ELECTRONICS DIGEST

Volume 6 No. 3

Welcome to the Winter 1985 issue of Electronics Digest.

This edition is dedicated to and designed for that rarest of connoisseurs, the hyper-audiophile. Within these pages are gathered the best of the many splendid audio projects published in Electronics Today International during the past five years. There are no gimmicks, no fancy gadgetry, just down to earth honest audio equipment offering the sort of performance that, too often, isn't available from commercial equipment at comparable costs.

Many hundreds of kits for these amplifiers, pre-amps and speaker systems have been sold and constructed during the past five years. Unfortunately complete kits are no longer available in some cases, but as far as possible we have arranged sources of supply for the more difficult-to-find components and these are noted in the Buylines. In other cases components or, with some of the more recent projects still, kits are readily available.

The projects selected for this issue have been designed by engineers with years of experience in audio design and engineering. If you are a regular reader of Electronics Today International then the names mentioned on our cover will need no further introduction. If you are unfamiliar with these men and their work then be assured that their designs are known and respected by thousands of people like yourself — hyper-audiophiles one and all.

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CONTENTS

This volume contains reprints from Electronics Today International, also published by Argus Specialist Publications. Other work permitting, we are prepared to attempt to answer readers' queries over difficulties in managing to get these projects working. However, we expect readers to make reasonable efforts themselves, such as checking suppliers' advertisements in other publications for sources of components, and making and attempting to interpret diagnostic measurements. We also expect readers to be prudent in their choice of constructional project, bearing their own real capabilities in mind. All readers' queries must be written and accompanied by an s.a.e. — we are not able to answer telephone enquiries, as these seriously disrupt our work. We are not able to advise on modifications.

John Linsley Hood's

AUDIO DESIGN AMPLIFIER

Many amplifiers have claimed to be the best ever published for the home constructor; but take it from us — this is.

In an earlier series on Audio Circuit Design I explained, in as simple a manner as I could, how circuits were designed, and their values specified, to do a specific job — in an engineering sense. Fifteen years or so ago, this would have been all that the user would have asked, and he would probably have been delighted with the performance given by what would be considered a run-of-the-mill design exercise. However, things have now changed.

As I mentioned in the article which dealt with audio amplifiers, I have an ambivalent attitude towards the whole 'hi-fi' scene, in the sense of that conspiracy which appears to exist between the editorial staff of 'hi-fi' journals, and the manufacturers of 'hi-fi' equipment, by which fulsome praise is lavished only upon rare items of audio exotica — which must be very expensive, in addition to being relatively infrequently seen — and staff writers are flown half way round the world to see, hear and inwardly absorb the latest propaganda in the cause of the most recent technological miracle.

Meanwhile, the bulk of the readers of the magazines continue to use, and frequently to enjoy, the equipment which they bought at a modest price from their local dealer or discount warehouse, and which never received any rave reviews from anyone, except a consumer magazine which said it was good value for money!

I am not so calloused, mentally, that I cannot recognise that some of these hi-fi exotica are indeed very good, and well designed and made, to boot. However, this presents a problem to the designer of any kit which is to be described in an electronics constructional magazine. Whatever it is, it must provide an incentive to the would-be constructor. Not only must it be sensible value for money, but it must also offer some quality advantage over the nicely made and prettily finished units offered at such tempting prices in the local High Street.

One advantage, which it is perhaps a little unkind to stress, is that the things you build yourself are repairable by you — the others may not be. However, if you are known, or mistakenly thought, to have any skills in the repair field, you are likely to have to fend off a kind of fan-club of friends and relatives who have bought some pretty tin-ware a few years ago, and now can't find a dealer who wants to know about it.

The other advantages may be those which are concerned with audio quality, in its various aspects. You may be able to include amenities or facilities which only you may want, but which are absent from the less expensive commercial gear, or it may be that you can gain some advantages in sound quality. This latter task is made a bit easier, certainly in respect of the equipment at the cheaper end of the market, by the fact that the need for a low sale price forces the manufacturer into the use of specialised, custom designed, circuit hardware which does an adequate though not marvellous job.

Many of the circuit designs which have been offered for the DIY constructor have relied for their appeal on the provision of a lot of electronic facilities, and I have been down this road myself, as testified by the JLH domestic preamp, published elsewhere. However, while it was fun to design and build the bits of gadgetry included in this design, the fact remains that most of these facilities are very seldom used. So, since I know that I can dispense with most of these, with very little real loss, and since I suspect that I could do those things which remain just a little bit better than I have done them so far, my intention here is to offer a fairly simple design in which all the small practical quality improvements are incorporated, in the hope that the final unit, within the power budget decided upon, will equal or exceed in sonic quality anything available anywhere else, at any price.

This may sound both vain, and impracticable; well perhaps it may be. But my problem, in the evolution of this design — and the problem of any other designer — is that unless one has a reasonable chance of matching the quality of the best, it is hardly worth while cluttering up the printed pages with yet one more design.

Design Philosophy

Like many of my readers, I suspect, I have read a lot about the recent tend in hi-fi thinking, in respect of Class A operation, and valves, and enormous power outputs, and the importance of connecting wires the right way round. Since I know that the people who espouse these causes are neither foolish nor easily led, I have had to try to work out a rational explanation for this collective attitude, in the hope of some design guidance emerging from this. For what it is worth, here it is.

Most of the audio quality judgments on audio amplifiers and ancillary equipment, made by the writers on this subject, are made on the basis of extended listening trials, most of which are at very high sound levels, with music of a type which has relatively few quiet passages. Valve amplifiers have the great advantage, because of their inherent tendency to 'soft' clipping, that when they are driven into overload they sound much less awful than transistor designs which have a 'brick-wall' clipping characteristic. Also, with valves, their distortion products — of which there are usually quite a lot — are mainly 2nd and 3rd harmonic: these can, curiously, tend to enhance the sound of certain music, to make it sound 'richer'. Also, for practical reasons, not a lot of NFB can be employed, which makes LS load compatibility less demanding of design. Finally, the output transformer, which does so much to impair the electrical quality of the amplifier, does at least ensure that it can push a lot of current into a low impedance LS load.

Since most of the reviewers' auditioning is apparently done at very high sound levels, the preference for high powers is also understandable.

The case for class-A is harder to fault. With junction transistors, particularly, the sluggishness in operation of the higher current types makes crossover distortion an ever present problem, in class-B (no quiescent current) or class-AB (some zero signal level quiescent current) operation. Class-A (standing output stage current the same for zero or maximum power) operation makes for much better power output transistor HF response, and also has the big advantage that the HF characteristics of the power transistors are just as good at low signal levels as they are at high ones.

However, a class-A amplifier is unavoidably inefficient, with efficiencies in the range 25-30% being normal. This means that an 80 watts/channel class-A stereo amplifier must dissipate, perhaps, 640 watts of heat. This either implies enormous heat sinks, to keep the operating temperature down to 80 to 100°C, or fan cooling, which is noisy. Either way, such an amplifier will make quite a contribution to room heating. Now think of what life would be like with a 200 WPC class-A system! Sliding-bias class-A systems have often been tried, but never liked.

MOSFETs To The Rescue

So far as I am concerned, the availability of power MOSFETs is a nearly complete solution to the power amplifier design problem. The recent design types of this kind are so fast that the difficulties of output stage sluggishness are abolished, and the nature of the device ensures that the HF response is the same at all drain current levels, thereby avoiding the normal class-AB junction transistor problem of subtly different sound quality at low and high sound levels.

Power MOSFETs do have design problems, which is why, inspite of their many and conspicuous virtues, in comparison with junction transistors and transformer coupled valves, relatively few commercial designs exploit these qualities.

As for the third hi-fi fetish, connecting wires, this is less to do with the wires themselves than with the connections made to them, in interconnecting the component units of the complete audio system. It makes little sense to go to a lot of trouble in the circuit design, and then put the results in jeopardy by the use of cheap connectors. For the LS circuit, in particular, there is a great deal to be said for solid screw-down terminals.

So — to summarise the design thoughts so far. The amplifer should employ power MOSFETs in a suitable design, and have good quality wiring connectors. The preamp should have only the necessary facilities, but those that are there should be as good as possible.

The Necessary Facilities

In my own experience, a good quality RIAA input stage, an input selector switch, a volume control, a balance control, a rumble filter and some reproducible means of modifying the relative levels of bass and treble response — where both of the latter circuits must be capable of being switched out — are all that are really essential in the preamp. However, a separate class-A headphone amp is a desirable addition, and if this is included in the preamp, this unit can work on its own. You will infer from this that I prefer the preamp and the power amp to be separate units. This form of construction does make life a lot easier for the constructior. Also, for reasons of practical convenience, I think that it is sensible to house the moving coil head amp, if used, in its own separate enclosure. It can still be powered from the preamp to avoid the inconvenience of battery replacement.

So far as the power amp goes, I

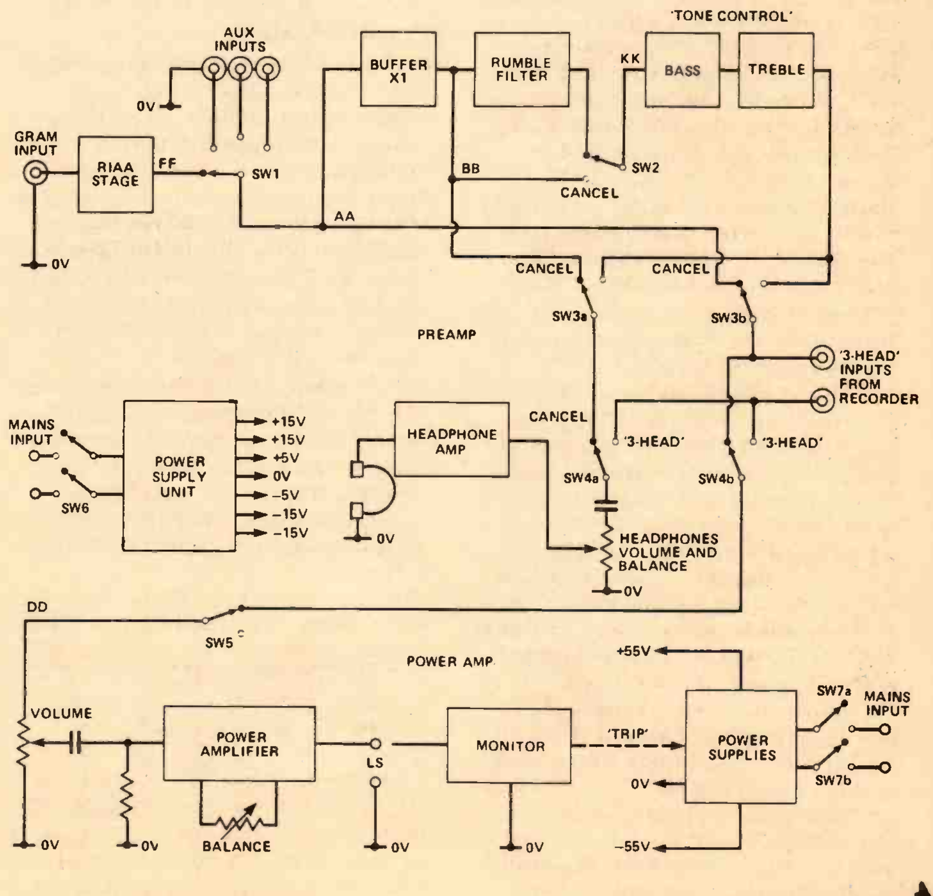


Fig. 1 Layout of the pre-/power amp system.

think that there is a lot to be said for designing this so that it has enough sensitivity (say 150mV for max output) that it can receive signals directly from ancillary units (cassette recorders, tuners and the like) without further amplification — on the grounds that the less one handles signals, the better the final result is likely to be. This implies that the volume control will be at the input to the power amp, and the channel balance control will also require to be included somewhere within the power amp. Fortunately this is easy to organise.

This arrangement also implies that the preamp needs to have provision for a 'straight through' path from the input selector to the power amp input position. The final layout is shown in Fig. 1. A minor point, occasionally exhorted by the hi-fi pundits, is that the overall system is non-inverting in signal phase, and that switching out sections does not affect the overall phase polarity.

Both the power amp and preamp are operated from stabilised power supplies. In the preamp case, these are voltage regulator ICs, and in the power amp, where higher voltages and currents are necessary, a discrete component unit is employed. LS protection is provided without the need for relays or fuses in the output line to the LS. (There are good relays with gold plated contacts, and some fuse holders also are soundly made, however, if one can do without them this must be better). This is accomplished by monitoring the DC offset on the LS line, and switching off the PSU electronically if this exceeds some predetermined value, averaged over a fraction of a second.

This doesn't confer on the circuit the useful facility of disconnecting the LS for a few seconds, following switch-on, to remove the normal switch-on 'plop', for which a relay is so useful, but it is possible, as an option, to connect a clamp circuit across the power amp input, to hold this down to the 0V line for a few seconds after switching on. A junction FET will do this job very well, since it conducts, bidirectionally, until a voltage is applied to its gate to cut if off, when it will become a very good quality open circuit.

This concludes the outline of the design 'architecture'. The only other point which seems worthwhile exploring before we get down to the detailed consideration of the circuitry is what kind of gain blocks we should use. I think that some of the new '741' pin connection op-amps such as the TL071, and the LF351 (or the PMI OP27 if money is less of a consideration) make excellent audio gain blocks. Moreover, for the convenience of the stereo enthusiast, these are available as dual op-amps in their TL072 and LF353 versions. Although I indicated earlier in this series that I thought that it was possible to do this job a little bit better by the use of discrete component 'gain blocks', the advantage is small, and the IC is simpler and more cost effective.

So, what I propose is that these 'discrete component' units should be restricted to the two input gain blocks in the RIAA stage, where their qualities may best be seen. In the case of the tone-control stage, or the rumble filter, my feeling is that if it is necessary to use these signal modifying elements there is an implied admission that the signal is less good than one would wish anyway, so the very slight tonal penalty (and it really is very small) which would be paid by using an op-amp is not likely to be enough to justify more complex and costly alternative circuitry.

RIAA Stage

Although in normal circumstances, I would prefer a two-stage input system, using two consecutive active stages, there is no tonal difference between the use of a passive and an active network as the second stage integrator — it is simply that the passive network will have an attenuation of 10 at 20kHz, which require the first gain block to have an output at least 10 times greater at this frequency.

However, in the circumstances of this preamp where the output signal level required for maximum power amplifier output is only 150mV, this is not a significant problem, especially in view of the 10V RMS output capability of the gain block used. So, since using the passive net-work to give the second part of the RIAA attenuation curve will save 16 transistors, it seems a sensible move.

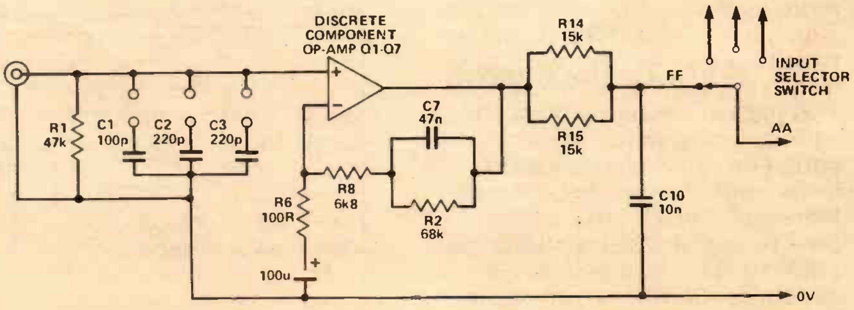
The advantages of breaking down the composite 30Hz - 1kHz, 1kHz - 20kHz RIAA equalisation curve into two separate stages were discussed in earlier articles of mine, and by other writers elsewhere. Unfortunately, there still remain designers to be convinced, I am sorry to say, and it is difficult, by remote control, to have them listen to the two choices so that one may say 'There you are. It does sound better, doesn't it?'. So, may I make the argument that very seldom, in human experience, can one make any device do two different jobs at once with as good a performance as two separate more specialised units. Why therefore should one expect a single gain stage to do two separate equalisation functions simultaneously with equal — let alone better — results than when these functions are separated.

The final gramophone PU input stage therefore becomes as shown in Fig. 2 and its complete layout as shown in Fig. 3, with the gain block instead of the schematic op-amp diagram.

Since for optimum results many PU cartridges require a measure of capacitative loading in addition to the 47k input resistor, a group of three capacitors are mounted on the board to allow the choice of input capacitance values from 100pF to 540pF by simple bridging of pins on the board.

(A reader has subsequently pointed out that the RIAA stage will only work as described into an infinite impedance load: a suitable buffer amplifier is described in Part 4).

Fig. 2 Simplified circuit of the PU input stage.



PARTS LIST — RIAA STAGE

RESISTORS (all 2% 0.4W metal film)	
R1,101	47k
R2,102	10k
R3,5,103,105	15k
R4,104	2k7
R6,11,106,111	100R
R7,107	100k
R8,108	6k8
R9,13,109,113	47R
R10,110	680R
R12,112	68k
R14,15,114,115	15k
CAPACITORS	
C1,101	100p polystyrene
C2,3,102,103	220p polystyrene
C4,104	470μ 6V3 low ESR electrolytic, tubular
C5,105	470μ 6V3 low ESR electrolytic, PCB mounting
C6,106	100p polystyrene
C7,107	47n polycarbonate, 1%
C8,9	470μ 16V electrolytic
C10,110	10n polycarbonate, 1%
SEMICONDUCTORS	
Q1-5,7,101-105, 107	BC416
Q6,8,106,108	BC414
MISCELLANEOUS	
PCB, wire, etc	

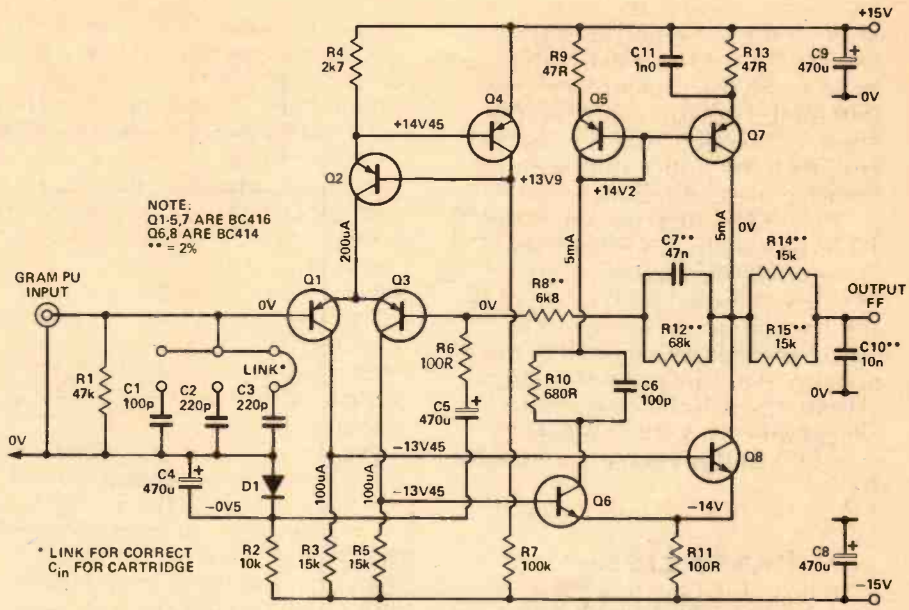


Fig. 3 Full circuit of the PU input stage; note that if you're cutting costs by using 5% resistors, components marked with ** must be 2% or better.

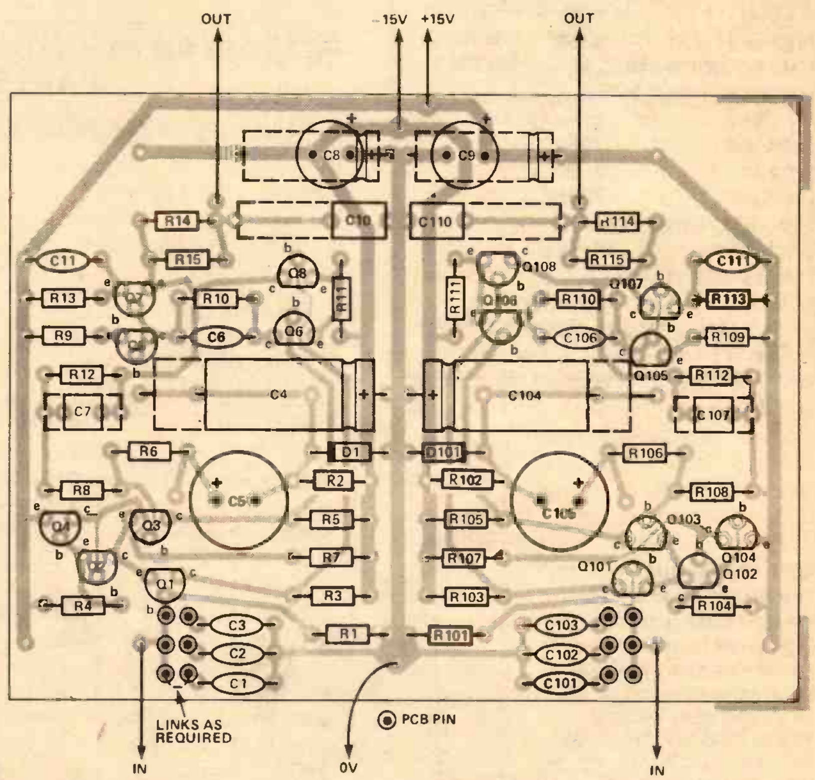


Fig. 4 Overlay diagram of the PU input stage — note the allowance for different capacitor sizes.

Tone Control Stage

My experience with my rather more elaborate domestic preamp which has a tone control circuit which is capable of modifying the frequency response, in a series of 3dB plateau steps, up or down, at various frequencies, has encouraged me in the belief that this is the kind of tone control to have. However, I do not use all the possibilities it offers, and most of the time it is switched out of circuit. So, in the light of exerience I feel that a simpler, cheaper, and easier to build unit would have served me just as well.

My second thoughts on this stage are shown in Fig. 5. Because two inverting stages are employed, in cascade, the system is non-inverting in phase, and the cancel switch should make no audible difference to the sound if this stage is swtiched in or out when in its flat response position. Each switch, S2 - S9, generates a click-free modification to the frequency response, in a controlled and reproducible manner. (Note, each 'lift' switch operation should be accurately cancellable by the equivalent 'cut' button, to restore the status quo, as a test of the correct operation of this stage.)

I have only aimed at a small (3dB) step increment or decrement in frequency response given by this stage, because the intention in this design is to compete in the upper audio bracket. If a very large treble or bass cut or lift is needed, it would seem to imply that there is is something badly amiss elsewhere in the system. It would, I think, be better to try to remedy this where it exists than to try to make the preamp compensate for it. The 3dB value has the merit, in practice, that is is just big enough to be noticeable, without being so big that it is intrusive.

Two LF353 dual op-amps are used to operate this circuit, and these are fed from the + and − 15V lines derived from a pair of IC voltage regulators, and used to power the remaining units in the preamp. Push-on, push-off switches actuate the frequency steps and cancel functions. The frequency response of this circuit is shown in Fig. 7.

PARTS LIST — BUFFER/FILTER

RESISTORS (all 2% 0.4W metal film)	
R1,101	100k
R2,102	330R
R3,4,5,103,104,105	6k8
CAPACITORS (all polycarbonate)	
C1,101	470n
C2,102	100n
C3,103	2µ2
C4,104	220n
SEMICONDUCTORS	
IC1,101	LF353
MISCELLANEOUS	
SW2	3p (min), 2w push-on, push-off
PCB, wire, etc	

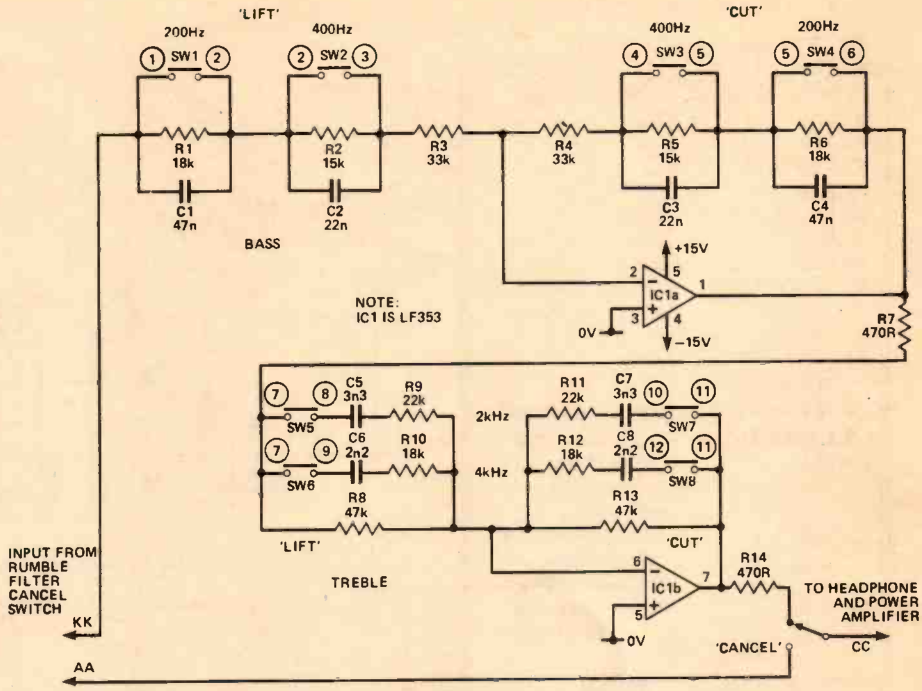
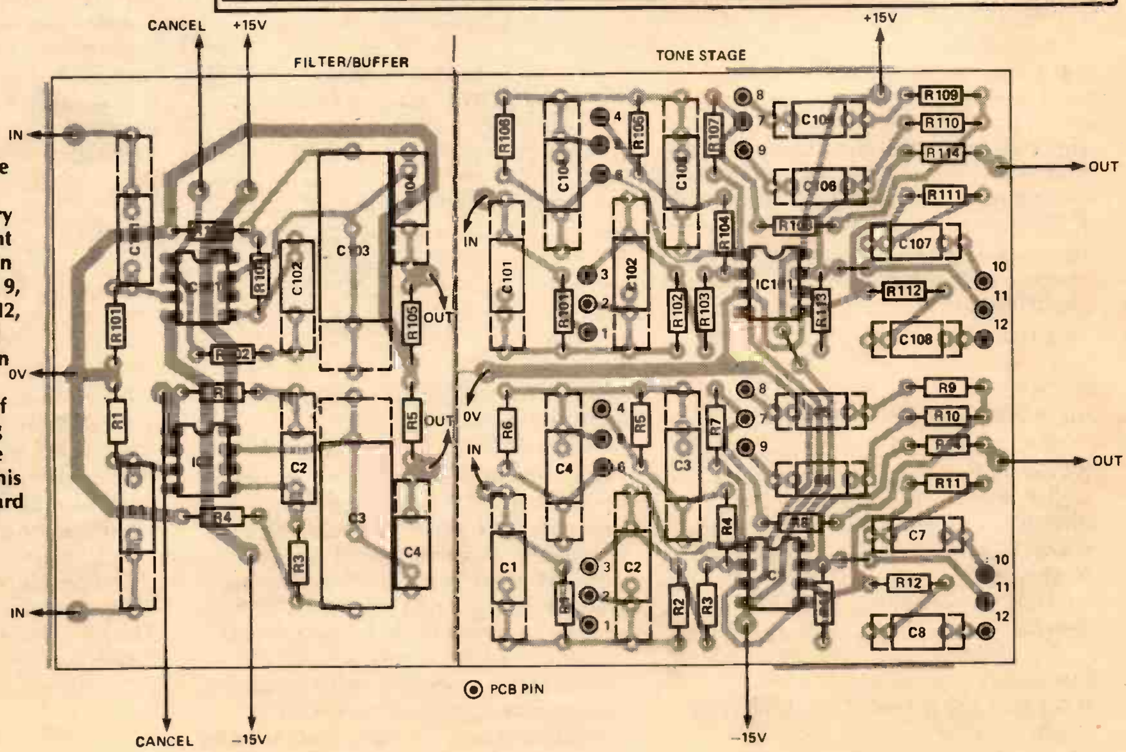


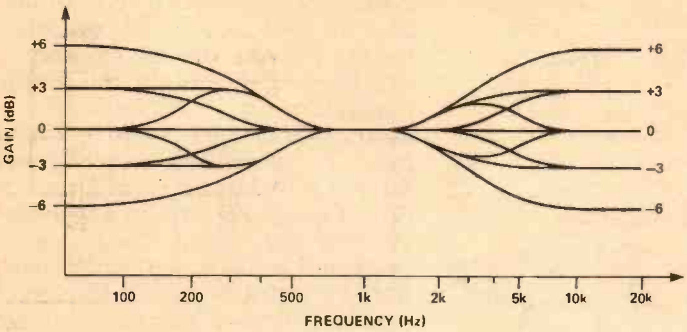
Fig. 5 Circuit diagram of the tone stage.

PARTS LIST — TONE STAGE

RESISTORS (all 2% 0.4W metal film)	
R1,6,10,12,101, 106,110,112	18k
R2,5,102,105	15k
R3,4,103,104	33k
R7,14,107,114	470R
R8,13,108,113	47k
R9,11,109,111	22k
CAPACITORS (all polycarbonate)	
C1,4,101,104	47n
C2,3,102,103	22n
C5,7,105,107	3n3
C6,8,106,108	2n2
SEMICONDUCTORS	
IC1,101	LF353
MISCELLANEOUS	
SW5-12	DPST panel-mounting switches.
PCB, wire, etc.	

Fig. 6 Overlay diagram of the buffer/filter stage (left) and the tone stage (right). Be very careful not to get component numbering confused! N.B. In the tone control section R8, 9, 10 should be linked to R11, 12, 13 and R108/109, 110 should be linked to R111, 112, 113. In the buffer filter stage the connection between pin 4 of IC2 and −15V rail is missing on this overlay diagram. The foil pattern reproduced in this issue is correct, as is the board from our PCB Service.





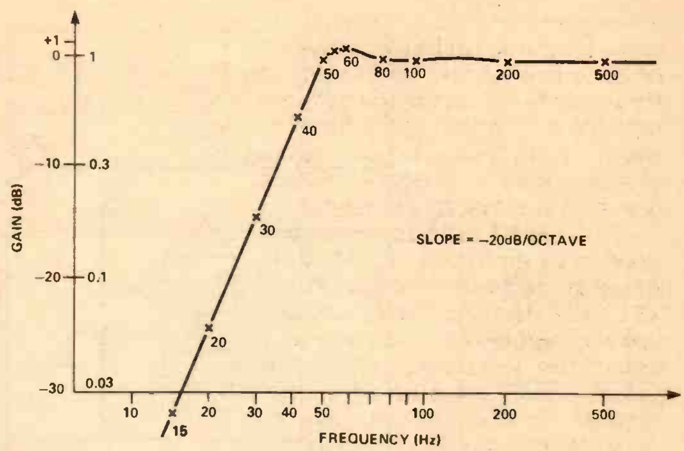
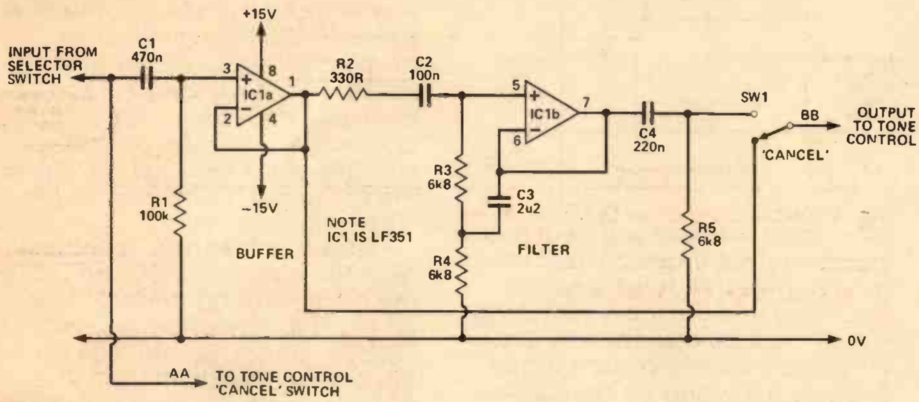


Fig. 7 (above) Response of tone stage.
Fig. 8 (below) Circuit of the buffer/filter.
Fig. 9 (right) Measured rumble filter response.



The Rumble Filter Stage

Some otherwise very good records can suffer from rumble in the recording, and even those played by the BBC are not always free from this fault. If one is lucky enough to have LS units with a good low frequency response, this can be a very irritating problem. My practice, in the past, has been to choose the LF turn-over point at 30-32Hz, on the grounds that, with a −20dB per octave filter slope, this should adequately deal with the rumble components in the 5-8Hz region. Well, I suppose it would, if this is where they were. Unfortunately, experience shows that the really irritating LF noises are often in the 20-25Hz region, and a turn-over frequency of around 50Hz is really needed to get rid of them.

So, where one has this nuisance, it is, in reality, much better to be without it than to try to hold on to such signals as may occur in the half octave between 32 and 50Hz.

The filter block employed is a bootstrap filter circuit and the frequency response given by the circuit of Fig. 8 is shown in Fig. 9. Again, an LF353 dual op-amp is used to implement this stage, and, as before, a push-button cancel switch is wired to bypass it when better quality programme material is available.

The Headphone Amp

If the preamp is a separate unit from the power amp, it is a very useful thing to have a small headphone amp capable of driving a couple of pairs of phones, within the preamp box. However, if this amplifer is to be an accurate monitor of the signal delivered to the power amp and if, in the sort of architecture proposed for this unit, in many cases the signal from the auxiliary units will be routed directly to the power amplifier, the standard of accuracy and quality of the headphone amp must, if anything, be higher than that for the power amp itself.

Fortunately, the headphone amp has a much easier job to do, in that neither the output power requirements nor the load characteristics are so severe, since headphones typically have a load impedance of 100-2000 ohms, and only require 1-2V max RMS, for normal output. There are of course electrostatics, which may demand 5-10 watts, at loads down to a few ohms, but these are best driven from the power amp anyway, and the '8 ohm' headphones, will require a very low drive voltage anyway.

Since only a low power output is required, a class-A stage is perfectly feasible. Because only smallish output transistors are needed, 10MHz F_t devices are easily found, and, in any case, class-A operation makes the HF response good. The only other thoughts which commend themselves are that the design should be completely symmetrical, and direct coupled to the output, and that where NFB bypass capacitors of electrolytic type are used these should have a polarising voltage across them. It will also help sound quality if the amplifier has few stages, using discrete components, and no slew-rate limiting internal HF roll-off components are needed.

A design which meets these requirements, and gives an excellent sound quality, is shown in Fig. 10, with a suitable PCB layout in Fig. 11.

The basic amplifer system is as shown in the very simplified layout of Fig. 12. In this, a pair of push-pull input transistors, Q1 and Q2, drive a push-pull pair of output transistors, Q5 and Q6. Negative feedback is taken from the output point to the emitters of Q1 and Q2, and the load is connected between the joined collectors of Q5 and Q6 and the 0V line. For adequate class A operation the output transistors should pass, say, 100mA each. With a ±15V supply, this would mean 1.5 watts dissipation, so a smallish heatsink, perhaps 1.5" square, will be needed

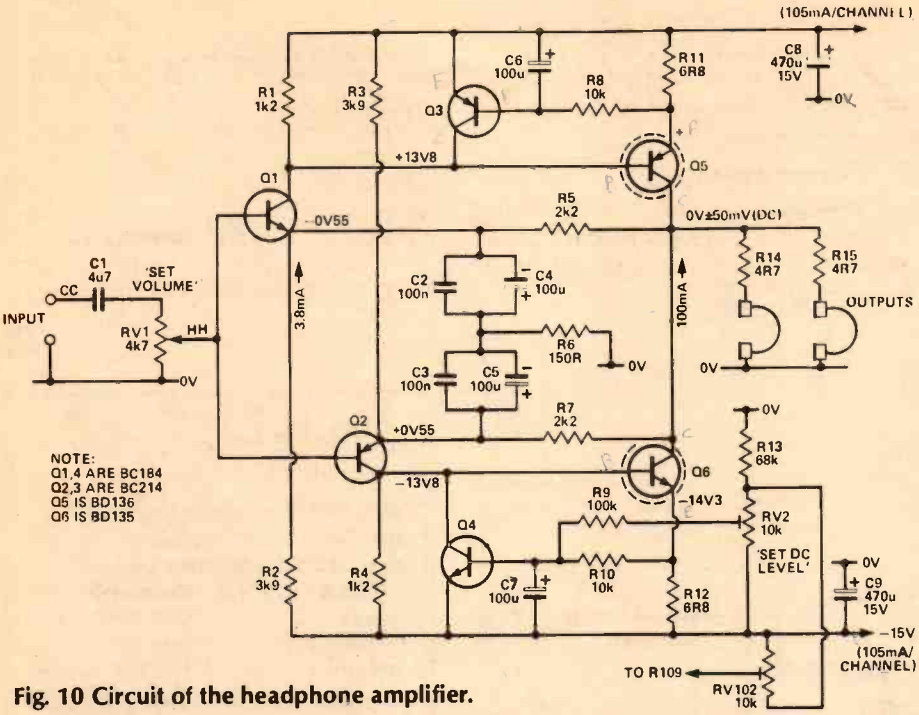


Fig. 10 Circuit of the headphone amplifier.

for each.

If the output transistors have a minimum current gain of 50, then each may require a maximum current input to their bases of 2mA. In order to provide this, with a bit to spare, the input transistors, Q1 and Q2 should normally pass about 4mA. If these have a current gain of 150, their base currents will be .004/150 = 26.7uA, which gives an input impedance of about 19k. The input gain control (and since this has to provide a 'balance' feature too, this should be the twin concentric spindle type) must therefore be a good bit less than this: a value of 4k7 will be fine. Unfortuantely, some of the aux input signal sources will have too high an output impedance to be able to drive this. It is therefore necessary that the input switching (Fig. 1) shall be organised so that the input buffer (incorporated to generate a low source impedance for the rumble filter and the tone control stages) is used also as the headphone amp input buffer, where otherwise a 'straight through' signal path would have been used. (A 3-head recorder system will normally have a 'line' output impedance (600 ohms) so this can drive the headphone amp without problems.)

Returning to the headphone amp circuit, we must now provide a source of emitter current for Q1 and Q2, and a means of controlling the current through Q5 and Q6. Looking now at the full circuit diagram of Fig. 10, the emitter current for Q1/Q2 is derived from the ± 15 volt lines through R2 and R3. For a 14.5V drop and 4mA flow, this would require a resistor value of 3625 ohms. The nearest preferred value is 3k9, which will pass a current of 3.7mA, though some 250uA will also flow through R6 and R8.

Looking now at Q5, (the circuit operation for Q6 is the same), a small resitor (6R8 ohms) in its emitter circuit senses the current flow. If this is too high, a forward bias is applied to the DC amplifier transistor Q3, through R8, (C6 removes all audio signals from this point), which will cause Q3 to conduct and steal drive current from Q5 base, holding the collector currents of Q5 (and Q6 for which the operation is identical) to the chosen average value.

Negative feedback is applied from the outputs of Q5 and Q6 to the emitters of Q1 and Q2. This gives a measure of DC output voltage control, but this can be fine-trimmed by R9, R13 and RV2, which operate to adjust the collector current of Q6 relative to Q5. A

PARTS LIST — HEADPHONE AMP

RESISTORS (all 2% 0.4W metal film)	
R1,4,101,104	1k2
R2,3,102,103	3k9
R5,7,105,107	2k2
R6,106	150R
R8,10,108,110	10k
R9,109	100k
R11,12,111,112	6R8
R13	68k
R14,15,114,115	4R7
RV1	2k2 twin concentric stereo log pot
RV2,102	10k lin preset, horizontal
CAPACITORS	
C1,101	4µ7 non-polarised
C2,3,102,103	100n polycarbonate
C4,5,6,7,104,105, 106,107	100µ 6V3 low ESR electrolytic, PCB mounting
C8,9	470µ 16V electrolytic
SEMICONDUCTORS	
Q1,4,101,104	BC184
Q2,3,102,103	BC214
Q5,105	BD136 or BD538*
Q6,106	BD135 or BD537*
MISCELLANEOUS	
PCB, wire, etc	

* BD538 and BD537 should only be used together.

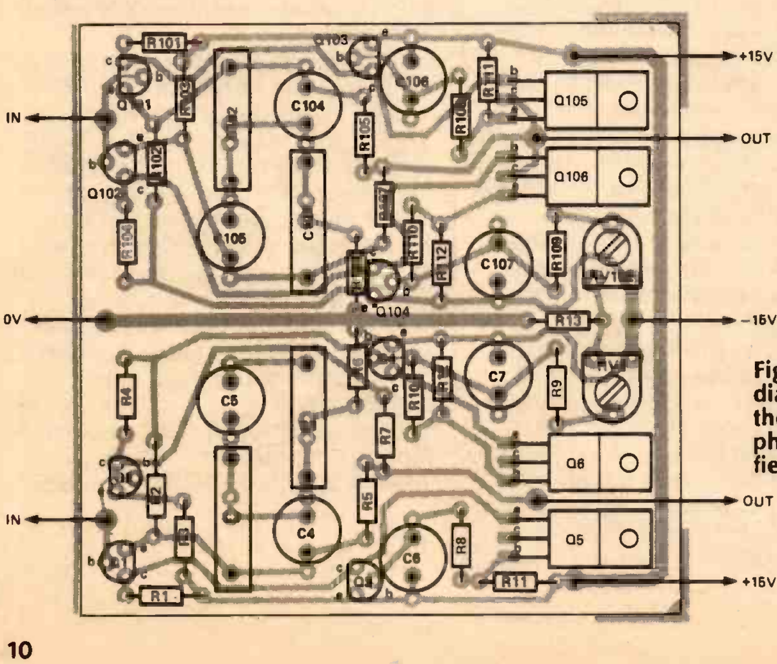


Fig. 11 Overlay diagram for the headphone amplifier.

DC output level of 0V± 50mV is adequate. Because the bases of Q1/Q2 are joined together, their emitters will sit at −0.55V and +0.55V respectively, which provides a standing 0.55V potential across C2/C3 and C4/C5. C4/5 should be low ESR aluminium electrolytics bypassed by C2/C3 polypropylene or polycarbonate 100nF types.

On typical headphone load impedances, the output THD is substantially that of the input signal, as is the transient response.

Power supplies

These are quite straightforward, and use a 15-0-15V toroidal transformer, in the interests of low hum field, a bridge rectifier, and two pairs of 15V IC voltage stabilisers. Low equivalent series resistance electrolytic capacitors are used to bypass the output DC lines to the 0V rail, and similar capacitors are used as bypass elements at the supply line connections to the preamp circuit modules. A ±5V take-off point, from another pair of stabilisers is employed, optionally, to power a MC head amp module.

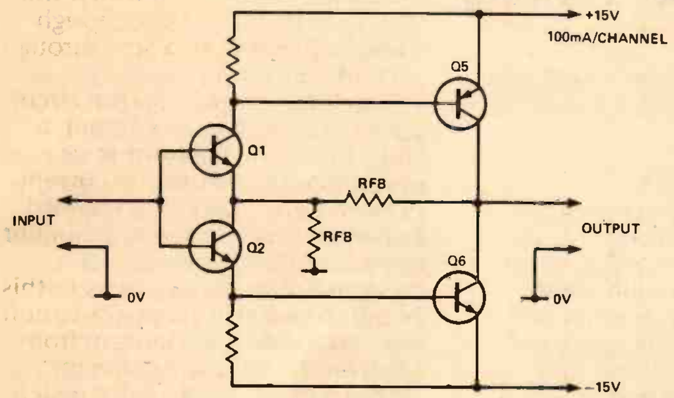
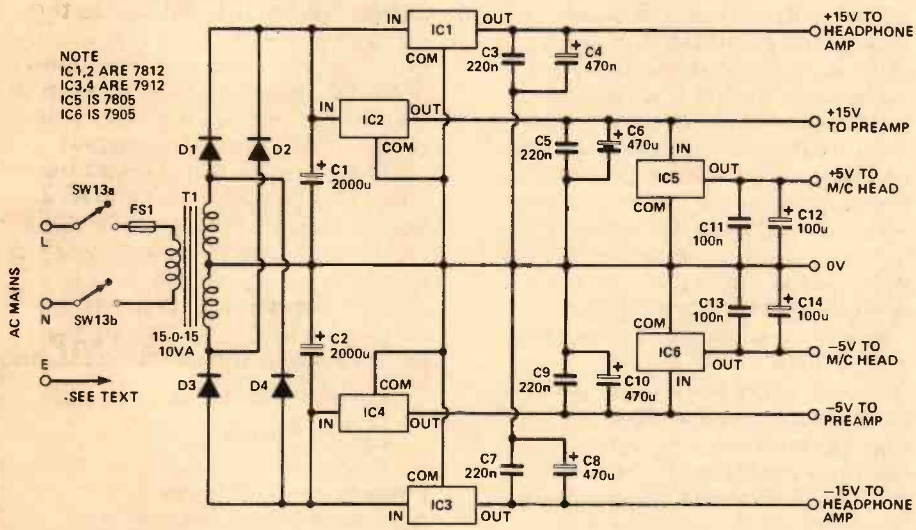


Fig. 12 (left) Simplified headphone amp.

Fig. 13 (below) PSU circuit.

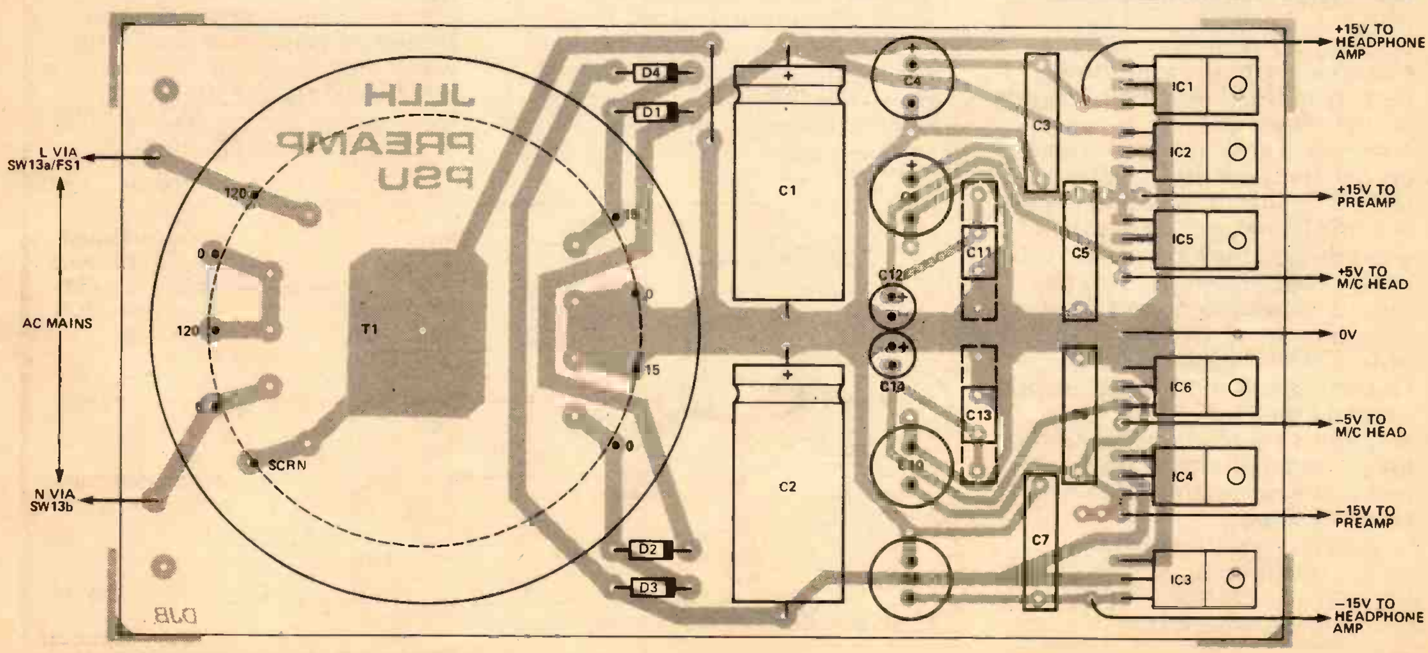
Fig. 14 (bottom) PSU overlay.



PARTS LIST — PSU

CAPACITORS	
C1,2	2000µ 25V tubular electrolytic
C3,5,7,9	220n
C4,6,8,10	470µ 16V electrolytic, PCB mounting
C11,13*	100n
C12,14*	100µ 10V electrolytic, PCB mounting
SEMICONDUCTORS	
IC1,2	7815
IC3,4	7915
IC5*	7805
IC6*	7905
MISCELLANEOUS	
T1	15-0-15V 10VA mains transformer, toroidal PCB mounting
FS1	100mA fuse + panel-mounting holder
SW13	double-pole mains switch to suit

PCB, wire, etc.
* = not needed if M/C head amp not used.



AUDIO DESIGN AMPLIFIER

Part 2: The Power Amplifier

In the previous article, describing the accompanying preamplifier, the basic design requirements of this power amplifier were outlined. These were: that it should offer an audio quality which was as good as the best commercial unit on the market, if only because there isn't any point in aiming lower than this; that is should have an input sensitivity and impedance which were both sufficiently high that signals from auxiliary sources could be routed directly to it, without manipulation by the preamplifier; and that it should be direct-coupled to the LS units.

Several other things followed on from this basic general specification: for example, if it is intended to be possible to route signals from auxiliary inputs directly to the power amplifier, to avoid any possible degradation in quality by preceding stages, then the power amplifier needs to have gain and balance controls on its input, rather than situated in the preamp. Another feature which is implied in this design spec is that the output stage should be based on the use of power MOSFETS, because they can offer a sound quality which is at least as good as that of bipolar transistors operated in class-A without the enormous penalty of the thermal dissipation of such designs.

I have a great liking for valves, myself, because they can be pretty to look at, they don't mind getting hot (in class-A use), and, with a good design, they are pretty well burst-proof. However, they need output transformers, and these are invariably so destructive of the potential performance of the circuit, especially in transient response, that I feel, sadly, that valve amplifiers are about in the same league as an oil tanker with sails and masts, a romantic idea overtaken by events.

Some other things which I hadn't dwelt upon, but which are necessary to consider if one is after the ultimate quality league, are stabilised power supplies, direct coupling, and the maximum practicable symmetry of the drive circuitry.

Stabilised PSU?

Looking at these in turn, the advantage of a stabilised PSU is that it will give a somewhat more solid bass response (mid range and treble response are more influenced by the circuit design of the amp and its feedback loop characteristics), and that the power output is identical under steady-state and transient conditions. In some ways this is an advantage, in that it will make power output specs less dependent on measuring conditions, and can help deliver more power into lower impedance loads. In some ways, though, it is less beneficial, because the simple power supply with output capacitor can, for a brief time, which is all that is needed on some transients, provide a higher peak power.
Many of these advantages can be gained, at lesser expense, by feeding the relatively low current, class-A, gain stage of the audio amp from its own PSU, separate from the power supply which feeds the output devices. However, there is yet another possibility in a stabilised PSU system which has finally swung my preference that way, and that is that it can be made to perform a LS protection system.

With any direct coupled amp, in which the output stage midpoint is taken directly to the LS units, there is a danger that an output device failure will damage the LS drivers, so a fuse, or a relay to disconnect the LS line, is a necessary precaution. Unfortunately, fuses and relay contacts tend to impair the electrical integrity of the circuit, which is made more apparent by the relatively high currents which are flowing in these paths. Gold-plated relay contacts do not impair the performance too much, provided that the thickness of the plate layer is adequate to survive the duty, but it would be better still to do without them.

Therefore, in this circuit I have chosen to provide the LS protection function by monitoring the DC offset at the LS terminals, and using any excess voltage detected at this point to electronically disconnect both of the output stage power supply lines, with a suitable warning that this has happened.

Drive Symmetry

A further design aspect in the power amp which I have not yet discussed is that of drive symmetry. Ideally, any power amp should be capable of operating with equal facility in either polarity direction. This becomes of importance where large voltage swings are likely, which is in the final

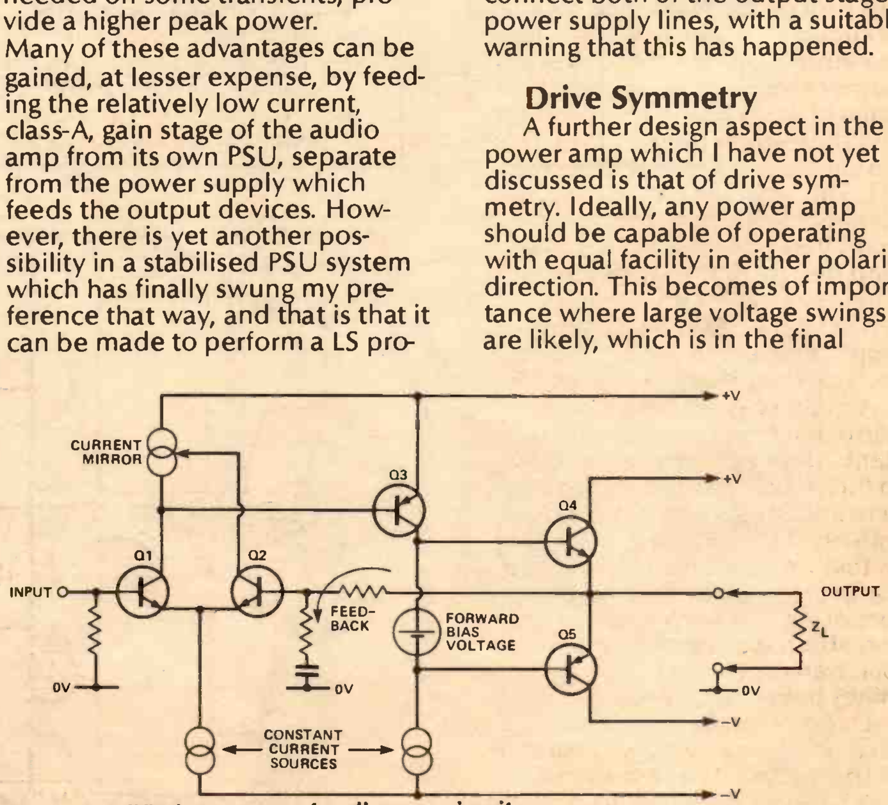


Fig. 1 Simplified structure of audio amp circuit.

class-A driver stage of the power amp, and in the output transistor pairs, Q3 and Q4 and 5, respectively, in my schematic circuit of Fig. 1, which is, itself a simplified circuit.

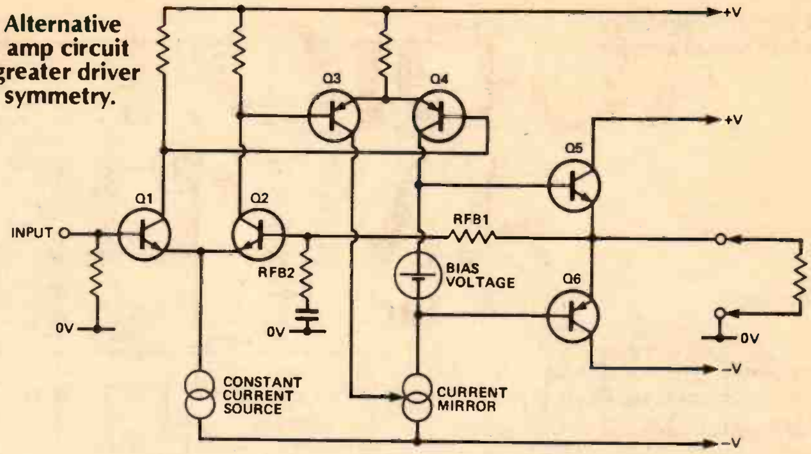
It isn't too difficult to make the output stages themselves quite symmetrical — within the limitations imposed by the transistors, which, in the case of the devices chosen, don't take effect until we get up to very high frequencies — but this is not true of the driver stage, Q3, and its constant-current source load. This is the point at which a conflict of requirements becomes apparent. If the biassing of the output stage is to remain constant, the load for Q3 must have constant current source characteristics, but it also must behave as an effective dynamic load for the amplifier stage Q3.

If the load on Q3 were purely resistive, there would be no great difficulty in satisfying this requirement, but there is, inevitably, some capacitance at this point, due to the output stage loading, and it then becomes essential that the current flow through the constant current source shall be able to charge this capacitance, as the voltage at Q3 collector falls, at a rate which is greater than the fastest negative-going rate of change called for by the incoming audio signal.

An apparently neat answer to this problem is given by the kind of circuit shown in Fig. 2, in which the input long-tailed pair drives a further symmetrical push-pull stage of amplification, Q3 and Q4, and the current mirror driven by Q3 provides a dynamic load for Q4. This was first introduced by National Semiconductors in the mid-1970's, in their LH0001 op-amp design, and adopted by Hitachi as the recommended driver stage for MOSFET power amplifiers using their devices.

However, there are snags. The first of these is that the current mirror load isn't any kind of constant current source, which leads to further consequential problems in maintaining output stage bias stability. The second is, surprisingly, that on close examination and comparison of the two systems, that of Fig. 1 is both more linear and also has a superior reactive load transient response — other things being equal — to that of Fig. 2. This is possibly the reason why such an obviously elegant solution to this problem has not found much favour in the minds of the IC designers, whose products overwhelmingly favour the Fig. 1 scheme, which is the layout I have ultimately returned to, with the implicit requirement that Q3 current must be adequate.

Fig. 2 Alternative audio amp circuit with greater driver stage symmetry.



MOSFETisation

There are, however, some further improvements which can be made to this circuit, and of these, the major one is the replacement of the small signal transistors by low power versions of the power MOSFETs, which are now available. These are both faster and more linear than the equivalent bipolar junction transistors, and, in principle, all of the bipolar transistors could be so replaced, with suitable adjustments to the circuitry, as shown in Fig. 3.

The current mirrors and constant current sources perform functions that do not benefit from 'MOSFETisation', and the higher mutual conductance of the input bipolar devices is definitely useful in maintaining a high circuit gain. However, N-channel MOSFETs are faster than P-channel equivalents, because electrons travel faster than holes, so to make it possible to use an N-channel device for Q3, the input stage must be recast to use PNP transistors for Q1 and Q2, rather than NPN types. Another possible improvement would be to use small-power MOSFETs to make Q4 and Q5 into compound output pairs.

In this form, the circuit gives an excellent performance. However, I am all in favour of simplicity, and with the small-power MOSFET final class-A stage, a sufficiently high stage gain is available for the output MOSFETs to be used as simple source-followers. Moreover, careful tailoring of the output and driver circuitry allows the removal of the output inductors normally essential in this style of circuitry. The final circuit layout is shown in Fig. 4.

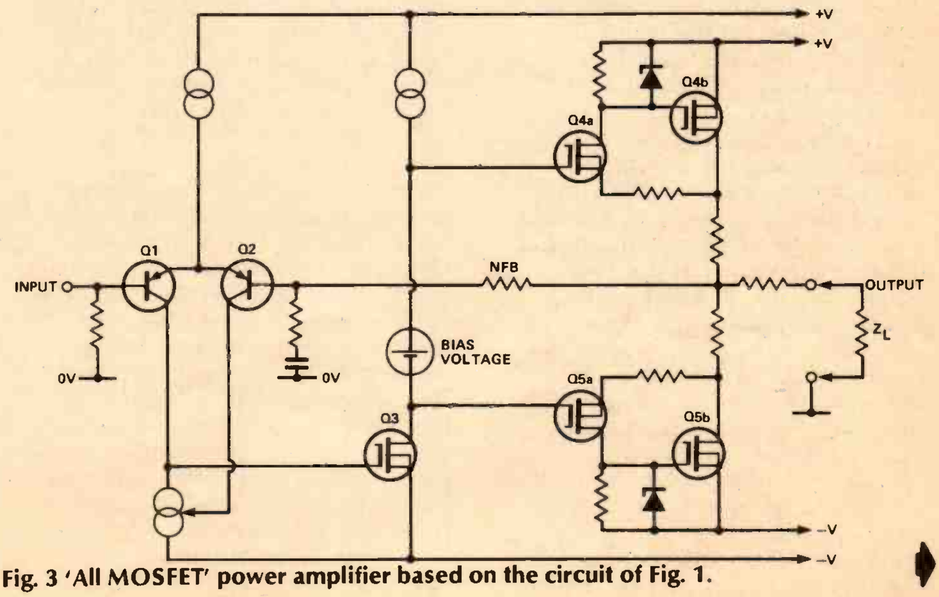
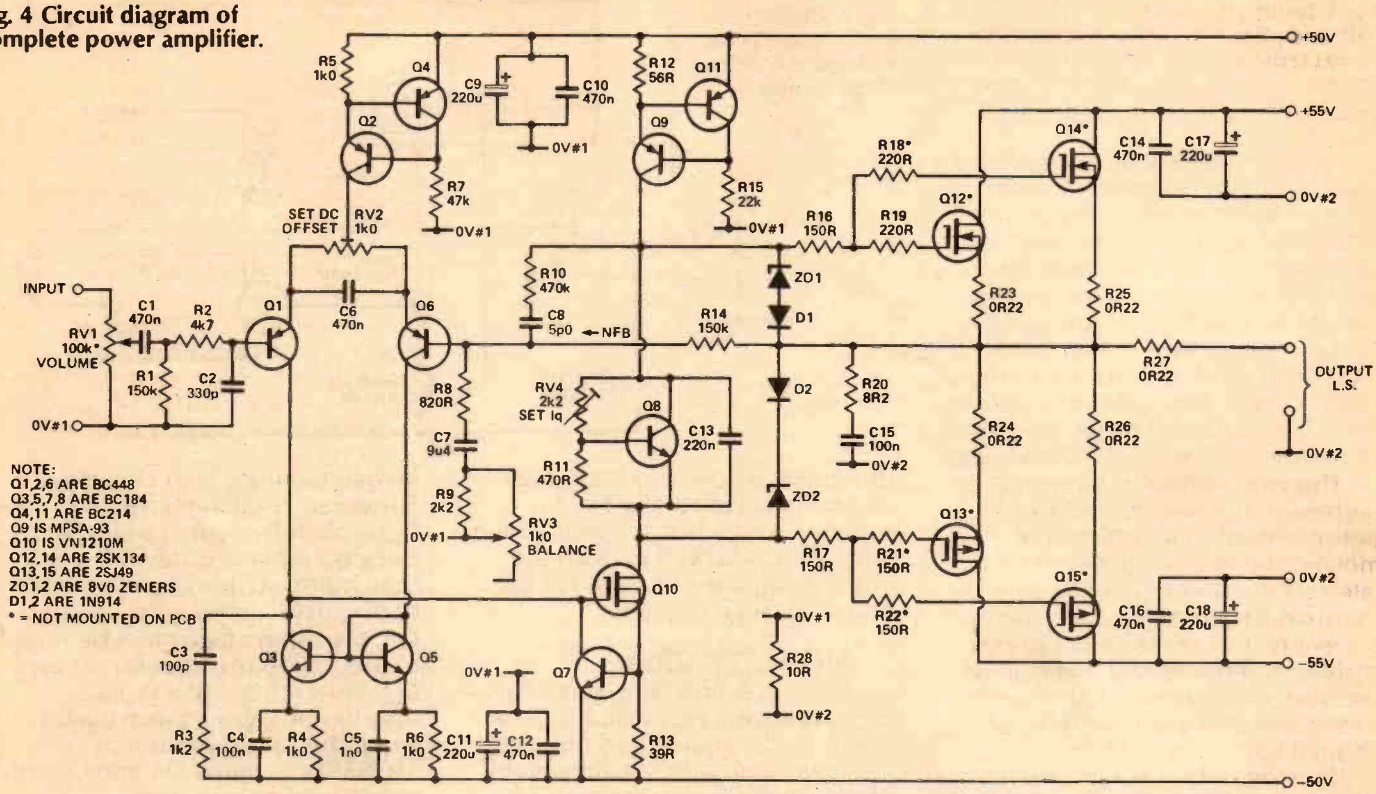


Fig. 3 'All MOSFET' power amplifier based on the circuit of Fig. 1.

Fig. 4 Circuit diagram of complete power amplifier.



