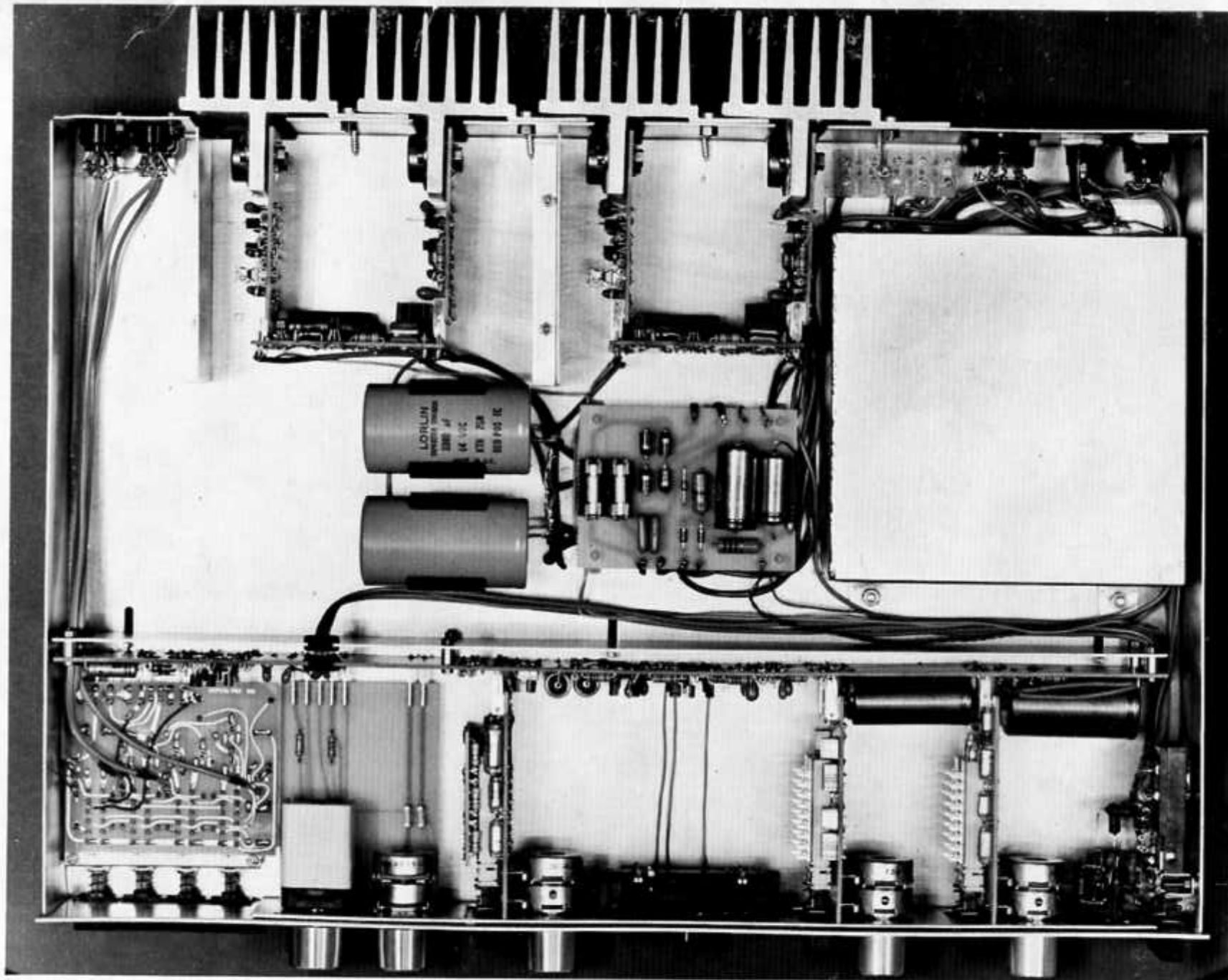


Linsley Hood 75 Watt Amplifier

De Luxe



by POWERTRAN ELECTRONICS



The subject of the photographs is a prototype and as such is subject to alteration without prior notice.

HI-FI NEWS

75 WATT AMPLIFIER

Extracts from Hi-Fi News articles.

BY J. L. LINSLEY-HOOD

IT is often difficult, in retrospect, to decide the point at which an idea took concrete shape, since many of the contributing factors have been lying around in one's mind for some time waiting to join up, and this is certainly so in the case of the equipment described here.

I have thought for some time that the most satisfactory arrangement for the manufacturers of high quality loudspeaker systems would be to incorporate the power amplifiers with the speaker drive units. In this way the amplifier could be suited precisely to the power handling capability of the speaker system, the need for short-circuit protection would be removed, the frequency crossover between units could be accomplished electronically, at high impedance at the amplifier input (preserving the benefits of low output impedance damping from the amplifier), and best of all, the whole system could be equalised for an optimally flat response in the high impedance circuitry, with troughs and bumps electronically levelled.

During some discussion of these points with a loudspeaker manufacturer at an audio show, the question was raised of how much power one should provide if the amplifier is to cope with foreseeable future trends in transducer characteristics, and a figure 75 W was suggested as a suitable target value. It was not, I think, seriously implied that the listener would want to run his system regularly at power levels of this order—although this might arise in large halls or in sound reinforcement systems—but that powers of this order could be of great use, under transient conditions, in providing the necessary rates of acceleration for the realistic reproduction of sounds with rapidly rising wavefronts.

About two years ago Mr E. J. Jordan mentioned in a lecture that it was a fairly widely accepted belief among loudspeaker designers that directly-coupled amplifiers sounded better than similar transformer or capacitor coupled systems, although the reasons for this had not been established. It seemed probable

at the time that this was a function of the output impedance characteristics of the amplifier, as it was known that this parameter varied quite widely in commercial designs at the ends of the audible spectrum, and this could be important at low frequencies in damping system resonances. But whatever the reasons, if one could remove the output capacitor of a conventional transistor power amplifier one could eliminate a costly, bulky and possibly non-linear component, an idea that seemed very interesting.

Moreover, the normal circuit arrangement of transistor push-pull output stages is of a form which could lend itself, very simply, to a DC-coupled output if a split power supply (equal positive and negative voltage lines) were to be used with the centre-line earthed, providing the return path for the loudspeaker unit. The inherent snags with this arrangement, which had possibly deterred designers in the past, are that the DC output voltage level of the amplifier must remain accurately at zero, and at least one of the electrolytic capacitors in the circuit would have to operate at near zero voltage—not good for the normal aluminium electrolytic. The first of these problems is soluble by known circuit techniques, and the second can be dealt with neatly by the use of a tantalum electrolytic. These components have been around for some time but have been rather too dear for amateur use. However, the recent introduction of plastic-encapsulated bead tantalum electrolytics at only a few tens of pence each makes these into attractive circuit components.

With these thoughts in mind it seemed probable that a power amplifier module could be constructed with very few bulky components apart from the output transistors and power supply, and with such advantages in performance as there might be in a directly coupled system.

For the power levels envisaged, a Class-B or Class-AB design was the only possible choice,

and it was clear that at the rail voltage necessary to provide 75 W into a nominal 8-ohm LS load, the use of a fully complementary output pair would not be practicable on the grounds of expense. Since the intention of the design was to provide an amplifier which would be at least equal to (if not better than) the best of currently available Class-A transistor power amplifier designs in terms of aural performance—a fairly taxing (though not impossible) task using the complementary output transistors available at lower voltage and power levels than those envisaged—it was also clear that there would be some fairly knotty design problems to overcome in the process.

A considerable amount of research has been done in recent years by various workers^{1,2,3,4} to determine why amplifiers with good technical specifications do not always sound as good as they should, and most of the problems have been found to lie in the power amplifier output stage, with the commercially very popular quasi-complementary Class-B arrangement being particularly suspect in its crudest form. Fortunately, developments of this system due to Shaw² and Baxandall⁵ offered at least the possibility that an output stage employing only *n-p-n* silicon transistors in a Class-AB system might offer the quality required.

In the event, a large number of output transistor arrangements were investigated in combination with suitable driver-stages; all of these fell far short of the sort of performance standard desired, although the broad requirements of output power capability, bandwidth and DC-coupled output were easy enough to provide. Matters rested at this stage for some months, and the whole project might have been abandoned as an interesting but unprofitable idea were it not for the fact that Rex Baldock had asked me to give a talk to the BKSTS on audio amplifiers and I had offered (in a moment of premature optimism) to demonstrate this design.

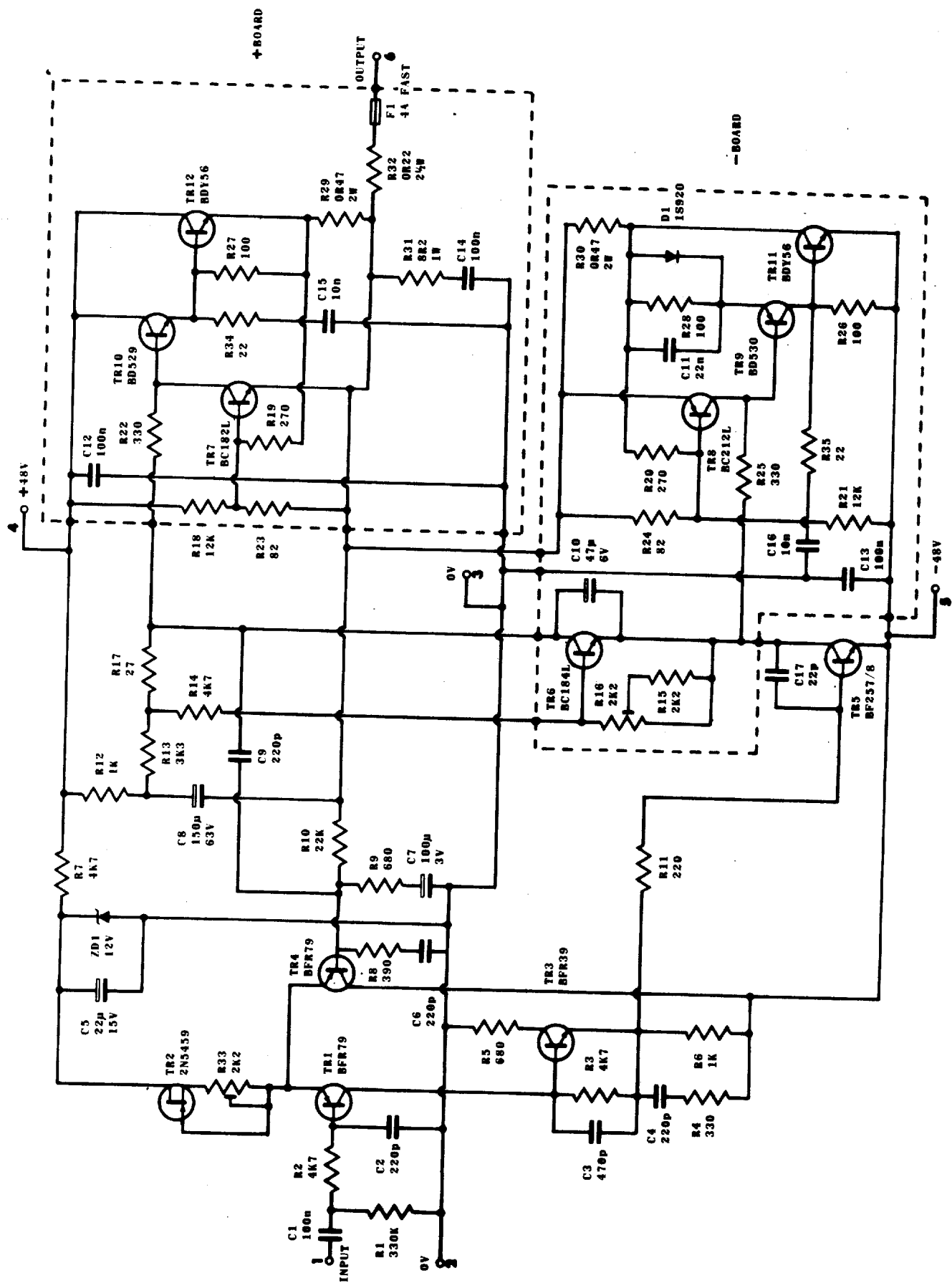
Since, in retrospect, each of the earlier designs had had some interesting features which it seemed could be combined with one another, one last version was evolved and tried—without, I confess, much optimism since most of the previous designs had seemed as good on paper and slide-rule and yet had not come up to expectations for reasons which were obscure. To my delight (and great relief) this time everything seemed to go right and the performance in every major respect came up to the specification. This, with a little tidying up of minor details, is the design to be described in these articles, and shown in circuit form.

This has now been built in a variety of power output ratings, with different transistors to give 20, 30, 50 and 75 W continuous sine-wave output into 15- or 8-ohm loads, and in every case the performance has been impeccable—so it seems that the design is reasonably free of temperament.

Most of the commercially available *pre-amplifier* designs used with transistor power amplifier units employ a now very conventional arrangement of components, and with a few notable and praiseworthy exceptions tend to a rather indifferent performance, which would render pointless any substantial improvements in the power circuits, since the preamp would then be, if it isn't already, the weakest link in the chain. In fairness, it should be noted that, in the main, existing run-of-the-mill preamps have some very good features: low hum and noise levels and the fact that the fairly substantial quantities of harmonic and transient distortion which they *can* produce don't normally lead to a tonally objectionable sound (in contrast to the comparable power amplifier) because the non-linearity distortion is mainly second harmonic, which is audibly tolerable even in fairly large amounts. Nevertheless, the greater transparency of sound arising from the lower harmonic and intermodulation products in a well designed preamp system is well worth having, and there have been some good designs produced for the quality-conscious amateur.

In general, the main shortcomings of mass-produced preamps are: the provision of inadequate filtering (especially in respect of 'rumble' and LF 'rubbish' removal); and the use of a single transistor stage in a negative feedback tone control circuit at a signal level which leads to excessive harmonic distortion when any substantial amount of bass or treble lift is employed. The attempt to get a quart out of a pint pot in the input equalisation stages can also lead to poor distortion, overload, equalisation and input impedance characteristics—particularly in respect of ceramic pickup and other piezo-electric inputs.

Low distortion levels can be attained in small-signal circuit stages by the use of high-gain configurations, with appropriate choice of signal transfer and operating voltage levels, and retention—under 'worst case' conditions—of adequate negative feedback and stability margins. Equally important, and frequently overlooked, is the need to ensure that no unwanted signal components pass into the signal circuitry via the HT line or inadvertent common earth paths. Low noise levels can be ensured



Power amplifier circuit diagram

by the use of correct transistor types at optimum current and output voltage levels, and collector voltage and input impedance conditions. The last major requirement, that of negligible hum, depends heavily on correct lead and component layout, and if changes are made in these its elimination can prove a tedious business of trial and error.

