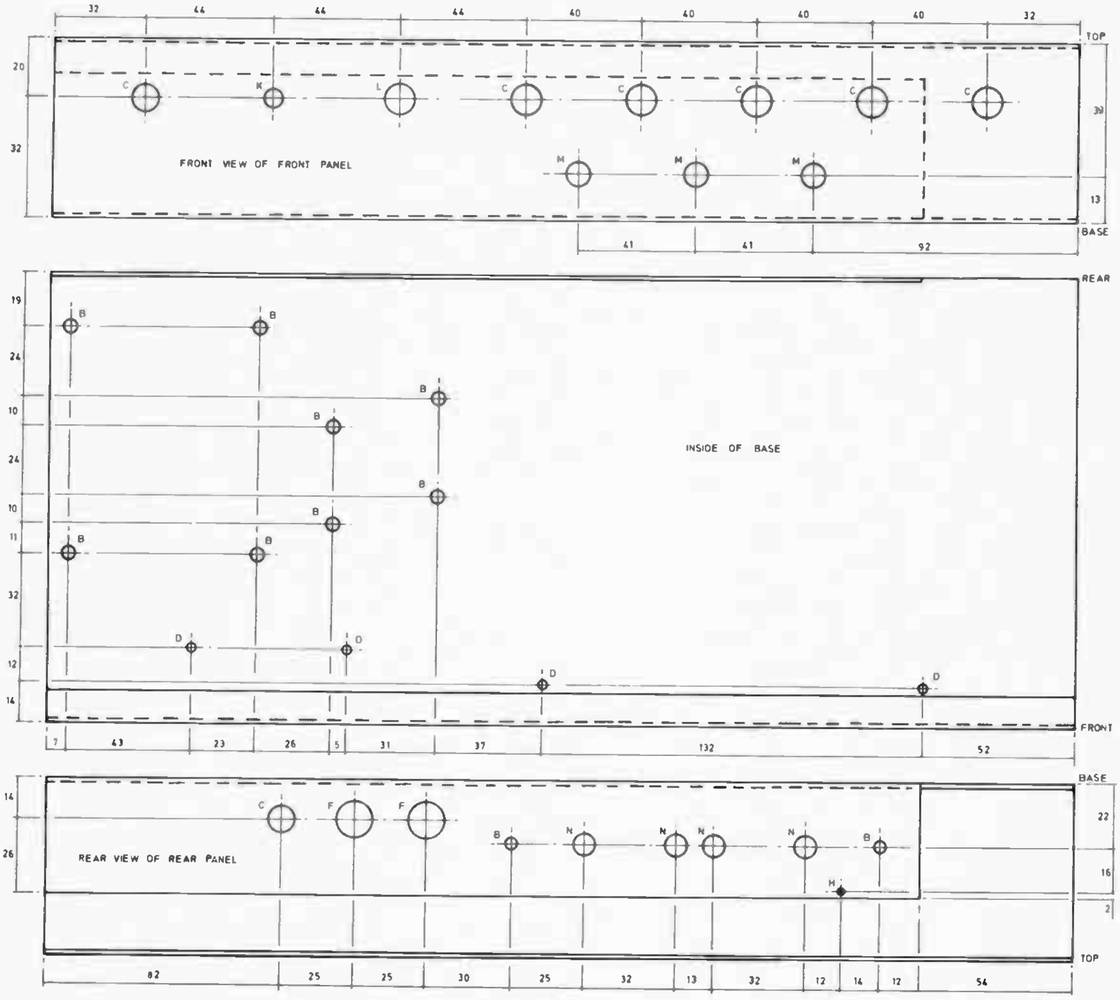


Fig. 15. Drilling details of main chassis

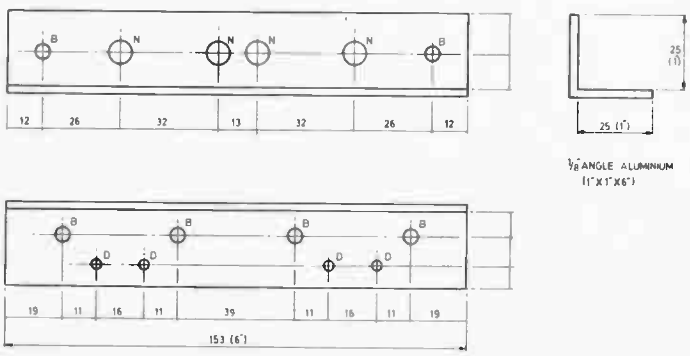


Fig. 16. Drilling details for the output transistor heatsink. "N" holes should