Conflicting Requirements

In every audio power amplifier circuit design there is a conflict between the requirements of low harmonic distortion, smooth transient response, and reactive load stability. This arises because low harmonic distortion demands both that the basic structure of the circuit, and its component elements, shall be such that it has high intrinisic linearity, and that the negative feedback loop will provide an effective measure of linearity enhancement. However, a smooth transient response, and good reactive load stability both require that there is a good phase margin in the feedback loop at the point at which the amplifier gain has reached unity. This comparison is shown in Fig. 5.

The loop gain characteristics shown in curve (a), in which the gain is maintained at a high level to as high a frequency as possible, and then rolled off rapidly so that it is less than unity at the 180° phase shift point (if it is unity at this frequency the amplifier will oscillate uncontrollably), will give better THD (because the amount of feedback applied at higher frequencies is greater) than the type of characteristic shown in curve (b). On the other hand, the kind of amplifier response shown in 5(b) will have much better reactive load stability on 'awkward' loudspeaker loads, and will generally be more predictable, and 'smooth' sounding, in spite of rather worse THD.

Obviously this is one of the occasions where one wants to have the cake and eat it, and if one is a commercial manufacturer, one is more or less forced to adopt the 'low THD' choice, because this will be measured and quoted in the test reports, with the — to my mind — very important reactive load transient response taking pot luck; after all, this isn't a quotable parameter.

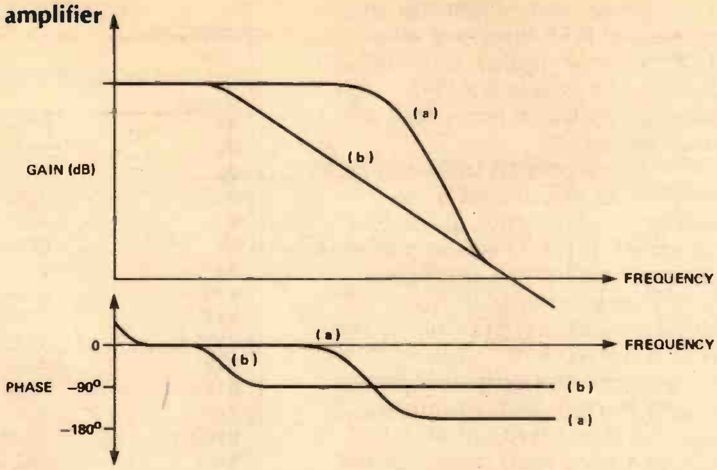
Since I am in the happy(?) position that I design amplifiers for my own use and pleasure, and not for sale, I am more concerned with how they will sound than how they will measure. Nevertheless, I am an engineer, and I have a normal engineers pride in doing things competently — which means, in practice, that I cannot call the job done until I have at least equalled, if not improved upon, the best performance I have so far come upon, in my own or in commercial designs. (Yes, I do look at, and test, whatever commercial units come my way, and I study their circuits to see if I can learn anything from these, in the way of clever engineering or crafty pieces of circuitry. Sadly, my feeling is often that elaborate and expensive paths have been adopted to achieve a result which could have been done as well or better with more simple and economical means.) For the record, the performance of this circuit, in respect of the THD levels obtained, *without sacrifice to transient response*, is the best I have achieved so far. I do not, at this moment, want to try to better it!

The THD figures are quoted in Table 1, and the way in which the THD varies with power and frequency, at max. output, is shown in Fig. 6a and 6b. I show the THD vs. power output at 10kHz, because, on the prototypes, the THD at 1kHz is, at all power levels below clipping, below the residual circuit and measurement apparatus background noise level. Such distortion products which can be extracted from this noise floor can be shown to originate in the signal source, and are around the 0.002% level (−94dB).

Circuit Analysis

As mentioned earlier, the design decision in the concept of this amplifier was that its input impedance and sensitivity should be such that it could be driven directly from the sort of input signal, in magnitude and impedance, which could be expected from typical auxiliary units — tuners, cassette recorders and the like. In practical terms this implies an input sensitivity of about 150mV and an input impedance greater than 100k.

Fig. 5 Feedback amplifier gain and phase characteristics.



HARMONIC	DISTORTION (%)
2nd	0.021
3rd	0.003
4th	0.0015
5th	0.0007

Table 1 Harmonic analysis at 10 kHz (80W/8 ohms).

This determines the input impedance requirements of the input transistor stage, which can be met, adequately, by an input long-taled pair of reasonably high gain transistors operating at a collector current of 250uA. At this collector current, the typical current gain of the devices chosen is 250, giving a base current of 1uA, and a Z_{in} of about 330k.

To ensure that the input stage has a good DC balance, so that the output offset voltage of the amplifier is close to zero, the base circuit DC resistances for the input long-tailed pair (Q1 and Q6) are made similar, at 150k, and a 1k0 DC-offset adjust pot, RV2, (1k0 cermet) is connected in between the two emitters. This is adjusted so that the output voltage of the power amp is within about 50mV of 0V.

The input signal to the power amp is derived from the 100k gain control, RV1, via C1 and R2 — which acts with the 330pF input capacitor to lessen the sensitivity of the circuit to impulse noise or radio breakthrough. The current feed for the input stage is derived from the +50V line by the constant-current source, Q2 and Q4, through which the current flow is set to 500uA by the resistor R5 (1k0), and the collector load for the input stage is provided by the current load for the input stage is provided by the current mirror configuration of Q3 and Q5. By using high current gain transistors in this position (their operating collector voltages are very low) the current flow through Q3 is forced (by the action of the overall DC negative feedback loop in the amplifier) into a very close equivalence to that through Q5.

The action of the bypass capacitor across the emitter resistor of Q3 is to increase the output impedance and effective dynamic gain of this current mirror — an option which is available to us because we are driving a very high impedance following stage: the small-power MOSFET, Q10, whose gate circuit is effectively an open-circuit, apart from some 75pF of gate-source capacitance. The phase-correcting network, C3 and R3, together with the small emitter resistor bypass capacitor C5, adjust the HF phase-angle of the feedback in the 1MHz region, which is where the amplifier would otherwise approach a critical stability threshold. It will be appreciated that, with circuits operating in these frequencies, the layout of the components and interconnecting wiring has a great influence on the gain/phase characteristics of the system, which are optimised only for the PCB layouts employed. So, if you use a different layout, C3, C5 and R3 may need to be different!

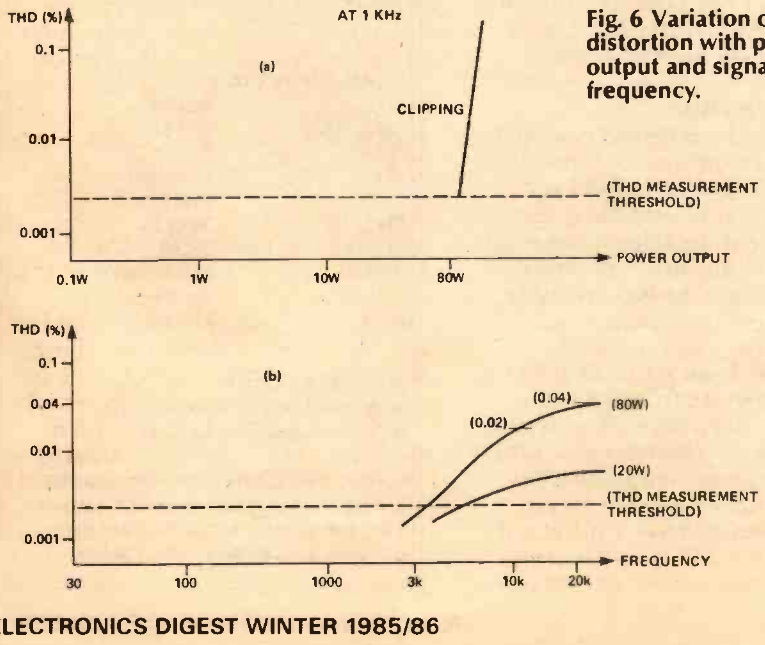


Fig. 6 Variation of harmonic distortion with power output and signal frequency.

Driver Stage

The second, class-A amplifier stage, using the MOSFET Q10, is quite straightforward in operation. The operating current is held at 10mA by the constant current source Q9 and Q11. If the current exceeds this value, the voltage drop across the 56R resistor R12 exceeds the 0.56V turn-on voltage for Q11, and it steals more of the base current fed to Q9 through R15. If the output current from Q9 falls, the converse occurs, and Q9 is turned on more fully. This constant current source protects the operation of this stage from an inadvertent output short-circuit, during a positive-going voltage excursion. A similar protective function is performed, in respect of an output short-circuit during a negative-going voltage excursion, by Q7 and R13. If the current through Q10 and R13 exceeds 14mA, Q7 will turn on and clamp the gate voltage of Q10. The actual class-A standing current through Q9 and Q10 is set at 10mA, as the largest practicable current flow compatible with the 625mW dissipation of Q9 (Q10 can dissipate 1W). Note that the collector/drain tracks on the PCB are broadened to assist in heat removal from these devices.

The choice of the class-A stage DC operating voltage (±50V) is determined only by the need to provide an adequate voltage swing to the output stage MOSFET gates.

For an output power of 80 watts into an 8 ohm load, an RMS voltage swing of 25.3V RMS is needed. This is equivalent to peak-to-peak voltage swing of 71.55V. However, it must be remembered that, at the peak output currents demanded (4.47 amps), the MOSFETs will require a 6V source-to-gate voltage. Also the circuit of Q9 and Q10 will only swing to within 2V of the positive or negative supply rails. Finally, at 4.47A, the voltage drops through R23, R24 and R27 will amount to 1.78V on each half cycle. Adding these together, we get 71.55 + 2 + 2 + 1.78 + 2x6V = 89.33, so ± 50V will be quite adequate.

The necessary forward bias for the output MOSFETs is generated by the 'amplified diode' circuit of Q8, which is bypassed by a small, non-polar, capacitor in the interests of HF symmetry, as is the zero DC offset adjust pot RV2.

Although the circuit will operate satisfactorily with a single pair of output MOSFETs, more power from the same HT supply voltage, an improved THD performance, and better low signal level, pure class-A, performance can be obtained, at a relatively modest extra cost, by doubling-up the output MOSFETs. These can be paralleled quite easily, provide that they have separate source and gate resistors. Since it is preferable for the gate resistors to be mounted close to the MOSFET gate pin connections, these are not included on the PCB.

Earthing

In order to avoid unwanted earth-loop effects, between the low-current input signal earth lines, and the high-current output earth lines, the '0V' lines at the inputs and outputs of the amplifier boards are separated, but joined on the PCB by a low-power 10 ohm resistor, R28. Each supply rail is decoupled, on the board, to its appropriate '0V' line by a 220uF/470nF electrolytic/non-polar combination.

Output transistor input over-voltage protection is given by the ZD1/D1 and ZD2/D2 networks connected between the outputs of the driver stage and the output of the amplifier, which limits the maximum forward gate drive voltage to 8.5V. The output 'buffer' resistor R27 serves two functions. These are to assist in rejecting externally generated signal voltages on the LS line, due for example to dynamic delayed echo effects within the LS units, from the amplifier internal NFB line, and also in allowing the amplifier, unusually in the case of a power MOSFET unit, to operate without an output LS line inductor.

The reactive load transient performance of this circuit is extremely good, in spite of the low level of HF THD. This is in part due to the 'tuning' of the amplifier phase characteristics in the 100KHz — 300KHz region by the R10/C8 network. By altering R10 one can tune the output to give a virtually impeccable square wave response (i.e., identical with or without added load capacitance) over the range 8R//100n to 8R//2.2uF — for R10 values from 220k to 600k. The mid-range value I have chosen is about optimum for 1uF//8R, though the actual differences in performance on either side of this value are very small.

Channel Balance Adjustment

I have chosen in this design to adjust the relative gain of the two channels by alteration of part of the low-signal level NFB resistor arm using R9 and RV3. With a two gang 1k0 pot, one half of which is connected in each channel in a reciprocal fashion, a ±6dB gain adjustment of each channel with reference to the other, is provided. A two gang pot. is essential to prevent inter-channel breathrough.

However, I am aware that this is a point of some controversy among users, some of whom very much prefer that each channel should be capable of reduction to zero output. For those who prefer this style of operation, I would recommend that a twin-spindle, concentric, input volume control is employed, RV3 be deleted, and R9 replaced by a 390R resistor.

Construction

A suitable PCB layout is shown in Fig. 7. As mentioned above, the layout employed will affect the performance at HF, and the consequent phase shifts within the feedback loop. Therefore, I strongly urge that the suggested layout is retained.

General Considerations

It has been demonstrated to me, in relation to an earlier design of mine, that the component types employed can have a significant effect on audible quality. In particular, the capacitor employed in the NFB loop (C7) is a very sensitive component, where a consider-

PARTS LIST

RESISTORS (metal film, 0.3W, unless stated)

R1	150k
R2	4k7
R3	1k2
R4,5,6	1k0
R7	47k
R8	820R
R9	2k2
R10	470k
R11	470R
R12	56R
R13	39R
R14	150k
R15	22k
R16-21	220R
R22	8R2 2.5W WW
R23-26	OR22 2.5W WW
R27	OR22 2.5W WW
RV1*	100k log stereo pot.
RV2	1k0 lin cermet preset, open horizontal
RV3*	1k0 lin stereo pot.
RV4	2k2 lin cermet preset, open horizontal

CAPACITORS (radial lead, stacked film polyester unless stated)

C1,6	470n
C2	330p polystyrene foil
C3	100p polystyrene foil
C4	100n
C5	1n0 polystyrene foil
C7a,b	9μ4 (2x4μ7 parallel) or 10μ (single) polycarbonate
C8a,b	5p0 (2x10p series) polystyrene foil
C9,11	220μ electrolytic
C10,12	470n
C13	220n
C14,16	470n
C15	100n
C17,18	220μ

SEMICONDUCTORS

Q1,2,6	BC448
Q3,5,7,8	BC184
Q4,11	BC214
Q9	MPSA-93
Q10	VN1210M
Q12, 14	2SK134
Q13,15	2SJ49
ZD1,2	8V0 Zener diodes
D1,2	1N914

MISCELLANEOUS
PCB, heatsinks to suit (see next article), connecting wire, etc.

*** Note: items marked with an asterisk are common to both channels, so only one is required; two of all other components will be required for stereo.**

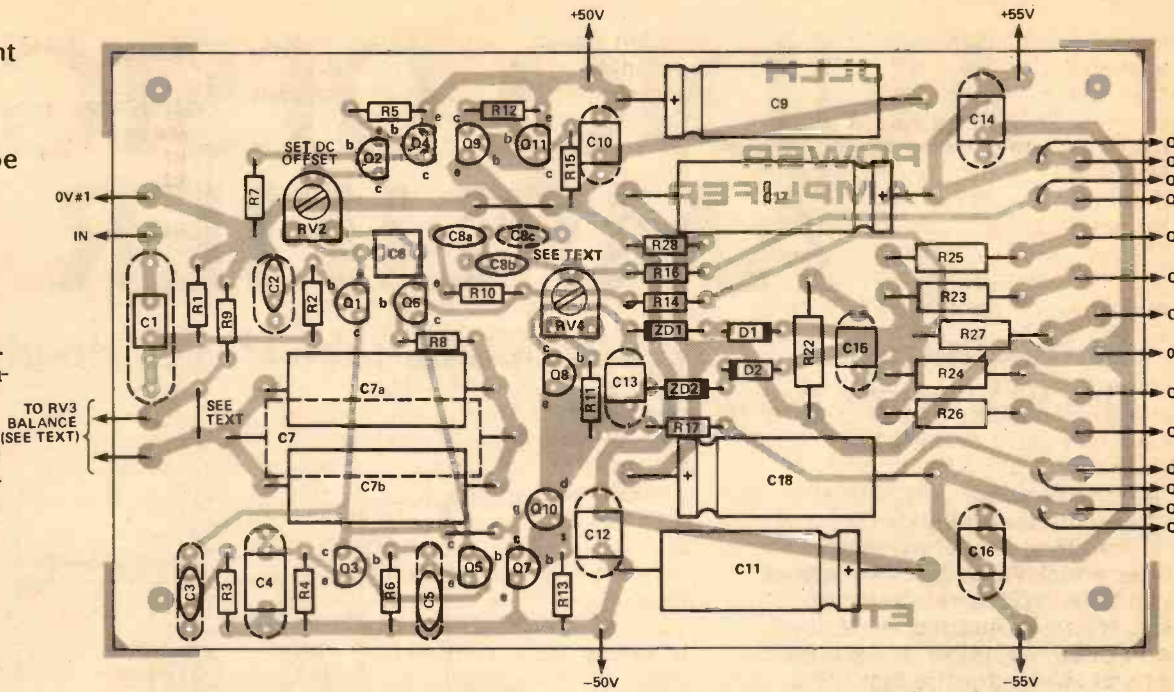


Fig. 7 The recommended PCB layout. Please note that the a 'c' and 'e' labels on Q9 are reversed. The underlying PCB pattern is missing the link between the cathode of 2D1 and R16, RV4 wiper etc. This has been corrected on the foil pattern reproduced elsewhere in this issue, and on the board supplied by our PCB Service.

able improvement in sound quality — not readily measured instrumentally — can be gained by the use of non-polar rather than, for example, a polar (tantalum bead or aluminium electrolytic) type. Polypropylene capacitors are probably the best choice, but these are bulky and difficult to obtain in large values, so I have designed this unit around the second best choice in this position, polycarbonate, and C7 is built up from two 4u7 polycarbonate capacitors connected in parallel. (10u polycarbonate capacitors are fairly rare, but if you can obtain them, one of these can be used instead.)

With the values chosen for R8 and R9, this gives a low frequency −3dB gain point of 14Hz, which is adequately low. The resistor types should be metal film 0.3 watt, or wirewound, as appropriate, and C8 is two 10pF polystyrene foil capacitors connected in series.

The other larger value capacitors, apart from the supply line decoupling electrolytics, are radial lead, stacked film, polyester types.

AUDIO DESIGN AMPLIFIER

In this third part of the description, John Linsley Hood describes the PSU and a power meter.

In the previous part of this article, referring to the power amplifier, I outlined the advantages which arose from the use of a stabilised power supply unit, which had persuaded me that this kind of arrangement was essential if I was aiming for the highest standard.

I was, indeed, responsible for a bit of propaganda in this cause in an earlier article (ETI May, 1983) describing such a stabilised PSU unit. Inevitably, therefore, my thoughts returned to this as a useful working design, though, in this case, I wanted to add somewhat to the facilities offered by the earlier design.

These additions are a pair of stabilised, lower current, power supplies to drive the earlier, class-A (voltage gain) stages of the power amplifier, and a DC offset monitoring facility which could be used to detect any abnormal DC voltage present on the LS output terminals — as might arise, for example, in the event of a catastrophic failure of one of the output devices — and switch off the high current sections (+ve and −ve) of the PSU, before any damage could occur to LS units or the like.

Since the power supply described previously has a re-entrant output characteristic (which means that the DC output current will decrease as the output voltage falls to very nearly zero output current into a short circuit), it will also perform the function of overload protection for the PA in the event of an abnormally low impedance output load. I happen to know that this works, since during bench testing, to see just how much power I could get out of a single channel driven just short of clipping (117 watts, as it turned out) and how well the PSU would hold the line voltage under these conditions (−1 volt) the soldered connections holding my load resistor melted off, the resistor dropped onto the floor, and the two liberated lengths of wire connected to the output terminals promptly soldered themselves together! After I had restored the load, everything was still perfectly functional, and apparently unruffled by the event.

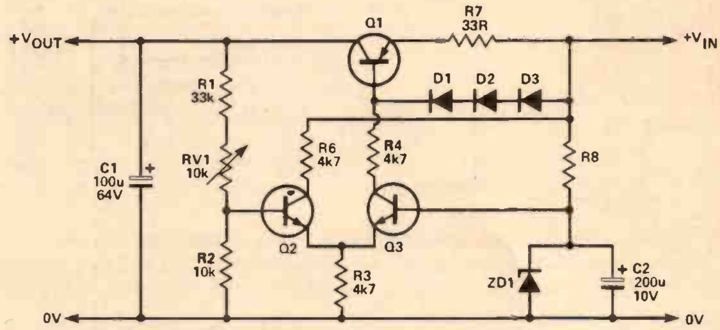


Fig. 1 Low-current stabilised PSU.

Experimental work, and inward deliberation, has convinced me that it is very advantageous to separate out the power supply lines feeding the output and the class-A stages of a power amplifier — indeed I think it is a false economy not to do this — and if one is using a stabilised PSU, it makes sense to put in a few more components to generate a pair of independently stablilised lines for the early stages.

Since the current requirement at this stage is quite small, typically about 12mA per channel, no problems of 'secondary breakdown' will arise in the series control transistors, so a simple constant-current overload characteristic will suffice, at 35-40mA total output. This will prevent anything inconvenient happening in the event of an accidental output short-circuit across these DC supply lines, as can so easily happen during setting up or testing.

I have shown the circuit I have adopted in Fig. 1. Once again the input and output voltage requirements prevent the use of an IC voltage stabiliser, though I guess that 60-80V input voltage IC stabilisers will be on the market (at a price) within the next few years. As in the higher current supply previously described, the pass transistor, Q1, is turned round so that the output current is drawn from its collector. This allows the forward base bias current to be derived from the 0V line, rather than from the forward voltage drop across this transistor. This makes for more efficient working and allows a much lower minimum voltage differential between input and output.

This last factor is important, because although the output voltage is very smooth, the input voltage across the power supply reservoir capacitors will show a fairly large 100Hz 'sawtooth' waveform, of 5 to 10V P-P amplitude, when a significant amount of current is drawn from it. The stabiliser circuit must work as well at the minimum input voltage represented by the bottoms of these input voltage waveforms (see Fig. 2) as at their peak.

Circuit Operation

This method of operation of the circuit is quite straightforward: a 10 volt reference voltage is generated across ZD1 and C2 by

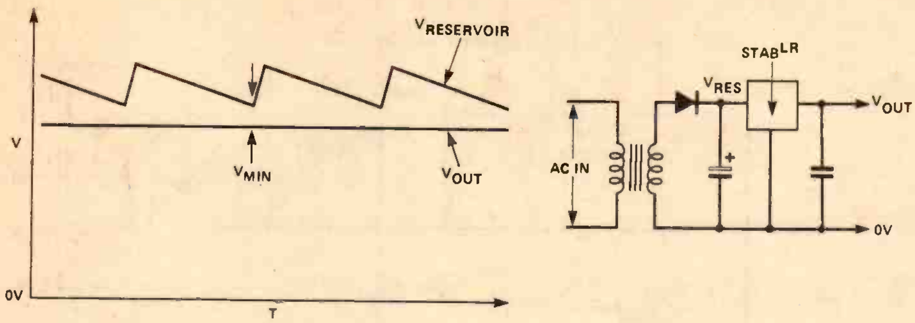


Fig. 2 The effect of ripple on stabiliser input — output voltage.

current flowing through R8. This is applied to one of the long-tailed pair of transistors Q2/Q3, and turns Q3 on. This passes current through R3, Q3 and R4 into the base of Q1, which causes Q1 to conduct and feed current to the output. A proportion of the output voltage, developed across R1, RV1 and R2 is applied to Q2, and if this exceeds the 10 volt reference fed to Q3, the current flowing through the 'tail' resistor, R3, will be progressively diverted away from Q3 and Q1, and will, instead, pass through Q2 and R6.

By this means, the voltage permitted at the output of Q1 is controlled so that the current flowing through R2 (which is, in turn, controlled by the values of R1 and RV1) produces a 10 volt drop across it (remember, V=IxR).

Overload (over-current) protection is obtained by putting a resistor R7 in the emitter circuit of Q1, and three small diodes (D1, D2 and D3) between the DC input and its base. Q1 will require about 0.6V forward bias to conduct, while the diodes will conduct at about 0.55V each. This limits the voltage which can develop across R7 to 1.65-0.6V=1.05V. If the voltage tends to exceed this value, Q1 will run out of forward bias, and will progressively turn off. With a value of 33R for R7, the circuit will limit at about 35mA, under output short-circuit conditions, which makes it effectively disaster proof.

To calculate the circuit component values, we first select a pass-transistor, Q1, as a device which will withstand 70 volts input, and carry the necessary current: a BD538 will serve. This has a minimum H_{fe} of 40 at 100mA, so it will need, say, a 1mA base current. Therefore let us make Q2 and Q3 both pass 1mA normally. This requires a 'tail' resistor of 4k7 (R3). The output voltage divider chain is chosen to pass about 1mA and give +10V at Q2 base when the output voltage is +50V. R4 and R6 are just protection resistors to prevent damage if a faulty transistor should be inadvertently installed in construction. RV1 is adjusted to set the output voltage to +50V. A mirror-image of this circuit is used to provide the −50V supply.

LS DC offset protection

I have made use of the two transistor 'thyristor' circuit shown in Fig. 3 to provide the offset protection function. (Note that component numbers here refer to Fig. 3) In this arrangement, Q1 and Q2 are both normally non-conducting. However, if an input voltage is applied to Q2, even briefly, it will conduct and feed current into the base of Q1. This will make Q1 conduct, which, in turn, will feed current into Q2, which holds the circuit on, or 'latched'.

In order to make the circuit respond only to long-term averaged DC offsets, a 1M0 and 2μ2F input integrating circuit is connected to the LS outputs, with an emitter-follower transistor Q3 interposed as an impedance conversion system. A similar circuit, with Q4, R4 and C2, can then monitor any offset occurring on the other channel. To avoid quadrupling C1 and C2, the offset voltages averaged across these are taken to a mirror image circuit controlling the other half of the PSU. The circuit I have shown is for the positive half of this.

When Q1 and Q2 are latched, the voltage drop across them falls to about 0.65V, and they will stay in the latched condition until the power supply to them is removed by switching of the equipment. it is possible to provide a momentary reset by S1, R5 and C3. If the fault persists the circuit will cut-out again almost at once. I prefer to switch off in the event of failure, so I haven't provided this facility on my prototype. The output of this 'thyristor' is taken to a point on the main PSU where a 0.65V clamp on the circuit voltage will cause the system to cut off.

A simple resistor and zener diode network, shown in Fig. 4., monitors the relative voltages on the +ve and −ve supply lines. If these differ by more than 20V, as will happen if one of the supplies is cut off, it will then turn the other line off as well. Since the tripping of one of the DC offset monitor circuits will automatically trip the other, the power supply failure warning can be given by a LED, in seies with a zener, and a suitable limit resistor, between the reservoir and the output on either DC line, so that the LED will light if the difference between input and output voltage exceeds 30V. This will happen briefly on switch-on, because the power supply has a slower rate of voltage rise (deliberately) than the voltage rise across the reservoirs. However, the LED will extinguish, in the absence of any fault condition, in a few seconds, when the supply lines have reached their proper operating voltage.

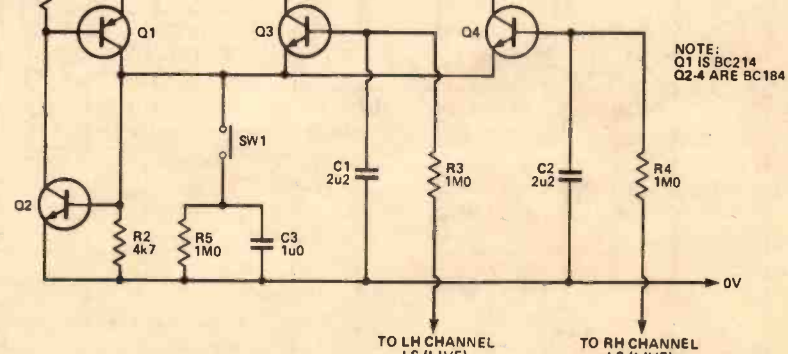


Fig. 3 Amplifier output DC detection and trip circuit.

The Full Circuit

The complete circuit diagram, apart from the transformer and rectifiers, of the power supply is shown in Fig. 5. The low current supplies, built around Q1, Q2 and Q3, with their mirror-images (Q4, Q5 and Q6), are as have been described above. The protection circuitry (Q9, Q10, Q11, Q12, Q13, Q14, Q15 and Q16) in its two mirror-image forms, is also as described above. The rest of the circuitry, comprising the twin high-power stabilised units, is largely as described in May 1983, but I will run through its operation to explain the method of the cut-out trip function, and to avoid difficulties for those who missed the May '83 issue (Shame! — Ed.).

Taking the positive-line supply section, a power Darlington transistor, a Motorola MJ2501, is used as the series control or 'pass' device. This is a moderately beefy component, with a maximum current of 10 amperes, an 80V$_{ceo}$ rating, and a maximum dissipation of 150 watts. The 'safe operating area curve' is shown in Fig. 6, and the actual output currents, with voltages, given by the PSU are as shown, for two different values of R15/R16.

To check on my calculations with these I have run the PSU into a low resistance (0.1 ohm) ammeter, which gives an effective output short-circuit, with the transformer fed from a 'variac. I have also, inadvertently, made screwdriver-type shorts from supply lines to chassis, without any disasters. This is *not* a practice I recommend, but it does happen, especially if one is developing or debugging a new circuit and one forgets to switch off.

The pass transistor Q17/Q20, is normally turned on by current flow from the 0V line through a control transistor, Q18/Q19, and a current limit resistor, R29/30. The control transistor is itself made conducting

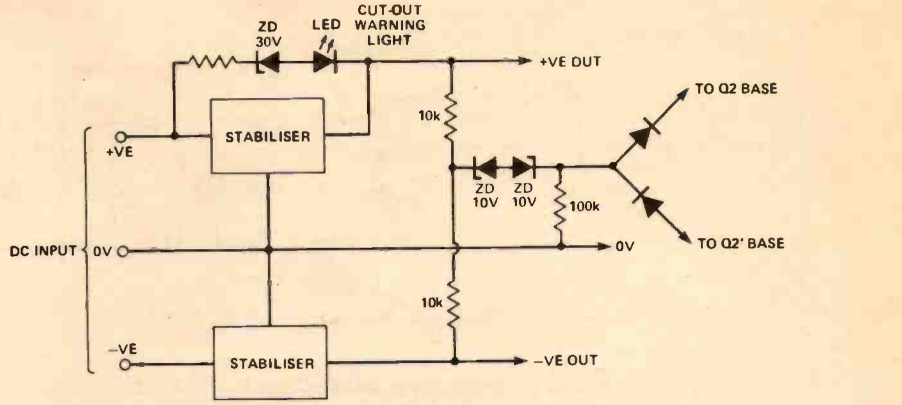


Fig. 4 Method of making both power supplies cut out simultaneously.

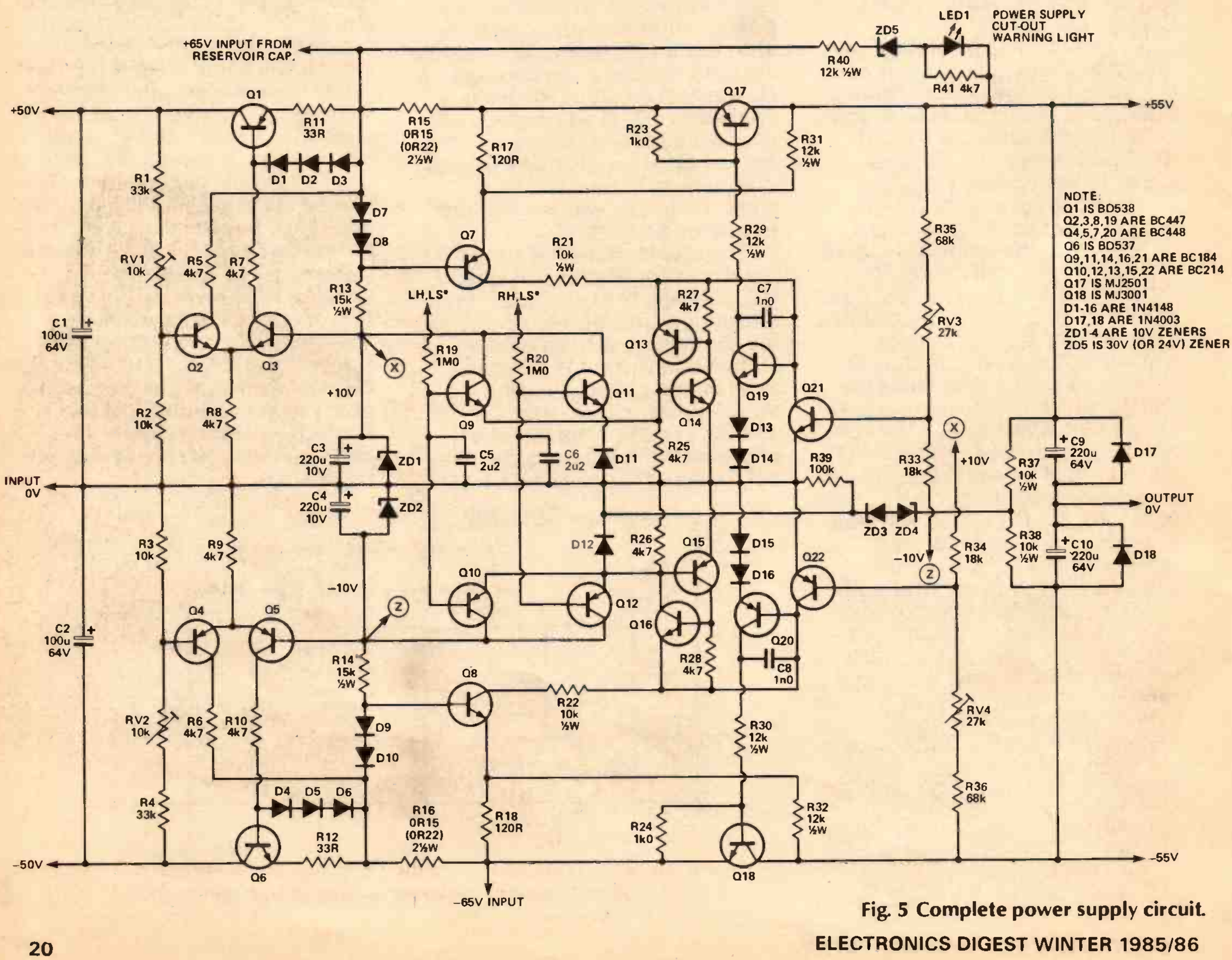


Fig. 5 Complete power supply circuit.

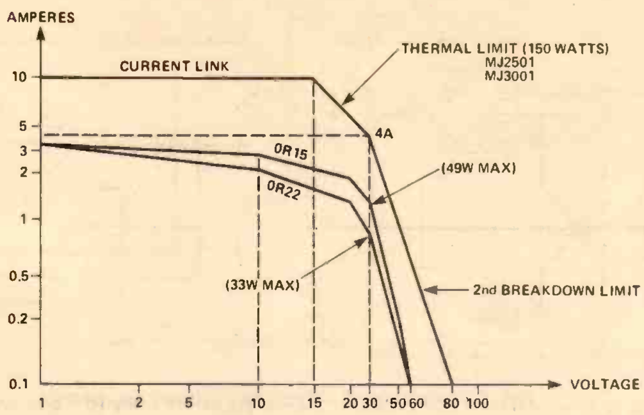


Fig. 6 Safe operating area curve for MJ2501/3001 and output current/voltage limits for PSU.

by a current flow from the input line through the protection transistor, Q7/Q8. A further transistor, Q21/Q22, sits between the 0V line and the base of the control transistor. This monitors the potential developed at its base from the voltage dropper chain, R35, RV4, R33/R34, RV3 and R36, connected between the output voltage line and the internal zener reference potential. If the output voltage should increase, this transistor is turned on more, and 'steals' more current from the control transistors base supply. This in turn reduces the current flow through the pass transistor, to oppose the detected increase.

Because there is a very high loop gain in this three transistor amplifier loop (Q17,Q19,Q21) — much higher than that of the low power supply which has a much less onerous job to do — some HF loop stabilisation is needed and this is provided by the small capacitors C7 and C8.

The current limit transistor, which sits astride the supply to the control transistor, is normally turned on by a forward voltage developed across the diodes, D7 D8/D9 D10, in the path to the zener supply. However, if too much current flows through the circuit this foward bias will be diminished by the voltage drop occurring across R15/R16, and will ultimately switch this transistor off again. A similar function is carried out, in respect of the voltage across the pass transistor, by the two resistors R32 and R17/R33 and R18. Acting together, these current flow and voltage sensing networks generate the limiting characteristics shown in Fig. 6.

In order to help the operation of the cut-out circuit, a pair of diodes, D13,D14/D15, D16, have been added in comparison with the original circuit. This means that the base potential of the control transistor normally sits at about 1.65V with respect to the 0V line. When the trip circuits operate, this is clamped at 0.65V, and the control transistor and the pass transistor are both cut off. The LED is then illuminated, to indicate a fault condition.

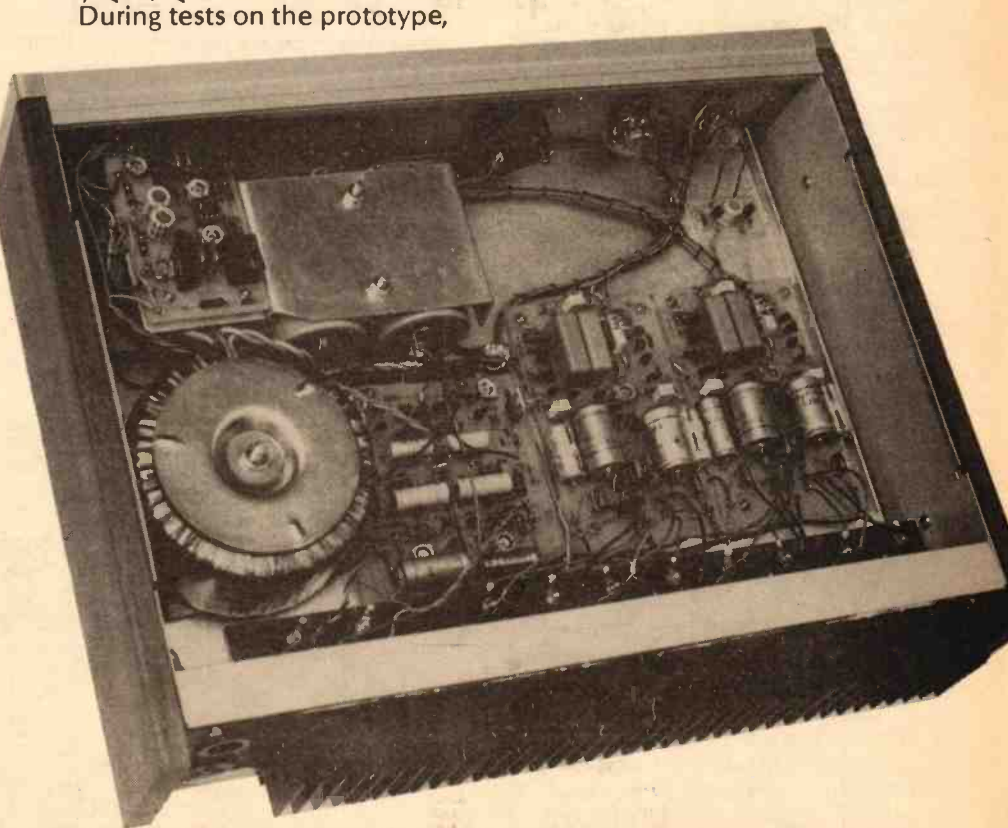
As mentioned above, the power supply can be momentarily reset by applying a discharged condenser between the 0V line and the bases of the trip transistors, Q14/Q15.

During tests on the prototype, the output voltage of the PSUs, under quiescent conditions, were constant for mains input voltages varing between 170V and 260V RMS, and the output AC ripple was less than 3mV. The measured voltage drop, from minimum to maximum measured load (one channel driven at 117 watts) was less than 1V.

Setting Up The Amplifier

Normally my amplifiers start more or less as a plain sheet of aluminium, of a bit larger than the expected necessary size to allow for oversights, on which the bits and pieces are fixed in a way which looks sensible when all of them are eventually to hand, and working as I hope. The result inevitably looks a bit less polished than the commercial equivalent. In this instance, I was provided with some nicely made metalwork into which I fitted the various PCBs which I had previously made, along with the other essential major components, in the best practicable arrangement in relation to the plugs, sockets and controls.

The result, shown in the photograph is perhaps a little less neat, on the inside, than I would



The interior of the prototype: along the top (L to R): meter driver PCB (mounted over on/off and mute switch), reservoir caps, switch-on muting PCB; bottom: transformer, PSU, 2 x power amps

expect the final kit version to be.

Externally I am very pleased.

I have mounted all the ten power transistors (eight from the amplifier, and two from the power supply) on a length of substantial gauge angle aluminium which is clamped to the back plate of the amplifier. On the outside of this back plate, four Redpoint heat sink blocks are mounted, side by side, to give a heat sink 32cm long by 5cm deep with total fin length of 3cm. This heat sink has a calculated capacity of 0.4°C/watt, and gets only mildly warm in use. This arrangement, in which the transistors are mounted horizontally inside the box, is one which I prefer, since it protects the exposed cases of the transistors from inadvertent electrical contact, and makes their connections easy to join. The white silicone/zinc oxide heatsinking paste should be applied to all the joins through which heat is to pass.

I have used 4mm insulated terminal binding posts (10 amp rating) mounted on the rear panel, for the LS output connections, and these are joined to the output pins on the PA PCB by twisted pairs of 24x0.2mm PVC insulated cable (4.5A rating). The 0V pins at the output of the PA boards are taken, using the same type of wire, to a conveniently positioned chassis earth point, which should not be too far away from the reservoir capacitors.

I have shown the mains input, transformer, and reservoir capacitor circuit and suggested layout in Figs 7 and 8, and I have indicated, by heavy lines, which of the connections it is preferred should be short, and of the thickest gauge of stranded wire which it is practicable to solder. The important thing to remember is that the wires from the capacitor tags to the earthing post are carrying heavy currents and will have significant voltage drops along them. They should therefore go directly to the earthing post, and nothing else should be joined to the lug on the capacitor case.

The output 0Vs from the PAs, and the input and output 0V lines from the power supply unit, are similarly taken directly to this post, with as substantial a gauge of wire as reasonable. The input earths for the amps. are commoned both at the input phono sockets and at the gain control, and joined to the earth post with a single wire. By this means, the heavy pulsating

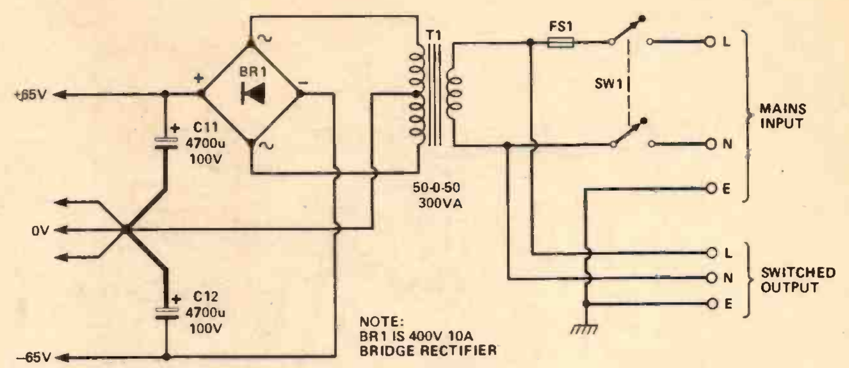


Fig. 7 Mains input circuit for power supplies.

currents in the output DC supply and return lines are kept out of the input signal path, where they can introduce significant amounts of distortion, and impair the performance of an otherwise impeccable amplifier.

Since the input sockets are also mounted on the back panel of the amplifier, it is necessary to screen these so that they do not pick up capacitatively coupled signals from the cases (which are connected to the output) and wiring associated with the output MOSFETs. It is also necessary to isolate these input sockets from the chassis earth, to avoid earth path signals which could contain both hum and distortion inducing voltages. I solved this problem on the prototype by making up a little tin box, with soldered corners, on which the input phono sockets were mounted, and which itself was held to, but insulated from, the back plate.

Signal Muting

This is a facility for which there is provision on the PA PCBs, but which I did not describe in the last part of the series. This employs the circuit layout shown in Fig. 10. In this a normally closed push switch (two-gang) is inserted in place of the link shown on the PCB. This is bypassed by a 1nF capacitor and a 470k resistor, so that when the switch is opened, the gain of the amplifier is reduced from 122 to 1.3, at all frequencies below about 100kHz — which are safely supersonic.

The 1nF capacitor is there to avoid jeopardising the feedback safety margins at HF which are a lot less at unity gain than at 122.

By the use of this control, the amplifier can be effectively 'muted' during switch-on, to minimise plops, or during other operations where it may be desired to avoid unwanted noises. I have suggested this technique, as an option, since my decision not to use a relay has removed the otherwise attractive option which this offers to disconnect the LS lines until the amplifier has had a chance to settle. The 470k resistor across the mute switch gives C7, in the feedback line, a chance to charge, over a few seconds, to its normal operating DC level.

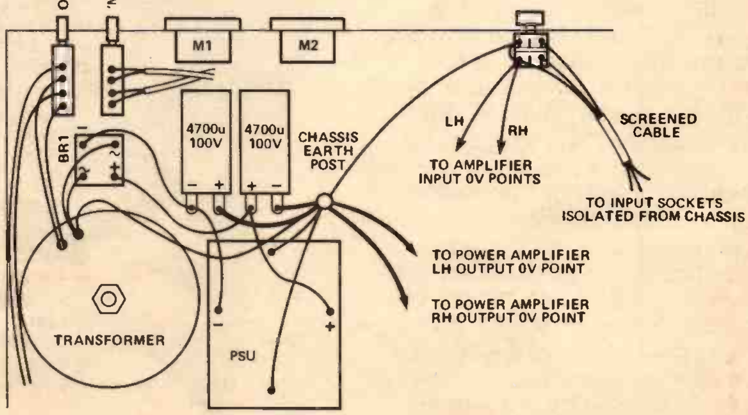


Fig. 8 Suggested lay-out of earth (0V) wiring for power amplifier and power supplies.

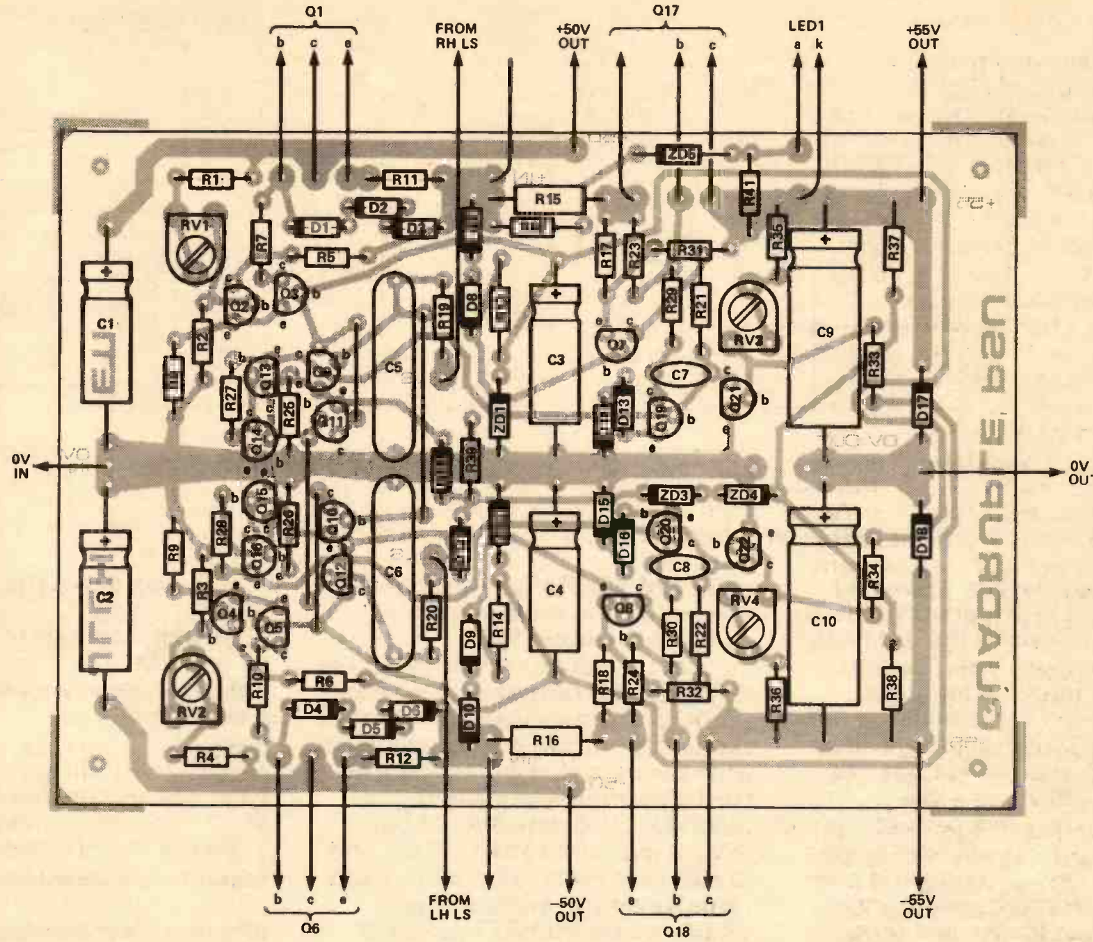


Fig. 9 Overlay diagram for the PSU.

PARTS LIST — PSU

RESISTORS (all ¼W 5% unless stated)	
R1,4	33k
R2,3	10k
R5-10	4k7
R11,12	33R
R13,14	15k ½W
R15,16	0R15 (or 0R22, see text)
R17,18	120R
R19,20	1M0
R21,22,37,38	10k ½W
R23,24	10k
R25-28	4k7
R29-32	12k ½W
R33,34	18k
R35,36	68k
R39	100k
R40	12k ½W
R41	4k7
RV1,2	10k horizontal preset
RV3,4	27k horizontal preset
CAPACITORS	
C1,2	100µ 64V axial electrolytic
C3,4	220µ10V axial electrolytic
C5,6	2µ2 63V polyester, radial
C7,8	1n0 disc ceramic
C9,10	220µ 64 V axial electrolytic
C11,12	4700µ 100V can

SEMICONDUCTORS	
Q1	BD538
Q2,3,8,19	BC447
Q4,5,7,20	BC448
Q6	BD537
Q9,11,14,16,21	BC184
Q10,12,13,15,22	BC214
Q17	MJ2501
Q18	MJ3001
D1-16	1N914 or similar (16 off)
D17,18	1N4003
ZD1-4	10V zeners, 400mW
ZD5	30V (or 24V, as available) zener, 400mW
BR1	400V 10A bridge rectifier
LED1	single LED to choice
MISCELLANEOUS	
T1	50-0-50 (or 48-0-48) V 300VA mains transformer
FS1	1A mains fuse and holder
SW1	mains switch to choice
PCB; mains input socket; mains output socket; wire, etc.	

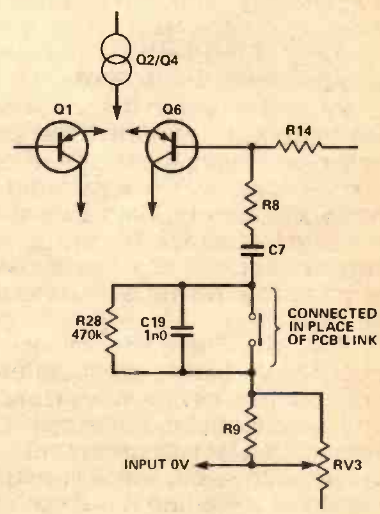


Fig. 10 Circuit arrangement for amplifier muting.

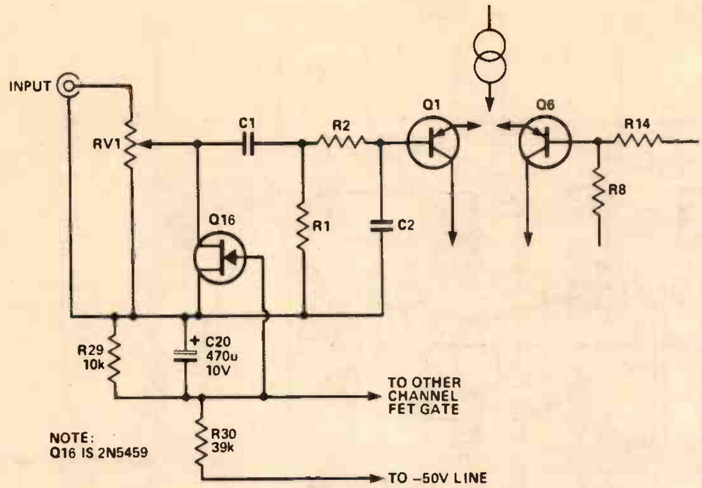
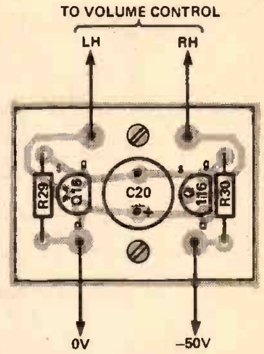


Fig. 11 FET input clamp circuit.

Fig. 12 Overlay for circuit of Fig 11.



Additional facility to which I had referred in an earlier article, as a possible option, is the use of an FET as a normally open switch across the amplifier inputs, as shown in Fig. 11. Normally, at the moment of switch-on C20 will be uncharged, and the FET, Q16 (a 2N5459), will act as a low-impedance resistive path across the inputs, which will effectively zero the volume control and prevent the amp from producing distorted signals for the few seconds during which the DC supply lines from the power supply rise up to their final operating voltage. The FET bias is derived from the −50V line, and lags behind this in its rate of voltage rise, as C20 charges through R30, towards its final operating voltage of −10V, at which the FET is fully cut-off, and is effectively removed from the signal circuit.

Power Amplifier Quiescent Current

I had omitted to discuss this, inadvertently, from the description in the previous part of this article. The optimum value, if twinned MOSFETs are used in the outputs, is 250mA/channel. The amplifier can be operated, at a lower maximum output power but without any other penalties, with a single N-P-MOSFET pair. This will give about 65W. In this case a quiescent current, per channel, of 120-150mA is required. With the circuit shown, the 250mA quiescent current allows 0.5 watts of output in pure class-A, and it is surprising just how much of ones programme, in almost everything except heavy rock or reggae, falls below this level. (To organise the circuit with single MOSFETs, just delete one pair of N- and P-channel devices from each output four.)

On this score, on tidying up the wiring to the output power MOSFETs, it became clear to me that its actual layout was a bit over-critical. I therefore propose that the gate resistors, mounted close to the MOSFET gate pins, should be increased from 150R (16/17) and 220R (18, 19/21, 22) to 1k0 each. This solves the awkwardness. When single MOSFET pairs are used, this problem doesn't arise.

As can be seen from the photograph of the prototype power amplifier internal layout, I have laced quite a few of the input cables together, in the interests of neatness and in keeping them together in a safe position. Please **do not** do this with the output wiring or the wiring to the MOSFET pins, which should be spaced out, but not more parallel than inevitable. MOSFET pairs are likely to see parallel wiring to their pins as an invitation to oscillate (this problem is even worse with the recent very fast T-MOS devices, and I decided that these were not sensible for use by DIY amplifier builders, in spite of their otherwise superb technical possibilities).

Output Power Meter

It is certainly a useful feature to have a pair of channel power output meters mounted on the front of a power amplifier. However, that is where agreement ends. If the meters, which should be peak reading, with a fairly slow decay rate, have a scale which is linear in voltage it will result in the necessary calibration for power output being very cramped at the top end, since $P = V^2R(\text{load})$. It will also require the meters to be hand calibrated, which isn't an easy thing to do oneself if the result is to be neat-looking. On the other hand, the circuit is simple to organise.

If the purpose of the meters is to make the user aware of his proximity to the amplifier overload margins, so that he can use it within its limits, it is much more satisfactory to have a measuring circuit which is linear in terms of power output. This also solves the problem of a neat scale calibration. I have therefore adopted this approach based on a 100uA meter movement, scaled 0-100 as watts. This makes it very easy to see where one is operating in relation to the overload threshold, but it does mean that the meters will be sitting near the zero mark for most of the time (unless one likes ones music very loud!)

The circuit I have adopted is shown in Fig 13. In this I have used a junction FET as the 'square law' element, in the input limb to an inverting mode IC amplifier. The gain of the amplifier depends on the ratio of the impedance of Q1 to the resistance of R5 and RV1. When the FET has zero bias, its AC impedance is low, and the amplifier gain is (relatively) high.

PARTS LIST — POWER AMPLIFIER

There are additional parts to implement the switch-on mute.

R29	**10k**
R30	**39k**
C20	**470μ 10V PCB electrolytic**
Q16,116	**2N5459**

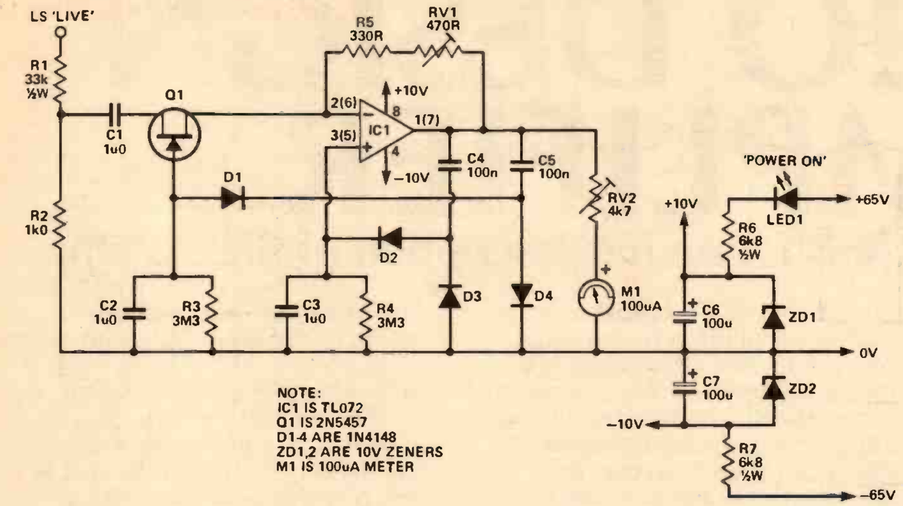


Fig. 13 Peak-reading linear scale power meter (8 ohms load).

PARTS LIST — POWER METER

RESISTORS (all ¼W 5% unless stated)	
R1,11	33k
R2,12	1k2
R3,4,13,14	3M3
R5,15	330R
R6,7	6k8 ½W
RV1,11	470R horizontal preset
RV2,12	4k7 horizontal preset
CAPACITORS	
C1,2,3,11,12,13	1µ0 polyester
C4,5,14,15	100n polyester
C6,7	100µ 16V PCB electrolytic
SEMICONDUCTORS	
IC1	TL072
Q1,11	2N4557
D1-4, 11-14	1N914 or similar (8 off)
ZD1,2	10V 400mW zener
LED1	single LED to choice
MISCELLANEOUS	
M1,11	100µA FSD moving coil meter to choice
PCB, wire, etc.	

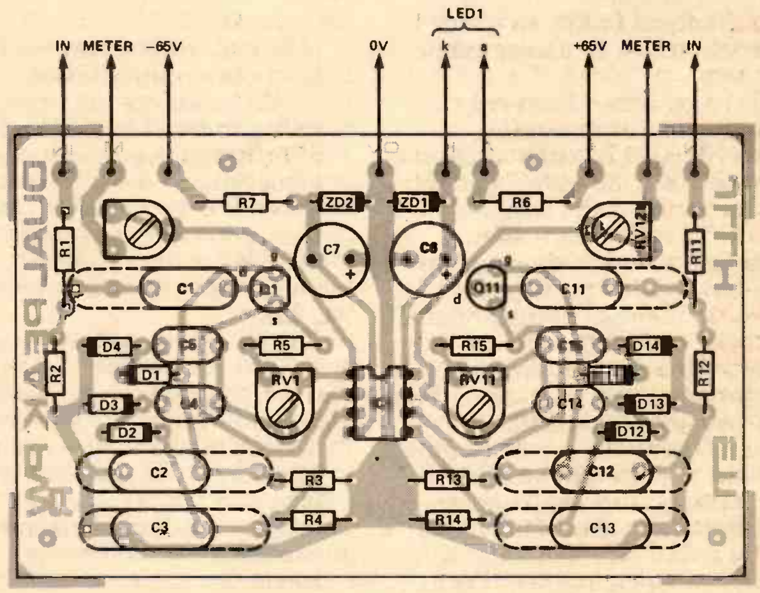


Fig. 14 Overlay diagram for the power meter.

When an AC signal is applied to the input, via R1, R2 and C1, the amplifier output is rectified and applied as a positive-going voltage to the non-inverting input of the op-amp (which makes its output, and consequently its inverting input voltage also move +ve), and as a negative-going voltage to the gate of the FET, in relation to its positive-going source and drain. This biases the FET to a higher impedance and reduces the gain of the amplifier. The large the input signal, the lower the gain of the amp, and the higher the bias voltage.

Although FETs vary a bit from one to another, every one of about a dozen Motorola 2N5457s could be adjusted to give a reasonable square-law characteristic. The technique is to apply a measured input voltage (Vin(RMS)= $\sqrt{}$ P.Rload) — for example 12.65V RMS for 20 watts into 8 ohms, and 26.8V for 90 watts — use the 'linearity' pot, RV1, to set the power reading at say 20 watts, and use the 'scale' pot, RV2, to set the meter reading at the high end. This will need to be done iteratively, going from one to the other and back again, since they influence each others readings. However, one wins in the end. I have shown in Table 1, below, the results on my prototype using 20W and 90W as the adjustment points.

V (rms)	P. (8 ohms)	Meter reading
4.0V	2W	2W
6.32V	5W	5W
8.94V	10W	10W
10.95V	15W	14W
12.65V	20W	20W
15.5V	30W	31W
17.9V	40W	41W
20V	50W	52W
22V	60W	63W
23.7V	70W	72W
25.3V	80W	82W
26.8V	90W	90W
28.3V	100W	95W

Table 1 Calibration of the prototype power meter.

AUDIO DESIGN AMPLIFIER

John Linsley Hood finishes up his description of the system.

The Editor of ETI had decided, and this was a decision I gratefully accepted, that if this amp and preamp was to be a contender for the top, then it also must look the part. Since any DIY metalwork would obviously not meet this requirement, a professional case-maker had to be brought in, and through the good offices of ETI, Newrad Instrument Cases Ltd, were called to my help.

This has resulted in a very elegant looking amp, in satin finished metalwork with wooden side panels, but led to the sort of complications which can arise when the circuit designer and manager of the body shop live in offices a hundred miles apart. Fortunately, in the case of the preamp (no pun intended) the circuit boards and metalwork settled down together very happily, as shown in the photograph.

To avoid possible earth loops, I have linked all the earthy sides of the rear phono sockets together, and tied these to the main chassis plate by a very short length of wire at a point adjacent to the pick-up inputs. The earthy side of the phono inputs is also taken directly to the pin on the RIAA Input board, which I have mounted as close as practicable to the pick-up phono sockets.

The power supply board, mounted at the RH rear of the chassis is positioned close to the mains inputs, and as far removed from the inputs as sensible. I have used the + and − 15 volt and 0V points on the PSU PCB as distribution points to take wires to each of the active modules (ie, the active boards are wired for supply purposes to the PSU, not to each other).

Because the PSU and the headphone amp both require to dissipate a small amount of heat, I have tied the case clips of the transistors and voltage regulators, through appropriate insulating hardware, to a 'Z' shaped strip of metal, clamped, in turn, to the main chassis plane. This has proved in practice to be quite adequate to ensure that all devices keep cool.

The small input buffer stage (my apologetic afterthought) is mounted immediately behind the input selector switch, and I have adopted the option of taking all signals through it, so that the whole internal signal wiring is at a low impedance, and therefore largely immune to unwanted pick up.

The LEDs which serve as function reminders are all connected through the appropriate selector switches from 0V to +15V, via a 3k3 resistor in series with each. If one is sitting on the opposite side of the room, it is useful to be able to check that one hasn't inadvertantly left the tone control or rumble filters in circuit after the need has passed.