Provision is made in the preamplifier part of this project for a very low distortion, wide-range tone control unit, an effective steep-slope rumble filter whose frequency can be adjusted if necessary to suit turntable or LS/room characteristics, a variable-slope switched frequency treble filter, comprehensive input equalisation facilities, and a versatile stereo/mono selection arrangement, which has been found useful for comparisons between dissimilar speaker units as well as for the more prosaic purposes of channel balance. The hum and noise levels of the completed circuit are very low and the worst-case distortion figures throughout are less than 0.02%, which means that they are difficult to determine in the presence of even the very low residual background noise.

The reasons for the choice of the major design features of the power amplifier were discussed in general terms last month, and it was decided that the design target should be a directly-coupled low-distortion amplifier capable, if required, of a power output of up to 75 W per channel and that the total component cost, per channel, should not be much. The reasons for the choice of the relatively high output power figure was that it would allow the use of inefficient loudspeaker systems and still provide enough power, under transient conditions, to allow rapid rates of attack. This is an important feature in the realistic reproduction of percussive sounds and is one of the major, if residual, reasons (in my opinion) for the superiority of direct microphone input systems over high quality disc recordings in which tracking and cutting considerations limit possible input 'rate-of-change' characteristics. Moreover, if it should prove possible to design a high power amplifier which was reasonably inexpensive, and if the distortion levels at low output powers were of a properly low order there would be no technical or economic reason for not using it at 1 W or even 1 mW, with the comforting assurance that there was plenty of power in reserve if it should ever be needed.

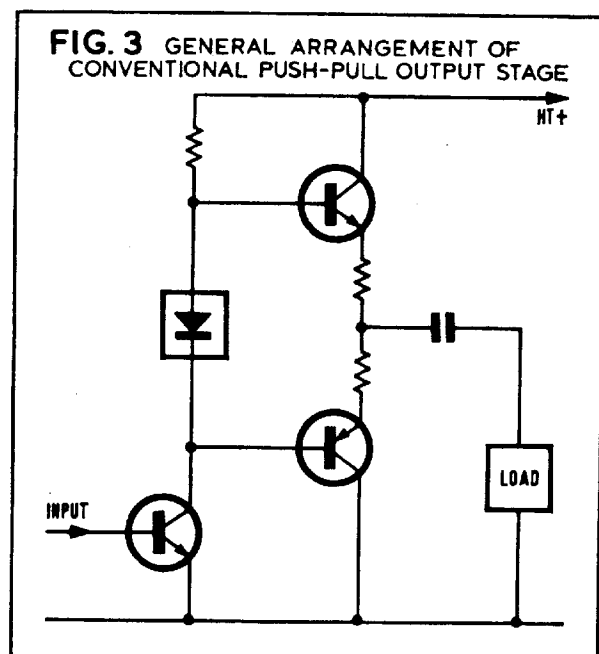
The choice of direct coupling was mainly on the grounds that it would save the bulk and cost of an output coupling capacitor between the amplifier and the loudspeaker, but it was

also possible that there might be an audible advantage arising from the improved damping characteristic at very low frequencies. In the event, I am now convinced that directly coupled systems *do* sound better at very low frequencies, given suitable speakers, because of a significantly improved 'ambience', which is particularly noticeable on certain organ and large scale orchestral works, and the better the LF response of the speakers the more worthwhile this may be.

Economic considerations dictated the use of either a Class-B or Class-AB output stage using only *n-p-n* silicon transistors, and present 'state-of-the-art' considerations suggested: a full output power total harmonic distortion of less than 0.01%, which must not worsen at lower power levels; a main amplifier bandwidth of at least 10 Hz—40 kHz (with a directly-coupled system the lower end of this is, of course, very easy to achieve); and a transient response which should approximate to that of a simple R-C integration network, for minimum phase-distortion. The transient performance should be completely unaffected by any type of loud-speaker load characteristic. That is to say that the output-waveform from, say, a square-wave input should be completely devoid of ringing or overshoot on any type of load. Finally, the system should be short-circuit and overload proof.

Although this is a formidable specification, far better in fact than the performance of any commercially available amplifier of which I am aware, technology in this particular field is advancing at such a rate that any lesser specification would soon be overtaken by the march of progress. However, the attainment of such a specification presents a number of major difficulties, and the key to the solution of these lies in the design of the output stage. The evolution of this is therefore considered in some detail.

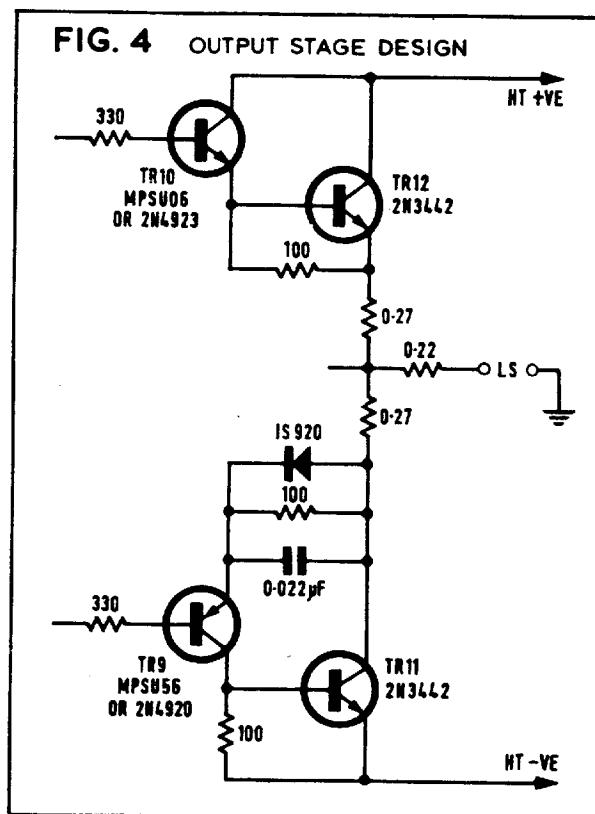
If one considers a notional push-pull transistor output stage of the type shown in fig. 3, the output voltage and current swings required can be calculated (albeit for an ideal resistive load) without difficulty. It is found that for 75 W power into an 8-ohm load, an RMS output of 24.5 V is required. For a sine-wave, this is equivalent to a peak-to-peak voltage of about 69 and a peak current into the load of some 4.3 A. At these output current levels, the voltage drop across each half of our notional output stage (allowing for the driver transistors) will be about 4.5 V, and the total available HT must therefore be at least 78 V DC.



From these considerations, the output transistors employed must, at the very least, have a collector-emitter operating voltage of 80 V and a current handling capability of at least 5 A. Unfortunately, a study of the secondary breakdown characteristics of output transistors (these are the ones involved when smoke starts to appear from the associated resistors) shows that they are capable of handling only very small currents at voltages near to their working maximum and therefore, in order to avoid sailing too close to the wind, it is desirable to choose a power transistor whose peak current handling capacity and breakdown voltage are both considerably in excess of these minima. V_{ce} and I_c (max) of 100 V. and 10 A respectively would be ideal. Equally unfortunately, a study of the catalogues and price lists of the semiconductor manufacturers show that *n-p-n* transistors of this type are rare, and *p-n-p* types are rarer still and excessively expensive. The use of a fully complementary output stage is therefore just not a practicable possibility within our original terms of reference at these power levels, and even an *n-p-n* only stage could be expensive, if the transistors are chosen imprudently.

The circumstance which has simplified this part of the problem is the current availability of 2N3442 devices. These have a 140 V_{ce} rating and a current capability of 15 A and are relatively inexpensive. The circuit to be described was therefore designed around these devices, although it has subsequently been used at other power levels with different transistor types, in arrangements which are described later.

Several output stage configurations using only *n-p-n* silicon power transistors were investigated in the earlier stages of the evolution of this design, and of these by far the best, in terms of performance and economy in the use of components, was a derivative of a design due to P. J. Baxandall⁵, and is shown in its final form in fig. 4. The 0.022 μ F capacitor across the diode in the emitter circuit of Tr9 compensates for the load capacitance in its collector circuit due to the Miller capacitance of the base-collector junction of Tr11, and gives a small improvement over the circuit due to Mr. Baxandall, in which this capacitor is not used. The distortion waveforms given by the final design of amplifier (all well below 0.01% RMS) for a 60 V peak-to-peak sinewave output, are shown in figs. 5, 6 and 7. In the first of these, the characteristic crossover point notches of the standard quasi-complementary output arrangement can be seen quite clearly, in spite of the very large amount of negative feedback employed (and this should be a sufficient answer to those who think that enough negative feedback can straighten out the hind leg of a donkey). In fig. 6 the considerable improvement given by the use of Mr. Baxandall's modification, a forward biased diode connected across the emitter resistor of Tr9, can be seen.



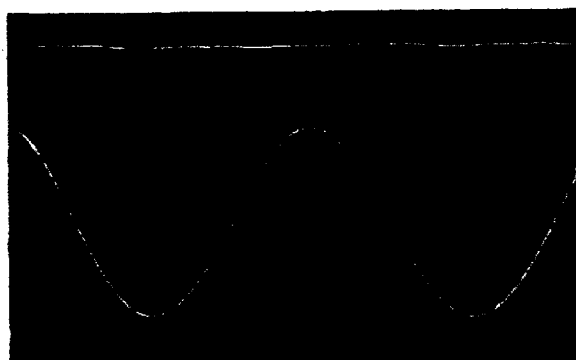
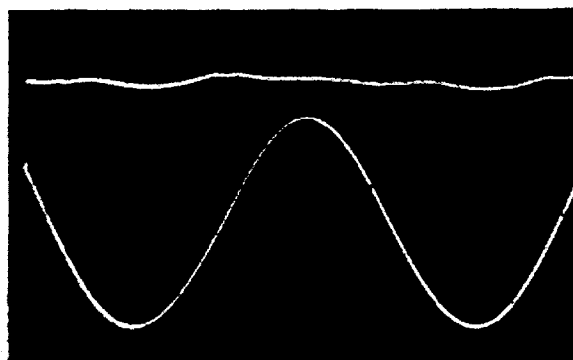
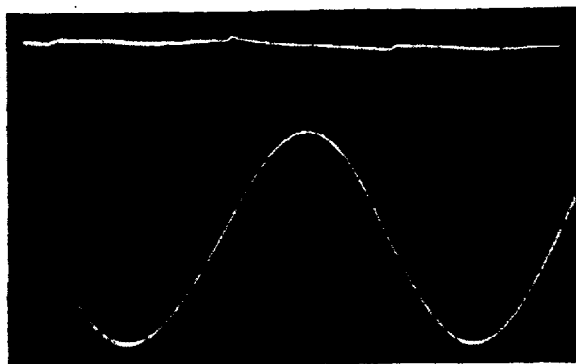


Fig 5 (top left) Distortion characteristic of normal quasi-complementary pair.

Fig. 6 (top right) Improvement in distortion characteristic due to inclusion of P. Baxandall's diode in lower driver transistor circuit of output pair. (Distortion meter sensitivity X4 as compared with fig. 5.)

Fig. 7 (left) Distortion characteristics of final output stage arrangement, (distortion meter sensitivity as fig. 6).

By this addition the notches are reduced to a much lower peak amplitude. Finally, the addition of the capacitor across the diode reduces the amplitude of these small irregularities still further, fig. 7, and gives an *n-p-n* only system which appears at least equal to its fully complementary counterpart.

The final complete power amplifier circuit is shown in schematic form in fig. 8, and its method of operation will be explained with reference to this simplified diagram.

In this, the input signal is fed to one of the transistors (Tr1) in a 'long-tailed pair' which are arranged to have a very high dynamic impedance 'tail' circuit load, which gives very good static and dynamic balance between the two input halves, and the negative feedback signal, with which the input is compared, is fed to the other (Tr4). Since the circuit seeks to establish a condition of balance between the two halves, and the base of (Tr1) is connected by a DC path to the earth line; this is achieved when the point by which the DC potential of the base of the other transistor (Tr4) is determined is also at zero volts. Since this point is the output connection of the amplifier it follows that the system will operate automatically to adjust the output point to a potential which is, for all practical purposes, at a DC earth line voltage. The loudspeaker system can therefore be connected between this point and earth without any DC current flowing, and it is there-

fore unnecessary to interpose any blocking electrolytic capacitor.

An additional advantage accruing from this system (quite apart from any audible improvement which there may be) is that if the HT+ and HT- lines are equal in voltage, then the output point automatically adjusts to be at the mid-point of the HT voltage, which is desirable for maximum output and minimum distortion. Moreover, if the output point starts and ends at earth line potential there should be no 'plop' at switch-on or switch-off. In practice, this condition may not be met precisely because of differences in the charge and discharge rates of the smoothing capacitors in the positive and negative lines, but it is now a minor rather than a major problem.

The second amplifier stage Tr5 is arranged to be an *n-p-n* transistor, rather than the more commonly used *p-n-p* type. This allows the input devices to be of *p-n-p* type where their lower noise level is advantageous, but more importantly, allows the use of a very high voltage working device in the very critical high level amplifier stage. For this a BF 257 (160 volt V_{ce} rating) transistor is employed, and this contributes significantly to the linearity of the system at high output voltage swings.

Tr6 is employed as an amplified diode in the collector circuit of Tr5 to provide the correct biasing levels for the output transistor pairs. A BC 108 or BC 109 is used in this position and

it should be arranged so that its can (which is connected to the collector) is in thermal contact with the earthed output transistor heat sink, but insulated electrically from it. The most convenient way to do this is probably to cement the transistor into a small piece of metal with a suitable sized hole bored in it, and then to affix this with a suitable insulating washer to the heat sink itself. A simpler, and probably adequate, alternative is to use a plastic-encapsulated transistor, such as the BC 184L, cemented into a hole in the heat sink.

The overall AC gain of the amplifier is determined by the ratio of the two resistors RFB1 and RFB2, and with the values shown the gain is 33. Maximum output is then given by an input signal of 0.75 V RMS. At DC the gain is unity, as explained above, for the requirement of stabilising the output voltage level, and at low frequencies the gain is determined mainly by the value of the capacitor in series with RFB2. The value shown gives a 3dB point of 3 Hz, which is more than adequate for audio purposes. The upper frequency roll-off point is mainly determined by the 4.7 K/220 pF input integrating circuit in the base circuit of Tr1.