be drilled using the mounting plate of Fig. 2.5 as a template

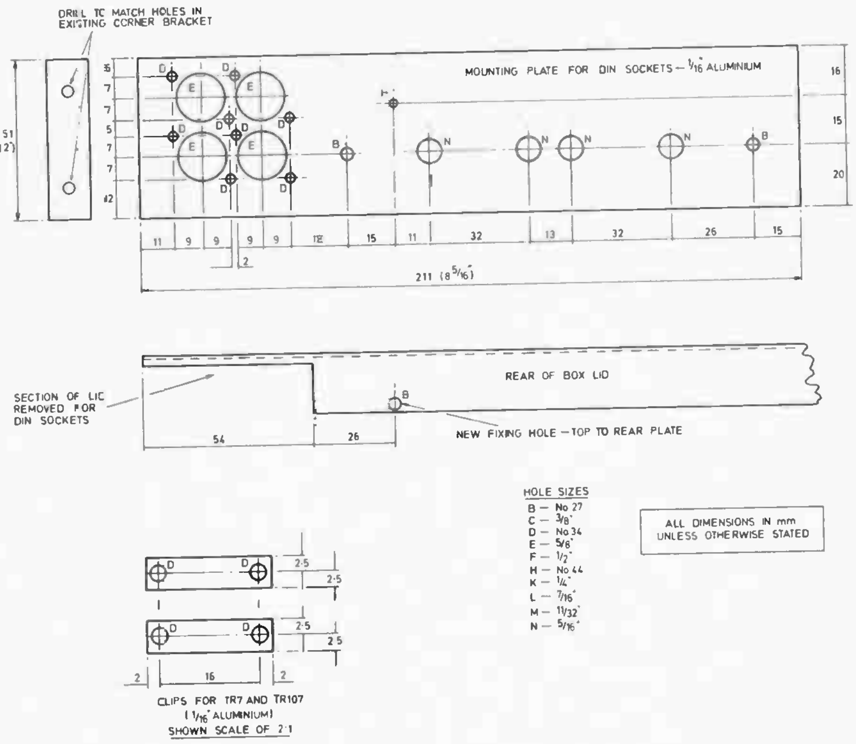


Fig. 17. Drilling details for DIN socket mounting plate, lid and small heatsinks for TR7 and TR107