Because ETI and Newrad have gone to some trouble to ensure that the completed unit is a pleasing assembly, I have tried to keep the wiring neat and have laced it together in bundles, with appropriate colour coding for functions, where there would not be any possiblity of unwanted cross coupling. Do not, for example, lace up inputs and outputs, unless these are carried in screened cables.

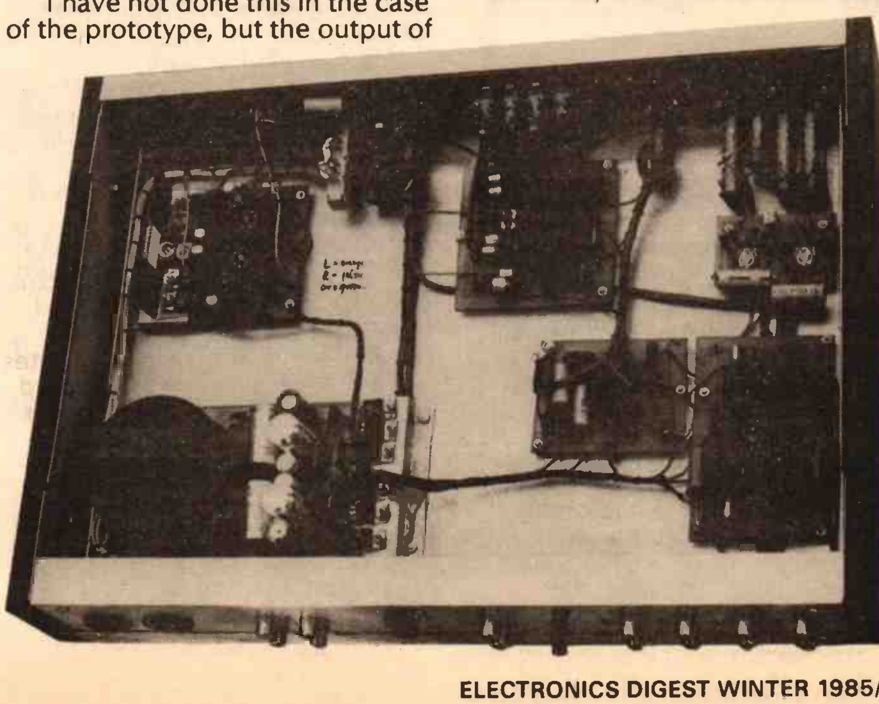
I have not done this in the case of the prototype, but the output of the headphone amp could be taken to the rear of the unit to provide a higher signal level output to a more normal power amp unit.

The headphone amp has its own + and −15V supply, which is separate from that of the rest of the preamp, and it also has a separate connection on the 0V line to the 0V output point on the PSU.

The Proof Of The Pudding

This lies, it is said, in the eating. So, after all this effort, how does this amp and preamp combination sound? Unfortunately, doing all the important things right and getting a good technical specification is not in itself a cast iron guarantee that the sound will be well, if only because no-one can be quite sure that they know all the important things or what is necessary to specify. For these reasons, all power amplifiers and preamplifiers sound very slightly different — from one design to another — though there does, in my experience, tend to be a family likeness between the designs of one particular designer in terms of sound quality.

To be sure, these differences are small, and tend to make them-



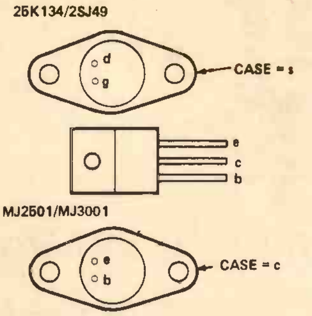


Fig. 1 The pin-outs of some of the off-board transistors.

selves more apparent after a few hours or a few days acquaintance with a new system. This coming to terms is greatly helped if the environment, the music in question, and the ancillaries are familiar. I do not know whether I am speaking for other designers when I say that I am always a little apprehensive on the first trial, to be sure that all is as I hope. In this particular instance I am very well pleased. I have heard a lot of amplifiers. I think that this is the best I have heard yet. Moreover, this opinion is shared by some of my friends whose judgement I value, and I have used as guinea pigs in listening trials. The particular, and unexpected, quality which this design has shown, apart from a surprisingly solid bass (which could be simply the benefit derived from the fully stabilised power supplies — this is the first time I have used one in my domestic amps) is an extraordinary degree of sound detail and 'transparency', of a kind which I have only ever found in the past with headphone amps.

The effect of this is to disclose a wealth of previously unremarked minor aspects and incidental noises from instruments, all of which tend to add to the vividness of the fantasy world created in ones living room by the artistry of the programme or record producer.

Obviously, no author will want to report that his efforts have been unsuccessful, and I am very pleased therefore that I can be both truthful and complimentary. I hope that in time this verdict will be shared by others.

BUYLINES

The Audio Design Amplifier system contains a number of unusual and/or hard-to-get parts, most particularly the power amplifier semiconductors, presenting the constructor (not to mention the editor) with a Snark hunt for the necessary components. Happily, Hart Electronic Kits have agreed to provide a set of the more elusive components, including the semiconductors. Write to Hart Electronic Kits Ltd, Penylan Mill, Oswestry, Shrops. SY10 9AF or phone 0691-652894 for prices and availability.
The original cutom-built cases are still available from Newrad Instrument Cases Ltd; their new address is Unit 19, Wick Industrial Estate, Gore Road, New Milton, Hants. BH25 6SJ: phone 0425-621195. Prices were not available at the time of going to press — please contact Newrad for up-to-date information.

Odds And Ends

A question which inevitably arises with any design is the extent to which active components can be interchanged. In general, within limits, devices should be interchangeable without much overall effect on the performance. These limits are:

1. **Working Voltage** — don't use a 20V max. transistor where the line voltage is 50V, but the converse is OK
2. **Current Gain** — if a chosen device has a current gain in the range of 250-400, one with a gain of 40 may give disappointing results; one with a gain of 120 would probably be satisfactory.
3. **Noise Figure** — some devices are specifically chosen for low noise (these usually have a high current gain too, but this will not, by itself indicate low noise); this is usually important only at the front ends of preamps.
4. **Gain Linearity** — this is usually important in output devices, and may influence the choice of particular types.

Also in output devices, the HF characteristics will have determined the type of feedback compensation employed. It is usually as well to stick to the author's recommendations here.

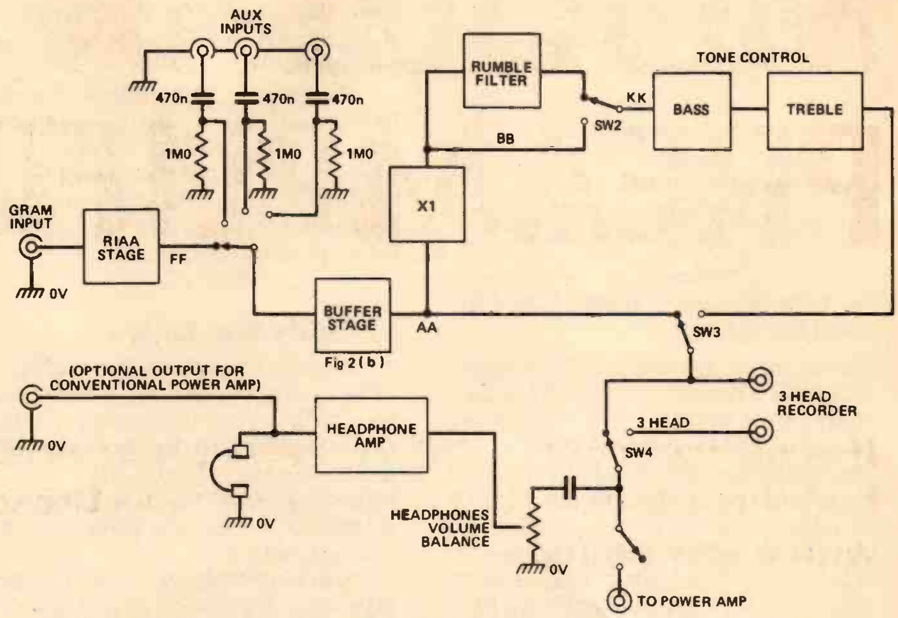
In my own case, and I suppose I am typical, I have certain device types which I keep in the boxes in my workshop, and which I buy in 100-off quantities when the stocks need replenishing (because this is cheaper). Therefore, I tend to use these devices in my designs, simply because they are to hand — not necessarily because they are any better. Whether substitutes will work as well I cannot say, and cannot easily test — but I'd guess that they will. Often in the evolution of a design I will have swapped types around a bit, to make sure that my first choice was the best. I do not recall that I have ever found much difference.

Ferrite beads are sometimes advocated as a simple way of cutting out unwanted RF breakthrough. Treat these with care. If no significant current is flowing in the wires around which they are threaded, they will do no harm, but in output stages they can be disastrous. For example, a single ferrite bead around one LS lead will worsen distortion at 10 watts and 20KHz from 0.015% to 0.4% ! Just like that!

Finally although I had no idea that the outcome of my series on Audio Design would be that I would end up with the nicest, and best-looking amplifier I have yet owned, I hope that the explanations and calculations I have attempted will have dispelled any beliefs that good results arise from some kind of magic. They are the outcome, all being well, of sensible layout structures and the right answers to the sums which can be made to relate to them. Nothing in this field is sacred, and no-one is ever absolutely right in the choices made. If you know the reasons for the choice and the sums that have been done, you can do the same sums, and maybe improve on the results.

CORRECTIONS

A reader has, very properly, pointed out that my RIAA stage, Fig. 3 (and 2) ETI June 1984, will only work as claimed, (and as I ruefully admit, as calculated and measured) into a load which has effectively an infinite impedance. With the actual load resistances implied by the circuit layout shown in Fig. 1, this condition is not met, and the 75 us second integration characteristic of the RIAA spec is impaired. The best answer to this problem is to feed the RIAA stage into a buffer circuit which does look like an infinitely high impedance. Two possibilities exist for this: 1. to use a pair of FET input ICs as unity gain voltage followers, (a TL072 or a LF353 would do this nicely) or 2. since I prefer at this point to avoid ICs, to make a discrete component buffer stage. These two options are shown in Fig. 2a and b. The small bipolar-FET symmetrical compound source follower circuit works extremely well, with negligible steady-state or transient distortion and I am tempted to suggest that this should follow the input selector switch as shown in Fig. 3, as a universal input buffer, which would allow all the subsequent signal wiring to be at a low impedance.



3 Alternative lay-out of preamp using discrete component buffer stage.

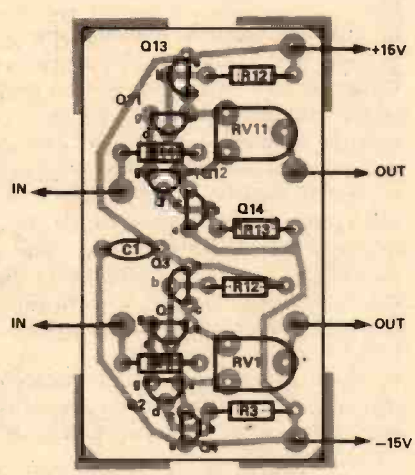


Fig. 4 The overlay diagram of the discrete component buffer.

PARTS LIST

RESISTORS	
R1,11	330R
R2,3,12,13	4k7
RV1, 11	1k0 lin horizontal preset
CAPACITOR	
C1	470n polyester
SEMICONDUCTORS	
Q1	2N5457
Q2	2N5460
Q3	BC212
Q4	BC184
MISCELLANEOUS	
PCB	

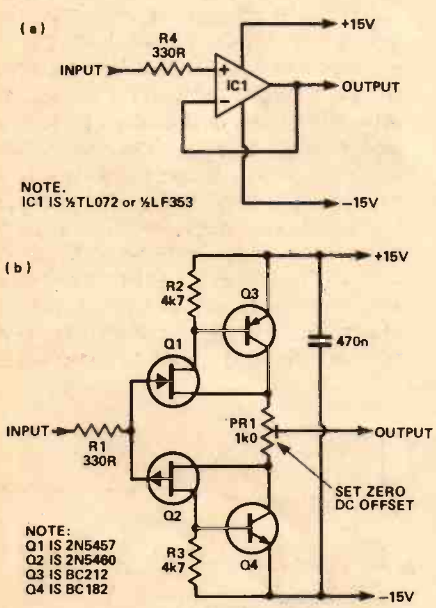


Fig. 2 RIAA stage output buffer options (one channel only shown): (a) using an op-amp, Zin in excess of 1000 Megohms; (b) using discrete components, Zn in excess of 100 Megohms.

MODULAR PREAMPLIFIER

If walls had ears. . . they'd certainly appreciate ETI's audio building blocks. Barry Porter surveys the ground and comes up with a few plans.

Pre-amplifiers come in all shapes and sizes, yet all are designed to perform the same basic function — to select the output of a given signal source and apply it to a power amplifier at a level that can be adjusted to give the required power output.

It is evident that no pre-amplifier on the market satisifies everyones requirements. Some people must have tone controls while others will not entertain them at any price. Some want moving coil cartridge inputs, others do not . . . What is really required is a pre-amplifier that can be constructed to suit individual needs — a sort of audio Lego kit!

After much deliberation it was decided to design just such a unit, using a mother board system with the active circuitry on individual plug-in boards. This makes it possible to build a pre-amplifier to virtually any configuration by just changing the mother board, which consists primarily of busses and interconnections between the boards carrying the signal circuits.

To make this system as flexibile as possible, the individual building blocks have been broken down into the following categories:-

1. Disc amplifier (Moving coil or magnet)
2. Unbalanced output stage (With provision for active balance control)
3. Balanced output stage
4. Tone Controls
5. Headphone Amplifier
6. Muting Relay Control
7. Power Supply

The first pre-amplifier to be described will use units 1, 2, 6 and 7, so a description of these circuits will be given first, followed by details of how to link them together to make a pre-amplifier that will out-perform many manufactured units that cost the proverbial arm and a leg.

It will be noticed that all signal circuitry is based on the use of operational amplifiers, and as this may raise a few eybrows, a short sermon in defence of this practice is called for. . . In the eary days of integrated circuits, there was created a device known as the 741. Although this humble chip performed well at low frequencies, its limitations at the upper end of the audio spectrum rightfully gained it a reputation for sonic nastiness. Fortunately, progress and evolution have been quite active, and about five years ago a far superior device emerged. Called the 5534, this quickly became the standard IC of the professional audio industry, which consumed them in great quantities. So it is safe to assume that any recent recording has passed through quite a number of these devices — as many as two or three hundred in the case of a multi-tracked original — so no excuses are offered for using a few more in the reproduction chain.

Perhaps comment should be passed on one or two other practices. Following a recent discussion with Martin Colloms, the author carried out some tests which showed that certain types of capacitor can degrade the sound of audio circuits. In particular, the common polyester types proved to be unsatisfactory, as did the standard type of aluminium electrolytic, especially when used without a defined polarizing voltage. For this reason, all small value capacitors should be either polystyrene or polycarbonate (polypropylene are marginally better than polycarbonate, but are difficult to obtain, expensive and very large in size) and the inter-stage and feedback shunting electolytics should be of non-polarized

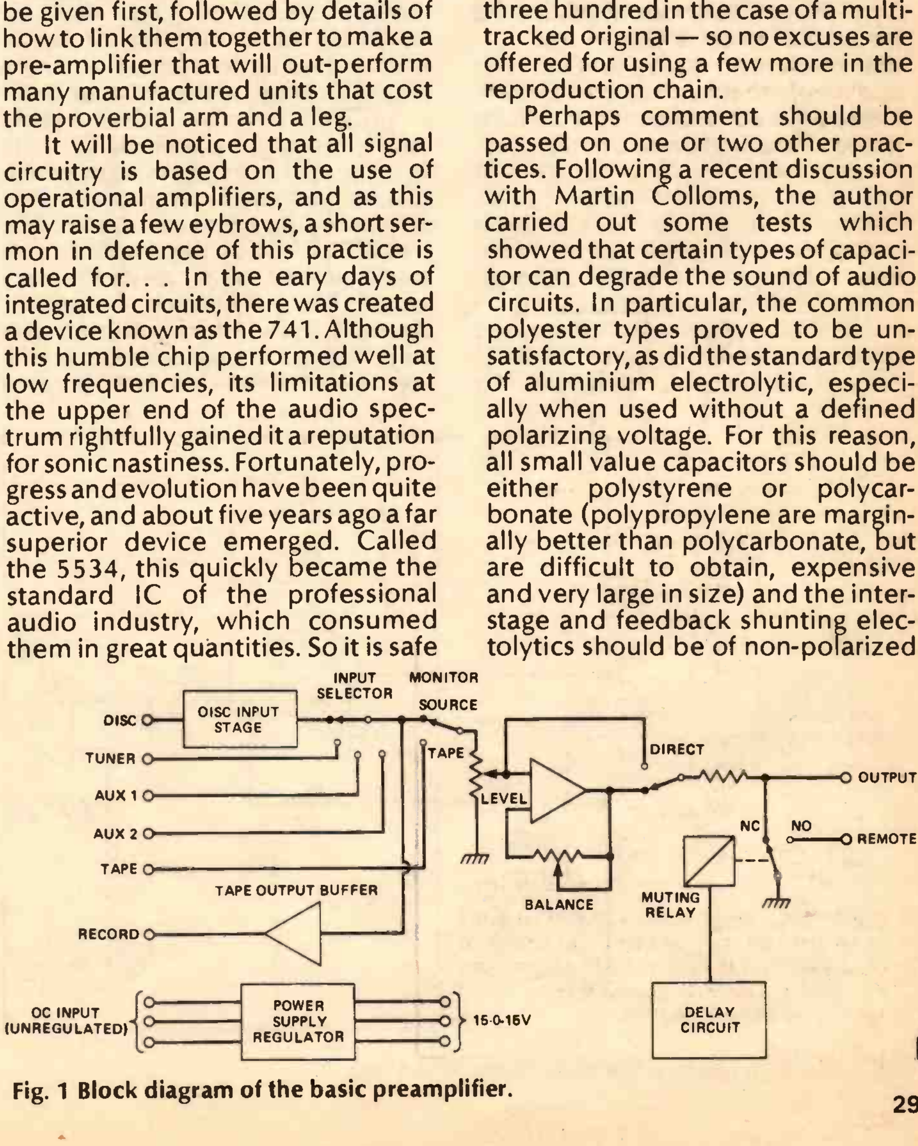


Fig. 1 Block diagram of the basic preamplifier.

construction and should be paralleled with a smaller value polycarbonate, which helps to flatten the high frequency impedance curve.

With that little lecture out of the way, we come to the design of the pre-amplifier, shown in block diagram form in Figure 1. It consists of a disc amplifier stage which may be for moving coil or moving magnet cartridges, input selector switch, level control, active balance control stage and muting circuit. A switch marked 'Direct' may be used to bypass the balance stage, so that if a source with sufficient output capability, such as a Compact Disc player, is connected to an auxiliary input, it may be routed to the power amplifier with only the ganged level control in it's path. When the balance stage is in circuit, the input sensitivity of the tuner and auxiliary inputs is 200 mV for 1.0 V output. The input sensitivity of the disc amplifier can be set to match the cartridge in use, and the input loading may be set to any suitable value of resistance and capacitance. The overload margin of the disc stage is 32dB at all frequencies which should be ample, even with the hot cuts that are sent to annoy us. On tuner and auxiliary there will be no overload problems, as the level control is placed in front of the active circuitry. A note of warning here though — it has been found that the best value of level control is 10k ohms, as this is not likely to cause problems when the direct path is used. Unfortunately, some equipment requires a greater load than this for correct operation, so the choice of level control value should be made with due consideration to this.

Now to the individual circuits that are to be used, starting with the most critical which, of course, is also the most difficult to design and engineer.

Disc Amplifier Stage

In simple terms, this circuit has to amplify the output of a pick-up cartridge to a higher, more manageable level, and at the same time apply equalisation to the RIAA standard, this being defined by three time constants: 3180 μs, 318 μs and 75 μs (corresponding to 50.05 Hz, 500.5 Hz and 2.122 kHz).

The amount of amplification will depend upon the output voltage of the cartridge in use, moving coil types typically requiring some 20-25 dB more gain than moving magnets. To give some idea of the magnitude of the problem, a moving coil cartridge with a nominal output of 0.2 mV at 1 kHz requires amplification by more than 9000 at 20 Hz to give 200 mV at the amplifier output. The RIAA equalisation curve which, relative to 1 kHz, rises to +19.27 dB at 20 Hz and drops to −19.62 dB at 20 kHz, may be obtained in a number of ways. Active feedback around a single amplifier stage is the most popular — possibly because it gives a good specification on paper, particularly in respect of overload margin, which is constant with frequency. This configuration has some drawbacks, usually caused by the amplifier output stage having to drive the very capacitive feedback network. Another type of circuit that has become quite popular uses a passive network between two stages of amplification. Subjectively, this method proved quite successful, but it suffers from inferior noise performance and greatly reduced high frequency overload characteristics.

The circuit to be described may be termed a 'hybrid', in that it has part active and part passive networks. In order to keep noise to a minimum when using a moving coil cartridge, the technique of forming the input stage from several transistors in parallel has been employed. The LM394 integrated circuit contains 100 individual devices divided into two sets of 50 each. By joining the connecting leads of each set together, all 100 transistors are used. When the circuit is operated with a moving magnet cartridge, the LM394 can be replaced by a similar dual device containing a pair of normal, bi-polar transistors. Not only does this help the economics, it should also be quieter as a better impedance match is likely.

Figure 2 shows the complete disc input circuit, and Table 1 gives the component changes necessary in order to adjust the input sensitivity to allow operation with a wide range of cartridges. The input components, R1 and C1, should be chosen to accurately load the cartridge in use, but as a general rule should be 100R and 22n for moving coil cartridges and 47k and 220 pf for moving magnets.

The operation of this circuit is quite straightforward. The LM394 is set at 2mA collector current, which causes the inverting input of IC2 to be at 6.3V. The non-inverting input is therefore biased to be at the same voltage. Feedback is applied to the

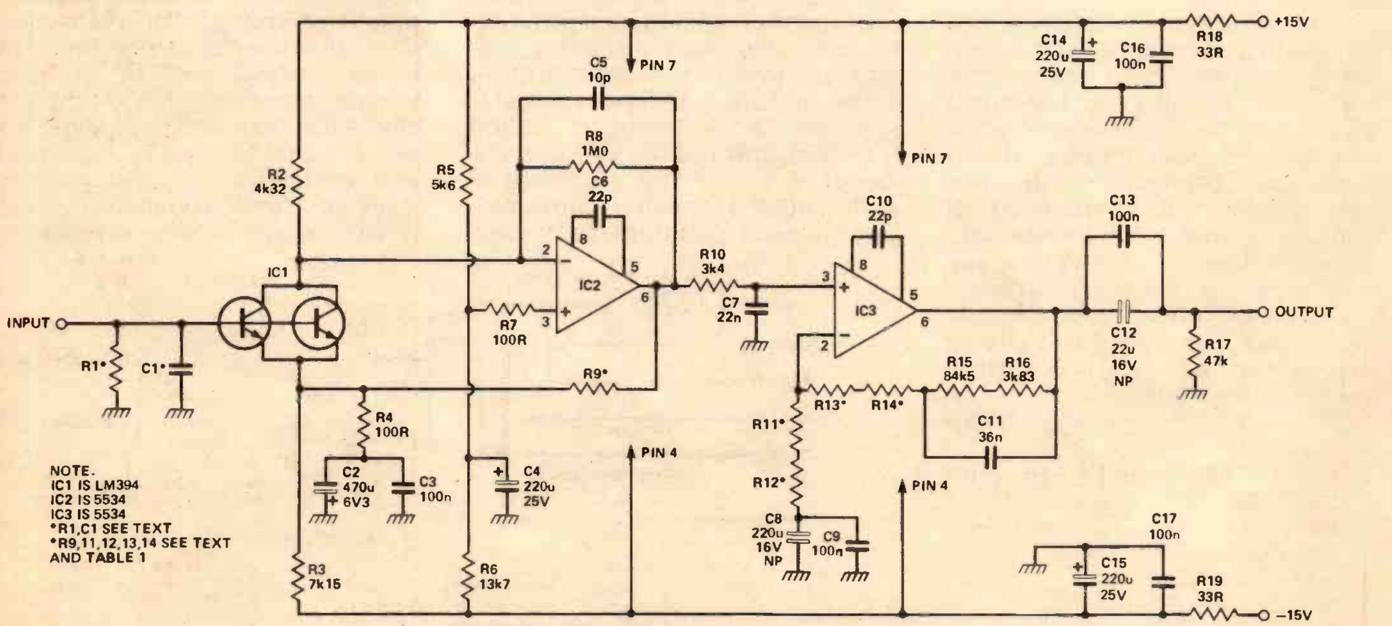


Fig. 2 Circuit diagram of the disc preamplifier stage.

emitter of the LM394 by R9, which, together with the 100 ohm shunt resistor R4, sets the gain of the first stage. This may be calculated from:

$$A(dB) = 20 \log (1+\frac{R9}{100})$$

An important consideration at this point is signal overload. This is measured by applying a range of frequencies that have been subjected to inverse RIAA equalisation, and is stated as the amount that the input level can be increased above its nominal rating before the output signal is clipped or severely distorted. As the rated output of the disc amplifier is 200 mV and clipping occurs at about 8.0 V, the overload margin should be:

$$20 \log (\frac{8}{0.2}) = 32dB$$

at all frequencies. As no equalisation takes place in the first stage, care has to be taken to ensure that high frequency overload does not occur — remember that the 20 kHz input level will be at +19.62 dB — which means that the rated input level multiplied by the first stage gain must not exceed (19.62+32.0) = 51.62 dB below 8.0 V, or about 21.0 mV. A quick calculation based on the gain settings given in Table 1 shows this to be the case, and consequently a full 32.0 dB overload margin will be maintained throughout the audio spectrum.

Following the input stage, a passive network provides the 75μs time constant (3400x22n = 74.8μs) and a network in the feedback circuit of IC3 gives the 3180μs and 318μs breakpoints. The 36n capacitor, C11, with R15+R16 fix the first point (36nx(84.5k+43.83k) = 3179.88μs), and the same capcitor together with the series-parallel combination of R13, 14, 15 and 16 fix the second. Table 2 gives details of the RIAA characteristic over a range of frequencies so that performance of the disc stage may be checked for accuracy, which, with the specified components will typically be to within 0.1 dB between 20 Hz and 20 kHz. Stray capacitance may affect the extreme high frequency response, but the prototype

Input		1st Stage Gain	R9	R11	R12	R13	R14
0.1 mV	Moving coil	41.0 dB	11k0	536R	21R5	8k87	412R
0.2 mV		35.0 dB	5k6	536R	21R5	8k87	412R
0.3 mv		31.48 dB	3k65	536R	21R5	8k87	412R
0.5 mV		27.0 dB	2k15	536R	21R5	8k87	412R
2.0 mV	Moving magnet	20.0 dB	887R	909R	82R5	8k06	806R
3.0 mV		16.48 dB	562R	909R	82R5	8k06	806R
5.0 mV		12.0 dB	301R	909R	82R5	8k06	806R
8.0 mV		7.96 dB	150R	909R	82R5	8k06	806R

Table 1 Component values for the disc amplifier stage.

f(Hz)	dB	f(Hz)	dB	f(Hz)	dB
0	+19.911	100	+13.088	10k	−13.734
5	+19.868	200	+8.219	15k	−17.157
10	+19.743	500	+2.648	20k	−19.620
20k	+19.274	1k	0 (Ref)	30k	−23.117
30	+18.593	2k	−2.589	50k	−27.541
40	+17.792	3k	−4.740	100k	−33.556
50	+16.946	5k	−8.210		

Table 2 RIAA equalisation characteristics of the disc amplifier.

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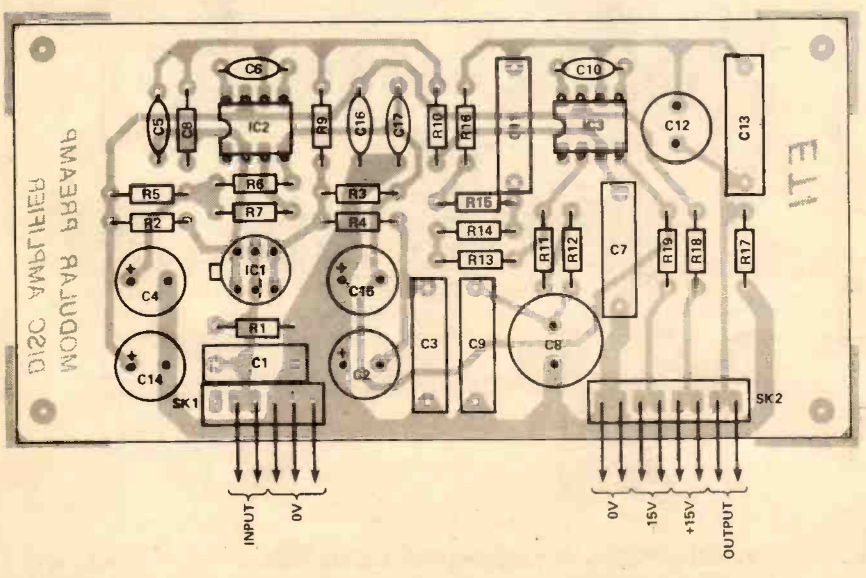

Fig 3 Component overlay of the disc amplifier PCB.

PARTS LIST

Disc Amplifier (one channel only)

Resistors (all 1% metal glaze or metal film)

R1	47k	see text
R2		4k32
R3		7k15
R4, 7		100R
R5	200	5K6
R6		13k7
R8		1M0
R9		see text
R10		3k4
R11, 12, 13, 14		see text
R15		84k5
R16		3k83
R17		47k
R18, 19		33R

Capacitors

C1	220p 24	see text
C2	32	470u 6V3 radial electrolytic
C3,9, 13	108	100n 250v polycarbonate
C4, 14, 15	78	220u 25V radial electrolytic
C5		10pf 160V 2½% polystyrene
C6, 10		22pf 160V 2½% polystyrene
C7	44	22n 63V 1% polystyrene
C8	22	220u 16V non-polarised electrolytic
C11	33+33 28 106	36n 63V 1% polystyrene
C12	40	22u 16V non-polarised electrolytic
C16, 17		100n 100V polyester

Semiconductors

IC1	LM394
IC2, 3	NE5534

Miscellaneous

SK1	6 way PCB socket
SK2	8 way PCB socket
PCB:	

was still accurate to within 0.5 dB at 100 kHz, which should not cause undue concern. There is no particular merit in having equalisation this precise, but as the network design is so simple, there seems no point in not keeping things as tidy as possible.

Unbalanced Output Stage

This stage, which incorporates an active balance control, is placed immediately after the main level control. The same circuit is used, without the balance facility, as a tape recorder output buffer. As shown in Figure 4, it has been designed to allow a limited amount of imbalance between channels — a maximum of 10 dB — with sufficient gain to raise the 200 mV input signals to the 1.0 V rated output of the pre-amplifier. The balance control operates by changing the amount of feedback around the amplifier stage.

This method preserves a good signal to noise ratio and overload margin while giving a well controlled image shift. Table 3 shows the amount of imbalance between channels at different positions of the control, and it will be seen that the calibrations are typically accurate to within 0.5dB.

Note that the output capacitor of the stage is within the main feedback loop of the amplifier. This is done for two reasons — it helps to counteract any effects introduced by the capacitor, and it avoids the danger of any DC voltage appearing on the control potentiometer which would introduce noise whenever the control was operated. There have recently been some suggestions that all passive components, such as switches, connectors and potentiometers should be provided with a DC bias voltage, as this helps to provide clean contacts for improved signal transmission. The theory behind this may be well founded, but long experience with recording studio mixing consoles, where it is quite common for DC to appear in all sorts of unwanted places, has shown that the working life of components subjected to this treatment is drastically reduced, so they need replacing much earlier than similar ones that have remained free of DC voltages.

The same stage is also used as a tape recorder buffer, with points A and B linked together and changed values for R4, R6 and C2. R4 should

Control Calibration	Imbalance
2	1.81 dB
4	3.67 dB
6	5.61 dB
8	7.69 dB
10	10.01 db

Table 3 Characteristics of the balance control

Record Output Level	Gain	R6	R4	C2
499.5 mV	7.95 dB	1k50	1k	100u
1.0 V	13.99 dB	4k02 3k9	1k	100u
1.2 V (0 VU)	15.72 dB	5k11 5k6	1k 1	100u

Table 4 Component values for the tape output buffer.

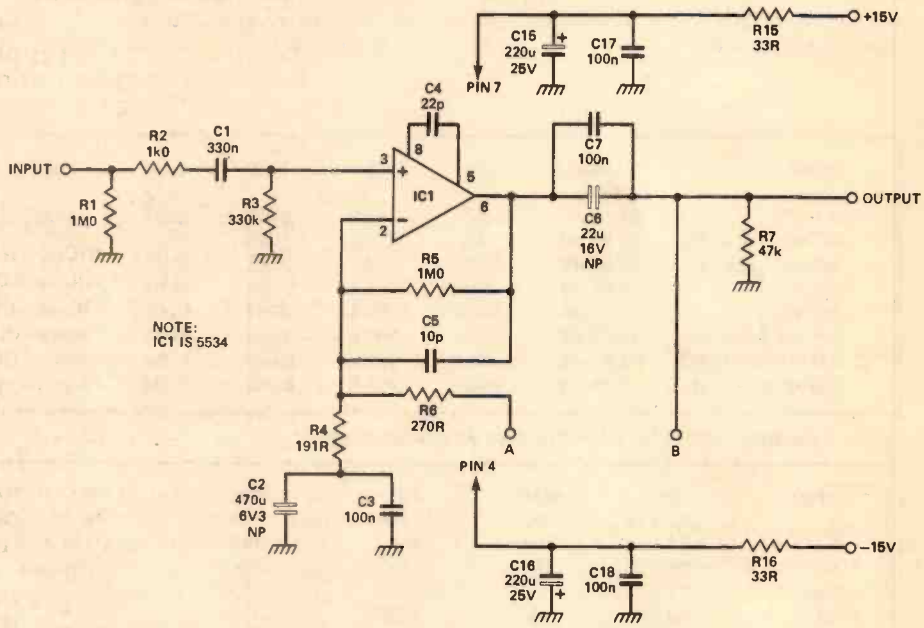

Fig. 4 Circuit diagram of the unbalanced output stage.

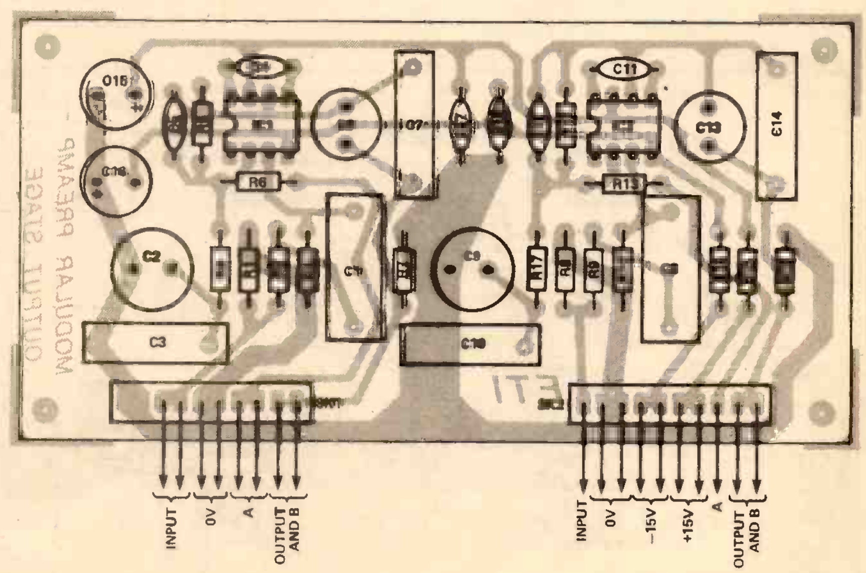

Fig. 5 Component overlay of the unbalanced output stage PCB. Note that this board is for stereo operation and therefore carries two complete output stages.

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become 1k ohm, C2 should be changed to 100μ 16V non-polarized and R6 chosen to give the required output level according to Table 4.

The importance of buffering the tape recorder output, as shown in Figure 1, must be stressed. Without the buffer stage, the recorder input would be connected directly to the main signal path of the pre-amplifier. There is nothing wrong with that, provided the recorder is switched on, but some recorder input circuitry can appear very non-linear when it is not powered, and this could introduce high levels of distortion into the pre-amplifier. The buffer stage also presents the opportunity to introduce gain into the record chain, and avoids the danger of a recorder with a low input impedance loading the output of any other piece of equipment that is connected to the pre-amplifier.

The output levels given in Table 4 should be suitable for most applications, but any other gain setting can be used, calculating R6 from:

$$R6 = 1000(G-1).$$

Muting Relay Control

Although not absolutely essential, this simple circuit will more than pay for itself the first time you switch your equipment on in the wrong sequence and it saves your speakers, eardrums or central nervous system from instant destruction. The circuit (Fig. 7) controls a relay which, in its relaxed state, shorts the main signal outputs to earth via R4 and R5. This means that the signal does not normally pass through the relay contacts, and is therefore unaffected by their presence. When power is applied to the pre-amplifier, the relay remains relaxed until the voltage on the base of Q1 has risen to 0.6V when Q1 and Q2 turn on, opening the relay contacts that are shorting the signal to earth. The 560 ohm resistors are in series with the output signal to prevent the output amplifier stages from being overstressed by the short circuit condition. Note that power to the relay is supplied from both rails, so that if one rail is not present, the output will remain muted. This eliminates a situation where, if the negative rail has failed, a 5534 will sometimes oscillate at a frequency approaching that of Heathrow Air Traffic Control, causing havoc to tweeter voice coils, bats and Boeing 747s.

The 6.8 V zener diode ZD1 will normally be connected to earth at a convenient point but, if headphone amplifiers are to be fitted, it may be earthed via the break contacts on the phones jack socket. Insertion of a jack plug will then de-energise the relay so that the main outputs are muted, and also apply power to the headphone amplifiers so that the programme is played only through the headphones. The relay is also used to provide remote switching for use with active speakers and other ancillaries.

Power Supply

This is based on IC Regulator types 7815 and 7915 (Fig. 8). Ideally, these should be mounted within the pre-amplifier with the transformer and main smoothing capacitors separately housed and placed some distance away to minimize the danger of hum pick-up. If the complete supply is contained within the pre-amplifier case, it is essential that a toroidal mains transformer is used, as the problems of screening will be considerably reduced.

Although the power supply appears simple, one or two tips may be in order. The 0.1μf capacitors, C4,5,8 and 9, should be mounted as close to the regulators as possible, and care should be taken to establish a single earth path from the transformer centre tap to the OV output. Contact suppressors should be connected across the mains switch as shown. The power 'On'

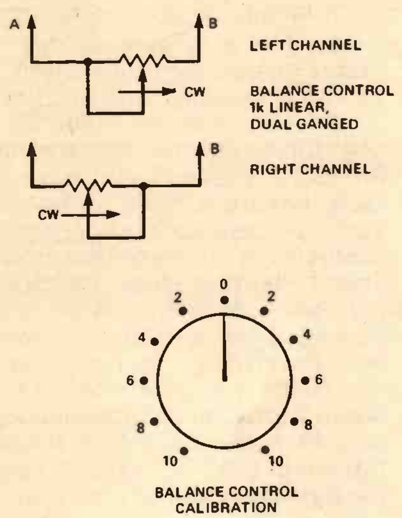


Fig. 6 Balance control connections and calibration.

PARTS LIST

Unbalanced Output Stage

Resistors (all 1% metal glaze or metal film)

R1, 5, 8, 12		1M0
R2, 9		1k0
R3, 10		330k
R4, 11	180 R	191R
R6, 13	220 R	270R
R7, 14		47k
R15, 16		33R

80

Capacitors

C1, 8	68436	330n 250V polycarbonate
C2, 9	96	470u 6V3 non-polarised electrolytic
C3, 7, 10, 14	72	100n 250V polycarbonate
C4, 11		22pf 160V 2½% polystyrene
C5, 12		10pf 160V 2½% polystyrene
C6, 13	32	22u 16V non-polarised electrolytic
C15, 16	26	220u 25V radial electrolytic
C17, 18	14	100n 100V polyester

Semiconductors

IC1, 2	95	NE5534

Miscellaneous

SK1, 2	10 way PCB sockets
PCB:	

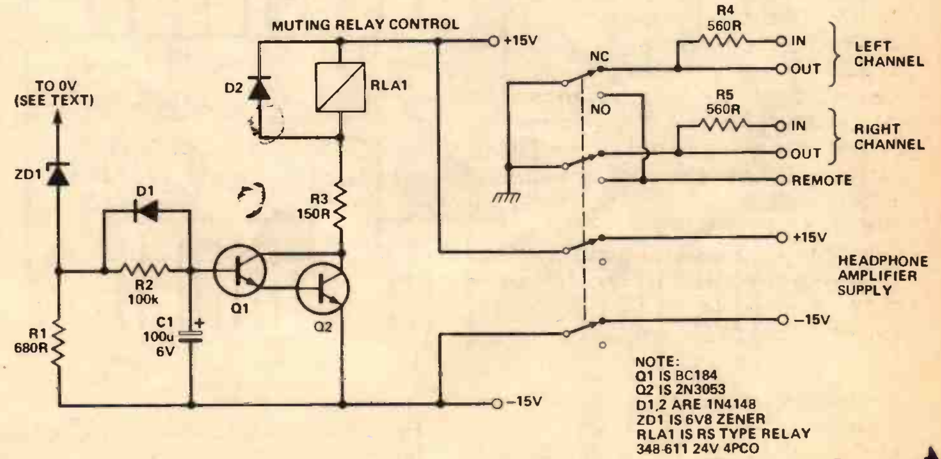


Fig. 7 Circuit diagram of the muting relay control.

LED is connected between both stabilized rails so that it acts as an indicator that all is well in the power supply department. The regulators do not require heatsinks when only the basic unit is constructed, but if headphone amplifiers are included, IC1 and IC2 should be mounted on standard T0220 finned heatsinks, or attached to the metal chassis using the insulating washers supplied with the devices.

Construction

The individual 'building block' circuit boards should be fitted with interconnecting sockets which mate with matching plugs on the mother board. The contacts of the recommended connectors are on a 0.1" pitch so that if necessary, Vero board can be used for the mother board, and indeed, for the plug-in boards as well if you do not want to go to the expense of obtaining proper printed circuit boards.

Suitable cabinets are available from such suppliers as West Hyde Developments and Maplin, who can also provide the front panel controls There are no special requirements for these except that they are of a reasonable audio grade. The input selector switch, which needs to be a 2 pole 4 position type, should be purchased as a 4 pole variety so that each contact can be doubled up for reliability. It is not necessary that the switch contacts are gold plated, providing they have a good, firm wiping action, so any build-up of deposits is removed with each operation. The potentiometers should be good quality carbon or cermet types with multi-contact wipers. If they can be obtained, the special audio controls from the larger Japanese suppliers, such as Alps, are of very high quality, and are not expensive. It has already been suggested that the best value for the level potentiometer is 10k ohms, although if this is likely to cause loading problems, 25k is acceptable. It is worth remembering that the Japanese D law is preferable to the usual logarithmic law, as it gives a much smoother control of the output. To avoid the image shifting with different settings of the level control, the two sections should be matched to within 1dB over most of their travel. The balance control should not suffer from this shortcoming, as linear potentiomenters are normally made to tighter tolerances, but as a general rule, the better quality the component, the greater is the chance of accuracy.

If lever or toggle switches are used for the tape monitor and direct functions, these should have contacts that are suitable for low level audio use, and again, suppliers such as Alps seem to have got the problem licked. Phone sockets used for signal connections should preferably be gold plated, and must be isolated from the chassis. This can prove difficult if the rear panel is thicker than about 1.0 mm, as most available sockets do not have sufficient length of threaded bush to pass through a thick panel as well as the insulating

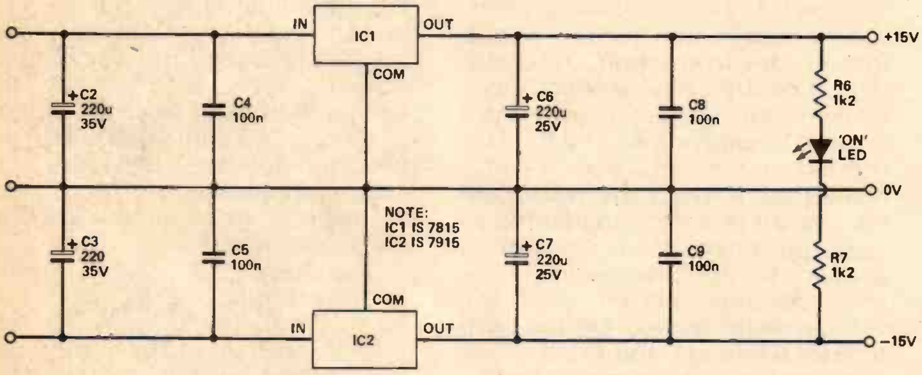


Fig 8. Circuit diagram of the power supply.

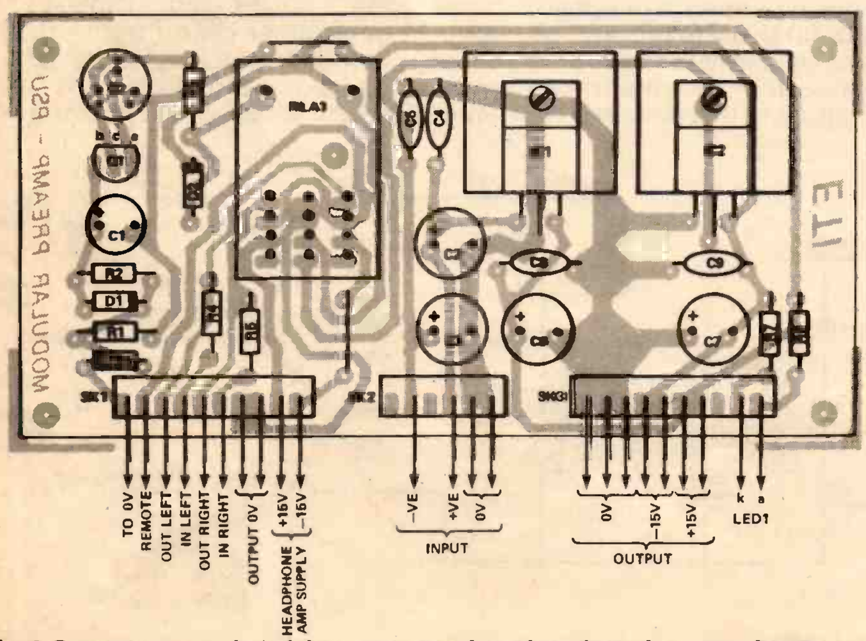


Fig. 9 Component overlay of the power supply and muting relay control PCB.

PARTS LIST

Power Supply and Muting Circuit

Resistors		
R1		680R 5%
R2		100k 5%
R3	35	390R 1W 5%
R4, 5		560R 1% metal glaze or metal film
R6, 7		1k2 5%
Capacitors		
C1	7	100u 25v radial electrolytic
C2, 3	44 ~~26~~	220u 35V radial electrolytic
C4, 5, 8, 9	28	100n 100V polyester
C6, 7	26	220u 25V radial electrolytic
Semiconductors		
IC1		7815 35
IC2		7815 35
Q1		BC184 15
Q2	30	2N3053
D1, 2		1N4148
ZD1	10	6V8 400 mW zener
		380
Miscellaneous		
SK1,3		10 way PCB sockets
SK2		6 way PCB sockets

PCB: heatsinks and nuts, bolts, etc to suite (see text); 12V 185R 4 pole changeover realy, continental series.

washers. One solution is to mount the connectors onto a piece of fibreglass circuit board material and drill clearance holes in the panel.

The internal construction is quite straightforward. Vertical guides should be mounted on the mother board to locate and support the individual circuit board, and care should be taken to ensure that the signal earth is not connected to the chassis at any point — for example, chassis mounted power supply capacitors should be checked to make certain that their cans are not earthed by their mounting clips.

Individual screened wires should be used to connect the rear panel sockets to the mother board, with twisted lenghts of 16/02 stranded wire carrying the DC supplies. If a separate power supply unit is used, a 7 pin DIN socket should be employed to connect this to the preamplifier, but be sure to use the heavy, cast type, as the common lightweight ones can easily be plugged together upside-down, with obvious consequences.

A major problem with home constructed equipment is usually caused by the earthing techniques employed. There is no secret path to success — just use the same method that manufacturers use — its called trial and error! There are one or two rules to follow, such as making the earth follow the signal and separating the signal output and power supply earths, but once the unit is working, the best earthing arrangement is usually found by ear. Figure 11 shows, in very simplified form, an earthing arrangement that was successful in the prototype. The most difficult job was to prevent the unit from being extremely sensitive to external hum fields, such as those generated by the mains transformer in a typical power amplifier. Eventually a cure was found by breaking the rule that the signal earth connects to the chassis at the input of the most sensitive circuit. For instance, the most satisfactory arrangement was to connect the earth at the output of the disc amplifier stage to chassis. This eliminated all traces of hum pick-up, but turned the preamplifier into a very good radio receiver. This was cured by connecting a 1000pf ceramic capacitor between the shell of the disc input sockets and the chassis (and in doing so, losing a very interesting CB conversation between Naughty Nora and The Cannonball!).

Once construction is complete, power should be applied to the mother board without the individual boards present. Providing all is well, the power supply regulator board should be plugged in and the busses checked to ensure that the 15-0-15 V supplies are operating correctly, which means that each rail should measure between 14.5 and 15.5V with respect to earth. The muting circuit should next be tested — correct operation being indicated by the relay operating about 5 seconds after power is applied.

The remaining circuit boards may now be plugged in and the complete signal path checked. If the unit appears to be working correctly, leave it switched on for ten to fifteen minutes, then place a finger tip on each IC in turn. None should be more than warm to the touch, including the supply regulators. If any device is hot, this is a sign that excess current is being drawn, so start a systematic search for a wrong component or assembly fault. If any of the 5534 ICs are not working, check the DC voltages on the connecting pins. Pin 7 should be at +15 V and Pin 4 at −15 V. If these are correct, Pins 2, 3 and 6 should all be approximately at 0 V, and if this is not the case, the IC may be suspect. The easiest way to proceed is to change the device for a new one, but if this does not bring about a cure, the fault is likely to be with one of the associated passive components — for example, the small value polystyrene capacitors can be prone to shorting if too much heat is applied during soldering. Should the one between Pins 5 and 8 of a 5534 become shorted, the IC will not work, and will give every indiction that it is faulty.

Once all is working, the unit should be connected to a power amplifier and speakers and the system checked for excess hum. The chances are that there will be plenty of this in evidence, so the trial and error procedure must be adopted until it is elminated. Try earthing different parts of the pre-amplifer signal earth path to the chassis with a short length of thick wire. Once the position that gives minimum hum has been found, short out the disc input sockets and check that the hum output has not increased. If it has, continue with the trial and error

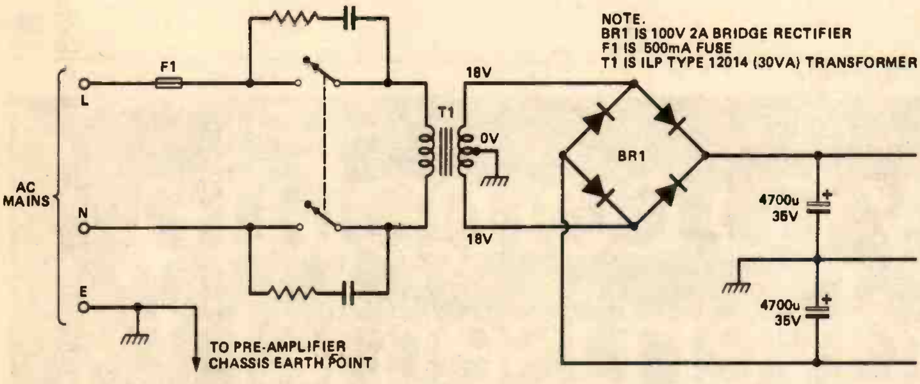


Fig. 10 Suggested circuit to feed the regulated power supply.

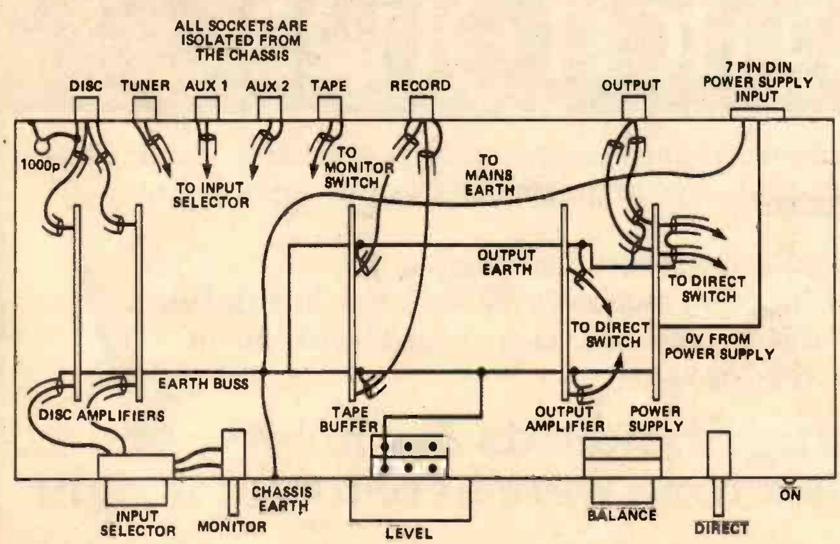


Fig. 11 A possible earthing arrangement.

exercise until optimum earthing has been achieved, then install permanent wiring where necessary.

Once it's working, what can you expect? It is nice to report that the performance of the prototype was well up to expectation. Distortion was virtually unmeasurable, being equal to the test equipment residual on the auxiliary inputs (0.0018%), and well down into the noise on the moving coil inputs. Signal to noise of the moving coil stage was better than −75dB (A weighted, 0.5 mV input, 0.5 V output). The RIAA equalisation curve was within 0.15 dB between 20 Hz and 20 kHz, and crosstalk was better than −65 dB at 20 kHz and −83 dB at 1 kHz.

Subjectively, the unit sounds clean and analytical. Hum and noise never intrude and dynamics are handled with an ease that leads the listener to believe that it will never overload; in fact it is all that a good pre-amplifier should be, in that it is the quality of the source material that decides the quality of sound coming from the speakers.

BUYLINES

The PCB plugs and sockets are available from both Maplin and Ambit, although neither carries a full range so you might have to buy from both. The relay is available from Watford Electronics. Note that no provision has been made for the use of a relay socket, so if you prefer to use one you will have to adjust the pads on the PCB. You could, of course, use any other four pole relay with a coil operating voltage of 30V or less by adjusting the layout and the value of R3. Several different types of heatsink would be suitable or if you prefer, you could make your own quite simply. Some of the 1% tolerance resistances are in the E24 range which is widely available. However most of the components for the Modular Pre-amp, including the 1% E96 resistors, the non-polarised electrolytics and the high tolerance polystyrene capacitors are available from Millhouse Electronics, 15 Thieves Bridge Road, Whatlington, Kings Lynn, Norfolk PE33 0HL. Write, or phone 0366 382165 for up-to-date prices and delivery details.

All the printed circuit boards for this project are available from our PCB Service, and they are reproduced in this issue for those constructors who like to make their own.

The second part of this article contains details of the mother board and the three remaining modules.

MODULAR PREAMPLIFIER PART TWO

Some people are not happy unless they have a mass of controls not dissimilar to Concorde's flight deck on their side-board. The circuits here can help to extend the basic design, built using the modules featured last month, to include parametric tone controls, balanced output stage and a headphone amplifier.

A block diagram of the extended preamp is shown in Fig. 1, and apart from the increased circuitry, the main difference between this and the 'basic' design is that the volume control is no longer at the input. This obviously means that consideration has to be given to the possibility of overload. In Part 1, it was shown that the disc overload margin could be maintained at an adequate 32dB if the output level of the disc input stage was 200mV for its rated input. In order that this margin is not reduced, no gain may be introduced before the volume control, and the gain necessary to give a 1.0V output should be made up by the output stage.

The input switching and tape buffer remain the same as in the simpler unit, but after the monitor switch, a unity gain buffer amplifier presents a high input impedance to the line inputs, and provides a low impedance drive to the tone control circuitry. The attenuator on the compact disc input is not to avoid overload, as the input and tone control circuits will easily handle the output of any CD player. Its purpose is to limit the signal level so that the volume control is not operated at a setting where its inter-track balance is not likely to be better than 3 or 4dB.

Tone Controls

Although fully paid-up members of the Flat Earth Society would have us believe that any form of signal processing is guaranteed to make a pig's ear of the emotional experience of listening to a group of musical morons twanging guitars and wailing in voices more suited to Billingsgate than Covent Garden, it should be remembered that most recordings are subject to considerable amounts of 'equalisation', usually to satisfy the producer's requirement for a particular type of sound. No allowance is made for the introduction of random phase shifts, or that the intricate

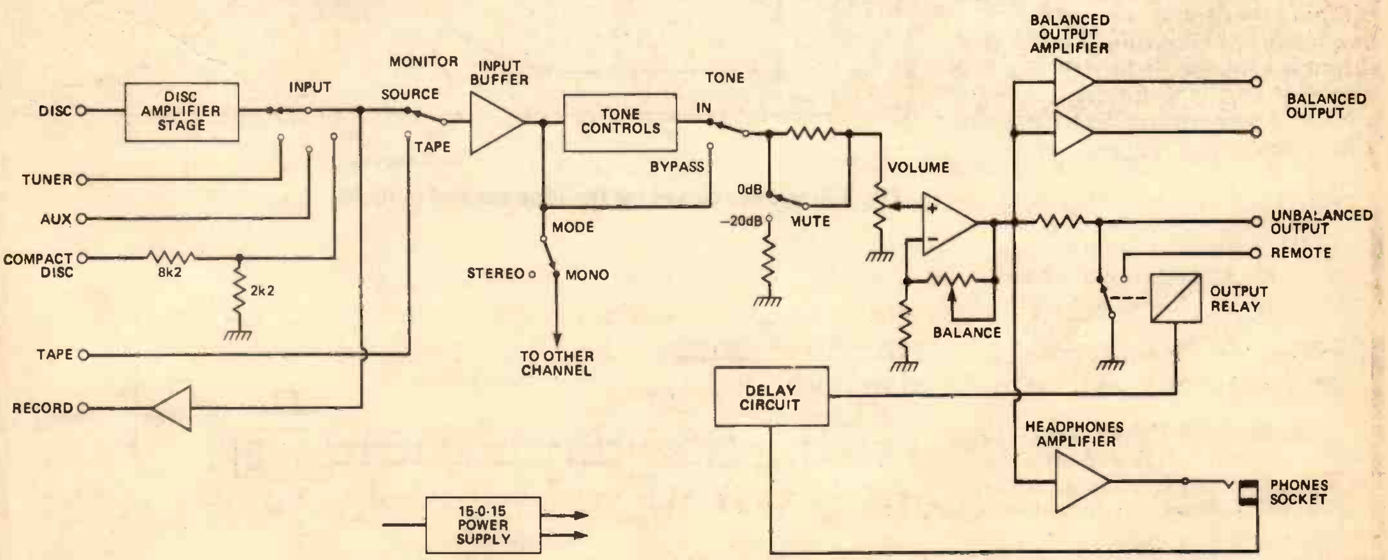


Fig. 1 Block diagram of the extended preamp.

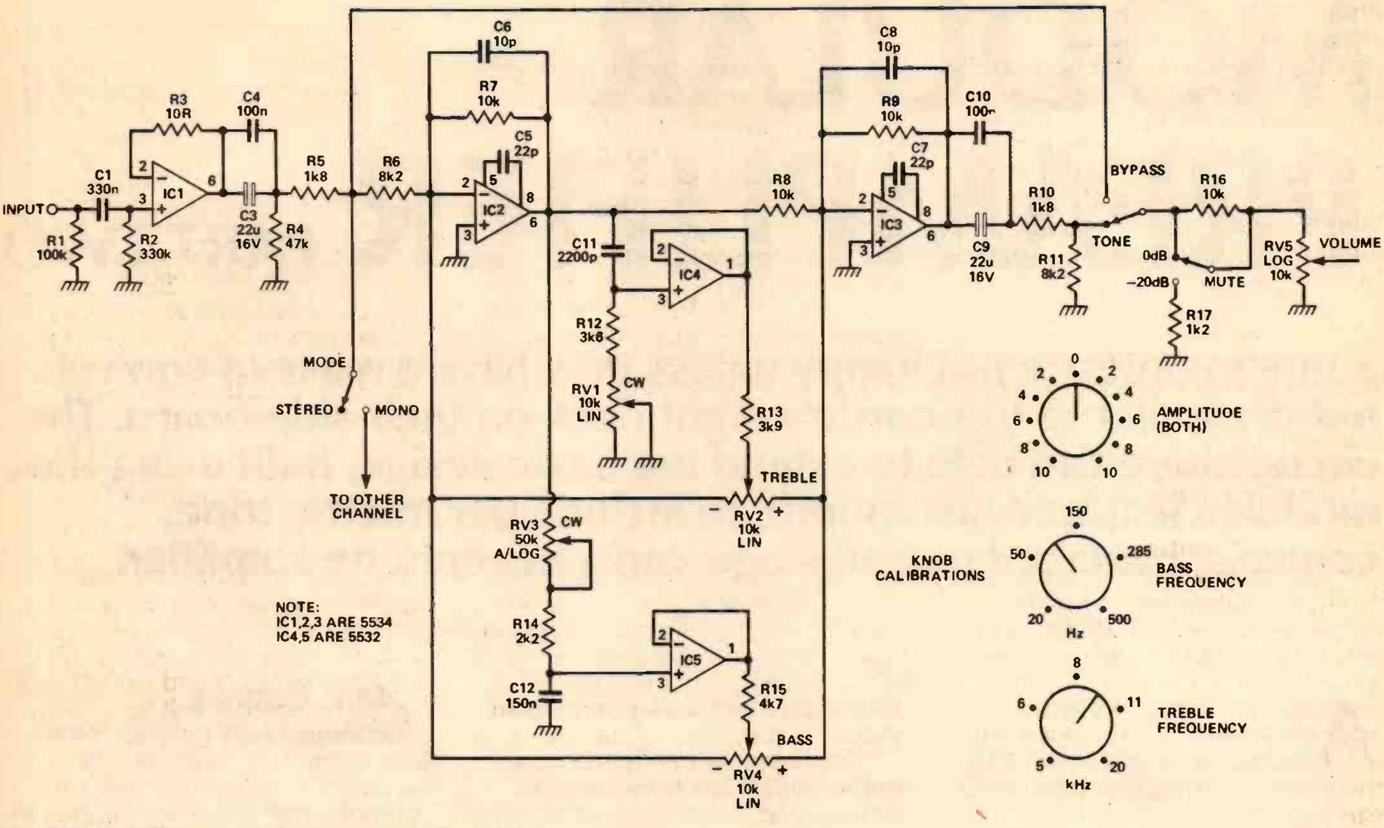


Fig. 2 Circuit diagram of the tone control module.

relationship of harmonics is sent on a one way trip to the cleaners. The object is to change the sound to make it more satisfactory, and if tone controls are used in the replay process for exactly the same purpose, surely no-one has the right to complain?

The type of tone control fitted to most hi-fi equipment is far from ideal, usually being much too dramatic in operation — for example, if it is required to lift frequencies below about 100 Hz, the effect is usually to lift, by varying amounts, everything up to at least 1 kHz, and even higher.

The circuit shown in Fig. 2 is somewhat more sophisticated than usual, possessing in addition to the normal lift and cut controls, adjustment of the turnover frequencies of the two sections.