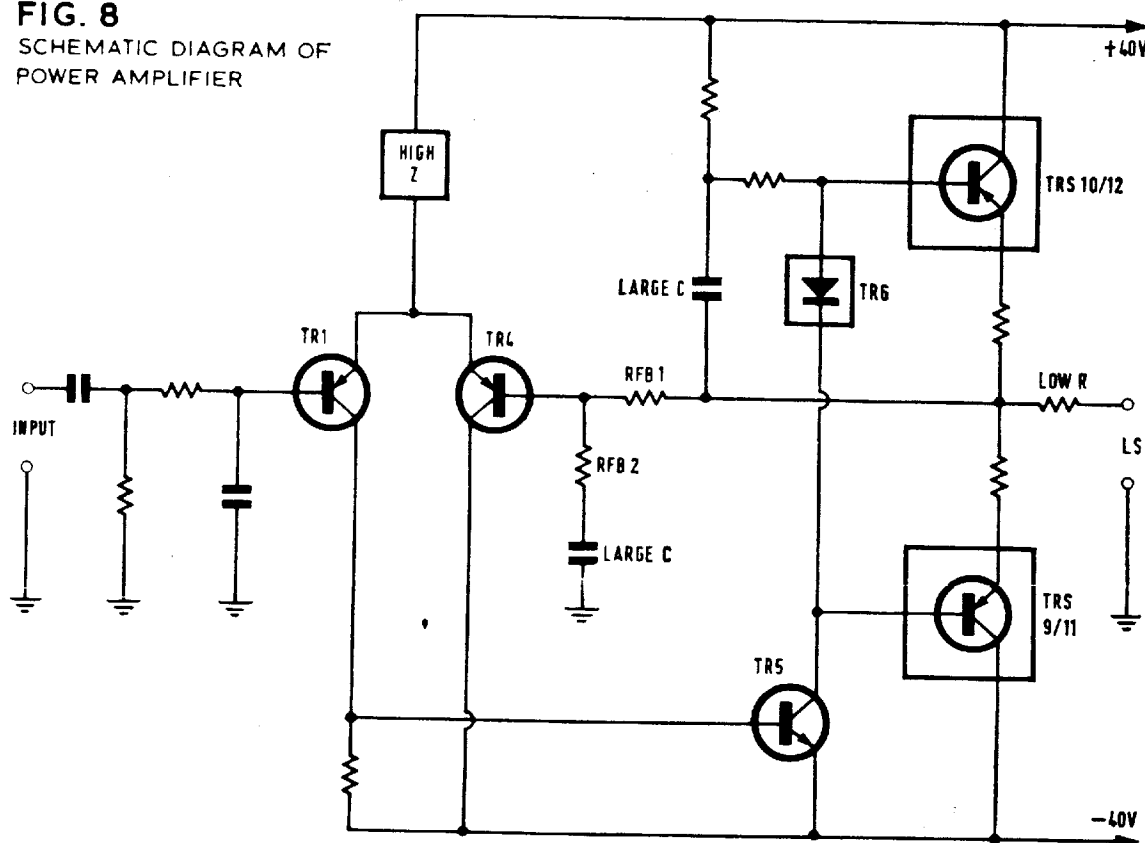
although the loop gain of the amplifier is deliberately reduced at high frequencies by the 220 pF feedback capacitor from the collector of Tr6 to the base of Tr4.

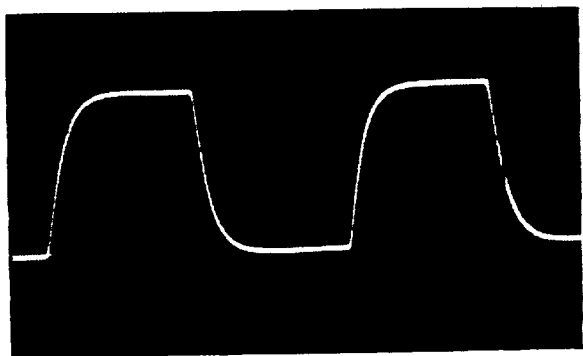
The final point of note on the simplified diagram is the 0.22 ohm resistor in the output to the loudspeaker. This serves to prevent the DC negative feedback within the amplifier, which stabilises the DC working point voltage, from disappearing in the event of an inadvertent short circuit across the output.

The complete circuit of the power amplifier was shown in fig 1 last month. It is seen from this that FET (Tr2) is used as a constant current source to provide the necessary high impedance 'tail' for the input long-tailed pair, and that this is, itself, decoupled from the HT line. In order to increase the gain of the first amplifier stage, and thereby increase the amount of overall negative feedback available, the dynamic impedance of the first transistor collector load is multiplied by a DC 'bootstrap' circuit consisting of the emitter-follower Tr3. This also provides a low impedance drive to the amplifier stage, Tr5, which is desirable.

Transistors Tr7 and Tr8 provide a fast acting overload protection circuit, and limit the input

FIG. 8
SCHEMATIC DIAGRAM OF
POWER AMPLIFIER





Transient performance of power amp, 10 kHz square-wave input; 30 V peak-to-peak.

Fig. 9 (top left) With 8 ohm resistive load. Rise time 5 μ S.

Fig. 10 (top right) With 8 ohm resistive load in parallel with 2 μ F.

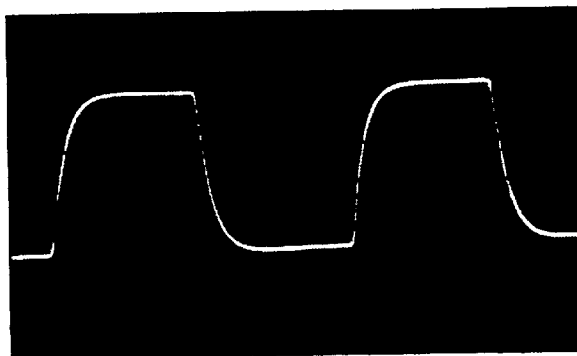
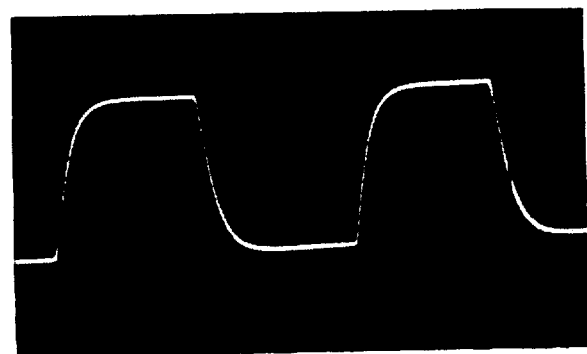
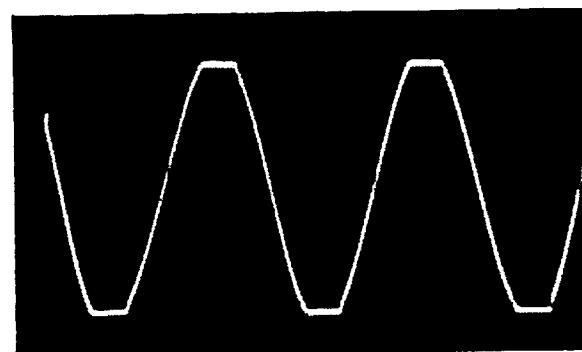


Fig. 11 (lower left) With simulated ELS load.

Fig. 12 (lower right) Overload characteristic of power amp. Sine-wave at 1 kHz into 8 ohm, 2 μ F load. 72 V peak-to-peak.



drive to the output stage in the event of excess current or voltage demands into, for example, an inadvertently short-circuited output. The base emitter-capacitor of Tr3, the resistor-capacitor network from the base of Tr4 to earth, the resistor-capacitor networks across the emitter resistor of Tr3 and the LS output, and the 0.1 μ F capacitors from the emitter of Tr11 and the collector of Tr12 to earth are all provided to tailor the HF-gain and -phase relationships, and to ensure that the behaviour of the amplifier under transient conditions and reactive loads is beyond reproach. I believe that this parameter is a vital and often overlooked factor in ensuring that the tonal quality of the system is good under all loudspeaker load characteristics.

The output fuse is provided to protect the *loudspeaker* and the value should be suited to the capabilities of the units employed.

Because a very high level of negative feedback is employed in this circuit (the open loop gain is normally in excess of 250,000—giving a

feedback factor of about 8,000 or 78 dB at 1 kHz) the layout of the circuit is moderately critical for optimum results, and it is recommended that a printed circuit board be used, for which an appropriate layout of components and circuit design is shown.

It is a difficult task, even for an experienced engineer, to construct a complete amplifier system and have it work first time off, so it is suggested that the best way to go about this is first to cut out and fit together the case of the amplifier, drilling out all the necessary holes for plugs and sockets, power transistors etc. Then the mains transformer, rectifiers and smoothing capacitors can be mounted on the rear plate of the chassis and tested to make sure that they give the correct order of output voltage and polarity. At this stage the power amplifier panels can be built and connected up, one at a time, to the power supply, and the quiescent current for each half set.

The value should be checked, and adjusted if necessary, once again, after the unit has been in operation for five or ten minutes. This current setting should then be regarded as the final one. The power amplifier panels can then be mounted on the chassis as shown and the output stages tried out on a loudspeaker, if required, using an input signal from a portable radio or some similar source. Then, once the constructor is satisfied that the power supply and output stages are working correctly, construction of the preamplifier can proceed. This should be built in reverse order with the stage immediately preceding the main amplifier (in this case the tone control stage) built and tested with the main amplifier first, and then the one preceding this, and so on. This way the point at which any fault occurs can be identified and corrected before proceeding further.

The performance of the main amplifier in its final form is summarised in table 1.

**TABLE 1:
PERFORMANCE SPECIFICATION**

Power output	75 W into 8 ohms
Distortion at all power levels	0.01% maximum
Stability	Unconditional
Input impedance	220 K
Output impedance	0.22 ohms
Half-power bandwidth	3 Hz-40 kHz
Rise-time	10 μ S
Intermodulation distortion (at all power levels)	0.05% maximum
Squarewave performance at 1 kHz	Less than 0.2% error with resistive or reactive loads
Voltage gain	33 (31 dB)
Feedback	78 dB typical

With the present state of the art in commercial audio amplifier systems the pre-amplifier circuitry has tended to follow certain rather stereotyped design forms, and although these have undoubted advantages, mainly in terms of economy of components and low noise levels, they do not give the performance—in terms of harmonic and intermodulation distortion and the filtering of unwanted noise components—of which the best of modern transistor circuitry is capable.

In this respect the DIY enthusiast for whom a variety of good designs are available, may have a substantial advantage over his wealthy neighbour whose elegant satin-chrome front panel can sometimes conceal a motley collection of cost-effectiveness compromises, and in my own experience the cumulative effect of

a string of commercially acceptable design short cuts certainly shows in comparison with the sort of no-compromise design one can build for oneself. (The look of astonishment on the face of one's neighbour who has come to patronise, and will leave to advertise, must certainly make up for a lot of late nights with screwdriver and soldering iron.)

One of the main problems which faces the engineer in the design of low level signal circuitry such as an audio preamplifier, even if cost is not a major consideration, is that the requirements of low harmonic distortion and low noise level conflict, and since the manufacturers of commercial 'Hi-Fi' equipment know that the background noise level of their amplifiers, particularly at low settings of the volume control, is important to the customers who buy (and the dealers who sell) their products, it is hardly surprising that the design balance is normally tipped in favour of low noise, even at the expense of other things.

In more specific design terms, it is normal commercial practise to concentrate the bulk of the signal amplification in as few stages as possible (usually a two transistor feedback pair), prior to the volume control, and to have little or no gain before that of the following power amplifier. If the signal level at volume control for maximum amplifier output is 0.5 V RMS, which is fairly typical, the input transistor pair has to provide, from a low signal level magnetic pick-up input, a gain of about 150 at 1 kHz, and if the RIAA recording characteristic curve is to be properly equalised, a gain of about 1500 is then required at 40 Hz. Even when possible, it leaves little loop gain available for negative feedback, and distortion levels at 0.5 V output at 40 Hz can approach 1%, with a consequent substantial reduction in the clarity of reproduction of signals, such as organ accompanied choral works containing both bass and treble.

In addition, the 26dB (20 \times) overload margin frequently claimed is hardly within the realms of possibility at this sort of signal level since 0.5 V RMS is equivalent to 1.4 V p-p, and 26 dB up on this is 28 V p-p, hardly practicable with the normal 15 volt power supply!

Further, the typical single transistor feedback tone control circuit, which is widely used as a unity gain stage following the volume control, has a poor performance both in respect of harmonic and IM distortion except in the condition in which it is inoperative (i.e. set 'flat'). Harmonic distortion of 0.3%, and

worse IM figures, are typical from such an arrangement in the regions affected by bass and treble lift. In this respect the practice of manufacturers and reviewers of quoting THD figures at 1 kHz, with tone controls set level, is as misleading to the potential purchaser as the (happily now discontinued) practise of quoting distortion figures of power stages at full output with a resistive load, which concealed both the shortcomings of early quasi-complementary transistor output stages and also the lack of reactive load stability.

Finally, the filter facilities provided—even in quite expensive apparatus—are often laughable, such as so-called ‘rumble’ filters which begin to operate at 200 Hz (or even 2 KHz!). Even if treble filters can be dispensed with, the availability of a proper ‘rumble’ filter (i.e., one which is flat down to some 30–35 Hz, and then attenuates rapidly below this frequency) is important to good results, since even if one’s turntable is good, a proportion of records (and tape recordings) have rumble recorded on them, and the role of an efficient rumble filter in removing ‘boominess’ due to inadvertent airborne or structure borne (i.e. via floors or shelves) mechanical or ‘acoustic’ feedback, needs to be more generally understood.

The problems of overload capability and distortion levels can be reduced in scale if the volume control is placed at a lower signal level position, with less gain in front of it, but this solution then demands much more care in circuit and layout design to minimise circuit noise and hum pick-up. This consideration has been the principal one in the design of this particular preamplifier circuit, in which the main aim has been to keep harmonic and other distortion components to the lowest possible level and to provide a large effective measure of overload capability. The circuit and layout are given detailed attention to minimise hum and noise only after the above requirements have been met. A schematic layout of the preamplifier circuit is given in **fig. 14**, and the complete circuit is shown

INPUT EQUALISING STAGE

For the reasons given last month it was decided to operate the volume control with a maximum signal level of 150–200 mV. This gives an output overload margin of about 20 dB with the type of stage envisaged for this position, and in order to avoid any part of this being wasted in taking up the differences between, say, a low output and a high output magnetic P.U. cartridge or a high output or low output radio tuner, each of the input

channels has been given sensitivity adjustment. This also allows each of the outputs of, say, a stereo cartridge to be balanced accurately, and in practise this initial setting up can be done without difficulty by the use of the ‘mode’ switch.