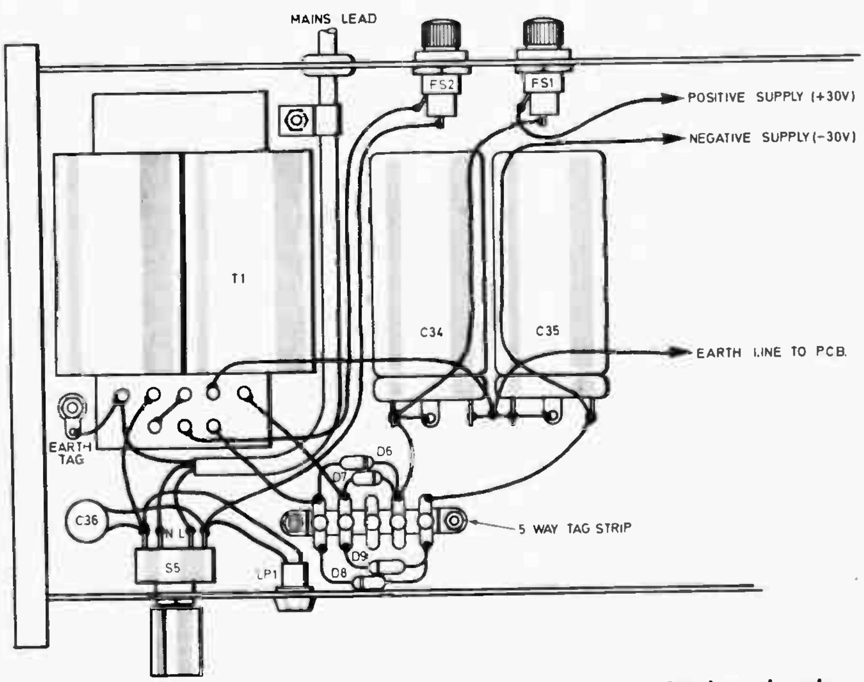
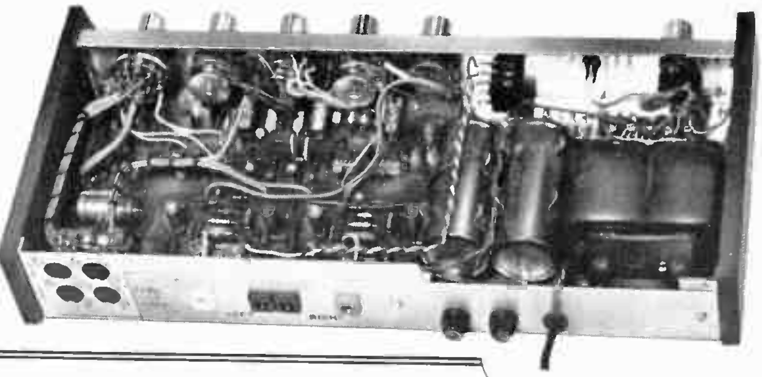


Fig. 18. Component layout for the power unit. The two hardwood end-cheeks are 7in × 2¼in × ⅝in