Operation of the circuit is quite straightforward. IC1 acts as an input buffer, presenting an input impedance of approximately 100k ohms to the line inputs of the preamplifier. The input of IC1 is AC coupled by C1, which together with R2 fixes the −3dB point at

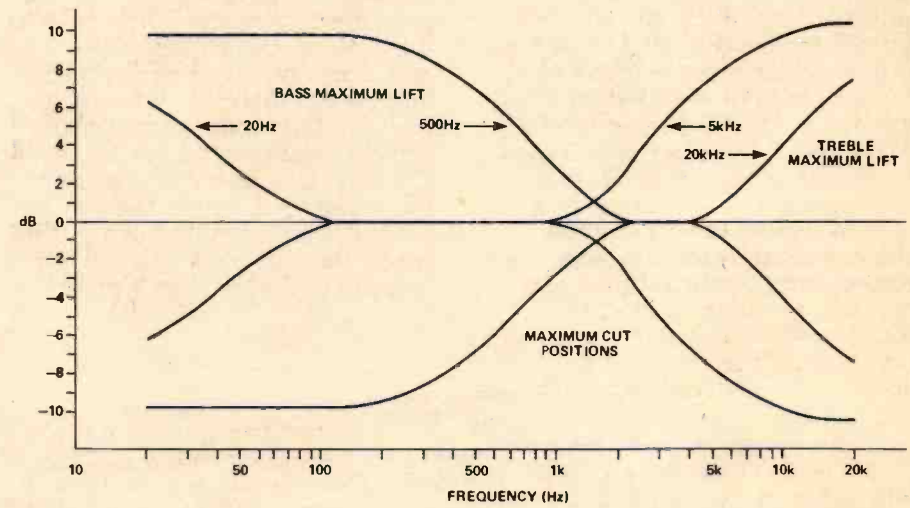


Fig. 3 Response curves for the tone control module.

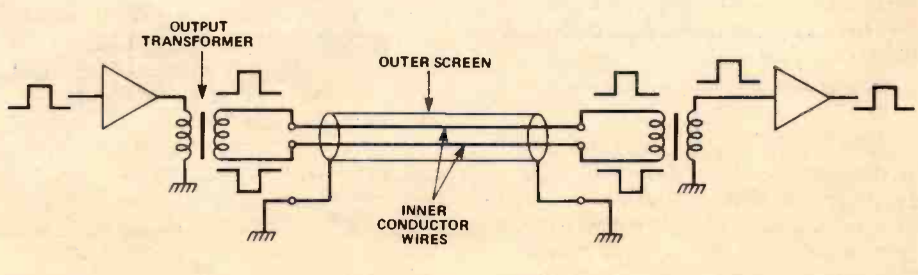


Fig. 4 The basic principle of balanced line operaton.

about 1.5 Hz — low enough to prevent objectionable low frequency phase shift. The output of IC1 drives the pair of inverting stages formed by IC2 and IC3, the input resistor to IC2 being split to allow mono summing of the two channels. The signal path is maintained at unity gain by the equal input and feedback resistors of the two stages.

The output of IC2 feeds two single-pole filters which are buffered by IC4 and IC5. The filter formed by C11, R12 and RV1 has a high pass characteristic with its −3dB point adjustable by VR1 from 5.3 kHz to 20 kHz. Operation of the treble control, RV2, decides the destination of the high frequencies that emerge from the output of IC4 — in the "cut" position they are applied as negative feedback to IC2, and in the "lift" position they bypass R8 giving additional gain to IC3. The amount of lift and cut is controlled by R13, the value specified giving a ±10dB variation.

The bass control works in the same way, except that a low pass filter comprising C12, R14 and RV3 selects the low frequency range which is variable between 20 Hz and 480 Hz.

This type of tone control is characterised by shelving response curves with no interaction between the bass and treble sections. The curves are shown in Fig. 3 which illustrates the range of the variable frequency controls.

As the tone control section is non-inverting from input to output, it can readily be bypassed as shown. To ensure that there is no change in level when the bypass switch is operated, the 2.9dB attenuator formed by R5, R6 and the volume control is duplicated by the addition of R10 and R11 at the output of IC3.

The mute switch has been added, as much for convenience as anything. When changing records or when the 'phone rings it is very useful to be able to reduce the overall gain without disturbing the volume control setting.

Output Stage

The volume control is followed by the unbalanced output stage, already described in part 1. However, in order to give an additional 2.9dB of gain (to counteract the loss in the tone control circuit) the values of R4 and R6 should be changed to 133R and 300R respectively. The balance control characteristics are very slightly changed by this, as shown in Table 1, but they will still remain accurate to the calibrations.

In normal circumstances, the unbalanced output stage is all that will be needed. The main advantage of using a balanced output stage is realised when the signal output from the preamp has to run more than a metre or two, where the balanced output will give a much better noise immunity, or in an intrinsically noisy situation, eg disco systems, where the lighting controller can interfere with the audio signal.

As far as the user is concerned, the only difference between balanced and unbalanced lines is that balanced ones use a three-wire connection per channel instead of the usual two. Balancing has been standard with professional audio equipment since the days of 2LO and the cat's whisker (and Angela Rippon?), where it is used to ensure reliable, hum-free connections over long distances.

The basic principle is shown in Fig. 4; the signal is carried along two wires with the outer screen acting as an earth connection. The signal is inverted in one wire with respect to the other, but any hum and noise picked up from external signal sources by the cable will have the same phase in both wires. The balanced input accepts the differential signal, but rejects the externally introduced common mode signal. The normal method of balancing has traditionally relied upon the use of special input and output transformers, but similar results can be obtained by applying standard operational amplifier techniques — after all, the common op-amp has differential inputs.

Control Callibration	Channel Imbalance
2	1.87dB
4	3.78dB
6	5.79dB
8	7.96dB
10	10.39dB

Table 1 Balance control performance when used with tone control.

A balanced output consists of two outputs of identical levels but with one signal phase-inverted with respect to the other. Fig. 5 shows three ways that this could be achieved, and any one of these methods would be quite adequate provided it was connected to a balanced input. It may be seen that in each case, attenuation is introduced to ensure that half the input voltage appears at each output, but with opposite polarities. Thus a 1 volt input would result in outputs of +0.5 V and −0.5 V, giving the required 1 volt between conductors.

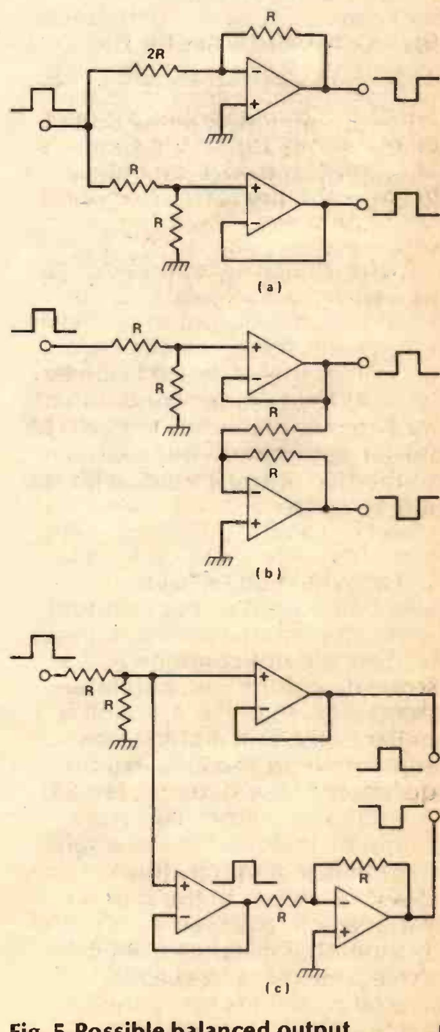


Fig. 5 Possible balanced output arrangements.

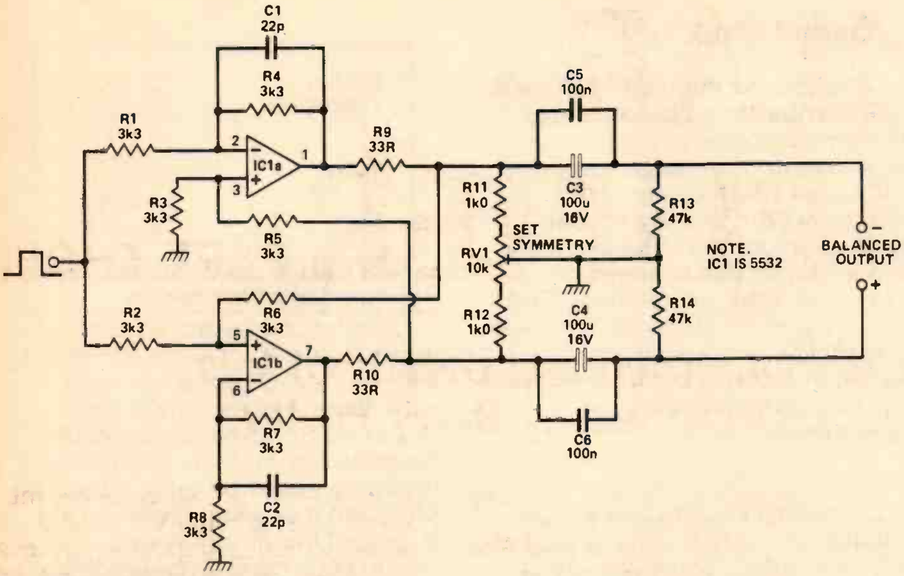


Fig. 6 Circuit diagram of the balanced output stage.

This is fine until the output is connected to an unbalanced input, when one of the balanced feeds gets shorted to earth, leaving a single output of 0.5 V. A difference of 6dB appears between balanced and unbalanced operation, which is not really acceptable. What is required is a method of increasing the gain of either output amplifier by 6dB whenever the other output is grounded, and such a circuit is illustrated in Fig. 6.

Here, the cross feed-back fixes the gain of each side at 0.5, but shorting either output to earth will remove this feedback, increasing the gain of the unshorted side to unity. As the balance condition of the circuit is extremely critical, the pre-set potentiomenter is used to set the two outputs equal with respect to earth.

The whole object of the balanced output is that it should be connected to a balanced input. As these are not common in domestic equipment, a suitable circuit is given in Fig. 7, which is similar to the line input stages common in professional studio equipment (alternatively, see ETI May 1983 — Editor). This stage should be included in any equipment that is driven by the balanced output of the preamplifier while placed a considerable distance from it — active speakers and speaker-located amplifiers being the obvious cases in domestic systems.

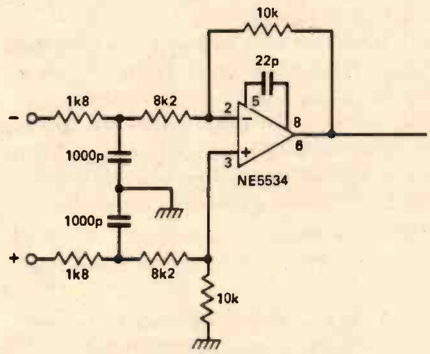


Fig. 7 A possible balanced input arrangement.

As shown in Fig. 1, the unbalanced output from the balanced line output stage is taken to the rear panel via the delay relay. This output will therefore be shorted to earth for about five seconds when power is first applied, and will also be cut off whenever a pair of headphones is plugged into the front panel-mounted jack socket.

Headphone Amplifier

Although most moving coil headphones have impedances in the 150 to 600 ohms range, the odd maverick pair are around that are as low as 8 ohms, so to be universal a headphone amplifier needs to be able to supply somewhat more current than a normal op-amp can manage. Various power ICs have been tried, but they all draw rather high quiescent current, causing the power supply regulators to get a bit too warm for comfort.

The circuit given in Fig. 8 uses an NE5534 to drive a pair of complementary transistors which are turned on by the voltage drop across R8. When the amplifier is in its quiescent state, the output stage is turned off, so minimum standing current is drawn. The transition point of the output devices does introduce some cross-over distortion, but this is kept within reasonable limits by negative feedback action, and the performance is more than adequate for headphone listening. The output resistor, R11, may be changed to suit the type of headphones in use, possibly requiring an increase to about 200 ohms with some medium impedance models.

Constructional details, overlay diagrams, etc for these modules will be given next, along with further information on the mother board.

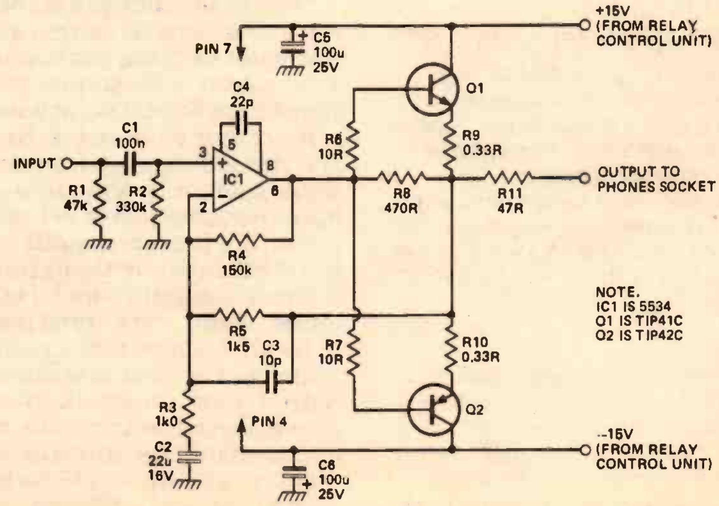


Fig. 8 Circuit diagram of the headphone amplifier module.

MODULAR PREAMPLIFIER PART THREE

In this final part, we give constructional details of this expandable audio project. Designed by Barry Porter.

As with the smaller preamplifier described in December, assembly is based on the use of a mother board with the individual modules plugged into mating connectors. The pins for these are on a 0.1" pitch, so it is quite acceptable to use a length of veroboard to carry the interconnection busses between modules.

Details of the disc amplifier, muting relay control and power supply were given in part 1, and will not be repeated here. If it is required that insertion of the headphone jack plug should cause the output relay to cut off the unbalanced output, the 6V8 zener diode in the delay circuit should be connected to earth via the common contact switch on the jack socket as shown in Fig. 10. So that the headphone amplifiers are not powered when they are not in use, their supply voltages should be obtained from the switched rails of the delay relay.

BUYLINES

Most of the components for the Modular Preamplifier, including the 1% EG6 resistors, the non polarised electrolytics and the high tolerance polystyrene capacitors, are available from Millhouse Electronics, 15 Thieves Bridge Road, Whatlington, Kings Lynn, Norfolk PE33 0HL. Write or phone 0366 382165 for up-to-date prices and delivery details.

All Modular Pre-amp printed circuit boards are available from our PCB Service, and they are reproduced in this issue for those constructors who like to make their own.

Other constructional comments in part 1 may be applied to this larger unit, which may be built into one of the standard rack-sized cabinets obtainable from a number of suppliers.

Once the preamplifier is working (again, see part 1) the output balance pre-sets must be adjusted to give equal voltages from the two outputs. The easiest way to do this is to temporarily connect two equal value, close tolerance resistors in series with the output and adjust the respective pre-set for zero volts at their junction when a 1 kHz signal is applied. (Fig. 11).

In use, the performance of this pre-amplifier is virtually identical to the more basic unit described in part 1. With the tone controls switched into circuit the noise increase is only about 1 dB with negligible additional distortion. The limited amount of control has caused no problems — in practice, if more than 10dB of lift or cut is required, it's not hi-fi you've got but a potential advertising copy for exchange and mart!

The situation that displays the advantages of the tone control most is when small bass-light loudspeakers are being used. Applying a limited amount of bass lift, with the frequency control set to about 50 Hz, will usually make it possible to increase the speakers' bass extension without encountering overload problems — something that is impossible to do when the turn-over frequency is fixed.

Although not detailed here, the individual 'building blocks' method of construction lends itself to a number of possibilities — for example, it is quite easy to modify the tape connections to allow for two recorders with cross dubbing, even providing balanced record outputs if required. A further enhancement could be to include record level controls on the front panel, with suitable VU or LED monitors displaying the signal level being sent to the recorder. Indeed, with a little thought, that Concorde flight-deck look might not be too far away . . . .

Some Changes

There have been some relatively minor changes between the circuit diagrams published in the previous parts and the PCB layouts printed here. They are:

on the tone control module:
C2 was left off the circuit diagram in error; this is a compensation capacitor for IC1 (as C5 is for IC2) and is included on the overlay;
C13 and C14 have been added in the leads to the wipers of RV2 and RV4; these are to prevent any offset voltages being passed around and amplified;
IC4 and IC5 have been combined into a single dual op-amp rather than two single op-amps;
on the balanced output stage:
IC1a and b have been interchanged;
on the headphone amplifier:
the input filtering to IC1 has been changed and the values of R1 and R2 are different; however, the PCB allows the original circuit to be used if desired;
on all modules:
supply line decoupling capacitors have been added; these were not shown on the circuit diagram last month (except for the headphone amplifier, where unpolarised capacitors have been added in parallel with the existing electrolytics).

Note also that the tone control stage is split over three boards for stereo operation. Unfortunately, it wouldn't quite fit onto two, so it was decided to split off the filter sections so that at some future date, constructors could alter

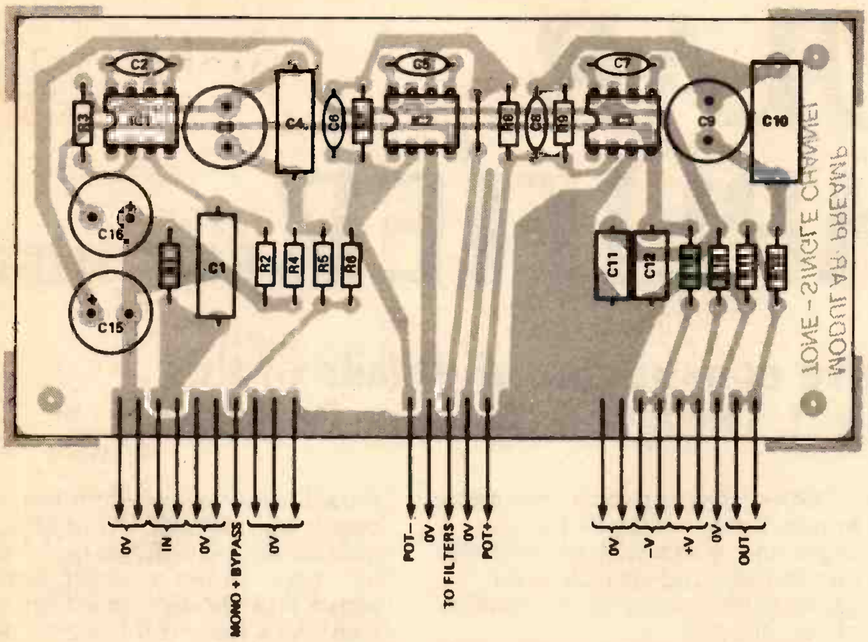
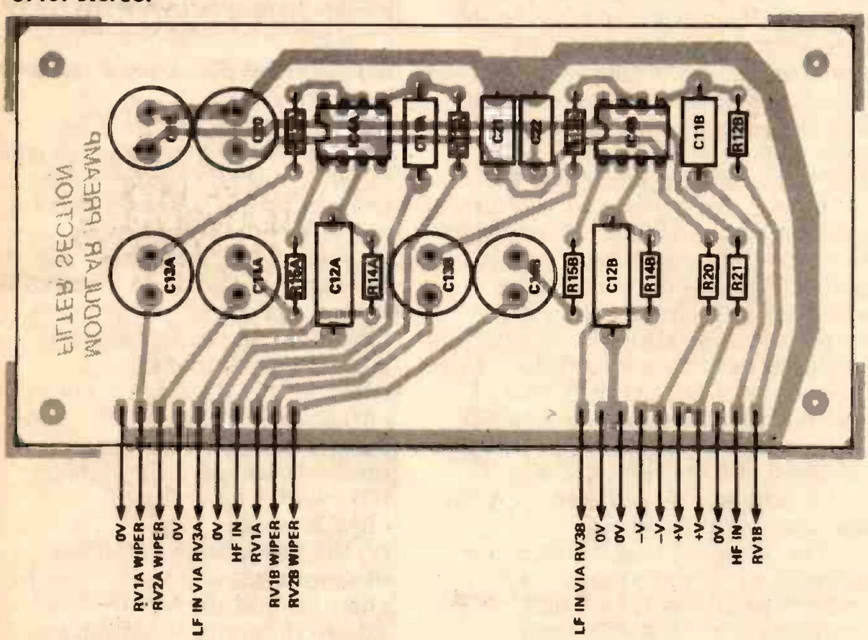


Fig. 1 PCB overlays for the tone control: the main board above is for a single channel, so two of these are required, whilst the filter board (below) is a stereo board, so only one of these is required. Note carefully which parts you need two of for stereo.



PARTS LIST — TONE MODULE

RESISTORS

R1*	100k
R2*	330k
R3*	10R
R4*	47k
R5*,10*	1k8
R6*,11*	8k2
R7*,8*,9*,16*	10k
R12*	3k6
R13*	3k9
R14*	2k2
R15*	4k7
R17*	1k2
R18-21	33R
RV1**,2**,4**	10k lin
RV3**	50k anti-log
RV5**	10k log

CAPACITORS

C1*	330n 250V Mullard polyester
C2*,5*,7*	22p 2½% polystyrene
C3*,9*,13*,14*	22µ16V PCB non-polarised electrolytic
C4*,10*	100n 250V Mullard polycarbonate
C6*,C8*,C11*	10p 2½% polystyrene
C12*	150n Mullard polycarbonate
C15*,16*,19,20	220µ 25V PCB electrolytic
C17*,18*,21,22	100n polyester

SEMICONDUCTORS

IC1*,2*,3*	NE5534
IC4*	NE5532

MISCELLANEOUS
PCBs: 2 off tone, 1 off filter; edge connectors: 6 off 10 way, 2 off six way

* R1 to 17, C1 to 18 and IC1 to 4 are required in both channels so two of each are required for stereo.
** RV1 to 5 could be stereo potentiometers or two single potentiometers each for stereo, as required.

these, for example, to include a 'mid' control.

Swings And Roundabouts

It is possible to modify the component values of the **disc amplifier** and of the **unbalanced output** stage so that it is not necessary to use E96 series resistors. This will actually give a less accurate response technically (degrading the error on the RIAA characteristic to 0.3dB), but for most people this will not be that noticeable (if it is noticed at all!)

For the disc amplifier, the modified component values are as follows:

R2	4k7
R3	6k8
R6	12k
R10	7k5
R11	560R or 1k0*
R12	39R or 82R*
R13	6k8 or 8k2*
R14	3k3 or 1k5*
R15	82k
R16	15k//330k (T3 = 3179.5µs)
C7	10n
C11	33n

* For R11, 12, 13, 14 the first figure given is for moving coil cartridges and the second is for moving magnet.

The value of R9 that should be used will depend on the required sensitivity of the input stage; for **moving coil** cartridges, the appropriate values are as follows:

R9 Value	Sensitivity
10k	0.11 mV
5k6	0.2 mV
3k3	0.33 mV
2k2	0.49 mV

For moving magnet, the following values are appropriate:

R9 Value	Sensitivity
820R	2.17 mV
560R	3.03 mV
270R	5.41 mV
150R	8.0 mV

Additionally, IC1 (LM394) can be replaced with a parallel pair of

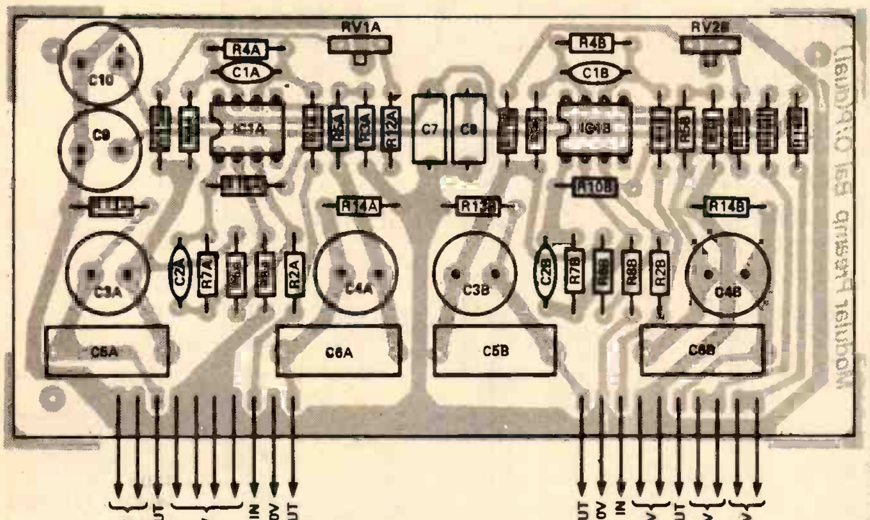


Fig. 2 Overlay of the PCB for the balanced output driver — this is a stereo board, so only one is required, but note the components that you have to obtain two of.

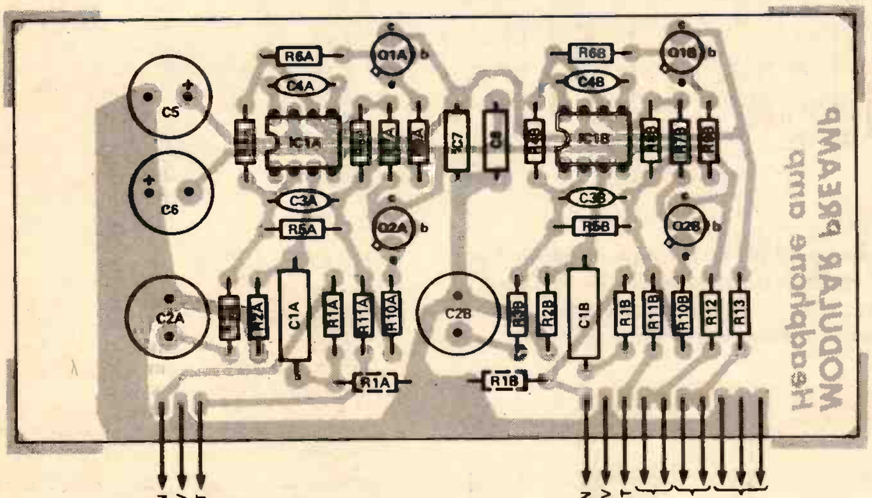


Fig. 3 Overlay diagram for the headphone amplifier; again, this is a stereo board, and again, you will have to sort out which components you need two of.

2SD786 transistors, which are somewhat cheaper than the LM394.

For the unbalanced output stage, the modified component values will depend on whether it is to be used as a tape output buffer or as an output stage to feed the power amplifier. For use as a tape output buffer, the values shown in Table 1 apply.

Record Output Level	Gain	R6	R4	C2
499.5 mV	7.95 dB	1k5	1k0	100µ
976.9 mV	13.77 dB	3k9	1k0	100µ
1.2 V (0 VU)	15.65 dB	5k6	1k1	100µ

Table 1 Revised component values for the tape output buffer.

PARTS LIST — BALANCED OUTPUT MODULE

RESISTORS	
R1*,2*,3*,4*,5*,6*,7*,8*	3k3 1%
R9*,10*	33R 1%
R11*,12*	1k0
R13*,14*	47k
R15,16	33R
RV1*	10k min vertical preset
CAPACITORS	
C1*,2*	22p polystyrene
C3*,C4	100µ16V PCB non-polarised electrolytic
C5*,6*	100n Mullard polycarbonate
C7,8	100n 250V Siemens polyester
C9,10	220µ 250V PCB electrolytic
SEMICONDUCTORS	
IC1*	NE5532

MISCELLANEOUS
PCB; edge connectors: 2 off 10 way

* R1-14, RV1, C1-6 and IC1 are required in both channels, so two of each of these components are needed for stereo.

PARTS LIST — HEADPHONE AMPLIFIER

RESISTORS	
R1*	47k
R2*	330k
R3*	1k0
R4*	150k
R5*	1k5
R6*,7*	10R
R8*	470R
R9*,10*	4R7
R11*	47R
R12,13	33R
CAPACITORS	
C1*	100n 250V polyester
C2*	22µ 16V PCB non-polarised electrolytic
C3*	10p polystyrene
C4*	22p polystyrene
C5,6	100µ 25V PCB electrolytic
C7,8	100n polyester
SEMICONDUCTORS	
IC1*	NE5534
Q1*	BC411 or similar NPN
Q2*	BC461 or similar PNP

* R1-11, C1-4, IC1 and Q1,2 are required in both channels so two of each are required for stereo.

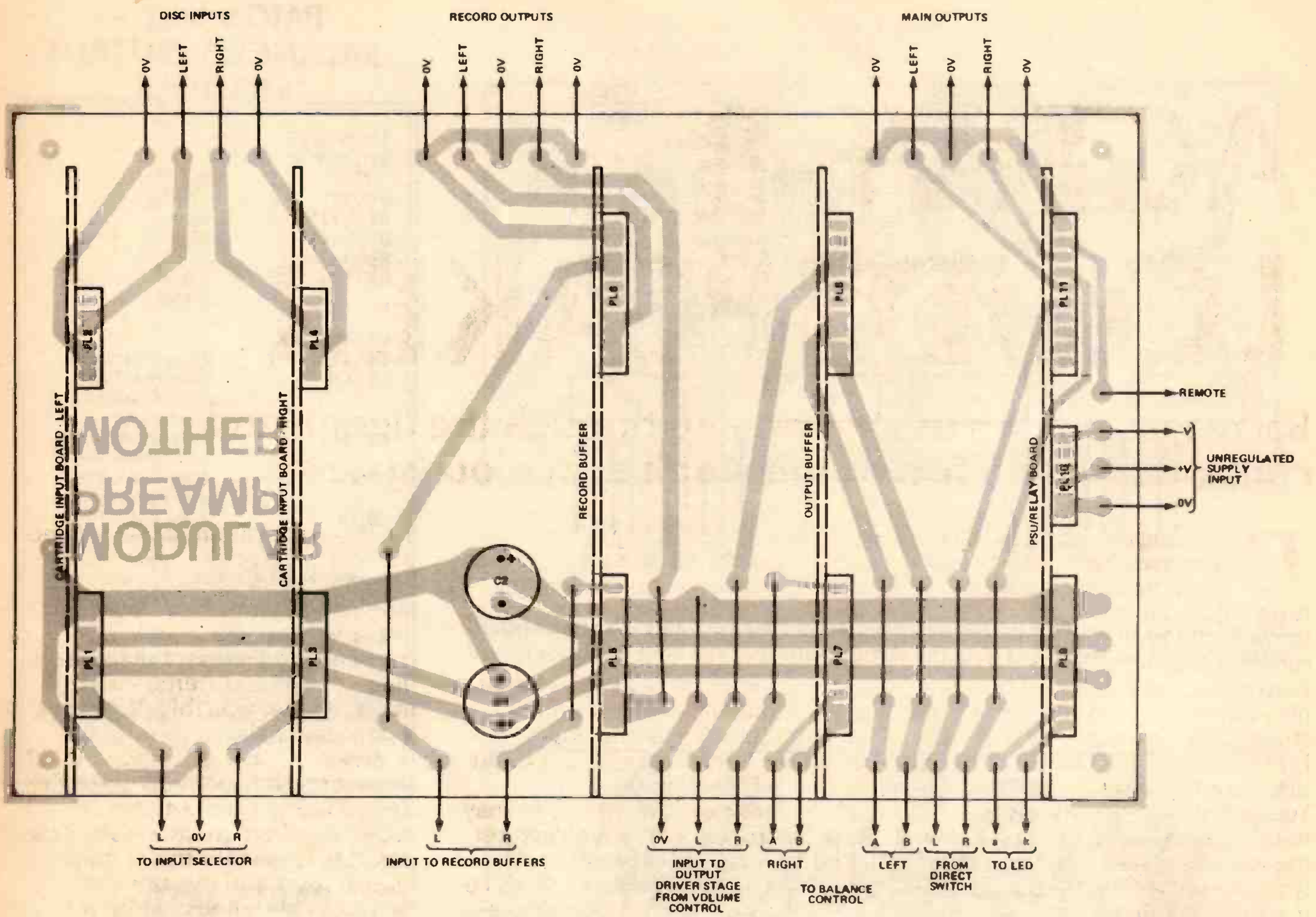


Fig. 4 Here is the overlay diagram for the mother board of the preamp as featured in the first part of the description. We have not reproduced a lay-out for the mother board of the extended system because the whole idea is for it to be adaptable to your needs — so everyone can make up their own, customised preamp using the same basic blocks.

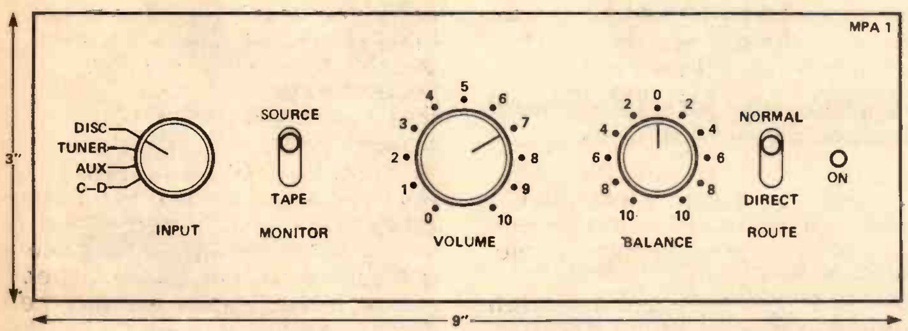


Fig. 5 A suggested front-panel lay-out for the small preamplifier.

Control Callibration	Imbalance
2	1.94dB
4	3.93dB
6	6.02dB
8	8.29dB
10	10.88dB

Table 2 Characteristics of the balance control.

For use as an output stage, R4 can be 180R and R6 can be 220R. The revised balance control characteristics are shown in Table 2.

After all that, all that remains for us to do is to wish you happy listening!

PARTS LIST — MOTHER BOARD

CAPACITORS

C1,2	220μ 25V electrolytic

MISCELLANEOUS

PL1,3	8-way edge plug
PL2,4,10	6-way edge plug
PL5-9,11	10-way edge plug

ADDITIONAL PARTS REQUIRED TO MAKE FULL PREAMPLIFIER (EXC. UNREG SUPPLY)

Input selector switch: 4-way (or to suit) 2-pole
Tape/source switch: 2-pole 2 way
Volume control: 10k log stereo (but see 'Construction', Part 1
Balance control: 1k0 lin stereo
Direct switch; 2-pole 2-way
Output resistor: to suit (100R suggested)
Indicator LED: to suit
Connectors for disc, aux, tape, tuner (as required) and output (also PSU); knobs, case, etc as required

ACTIVE-8 LOUDSPEAKER

Barry Porter takes us step-by-step through the design and construction of a two or three unit active loudspeaker.

Designing and constructing your own high quality audio equipment can be a very rewarding pastime, with some items in the reproduction chain representing a greater challenge than others. Loudspeaker building may appear to be quite simple and straightforward, but in practice this is not the case. The biggest problem confronting the Do-It-Yourself speaker builder is the need to take frequency response measurements. Audio manufacturers invest many thousands of pounds (or at least, claim they do!) in sophisticated measuring equipments, calibrated microphones, anechoic chambers and computer controlled analysers, and those that do not have their own test facility will spend many long hours in a hired laboratory during the design of a new speaker.

Obviously, the home constructor cannot hope to compete on equal terms with this, so what can be done when you are overcome with enthusiasm and the desire to create something that will justify your impulsive purchase of a complete Black and Decker outfit in 1976? If you are sane, you will take up fishing, so this is dedicated to non-angling, audiophile lunatics everwhere.....

The Active-8 has been designed as an active system, and no consideration has been given to producing a passive version. Throughout the following, sufficient details are given to allow the suggested dimensions to be modified or different drive units to be used. The less energetic may apply the principles to activating some existing speakers, but don't complain if the resulting guarantee invalidation brings on temporary insomnia or hot flushes. The design uses two drive units so that each speaker can be driven by a stereo power amplifier, but details of a tri-amplified version are also given.

It has long been accepted that active loudspeakers have many advantages over their passive brethren, some of which are listed here:
(a) electronic crossover filters may be constructed with much greater accuracy than passive networks, and may be configured to produce amplitude and phase characteristics that are often impossible to implement with passive filters;
(b) high level distortion is likely to be lower, as there are no inductors to drive into saturation;
(c) the direct coupling of amplifier outputs to drive units maintains maximum damping, thereby reducing unwanted resonance to a minimum;
(d) amplifier overload effects are greatly reduced because low frequency clipping is only reproduced by the bass unit, and often passes unnoticed;
(e) differences in drive unit sensitivities can be allowed for without introducing attenuation between the amplifier and driver simply by adjusting the gain in the signal path;
(f) low frequency equalisation can be introduced to extend the response, giving bass output equivalent to that of a larger speaker;
(g) time delay can be used to compensate for the positioning of the acoustic centres of the drive units in different vertical planes, thus preventing a directivity shift in the crossover region.

There are other advantages that are less easily defined, but subjectively, a good active system appears to handle wide dynamic range material with an ease that is not apparent with a similar, passive unit. Transient response is much better and stereo imaging more precise, possibly due to the lack of crosstalk.

Bearing in mind their potential superiority, it is perhaps surprising that so few good examples of active speakers are available. One possible reason for this is that loudspeaker and electronics designers are, almost without exception, totally separate breeds of animal. Few speaker designers are at home with present day filter and amplifier technology, whereas to most electronics designers, a loudspeaker is the result of a fair amount of mumbo-jumbo and an intravenous injection of BAF wadding. At a commercial level, loudspeaker manufacturers tend to be wary of anything that plugs into the mains as they are convinced that this is likely to bring about the instant destruction of their handiwork, and amplifier manufacturers, who are often "Cottage Industry" based, dare not think about the additional real estate required for the storage of lots of wooden boxes or the price of installing an anechoic chamber.

The few active speakers on the market that are both electronically and acoustically well engineered are invariably expensive, although there are examples around that would be better utilised by removing the drive units and turning the cabinets into condominiums for gerbils.

Before deciding to "Go Active", you may wonder if it is going to be worth the expenditure of energy, grazed knuckles and sawdust on the Axminster. The answer, from one who has been active for the past ten years or so, is a resounding YES, so brush up your 'O' level woodwork, comandeer the dining room table for a couple of weeks, and prepare yourself for the forthcoming revelation....

The Active-8

The design procedure of any loudspeaker may be divided into a number of distinct stages. In brief these are:

(a) decide cabinet size, drive units, bass loading etc;

(b) build prototype cabinet and take frequency response measurements of drive units mounted in place (No crossover network is involved at this stage);

(c) plot desired response of each unit, and by deducting this from the previous measurement, establish the required crossover network response;

(d) design the crossover filters and unit equalisation to be as near as possible to the target response established at (c);

(e) measure the complete system and correct any equalisation errors to achieve an output that is as flat as possible over the audio band;

(f) listen to lots of music — if subjective performance is below par, return to (a) (bit like snakes and ladders, isn't it?);

(g) when satisfied, invite all your friends along for a quick listen before your enthusiasm brings on acute turning of the volume knob, leading to terminal overdrive of one or more of the units.

Obviously, the steps that require response measurements are the most difficult for the home constructor, so for the Active-8 these have been done for you. If you decide to use drive units other than those recommended you have a problem, although some unit manufacturers are quite helpful at supplying anechoic response curves of their products in different sizes of enclosure. These can be reasonably accurate for bass units, but high frequency units should really be measured while fitted to a baffle of the right size as diffraction caused by the cabinet extremities can have a marked effect on the response. If you are activating an existing speaker, a good indication of the crossover response can be obtained by applying a 20 Hz-20 kHz sine wave to the speaker input and plotting the drive unit terminal voltages. This assumes that the overall response is acceptable in the passive mode, as any shortcomings will be repeated in the active network unless accurate acoustic measurements can be made.

Drive Unit Choice

Being a two unit design limits the bass driver diameter to 200mm, as anything larger would be distinctly unhappy operating up to the 2.5 — 3kHz region which is necessary to avoid overloading the high frequency unit.

Several low frequency units were considered, and four were selected for detailed examination and testing, these being from Peerless, Kef, Seas and Volt. The Peerless and Volt units were rejected for various technical reasons, leaving the Kef B200G and the polypropylene coned Seas PZ1 REX as main contenders for the job, with very little to choose between them.

Various high frequency units were tried, with the Kef T33A and Skanspeak D2008 coming out on top. The Kef T52 was not far behind, being preferred for its performance in the 2.5 — 5kHz region, but falling down at higher frequencies. In order to make the final choice cost and availability were entered into the equation, and the final design is based on the Kef B200G and T33A.

This all sounds quite simple, but of course the various combinations of bass and high frequency drivers all had to be mounted into cabinets, crossovers had to be designed and built and measurements made. To avoid littering up the love-nest with dozens of cabinets, a single pair were used, and the front baffles were duplicated with the necessary mounting holes for each pair of drivers. This meant that A-B comparisons could only be carried out between single combinations of units, but after a great deal of midnight oil had been burned, it was clear that the Kef units offered the best overall performance, although the Seas — Skanspeak combination handled transients with somewhat greater clarity. If you decided to use drive units of your own choice, make sure that you can obtain the necessary technical data for them. For the bass unit you will require the following parameters: free air resonance (f_S), driver Q (Q_{TS}) and suspension compliance (V_{AS}). For both units, you will require frequency response curves derived from anechoic or free-field measurements.

Bass Loading

A great deal of consideration was given to the type of bass alignment employed, resulting in what we in the trade call a "sonic breakthrough" which is what the rest of humanity recognises as a compromise that avoids having to make a difficult decision. The Active-8 has been designed as a reflex system, but with provision to blank off the tuning vent, plug in a circuit board and turn it into a closed box with active correction of the low frequency response.

The Active-8 in its reflex guise is happiest when used in a room of 60-100m³. In a room smaller than 60m³, the vent should be blanked off so that the extended bass is not overemphasised by the additive effect of room reflections. If you are fortunate enough to have a living room of more than 100m³, the equalised closed box will probably be preferable, but the final decision should be made after extended listening periods.

Cabinet Size

The B200G data sheet reveals the following information:

f_S = 27Hz
Q_{TS} = 0.37
V_{AS} = 90 litres.

Referring again to the aforementioned article, it can be calculated that the B200G requires a reflex cabinet volume of

V_B(enclosure volume)
= 67.66 litres

but for closed box operation, with a system Q(Q_{TC}) of 0.707 to give the flattest low frequency response:

$$V_B = \frac{V_{as}}{\left[\left(\frac{1}{\frac{1}{Q_{TC}} - 0.2}\right) \cdot \frac{1}{Q_{TS}}\right]^2 - 1}$$

= 22.77 litres

Unless you intend to pioneer a new type of expanding speaker

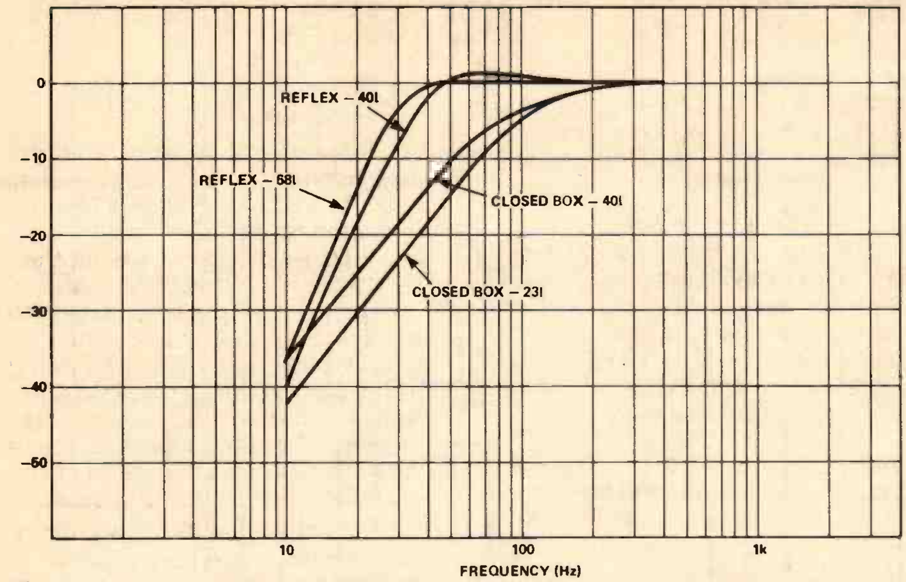


Fig. 1 LF response of the Kef B200G in optimum sized enclosures and in the Active-8 cabinet.

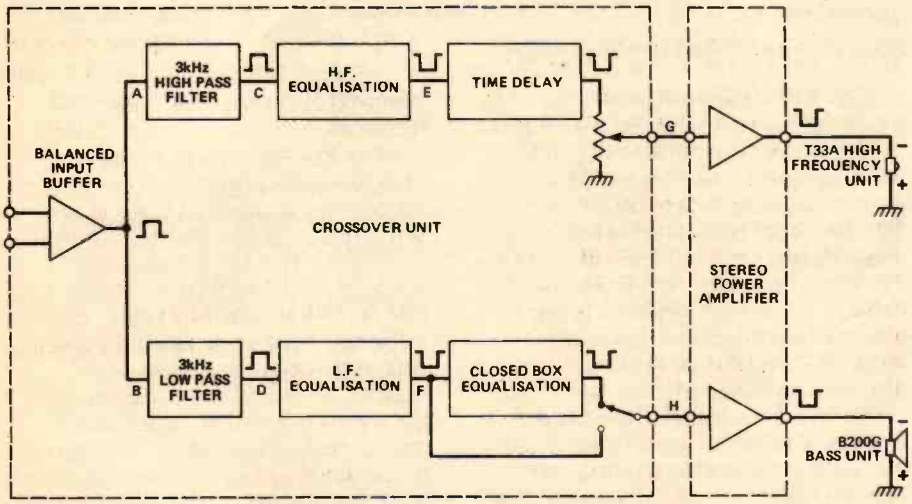


Fig. 2 Block diagram of the signal-handling stages of the Active-8 system.

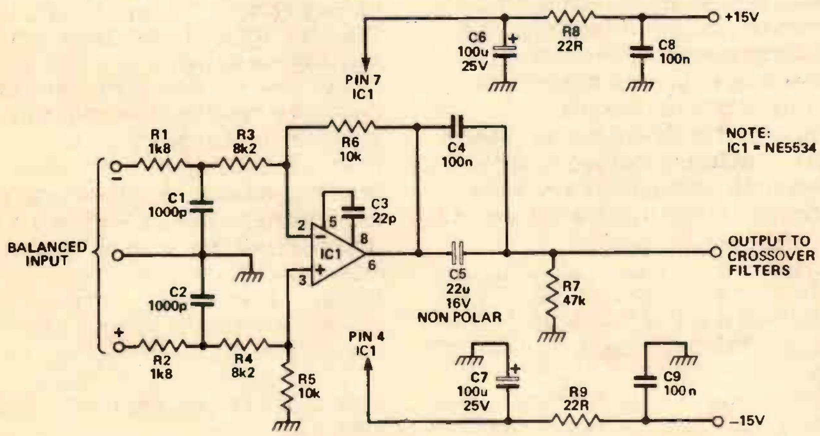


Fig. 3 Circuit diagram of the balanced input buffer.

cabinet, it is obvious that the Active-8 enclosure volume will have to be somewhere between these two extremes. The effect will be a hump in the response just above the low frequency roll-off point, whereas a larger than optimum closed box will have a Q_{TC} of less than 0.707, and will consequently exhibit an early roll-off with a gentle, rounded response shoulder.

After much calculating and plotting, the Active-8 enclosure volume was fixed at 40 litres. This gives a reflex response with a hump of:

$$R = 20 \log \frac{Q_{TS}\left(\frac{V_{AS}}{V_B}\right)^{0.31}}{0.4}$$

$$= 1.5\,\text{dB}$$

which is not likely to be objectionable. With closed box operation the system Q becomes:

$$Q_{TC} = \frac{1}{\left(\frac{1}{Q_{TS}\sqrt{\left(\frac{V_{AS}}{1.2V_B}\right)+1}}\right)+0.2}$$

$$= 0.557$$

Figure 1 shows a comparison between the Active-8 low frequency response and the same bass driver in optimum sized enclosure. It will be seen that the 40 litre curves are not far away from the optimum ones, so the choice is obviously about right.

The tuning vent should be a length of plastic rainwater pipe with a 75mm internal diameter (Dv). The cabinet is tuned to a frequency given by:

$$f_B = f_S\left(\frac{V_{AS}}{V_B}\right)^{0.31}$$

$$= 34.7\ \text{Hz}$$

which requires that the vent length is:

$$L_V = \frac{2340}{F_B^2\, V_B} \bullet D_V^2 - 0.731\, D_V$$

$$= 218.5\ \text{mm}$$

The Crossover Filters

A block diagram of the complete 'Active-8' system is shown in Fig. 2. It will be seen that each section of the crossover unit consists of a filter and an equaliser. Additionally, the high frequency path contains delay ciruitry to compensate for the acoustic centre of the T33A being about 38mm in front of that of the B200G, and the low frequency path has the facility to add bass equalisation for closed box use.

At the input of the crossover

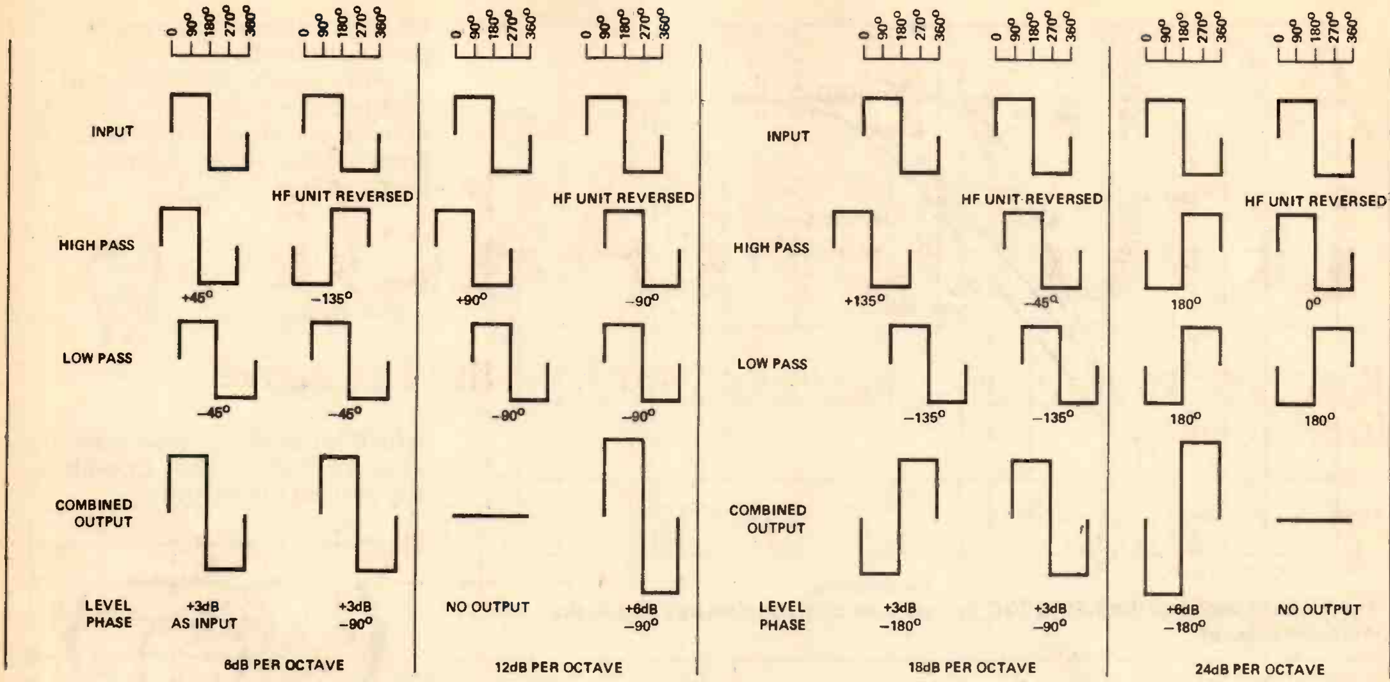


Fig. 4 The effect of 6, 12, 18 and 24dB per octave filters on signal level and phase.

unit is a balanced unity gain buffer stage, shown in Fig. 3. Until recently, only professional equipment had balanced interconnections but as operational amplifiers have become acceptable in top quality domestic equipment, some manufacturers have begun to appreciate the benefits of balancing and are making provision for balanced lines between pre and power amplifier or pre-amplifier and active speakers.

The input of the balanced buffer amplifier contains a degree of protection against radio-pick up by the connecting leads. Resistors R1 and R2 and capacitors C1 and C2 form a filter with its −3dB point at 88,4kHz — providing the signal source has a low output impedance. If used with a pre-amplifier with a high output impedance — say 10k ohms — this high frequency roll-off will move down into the audio range, so the value of the capacitors will have to be reduced to 150pF to avoid this.

The buffer amplifier output is AC coupled to the high and low pass crossover filters by C4 and C5. The non-polarised electrolytic is by-passed at high frequencies by C4 which should be a polycarbonate or polypropylene type. Carefully controlled listening tests have shown that polarised aluminium electrolytics, which are often used for inter-stage coupling, can cause effects which, although virtually impossible to measure, can be heard when an impeccable music source is used. During these tests, a bypassed non-polarised capacitor could not be detected, and for this reason, is used in the Active-8 whenever a large value component is necessary.

The traditional crossover filter is a 12 or 18dB per octave Butterworth stage, which has a number of shortcomings that have been eliminated in the Active-8 network.

The problem is this: the crossover should ensure that the combined output of both drive units remains constant at all frequencies. The effect of using 6,12,18 and 24dB per octave filters is illustrated in Fig. 4. It is important that both drivers are in phase through the crossover region, as any phase difference between them will cause their combined radiation pattern to tilt downwards, leading to colouration from increased floor reflections. This rules out the 6 and 18 dB per octave slopes; the 12 dB per octave filter with reversed connection of one drive unit or the 24dB per octave version both have the desired phase relationship between their outputs, but suffer from a 3dB jump in their combined response. In order to add two in-phase signals and arrive at a unity output, each signal must be 6dB down at the crossover frequency. This is easily accomplished with both 12 and 24 dB per octave stages by placing two 6 or 12 dB per octave filters in series. Both types are illustrated in Figure 5.

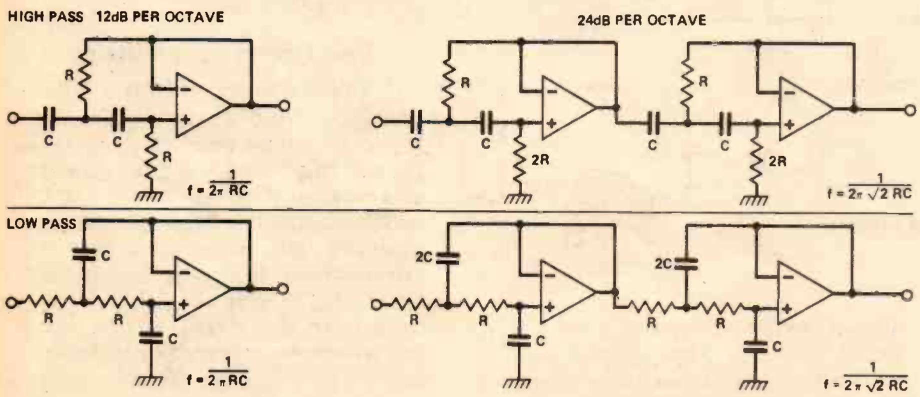


Fig. 5 Circuit diagrams of 12 and 24dB per octave high and low pass filters.

The discussion of the design process is completed in the next article, after which we will move on to describe the construction of the Active-8.

ACTIVE-8 LOUDSPEAKER

Barry Porter completes the design work on the ETI active loudspeaker.

The Active-8 was evaluated with both 12 and 24 dB per octave filters and no difference could be heard between them, so the 4 pole version was chosen as this gives slightly more protection to the high frequency unit by virtue of its steeper slope. It also has the additional advantage of reducing the level at the resonant frequency of the T33 — about 950hz — by about 40 dB, where its effects may be safely ignored. The response of both high and low pass sections is shown in Fig. 6 and the circuit of this part of the network in Fig. 7.

Drive Unit Equalisation

If each drive unit had a flat frequency response over its range of operation, life would be much more enjoyable for all concerned. Unfortunately this is not the case, so additional circuitry has to be used to correct the major inaccuracies. The Active-8 units were measured in free field conditions (free local playing field would be more accurate!) resulting in the plots of Fig. 8

Looking at the B200G response first, this shows a 6dB rise between 300 Hz and 3 kHz which the equalisation circuit shown in Fig. 9 cancels with reasonable accuracy, as the corrected plot shows.

The T33A also exhibits a response that rises with frequency, so a similar circuit is used to counteract this.

It will be seen from Fig. 8 that the T33A is slightly more sensitive than the B200G — about 3dB if the low frequency output at 1 kHz is compared to the 10 kHz output from the high frequency unit. This difference will be corrected at a later stage by placing a 3dB attenuator in the high frequency signal path.

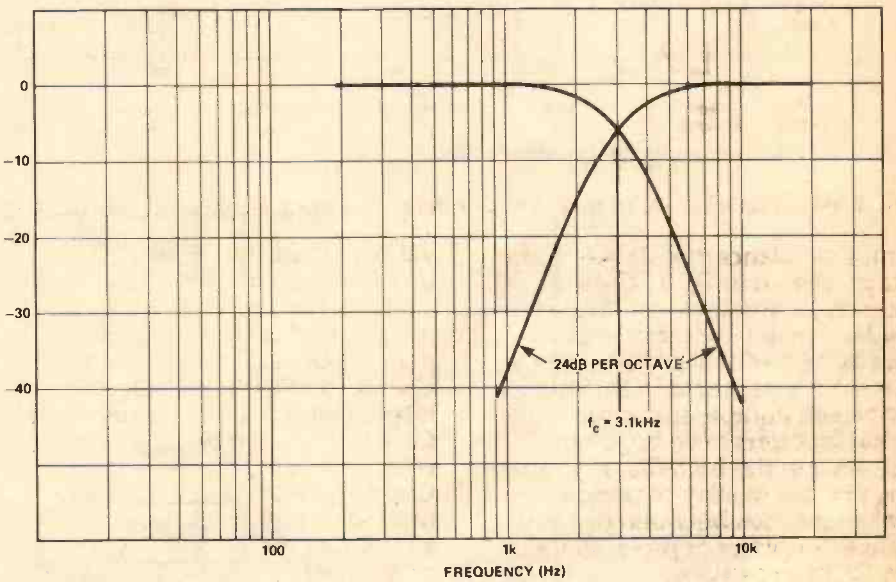


Fig. 6 Crossover filter response.

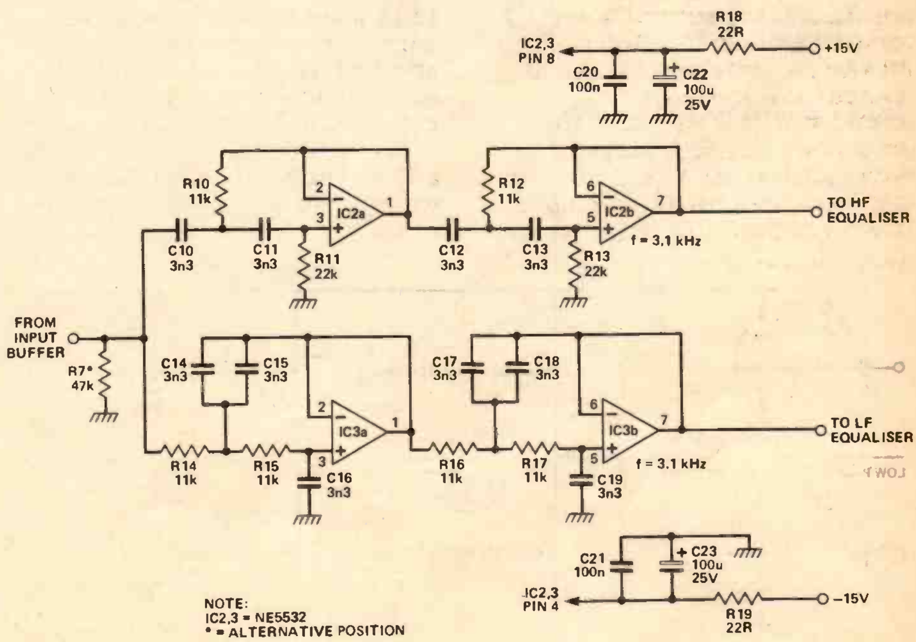


Fig. 7 Crossover filter circuitry.

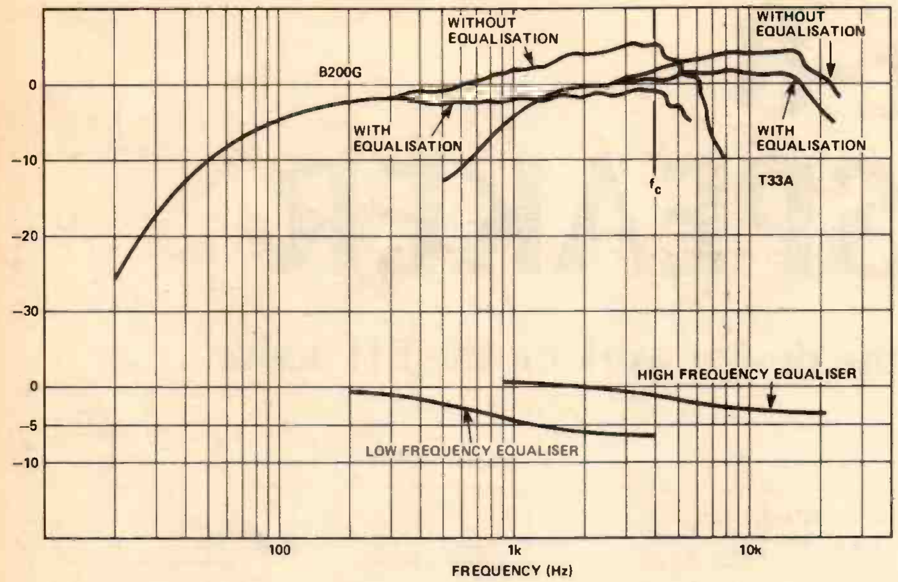


Fig. 8 Free-field response of the drive units.

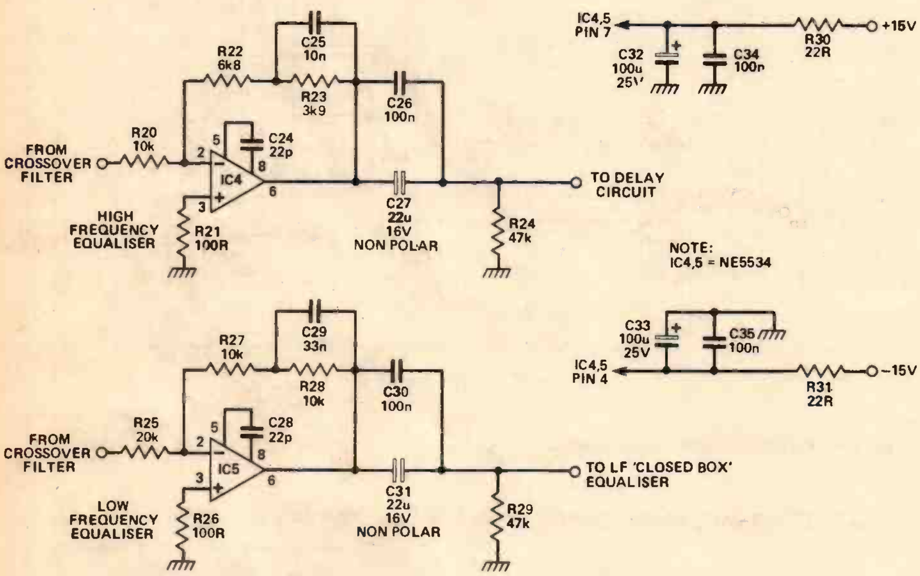


Fig. 9 Equaliser section circuitry.

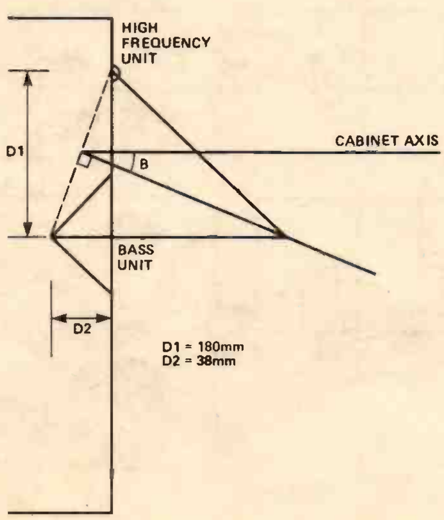


Fig. 10 The effect of the displacement of the speaker coils.