By far the easiest way to achieve this type of variable input gain is by the use of an operational amplifier system as shown in **fig. 14**, and the ‘Liniac’ circuit arrangement¹ provides a very good high gain, low noise and low distortion building block for this purpose. Since the typical open loop gain of the Liniac, at low frequencies, is 3000, and the maximum closed loop gain required to give full output from a low output magnetic P.U. is only 500 at 40 Hz., a feedback factor of 6 is still available under worst-case conditions. This reduces the typical distortion of this type of stage (0.06% at 0.2 V RMS) under such conditions to around 0.01%—about 100× better than commercial practice! A high impedance (2 megohm) input is also provided for a ceramic cartridge input, with proper equalisation for the 500 Hz–2 kHz fall in response due to the RIAA recording amplitude characteristics and also due to the self-capacitance of the cartridge at low frequencies. The other inputs have flat response.

FIG. 14 SCHEMATIC DIAGRAM OF PREAMPLIFIER

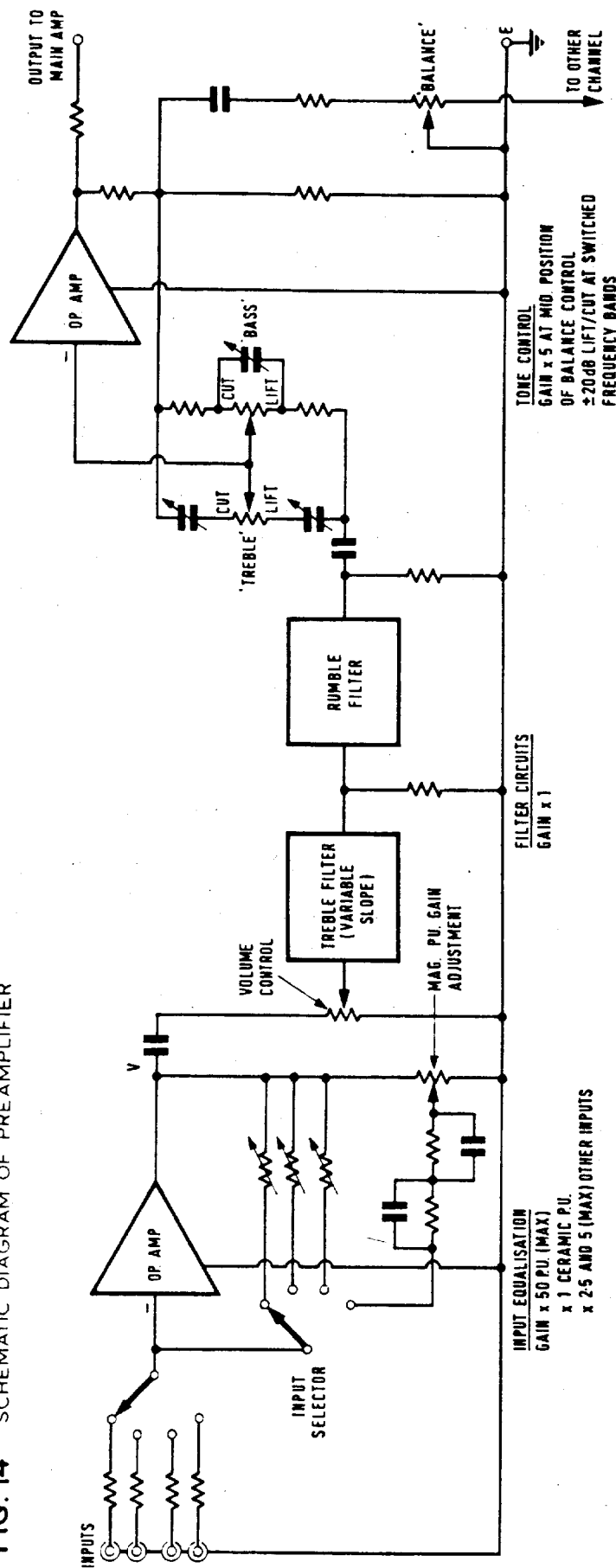


FIG. 19 ALTERNATIVE BASS CONTROL

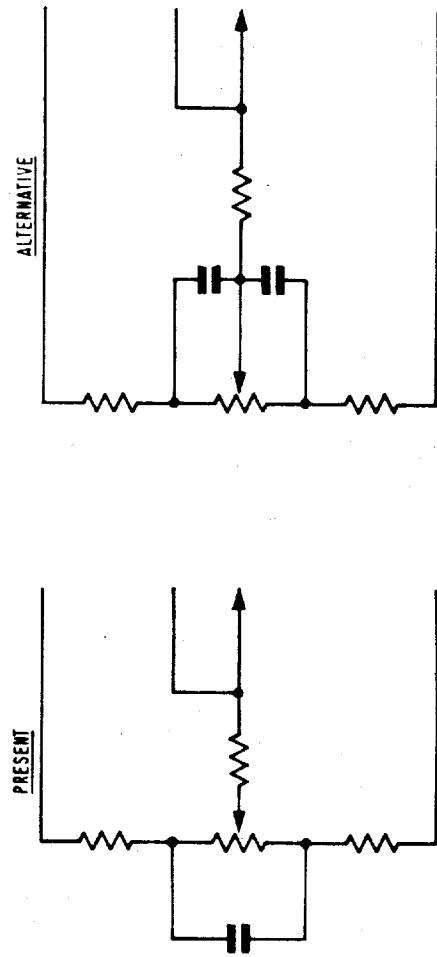
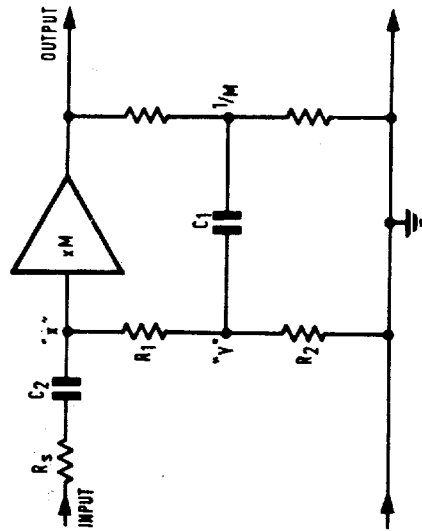
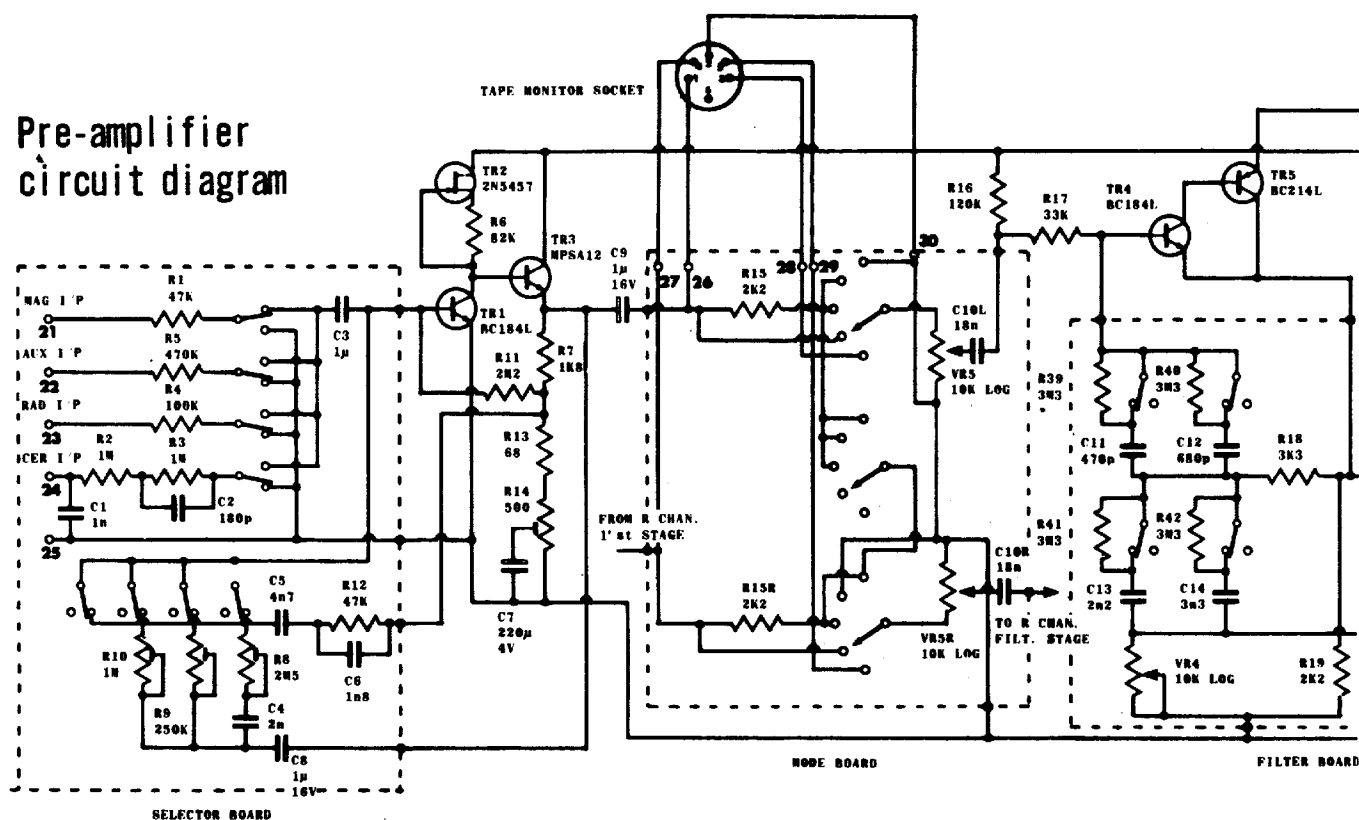


FIG. 20 'BOOTSTRAP' FILTER CIRCUIT



Pre-amplifier circuit diagram



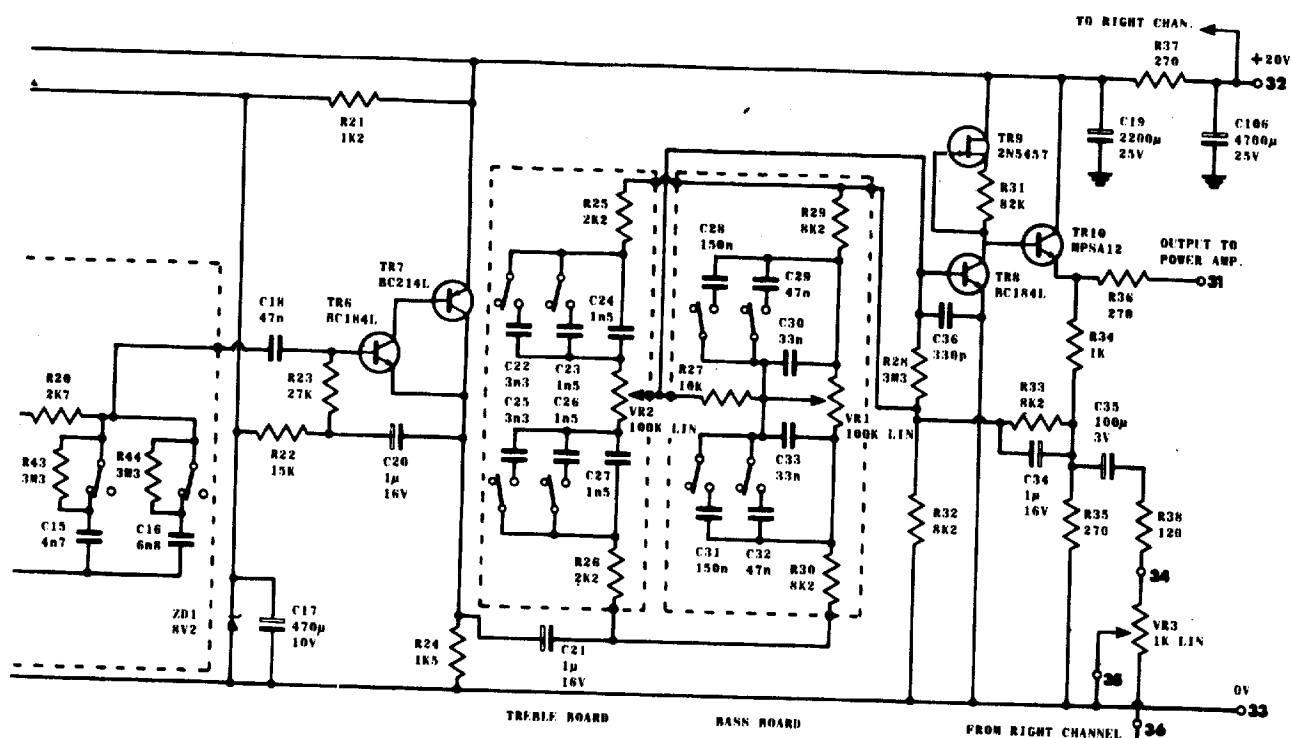
FILTER CIRCUITRY

Two filter stages are provided between the volume control and the tone control stage. These are of the 'H' or 'bootstrap' form and are capable of giving an ultimate attenuation rate of 18 dB/octave, with unity gain in the normal passband. The first is used as a variable-slope treble filter, having a switched operating frequency. At its maximum resistance position the slope control gives a characteristic which is nearly flat. A logarithmic control is preferable, arranged so that the maximum resistance position is fully clockwise. With this connection the 6 dB slope is at about half rotation, and maximum cut is fully anticlockwise.

This is followed by a steep-cut rumble filter having a fixed turn-over point at about 32 Hz. (During the development of this circuit I experimented with a variety of roll-off frequencies from 20 Hz upwards, which one can easily get by playing about with the component values, but I think that the 32 Hz position is optimum.)

The stage has a low output impedance, necessary for the proper operation of any negative feedback tone control stage. To minimise noise and distortion arising from these two stages a two-transistor high gain negative feedback

combination is used as a unity gain arrangement instead of the more normal single emitter-follower. Their bias is derived from the 8-V positive line feeding the equalisation circuit. Handling capability of these filter stages is very high, and they could have been sited between the tone control stage and the main amp without degrading the signal. In this case, however, an additional impedance converting system would have been necessary between volume and tone control stages, and this was avoided since I believe that, in general, the fewer the stages the signal has to pass through the better.