POWER SUPPLY MODIFICATIONS



HOLE SIZES
B—No. 27
D—No. 34
P—$\frac{3}{16}$in

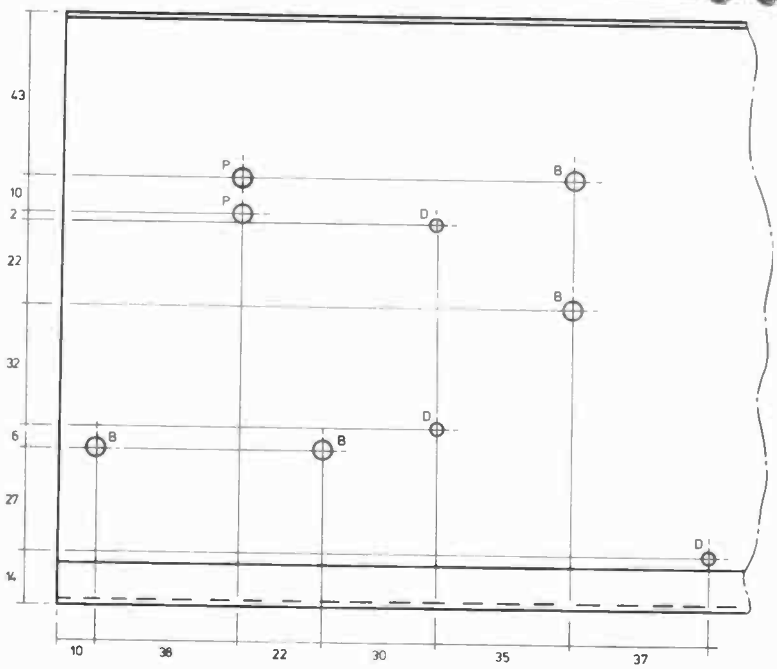


Fig. 19. Drilling details for the modified power supply component positioning

TI R.S. Components 50VA Toroid (20V) 20-0-20V 1·2A

Transformer should be mounted with two M4 screws

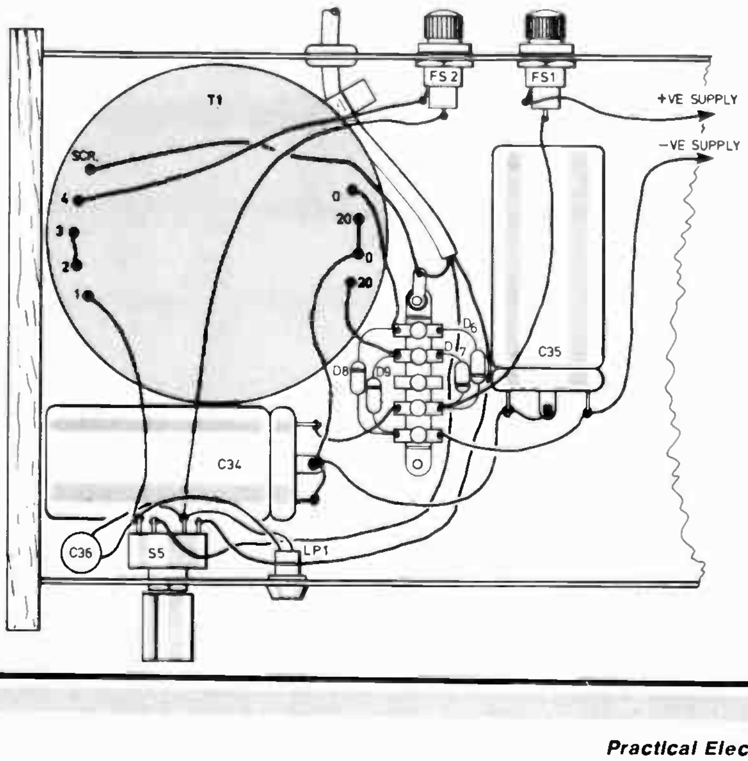


Fig. 20. Interwiring of the modified power supply

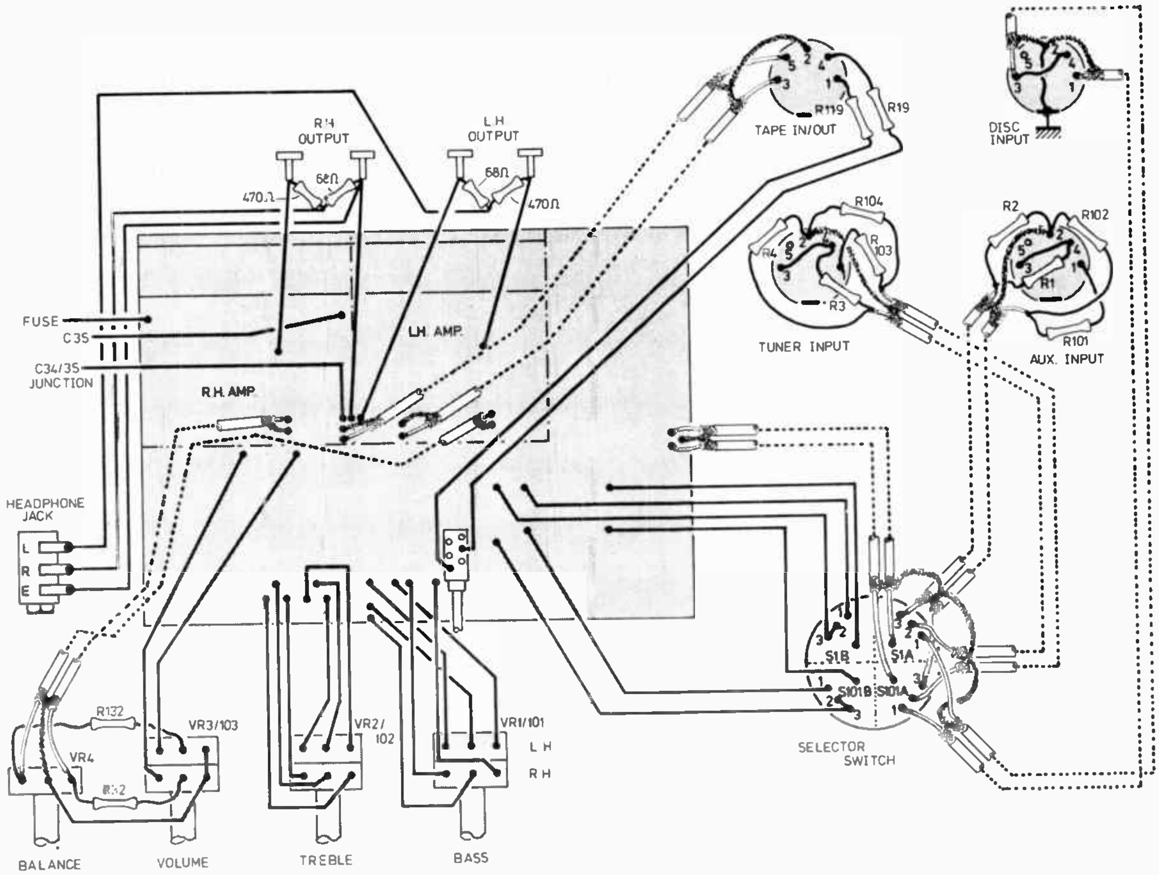


Fig. 21. External wiring to p.c.b.

LABELLING

If the drilling has been done carefully it may be possible to label the panels without any further preparation, but if scratches have appeared these should be removed by rubbing lightly with wire wool lubricated with soap and water. Rub in one direction along the length of the front and back panels until all the scratches have been removed.

The panel should then be dried and given one thin coat of protective lacquer, such as Letraset 101 or Letracote. This gives the panel a smooth surface to which the instant lettering will adhere well. The panels can then be lettered using Letraset or some similar instant lettering. Take great care to get all the words level and evenly spaced as a little carelessness will spoil the appearance of the amplifier. After the lettering has been completed it should be sprayed with two further thin coats of protective lacquer.

CHASSIS ASSEMBLY