Time Delay

Ideally, the two drive units should have their acoustic centres on a plane that is perpendicular to the speaker axis. This is not the case however, as the T33a radiates from a point approximately 38mm in front of the B200G. Referring to Fig. 10, it can be seen that the radiation pattern will be tilted downwards at the crossover frequency by:

$$B = \text{Atan}\left(\frac{D_2}{D_1}\right) = 11.9^\circ$$

This could be compensated for by mounting the T33A on a different plane to the B200G, but this would introduce a number of mechanical difficulties in avoiding diffraction effects from the cabinet edges. The alternative solution applied to the Active-8 is to delay the high frequency signal by the amount of time it takes sound to travel 38mm, which is:

$$t_d = \frac{D2}{V} = \frac{38 \times 10^3}{343}$$

A suitable delay circuit formed from cascaded all-pass filters, is shown in Fig. 11. Each stage gives a delay at the crossover frequency of:

$$t = \frac{2RC}{1 + (2\pi fRC)^2}$$
$$= 27.6 \text{ ms } (110.5 \text{ ms total})$$

The use of this delay ensures that both units are in phase along the cabinet axis, so no vertical directivity shift occurs over the crossover region. Colouration in the critical mid-range is therefore minimised, and the improved dispersion characteristics assist in the production of a very stable stereo image with a considerable presence of depth information.

The previously mentioned 3 dB attenuator in the high frequency signal path is formed at the output of the delay circuit by R44 and R45.

Closed Box Operation

Although the 'Active-8' may prove quite acceptable with reflex loading, there are certain advantages to be gained from replacing the vent escutcheon with a blanking plate and reverting to closed box operation.

Although curve A in Fig. 1 may not look too promising, especially if your musical taste runs to material with more than its share of bass emphasis, remember that this is the anechoic response. Under normal listening conditions, room boundary reflections will give a perceived increase in low frequency output.

Closed box response rolls off at 12dB per octave, and therefore exhibits less transient overshoot and ringing than the 24dB per octave reflex response. Although the Active-8 will give good performance when used as a closed box in a small listening room, it will not have sufficient bass output for use in larger rooms. The technique employed to resolve this problem works like this:— as we have seen, the closed box response rolls off at 12 dB per octave, so if circuitry is placed in the low frequency signal path that introduces a counteracting 12dB per octave lift, the acoustic output of the speaker will remain flat at lower frequencies. Obviously, the equalisation cannot continue to rise in level, so at the point where it flattens the speaker roll-off will start, still retaining a 12dB per octave sope and with a Q value that is decided by the electronics. A suitable low frequency equalisation circuit is shown in Fig. 12. The Active-8 values are based upon the following parameters.

$$f_o = 48\,\text{Hz}$$
$$Q_o = 0.505$$
$$f_P = 13.2\,\text{Hz}$$
$$Q_P = 0.5$$
$$A_{DC} = 22.4\,\text{dB}$$

This gives a considerable increase in bass output without too much danger of either the circuitry or bass unit running out of headroom. As an experiment, the author applied the same low frequency equalisation technique to a pair of large domestic speakers with 300mm bass drivers, but kept the response flat to about 5Hz. The bass was certainly impressive, although analogue records could not be played due to turn table and cutting lathe rumble causing excessive cone movement. Both analogue and digital master tapes caused no problems, and it was clear that, although there was no musical information at very low frequencies, the extremely good phase characteristics of the speakers gave weight and solidity to the lower register that is lacking in all but the largest studio monitors.

The bass equalisation circuit is

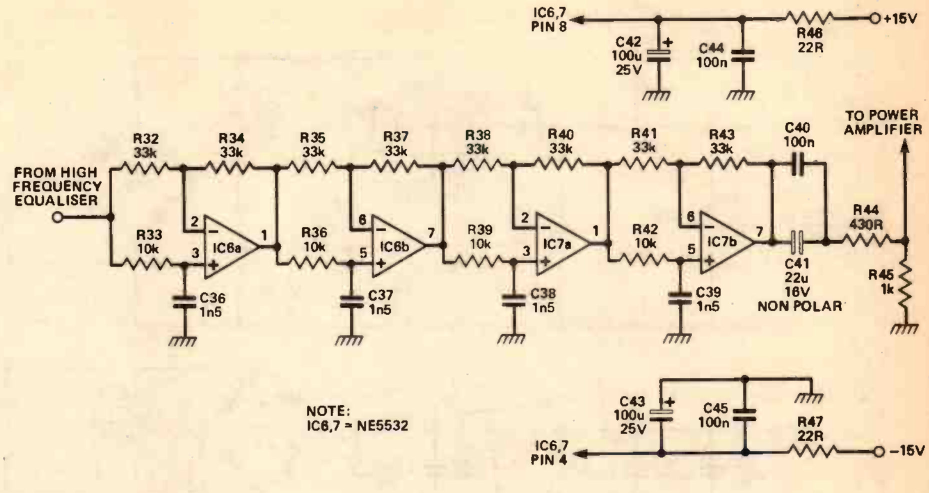


Fig. 11 Delay circuit.

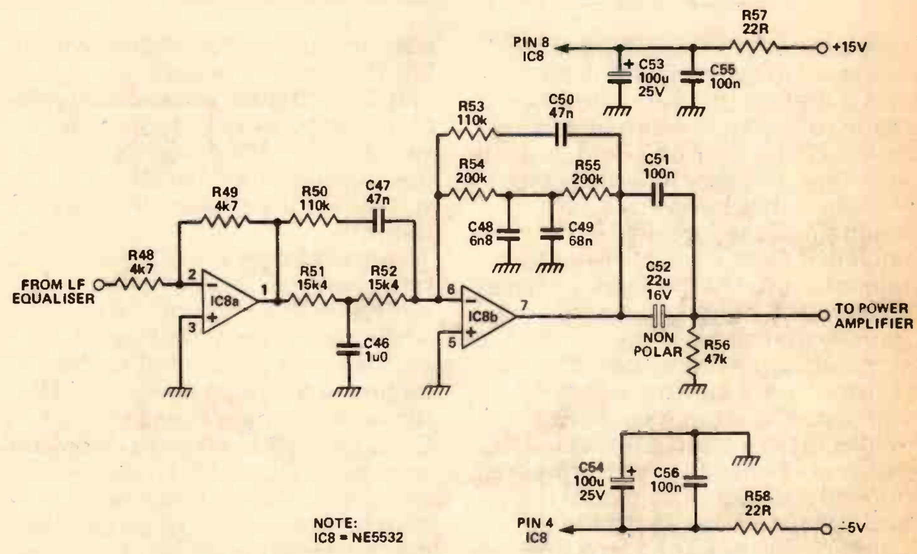


Fig. 12 Low-frequency equaliser for closed-box operation.

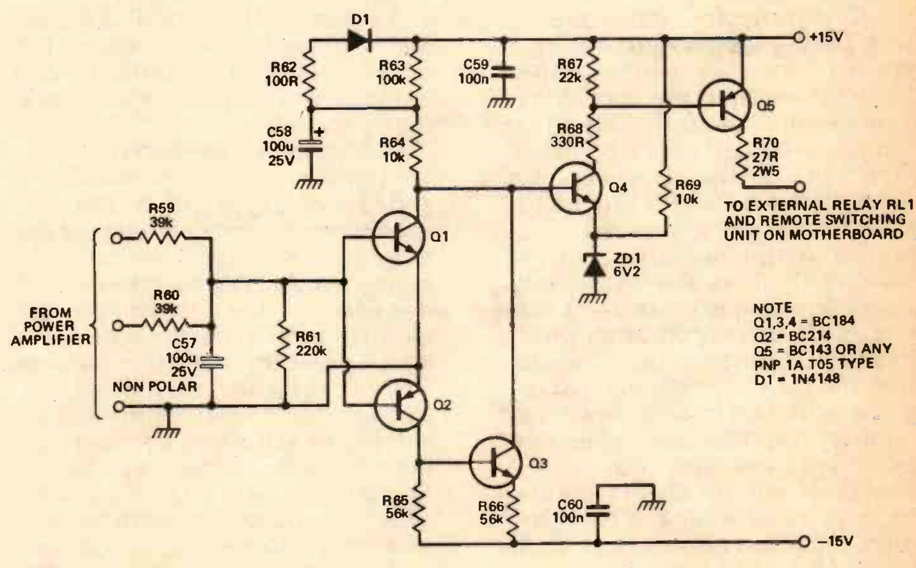


Fig. 13 Protection unit.

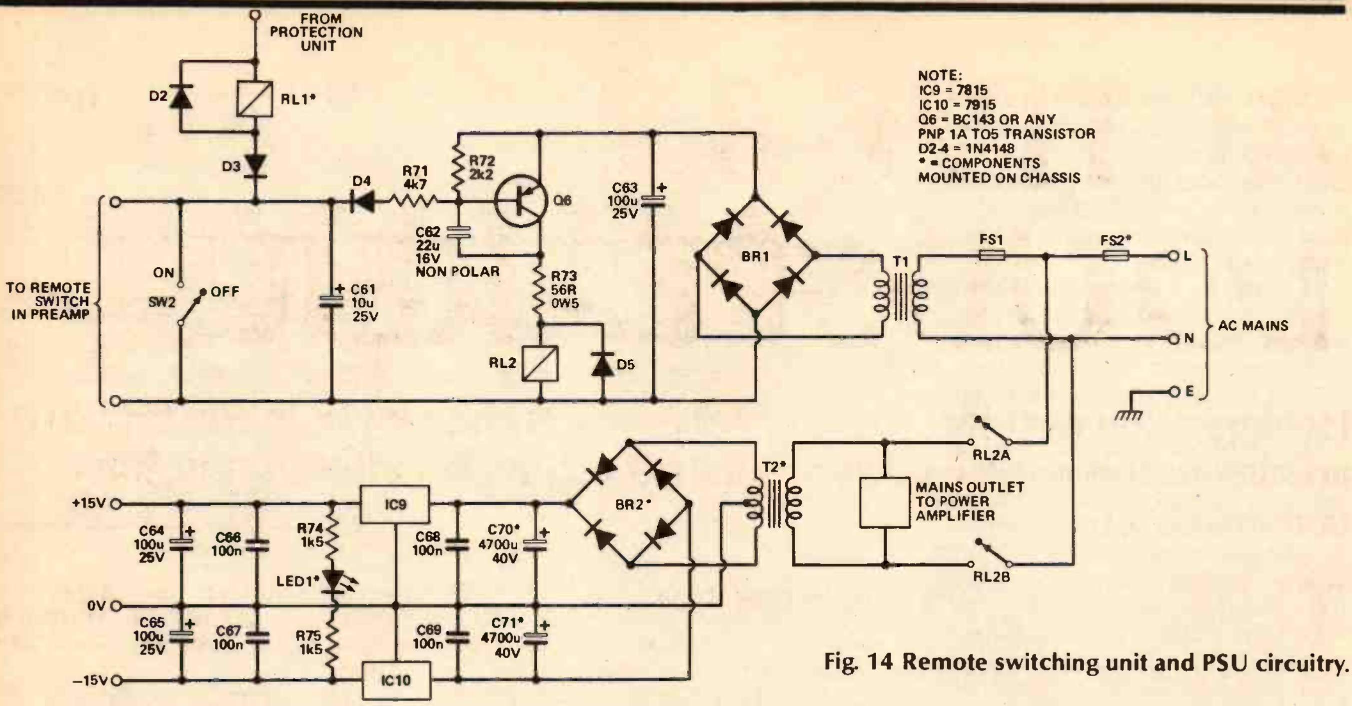


Fig. 14 Remote switching unit and PSU circuitry.

only used when the speakers are operated in the closed box mode, and therefore provision must be made to bypass it when necessary. As the circuit of IC8b (Fig. 12) is inverting, a further inverting stage — IC8a — has been added to maintain phase integrity. The choice of Q_P at 0.5 was made to minimise the low frequency phase shift, with f_P being set at a frequency that allowed the use of standard capacitor values. The resistor values are shown as calculated, but the nearest E24 values may be used with no noticeable change in performance. Similarly, the 75 nF capacitor is approximated by the paralleled combination of 68 nF and 6.8 nF (C48, 49).

Switch-On Delay

One problem encountered with the prototype Active-8 speakers was that switching them on or off required the adoption of a procedure not far removed from doing a pre-flight check on the family 747. It was all too easy, for example, to switch off the preamplifier while power was still applied to the filter units and power amplifiers, the reward being a superb example of transient handling as the drive units attempted instant self-destruction. To avoid this, the power amplifiers had to be switched on last. And switched off first. In spite of several feet of advisory Dymo tape, this sequence was not always adhered to, so to avoid wear and tear on drive units and nerves, the circuits of Fig. 13 and 14 were incorporated.

Together, these units provide remotely controlled mains switching, delayed connection of amplifier to drive units, disconnection of drive units before mains switch-off and continuous protection against excessive DC voltages at the power amplifier outputs.

Operation of these functions is best understood by considering a switch-on — switch-off cycle. The small transformer, T1 (Fig. 14) is permanently connected to the incoming mains so about 17 V DC sits on smoothing capacitor C63. Q6 is held off by R72, so the mains switching relay, RL2, is de-energised. The unit may be switched on locally, or by earthing the remote connection at the pre-amplifier, either action turning on Q6 by the pull-down of D4 and R71. RL2 is therefore energised, and contacts RL2A and RL2B apply mains voltage to the crossover unit power supply transformer, T2, and to the power amplifier via a mains outlet socket.

When the protection unit (Fig. 13) initially receives power, Q4 and Q5 are turned off, and the speaker drive units are disconnected due to RL1 being de-energised. As C58 charges up through R63, the bass voltage of Q4 rises until it reaches 6.8 V, at which point the transistor turns on. The current which then flows through R67 and R68 turns Q5 on and RL1 is activated, connecting the drive units to the amplifier. This take about 6 seconds, which allows all voltages to settle and switching transients to disappear.

In operation, any excessive DC voltage appearing at the power amplifier output will be detected by Q1 or Q2. A positive voltage will turn Q1 on, pulling the base of Q4 down, so that both Q4 and Q5 are turned off, as a result of which RL1 will disconnect the drive units. If the offset voltage is negative, Q2 will be turned on. Current flowing through R65 will turn on Q3 which will pull down the base of Q4. Again, Q5 will also turn off, de-energising RL1 and disconnecting drive units.

At switch-off, the remote connection is removed from earth, immediately causing RL1 to revert to its relaxed state as its OV path via D4 (Fig. 14) is broken. The drive units are therefore disconnected before the mains is switched. Q6 is held on for a short time by C62, so RL2 cannot switch the mains until the amplifier outputs are well and truly broken by RL1.

All this means that the Active-8 units can be switched on and off without fear of the clicks, bangs and thumps that are so often the hallmark of home produced equipment. The remote connections of each speaker can be joined together and taken by a single wire to the pre-amplifier where a single pole switch can be used to operate the speakers.

Next: Construction.

ACTIVE-8 LOUDSPEAKER

Having discussed the design process at some length in the first two articles in this series, Barry Porter now turns his attention to the construction.

The time has now arrived to do your Chippendale impression, having first checked that your home insurance policy covers such unlikely events as drilling holes in the dining table and spreading the latest high technology glue over the G-Plan.

Figure 1 shows the cabinet construction in sufficient detail to enable the average DIY duffer to build a pair without too many tears. The exact method of assembly is not too important, providing the final result is rigid and all joints are airtight.

Do not attempt to cut the veneered panels with a handsaw — find a supplier who is willing to cut them for you using a high speed circular saw. This will ensure that the edges are square, which is essential if you want the final result to have a professional appearance. The bare edges should be covered with matching strip veneer. Avoid the type that is ironed on — plain veneer is much better, and when attached with contact adhesive is less likely to drop off.

The most difficult part of the cabinet construction is preparing the front baffle. The drive units should be recessed using the templates provided to mark out the panel before commencing. If you do not have the correct tool to route out the rebated areas, it can be done by cutting through the veneer with a sharp knife, and forming the rebate with a chisel.

Recess an area around the tuning vent to a depth of 3.5mm using the T33 template. This is for the interchangeable escutcheon which is replaced with a similar sized blanking plate for closed box operation. These plates should be made from 10SWG alloy with a 75mm diameter hole in the escutcheon. They should have four fixing holes in the same positions as the T33.

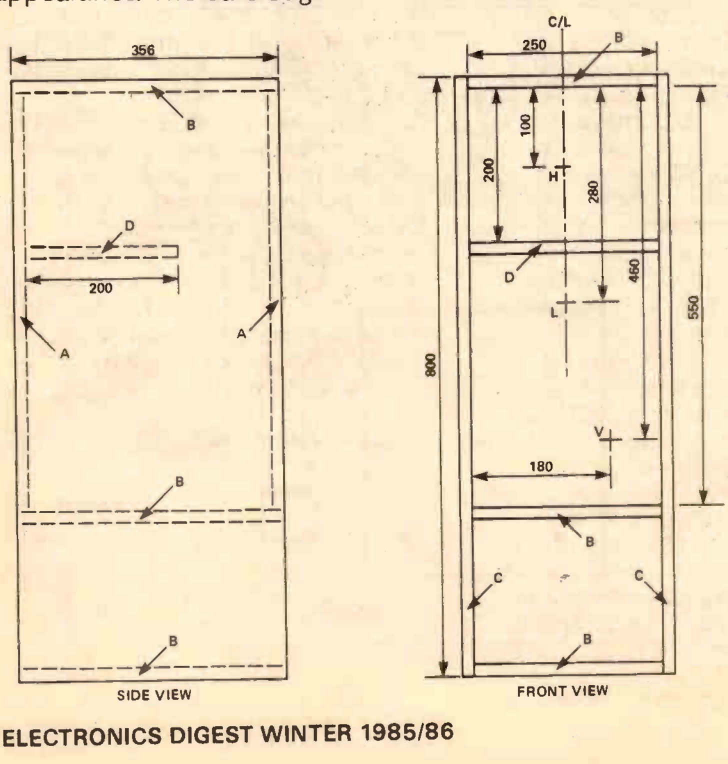


Fig. 1 Cabinet construction details.

ALL JOINTS TO BE GLUED AND AND STRENGTHENED WITH 25mm SQUARE SOFTWOOD (NOT SHOWN)

HOLES
H = T33A
L = B200G
V = TUNING VENT

CAPACITY = 44l gross
APPROX. = 40l net

The escutcheon and both drive units should be attached by M4.5 x 20 machine screws with Tee nuts fixed to the inside of the baffle. Do not be tempted to fix the units with woodscrews, even though you will find some enclosed with the T33; use these for hanging your holiday snaps on a wall instead.

Before fitting the drive units, half fill the cabinet with BAF wadding or medium density polyurethene foam blocks, taking care not to pack the area behind the vent too tightly and leaving a six inch clearance around the base unit.

A suitably sized hole for the lead-out wires should be drilled through the cabinet bottom and made airtight by a liberal application of glue from the inside. The cable should be firmly attached to the inside of the enclosure with 'P' clips so that, if it is accidently pulled, the connecting tags on the drive units will not be damaged.

If you know that the amplifier you will be using has a common output earth terminal a 3 core connecting lead is sufficient, but if the speakers are ever likely to be used with unknown amplifiers, separate connections to each drive unit should be made available. Four core mains cable is quite suitable for this purpose, as only short lengths are used.

The prototype cabinets were fitted with 50mm dual wheel Kenrick castors, which certainly eased the job of moving them around during construction. If you are a believer in the theory that loudspeakers should be mounted on reinforced concrete pillars that descend half a mile into the bedrock (when surely seismic disturbances will influence the sound quality?) the castors can be replaced with carpet piercing spikes. In practice, no advantage has been found by doing this, which pro-

PARTS LIST — MOTHER BOARD & PSU

RESISTORS (all¼ W 1% metal film unless otherwise stated)	
R1,2	1k8
R3,4	8k2
R5,6	10k
R7	47k (see text)
R8,9	22R
R71	4k7
R72	2k2
R73	56R ½W
R74,75	1k5
CAPACITORS	
C1,2	1n0 polystyrene
C3	22p polystyrene
C4	100n polycarbonate
C5,62	22µ 16v non-polarised electrolytic
C6,7,64,65	100µ 25V radial electrolytic
C8,9,66-69	100n polyester
C61	10µ 25V radial electrolytic
C63	1000µ 25V radial electrolytic
C70,71	4,700µ 40V electrolytic (see text)
SEMICONDUCTORS	
IC1	NE5534
IC9	7815
IC10	7915
Q6	BC143 or any PNP 1A T05 transistor
BR1	50V 2A bridge rectifier
BR2	209V 5A bridge rectifier
D2-5	1N4148
LED1	any panel-mounting LED
MISCELLANEOUS	
FS1	200mA anti-surge fuse and PC-mounting holder
FS2	1A anti-surge fuse and panel-mounting holder
RL1	DPCO relay, 12V coil
RL2	DPCO relay, 12V coil, mains contacts
SW1	SPCO PC mounting
SW2	SPDT slide, toggle, etc.
T1	6-0-6V 3VA PC-mounting transformer
T2	12-0-12V, 500mA transformer
PCB; 5 off 10 way PCB plugs; relay holders as desired; mains outlet; audio connectors; cable, cable clips, etc.	

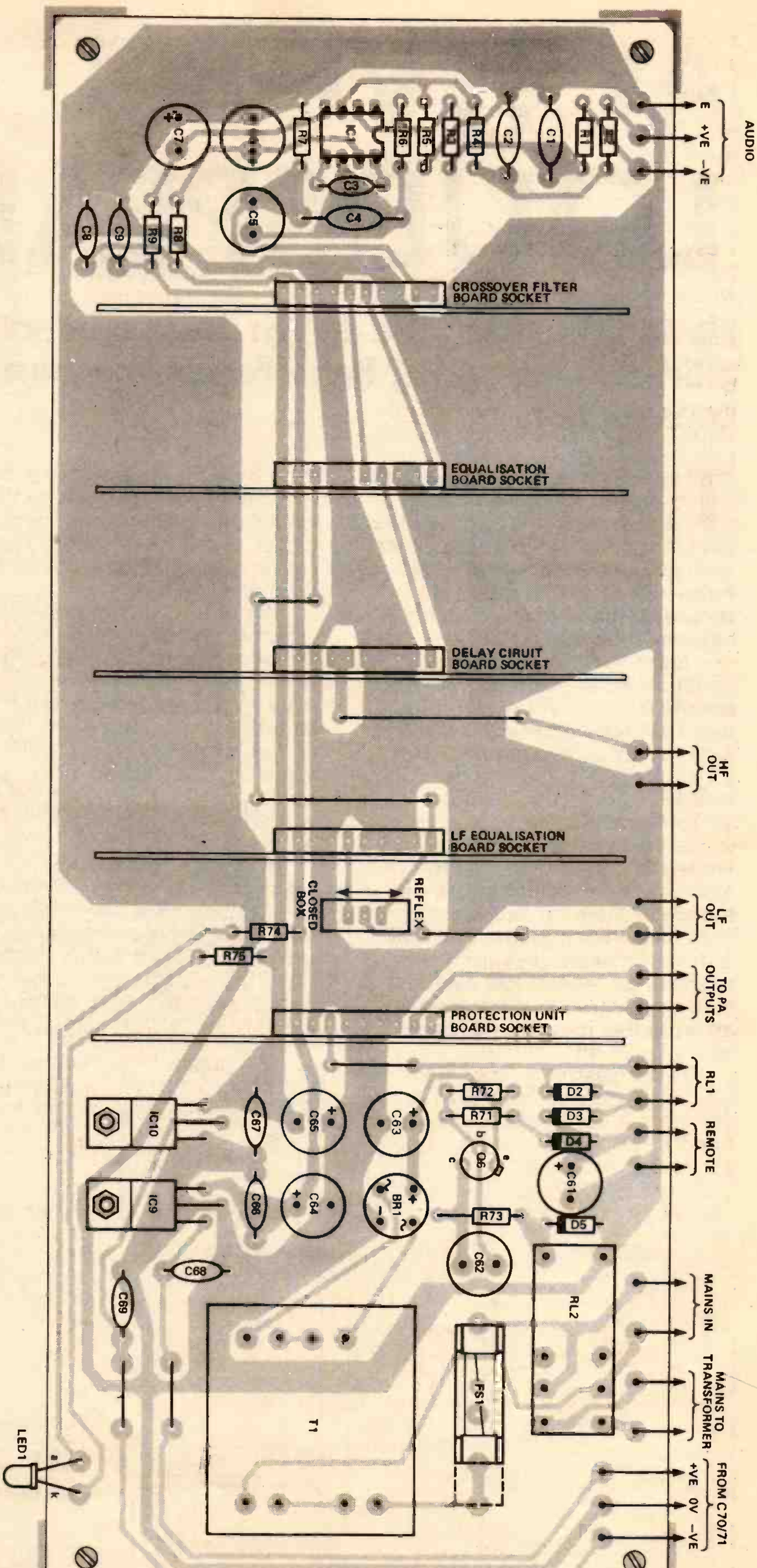


Fig. 2 Overlay of the mother board.

bably says something about the author's hearing or the quality of his living room carpet.

Should the final appearance require the drive units to be hidden from small sticky fingers or an inquisitive budgerigar, suitable grilles can be made from 12.5mm plywood covered with open weave material. Cut away the centre of the plywood to the point where the remaining frame is well clear of the units, while retaining sufficient strength to stay flat — about 50mm should cause no problems. Give the frame a coat of matt black paint before glueing or stapling the covering material into place. Attachment to the speaker can be by the usual plastic socket fixings or, less attractively, strips of adhesive backed Velcro. If you intend packing your amplifier and filter unit in the open area at the bottom of the cabinet, the grille should be made long enough to hide the inevitable jumble of connecting cables from sight.

Once the cabinets are built and drive units installed, they may be set aside while work proceeds with the active crossover filters.

Construction

To make the filter unit as flexible as possible (not in the bendy sense), the mother board method of construction is employed. Although this is more costly in terms of the number of individual circuit boards, it does mean that changes can be introduced quite easily, and fault correction usually means replacing a plug-in board with a spare one.

The suggested mother board layout is shown in Fig. 2, and its overall size allows considerable freedom of choice when selecting a suitable enclosure.

Some circuitry has been placed on the mother board, namely the input amplifier, power supply stabilisers and remote mains switching unit, but the filter and equalisation stages are mounted on the plug-in cards shown in Fig. 3 (except for the delay unit, which will be given next month due to lack of space). Note that two locations are shown for R7; it should normally be placed at the output of the buffer, but if, for any reason, the buffer is not to be used, it should be placed at the input of the crossover filters instead.

Components not mounted on the circuit boards include the connectors, power supply transformer, rectifier bridge and smoothing capaciters. Ideally a toroidal mains transformer should be used, but as the circuitry is fairly tolerant of radiated hum fields a suitably specified frame type should not cause problems. The main smoothing capacitors should be at 4700uf 40V, but if space is not at a premium 10,000uf components can be used to advantage. These should be firmly clamped to the cabinet, and the rectifier bridge mounted directly to the supply rail terminals.

The type of signal connector used will probably be decided by the constructors financial status, but the input signals should be connected via multi way sockets with at least four pins. Good quality DIN connectors are acceptable, but professional XLR connectors are preferable. The accepted standard is that signal inputs use chassis mounted sockets, the type required for the 'Active-8' inputs being termed XLR-4-31. These should be wired to the following convention:

Pin 1 — Signal earth
Pin 2 — Signal +
Pin 3 — Signal —
Pin 4 — Remote switching

If you use an unbalanced feed from your pre-amplifier, this should be taken to Pin 2, with Pins 1 and 3 joined together in the XLR cable plug.

The filter unit output is at low impedance, and the connection to the power amplifier should be unbalanced, using an insulated phone socket. If you want to stick with professional connectors, the outputs should employ XLR-3-32 chassis mounting plugs with signal on Pin 2 and Pins 1 and 3 joined to earth.

Whether used balanced or unbalanced, the input amplifier is DC coupled to the signal source, working on the assumption that pre-amplifier output stages are normally AC coupled. If you use one that isn't, it is quite permissible to place 10 or 22μf non polarized capacitors between the input socket and circuit board (not forgetting to place a 0.1μf polycarbonate in parallel).

The internal wiring is shown in Fig. 4, and should present no problems. (On reflection, if you have progressed this far without being removed by men in white coats, you will probably build the filter unit blindfolded!)

Next we will conclude the construction details with the delay section overlay, and give Buylines (so don't phone us!). We'll also give some suggestions for extensions.

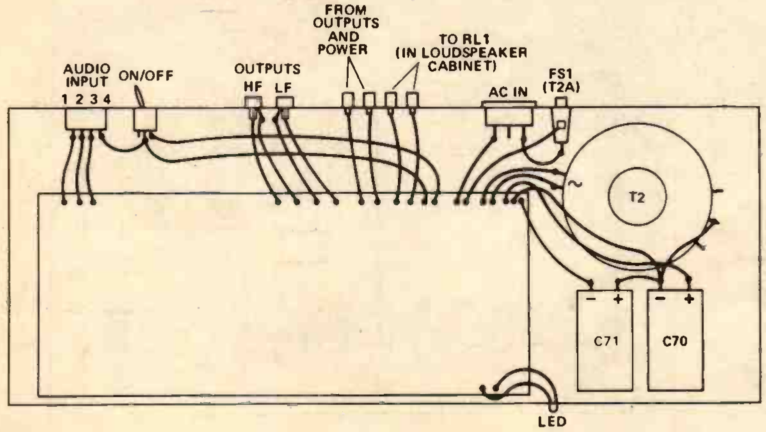


Fig. 4 Internal wiring diagram (note — BR2 not shown).

PARTS LIST — EQUALISER

RESISTORS (all ¼W 1% metal film)	
R20,27,28	10k
R21,26	100R
R22	6k8
R23	3k9
R24,29	47k
R25	20k
R30,31	22R
CAPACITORS	
C24,28	22p polystyrene 2.5%
C25	10n polystyrene 2.5%
C26,30	100n polycarbonate
C27,31	22μ 16V non-polarised electrolytic
C29	33n polystyrene 2.5%
C34,35	100n polyester
SEMICONDUCTORS	
IC4,5	NE5534
MISCELLANEOUS	
PCB; 10way PCB socket.	

PARTS LIST — LF EQUALISATION CIRCUIT

RESISTORS (all ¼ W metal film)	
R48,49	4k7
R50,53	110k
R51,52	15k4
R54,55	200k
R56	47k
R57,58	22R
CAPACITORS	
C46	1µ0 polycarbonate
C47,50	47n polycarbonate
C48	68n polycarbonate
C49	6n8 polystyrene
C51	100n polycarbonate
C52	22µ 16V non-polarised electrolytic
C53,54	100µ 25V radial electrolytic
C55,56	100n polyester
SEMICONDUCTOR	
IC8	NE5532
MISCELLANEOUS	
PCB; 10-way PCB socket.	

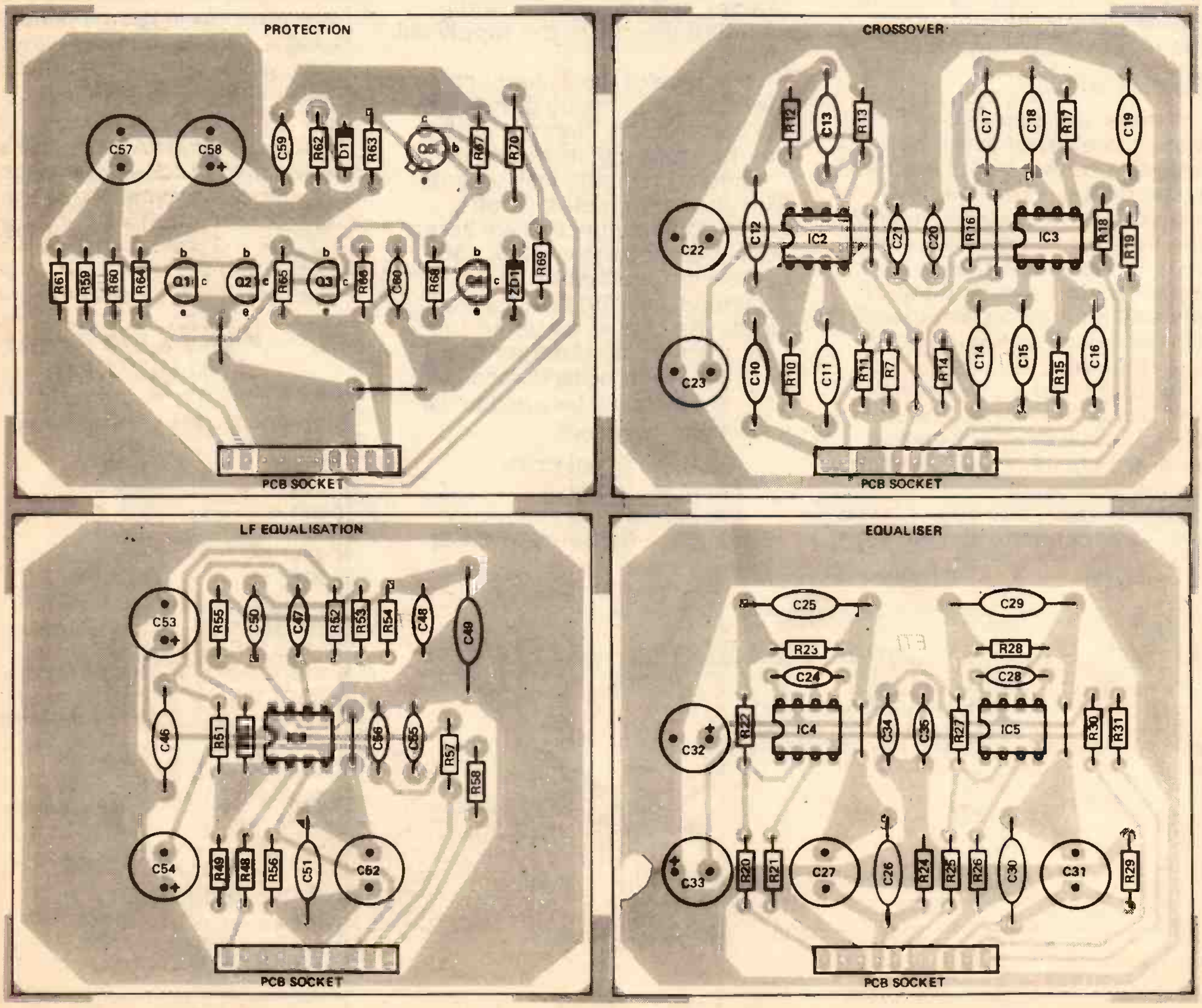
PARTS LIST — PROTECTION UNIT

RESISTORS (all ¼ W metal film unless otherwise stated)	
R59,60	39k
R61	220k
R62	100R
R63	100k
R64,69	10k
R65,66	56k
R67	22k
R68	330R
R70	27R 2.5W
CAPACITORS	
C57	100µ 25v non-polarised electrolytic
C58	100µ 25v radial electrolytic
C59,60	100n polyester
SEMICONDUCTORS	
Q1,3,4	BC184
Q2	BC214
Q5	BC143 or any 1A PNP T05 transistor
D1	1N4148
ZD1	6V2 300mW Zener
MISCELLANEOUS	
PCB; 10way PCB socket.	

PARTS LIST — CROSSOVER FILTERS

RESISTORS (all ¼ W 1% metal film)	
R7	47k (see text)
R10,12,14,15,16,17	11k
R11,13	22R
R18,19	22R
CAPACITORS	
C10-19	3n3 polystyrene 2.5%
C20,21	100n polyester
C22,23	100µ 25V radial electrolytic
SEMICONDUCTORS	
IC2,3	NE5532
MISCELLANEOUS	
PCB; 10-way PCB socket.	

Fig. 3 Overlay of the protection, LF equalisation, crossover filters and equaliser sections.



ACTIVE-8 LOUDSPEAKER

Warning! This introduction contains a pun which may be harmful to readers of a sensitive disposition! Barry Porter sets his active imagination to work once more and brings this series of articles to a tri-amp-hant close (Ouch! — Ed.)

Once completed, the units should be tested. Initially, remove the plug-in boards, switch on and ensure that the correct voltages appear where they should. Having established that the mother board is operating correctly, in particular that the 15-0-15 V supply rails are present, the plug-in boards should be inserted one at a time. It should be possible to connect a signal generator to the input and verify that each board is working by checking its output. If any problems appear, make sure that the IC voltages are correct — namely that +15 V and −15 V are on the supply pins and that both inputs and the output are within a few mV of 0 V. Non-working stages should be carefully inspected for faulty soldering and component insertion, and if no obvious error can be seen, the IC should be changed.

Once everything is working, the response of the two outputs should be plotted and compared to similar measurements taken from the second unit. If these agree to within about 0.25 dBm, it is safe to assume that no major errors are present, and proceed with the final connection to the speakers.

The high and low frequency outputs of the filter unit are connected to the two channels of a stereo power amplifier. A number of factors will probably decide the choice of amplifiers, not the least being cost. It is important that the four power stages of a stereo pair of Active-8 units are as identical as

PARTS LIST — DELAY UNIT

RESISTORS (all ¼ W 1% metal film)

R32, 34, 35, 37, 38, 40, 41, 43	33k
R33, 36, 39, 42	10k
R44	430R
R45	1k
R46, 47	22R

CAPACITORS

C36-39	1n5 polystyrene
C40	100n polycarbonate
C41	22µ 16 V non-polarised electrolytic
C42, 43	100µ 25 V radial electrolytic
C44, 45	100n polyester

SEMICONDUCTORS

IC6, 7	NE5532

MISCELLANEOUS

PCB; 10-way PCB socket.

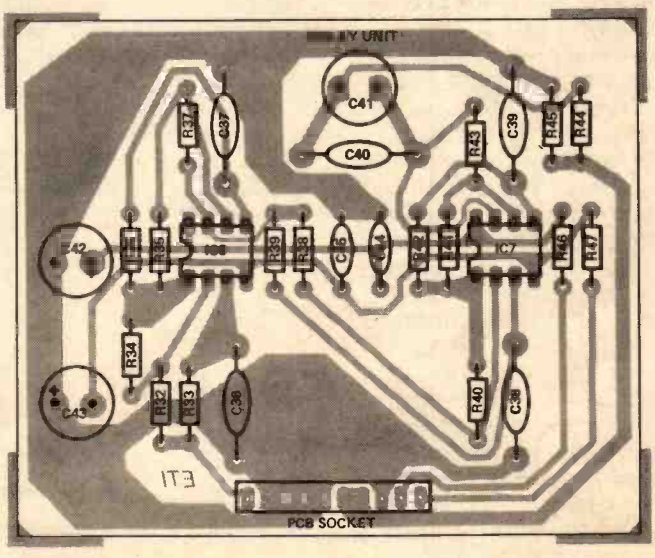


Fig. 1 The missing link — the PCB overlay for the delay unit.

possible. Regarding amplifier power, the speakers will operate at their best when driven by good quality units in the 100-150 watts region; anything below 50 watts per channel should be avoided, as transient clipping is likely to happen too often for comfort. At the top end, providing they are used with caution, there is no reason why 200 or 250 watts should cause any problems.

Before making the final connections the protection relay RL1, should be fitted — preferably inside the cabinet where, if an octal based version is used, the base can be screwed to the cabinet with 20mm chipboard screws passing through 10mm tubular spacers.

Once everything is connected up, the complete unit should be tested, making sure that both relays operate correctly so that a delay of about 6 seconds occurs at switch-on, and RL1 is released before RL2 when the units are switched off.

If everything is working, connect the speakers to your pre-amplifier using good quality screened cable. When fed from a balanced output, the connecting cable should contain a twisted pair of conductors within an outer screen. The conductors carry the signal to the inverting and non-inverting inputs, the screen being connected to the 0V contact. For unbalanced operation, the signal should be applied to the non-inverting (+) input, and the inverting (−) contact of the connecting plug should be connected to the cable screen. If you are using a pre-amplifier with a high output capability it may be advantageous if there is less gain in the system, and this can be achieved by leaving the inverting input unconnected. Some amplifiers (such as the Quad 303 and 405) invert the signal phase, so if you are using such a power stage the overall phase integrity may be maintained by connecting the pre-amplifier output to the inverting input of the buffer amplifier, with the non-inverting and N contacts joined to the screen of the connecting cable. Of course, if your pre-amplifier is also of the inverting type, this will cancel the power amplifier inversion, in which case the non-inverting input of the buffer should be used.

All that now remains is to put stylus to groove, sit back, and discover the joys of being 'Active-8-ed'!

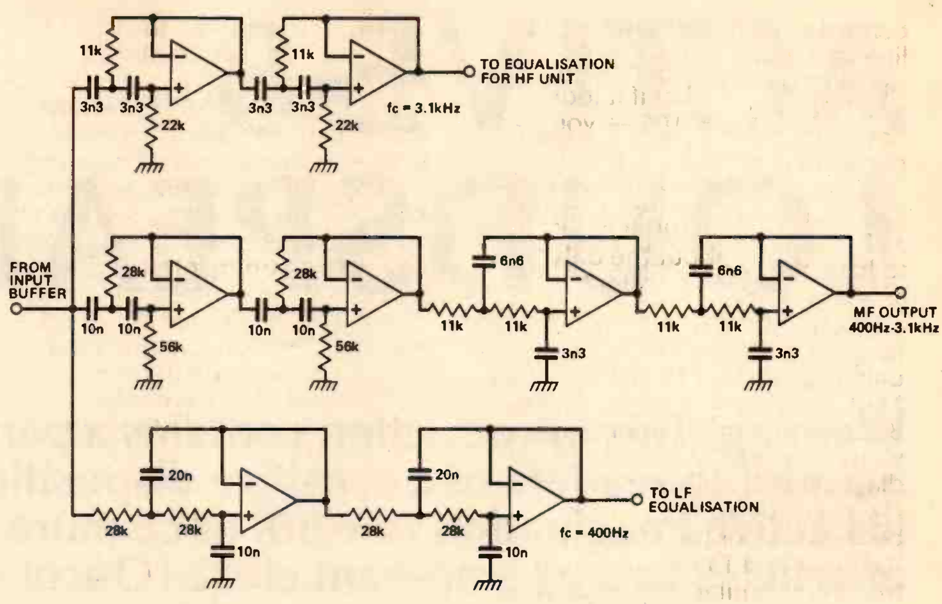


Fig. 2 Basic circuit diagram for a three-way cross-over. You'll have to work out the details (and the PCB) yourself.

Three Ways To Improve The System

The Active-8 was designed to be used with a stereo power amplifier providing power for each channel, which limited the number of drive units that could be used to two. Experimentation is the essence of speaker building (it is one form of building that doesn't require planning permission, except of the matrimonial kind) and most speaker builders go through a phase of Bigger is Better thinking. If, for reasons of sound output level or to impress the next door neighbours, you decide to use a larger than 200mm bass driver — say a 250 or 300mm unit — you will have to start thinking in terms of tri-amplifcation and mid range units. Although a few manufacturers claim to have produced 300mm units that will operate up to 2 or 3 kHz, in practice they leave much to be desired, so the additional complexity of adding a mid range driver is certainly worthwhile.

There are several good units available, but the author has always favoured the KEF B110 in its high power handling form (KEF part no. SP1057).

Depending upon the parameters of the chosen bass unit, you are likely to be using a cabinet of 60 to 120 litres. The basic rules are to keep the cabinet as narrow as possible with drive units close together and vertically in line. If your cabinet building ability is above average, you may like to consider putting the mid and high frequency units in a small enclosure separated from the main cabinet, which allows the acoustic

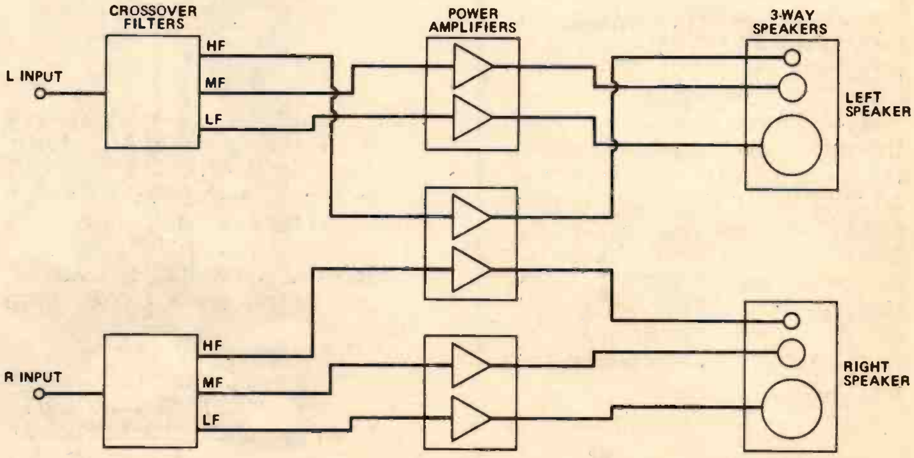


Fig. 3 Suggested set-up for a tri-amped system.

centres of all three units to be in-line and removes the need for signal delay. So what if it looks like a B&W 801 or KEF105 — you're not planning to go into competition with them, are you?

Using a 300mm bass driver such as the KEF B300.B or SEAS 33F-ZBX/DD (about the only unit to have the same transient attack as a JBL 15" monitor, but at about a quarter of the price), a B110 mid range and T33A high frequency unit, the network filters should be 24db per octave using the series Butterworth arrangement previously explained. A basic circuit diagram of the filters is shown in Fig. 2. Equalisation should not be required for the B110, but the bass unit and T33A will require treatment similar to that provided in the Active-8.

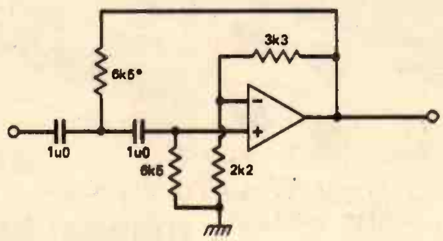
As the name suggests, you will require three stereo amplifiers for a tri-amplified set-up. These should all be of the same type to avoid system gain differences, and should be connected as shown in Fig. 3. Note that we have given no constructional details or PCB layout for this modification — it is intended purely as a starting point for those wishing to experiment further.

Sixth Order Bass Alignment

One of the drawbacks of the equalised closed box form of the Active-8 is the rather excessive cone excursion caused by subsonic signals — and there are plenty of those to be found on the average analogue record. Most record playing systems have some degree of subsonic filtering, but often this is too gentle to be effective, or begins to roll-off at a frequency well into the audio band. If you want to obtain very low bass output without subsonic excursion problems, you may like to experiment with a sixth order alignment. The basic requirement for this is that the reflex cabinet resonance (f_B) is lowered by half an octave, and that an active two-pole filter is introduced into the signal path, this having a Q value of 2 and a cut off frequency the same as the revised cabinet frequency. The Active-8 therefore has its f_B reduced from 34.7 to a new value given by:

$$f_B(\text{new}) = \sqrt{\frac{f_B^{\,2}}{2}}$$
$$= 24.5\,\text{Hz}$$

This requires that the tuning vent length becomes almost 500mm which is likely to be a problem. A quick calculation shows that a vent with 50mm internal diameter should be 207mm long, which is a bit more manageable. You will find that if you select the appropriate grade of plastic pipe, one with a 50mm internal diameter will slide comfortably into a 75mm one. It also has sufficient wall thickness of glue the outer end to a new escutcheon, so it is quite possible to have interchangeable 4th and 6th order alignments.



* 6k8 is 2 x 13k in parallel

Fig. 4 Circuit for a second-order filter.

The 2nd order filter shown in Fig. 4 should be inserted in the low frequency path in place of the closed box equalisation circuit. It is tuned to 24.5Hz ($f = 1/2\pi RC$) with a Q of 2 being set by the gain of 2.5 from the relationship:

$$\text{Gain} = 3-(1/Q)$$

The main problem with a 6th order system is the amount of phase shift that it introduces. Although this can cause some types of bass sound to become less solid, there is no sign of this with low organ notes, so perhaps this alignment is best recommended to those who are turned on by that sort of thing.

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BUYLINES

1% metal film resistors are available from a number of suppliers in almost all of the values required, the only difficult item being the 15k4 specified for R51 and R52. We don't know of a supplier for this so we can only suggest you use two resistors in series and stand them on end on the PCB. A 13k and a 2k4, both available from Maplin, should do the trick; ordinary mortals may well find that a 15k 1% is perfectly adequate on its own. The NE5532 and NE5534 are available from Watford, Technomatic, Rapid, etc, and the PCB-mounting transformer and most of the capacitors are also widely available. Non-polarised electrolytics in radial form are not readily available to the amateur, but Maplin and Cirkit both stock 50V axial components which could be mounted end-on. These two companies are also among those which stock the PCB plugs and sockets used, but note that there are some interesting discrepancies in stocking habits here and that you may need to order from more than one supplier to get the matching plug and socket halves you need. RL2 is also a Maplin type, and any relay with the correct contact arrangement and coil voltage can be used for RL1. The XLR type audio connectors recommended in the text are available from numerous suppliers including Electrovalue, Cricklewood, Maplin and Cirkit, and the PCBs are all available from our PCB Service.

AUDIOPHILE FM TUNER

A high performance stereo tuner with preset and manual tuning facilities, LED tuning and signal strength scales. Optional extras are an automatic search-and-lock tuning facility and a stereo LED audio level indicator. Design by Ray Marston. Development by Steve Ramsahadeo.

The heart of this unusual FM stereo tuner project is a ready-built type 7254 tuner module available from Cirkit. This module has built-in varicap diodes and is tuned by an externally applied DC voltage. It requires a mere handful of external components, plus a regulated 12 V power supply, to make a ready-to-run tuner unit which switch selection of muting, AFC and mono/stereo functioning. The audio output of the module (about 200 mV RMS) can be fed directly to the input of any stereo audio preamp.

Construction

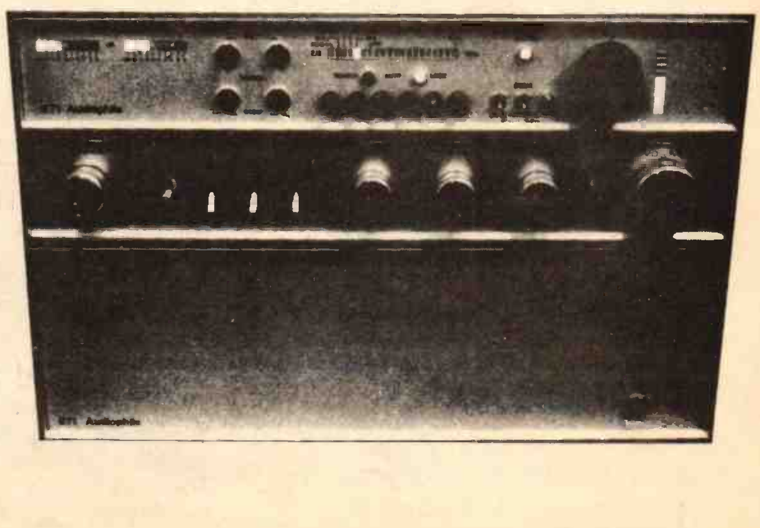
We strongly recommend that you build up the individual units of this project one at a time and interconnect and test them on the open bench before fitting the full set of modules (with or without the optional 'extras') into the final case. Start off by building the power supply unit and check that a stable 12 V is available at the output of IC1.

Next, refer to the basic wiring and interconnection diagram and temporarily wire up the 7254 module as shown,

Spot the Tune

In our Audiophile tuner unit we've provided the basic 7254 module with switch-selected tuning either with one of four preset pots for fixed station selection, or with a 10-turn pot for normal manual tuning. As an optional extra, you can also tune with a special search-and-lock circuit which automatically locates and locks on to strong FM broadcast stations. This circuit is provided with 'search' buttons that can be used to rapidly find the approximate location of a wanted station or reject an unwanted station.

We've designed our FM tuner to cover the 87.5 MHz to 104.5 MHz frequency range, with tuning indication provided on a linear 20-LED frequency scale. We've also provided the unit with a 10-LED signal-strength meter and with an optional stereo audio level indicator with 10 LEDs on each channel. The completed 'full option' unit is exceptionally attractive, with lots of fixed and moving lights, and gives a very impressive performance in the automatic 'search-and-lock' mode.



connecting the following components in place: R1, R2, R3, LED 1, SW1, SW2, SW3, SW4, PR1 to PR4, RV1, R4, R5. Note that on our prototype unit we used six interlocking push buttons (including one for the 'automatic' tuning mode) to make SW4 and mounted them on their own PCB, together with multi-turn preset pots PR1-4. Connect the unit to the 12 V power supply, connect an antenna (a few feet of wire should be adequate) and connect the audio outputs of the module to a preamp/power amplifier combination.

Switch the unit on, turn SW4 to position five (manual tuning), SW1 (muting) on, SW2 (AFC) off, SW3 (mono/stereo) to mono and check that the unit can be tuned with RV1. Check that interstation noise is suppressed to a reasonable level when mute switch SW1 is on. If necessary, trim the 100k pot on the 7254 module to obtain satisfactory muting. Check that the unit is functional in the four preset positions of SW4. Finally switch SW3 to the stereo mode and check that LED 1 turns on when a stereo station is tuned in and that stereo noise is not excessive. If necessary, trim the 10k pot on the 7254 to obtain the correct stereo operation.

Now make up the 20-LED frequency scale on PCB 'A'. Note that this board uses square LEDs, mounted horizontally. Fit the LEDs in place with great care, ensuring that their front faces are all in line and overhang the edge of the PCB by the thickness of the front panel of your proposed tuner case. When construction is complete, make the connections to the power supply and to pin 1 of the 7254 module. Switch on and adjust PR2 to set 1V75 between point 'A' and ground. Check that the frequency scale can be fully scanned via RV1 in the manual tuning mode, noting that the PCB is fitted upside down in the final tuner case, so that the display moves to he right with increasing tuning voltage and frequency.

Signal Strength Meter

Next, build up the optional LED signal strength meter on PCB 'C', noting that R39 is mounted on the underside of the PCB and that square LEDs are again used. In this instance the LEDs are mounted vertically on the PCB and special care must be taken to establish the LED heights, as follows. Make a mock-up of the signal strength meter section of your final front panel, complete with a cut-out for the 10-LED display, temporarily fit the PCB in its final 'fixed' position behind the front panel, push the two end LEDs into position on the PCB with their faces flush with the front of the panel and then solder into place. Now fix the remaining eight LEDs in place, using a straight edge across the two end LEDs to give the correct height adjustment. When construction is complete, connect the unit to the power supply and to the 7254 module, scan the band in the manual mode and adjust PR3 so that the lowest signal strength LED is barely on with the weakest of input signals; you'll find that only the lowest three to seven LEDs illuminate under most signal conditions.

Auto Search And Lock

The search and lock circuit is an optional 'extra' and requires some care in setting up. Construct the PCB as shown in the overlay, taking care to use good quality (low leakage) tantalum components in the C7 position. Temporarily remove the link from the completed PCB, connect the board into the tuner circuit, TURN THE AFC OFF (SW2), turn SW4 to auto position and switch the unit on. Check that the frequency scale can be scanned up and down using the fast and slow search buttons (PB1-4) and that a selected station remains in tune for a

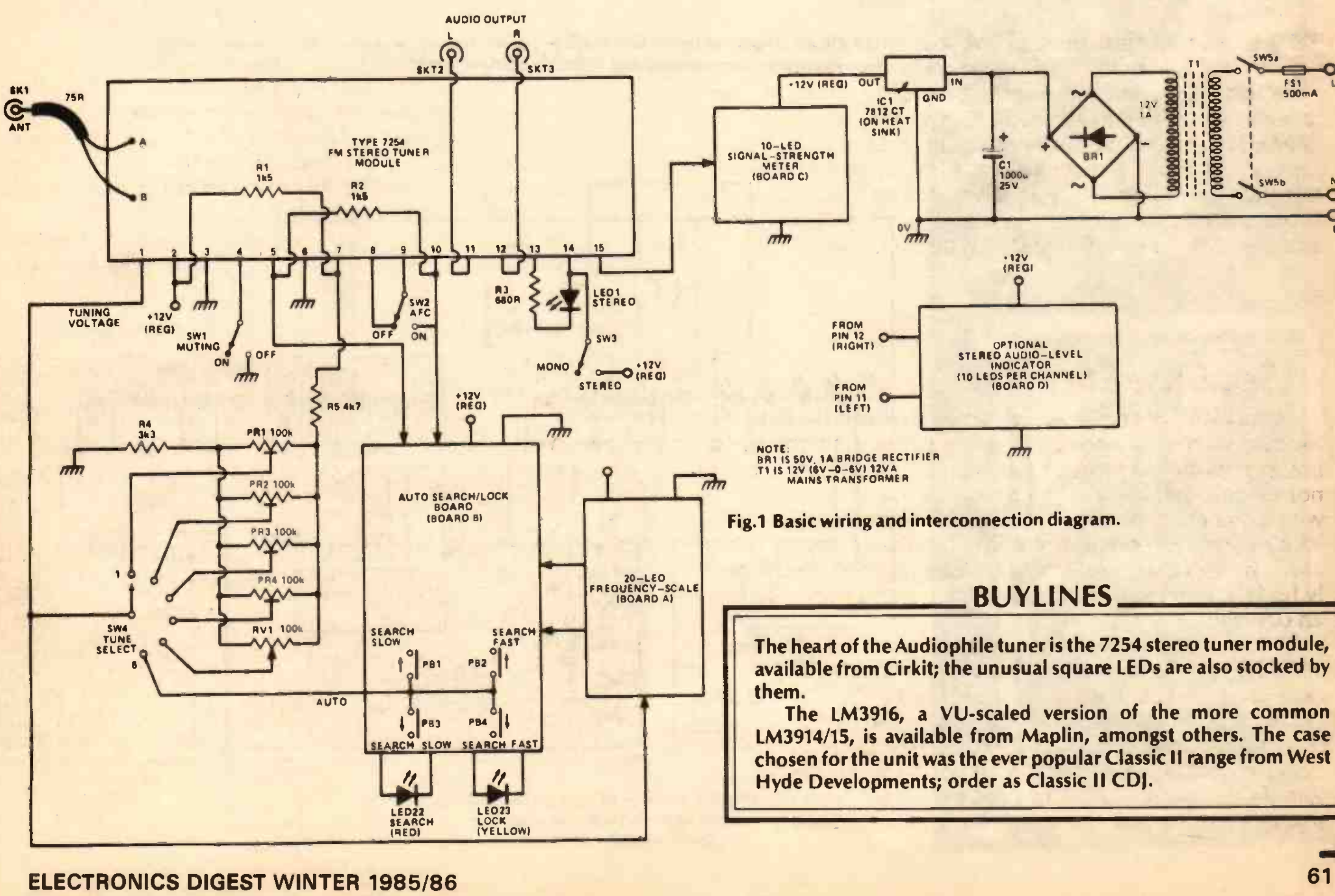


Fig.1 Basic wiring and interconnection diagram.

BUYLINES

The heart of the Audiophile tuner is the 7254 stereo tuner module, available from Cirkit; the unusual square LEDs are also stocked by them.

The LM3916, a VU-scaled version of the more common LM3914/15, is available from Maplin, amongst others. The case chosen for the unit was the ever popular Classic II range from West Hyde Developments; order as Classic II CDJ.

reasonable time (at least a minute) when the buttons are released. If all is well, insert the link on the PCB.

Switch SW4 to the manual tuning position, tune upwards (with RV1) towards a reasonably strong station and very carefully adjust PR2 so that the search LED (LED 22) is normally on, but turns off and switches lock LED 23 on just as the signal strength indicator goes past its peak reading position. Repeat the action several times, checking that the lock LED acts as an effective 'optimum tuning' indicator.

Now switch SW4 to the auto tuning position, drive the tuning scale fully to the left with the PB4 fast left button and then release PB4. The circuit will now start slowly scanning up the band (to the right), looking for a strong station. When a suitable station is located, the circuit will scan fractionally past the peak signal strength position and then lock on, driving on LED 23. The search LED will subsequently give an occasional flash as a correction pulse is generated to maintain correct tuning. If necessary, trim PR2 very slightly to obtain the same action.

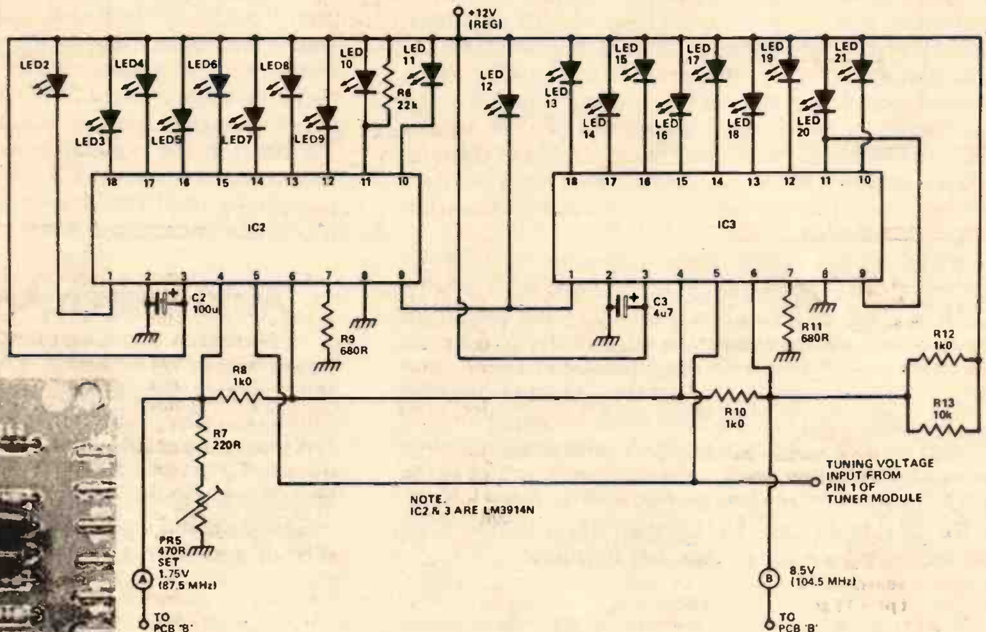
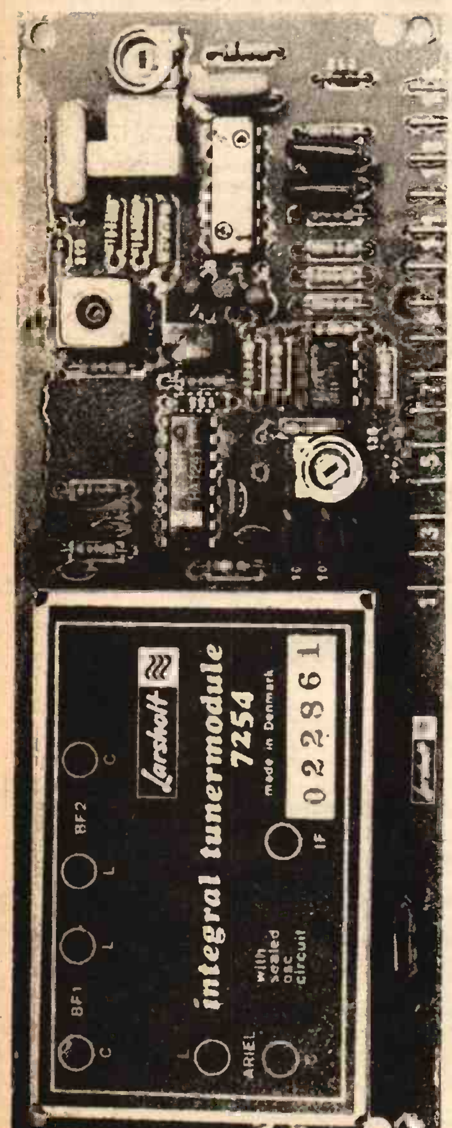


Fig.2 Circuit diagram of the 20 LED frequency scale. This circuit is driven by the tuner module (with its ancillary components) and so the component numbering follows on from the module.

The design is built around the assembled 7254 tuner module (below).