TONE CONTROL STAGE

A fairly conventional negative feedback tone control stage is employed, but its performance is substantially upgraded by the use of a further Liniac in this position. Since the reserve of gain in this case is enormous this stage is used to provide the necessary $5 \times$ gain between pre-amp volume control and main amplifier input. The worst case distortion is about 0.01%, a further $30 \times$ improvement on normal commercial practice. Variation of negative feedback in this stage is also used to provide the balance between channels.

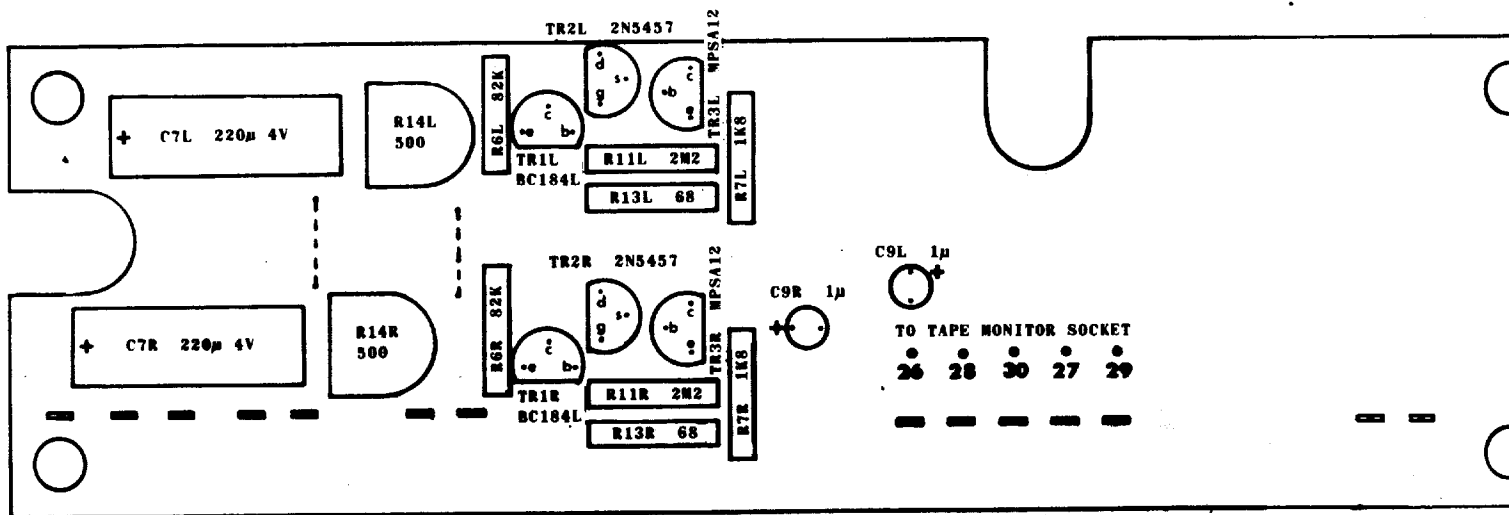
One development which I have included, and which I think could be applied more widely, is a switched operating frequency range for the bass and treble controls. This switching can be done by miniature DPDT toggle switches, and if these are of the centre null type, a range of three operating bands can be selected. The argument behind this is that, with good programme material, bass and treble controls will mostly be used to remedy shortcomings in the loudspeaker (or more rarely, the transducer) system. In this case only the extreme ends are likely to require lift or cut if the loudspeakers are of a high quality. If the programme material is poor, or the speaker units of more modest nature, lift or cut may need to be applied at regions nearer to the mid-

point of the frequency band. In the case of the units which I have in my own home, bass lift would normally be unnecessary, but the use of lift at the bottom of this range, with the $0.15 \mu\text{F}$ capacitors, produces a dramatic effect on bass drum beats or organ pedal notes, when a fairly substantial magazine such as HFN/RR held in the hand on the opposite side of the room can be felt to flutter. Modifications to the frequency curve can also be produced by judicious combinations of the tone control and treble filter characteristics.

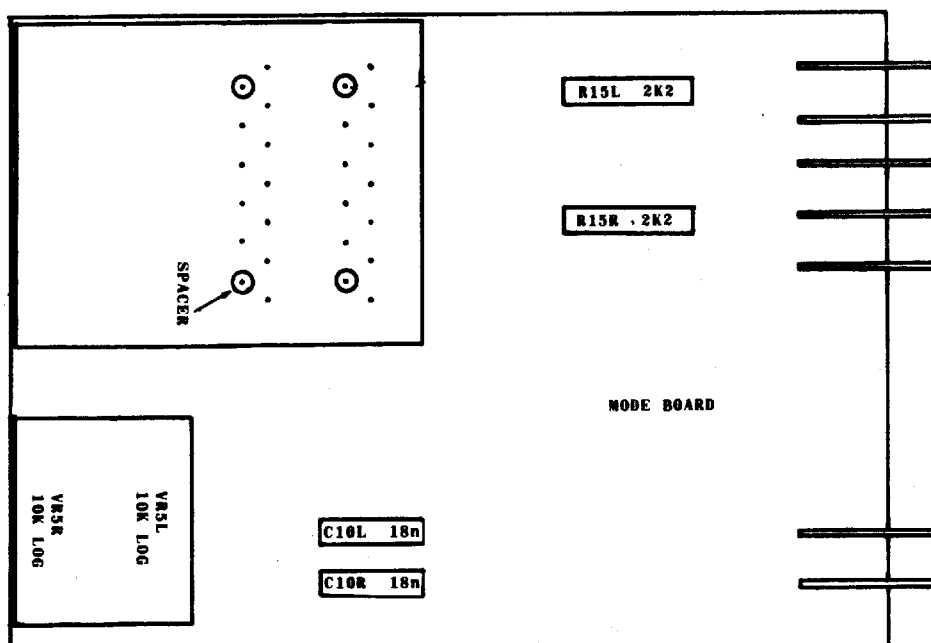
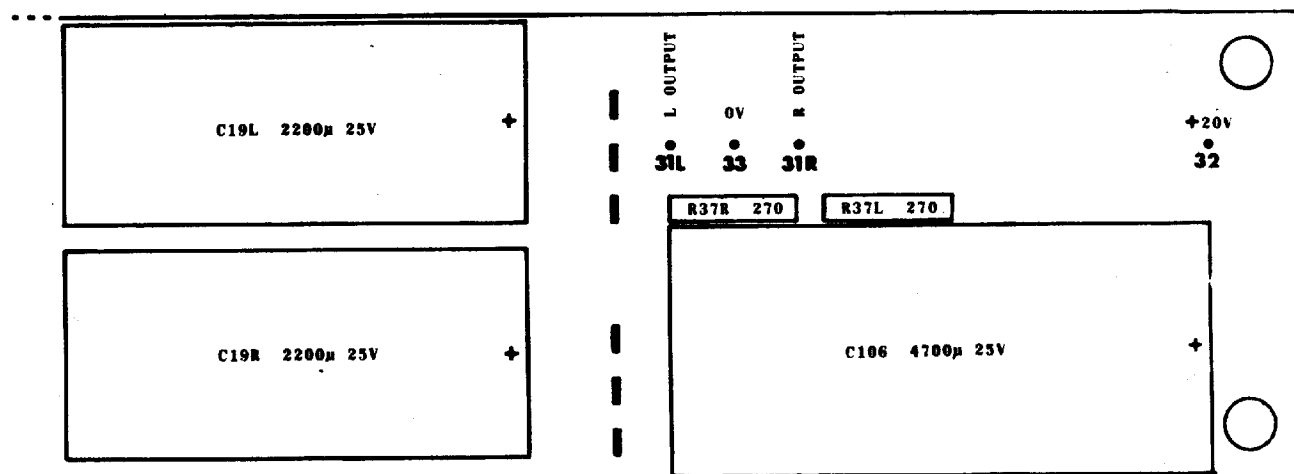
A tape record output is provided at the output of the tone control stage, at about the 100 mV signal level. The output impedance of both this output and that of the preamp itself is sufficiently low for these to be used at the end of a considerable length of screened cable without ill effect.

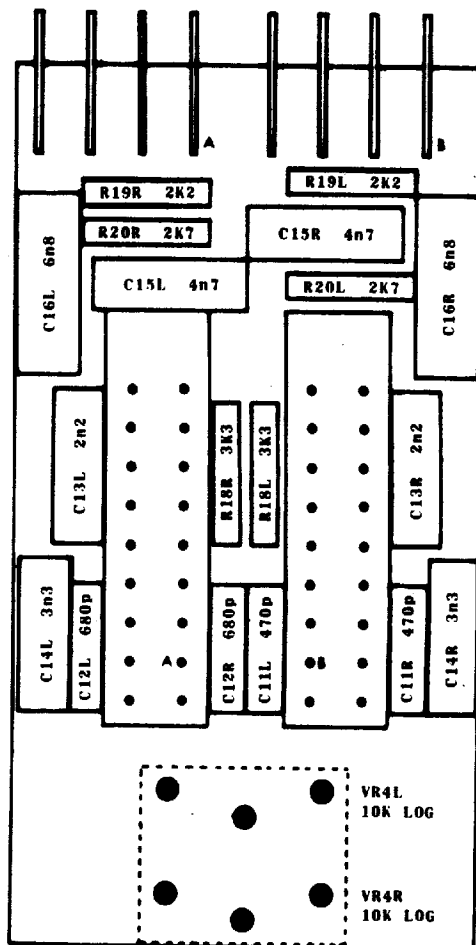
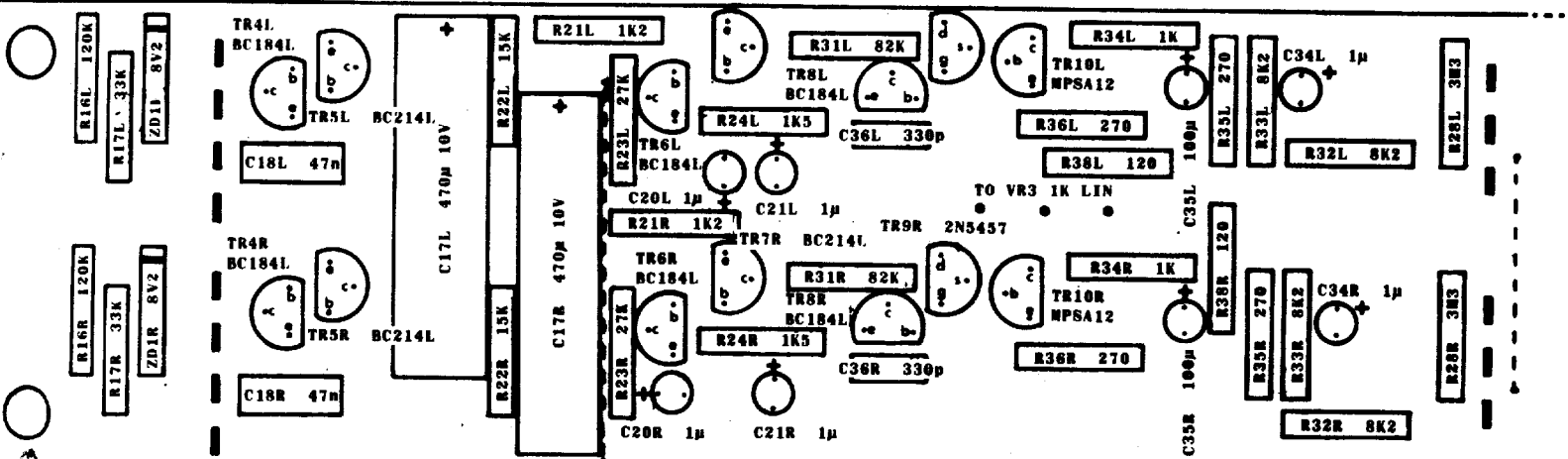
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1. J. L. L. H., *Wireless World*, September 1971, pp. 437-440.
2. *Ibid.*, May 1970, p. 207.
3. *Ibid.*, December 1970, pp. 609-610.



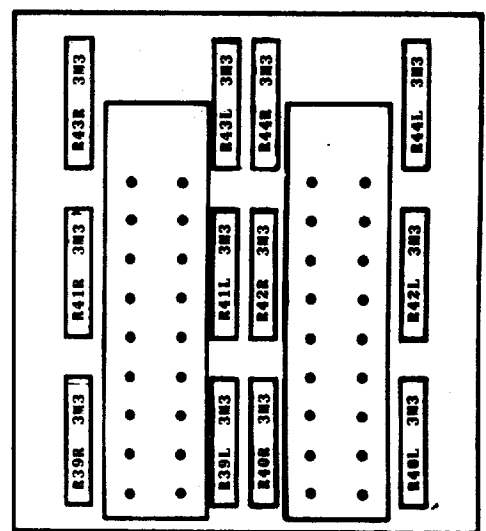
WIRE LINKS SHOWN AS DOTTED LINES





LINK SWITCH CONTACT A TO CONNECTOR A
LINK SWITCH CONTACT B TO CONNECTOR B

FILTER BOARD 1



FILTER BOARD 2

Pre-amplifier boards

The voltage at the live LS output point, with respect to the chassis, should ideally be 0V (DC) under no-signal conditions; ± 50 mV is a reasonable practical equivalent. All of the dozen or so early models achieved this. However, DC offset depends on the characteristics of the FET Tr_2 and on the matching of the current gains of Tr_1 and Tr_4 . (Because it was my wish to generate as low an impedance as possible at the emitter of Tr_4 , in order to achieve as high a gain as practicable from Tr_1 , I had chosen to operate Tr_4 at a larger current than Tr_1 and their base circuit resistors were chosen for this condition). If FETs other than the Motorola 2N5457 are used for Tr_2 , DC offset will probably be unsatisfactory.

However, in retrospect, I think I should have realised the probable component variability which would be found in practice and built in a 'zero offset' control. This can be done by using an FET which has a higher quiescent current than required (i.e. one which would give a normally negative DC offset at the LS terminal—if the one in use does this already, it will serve) such as the Motorola 2N5459 (MPF105) and then putting a 2.5 K variable resistor in its source lead. This resistor should be adjusted, after all the components have had a chance to reach their normal operating temperature, so that there is a zero DC voltage across the LS under 'no-signal' conditions.