After the drilling has been completed and the panels lettered the wooden ends can be screwed back in place. The mains transformer should then be mounted, followed by the two fuseholders and the clamps for the smoothing capacitors.

The interwiring of the power supply components is shown in Fig. 18.

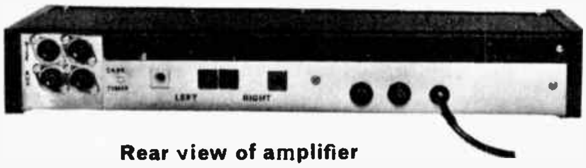
The four rectifier diodes should be soldered to their tapstrip and mounted in place. Any suitable piece of tagstrip with five or more tags can be used and the case should be drilled to suit. The rotary mains switch and the headphone jack can then be fitted.

The complete printed circuit board is fixed in place with two ¼in spacers at the front, and by the four 4mm output sockets and two 4BA screws at the rear. Take care that the printed circuit board does not short against any of the screws in the corner of the case. If the push-button switches do not quite line up with their holes they can be bent slightly on their tags.

SCREENED LEADS

The potentiometers and the selector switch are next fitted and the wiring up commenced. Screened wire should be used between the DIN input sockets and the selector switch, between the selector switch and the input stage, to the tape inputs, and between the balance control and the main amplifier inputs, but ordinary connecting wire is satisfactory elsewhere. All leads should be as short as possible and the output and power supply leads should be kept well away from the preamplifier part of the circuit and all leads carrying low level signals.