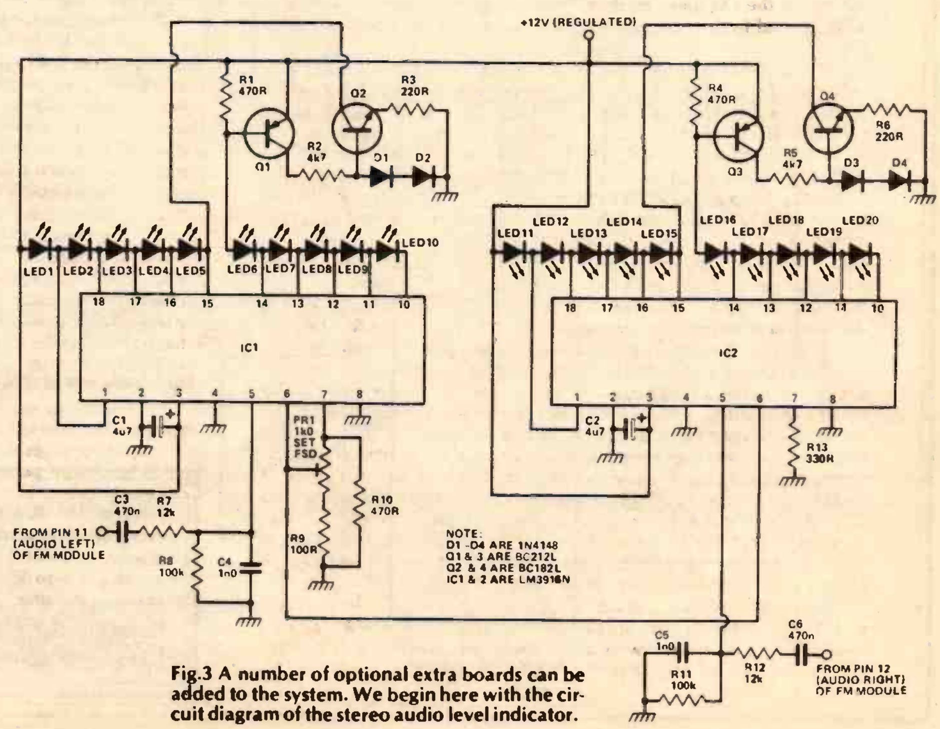


Fig.3 A number of optional extra boards can be added to the system. We begin here with the circuit diagram of the stereo audio level indicator.

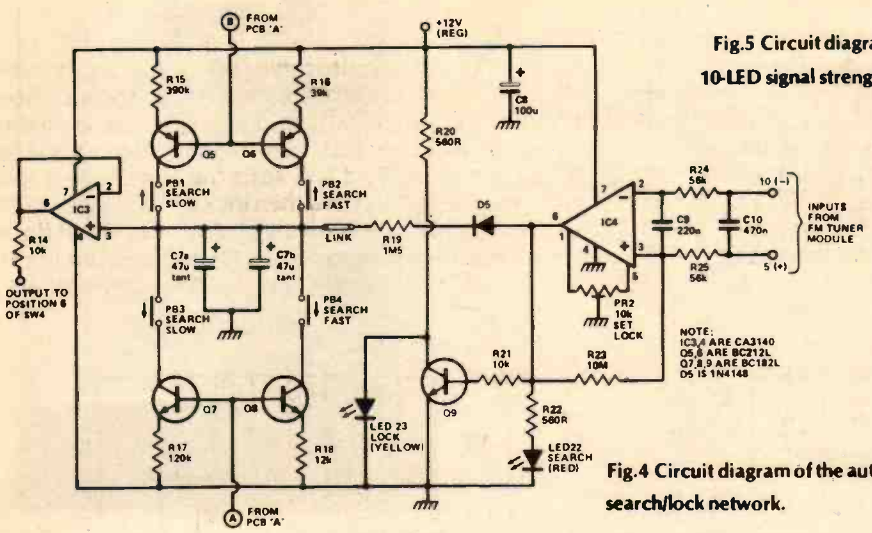
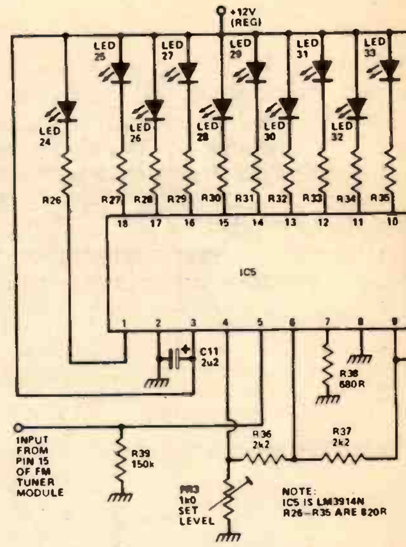


Fig.4 Circuit diagram of the auto search/lock network.

Fig.5 Circuit diagram of the 10-LED signal strength meter.



HOW IT WORKS

The heart of the ETI Audiophile FM Tuner is a ready-built type 7254 FM stereo tuner module with built-in varicap diodes, designed to be tuned with an external DC voltage applied to pin 1. In our application this tuning voltage can be derived from the 12 V regulated power supply line via the R1-R5-PR1-to-PR4-RV1-R4 potential divider or via an auto search-and-lock board. In either case, we use a 20-LED 'voltmeter' circuit to monitor the pin 1 voltage and to thus act as an effective frequency or tuning scale.

The 7254 tuner module has a number of useful output pins. Pin 7 includes an AFC voltage which can be used (via the tuning potential divider) to hold stations on-tune. A voltage proportional to the tuned signal strength is generated at pin 15 and is used in our circuit to drive a 10-LED indicating meter. Tuning meter 'nulling' signals are generated between pins 5 and 10 and are used in our application to control an automatic search-and-lock circuit. Decoded stereo output signals are available at pins 11 and 12 and can be used to drive an optional stereo audio-level indicator as well as for feeding audio signals to an external preamp power amplifier combination. The 20-LED frequency scale is simply an expanded scale LED voltmeter circuit designed around two series-connected LM3914 bargraph ICs operated in the 'dot' mode. The lower and upper limits of the voltmeter scale are determined by fixed reference voltages fed to pin 4 of IC2 (1V75) and pin 6 of IC3 (8V5) respectively via the PR6-R7-R8-R10-R12-R13 potential divider. The input to the voltmeter is fed to pin 5 of the two ICs from the pin 1 tuning voltage of the FM tuner module. This voltage is directly proportional to the tuned frequency, so the voltmeter acts as an effective tuning scale.

The 10-LED signal strength meter is also an expanded scale LED voltmeter designed around an LM3914 bargraph IC, but in this case the IC is operated in the 'bar' mode. The lower and upper limits of the voltmeter scale are determined by fixed reference voltages fed to pins 4 and 6 respectively of the IC. The input voltage to the meter is derived from the pin 15 'signal-strength' output of the FM tuner module.

The stereo audio-level indicators take the form of a pair of virtually identical 10-LED voltmeters with semi-log scales. Each voltmeter is designed around an LM3916 bargraph IC, a VU-scaled version of the LM3914. The full-scale sensitivity of the meters is set at a few hundred millivolts peak (to suit individual tastes) by preset PR1 and audio inputs are fed to pin 5 of each IC from the outputs of the 7254 tuner module via a simple filter network. The audio signals are AC-coupled and the meters respond to the positive halves of the waveform only.

The LEDs are connected to the LM3916 ICs in a most unusual manner in this particular application. The actual ICs are operated in the 'dot' mode, with only one output being low at any moment of time, but the LEDs are connected to each IC in two series-connected groups of five and produce a 'bar' form of display. The purpose of this configuration is to provide current conservation. In a normal 'bar' display, if all 10 LEDs are on and each LED consumes 10 mA, the total current consumption is 100 mA. In the configuration shown in the diagram the currents of each group of five LEDs pass through the series-connected LEDs, so that the circuit consumes a total of only 20 mA when all 10 LEDs are on and consuming 10 mA each.

To understand the operation of the circuit, let's take the example of IC1, remembering that only one output pin can be low at any moment. If pin 1 is driven low by an audio output signal, LED 1 goes on and draws (say) 10 mA from the supply. If pin 18 goes low, series-connected LEDs 1 and 2 both go on and each passes the same 10 mA of current. Similarly, if pin 15 goes low, series-connected LEDs 1-5 all go on and each passes the total of 10 mA that flows into pin 15 of the IC. If pin 14 goes low only LED 6 is turned on by the IC and passes 10 mA into the chip. This current flows via R1 and Q1 base, however, so Q1 conducts and turns on constant-current generator Q2, which, in turn, draws 20 mA (say) through the whole of the series-connected LED 1 to LED 5 'bar'; six LEDs are thus driven on under this condition and the circuit consumes a total of 20 mA. Similarly, if pin 10 goes low the whole LED 6 to LED 10 'bar' is driven on by the chip and the LED 1 to LED 5 'bar' is driven on by the constant-current generator, so all 10 LEDs are on and the circuit consumes a total of only 20 mA.

The search circuit is quite simple. To understand its operation, assume that the link is broken. Capacitor(s) C7 is a low-leakage tantalum type and is used to store a tuning voltage that can be fed to pin 1 of the 7254 tuner module via the IC3 high-impedance unity-gain buffer amplifier and via SW4. Transistors Q5 to Q8 are all wired as low-level constant-current generators and can be used to charge or discharge C7 via 'search' buttons PB1 to PB4 and thus scan the frequency band of the tuner module. When the buttons are released, C7 tries to store the tuning voltage, but in practice the voltage slowly leaks away via the capacitor imperfections and the operating frequency of the tuner very slowly decays downwards. The 'lock' section of the circuit is driven from the pin 5 and pin 10 outputs of the 7254 module. These pins are intended to drive an analogue tuning meter. If you connect a sensitive volt or current meter between these pins in the polarity shown (it is recommended that you do so to check circuit operation) you'll find that the meter will normally give a positive reading, but that the reading will null or go slightly negative when the 7254 module is precisely tuned to a reasonably strong broadcast station. In our circuit, the pin 5 and 10 outputs of the tuner module are fed to voltage comparator IC4 via a simple filter network in such a way that the IC4 output goes high when a station is not present or is off-tune, but switches low when a reasonably strong station is correctly tuned. Thus, when a station is not present the high output of IC4 slowly charges C7 via R19 and D5 and causes the 7254 tuning voltage to slowly increase as the module 'searches' for a station. Under this condition search LED 22 is driven on via R22. Q9 is also driven on via R21 and turns lock LED 23 off. When a reasonably powerful station is located and correctly tuned the output of IC4 switches low, so C7 is no longer charged via R19. D5 is back-biased under this condition, so C7 does not discharge back into IC4. Simultaneously, LED 22 and Q9 are disabled and lock LED 23 is driven on via R20. Once a station is 'locked' in this manner, C7 very slowly discharges via leakage currents, so the tuning voltage slowly decreases. As soon as a station goes fractionally off-tune, however, the IC4 output switches momentarily high and generates a brief 'correction' or 're-charge' pulse which brings the C7 tuning voltage back up to the correct value. This action is indicated by a brief flash of LED 22.

Note in the IC4 comparator circuit that a small amount of hysteresis is provided by R23. Also note that the trip point of IC4 can be varied over a narrow range with PR2, to compensate for slight offsets in the pin 5 to 10 outputs of the 7254 module and allow correct locking to be obtained. If a circuit fails to lock correctly, it may be that the offset of these outputs is excessive. This effect may be overcome by wiring a shunt resistor (value determined by experiment) between the relevant pins of the tuner module.

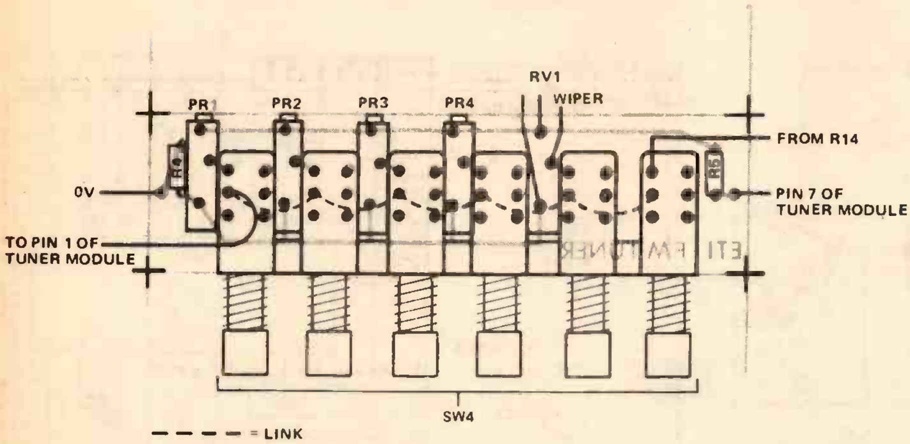
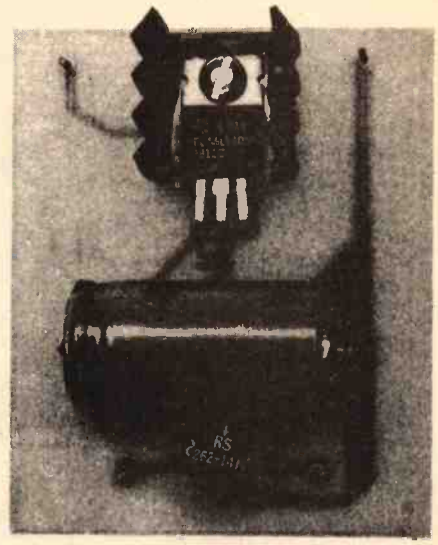


Fig.6 (above) Component overlay of the SW4 six-way switch assembly.



The power supply board — couldn't be simpler.

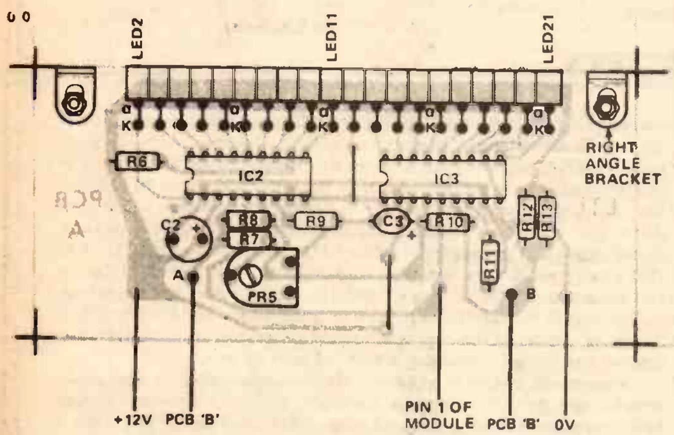


Fig.7 (above) Component overlay of the LED tuning scale.

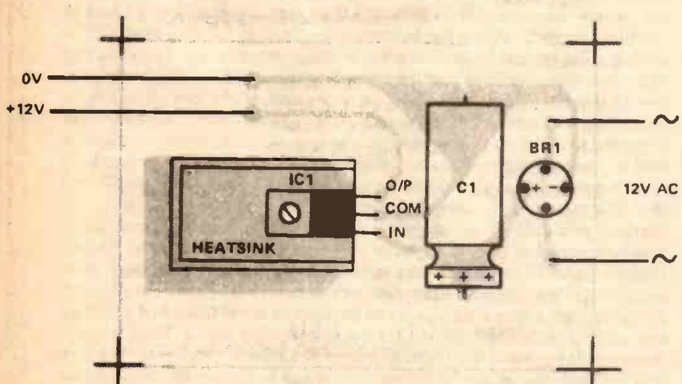


Fig.8 (above) Component overlay of the PSU.

PARTS LIST

FM Module and Tuning Scale

Resistors all ¼ W 5%

R1,2	1k5
R3,9,11	680R
R4	3k3
R5	4k7
R6	22k
R7	220R
R8,10,12	1k0
R13	10k

Potentiometers

PR1-4	100k ¾ in. 20-turn cermet preset
PR5	470R miniature horizontal preset
RV1	100k 10 turn linear

Capacitors

C1	1000u 25 V electrolytic
C2	100u 16 V electrolytic PCB type
C3	4u7 35 V tantalum

Semiconductors

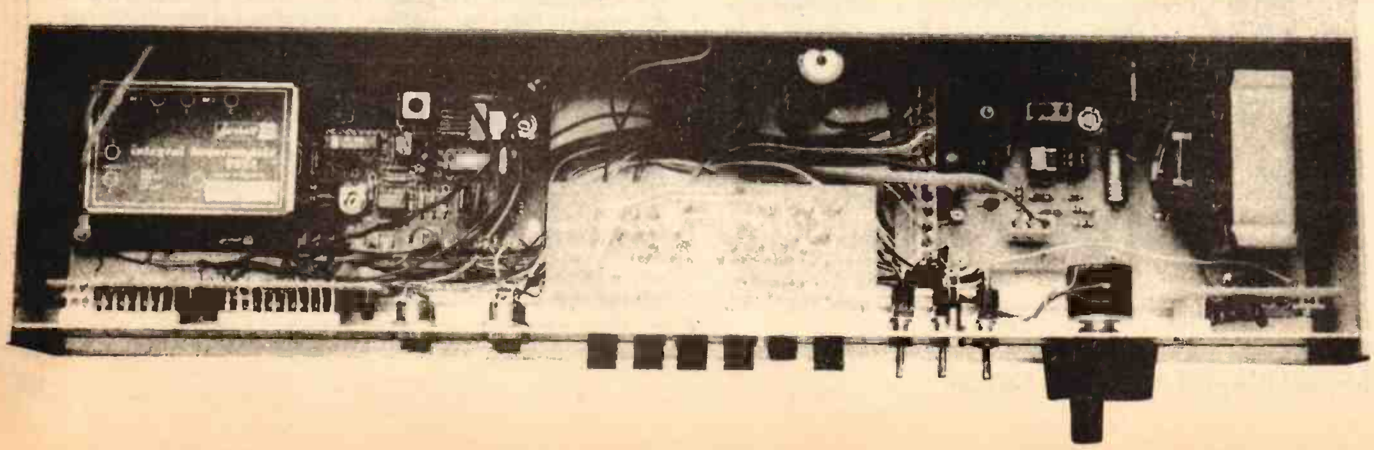
IC1	7812
IC2,3	LM3914N
BR1	50 V 1A bridge rectifier
LED 1	TIL209
LED 2-21	square LEDs (yellow)

Miscellaneous

T1	6-0-6 12 VA transformer
SW1-3	SPDT miniature toggle
SW4	2 pole changeover interlocking push button (6-way assembly)
SW5	DPDT miniature toggle (optional)

7254 FM module (see Buylines), PCB 'A', fuse and holder, phono sockets (X2).

And this is how the whole jigsaw goes together (below).



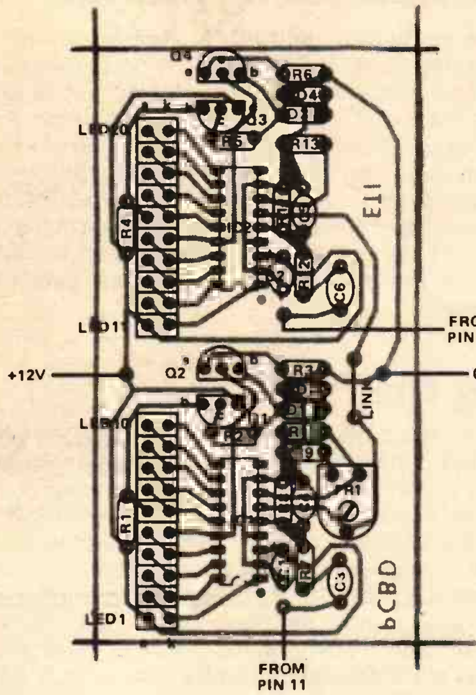


Fig.9 Component overlay of the stereo audio level indicator.

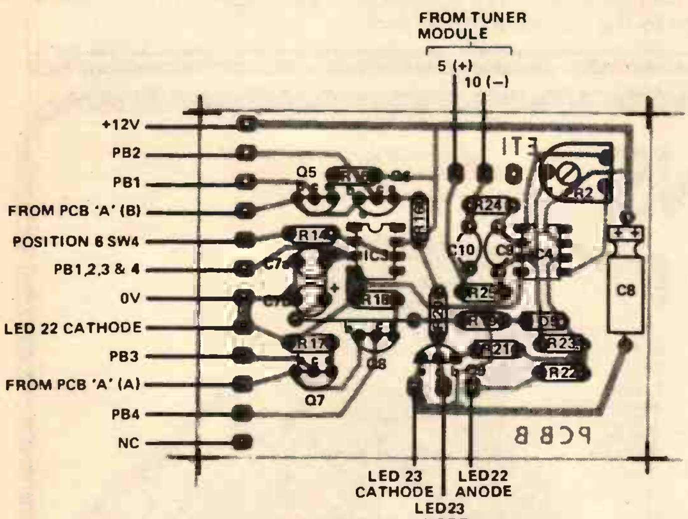


Fig.10 Component overlay of the auto search-lock network.

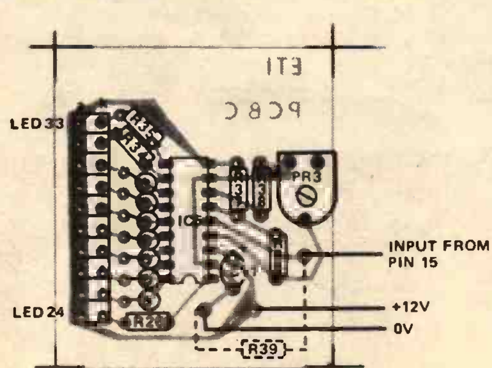


Fig.11 Component overlay of the signal strength meter.

PARTS LIST

Stereo audio level indicator

Resistors all ¼ W 5%	
R1,4,10	470R
R2,5	4k7
R3,6	220R
R7,12	12k
R8,11	100k
R9	100R
R13	330R

Potentiometers	
PR1	1k0 miniature horizontal preset

Capacitors	
C1,2	4u7 35 V tantalum
C3,6	470n polycarbonate
C4,5	1n0 polycarbonate

Semiconductors	
IC1,2	LM3916N (see Buylines)
Q1,3	BC212L
Q2,4	BC182L
D1-D4	1N4148
LED 1-20	square LEDs (red)

Miscellaneous	
PCB 'D'	

Auto search/lock

Resistors all ¼ W 5%	
R14,21	10k
R15	390k
R16	39k
R17	120k
R18	12k
R19	1M5
R20,22	560R
R23	10M
R24,25	56k

Potentiometers	
PR2	10k miniature horizontal preset

Capacitors	
C7a,b	47u 16 V tantalum
C8	100u 25 V electrolytic
C9	220n polycarbonate
C10	470n polycarbonate

Semiconductors	
IC3,4	CA3140
Q5,6	BC212L
Q7,8,9	BC182L
D5	1N4148
LED 22,23	TIL209

Miscellaneous	
PB1-4	momentary push buttons (see Buylines)
PCB 'B'.	

Signal Strength Meter

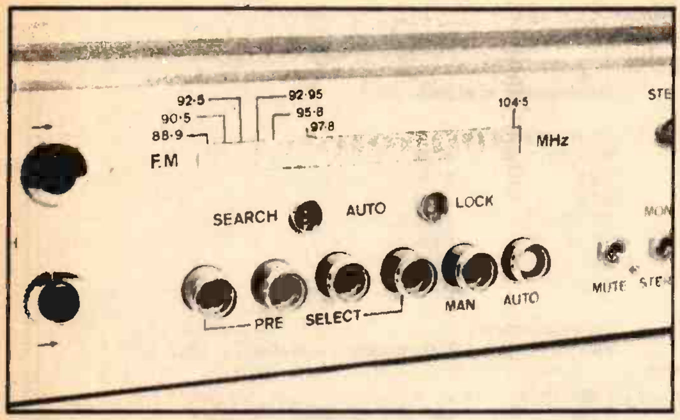
Resistors all ¼ W 5%	
R26-35	820R
R36,37	2k2
R38	680R
R39	150k

Potentiometers	
PR3	1k0 miniature horizontal preset

Capacitors	
C11	2u2 35 V tantalum

Semiconductors	
IC5	LM3914N
LED 24-33	square LEDs (green)

Miscellaneous	
PCB 'C'.	



The tuning scale is mounted above the switch assembly, which takes care of manual and auto timing. The search and lock indicators can also be seen.

When using the auto search-and-lock facility, note that the circuit always scans (searches) to the right and will only fully lock on to signals of reasonable strength (it may temporarily lock to weak signals). To rapidly locate a required station, the search buttons may be used to set the tuning scale slightly to the left of the known position. The search buttons can also be used to unlock from an unwanted station. Also note that the AFC facility must be turned off when the auto search-and-lock circuit is in use.

Stereo Audio Level Indicator

The stereo audio level indicator is an optional item and gives an attractive visual indication of the tuner's audio output signals. The circuit is built on PCB 'D' and uses 10 square LEDs on each channel. These LEDs are mounted vertically on the PCB, their heights being adjusted in the same way as for the signal strength indicator board. When construction is complete, wire the unit into place (connect its 0 V rail directly to the power supply common terminal) and give the unit a functional check. PR1 (on PCB 'D') is simply adjusted so that the display does not run off the scale when strong peak audio signals are present.

Casing The Tuner

When all of the modules have been fully tested on the open bench they can be fitted together in a suitable cabinet, noting that all supply connections must be taken directly to the power supply module (to avoid hum loops, etc.). If you decide to use the 'Classic II' case that we have used in our prototype, note the following constructional points.

The 'Classic II' case is provided with PCB mounting slots and these are used to hold the power supply and the signal strength and audio level boards in place. The basic 7254 tuner module, the SW4 PCB, the search-and-lock board (PCB 'B') and mains transformer T1 are all mounted on the case baseplate with ¼ inch stand-off pillars. The tuning scale (PCB 'A') is mounted on the front panel with angle brackets that are epoxied to the rear face of the panel.

SYSTEM A AUDIO AMPLIFIER

Look no further. This superb amplifier is quite simply the best. Designed to out-perform even commercial equipment, the System A combines ease of construction with Class A quality. Design and development by Stan Curtis.

The initial design brief for this amplifier — 'no compromise' signal reproduction but at the lowest possible cost — proved to be deceptively difficult! The first preamp design eliminated all switches and controls to leave a pick-up input socket, an output socket and a volume control, but such a layout would be far too spartan for even the most serious audio enthusiast. The minimum input requirements were thought to be pick-up, tuner and tape, with tape recorder/monitor output. A stereo-mono switch is unnecessary for serious listening, as are all the other controls that came to mind (except volume and balance!).

The next choice was between discrete or integrated circuits. Despite the obvious benefits and inherent simplicity of IC-based circuitry, I decided upon good old-fashioned transistor stages. Why? Several reasons:

1. If labour costs are disregarded (which they are in this case) the discrete transistor version costs less.
2. Discrete stages can be more easily optimised for a particular design requirement, and give a lower component cost and higher sound quality.
3. There is a purely emotional feeling that when using audio ICs, the designer hasn't really contributed very much to the final design!

(In fact the final circuits are, in effect, discrete component operational amplifiers, so something of the IC design philosophy has obviously rubbed off).

Pick-up An Input

Provision has been made for the preamp to be used with virtually any available pick-up cartridge, through the use of plug-in input circuit boards. Two input circuit boards have been designed although both use the same printed circuit layout. One is for moving magnet-cartridges and the other for moving-coil cartridges. The gain of both these modules can be varied to suit different cartridges by the change in value of a single resistor. Input loading (both resistive and capacitive) can be changed by the substitution of alternative components and, as a source of guidance, a comprehensive table has been produced showing the requirements for the majority of pick-ups currently on sale.

The whole of the preamp design is extremely flexible, permitting alterations to ensure compatibility with other equipment. The basic version has a nominal 775 mV output level and a 75 R output impedance.

PSUing Quality

The power supply is built into a separate case to achieve better screening as well as increasing the versatility of the system. This new 'Audiophile' system is conceived as a modular 'building block' concept offering a variety of facilities.

Provision has been made on the main PCB for the fitting of an output coupling capacitor (C15). Normally this shouldn't be necessary and the two pads should be joined by a wire link to couple the output directly to the power amplifier. A very small number of power amplifiers are totally DC-coupled, so any DC voltage on their input terminals would result in an unacceptable DC offset across the loudspeaker. In such a situation the capacitor should be fitted. Its value can be selected to suit the input impedance of the power amplifier; a value of 3u3, 35 V (tantalum) is acceptable with a 10k input impedance and 470n with a 50k input impedance. The capacitor polarity should be aligned to correspond to the residual DC offset at the output of the preamplifier.

Construction

Although no metalwork plans have been provided it will be seen that the prototypes have been housed in a simple, compact, and functional case consisting of an aluminium chassis and a substantial steel cover.

TABLE 1. SPECIFICATION

PREAMP		
Rated output level:	775 mV (0 dBm)	
Maximum output level: (20 Hz to 20 kHz)	7V8	
Total harmonic distortion (including noise)		
Auxiliary input, 775 mV output	20 Hz	0.01%
	1 kHz	0.01%
	20 kHz	0.01%
Pick-up input 1V5 output	20 Hz	0.02%
	1 kHz	0.02%
	20 kHz	0.02%
Pick-up input overload (ref rated input at 1 kHz)		
	Moving Magnet	Moving Coil
20 Hz	43 dB	44 dB
1 kHz	43 dB	40 dB
20 Hz	43 dB	32 dB
Input sensitivity (ref 775 mV output at 1 kHz)		
Auxiliary	65 mV	
Pick-up (moving magnet)	2.3 mV	
Pick-up (moving coil)	550 mV	
Noise level, 'A' weighted (ref 775 mV output at 1 kHz)		
Auxiliary	−90 dBA	
Pick-up (moving magnet)	−80 dBA	
Pick-up (moving coil)	−76 dBA	
Channel separation, pick-up input (unused channel loaded)		
	1 kHz	62 dB
	20 kHz	69 dB
RIAA equalisation accuracy: (20 Hz to 20 kHz)	±0.2 dB	
Frequency response: (auxiliary input)	±0.5 dB, 5 Hz to 35 kHz	

The above figures are for the standard version. The performance of the alternatives will vary in terms of sensitivity etc.

POWER AMP	
Biasing mode:	Class A
Rated power:	60 W RMS into 8R, 20 Hz to 20 kHz
Transient delivery:	150 W into 8R

Harmonic and intermodulation distortion: less than 0.06% at rated power output (20 Hz to 20 kHz), decreasing monotonically with decrease in power. Distortion is virtually unmeasurable at small signal levels.

Frequency response: (ref 0 dB at 1 kHz)	10 Hz — 1 dB 120 kHz — 6dB
Power bandwidth:	5 Hz to 60 kHz
Hum and Noise:	100 dB below 24 V RMS output (CCIR)
Sensitivity:	700 mV RMS for 60 W into 8R

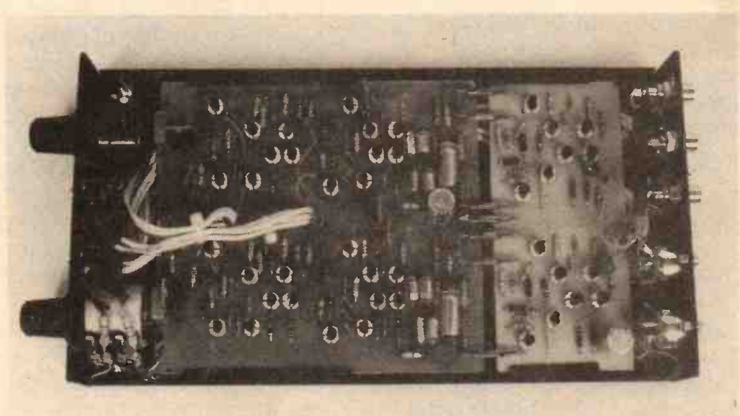
Negative feedback: the open loop gain is reduced by 22 dB by the application of overall negative feedback.

Transient intermodulation distortion: zero

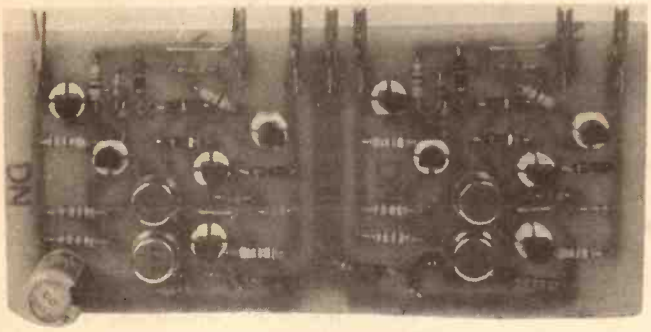
The preamplifier circuitry has been constructed on two printed circuit boards which plug together using high quality gold-plated connectors. The construction of these boards should present no difficulties if the layout is followed correctly. There is a certain amount of wiring using screened cable and it is essential that this be done neatly and correctly. A wiring diagram has been given which shows the loom in detail and this arrangement should be followed fairly closely. The ends of all screened cables should be sleeved to avoid the danger of stray strands shorting out the signal. Particular attention is drawn to the earth connections which are always a problem with stereo amplifiers. The arrangement as drawn works. Others might not! You may wonder why this wiring has not been incorporated on the PCB. This could have been done for ease of assembly but only at the cost of the loss of isolation between the various signal and supply paths. In this context it is interesting that one of the world's best regarded preamps, the ultra-expensive Levinson, uses several hundred dollars' worth of military grade, PTFE-insulated screened cable in the pursuit of signal isolation. However, our budget model uses common-or-garden screened cable to do the same thing! The use of this cable plus some care in layout results in a quite respectable figure for stereo separation at high frequencies.

It is recommended that the phono sockets for the pick-up inputs be gold-plated. These are expensive and difficult to obtain but, for optimum results to be obtained, they must be used. I have undertaken a lot of research into the effects of signal connections and have found that, while in theory both the gold-plated and nickel-plated contacts give equally good connections, in practice and over a period of time the gold-plating will prove its worth. I will say no more because a full summary of the problems associated with connectors would fill an article of its own.

Most of the transistors used are uncritical and the recommended types can often be substituted for, provided that due regard is paid to voltage ratings and so on. However, the 2N4401 first stage transistors are notably quieter than many alternative 'low noise' types (BC109 etc) and these should be fitted. The input transistors (Q1 and Q2) used for the moving-coil stage (module A-MC) are medium-power devices selected from the BC160 family. They are tested for low noise under the specified operating conditions. Transistors of this type could be fitted on a 'pot-luck' basis but this may lead to disappointment, frustration, and a need for a new nozzle on your solder sucker!



Inside the prototype preamplifier. Construction is on two boards, the main preamp module A-PR and the smaller input module. The latter is connected to the main board and the phono input by gold-plated connectors. This enables different input modules to be easily exchanged to match different cartridges. If you're *certain* you'll only ever be using one cartridge, you could dispense with the connectors and solder wire links instead.



This photograph shows the A-MC coil input module.

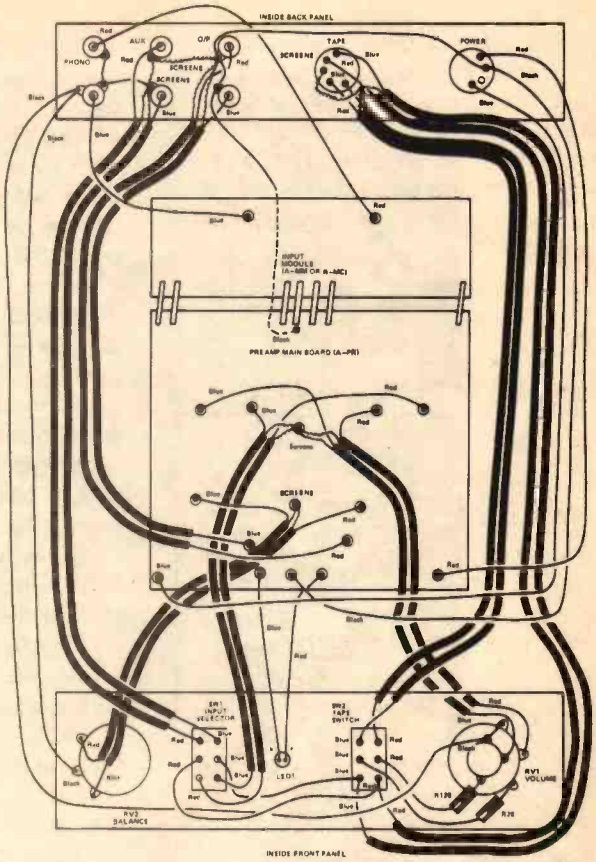


Fig. 8 Wiring diagram of the preamp. No wiring from the preamp main board crosses the input module; all cables are taken towards the front panel and back down either side of the case to the rear panel. See photos.

PARTS LIST

INPUT MODULE A-MM

Components are listed for one channel only — add for other channel.

Resistors (all ¼ W, 5%

R1		100k (see text)
R2		100k
R3, 4		6k8
R5		2k7
R6		560R
R7		3k9
R8		2k2
R10		220R
R11, 12		47R
R13		3k3
R9 is not used		

120

Capacitors

C1	120	10u 35 V tantalum
C2	48	1000u 16 V electrolytic (PCB type)

Semiconductors

Q1, 2	120	2N4401
Q3, 4	80	BC107 or similar
Q5	40	2N4403
Q6	60	MPSA06
Q7	60	MPSA56
D1-5	40	1N4148 or 1N914

Miscellaneous

Connectors, PCB.

INPUT MODULE A-MC

Components are listed for one channel only — add 100 for other channel.

Resistors (all ¼ W, 5% except where stated)

R1	100k (see text)
R2	1k0
R3, 4	1k2
R5, 9	270R
R6	2R2 2% metal film
R7	3k9
R8	56R
R10	220R
R11, 12	47R

Capacitors

C1	100u 6V3 tantalum
C2	1000u 16 V electrolytic (PCB type)

Semiconductors

Q1, 2	BSS15 (specially tested — see text)
Q3, 4	BC107 or similar
Q5	2N4403
Q6	MPSA06
Q7	MPSA56
D1-5	1N4148 or 1N914

Miscellaneous

Connectors, PCB.

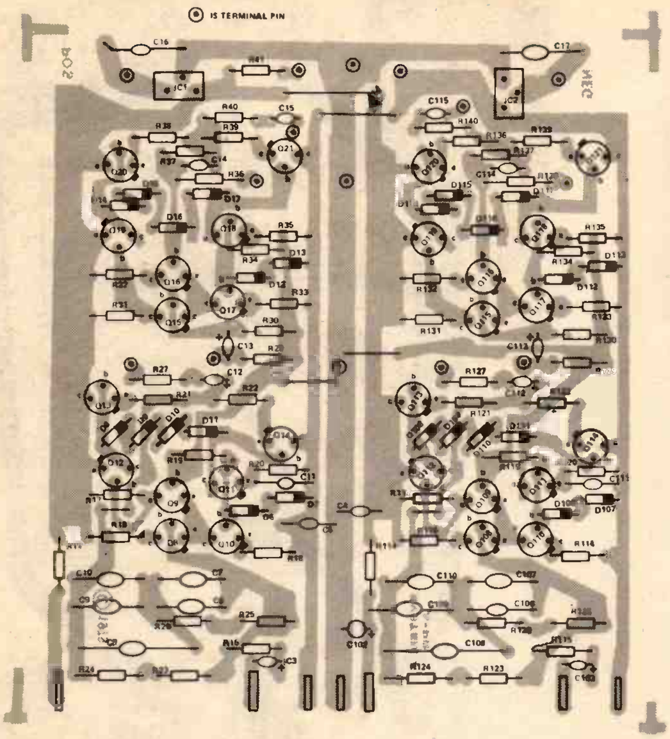


Fig. 9 The A-PR overlay. For off-board connections see Fig. 8.

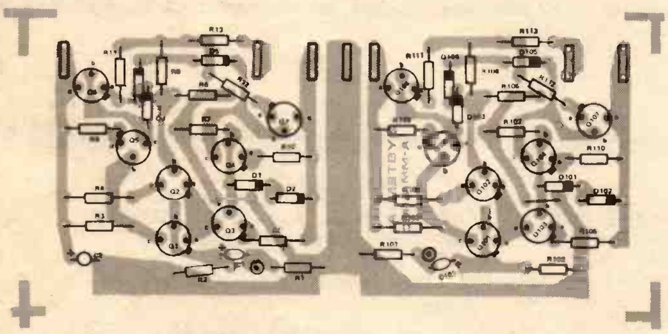


Fig. 10 Overlay for both phono input modules. Note that R9, R109 are replaced by wire links in the A-MM module.

Testing

The power supply should, because of its simplicity, present few difficulties. Before any connection to the mains supply, a visual inspection should be made to check the wiring, the polarities of the capacitors and rectifier, and not least the wiring of the mains switch. It never ceases to amaze me just how often mains switches are wired to short across the supply at switch-on. So take a little care and save a few bob!

With all checks completed, the fuse is fitted and a meter wired between the positive and negative output lines. The mains supply can be connected and for a 240 V nominal supply the meter should read 21 V (±2 V). Then measure the supply to 0 V to check that they are equal and that the LED is illuminated.

The preamplifier is fairly straightforward to test, albeit rather repetitive. The two power supply regulators are protected against excessive currents (eg shorts) and overtemperature, so they are unlikely to come to any grief providing they are correctly inserted into the PCB. Each of the amplifier stages on the main board can be isolated from the power supplies by the removal of wire links and, of course, the input module can be unplugged, so in the event of a fault the offending stage can be isolated.

PARTS LIST

PREAMP MODULE A-PR

Components are listed for one channel only — add for other channel.

Resistors (all ¼ W, 5% except where stated)

R14	39R
R15, 19, 34	3k3
R16, 17, 31, 32	6k8
R18, 33, 28	2k7
R20, 35	220R
R21, 22, 38, 39	68R
R23	330k 2% metal oxide
R24	2k0 2% metal oxide
R25	24k 2% metal oxide
R26	2k7 2% metal oxide
R27	47R
R29	150k
R30	100k
R36	330R
R37	10k
R40	75R
R41	4k7

Potentiometers

RV 1	50k logarithmic
RV 2	1k0 linear (preferably wirewound)

Capacitors

C3, 12, 13	10u 35 V tantalum
C4, 5, 11	100n 63V ceramic disc
C6, 7, 9	1n5 2% polystyrene
C8	6n8 2% polystyrene
C10	560p 2% polystyrene
C14	100p ceramic
C15	see text

Semiconductors

IC1	7815
IC2	7915
Q8, 9, 15, 16	2N4401
Q10, 11, 17, 18	BC107 or similar
Q12, 19	2N4403
Q13, 20	MPSA06
Q14, 21	MPSA56
D6-17	1N4148 or 1N914
LED1	TIL209 or similar

Miscellaneous

SW1, 2	DPDT slide switch

Connectors, PCB, phono sockets, DIN sockets, Veropins, screened cable, case, knobs to suit.

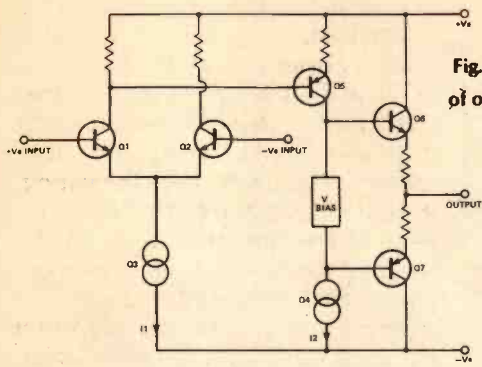


Fig. 1 Simplified diagram of one gain stage.

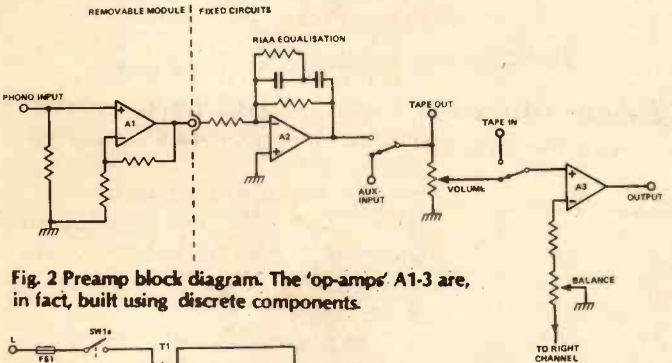
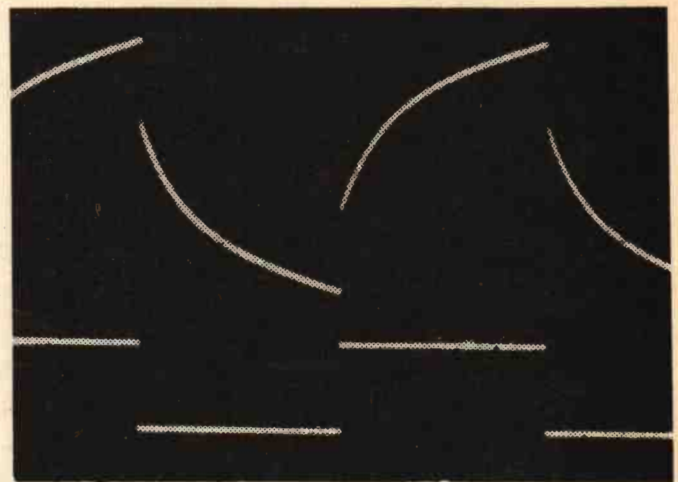


Fig. 2 Preamp block diagram. The 'op-amps' A1-3 are, in fact, built using discrete components.



Response of the series feedback equalisation stage to a square wave input signal.

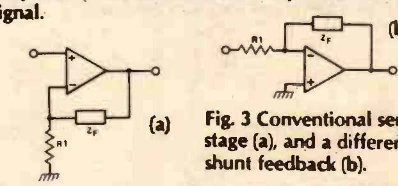


Fig. 3 Conventional series feedback equalisation stage (a), and a different configuration using shunt feedback (b).

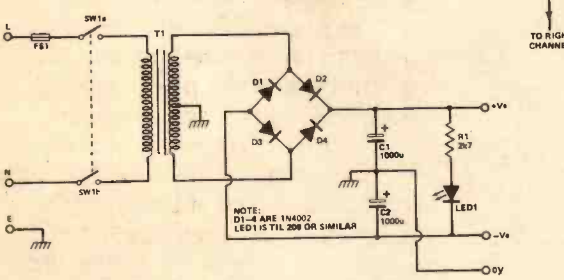


Fig. 4 Circuit diagram of the A-PSU preamplifier power supply.

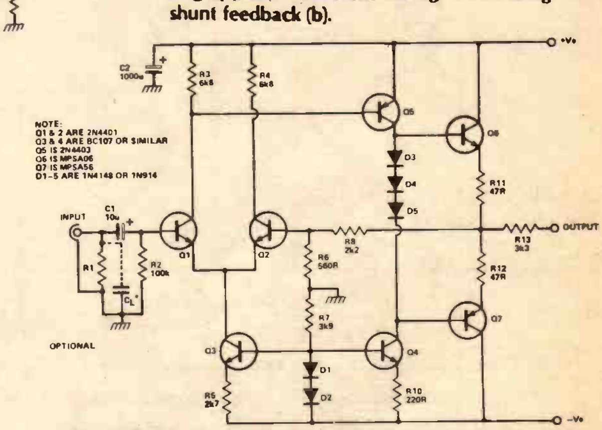


Fig. 5 Circuit diagram of the A-MM moving magnet module.

HOW IT WORKS

Each stage of the preamplifier uses a virtually identical discrete component operational amplifier. This is shown in simplified form in Fig.1. The input stage is a long-tailed pair composed of transistors Q1 and Q2 whose collector current is determined by a constant current source (Q3) and works out at about 100 uA for each transistor. This current has been chosen to give a low noise figure for this stage. The second stage is a voltage amplifier (Q5) which drives a constant current load (Q4) to set the standing current of this stage at about 2mA. The four series diodes bias on the complementary output stage (Q6,Q7) to give a quiescent current of 8 mA. This value of standing current ensures that all the amplifier stages continue to operate in the linear Class A region even when driving low impedance loads.

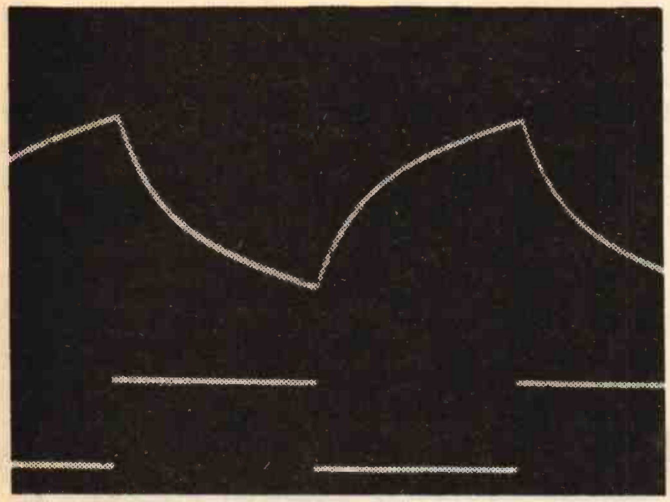
The moving-coil stage is virtually identical to the other op-amps except for the use of some different component values. Whereas the other stages are optimised for low noise when driven from medium impedance signal sources, the moving-coil cartridge can represent an almost pure resistance of between 2 and 10 R. To achieve a better noise figure medium-power transistors are used in the input stage, and each is operated at a collector current of slightly over 1 mA.

The three stages are arranged as shown in the system block diagram (Fig.2). The first stage can be either a moving-magnet or a moving-coil stage. Whichever is chosen, the gain and input loading are optimised to suit the pick-up cartridge in use. This stage has a flat frequency response and no feedback equalisation. It does, therefore, buffer the cartridge from the equalisation stage and so ensures that the cartridge loading is not frequency-dependent.

The second stage is the equalisation stage with the RIAA network wired in a shunt feedback arrangement. This stage has a voltage gain of 20 dB (x10) at 1 kHz and brings the signal level up to a nominal 50 mV before the switching circuits. After the volume control comes the third stage (A3) which is wired as a simple 20dB (x10) line amplifier. However, the feedback resistor is wired to ground through a potentiometer which acts as a balance control, giving a gain variation of 11dB on this stage.

Shunt Feedback

The purpose of the equalisation stage is to provide a fixed degree of frequency de-emphasis exactly complementing the RIAA specified pre-emphasis applied when a record is cut. Although the equalisation is normally specified over the band 20 Hz to 20kHz it was assumed that the response curve would be continued outside of the audio band. Most important, the replay response above 20kHz should continue to reduce with frequency until at some infinitely high frequency the output is zero. This requirement is disregarded by most audio engineers who concentrate primarily on the audio band performance, but the music signal reproduced from a disc contains transients whose frequency content can lie outside the arbitary audio band. (Question: why 20 Hz to 20 kHz? Answer: because it has always been so!) The conventional series feedback stage of Fig.3a is unable to provide an accurate transfer of these high frequencies. This is because the gain does not drop towards zero with increasing frequency but towards unity. The voltage gain of this stage is equal to 1 + (Z_F ÷ R1); so even if Z_F is made infinitesimally small the minimum gain cannot be less than unity. The same is not true of shunt feedback equalisation stage such as the one shown in Fig. 3b. Here the voltage gain is equal to Z_F/R1, so that as Z_F continues to reduce so the gain continues to drop until finally the minimum gain is determined by the signal leakage through the stage. The accompanying photos show the reproduction of a square wave through the two types of equalisation stage and it will be clearly



The response of the same stage when wired for shunt feedback.

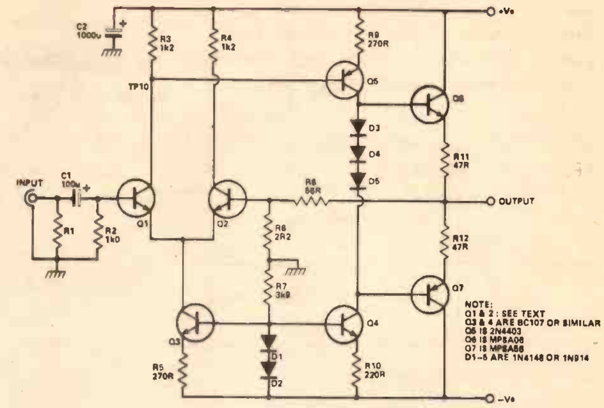


Fig. 6 Circuit diagram of the A-MC moving coil module.

2N4401 → BC637
4403 → BC638
BC 141 → 06
BC 161 → 56

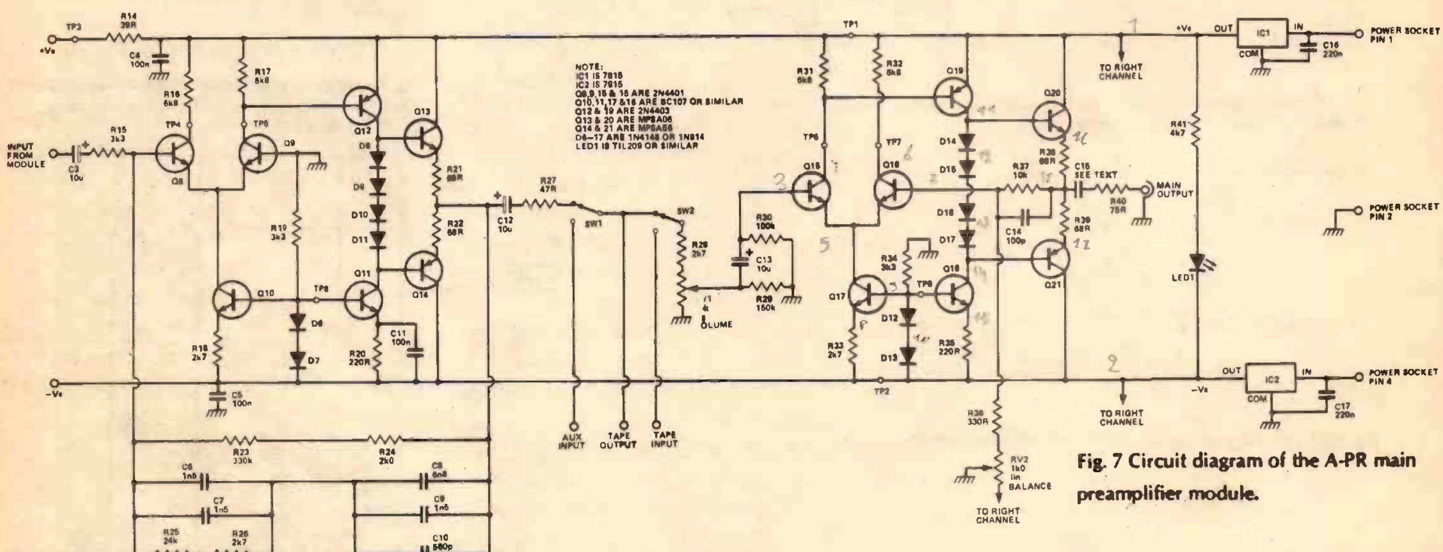


Fig. 7 Circuit diagram of the A-PR main preamplifier module.

HOW IT WORKS

seen that the series feedback arrangement imparts a degree of treble boost to the signal.

So why isn't the shunt feedback system commonly used in commercial pre-amplifiers? The answer is noise; to be exact, the noise generated by the series input resistor R1. Both input configurations use a nominally 47k resistor to load the cartridge, but in the series arrangement it is 'shorted-out' by the (approximately) 200R resistance of the cartridge. However, with the shunt arrangement this 47k resistor remains in series with the signal path and hence contributes a lot of Johnson (thermal) noise. It has been calculated that the maximum theoretical signal-to-noise ratios of the two stages (measured over the band 20 Hz to 20kHz and RIAA equalised) are:

Shunt feedback 58.5 dB

Series feedback 72

Both ref. 2 mV at 1 kHZ

This difference is enough, in our world of specmanship, to have consigned the shunt feedback stage to the dustbin for many years.

However, to get the best of both worlds I have gone back to the system I used many years ago at Cambridge Audio. This is the use of a linear series feedback input stage followed by a shunt feedback equalisation stage. The equalisation stage can now work under far easier conditions as the signal has some initial preamplification. Furthermore the input resistor (R1) now no longer needs to be 47k but can be a lower value chosen to set the stage gain. In this case it has been set at 3k3 and so its noise contribution is quite low.

Now we have an input arrangement which buffers the cartridge from the equalisation stage (and so makes the input loading indepedent of equalisation), continues the RIAA equalisation curve at high frequencies, and achieves the low noise figures typical of the conventional series feedback arrangements. Just as important, the shunt feedback sounds different (and in my opinion better), and that is the deciding factor. A revealing experiment is to wire one preamplifier in shunt and one in series feedback and (having equalised their gains) to listen to each in turn reproducing the 'off-record' noise. It will then be apparent that some preamplifiers emphasise such noises more than others.

Power Supply

The power supply circuitry is kept simple and consists of two integrated circuit regulators (IC1,IC2) which give a low ripple +15V supply to the circuits. The positive rail is further decoupled at the pickup stage by resistor-capacitor filters (R14,C2). The negative rail is adequately decoupled for this stage as the long-tailed pair(Q1,Q2) is fed through a current source, but the positive rail is connected directly to the collectors of this stage and so some additional decoupling is required. The decoupling capacitor needs to be of quite a high value to maintain a low impedance supply. If this value is reduced the low frequency distortion can become excessive.

The supply indicator LED is wired across both supply rails so that the absence of either one will cause the LED to go off.

The power supply module is also simple. The incoming mains supply is fused and switched and fed to a toroidal transformer. The centre-tapped secondary feeds a bridge-rectifier to produce a split rail supply across the two reservoir capacitors (C1,C2). The off-load voltage at this point should be a nominal ± 21V. Again the supply indicator LED is wired across both rails as a monitor.

Table 2. Voltages measured between test points and ground with Avometer Model 8. These voltages should be taken only as a guide.

TP1	+15V	TP6	+14V3
TP2	−15V	TP7	+13V6
TP3	+14V5	TP8	−13V8
TP4	+13V6	TP9	−13V8
TP5	+14V3	TP10	+13V

ADC	
ZLM	E
XLM−II	G
XLM−III	E
VLM−II	E
AKG	
P7E	G
P8E, P8E−S	E
AUDIO TECHNICA	
AT−10	F
AT−11E	F
AT−12E	F
AT−13EA	E
AT−25	B
AT−30	C
Signet MkIIIE	C
Signet TK5E	F
Signet TK7E	E
AZAK	
DC2100K	C
BANG & OLUFSEN	
MMC20CL	E
CORAL	
MC81	C
MC88	E
777E	C
777EX	C
DECCA	
Blue	E
London	F
DENNON	
DL103C	A
DL103S	A
DL103D	A
DYNAVECTOR	
10XII	E
20AII	E
ELITE	
MC555	B
EE1500	G
EMPIRE	
500D	E
2000 1E	G
2000 E4	G
600LAC	E
2000 X	E
EMT	
XSD15	A
ENTRE	
Entre 1	C
GOLDRING	
G900 IGC	F
GRADO	
FTE+1	G
KOETSU	
Koetsu	C
MAYWARE	
MC3L	E
MC2C	C
MICRO ACOUSTICS	
2002−E	E
NAD	
9000	E
NAGAOKA	
JT−R11	B
ORTOFON	
MC10	C
MC30	C
VMS 20E	E
REGA	
100	E
SHURE	
75−ED	H
M75EJ	H
M97HE	H
V15−IV	G
SONUS	
Blue	E
Gold Blue	E
SONY	
XL35	E
XL55	A
STANTON	
680	E
680EE	E
881	E
681EEE	E
SUPEX	
SD9015	E
900E	C
TECHNICS	
EPC205−C	E
EPC−300MC	D
ULTIMO	
10XII	E
20A	G
DV20C	A
DV KARAT	A

Table 3. Cartridge matching table.

Before connecting up the power supply it is a good idea to give the PCBs one final visual check, paying particular attention to transistor types, diode and capacitor polarities, and solder bridges on the PCB tracks. Now connect the power supply and monitor the supply lines. They should measure ±15 V (±0V6) and the LED should light up. The controls should now be set as follows;

Input:	PU
Tape:	OFF
Balance:	Central
Volume:	Minimum

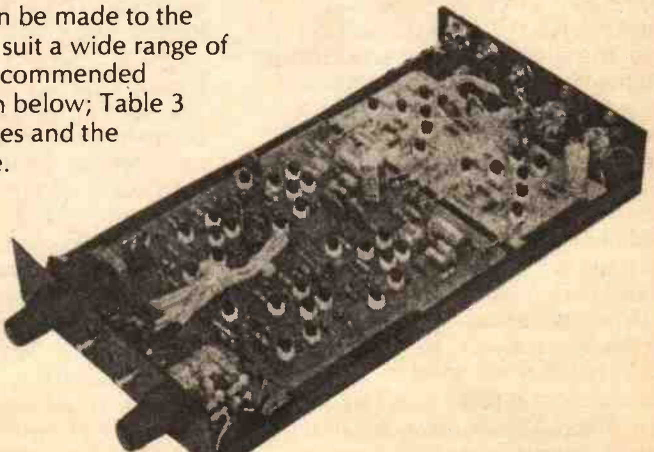
Now measure the DC voltage between earth and the junction of the two emitter resistors in the output stage of each amplifier. This voltage should be zero, but can be ± 2 V without any significant effect on the workings of the preamplifier (although the blocking capacitor will be necessary). That completes the DC tests. The preamplifier will now almost certainly work but if you have test equipment available it would be a good ideal to test each channel with an audio signal and to centralise the balance control.

The total current drawn from the negative supply is about 120 mA for the moving-coil version and 115 mA for the moving-magnet version; and about 15 mA less from the positive supply.

As an aid to fault-finding a list of test-voltages has been provided which can be used in conjunction with the main circuit diagram (Table 2).

Variations On A Theme

Alterations can be made to the input modules to suit a wide range of cartridges. The recommended changes are given below; Table 3 lists most cartridges and the matching module.



Moving-coil Cartridges

The gain of the A-MC input module can be varied by changing resistor R6. This resistor has a value of 2R2 to give a sensitivity of 550 uV on the standard version. Changing R6 to 0R6 (eg two 1R2 resistors in parallel) will increase the sensitivity to about 150uV. The input loading can be varied by changing resistor R1 from the standard value of 100R to any other value. The four recommended alternatives are:

A	550 uV sensitivity,	R1 = 1k0
B	150 uV sensitivity,	R1 = 100R
C	550 uV sensitivity,	R1 = 100R
D	150 uV sensitivity,	R1 = 1k0

Moving-magnet Cartridges

Again, the input loading of module A-MM can be changed by using an alternative value for resistor R1. An input capacitor C_1 can also be wired across R1 to lower the input impedance at high frequencies and so 'equalise' the output from some cartridges. The gain of the standard version is set by R13 and gives a sensitivity and vice-versa. The four recommended alternatives are:

E	Standard version
F	R13 = 8k2
G	C_1 = 180pF
H	R13 = 8k2 and C_1 = 180pF

Table 3 assumes that the cartridges are mounted in tone-arms which have a total cable capacitance of about 100pF and below.

Class A Power

There is one amplifier configuration that is universally accepted as the ideal for audio use: Class A operation. Many early amplifiers operated in Class A, but as output powers rose above 10 W the problems of heat dissipation and power supply design caused most manufacturers to turn to the simpler, more efficient Class B arrangements and to put up with the resulting drop in perceived output quality.