The FETs used, except as I noted in the original diagrams, should be Motorola MPF 103s or Motorola 2N5457s (a later type number for the same device). Having now tested the substitutes I do not find that they give an equivalent performance, so I would urge the use of the Motorola component where possible. The important one is used as the dynamic load in the first 'Liniac' stage, where the stage gain depends on the FET dynamic impedance. (It also depends on the current gain and output impedance of the amplifier transistor used—the lower the h_{oe} value the better

Sescosem 2N3442 output transistors were used in the prototypes. These have superior HF characteristics to most of the substitutes

which I have tested. Motorola's MJ481 and epitaxial-base 2N3055s have equivalent HF performance. Unfortunately this is a more critical factor than I appreciated when writing the articles since, by chance, all the output transistor types which I had tried (of the types noted above) had F_T values of 3 MHz or above. For an output transistor F_T below some 2 MHz, the transient performance under reactive load conditions will deteriorate, and if the HF performance is poor enough (F_T values below 0.8 MHz) harmonic distortion will be worse and the amplifier may even become unstable for small values of reactive load (say 0.1 μ F). Larger C values may be less bad in this instance.

Fortunately, there is a relatively simple solution to the problem which avoids the expense of replacing output transistors, and will provide a solution in all but (I guess) the very worst cases. That is to put a small RF choke, about 1 μ H inductance (18—20 turns of 24—26 SWG enamelled copper wire wound around the body of a 100 ohm, $\frac{1}{2}$ watt resistor, with the free ends of the wire soldered onto the leads of the resistor) between the output fuse holders and the LS socket.

The linearity of the power amplifier circuit up to the output stage is extremely good, and the design philosophy, for reasons explained later, utilises a progressive decoupling of the output stage from the main feedback loop as the signal frequency increases. This means that the output stages rely increasingly on the emitter-follower type loop negative feedback at higher frequencies, which means that, within reason, the higher the F_T of the output devices the better the circuit performance. Three very new transistor types are: the Sescosem BDY56 which is almost identical with, and is a direct replacement for the 2N3442, but with a *minimum* F_T value of 10 MHz; and the Motorola BD529 (*n-p-n*) and the BD530 (*p-n-p*) (available driver transistors (100 V, 100 MHz planar types) which are an effective replacement for the rather more massive but lower F_T TIP29C and TIP30C devices.

An experimental output PCB with these devices gave measured THD of about 0.003% from 3 kHz—300 Hz when the I_q was set to the optimum value (which appears to

be in the range 80—100 mA for BDY56s). This is around the background noise of the main amp itself, and would be well below the measurable level when the preamp noise contribution is added, so I think this is 'lily gilding'. An alternative transistor type which is very similar to the BDY56 and appears to give an identical performance is the RCA 2N5038, but with higher optimum I_q (120—150 mA).

While the use of very high F_t output transistor types certainly allows some very low THD figures to be obtained over a wide frequency range, which is most attractive to the perfectionist, there are a couple of snags. With these output devices (I am not referring to the BD529/30s) some care must be exercised to separate—physically—output and input leads to the PA, or interpose some screening, otherwise the small residual feedback capacitances could cause oscillation in the 10–20 MHz region. Also, the higher F_t transistors are not so rugged as the 3442s, and have a smaller reactive load safe operating area. So, while these high F_t transistors are an attractive proposition for someone starting completely from scratch, I would emphasise that the bulk of the Sescosem 2N3442s will give the performance indicated in the original article.

I suggested the use of high F_t output and driver transistors as a means of achieving even lower power amplifier THD figures, but pointed out the possibility of HF instability due to wiring 'stray' capacitances. This problem can be made much less likely by the connection of a 0.01 μF capacitor in a series with a 22 ohm resistor from the base pin of each output transistor to the '0' volt rail. The very high frequency 'break' points due to this network have no adverse effects on the performance, and assist in reducing HT line transmitted impulse noise.

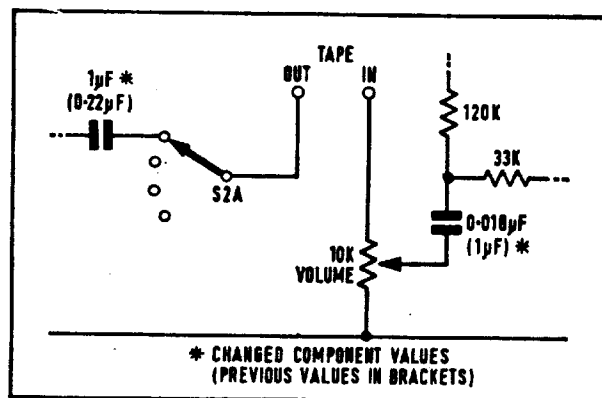
Mains borne 'clicks' and radio signal pick-up have troubled a number of users, and an almost complete solution to both of these problems, in conjunction with the above, is given by decoupling the earthy side of the inputs to chassis (either at the input sockets or where the earthed input cable screens connect to the pre-amp PCBs) by a small ceramic disc capacitor (0.01–0.05 μF), and by connecting a 330 pF capacitor between base and emitter of the BC 109 (or BC 184) in the tone control Liniacs

I should also have mentioned that the standard technique for preventing clicks on switching filter or tone control capacitors, where this is troublesome, is to connect high value resistors (of 3–10 megohm order) across the appropriate switch contacts, to prevent a sudden voltage pulse as the capacitor charges when the switch is closed.

Flattening the 75 W Amplifier Filter Response

HAVING RECENTLY completed a Linsley-Hood 75W amplifier, the following point has come to light and may be of interest to other readers.

The high-pass 'H' filter has the characteristic rise in response before cut-off as explained by Mr. Linsley-Hood. The overall response is made flat by preceding the filter with an RC circuit which is formed by the 10K volume control and the 0.22 μF coupling capacitor from the first Liniac.



If, however, the circuit is broken after the function switch to provide a tape recorder monitor point as suggested, the RC circuit is broken. The input to the tape recorder will have a flat response provided that the input impedance of the tape recorder is greater than about 40K, but the output from the tape recorder back to the amplifier volume control will suffer a bass boost due to the high-pass filter not being preceded by the requisite RC circuit.

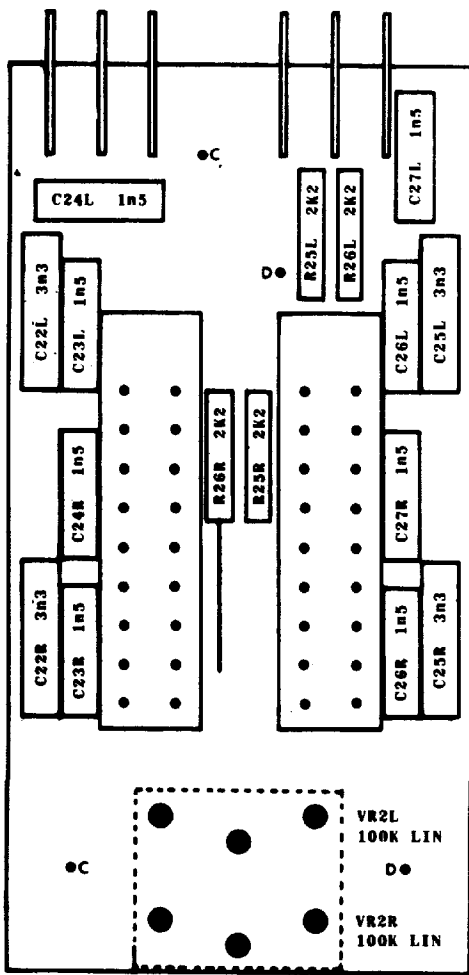
I have made the circuit changes as on the attached diagram to place the RC circuit after the volume control and give a flat response under all conditions; this has been confirmed by measurement.

Construction

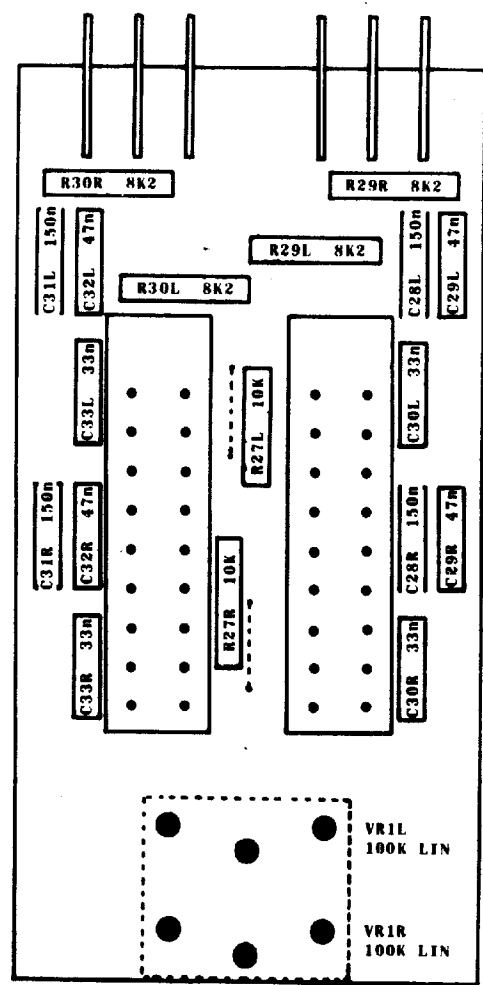
The order of assembly is as below. All references to left and right assume the unit to be with the front panel facing the constructor.

The correct size of bolt to be used is given in the 'Screwing Table'. The 1st size is Imperial and the 2nd size is the Metric size which may alternatively be supplied. Please note that dependent on the supply position longer than listed bolts may be provided in the kit. If in doubt about the position of any component then turn to the back cover of these notes and look at the photograph of one of our prototypes but do please remember that improvements are continuously being incorporated into our products and your kit may be slightly different.

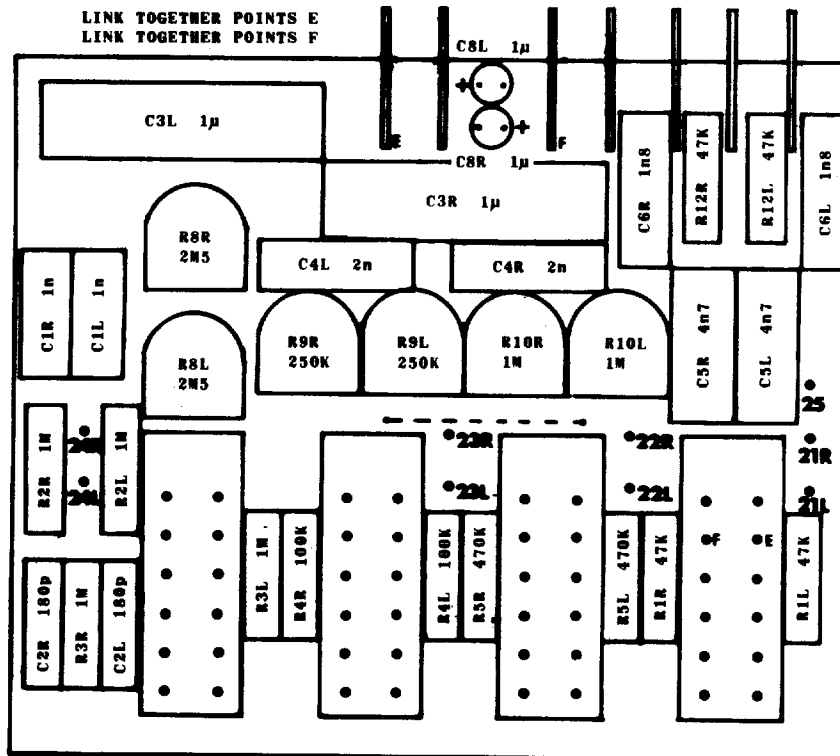
1. Fit the 4 DIN input sockets to the inner face of the left side of the rear of the chassis.
2. Fit the 2 way (loudspeaker O/P) DIN sockets to the rear of the chassis. Press until they lock into place.
3. Fit the DIN tape monitor socket above the O/P sockets.
4. Fit the mains input socket and fuse holder to the rear of the chassis. Place 1A5 antisurge fuse into the fuse holder.
5. Fit the mains transformer to the chassis with the wires coming out at the rear. On the bolt nearest the mains input socket fit 2 solder tags. These will be the sole earthing point on the chassis.
6. Fit the terminal block behind the mains transformer. The screw goes between sections 2 and 3 (counting from left to right).
7. Press 4 PCB support posts into the holes to the left of the Transformer.
8. Fit 2 capacitor clips to the chassis to the left of the PCB posts. Press the power supply capacitors (3300 μ F 63V) into the clips. The rear one is referred to as C103, the front one as C104.
9. Fit the slider control and jack socket (temporary fitting only) to the front of the chassis.
10. Fit the Mains + Phones switch to the right flange of the chassis. In the interest of safety, stick a piece of insulating plastic over the hole adjacent to the mains switch contacts.
11. Check the length of the potentiometer shafts, if necessary shorten to 18mm from the front face of the controls.
12. Assemble the printed circuit boards following the layout diagrams and noting the points below.
 - a) Soldering must be carried out carefully as hasty work here will always result in unreliability and examination of kits returned for servicing always reveals shortcomings in this area. Our recommended procedure for soldering is:
 - i. insert the component into the board
 - ii. bend its leads slightly — no need to flatten them
 - iii. trim its leads to 2mm from the board with cutters
 - iv. proceed to next component until all are inserted
 - v. hold solder in one hand and soldering iron in the other
 - vi. heat the component lead and circuit track simultaneously with the iron
 - vii. keeping the iron on the joint apply solder and ensure that the solder flows COMPLETELY over the joint.
 - viii. proceed to the next joint until all are soldered
 - ix. remove the flux from the joints by brushing with a stiff brush (e.g. a toothbrush) and a suitable solvent — acetone or cellulose thinners will work but keep away from the components as it could remove the markings from them and possibly damage them (particularly the polystyrene capacitors).
 - x. check the board when dry under a bright light with the aid of a magnifying glass (this is standard practice on production lines) for fine wisps of solder and dry joints. Even after very careful soldering a surprising number of faults can be revealed by this test.
 - b) before fitting any of the electronic components to the boards fit the terminal pins (as shown by labelled dots on the layout diagram) and connectors. Double sided pins should be used for the power amplifier main boards and the selector board as these will enable easy making of connections to the track side of the boards. Elsewhere single sided pins should be used. Take care to ensure that the connectors are fitted squarely to the boards. After soldering slowly bring the boards together periodically heating the soldered joints to allow the connectors to move slightly into accurate alignment. **AT NO STAGE MUST ANY FORCE BE APPLIED** as this will deform the contacts and unreliable connections could result.