The wiring for components not mounted on the circuit board is given in Fig. 21.



Rear view of amplifier

VOLTAGE TABLE Location	Voltage
Across D4 and D5	1·3V
Across R43	0·6V
Across R39	0·6V
TR6 collector	— 1·2V
TR8 collector	1·2V
D3 anode	12V
TR4, TR5 emitters	0·7V
D1 anode	—15V
D2 cathode	—15V
IC1 (pin 6)	0V
TR3 emitter	— 1·2V
TR2 collector	1V
TR2 emitter	9·2V
TR1 emitter	— 0·7V
C3	14V
C4	—14V
C34	30V
C35	—30V

All voltages are measured relative to chassis with no signal input.

A piece of thin insulating material should be glued inside the top cover over the terminals of the mains transformer to eliminate the possibility of the cover touching the terminals if anyone should lean on the top of the case.

TESTING

After all the constructional work has been completed, check the circuit carefully for errors, and turn the two bias potentiometers VR5 and VR105 fully anticlockwise.

The initial testing should be done with a 1A fuse (FS1) and with the speakers disconnected and the volume control set at minimum.

Connect a meter reading 0–500mA or 0–1A in the positive rail, between C34 and FS1, and switch on. The meter should read about 80mA. Now turn either VR5 or VR105 until the reading on the meter increases by about 10mA.

Leave the amplifier running for about 10 minutes and then readjust if necessary. Repeat this procedure for the other channel. Check the voltage across the output terminals of each of the amplifiers. This should be less than 0·1V.

The loudspeakers and the record player deck or tuner can now be connected and the amplifier tested on music. Fuse FS1 can be left as 1 amp if only speech and music operation is required, but it should be increased to 2A if the amplifier is to be driven at full power or if it is to be used with a transformer load (e.g. a Quad electrostatic loudspeaker).

VOLTAGES

A table of voltages (Table 1) is given to assist in fault finding. Due to the unregulated power supply there may be considerable variations in some voltages due to mains voltage variations and so differences of up to 20 per cent from the values given do not necessarily indicate a fault. Voltages are measured with the volume control at minimum using a meter of at least 20,000 ohms/volt.

EARTH LOOPS

Earth loops are a common problem with audio systems, usually evident as a loud hum on both channels. They arise when two pieces of equipment, such as an amplifier and a turntable, are both connected to a mains earth. Different voltages are induced in the earth leads of the two units and this causes a 50Hz current to flow in the braid of the screened cable connecting them. The fault can be cured by removing one of the earths, preferably not the amplifier, and the other equipment is then earthed via the amplifier.

FAULT FINDING

It is difficult to give specific advice as the number of possible faults in a newly built amplifier is very large and a fault in almost any component will affect the amplifier in some way. However the following general advice can be given:

Fault on Disc/Radio. Check Disc input stage.

Fault on Tape. Check Tape input and interconnecting leads between amplifier and tape deck.

Fault on all inputs. Check tone control, main amplifiers and power supply.

If you find any dud transistors do not assume that the transistor was the only fault. Transistors can easily be destroyed as a result of a fault elsewhere in the circuit and the circuit should be checked thoroughly for wiring errors or incorect components before replacing. Failure to do this may result in blowing up the replacements as well!

In the Components List R28, R128 should be 4·7kΩ. Fig. 20 shows C34 incorrectly wired. The common 0/20V leads from the toroid and C35 move from the centre tag to the lower tag. The upper and middle tags should be linked. In the Voltage Table, "D3 anode" should be "D3 cathode". ★

Note:

A glass fibre printed circuit board printed with component locations, and a kit of semiconductor devices for this project are available from Davian Electronics, PO Box 38, Oldham, Lancs., or Astro Electronics, Spring Bank Road, West Park, Chesterfield. (All components).