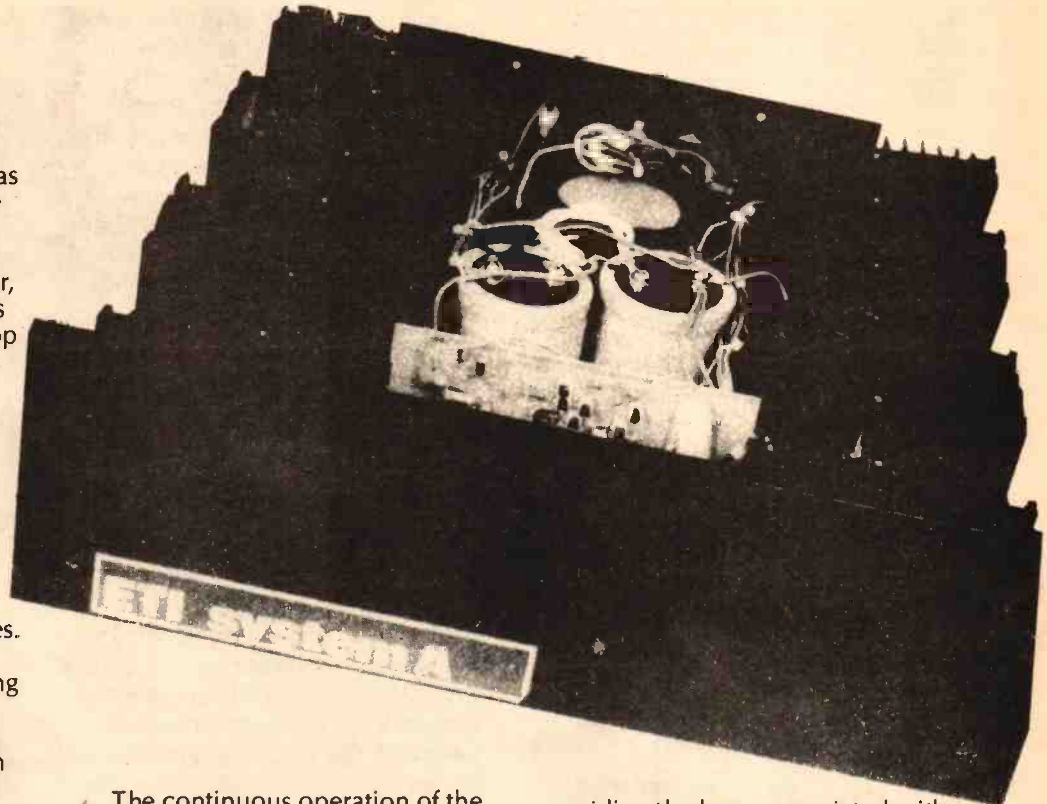
The System A applies the unchallenged excellence of Class A operation to the design of a reference amplifier free of the aberrations of commercially available models. Class A biasing is recognised as the ideal operating mode for an amplifier, offering the uncompromising accuracy demanded by dedicated audiophiles. The superiority of this amplifier depends on the output devices being constantly operated in their linear region, above cut-off and below saturation. Such operation results in the smoothest transfer function and the widest bandwidth.

The System A amplifier has a clarity and a tonal response that produces a superior perspective of depth with a sense of reality: instruments appear in precise position out of a silent background. The musical 'naturalness' of this amplifier is due to its lack of the constrictions of commercially desirable circuitry and the single-minded approach to a no-compromise sound quality.

Why Class A

The amplifier has an excellent technical performance even when operated in the conventional, but less desirable AB mode. With an open-loop (ie no overall negative feedback) distortion of around 0.1% (1 kHz) and a frequency response stretching well outside the audio band, the use of the large amounts of negative feedback (found in most commercial competitors) is completely unnecessary. However, extensive correlation between measurements and subjective performance using a wide variety of amplifier types led to the conclusion that Class A biasing is the optimum for audio amplifier performance.

When biased to Class A, the transistors are always turned on, always ready to respond instantaneously to an input signal; Class B and AB output stages require a microsecond or more to turn on. Thus Class A operation permits cleaner operation under the high-current slewing conditions that occur when transient audio signals are fed into difficult loads.



The continuous operation of the output stage in the linear collector region results in a more desirable distribution of distortion harmonics than is possible in Class B or AB because the non-linearites in the transfer curve are smoother and free of the abrupt transitions of Class B and AB. The gradual non-linearities resulting from Class A operation produce distortions of low orders; primarily second and third harmonics. These lower order harmonics tend to be far less offensive to the ear than high order harmonics, being far more musical in nature (they are predominant in the harmonic spectra of most musical instruments). Higher order harmonics tend to 'harden' the overall sound. Such is the linearity of the Class A Amplifier that a mere 22 dB of gain reduction is made in the form of negative feedback.

Each amplifier is a completely separate self-contained mono unit. The use of mono amplifiers, while costly in terms of components, provides the maximum stereo signal separation under dynamic operation with complete freedom from cross-modulation effects, giving an improvement in subjective depth and accurate instrument imaging.

A glance at the photographs will also explain why each amplifier is made as a mono-block. A stereo version would be just too heavy, unwieldy, and hernia-inducing for even the most dedicated audio fanatic (but if you know different...). Ideally each power amplifier can be located next to its respective loudspeaker and connected to it by very thick but short leads, thereby avoiding the losses associated with loudspeaker cables (30 A cable is suitable).

Protection — A Racket?

This Class A power amplifier is totally free of the usual protection circuits with their unavoidable colourations, distortions, and current-limiting characteristics. Instead we use an output stage having an exceptional power capability for an amplifier of such a low rating. With its substantial heatsinking this amplifier is capable of sustained operation with difficult loads.

The System A amplifier maintains complete control over the driven loudspeaker throughout its operating cycle. The true Class A operation avoids the inherent phase irregularity and inadequate current-sinking ability of comparable Class B and AB designs. The provision of an extremely low-impedance power supply gives the Class A amp a short-term current delivery and, equally important, current-sinking capability far in excess of any known Class AB power amplifier of similar rated output power.

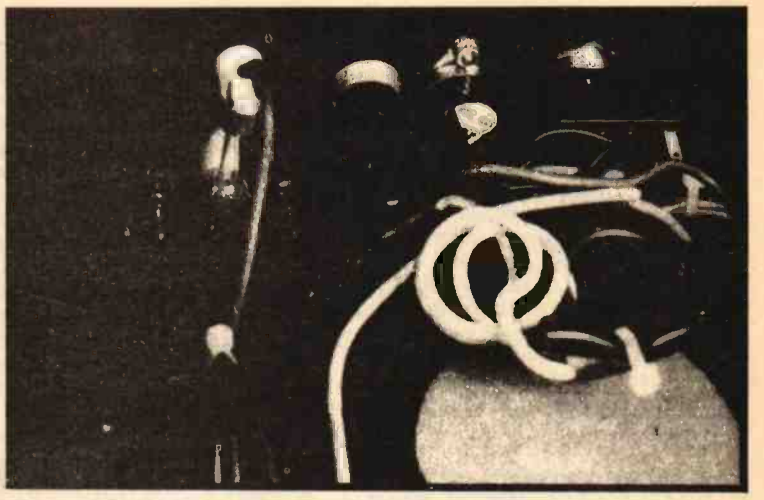
Amp of Substance

The output stage is quite substantial, using a total of six 250 W power transistors. Fairly 'old-fashioned' power transistors have been used (the MJ4502/802 family) in preference to some of the higher performance devices now available. They have been chosen because the die used to mount the semiconductor junction is of a large area; the device is quite rugged and can handle high currents. The short-term current capability of the output stage is, in fact, of the order of 90 A, somewhat in excess of the current capability of the wiring!

The power supply is equally substantial, using a 500 VA toroidal mains transformer and two massive computer grade reservoir capacitors. These components are expensive but essential. The rest of the construction is equally massive with a steel chassis supporting six very large heatsinks. However, construction is straightforward provided that the builder has strong arm muscles, and circuit alignment simple — there are but two adjustments — quiescent current and DC offset voltage nulling.

Construction

The constructional layout shown in the drawings and photographs should be followed as closely as possible. (With such high currents flowing down the cable forms, problems can easily occur if too many changes are made). The heatsinks and the power supply components are assembled onto the base-plate and wired up in accordance with the wiring diagram. The recommended wire types and gauges should be adhered to.



Close-up of fuse wiring on back panel.

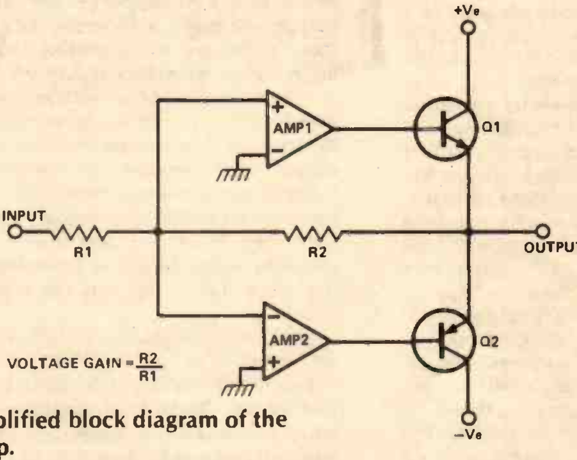


Fig. 1 Simplified block diagram of the power amp.

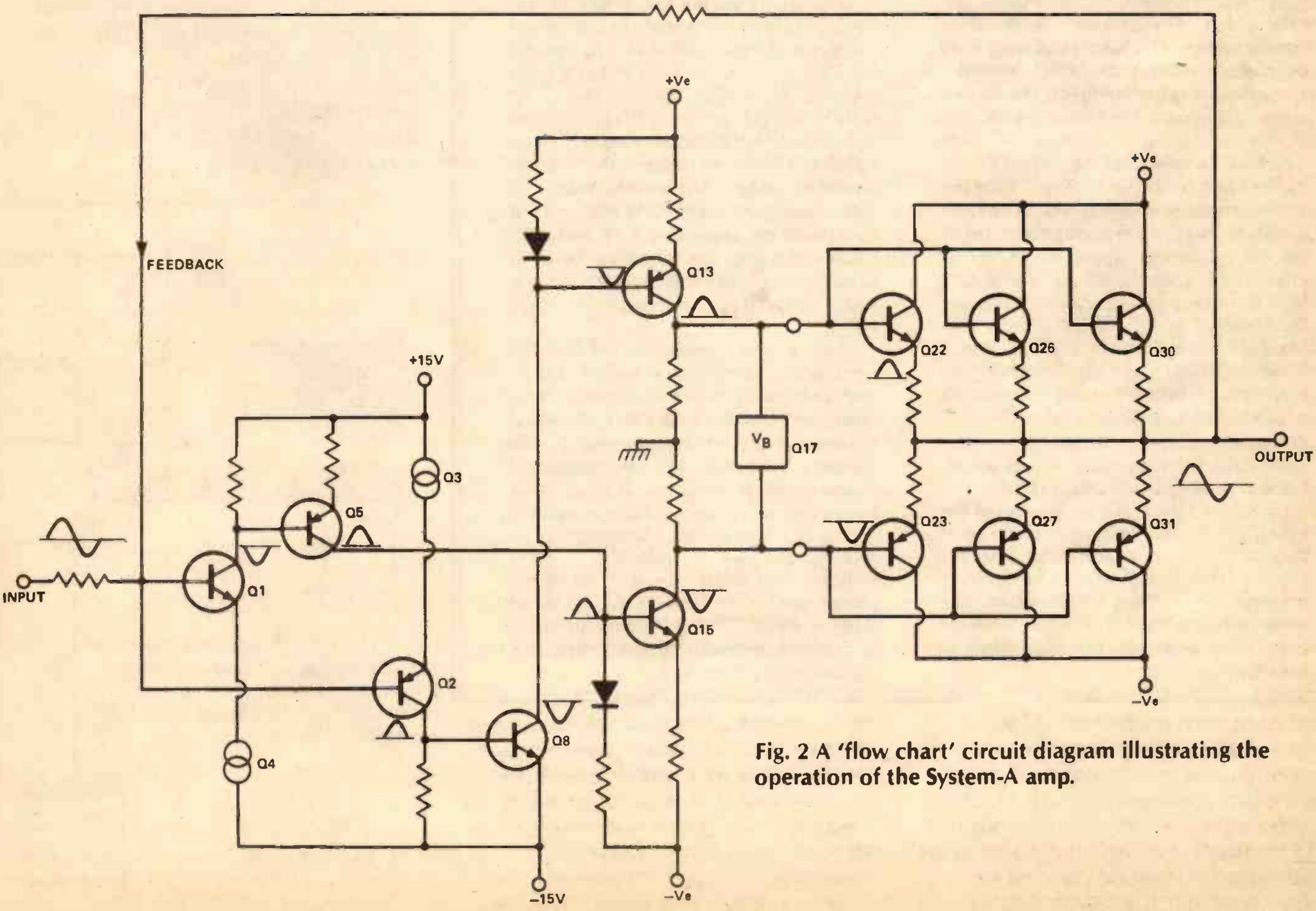


Fig. 2 A 'flow chart' circuit diagram illustrating the operation of the System-A amp.

HOW IT WORKS

This amplifier is basically simple, as can be seen from the block diagram (Fig. 1). Conventional complementary emitter-followers are driven by two separate voltage amplifiers arranged such that one handles the positive-going signals and the other the negative-going signals. A moderate amount of overall negative shunt feedback is then applied to stabilise the gain. To maintain a balanced and symmetrical treatment of the signal the performance of each 'sub-amplifier' should be the same. Furthermore these amplifiers have been designed to operate independently, without the need for the balancing signal currents from their 'mirror image' halves required in many so-called balanced amplifiers. The simplified circuit (Fig. 2) shows that each sub-amplifier consists of two voltage-gain stages. This stage is of a novel arrangement previously used in a Meridian amplifier and subsequently in amplifiers by Lecson and Syntec. In the redesigned form here, the first stage consists of a complementary two-stage common emitter (Q1, Q5) whose gain is about x 2.3. The second stage is a current mirror stage (Q13) which drives the voltage across a load resistor tied to 0 V. The gain of this stage is about x 200. Thus the overall open loop voltage gain is of the order of x 460 and so, as the closed loop gain is x 26.7, the reduction due to negative feedback is x 17.2 or about 24 dB.

Looking now at the final circuit (Fig. 3) it will be seen that the input amplifiers are powered from ±15 V supply rails derived from resistor-zener regulators (R14-ZD1, and R15-ZD2). The current through the first stage (Q1) is held constant, at about 0.36 mA by a floating regulator stage (Q3, Q4) which also provides temperature compensation. The gain of this stage is set by emitter resistor R4 which provides some local negative feedback. The second stage (Q5) is loaded by two series cascode transistors (Q6, Q7), the first having its base tied to ground and the second having its base tied to the — 15 V rail. Thus the maximum collector voltage swing on Q5 is greatly reduced, so reducing the effect of the base-collector capacitance (Miller effect) which would reduce this stage's high frequency bandwidth. In summary, the presence of Q6 and Q7 improves the bandwidth and linearity. The load on Q7 is one half (Q12) of the current mirror and can be visualised as a resistor in series with a forward-biased diode. The second half of the current mirror is a common-emitter stage (Q15, Q16), a simple voltage amplifier except that its collector current equals (or 'mirrors') the collector current of the other half (Q12). This stage is made up of two transistors in parallel which share the current. This arrangement was found to improve the linearity of the stage. The other 'sub-amplifier' (Q2 to Q14) works in exactly the same way but with opposite polarity.

The output stage uses the conventional Darlington emitter follower arrangement, but with three parallel pairs of driver and output transistors. A transistor (Q17) is wired across the bases of the pre-driver transistors (Q18, Q19), providing a bias voltage to set the standing current in the output stage. Q17 is mounted on the heatsink with the aim of keeping this current constant regardless of temperature. Preset resistor PR2 is used to set the value of this current.

It will be seen that both the current mirror stages are driven from power supply rails that are different from those feeding the output stage. The same supply could be used but the signal in the current mirrors would clip well before the output stage, reducing the available output power. In fact the supplies to the current mirrors are made sufficiently high that these stages are still operating in their linear regions when the output stage clips.

The output DC offset voltage is set to zero by preset PR1 in the input stage. In theory there should be no DC offset at the output but, because of component tolerances and consequent mismatching, there always is. PR1 is arranged to make the current in the first stage of one 'sub-amplifier' either higher or lower than in the other and so null out any residual offset.

A simple low-pass filter is created by an R-C network at the input (R2, C2) to reduce the bandwidth of the signal below that of the open loop amplifier and thereby eliminate the generation of any transient intermodulation distortion.

The power supply has to deliver two split rails. The main supply to the output stage is nominally ±40 V at 4 A, derived from the main transformer windings and rectified by bridge rectifier BR1. This rectifier can get very hot so it is bolted onto the chassis. The secondary supply is a low-current ±50 V to power the voltage amplifier stages. The output from the extra windings is rectified by BR2 and fed to smoothing capacitors C12 and C13. These capacitors are not wired between supply and ground but between the two supplies; this layout reduces their voltage rating.

The mains supply is fed to the transformer via an on-off switch, a fuse, and a thermal cut-out switch. Two neon indicator lamps are used. LP1 is connected between live and neutral and is the 'power' indicator; LP2 is connected across the thermal cut-out. If this cut-out opens the full supply voltage is applied across LP2 which then illuminates as an 'overtemperature' indicator. (This indicator has never operated yet in the prototypes.) Care should be taken to adequately sleeve and insulate all mains wiring and terminals to ensure safe and reliable operation.

Any bare wire ends should be sleeved using silicone rubber sleeving. This may seem an extravagance but your opinion will change shortly after a short-circuit wipes out £18 worth of transistors! A substantial soldering iron will be needed to solder together the power supply components. The use of a low-power iron will usually result in a selection of dry joints on these connections.

The coil L1 is wound onto the body of R40. This is not a critical procedure – about 17 to 20 turns of enamelled copper wire should do nicely. The gauge can be anything you have to hand, from 20 to 26 swg. Use some lacquer or epoxy to hold the wire in place on the resistor, scrape the enamel off the ends of the wire and solder them close to the resistor. The whole thing can now be soldered in place on the board.

Particular care should be taken in mounting the power transistors. Good quality insulating washers and bushes should be used and a generous smearing of thermal paste is essential. These transistors should be bolted to the heatsinks very tightly to ensure good thermal contact at all temperatures.

Assembly of the printed circuit board is straightforward enough using the component overlay as a guide. As usual, particular care should be taken to confirm the polarity and alignment of all capacitors, diodes and transistors; and to avoid putting mechanical strain on any of the components. After assembly the board should be checked on the copper side for dry joints and solder bridges. Such defects on power amps usually result in an expensive bang, so don't skip this admittedly tedious chore.

One final point regarding construction. Once the amplifier has been completed and tested, it should be switched on and allowed to reach its normal operating temperature (about 20 minutes). The amplifier should then be switched off and all the screws tightened up. Differences in thermal coefficients of expansion can result in some of the screws becoming slightly loose, particularly those holding the heatsinks to the top and bottom covers.

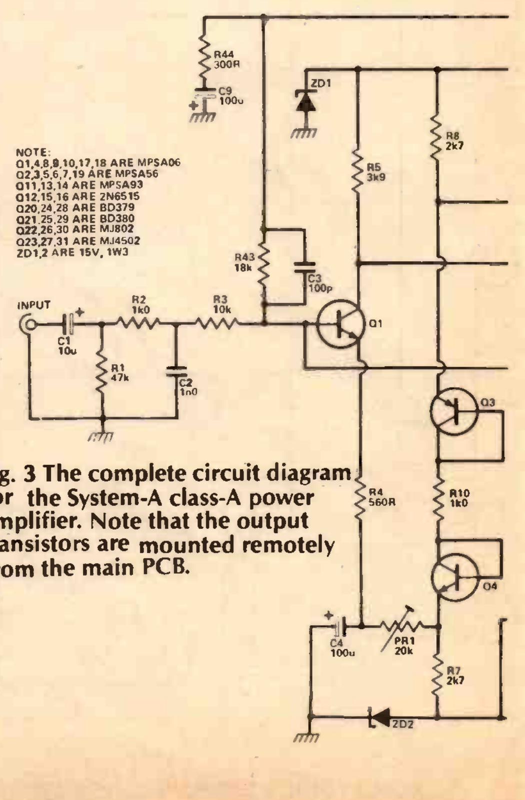
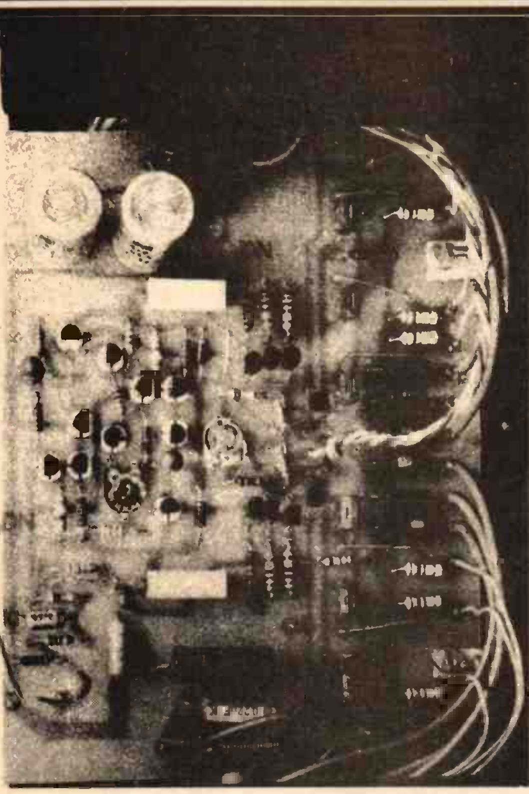


Fig. 3 The complete circuit diagram for the System-A class-A power amplifier. Note that the output transistors are mounted remotely from the main PCB.



Left: A detail shot showing how the PCB is wired into the case.

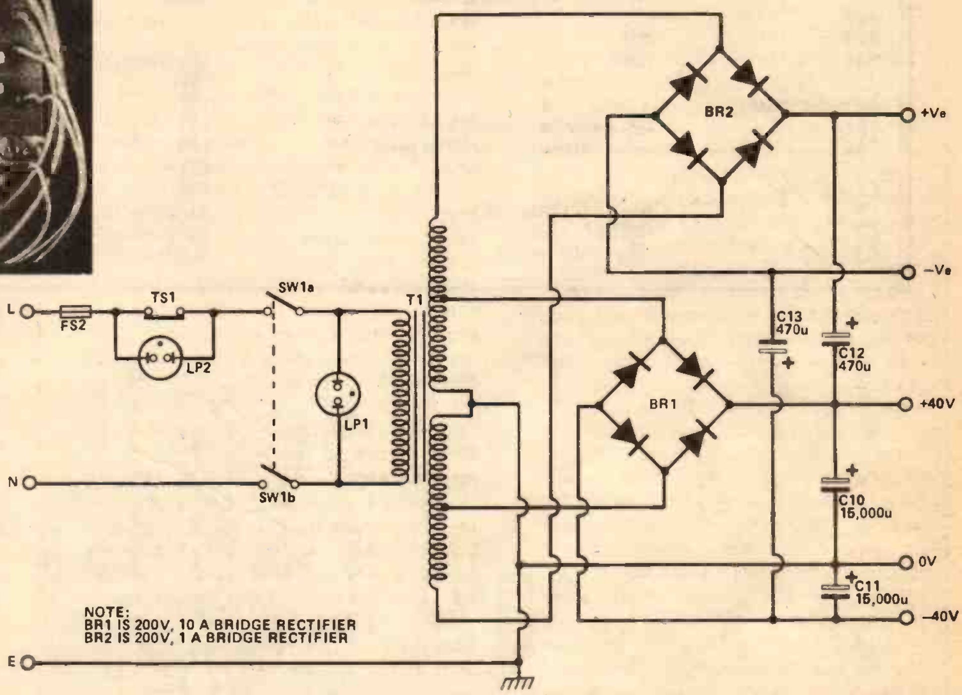
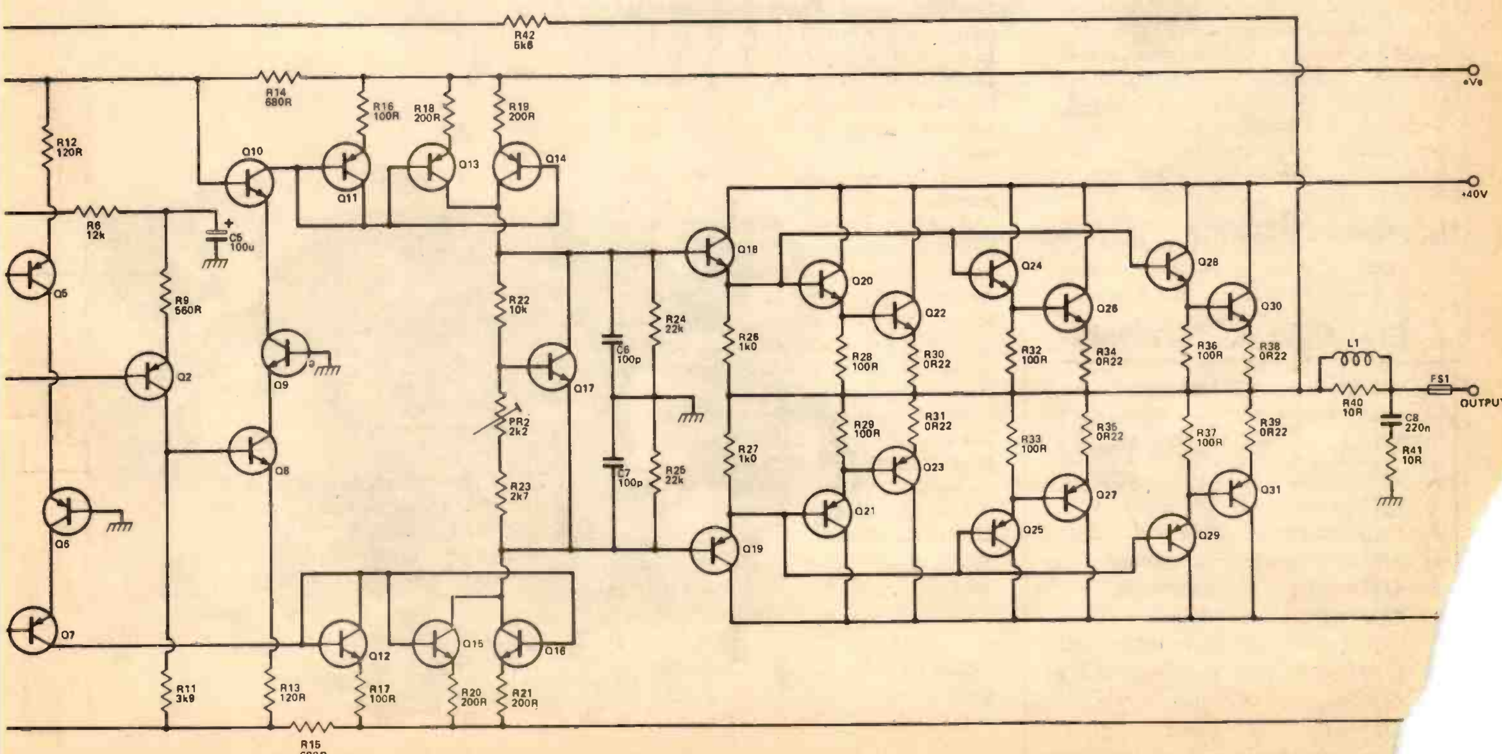


Fig. 4 The PSU circuit used to drive the power amps.



PARTS LIST

Resistors (all ¼ W, 5% except where stated)		C4,5,9	100u 6V3 tantalum
R1	47k	19 C6,7	100p miniature ceramic
R2,10,26,27	1k0	C8	220n polycarbonate
R3,22	10k	C10,11	15,000u 50 V electrolytic (Sprague type 36D)
R4,9	560R	C12,13	470u 63 V electrolytic (PCB type)
R5,11	3k9		
R6	12k	**Semiconductors**	
R7,8,23	2k7	Q1,4,8,9,10,17,18	MPSA06
R12,13	120R	Q2,3,5,6,7,19	MPSA56
R14,15	680R 4 W	Q11,13,14	MPSA93
R16,17,28,29,32,33, 36,37	100R	Q12,15,16	2N6515
R18,19,20,21	200R	Q20,24,28	BD379
R24,25	22k	Q21,25,29	BD380
R30,31,34,35,38,39	0R22 2W5	Q22,26,30	MJ802
R40	10R 1 W	Q23,27,31	MJ4502
R41	10R 2 W (not wirewound)	ZD1,2	15 V, 1W3
R42	5k6		
R43	18k	**Miscellaneous**	
R44	300R	SW1	DPST mains switch
Potentiometers		TS1	Thermal cut-out switch
PR1	20k miniature horizontal preset	LP1	Red neon
PR2	2k2 miniature horizontal preset	LP2	Orange neon
		FS1	1¼" 5 A–10 A (to suit loudspeaker)
Capacitors		FS2	20mm 3.15 A
C1	10u 35 V tantalum	Toroidal transformer, 1¼" chassis-mounting holder, 20 mm panel-mounting holder, phono input socket, loudspeaker screw-terminals, chassis and heatsinks, mounting hardware.	
C2	1n0 polystyrene		
C3	100p polystyrene		

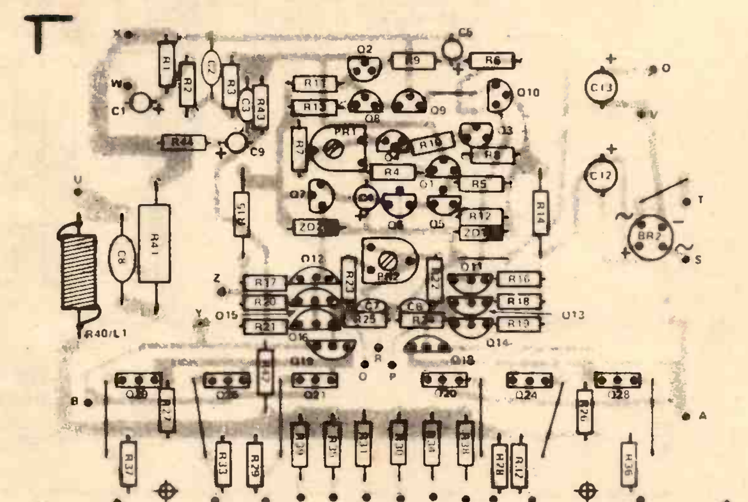
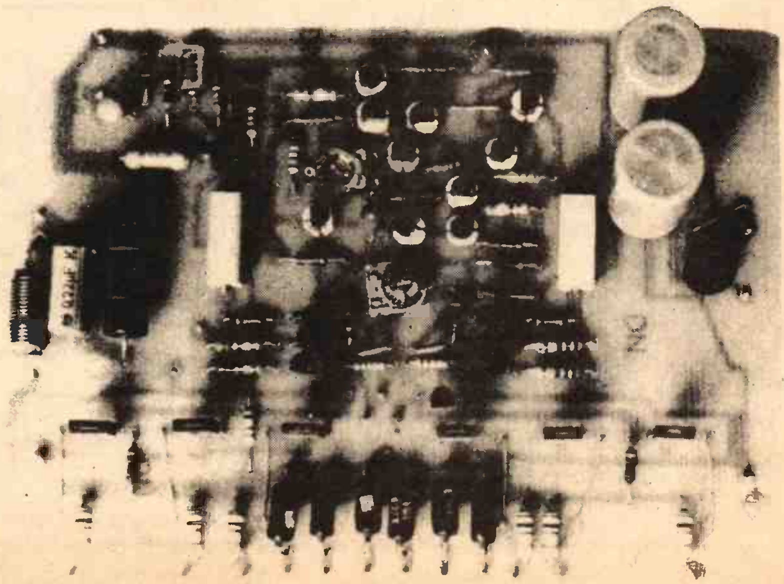


Fig. 5 Component overlay for the power-amp PCB.



PIN CONNECTIONS

A	+40 V	N	Q31 base
B	−40 V	O	Wire link to pin Y (underside of PCB)
C	Q30 emitter		
D	Q26 emitter	P	Q17 collector
E	Q22 emitter	Q	Q17 emitter
F	Q23 emitter	R	Q17 base
G	Q27 emitter	S	Transformer
H	Q31 emitter	T	Transformer
I	Q30 base	U	Output
	Q26 base	V	Wire link to pin Z (underside of PCB)
	Q22 base		
	Q23 base	W	Input
	Q27 base	X	Ground

Testing and Set-up

This amplifier is straightforward to test providing a logical sequence is followed. The first test is without the main PCB fitted and without the power-transistors connected to the power supply. Check that there is no leakage between the collector of any power transistor and heatsink using a high-resistance range of your meter. Next check the output transistor junctions (base-collector, base-emitter, collector-emitter and so on) at the PCB end of the wiring loom. If all is well the power transistors can be forgotten for the moment.

Next, the power supply. Fit a mains fuse; switch on and check that the voltage across the reservoir capacitors is ±40 V (within 2 V). Allow these capacitors to discharge and then fit the PCB assembly, connecting all the wires except those to the power transistors. Both the presets should be set to mid-travel and the power again switched on. The secondary supply rails can now be measured and should be about ±50 V. The output DC offset voltage (junction of R28, R29) should be measured and should be adjustable to zero by turning PR1. If the offset voltage cannot be adjusted you have a fault on the board.

If all is well, disconnect the supply and again wait for the power supply to discharge. Now connect up the power transistors to the PCB but with a current meter (able to measure greater than 3 A DC current) in series with the positive supply to the three collectors (Q22, Q26 and Q30). Ideally a voltmeter should be connected between the output rail and ground. Say a short prayer and switch on. You should find that PR2 (turned clockwise) will increase the current and PR1 should still adjust the DC offset voltage. Adjust the current to about 1 A and, using a loudspeaker and convenient signal source, quickly check that the amplifier works. If it does, the amplifier can be set up properly; but be warned that this takes several hours. Set the current to 3 A and allow the amplifier to heat up. The current will vary so adjust it *gradually* every 10 minutes or so until it is stable. The DC offset can now be nulled to zero but as this can interact with the current some alternate adjustments will be needed. After a couple of hours the amplifier should be stable and ready for use.

Stan Curtis has written an update for the System A which, unfortunately, could not be printed in this issue of Electronics Digest. However the Editor of ETI has kindly agreed to publish the update and it will appear in the February 1986 issue of Electronics Today International.

PARTS LIST

Resistor	
R1	2k7 ¼W 5%
Capacitors	
C1,2	1000u 25 V axial electrolytic
Semiconductors	
D1-4	1N4002 or similar
LED1	TIL209 or similar
Miscellaneous	
SW1	DPDT mains switch
Transformer (15-0-15, 20 VA), 1 A quick-blow fuse and fuseholder, case.	

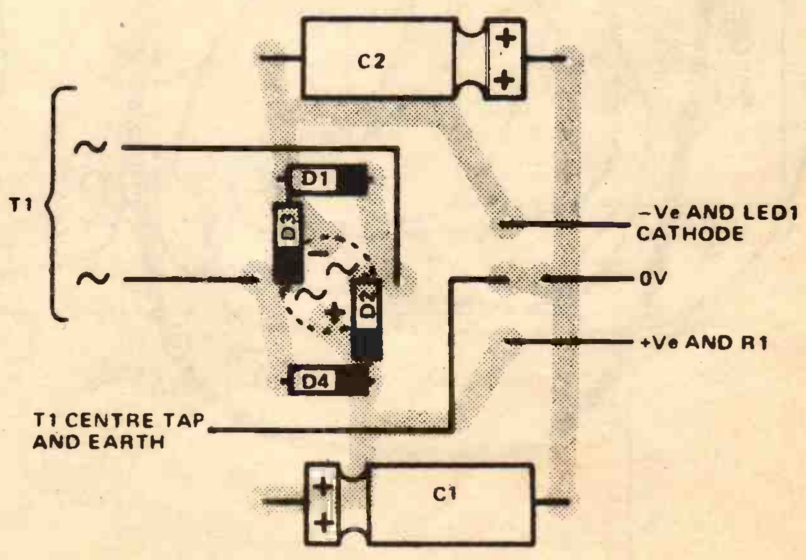
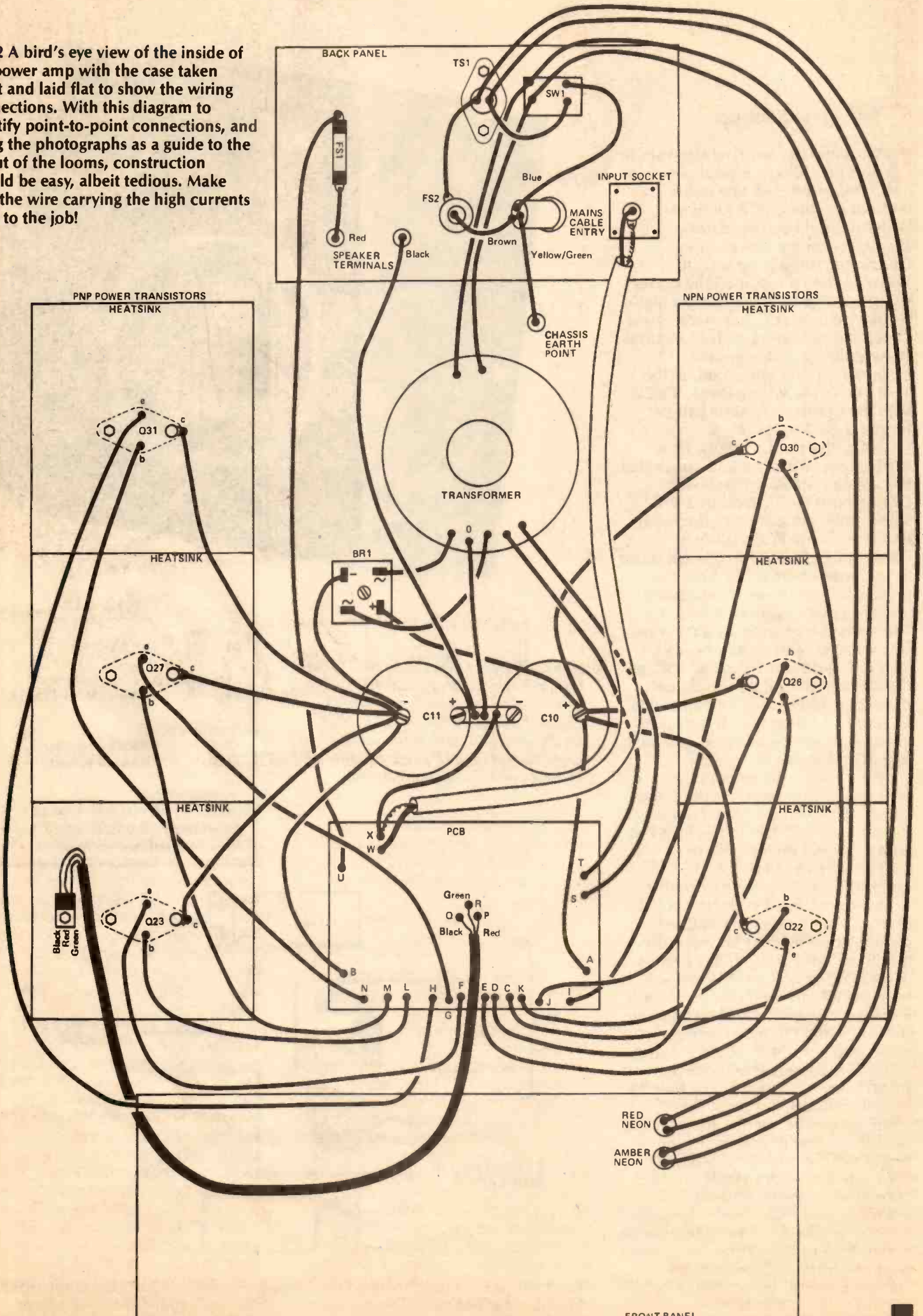
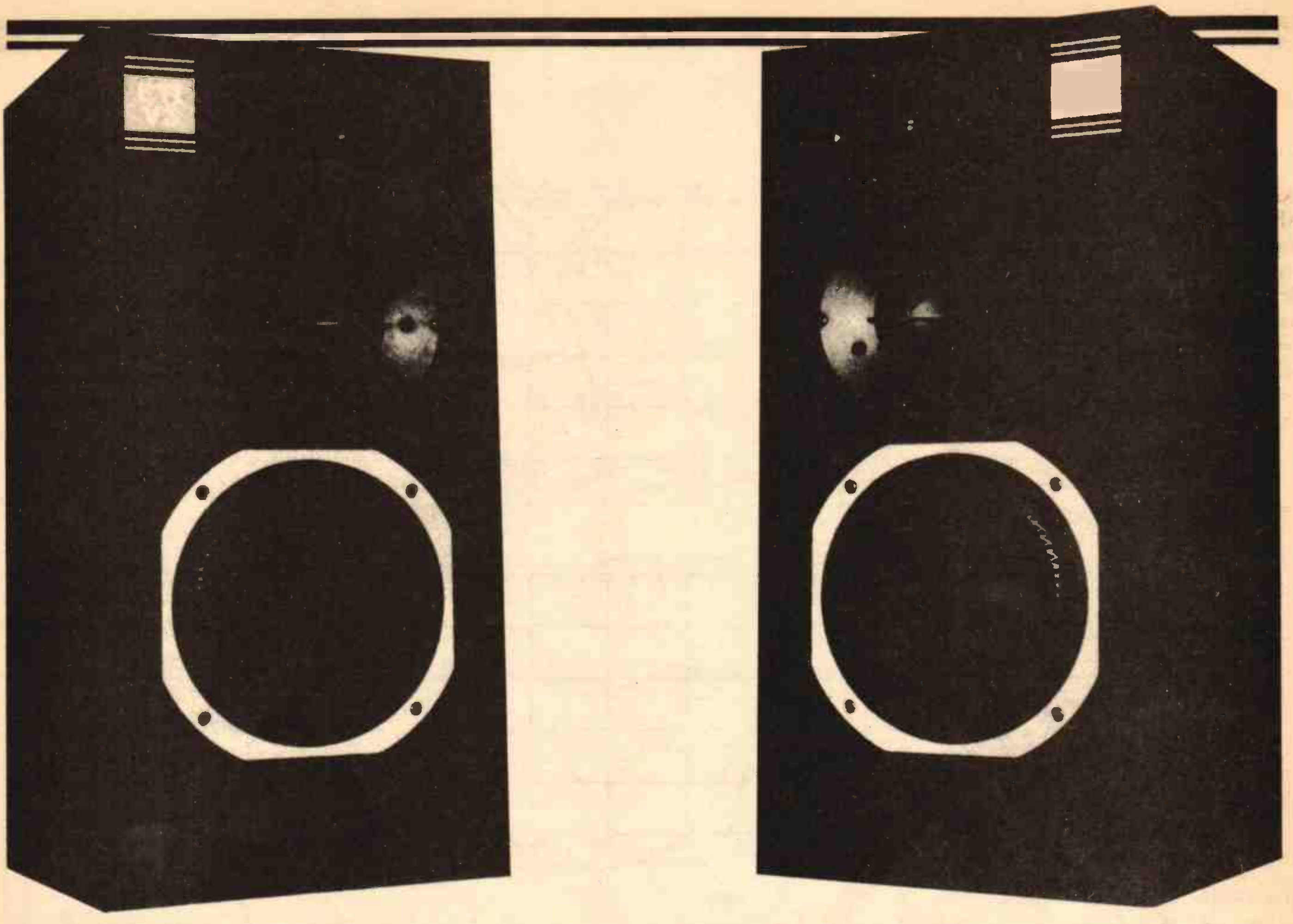


Fig. 1 The long-awaited preamp power supply overlay (and Parts List at top right). Provision has been made on the board for either four diodes or a small bridge rectifier, as shown.

Fig. 2 A bird's eye view of the inside of the power amp with the case taken apart and laid flat to show the wiring connections. With this diagram to identify point-to-point connections, and using the photographs as a guide to the layout of the looms, construction should be easy, albeit tedious. Make sure the wire carrying the high currents is up to the job!





V3 LOUDSPEAKER

As always the best in DIY hi-fi. Design and development by David Lyth.

The V3 loudspeaker system is a three unit design using a Volt 8" bass driver and a Philips dome midrange and tweeter; crossover points are 700 Hz and 5 kHz. A reflex enclosure is employed, constructed from 19 mm high-density chipboard with the units mounted in a mirror image configuration. A grille baffle is not necessary for other than aesthetic reasons because the driver units incorporate either a grille or similar structure protecting the 'software'. However, extensive research has revealed that small fingers can get at the midrange dome which will result in a recessive sound balance. Beware!

The V3 is a medium efficiency system with a wide frequency response and is capable of high power handling, especially in the bass. Distortion and colouration are very low and the smooth response gives good tolerance of aggressive or edgy material.

The cabinet uses high-density chipboard 19 mm thick and is simple to construct, with panels coming economically from standard sheet sizes. All the drive units are rebated and some care is necessary to get a good fit, although plywood could be used for the front panel to make this task easier.

Reflex Action

Reflex enclosures have been the subject of much controversy in the past because some designs work well and some are appallingly bad. The problem lies in the design itself and the approach taken here is to use tables generated by A. N. Thiele, an Australian. Thiele likened the combination of bass unit and enclosure to an electrical filter and used synthesis techniques from this field to build up tables enabling the designer to choose a cabinet to suit a particular drive unit. Knowledge of various drive unit parameters is necessary — for example, Q_T, V_{AS}, F_S. The system response options are those shown by Butterworth or Chebyshev high pass filters — Fig. 3 shows some characteristics.

However, the Thiele alignments are not the final solution to the design problem. The responses available (with the exception of the QB3) are flat down to the -2 dB point (give or take a ripple) and a correctly aligned system would show this response under free field or anechoic conditions. But who sits around listening to ideal electrical filters hanging 30 m above a field? Your private life is none of my concern, but I have found that when a pair of loudspeakers are listened to in an average sized room the bass response can sound unbalanced. This is because they are not 'looking' into omnidirectional space but seeing rather less than this depending on the wavelength of the sound reproduced and system-room positioning. There is a dramatic difference between the bass response of a system held in the middle of a room and that when placed in a corner. This is similar to the variation when going from true free field conditions to a listening room, and it is necessary to modify the 'correct' response to compensate for the bass

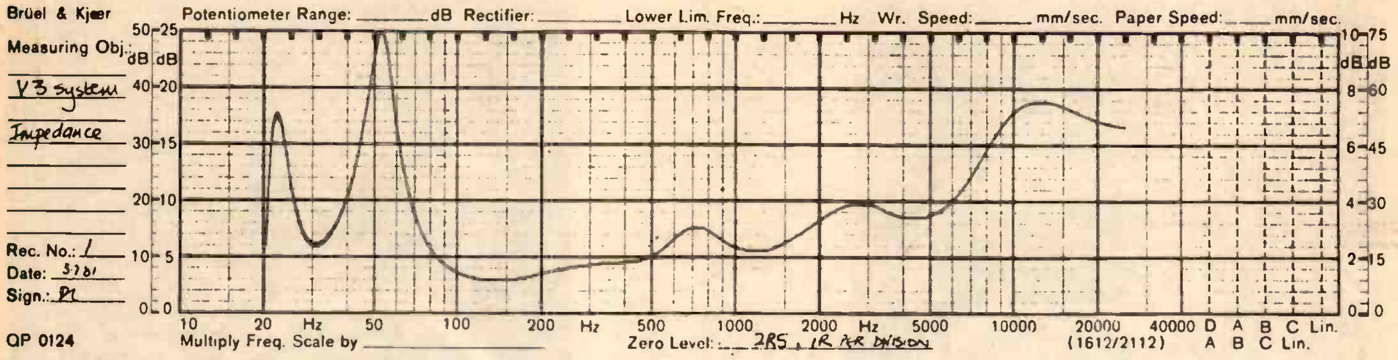


Fig. 1 Impedance curve for the V3 Loudspeaker.

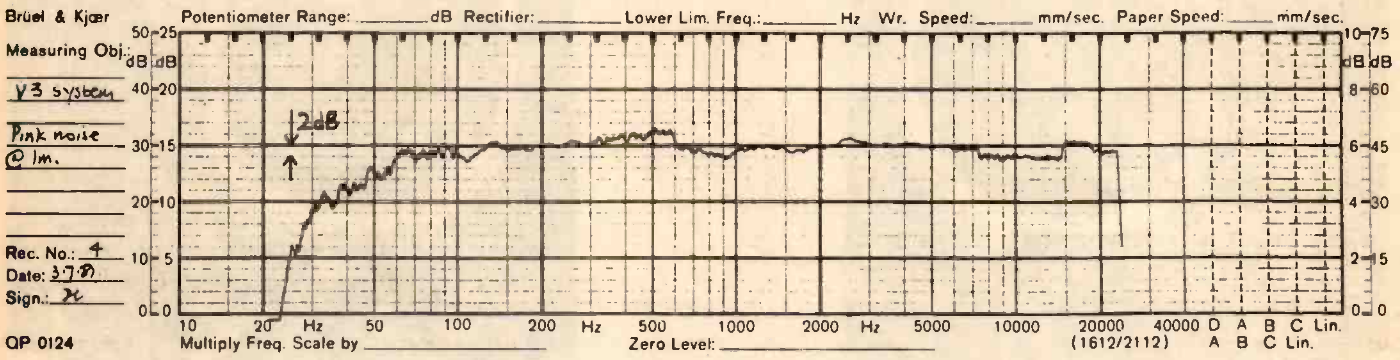


Fig. 2 Frequency response of the V3.

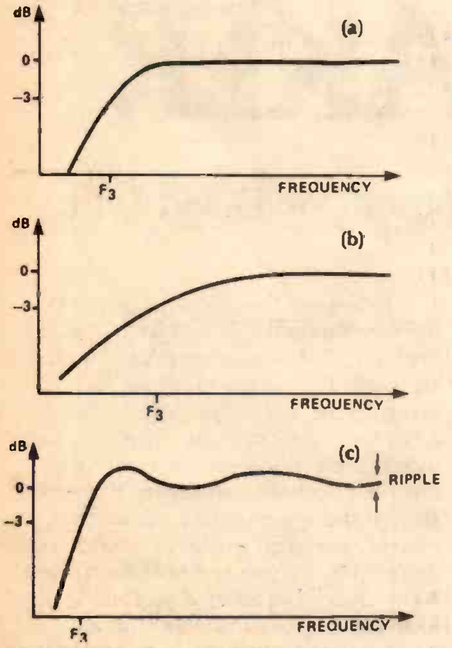


Fig. 3 Butterworth B4 response (a); Quasi-Butterworth response (b); Chebyshev C4 response (c).

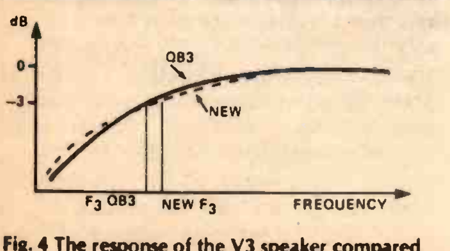


Fig. 4 The response of the V3 speaker compared to the Thiele QB3, showing the improvement in the bass.

boost provided by the room so that very deep bass is maintained but overemphasis of middle and upper bass is avoided.

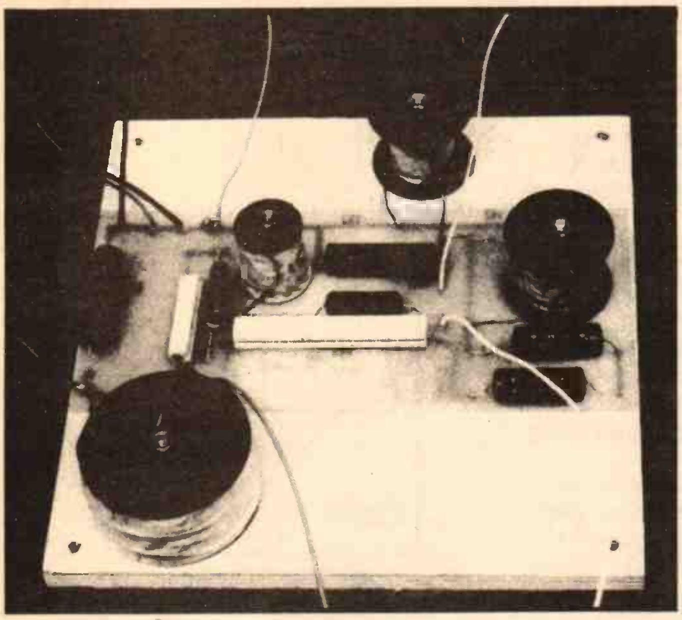
This means that we want a flat response in the room and not in free field conditions. The Thiele alignment that most closely approaches our desired free field response is the QB3. Initial work showed that a fairly small cabinet (volume about 29 litres) produced good results but there was still an overemphasis of middle bass in an average sized room. Enlarging the cabinet by 30% and reducing the reflex tuning frequency to 30 Hz gave a dramatic improvement, the bass output sounding even and with an excellent lower octave. In a way the obvious has occurred ie "a good big 'un will always beat a good little 'un". Excellent transient response is maintained and this is fundamentally due to the large magnet fitted to the B220 bass unit. Nevertheless the Thiele approach was an excellent starting point for the design. Figure 4 shows the effective improvement in the bass response. The system was always used on stands 250 mm high with free space underneath, placed with its back against a wall and kept out of corners.

Middle And Upper Class

A midrange dome was selected because domes have a fundamental advantage over coned units of the same size — their break up modes occur at higher frequencies. As the alternative to a 50 mm dome would have been a 75 mm cone unit, this means that the dome will operate with more diaphragm control to a rather higher frequency. The tweeter has a soft dome as opposed to the midrange's pulp dome, and incorporates a diffuser.

The midrange and tweeter have very smooth responses with low colouration and distortion. These two units are offset from the centre of the cabinet so that the pair are mirror imaged on the two speakers. This improves stereo imaging, mainly by reducing edge diffraction which is a major cause of poor stereo.

The worst case for diffraction is when the tweeter, for example, is mounted equidistant from the three nearest cabinet edges. Re-radiation will then take place at the same frequency for each edge and this will cause a discontinuity in the frequency response. By offsetting the unit the diffraction is smeared and reduced to insignificant levels. Elimination of the grille also helps because there is no grille baffle standing proud of the cabinet to present an obstacle to the surface sound wave, thus worsening diffraction. The only real argument for a grille is aesthetic and then a foam grille is the best solution.



After the components have been soldered to the PCB, the crossover board is secured to a plywood panel using the choke fixing bolts. To get the choke connecting leads the right length, you'll have to cut access holes beneath the PCB pads so you can solder the wires last.

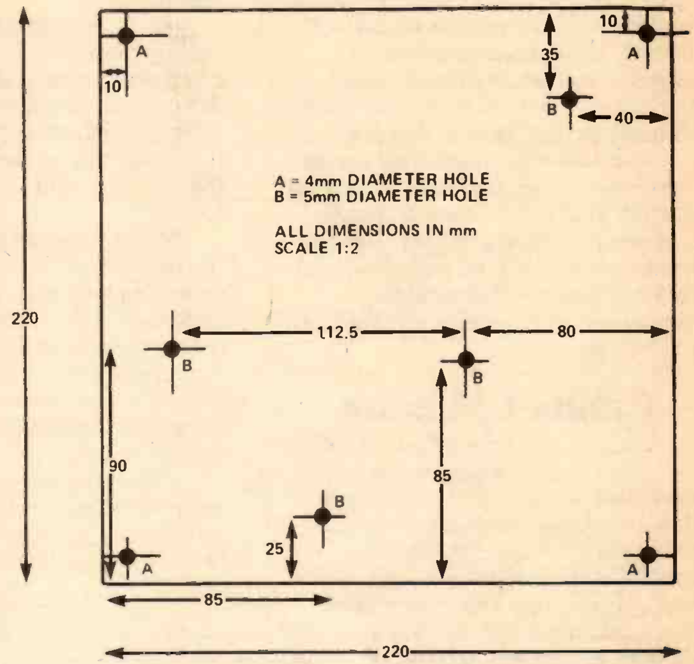


Fig. 5 This diagram shows the drilling positions for the choke and mounting screwholes on the crossover panel.

The Crossover

This uses air cored chokes exclusively because these have a better ability to pass transients than ferrite cored chokes, which can momentarily saturate on high power peaks. The chokes are well spaced to prevent any flux linkage between them. There are two 40uF capacitors feeding the midrange so that their combined voltage rating is great enough to prevent possible failure under high power drive.

The crossover is constructed on a PCB which is attached to a wooden crossover board by clamping it beneath the chokes which are bolted through with brass screws. The wire used should have a 6 A rating.

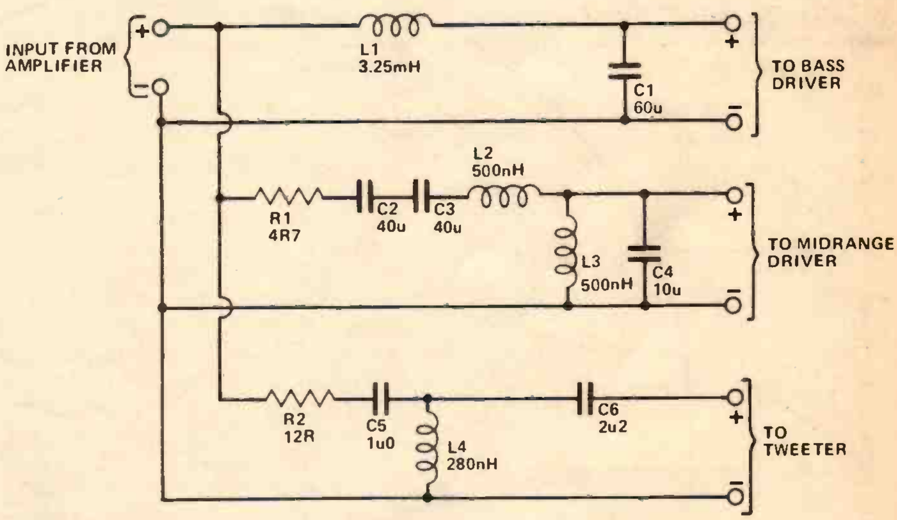


Fig. 6 Circuit diagram of the crossover.

Listening Tests

During the design of the V3 there were constant trips made between listening rooms and the test equipment in order to make modifications to the crossover or cabinet. Comparisons were made with the Yamaha NS1000M, Gale GS401A, KEF 105 II and Popular Hi-fi Boxers: each of these systems had particular strong points and it was a design aim to approach the low colouration and discrimination of these designs. Listening was done with material ranging from choral works through to heavy rock. The cabinets were set so that the dome units were on the inside of the pair. A front grille was never used and connection was by screened twin lead with the screen connected to the inner core, thus

HOW IT WORKS

The crossover uses second order filters throughout, except for the high pass tweeter section which is third order. The values diverge from those of the text book because the load presented by the drivers is not a constant resistance. Apart from the impedance rise at resonance for the midrange, all units exhibit a rising impedance characteristic over their possible operational range because of the voice coil inductance. This is compensated for in the low pass section feeding the bass driver by using a larger shunt capacitor than calculations show.

The band pass filter used for the midrange also includes a little response shaping in its function. The midrange unit has an impedance rise at its resonant frequency (which lies one octave below the crossover point) and to control this the shunt choke of 0.5mH is appreciably smaller than calculated. This is because the normally rising impedance characteristic of a high pass filter below the crossover point prevents amplifier damping from controlling a resonance in this area — this is the case with the Philips unit. By using the lowest possible value choke next to the unit that did not upset frequency response or drop overall system impedance it is possible to give the unit a degree of damping by simulating a low impedance drive around resonance. The net result is better control and increased power handling. This consideration dictated the choice of the lower crossover point.

The upper crossover point is chosen to match the radiation characteristics of the units as closely as possible and to provide the best integration. The tweeter is also attenuated to match levels.

making twin flex. If there is a magic in speaker cables, apart from common sense, then the greater surface area of the screen should reduce skin effect, the only thing that would have a deleterious effect on the electrical transmission — 13 A twin cable would probably do just as well, however.

Ancillary equipment tried varied from moving coil cartridges to moving magnets with either valve or a high-powered transistor amplifier. The end result justifies the time and effort and the V3 stands very favourable comparison with systems costing £400 upwards per pair.

Cabinet Making

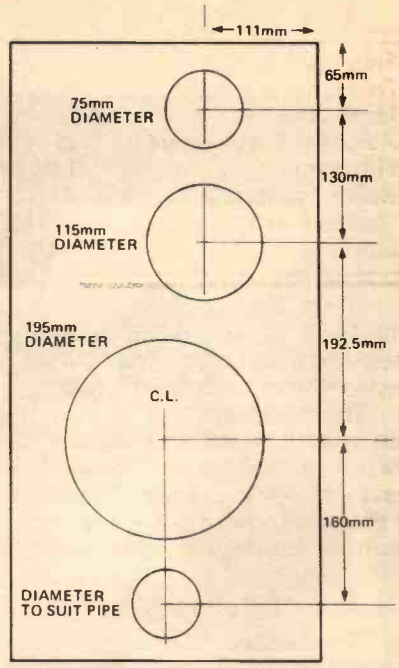
You will need the following: 24 + 1½" x No. 8 self-tapping screws and screw sink No. 8 — 1" long; flexible Plastic Padding, sandpapers, electric drill, jig saw, chisels, 7/32" drill bit, and wood glue. Butt joints are used throughout — providing reasonable care is taken this will provide a satisfactory cabinet. Rebates or internal battens are only necessary where high strength is required, which is not the case with a domestic system.

Internal battens, or a least small blocks, may be useful in helping to locate the side panels.

An important consideration when cutting the panels is to allow the edge of a panel to stand slightly proud (0.5 mm) of the mating panel surface when assembled so that the step produced can be sanded back — at the very least this should be toleranced for.

When it comes to rebating the unit the best plan is to cut the basic mounting hole and rest a unit in this. Draw round the outside edge and rebate to suit the flange thickness — it is not necessary to allow for any gasket thickness. It is more pleasing to the eye if the unit stands slightly proud rather than sub-flush.

The midrange and tweeter are secured by No. 6 x 19 mm pan head self-tapping screws. The bass unit is heavier and requires 2BA x 1" screws



FRONT PANEL (RHS SPEAKER) SHOWING BASIC CUTOUTS FOR UNITS.
PLASTIC PIPE IS 65mm INTERNAL DIAMETER AND 180mm LONG.
CUTOUT ON REAR PANEL FOR TERMINALS IS 30mm x 12mm.
BAF SIZE IS 1 SQUARE METER x 25mm THICK.
SCREWS SHOULD NOT BE MORE THAN 150mm APART.

Fig. 8 Template for the front baffle. The dimensions are important, so cut carefully!

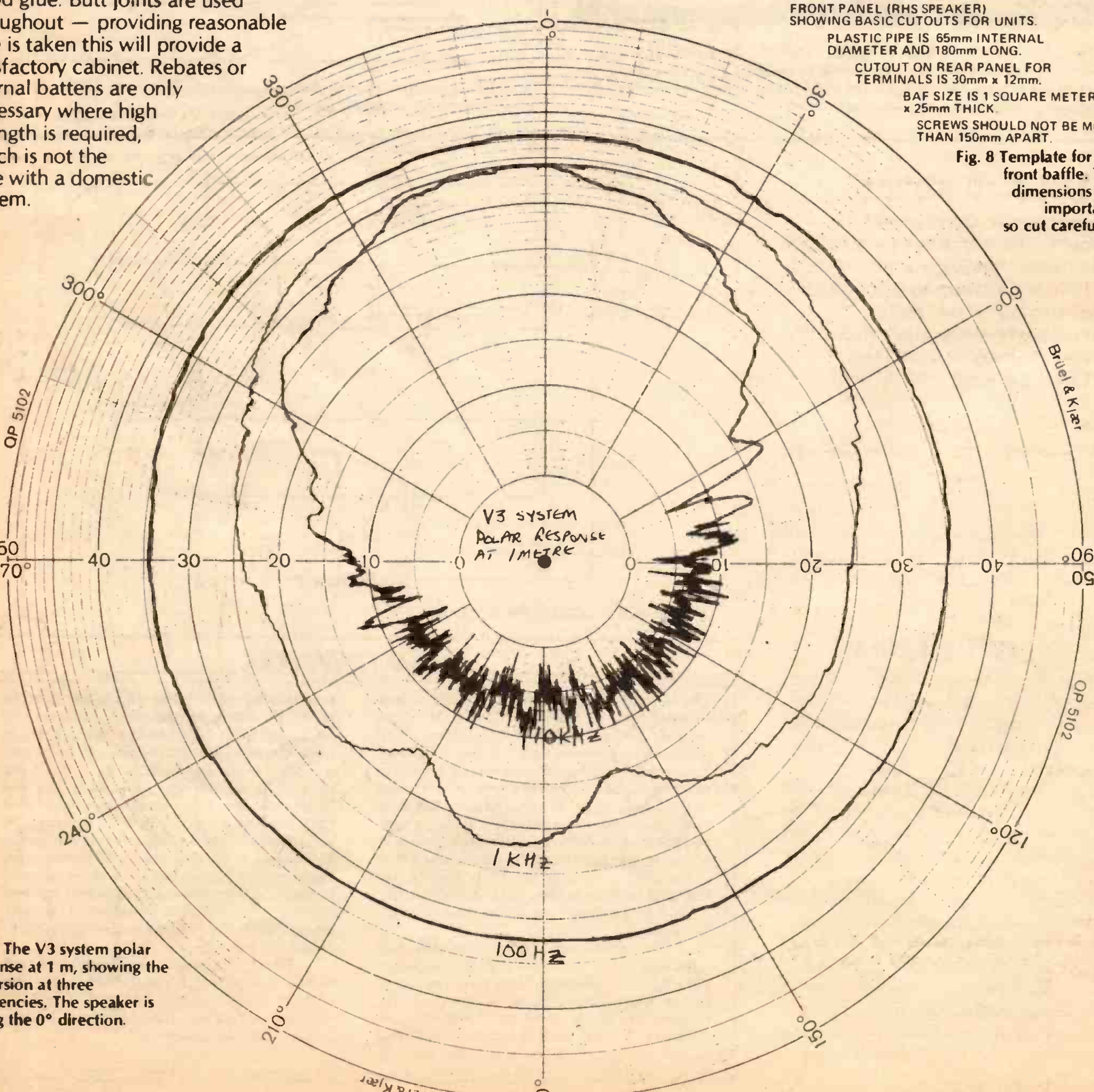


Fig. 7 The V3 system polar response at 1 m, showing the dispersion at three frequencies. The speaker is facing the 0° direction.