LINK TOGETHER POINTS C
LINK TOGETHER POINTS D
TREBLE BOARD

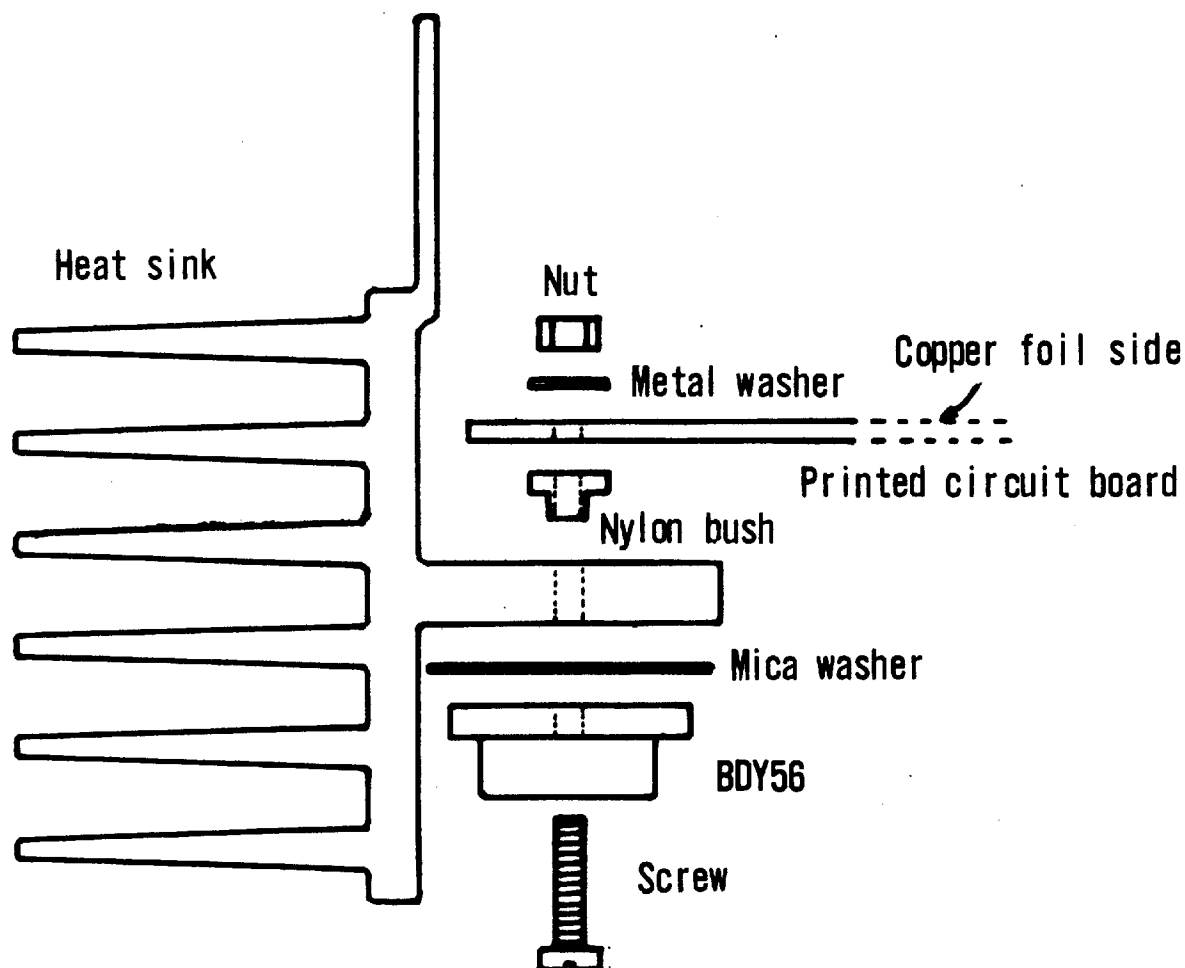


WIRE LINKS SHOWN AS DOTTED LINES
BASS BOARD

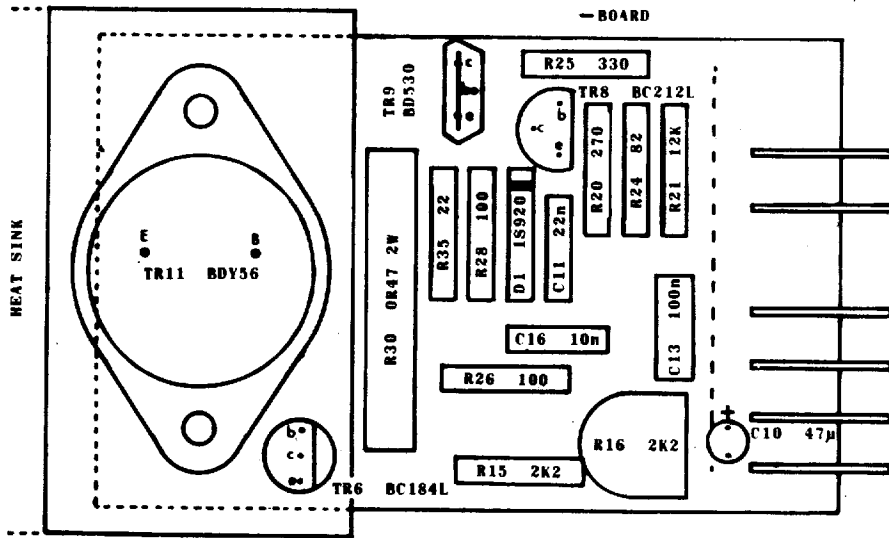


WIRE LINK SHOWN AS DOTTED LINE
SELECTOR BOARD

- c) the tags of the potentiometer may be either pre-formed in-line or staggered. If necessary carefully adjust with pliers, prior to fitting, any potentiometers which do not conform to the same arrangement as that of the holes in the boards.
- d) on the Bass, Treble and Filter boards the potentiometers are fitted to the track side of the board. Check that these controls are fitted squarely to the boards and then trim off the tags flush with the non-track side of the boards.
- e) before fitting the switches to the Bass, Treble and Filter Boards, remove the metal frame which holds the two sections together. After straightening the holding lugs on top of the switch, the sections can be easily prised out with a screwdriver. Fit these switches AFTER THE POTENTIOMETERS.
- f) do not fit FILT 2 board to the top side of the filter switch yet. If any re-work were necessary on FILT 1 board it could be quite difficult with the extra board in position. As this board holds only switch-click preventing resistors it is not necessary for testing purposes.
- male g) some makes of ~~selector~~ switch require spacers between the switch and the board. Where necessary 4 bushes are supplied in the switch pack. Fit these over 4 of the legs of the switch.
- h) the orientation of the transistors: follow closely the outlines on the layout diagrams.
- i) avoid damage, during soldering, to the transistors by standing them about 6mm above the board. This does not apply to the power transistors.
- j) the driver transistors (BD529, 530) are stood vertically. Fit a cooling tab to each one.
- k) fit the power transistors such that the heat sinks and PCB's are joined together according to the diagram below. Before assembly, check that there are no burrs around the holes in the sinks and after assembly check the insulation between the cans of the power transistors and the heat-sinks. A little silicon grease should be smeared onto both sides of the mica washer.

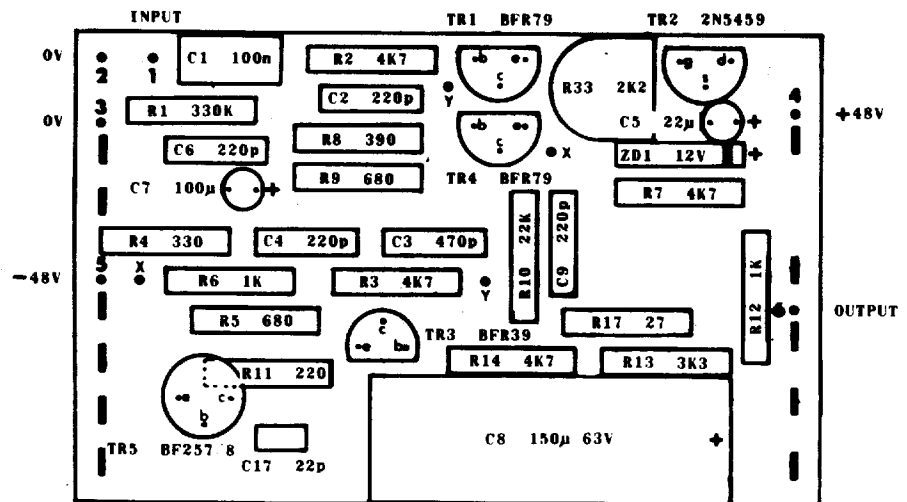


R16 Start fully,
anticlockwise
= high in value.



adjust for 80 mA
standing

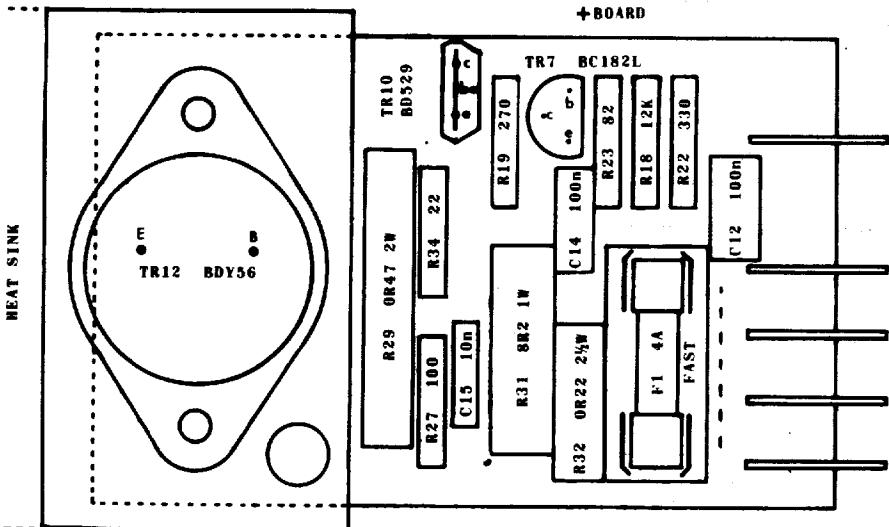
offset



LINK TOGETHER POINTS X
LINK TOGETHER POINTS Y

MAIN BOARD

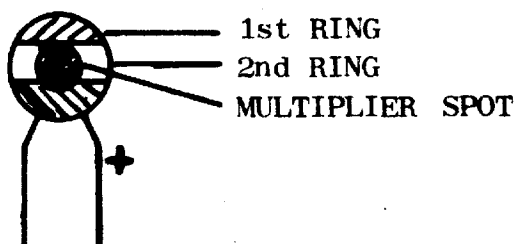
R33 Start fully
clockwise



WIRE LINKS SHOWN AS DOTTED LINES

Power amplifier boards

- l) fit the BC184L of each power amplifier after fitting the heat sink to the PCB. Smear silicon grease onto the transistor before pressing it into the hole in the sink.
- m) the cooling clip of the BF257 is anodised, which provides some electrical insulation, however, this should not be relied upon and the clip should be fitted with the gap next to the connectors, to prevent accidental contact.
- n) the polarity of the diodes: the end of the device with the stripe or start of the series of stripes, indicates the cathode (the end conventional current flows from). On the zener diodes the stripe indicates the positive end.
- o) electrolytic capacitors must be fitted with correct polarity. Where the positive end is not marked with a + sign then a ring around the can indicates the negative end. If the positive lead of the tantalum bead capacitors is not marked with a + sign then it is indicated as below as also is the colour code which may be used.



1st BAND 2nd BAND MULTIPLIER

COLOUR	1st & 2nd BAND/RING	MULTIPLYING FACTOR
BROWN	1	10
RED	2	100
ORANGE	3	1000
YELLOW	4	10000
GREEN	5	100000
BLUE	6	
MAUVE	7	
GREY	8	
WHITE	9	
BLACK	0	1
GOLD		1/10

- p) the striped, polycarbonate capacitors use the same colour code as resistors, reading from top to bottom the value is given in pF. Ignore all stripes after the first 3.
 - q) when fitting the capacitors to the boards rotate them to make visible the value of the components as this will simplify checking after construction.
 - r) the values of the pre-set potentiometers may not be marked so check with a multimeter, but bear in mind that these are wide tolerance components and some deviation from nominal values is to be expected.
 - s) where a dotted line between two points is shown on the layout diagram, the points should be joined together with short lengths of wire.
13. Connect in the transformer. The colour coding is as below:

0V	brown	33V	blue
117V	red	0V	white
0V	orange	33V	blue
117V	yellow	25V	mauve
		0V	black
screen	pink or green	25V	mauve

before soldering to the leads, the enamel must be carefully scraped off and the conductor tinned. Removal of the enamel can be speeded up by carefully warming the end of the wire in a flame before scraping. When the 0V leads are shortened it is **ESSENTIAL** that the 2 conductors in the sleeve are carefully soldered together. Unless this is done the winding will appear to have a discontinuity. After soldering, check the connections with a test meter across each complete secondary.

For 117V operation solder brown to orange wire and red to yellow wire.

screw red + yellow to term. block section 5
screw brown + orange to term. block section 4

For 234V operation, solder red to range wire.

screw red + orange to term. block section 1
screw yellow to term. block section 5
screw brown to term. block section 4

14. Solder a length of red wire between live of mains input socket and end tag of fuse holder and also a length of heavy black wire between earth of socket and the earthing solder tag.
15. Strip back 10mm from a length of heavy 2 core cable, remove exposed braid, cover end of braid with 2mm sleeving (stretching over the end of snipe nosed pliers simplifies this), strip conductors and solder.
 - red to bottom rear tag of mains switch
 - blue to bottom front tag of mains switch

far end: strip back end 30mm, twist braid together, trim to 6mm, solder to earth of mains input socket, strip red and solder to side tag of fuse holder, trim blue to 10mm strip and solder to neutral of mains input socket.
16. Strip back 10mm from a length of heavy 2 core cable, remove exposed braid, cover end of braid with 2mm sleeving and solder.
 - red to top rear tag of mains switch
 - blue to top front tag of mains switch

far end: strip back end 30mm, twist braid together and tin with solder.

 - screw red to term. block section 5
 - screw blue to term. block section 4
 - screw braid to term. block section 3
17. Solder transformer black, pink (or green), white leads to the earthing solder tag.
18. Press the power supply PCB onto its posts and connect to it following the table below.

transformer blue	1 to pin 17
transformer blue	2 to pin 19
transformer mauve	1 to pin 21
transformer mauve	2 to pin 22

 - black wire: pin 20 to earthing solder tag
 - heavy red wire: pin 15 to C103 + 've tag
 - heavy blue wire: pin 16 to C104 - 've tag
 - heavy black wire: C103 - 've tag to C104 + 've tag
 - heavy black wire: C103 - 've tag to earthing solder tag
 - black wire: terminal block section 3 to earthing solder tag.
19. Solder a length of thin red wire to the longer leg of the LED and a length of thin black wire to the shorter leg. Cover joints with 1mm sleeving, twist the wires together and at the far end solder the red to power supply pin "LED" and the black to pin 20. Leave the LED free in chassis for the time being.
20. Strip back mains cable 6mm, strip ends and solder to mains feed plug observing standard colour coding i.e.

Neutral	blue
Live	brown
Earth	green/yellow

feed cable through cover of plug and fit cover onto plug. Fit own choice of mains plug (not supplied in Kit) to far end of cable.
21. Plug in mains lead, switch on and check the voltages with respect to chassis. These are approximately:

pin 15	+	48V
pin 16	-	48V
pin 23	+	20V
22. Turn off power, remove mains lead and discharge the capacitors by connecting in turn between the above points and the chassis a resistor of about 100 ohms (supplied in pack 13) for a few seconds in each case.
23. Fit the heat sink/PCB assemblies to the back of the chassis. The PWR + board of each power amp. set is on the left.
24. Fit the left power amp. main board and connect power to the module as below:
 - heavy black wire: MB pin 3 to C103 - 've
 - heavy blue wire: MB pin 5 to C104 - 've
25. Connect between MB pin 4 and C103 + 've, with red probe on C103, a multimeter on 300mA scale. Connect a second meter, on 10V scale, between chassis and MB pin 6. Rotate R16 fully anticlockwise and R33 fully clockwise. The meters should now read 30mA and less than 1V. Adjust offset voltage to zero with R33 and slowly rotate R16 whilst watching the current reading. When the current reaches 80mA stop and wait. As the current creeps up slowly turn back R16 to

keep it constant. Do not be in a hurry at this stage as impatience or clumsiness could easily lead to a bill for replacing over-cooked transistors. After the current has shown no signs of changing for about ¼ hour turn off the power, unplug mains lead, wait until the current has fallen to zero, remove current meter and solder a length of heavy red wire between MB pin 4 and C103 + 've.

26. Repeat '24', '25', for the right channel.
27. If when carrying out the above the current is found to be high then switch off immediately to reduce the risk of creating further faults and re-check the assembly. If you do have this problem or the offset voltage is too high then the most likely faults are: tracks shorted with fine wisps of solder, dry joints and shorts between power transistors and heat sinks. Less likely are component faults but possibilities here include resistors cracked at lead outlets as a result of bending, polystyrene capacitors shorted through melting and open circuited base of small transistors resulting from thermal shock. Quality control of production of power devices is very tight and original faults here are most unlikely but if there has been a fault on the board leading to high power supply current being drawn then damage to them is likely and then will usually take the form of a collector-emitter short. Should there have been a fault resulting in a large offset voltage then C7 should be checked for damage from overloading it. Whatever fault has been found, particular care should be taken in rechecking the rest of the boards as faults in power stages can too easily lead to chain reaction further faults. Do not forget to check the protection transistors as damage to these may not show up on testing the amplifier but if they have been open circuited then disaster could be incurred if the loudspeaker leads are shorted.

Approximate voltages on power amplifier PCB

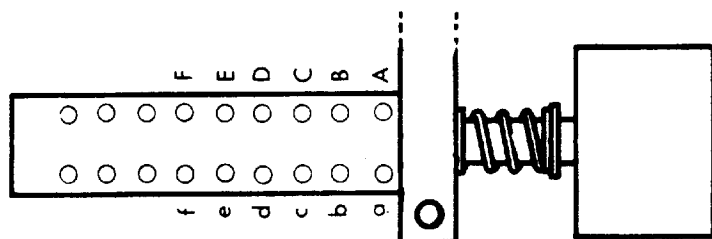
	Chassis to Tr1	base	0 volts
	Chassis to Tr1	emitter	+0.6 volts
	Chassis to Tr2	drain	12 volts
	Chassis to Tr3	collector	-0.4 volts
	Chassis to Tr4	base	0 volts
	Chassis to Tr5	collector	-1.1 volts
	Chassis to Tr9	base	-1.1 volts
	Chassis to Tr10	base	1.1 volts
	Chassis to Tr10	emitter	0.5 volts
	C104 - 've Tr3	base	1.2 volts
	C104 - 've Tr5	base	0.6 volts
	C104 - 've Tr9	collector	0.5 volts

-48V rail

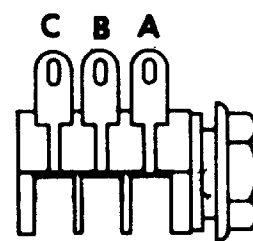
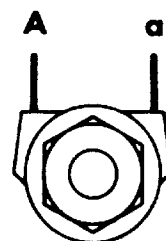
It sometimes happens that adjustment of R33 will not quite bring the offset voltage down to zero. This is due to variations in the parameters of the BFR79s and can be dealt with by interchanging TR1, TR4.

28. Complete the power amplifier wiring as below:

orange wire: left MB pin 6 to phones switch contact E
orange wire: right MB pin 6 to phones switch contact e
orange wire: left DIN O/P skt outer contact to phones switch contact D
orange wire: right DIN O/P skt outer contact to phones switch contact d
47R 2½ W resistor: JIC to phones switch contact f
47R 2½ W resistor: JIB to phones switch contact F
47R 2½ W resistor: JIC to JIA
47R 2½ W resistor: JIB to JIA
black wire: JIA to earthing solder tag

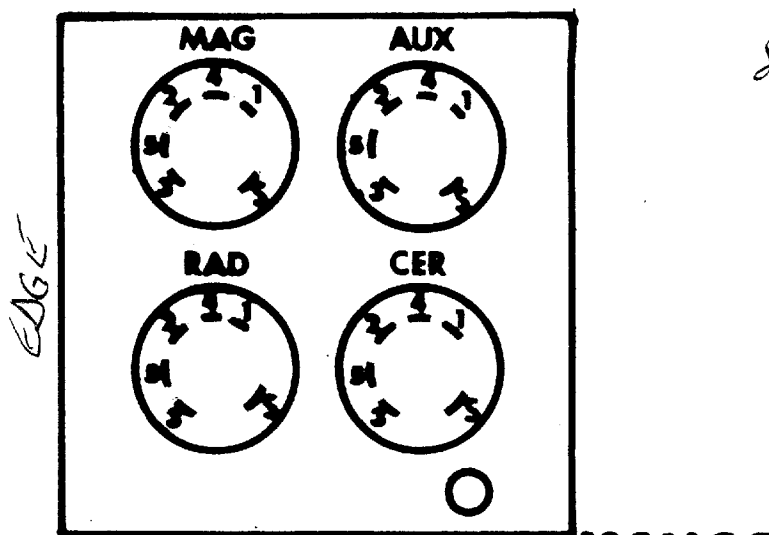


Push button switch



Jack socket

29. Fit the two screens (approx. 10cm x 4cm) to the chassis on the left side of each power amp. module.
30. Fit 6 6mm tapped spacers to the pre-amp mounting bracket (approx. 42cm x 4cm) on the same side as the flange and fit the pre-amp main PCB. Press a grommet into each 1cm wide slot of both the PCB and the bracket.
31. Fit 2 12mm tapped spacers to the chassis where the selector board will be fitted.
32. Place each of the small boards loosely in position.
33. Fit the main PCB/bracket assembly but do not tighten the screws.
34. Mate up the contacts, fit the selector switch to the tapped spacers, fit the 4 small boards to the front of the chassis by means of the nuts on the controls and then tighten up screws of the PCB bracket.



seen inside

Input DIN sockets

35. Connect in the pre-amp following the table below:
 - a) 2 way screened cable:
 - red to MAG DIN skt. 3
 - blue to MAG DIN skt. 5
 - braid to MAG DIN skt. 2
 far end:
 - red to SEL board pin 21L
 - blue to SEL board pin 21R
 - braid to SEL board pin 25
 - b) 2 way screened cable:
 - red to AUX DIN skt. 3
 - blue to AUX DIN skt. 5
 - braid to AUX DIN skt. 2
 far end:
 - red to SEL board pin 22L
 - blue to SEL board pin 22R
 - braid cut off and cover end with 2mm sleeve
 - c) 2 way screened cable:
 - red to RAD DIN skt. 3
 - blue to RAD DIN skt. 5
 - braid to RAD DIN skt. 2
 far end:
 - red to SEL board pin 23L
 - blue to SEL board pin 23R
 - braid cut off and cover end with 2mm sleeve

- d) 2 way screened cable:
 - red to CER DIN skt. 3
 - blue to CER DIN skt. 5
 - braid to CER DIN skt. 2
 far end:
 - red to SEL board pin 24L
 - blue to SEL board pin 24R
 - braid cut off and cover end with 2mm sleeve
 - e) join together MAG DIN skt. 2, MAG DIN skt. screen, AUX DIN skt. 2, AUX DIN skt. screen, RAD DIN skt. 2, RAD DIN skt. screen, CER DIN skt. 2, CER DIN skt screen with a length of bared wire.
 - f) 2 way screened cable:
 - red to pre-amp MB pin 26
 - blue to pre-amp MB pin 27
 - braid cut off and cover end with 2mm sleeve
 far end:
 - red to TAPE DIN skt. 1
 - blue to TAPE DIN skt. 4
 - braid to TAPE DIN skt. 2, screen
 - g) 2 way screened cable:
 - red to pre-amp MB pin 28
 - blue to pre-amp MB pin 29
 - braid to pre-amp MB pin 30
 far end:
 - red to TAPE DIN skt. 3
 - blue to TAPE DIN skt. 5
 - braid to TAPE DIN skt. 2
 - h) 1 way screened cable:
 - inner to pre-amp MB pin 31L
 - braid to pre-amp MB pin 33
 far end:
 - inner to left power amp MB pin 1
 - braid to left power amp MB pin 2
 - i) 1 way screened cable:
 - inner to pre-amp MB pin 31R
 - braid to pre-amp MB pin 33
 far end:
 - inner to right power amp MB pin 1
 - braid to right power amp MB pin 2
 - j) orange wire: slider pot contacts to PCB pins opposite (no crossing of the 3 wires)
 - k) red wire: pre-amp MB pin 32 to power supply PCB pin 23.
36. Fit a knob to the Mode switch and switch to S (stereo).
 37. Plug a speaker into the left channel output socket, plug a tuner or other signal source of similar level into the RAD socket, push in the 3rd switch from the left, set Bass, Treble controls to mid-way position, turn volume down to minimum, set all pre-amp pre-set controls to mid-way position, plug in mains lead and switch on.
 38. Gradually turn up the volume and check all controls to be functional. Turn down the volume, plug a magnetic pick-up into the MAG socket, press in the left switch and turn up the volume. Should the amplifier be heard to be clipping, turn down the volume and turn R14L anti-clockwise. If there is insufficient volume then turn R14L clockwise. Successively plug in the other inputs, set R8L, R9L, R10L to give consistent signals and turn the volume back down.
 39. Plug the speaker into the right channel output socket, repeat 38 for the right channel controls, plug a speaker into the left channel output socket and make further adjustments to the pre-sets, as necessary, to balance the two channels. At this stage the hum level will be quite high and will remain so until the front panel and cover are fitted.

40. Switch off, unplug mains lead and fit FILT 2 PCB to the filter control switch.
41. If either of the channels fails to produce an output then switch off, disconnect the leads from pins 31L and 31R and switch on again before making any voltage checks. When fault finding look for components in the wrong positions, tracks shorted with fine whisps of solder, dry joints, resistors cracked at lead outlets as a result of bending, polystyrene capacitors shorted through melting and open circuited base of transistors resulting from thermal shock in soldering. Unlike the power amplifiers there is little chance of chain reaction faults on a grand scale but if any of the tantalum capacitors have been subjected to reverse polarity or excessive voltage then check these for damage. If the use of a very insensitive pick-up results in insufficient volume even with R14 fully clockwise then a slightly lower value resistor (e.g. 47R) may be used for R13.

Approximate voltages on pre-amplifier PCB:

chassis to Tr1 base	0.6 volts	chassis to Tr6 base	8.2 volts
chassis to Tr1 collector	4.2 volts	chassis to Tr6 emitter	7.5 volts
chassis to Tr2 drain	8.2 volts	chassis to Tr6 collector	16 volts
chassis to Tr3 emitter	3.0 volts	chassis to Tr8 base	0.6 volts
chassis to Tr4 base	8.2 volts	chassis to Tr8 collector	6.2 volts
chassis to Tr4 emitter	7.5 volts	chassis to Tr9 drain	17 volts
chassis to Tr4 collector	16 volts	chassis to Tr10 emitter	5.0 volts
chassis to Tr5 emitter	17 volts		

42. Unscrew the front of the jack socket, press the black LED grommet into the front panel, press in LED and fit front panel by means of the screw under the Mode switch. Smear a little adhesive (epoxy type preferred) to each side of 3 of the jack socket fibre washers and refit the socket with these 3 washers between the socket and the chassis. When the glue has hardened unscrew the front of the jack socket, remove the front panel, glue plastic insulating sheet on the cover in the region of the power amplifiers, fit the cover and replace the front panel. Fit the knobs and taking care not to scratch the edges of the wood with the chassis, slide it into the cabinet and secure with two screws through the brackets in the rear of the cabinet.

Servicing

This kit has been designed to be extremely easy to service. Any of the boards can be removed simply for checking and rechecking without disturbing appreciably connections to other boards thereby making fault finding very straightforward and professional assistance is unlikely to be required. However, should it be necessary for the kit to be returned for problems to be sorted out, then please restrict the scope for product modifications by the Post Office by removing the case, knobs and slider control. Cut a stiff piece of chipboard or plywood a bit longer than the chassis + heatsinks, fix the chassis to the board with woodscrews through holes drilled in the chassis and pack firmly in a box with at least an inch of padding around the unit. Alternatively, modules may be sent for servicing. If this is done then send each one of the set i.e. all 3 of a power amp. or all 7 of a pre-amp. Unplug the boards before packing and remember that when the Post Office gets its corporate hands on your package it will throw it into a sack, throw more parcels on top of it and at various stages throw the entire sack into and out of their vehicles, so take great care in packing and use a VERY strong box. Details of servicing charges and turn-around times will be sent on request.

Tools required for constructing this kit.

- 1 Good quality miniature soldering iron preferably temperature controlled otherwise 15 watt only.
- 2 Pair of electricians pliers
- 3 Pair of pointed tweezers
- 4 Pair of wire strippers
- 5 Pair of side cutters
- 6 6BA/4BA open ended spanner
- 7 3/8" open ended spanner
- 8 Small hard bristled brush
- 9 Magnifying glass
- 10 8" Pozidrive screwdriver
- 11 Small standard screwdriver
- 12 2 Multimeters (10K/volt or better)