TECHNICAL SPECIFICATION

Frequency Range	30 - 20,000 Hz
Frequency Response	60 - 20,000 Hz ± 3 dB
Impedance	8R
Amplifier Requirement	25 W - 150 W
Efficiency	1 W for 82 dB at 1 m
Reflex Tuning	30 Hz
Crossover Frequencies	700 Hz and 5 kHz
Size	600 mm x 300 mm x 300 mm

WOOD CUTTING LIST

2 off 600 mm x 300 mm for front and back
2 off 262 mm x 300 mm for top and bottom
2 off 262 mm x 562 mm for sides
1 off 220 mm x 220 mm for crossover board
All panels are 19 mm high-density chipboard except for the crossover board, which is 12 mm plywood.

with T-nuts — use the 7/32" drill for these and hammer the T-nut into the back of the panel so it seats flat.

The mounting of the plastic tube is easy enough — you simply glue it into the hole at the bottom of the baffle, leaving the inner end free. The length of this pipe is critical, as it tunes the resonant frequency of the enclosure.

An Inside Job

The electrical assembly is straightforward. The completed PCB is bolted to a wooden board which is then glued and screwed onto the back panel of the cabinet — here it will give some extra panel damping. The BAF wadding is distributed about the cabinet such that the vent end is not obstructed and all surfaces other than the front panel are covered. Enough BAF is specified to allow for some to be positioned in the centre of the cabinet where it is at its most effective for stopping standing waves.

Ensure that leads to the drive units do not sit against the cabinet walls or drive units where they might buzz. Air leaks are definitely not allowed.

Good listening!

BUYLINES

A kit of parts comprising all the drive units, all the crossover components including the PCB, BAF wadding, T-nuts and 2BA screws, terminal panel and gaskets is available from Wilmslow Audio, 35-38 Church Street, Wilmslow, Cheshire, SK9 1AS. The cost of a set of parts for one pair of V3 speakers is £228.50 including VAT and carriage. Note that the woodwork is not included in the kit.

PARTS LIST

Resistors

R1	4R7 10%, 17 W wirewound
R2	12R 5%, 7 W wirewound

Capacitors

C1	60u 50 V non-polarised electrolytic
C2,3	40u 50 V non-polarised electrolytic
C4	10u 50 V non-polarised electrolytic
C5	1u0 polyester
C6	2u2 50 V non-polarised electrolytic

Inductors

L1	3.25mH (No. 17)
L2,3	0.5mH (No. 12)
L4	0.28mH (No. 15)

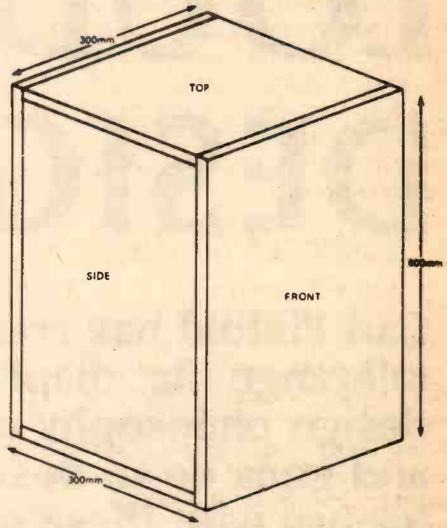


Fig. 10 This is how the panels fit together — if you did your sawing right!

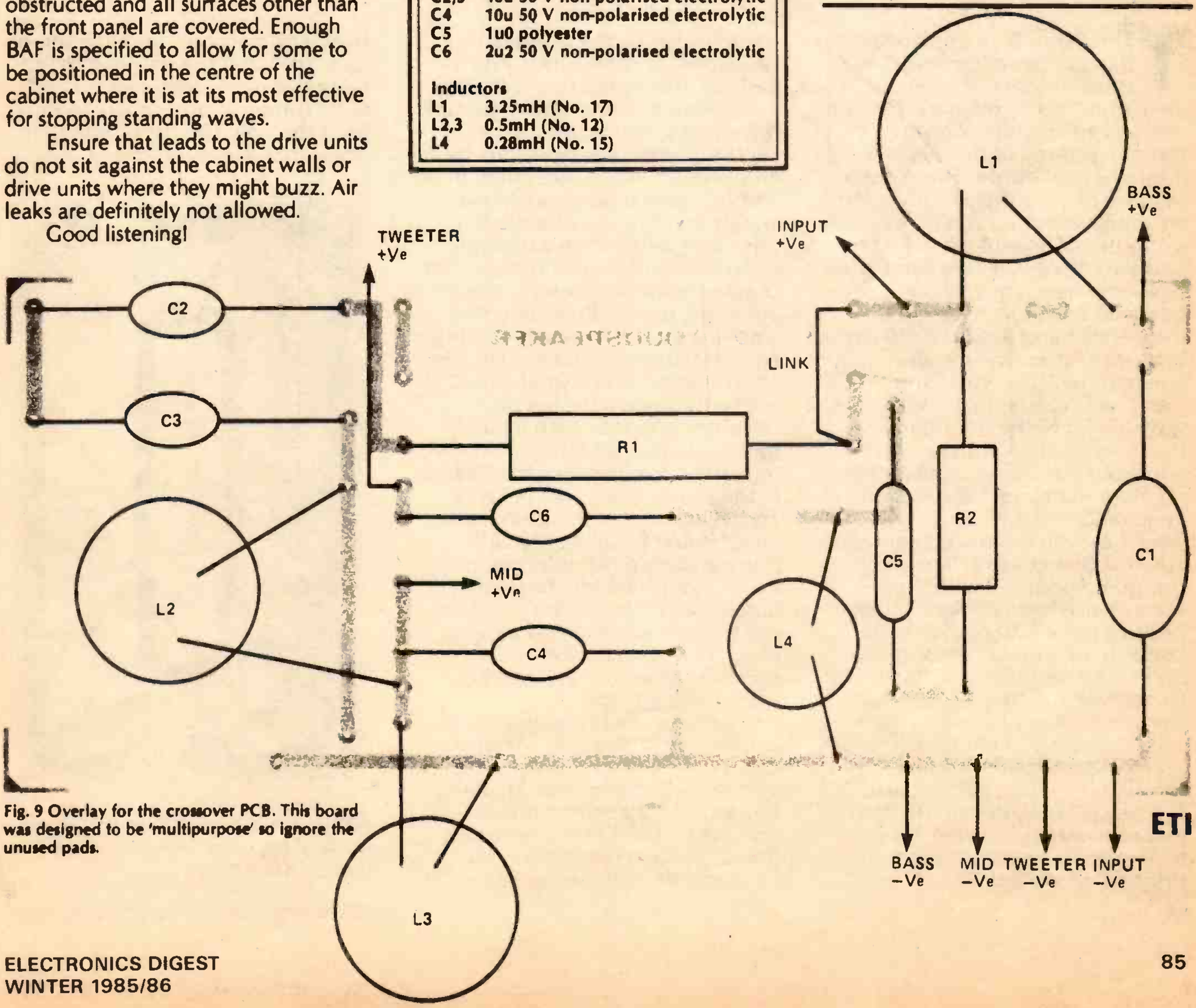


Fig. 9 Overlay for the crossover PCB. This board was designed to be 'multipurpose' so ignore the unused pads.

ETI

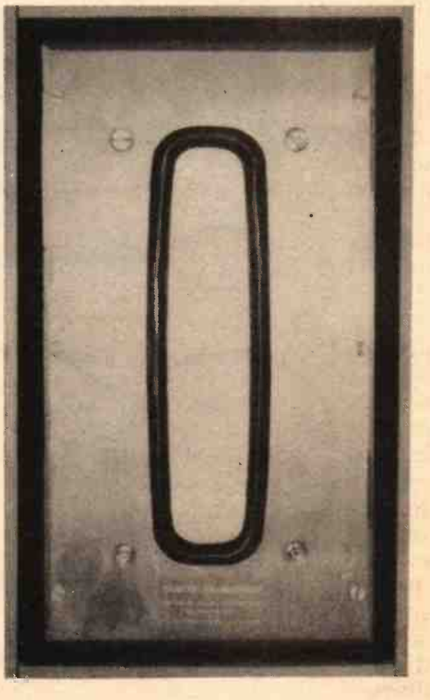
NOVEL LOUDSPEAKER DESIGN

Carl Pinfold has created quite a stir in the hi-fi world with his full-range, flat diaphragm loudspeakers. This article covers the design philosophy of both the drive units and their enclosure, and goes on to describe the construction of similar enclosures for use with these or any other small-cone-area drive units.

The majority of loudspeaker designs currently available use a number of separate drive units fed through a crossover so that each handles only a discrete portion of the audible frequency spectrum. There are a number of advantages with such an arrangement but there are also a number of drawbacks. Where only two drive units are used, the crossover frequency will be between 2.5 and 3.5 kHz, the frequency band in which the ear is at its most sensitive and discriminating. The cone diameter of the drivers can be matched to the wavelength of the frequency bands over which they are operating, but this leaves a tweeter of no more than 5 cm^2 diaphragm area to handle the band of frequencies which often contain the greatest power levels in music, further compounding the problems of crossover design. The use of drive units with conventional cone-shaped diaphragms generates further problems in itself. The frontal volume forms a resonant cavity which introduces megaphone-like colourations, while the limited speed of the sound through the cone material results in a phase lag between the inner and outer areas, causing the cone to 'break-up' and producing an irregular response.

In designing the Musician Loudspeaker drive unit, Carl Pinfold has attempted to overcome the limitations inherent in multiple drive unit systems using conventional cone loudspeakers by employing a single, full-range drive unit using a flat diaphragm. He argues that the use of a single drive unit rather than a multiple system with crossover makes a lot of sense with the present, almost universal, use of direct-coupled amplifiers since it preserves a high electrical damping factor. He goes on to suggest that a valid alternative to present design philosophies is to start with a good mid-range unit and then concentrate on extending its performance in the upper and lower octaves. This will lead to a sharper stereo image since the most crucial musical and spatial information is carried in the middle frequency range, and he believes the effect will be best achieved by making the sound source substantially narrower than the distance between human ears.

The result of his endeavours is a drive unit with a flat, 'lozenge shaped' diaphragm driven by a similarly elongated coil. The long thin shape fulfils the requirement that, when positioned vertically, the diaphragm is considerably narrower than the distance between human ears, and in addition it ensures that the diaphragm is fairly evenly driven since no part of its surface is more than 10 mm from the coil. The flat diaphragm does not have the stiffness of a



A Musician drive unit in an enclosure made of the cement-based inorganic plastic, NIMS 127.

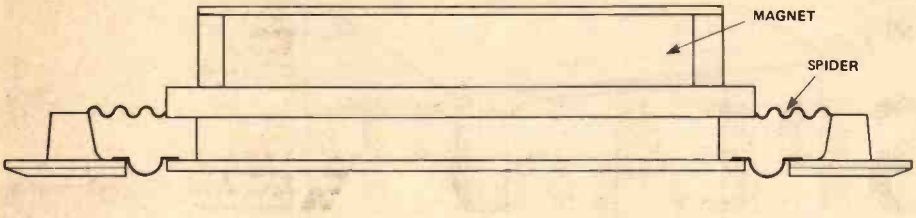


Fig. 1 Vertical cross-section through the Musician drive unit.

cone, but the more evenly-distributed drive overcomes this disadvantage to some extent, and the use of a stiff, lightweight, laminated plastic with a degree of self damping helps ensure that the inevitable flexing is not catastrophic.

The use of a large coil and relatively small diaphragm area does introduce certain limitations. Unless it is to make impossibly large excursions, a small diaphragm cannot produce high acoustic power levels. This prevents the unit delivering the power levels demanded by some heavy rock enthusiasts, but for most other tastes the sound level in a domestic room of average size should be satisfactory. The other limitation is the sensitivity. A low sensitivity is quite normal for small loudspeakers, and even with the substantial magnets used in the prototypes the sensitivity did not exceed 86 dB for 1 watt at 1 metre on axis. This simply means that the loudspeakers sould be driven by an amplifier offering in excess of 40 watts per channel, not an uncommon output level for a modern, high quality amplifier.

The Musician Loudspeaker, at the expense of introducing these minor limitations, has been designed to offer a sound which is largely free of colouration. There is little point, therefore, in mounting such a drive unit in a conventional resonant enclosure. There are two major sources of resonance in a loudspeaker enclosure, cavity resonance generated within the space and resonances set up within the panels of which the enclosure is constructed. Both of these sources have been considered and dealt with in arriving at a design which complements the low colouration of the drive units.

Considering first the question of cavity resonance, standing waves always occur in an enclosure which has flat, parallel

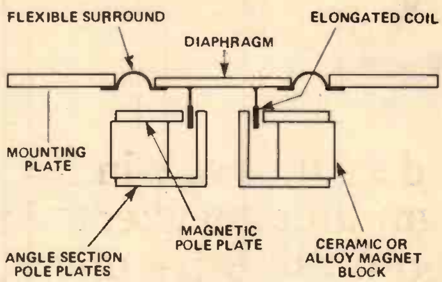


Fig. 2 Horizontal cross-section through the Musician drive unit.

interfacing surfaces. The wavelengths of the resulting resonances will be comparable to the distance between the faces.

Thus in a rectangular box 300 x 250 x 200 mm there will be resonances at approximately 1125, 1350 and 1700 Hz. Absorbent filling in the cavity will reduce these resonances but will not kill them altogether. The designers of musical instruments avoid the formation of standing waves by making their sound boxes in irregular shapes eg, the violin, the 'cello, the lute and the guitar. Reflections inside these soundboxes are so diffuse that standing waves cannot occur. An instrument with a rectangular sound box would produce some notes much more loudly than others.

It is inconvenient and impractical to house loudspeaker drive units in irregularly shaped enclosures but the same effect can be obtained by inserting an irregularly shaped object inside the cavity which similarly diffuses reflections between the inner surfaces. The 'sound splasher' used for this purpose is shown in Fig. 6 and simply consists of a number of lengths of square section timber glued together to form a spiral. BAF wadding is used in the normal way to fill the remainder of the cavity.

In a conventional box with mitred or butt jointed corners, each panel is rigidly held at all its edges and is readily excited into resonances, the frequencies of which will depend upon its mass, its stiffness and its dimensions. Again taking a lesson from the designers of musical instruments and noting that a piano string is damped by a resilient pad at one of its ends, the panels of the boxes are separated by a 1 mm gap and fixed together with cork fillets, giving them a degree of acoustic independence and providing a small but useful amount of damping.

The enclosure material must also be carefully chosen if excessive resonance is to be avoided. The best materials are heavy and stiff, and concrete and sand filled panels are among those which have been tried. Glass, ceramics and heavy metal plate are all better than timber or common chipboard, and very good results have been obtained using large (300 x 200 x 8 mm) Italian ceramic floor tiles. The constructional techniques outlined above can be used with all these materials.

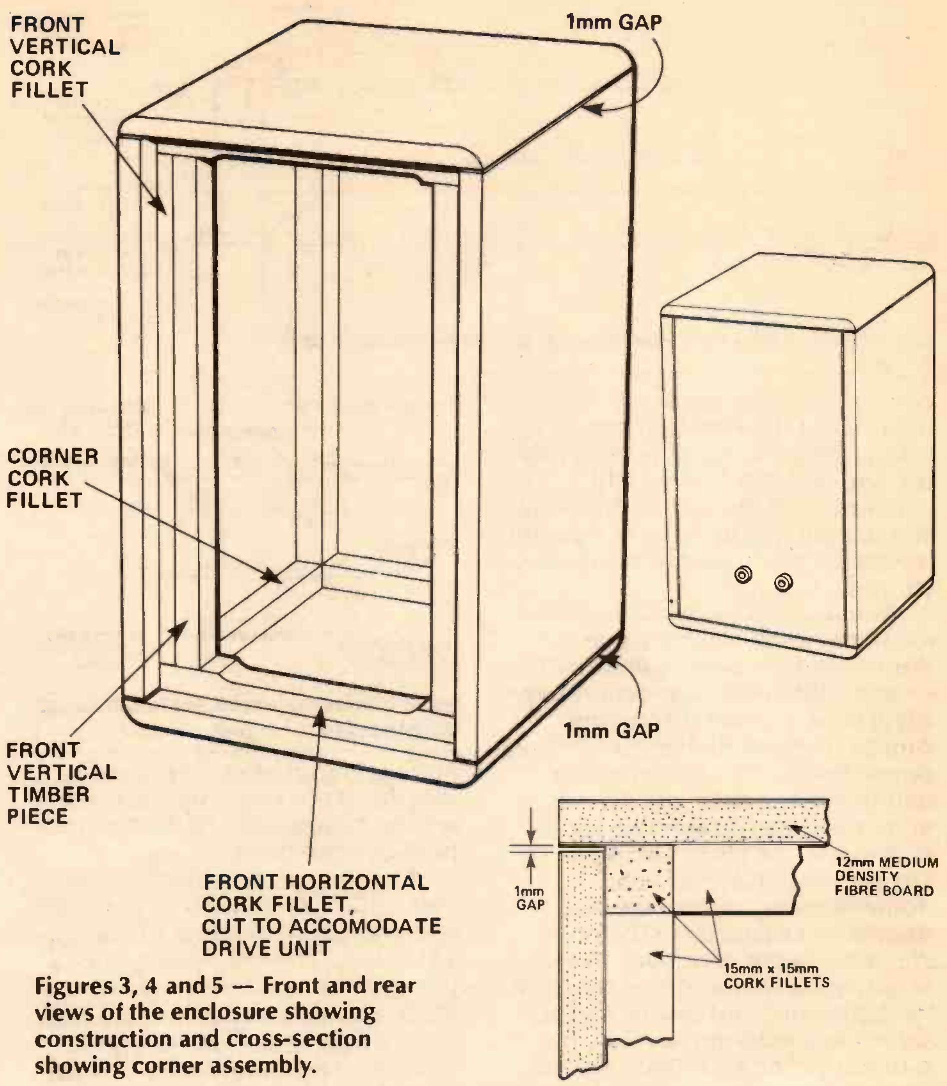
The material most favoured by Carl Pinfold is a cement-based inorganic plastic called NIMS 127 which has been developed by Professor Birchall working for ICI. Cement mortar normally has no tensile or compressive strength and is prone to crumbling, the result of having large spaces between its molecules. By mixing a polymer additive with simple cement, Professor Birchall has produced a material which has the strength of aluminium plate, can be readily machined, cast and extruded, and which can be produced in any colour simply by adding pigments into the mix. It is fireproof, waterproof and requires a minimum of energy in its manufacture, but its most attractive features from the loudspeaker designer's point of view are that it is both dense and stiff and hence acoustically very non-resonant. Unfortunately, this material is not yet available to the general public, so the enclosure design presented here is that of Carl Pinfold's 'Basic' enclosure which is constructed from Medium Density Fibre board. Ordinary particle board could also be used, but will not give quite such good results and will need to be at least 18mm thick.

The enclosure to be described is of the fully-sealed or 'infinite baffle' variety. It has been designed principally for use with the Musician drive units which have a free-air resonance at 42 Hz rising to a modest peak at 70 Hz when so enclosed. The peak is not pronounced, and there is a good output level at 50 Hz, at least as much as is to be expected from an enclosure of this size. Carl Pinfold expresses a perference for the well-damped bass provided by a totally air-tight, wadding packed box, but argues that room acoustics and 'speaker position are so crucial to frequency balance that there can be no general rule as to which loading method will be best for any given circumstances. He suggests that many of the differences between loudspeakers are less pronounced than the differences introduced by room acoustics and position, and that this may account for the wide variety of opinions about loudspeaker performance. There is no reason why the general building techniques described in this article should not be used in the construction of bass-reflex enclosures, although they would not be so easy to apply to transmission line designs. There is also nothing to stop you using this enclosure design with drive units other than the Musician unit, but note that you will then require a front panel since the Musician drive unit normally forms the front panel itself.

Construction

Begin by taking a suitable quantity of 12mm Medium Density Fibre (MDF) board and cut out the sides, top, bottom, back and, if you are using one, the front panel. Even though gaps are to be left in the finished enclosure for the reasons discussed earlier, it is still important to cut the panels as accurately as possible since this will give the neatest end result and avoid any awkward problems arising at the assembly stage. You should also prepare in advance the cork fillets, which can be obtained by cutting up an old (or even new) cork bathmat, and again you should cut it as accurately as your skills and the vagaries of the material will allow. The only other things you will need for the actual cabinet assembly are a suitable glue, such as Evo Wood Adhesive or another resin glue, some small strips of Formica or any other material about 1 mm thick, and lengths of tape or string to hold the panels together while the glue is drying.

Check the accuracy of your cutting by loosely supporting the panels in place next to one another and, if necessary, swap over matching panels so as to obtain



Figures 3, 4 and 5 — Front and rear views of the enclosure showing construction and cross-section showing corner assembly.

PARTS LIST

(one loudspeaker only)

Panels (all cut from 12 mm Medium Density Fibre board):		(2 off)	15 x 10 x 250
Back	280 x 165	Front Horizontals (2 off)	15 x 15 x 167
Sides	280 x 228		
Top/bottom	228 x 190	Timber pieces:	
Cork fillets:		Front verticals (2 off)	18 x 10 x 250
Corners (4 off)	15 x 15 x 165	'Splasher' (5 off)	18 x 18 x 130
Rear verticals (2 off)	15 x 15 x 220	Miscellaneous	
Rear Horizontals (2 off)	15 x 15 x 167	4 mm terminals, 2 off; one third of a square meter of BAF wadding; drive units; connecting wire; open-weave hessian for grille cloth.	
Front verticals			

the best fit. When you are satisfied that all is well, lay out the panels on a flat surface in their correct order, ie, as though in an 'exploded' diagram, with the back panel in the middle. Assemble the cork fillets on the back panel so as to form the frame, glue them in place, then offer up the side panels and space them away from the back and from one another using the thin strips of Formica. Assemble and glue the panels and the rest of the cork frame and then tape or tie the structure so that it cannot move whilst the glue is drying. With most adhesives it is best to allow overnight drying, but obviously this depends upon the type you use.

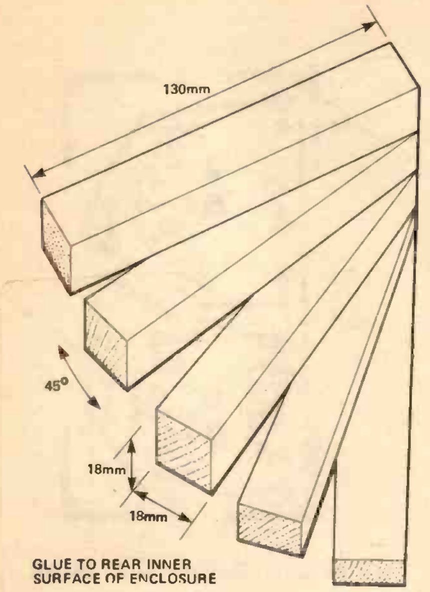
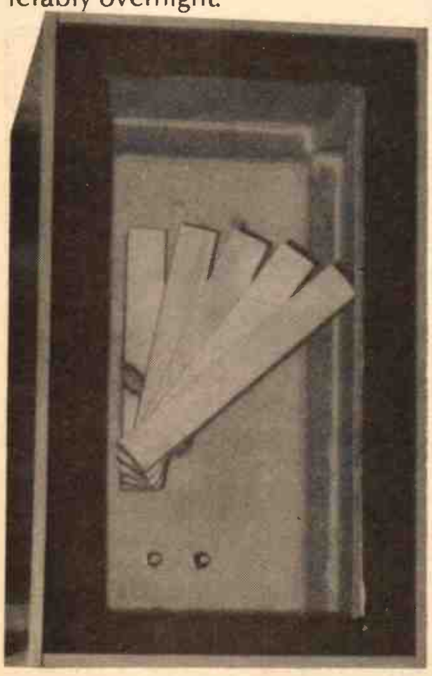


Fig. 6 The 'splasher'.

While the glue on the enclosure is drying, you can get on and produce the 'splasher'. As explained, this simply consists of five pieces of wood laid one-on-top-of-the-other at one end and staggered so as to form a spiral. You will need a clamp or a bench vice or some other means of holding the pieces in the correct position while the glue dries, although it should not be too difficult to devise another means of support if neither of these is available. As with the enclosure itself, the 'splasher' should be left to dry for a reasonably long time and preferably overnight.



Internal view of an enclosure made of NIMS 127 showing the cork fillets and the 'splasher'. The same construction is used for the MDF board version.

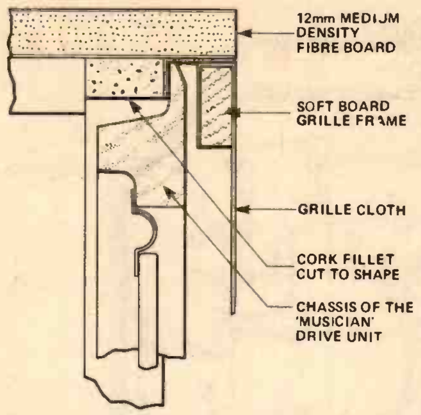


Fig. 7 Vertical cross-section through the enclosure with the Musician drive unit in position.

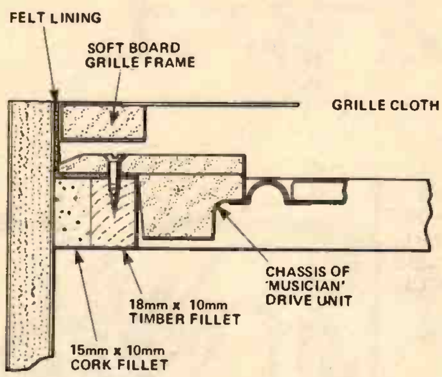


Fig. 8 Horizontal cross-section through the enclosure with the Musician drive unit in position.

When the glue on the enclosure is dry, remove the binding and then seal all of the inside joints with a flexible mastic sealant so as to make the final unit air tight. Depending upon the type of drive unit you are planning to use, identify the spot on the inside back panel directly opposite the driver coil and stick the 'splasher' down. The enclosure can be left on its back whilst the glue dries, but you will still need to tape the 'splasher' in place to stop it tipping over.

If you are using a drive unit other than the Musician loudspeaker, you can now glue the front panel into place. Remember to leave 1 mm gaps around the edges just as you have done with all the other panels. If you are using the Musician drive units, glue the two vertical timber pieces to the cork fillets at the front of the enclosure and support them in place until the glue is dry.

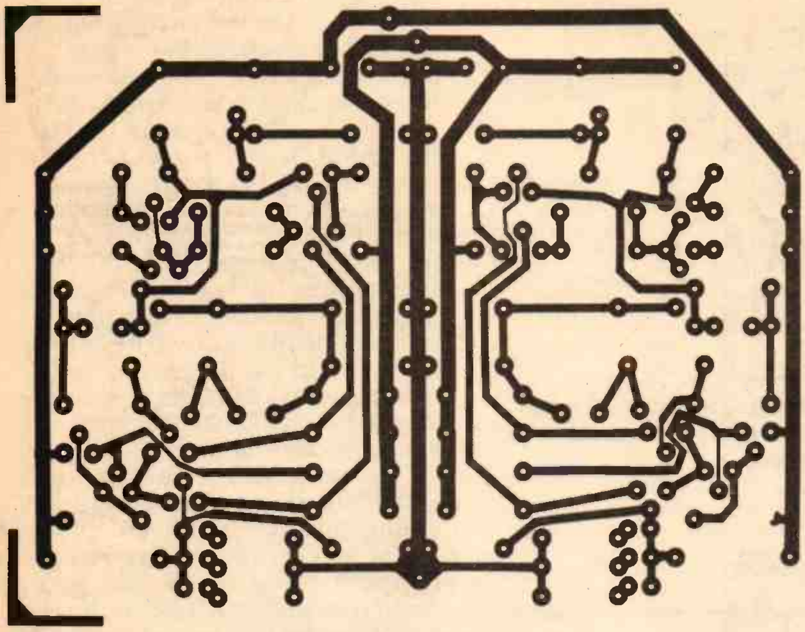
When the final glueing stage is over and the glue is dry, you can finish the exterior surfaces of the enclosure to your taste, either by staining the wood or painting it or by covering it with veneer or fabric. If you do cover the enclosure with something like veneer remember to expose the gaps between the panels and don't just cover over them. Stick a layer of felt to the inside faces of the front of the enclosure, and make up a simple wooden frame to hold the grille cloth. Provided you make the frame the right size, the cloth will bind against the felt and hold it in place and no other means of support will be necessary. The cloth itself should be as acoustically transparent as possible, and open-weave hessian or something similar is recommended.

The only tasks remaining are to install the drive unit and the input connectors and wire them up. Two 4 mm sockets are ideal as input connectors and only require two small (usually 5/16") holes. If you are using the Musician driver unit, position it within the front of the enclosure and line it up so that it is in the centre of the space. Six holes can then be drilled into the two vertical battens and wood screws used to hold the driver in place. If you are using any other drive unit you will have to devise your own mounting system but for obvious reasons you should avoid using a drive unit which can only be mounted from the rear of the panel. Finally, wire up the drive units to the terminals, loosely fill the cabinets with BAF wadding, secure the drive units in place and then sit back and enjoy them.

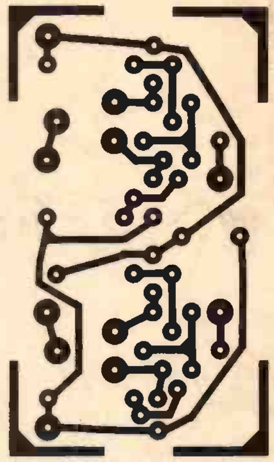
BUYLINES

Medium Density Fibre (MDF) board is widely available from DIY and timber shops, and if you don't have an old cork bathmat to hand you should have no trouble finding someone to sell you a new one. The BAF wadding and the hessian should both be available locally but if you encounter problems you could try Wilmslow Audio who certainly have the wadding and probably have a suitable grille cloth. The Musician drive units are available from Merseyside Acoustic Developments, Merseyside Innovation Centre, 131 Mount Pleasant, Liverpool L3 5TF, tel 051-709 0427, and cost £160.00 per pair inclusive of VAT, carriage and packing. If you don't feel like doing all the hard work but would still like a pair of Musician loudspeakers, they can be obtained from the same address, prices for the basic MDF board enclosure or the NIMS 127 enclosures are available on application.

PCB FOIL PATTERNS

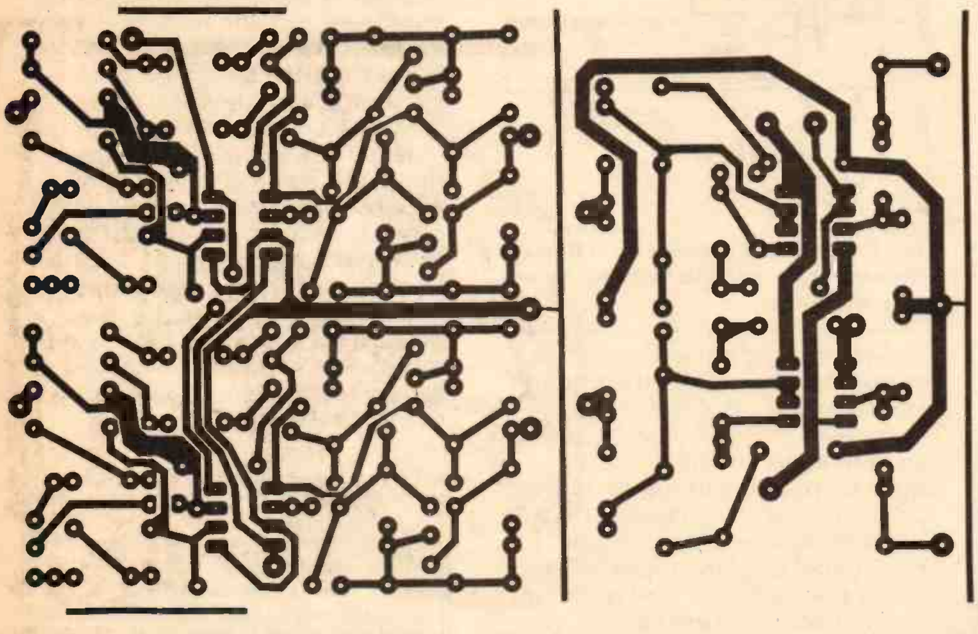


The RIAA input board of the JLLH amplifier.

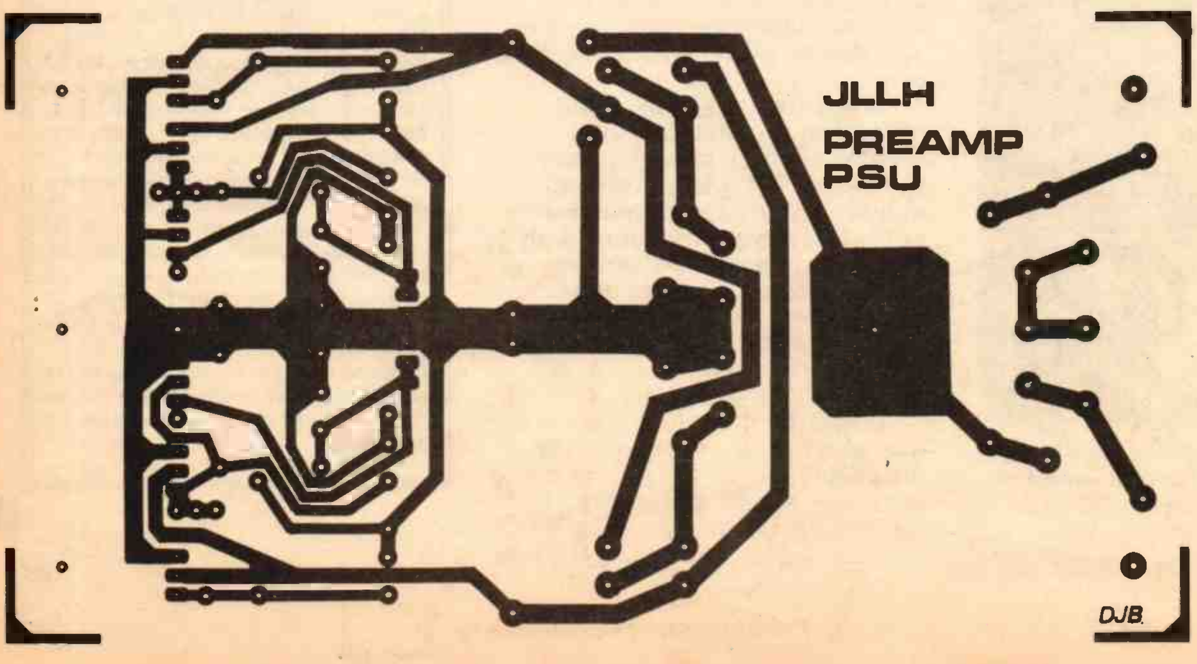


The Audio Design buffer board.

All the boards on these two pages are for the John Linsley Hood Audio Design Amplifier; all the corrections noted in the PCB overlay captions have been made on these boards.

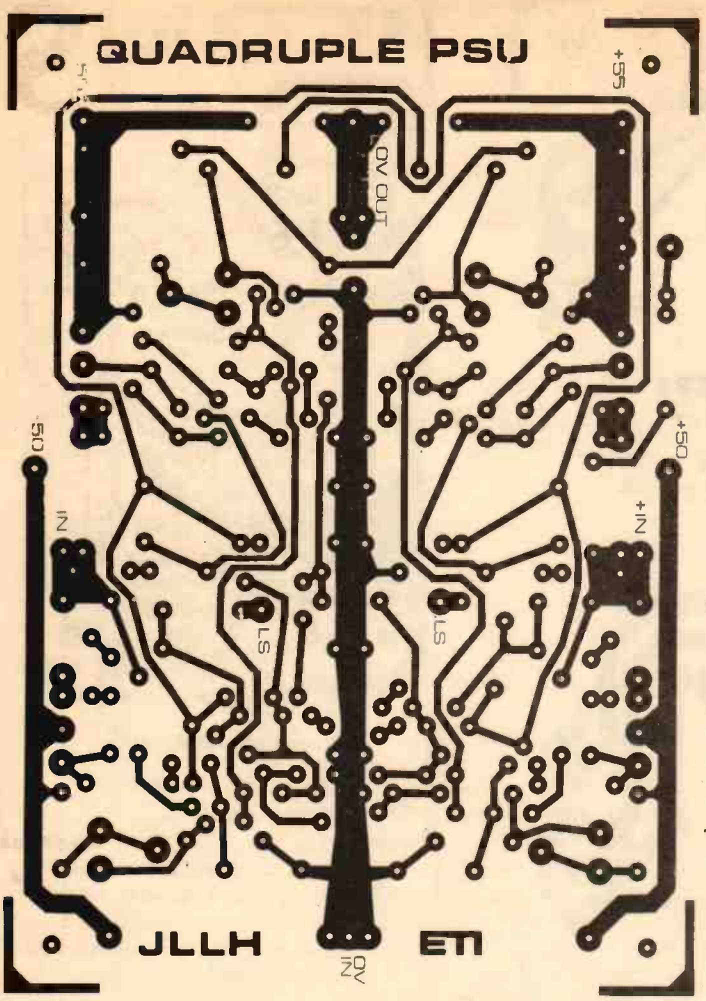


The buffer/filter and tone control board of the JLLH amplifier.

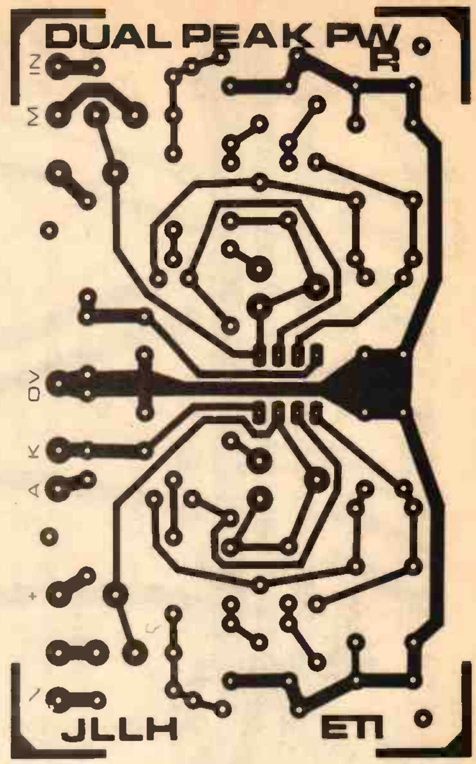


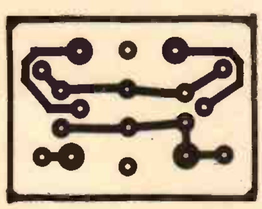
The PSU board of the JLLH amplifier.

the power meter.

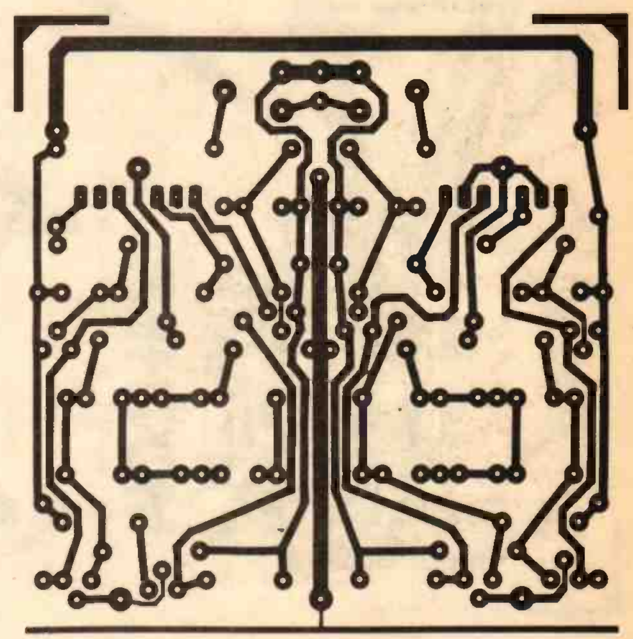


The Audio Design PSU board.



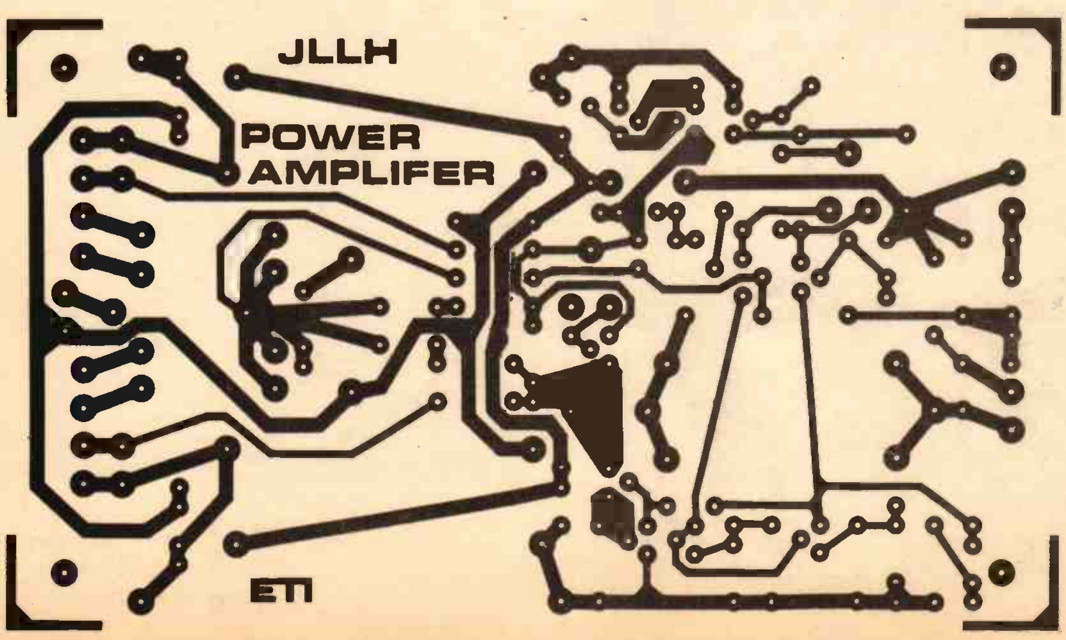


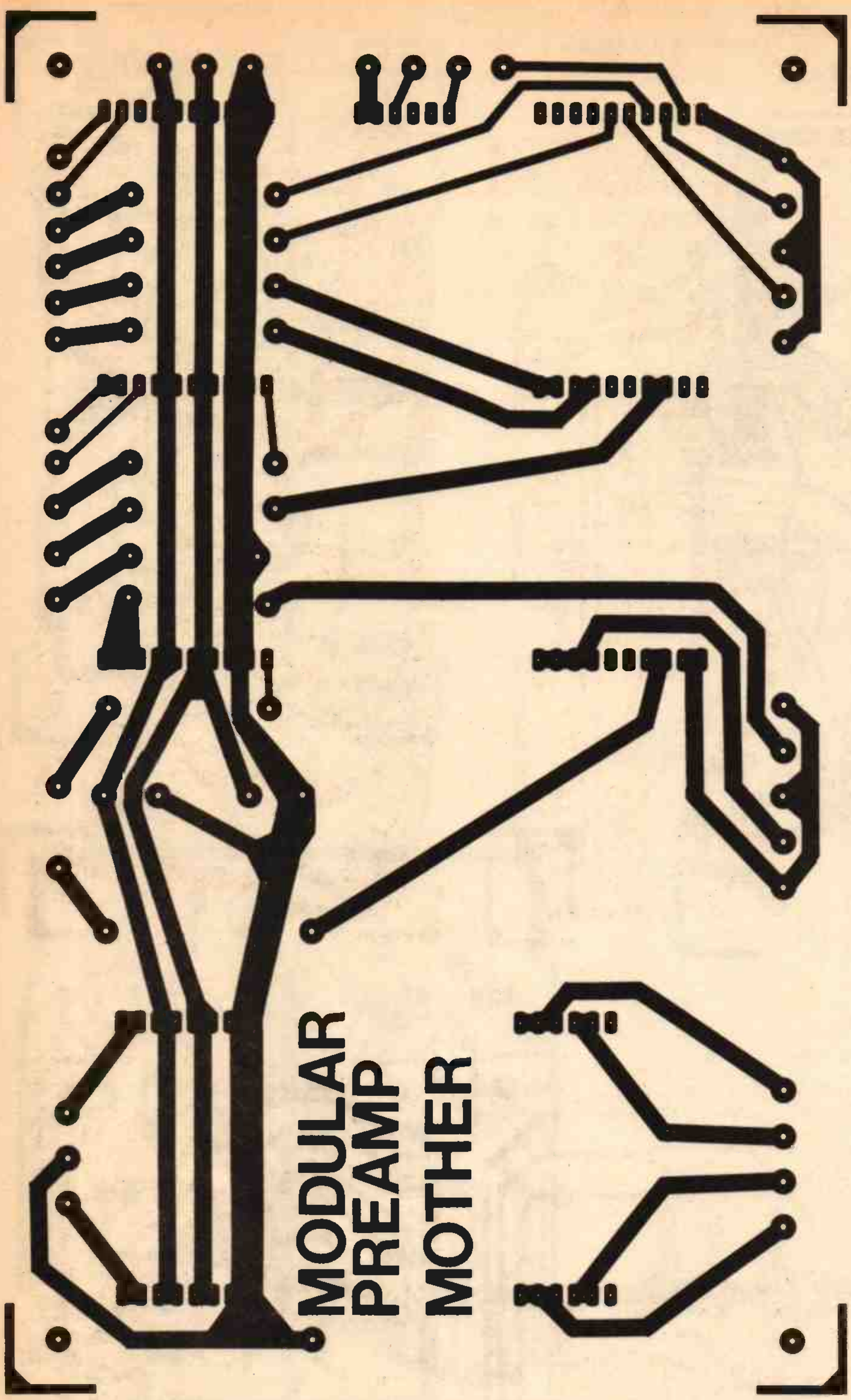
FET input clamp circuit.

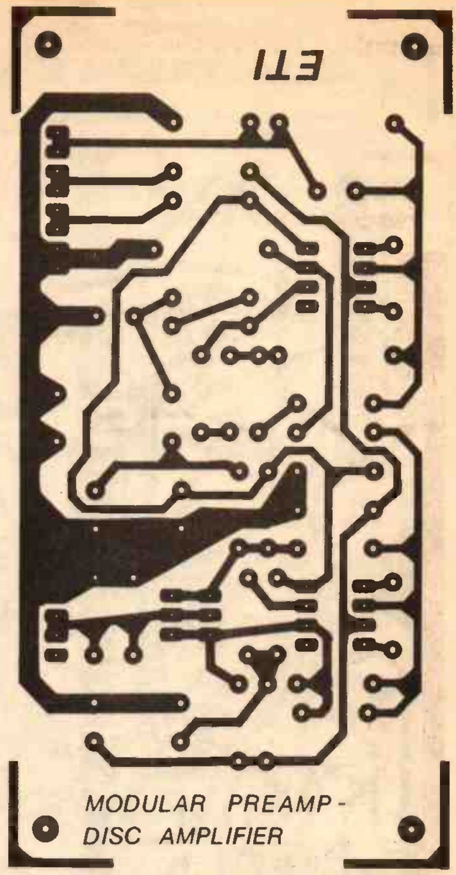


The headphone amplifier board of the JLLH amplifier.

The Audio Design power amplifier board.

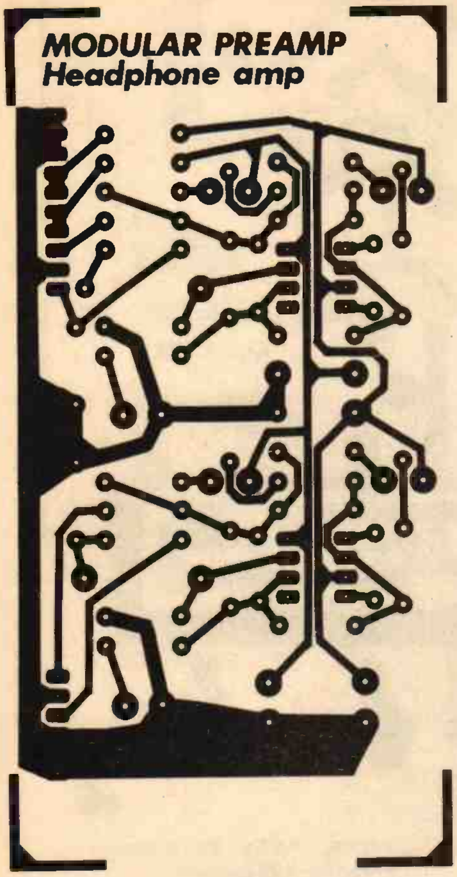


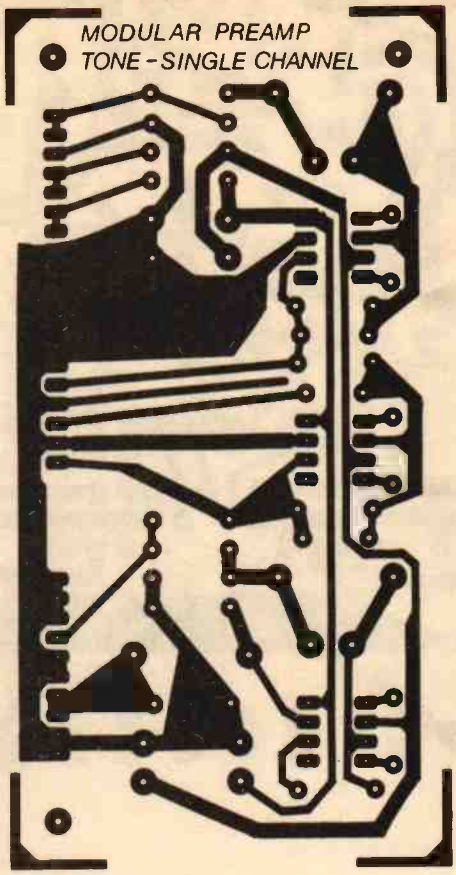


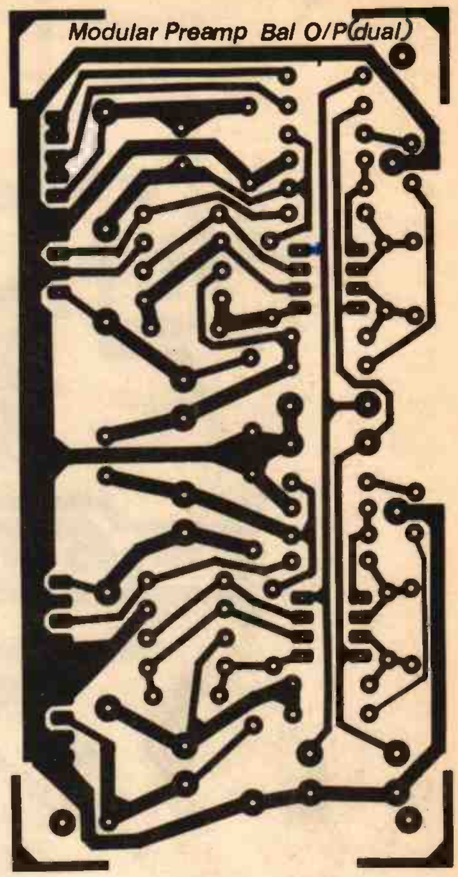


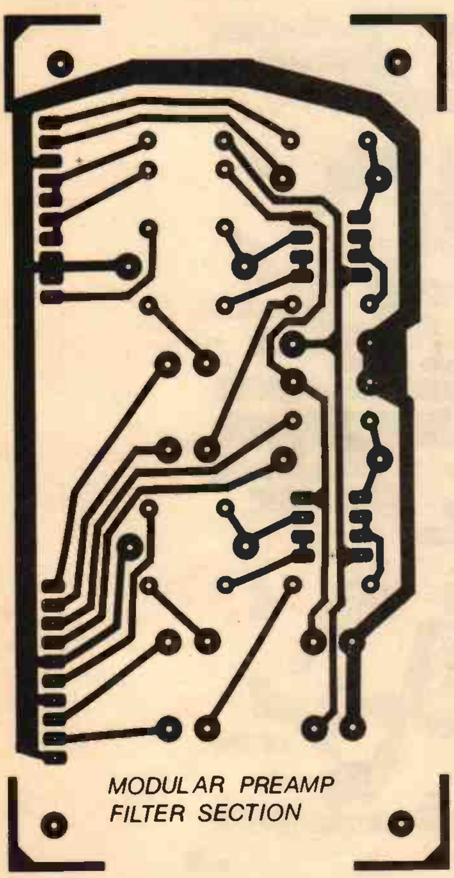
All the boards on this spread are for the Modular pre-amplifier.

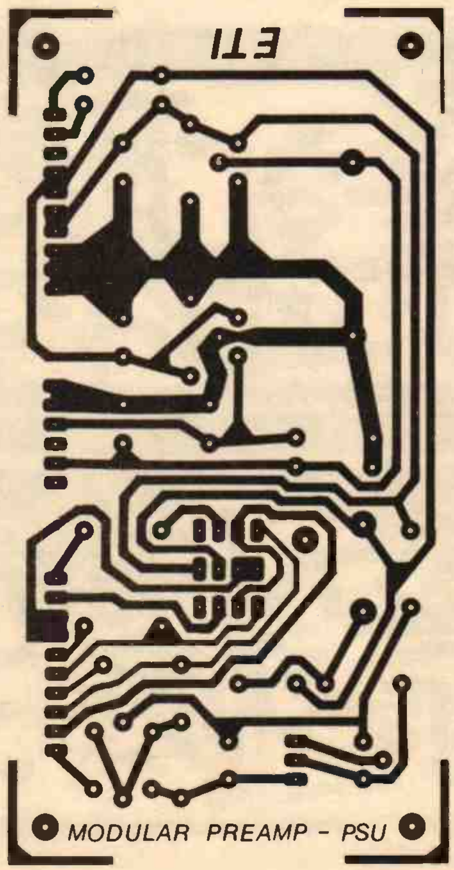
The Modular Preamplifier boards.

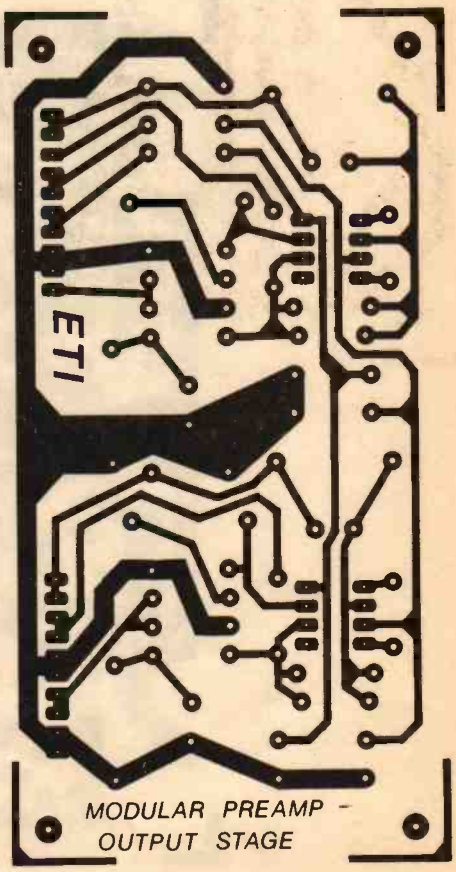


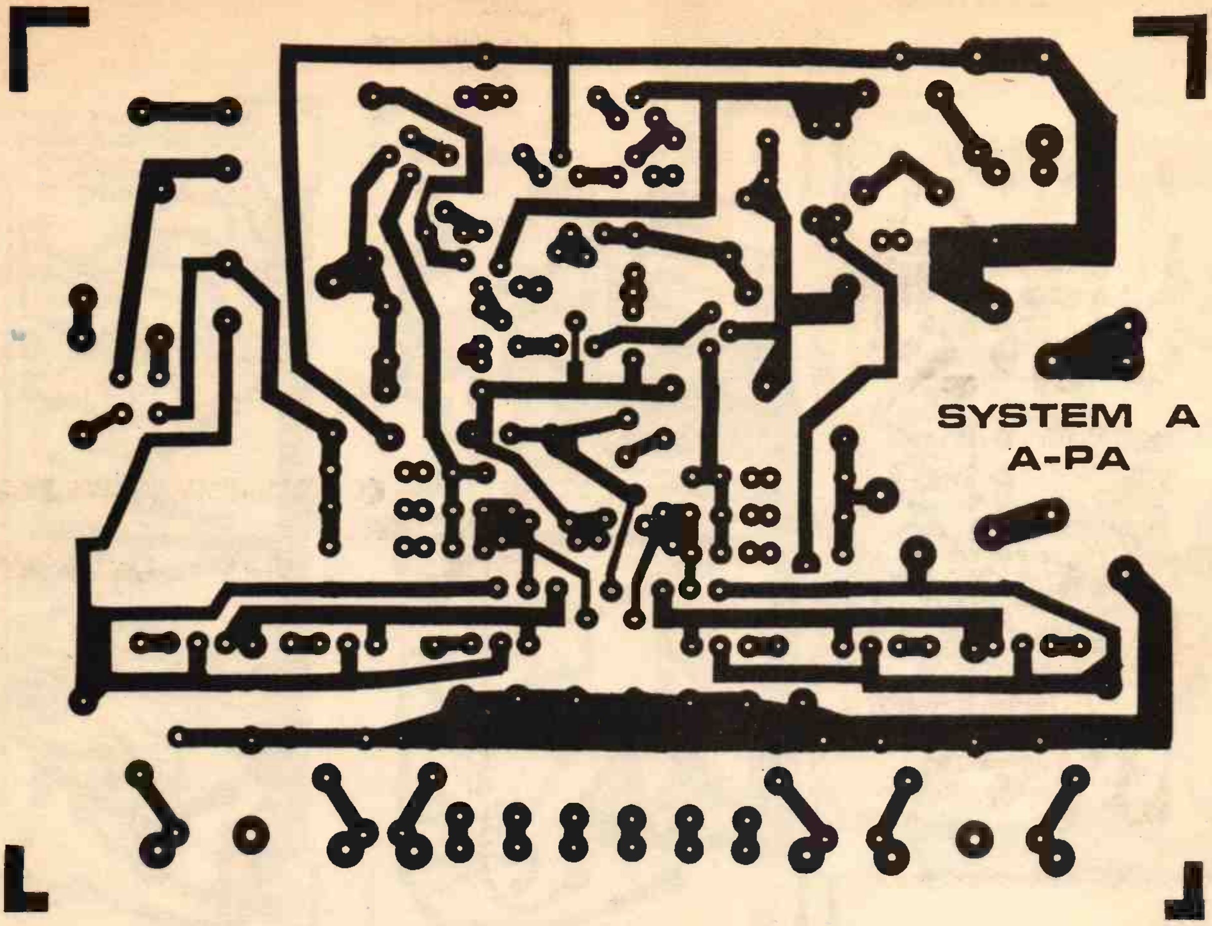




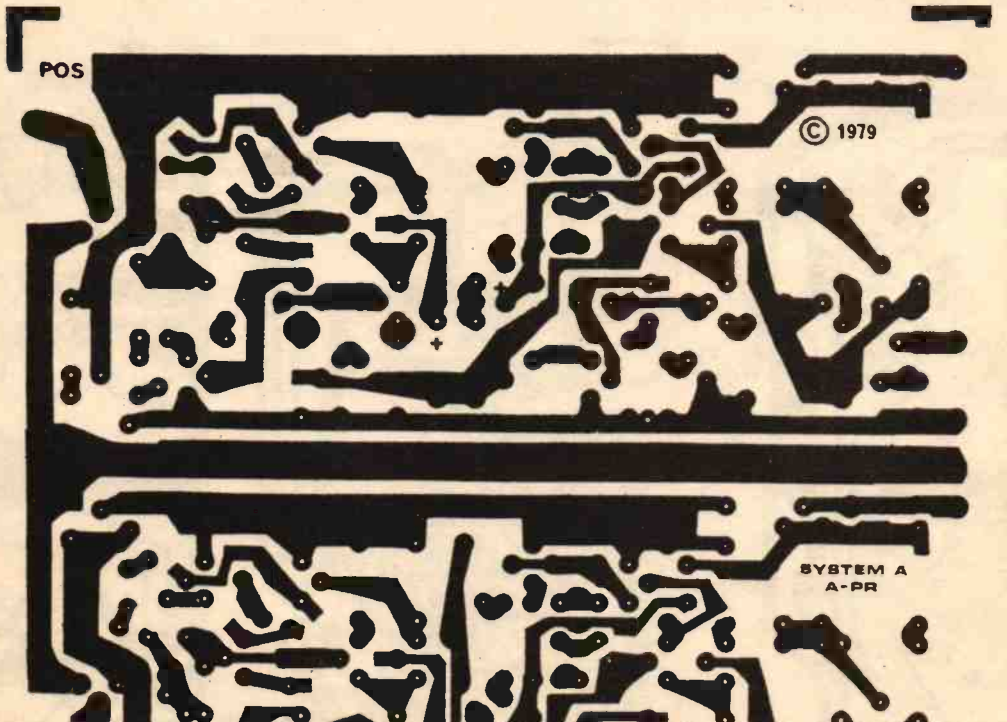




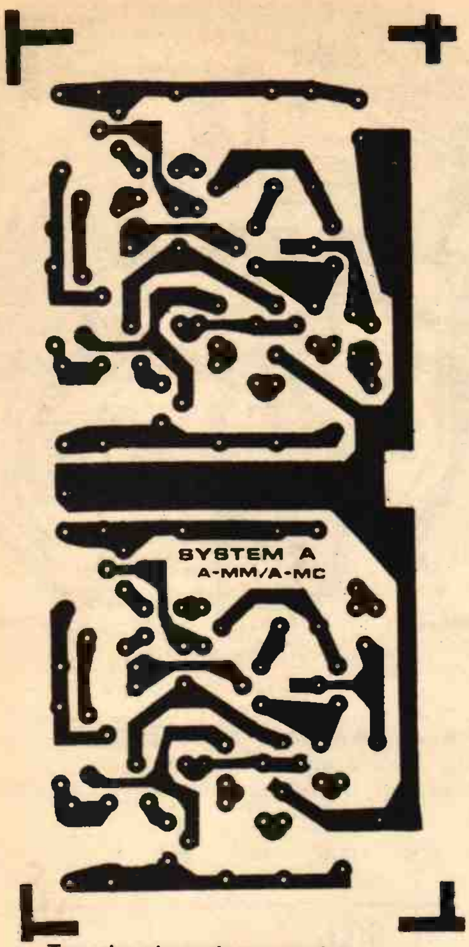




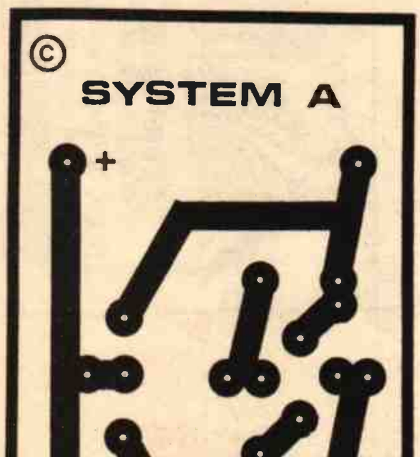
Above: the System A power amplifier foil pattern.

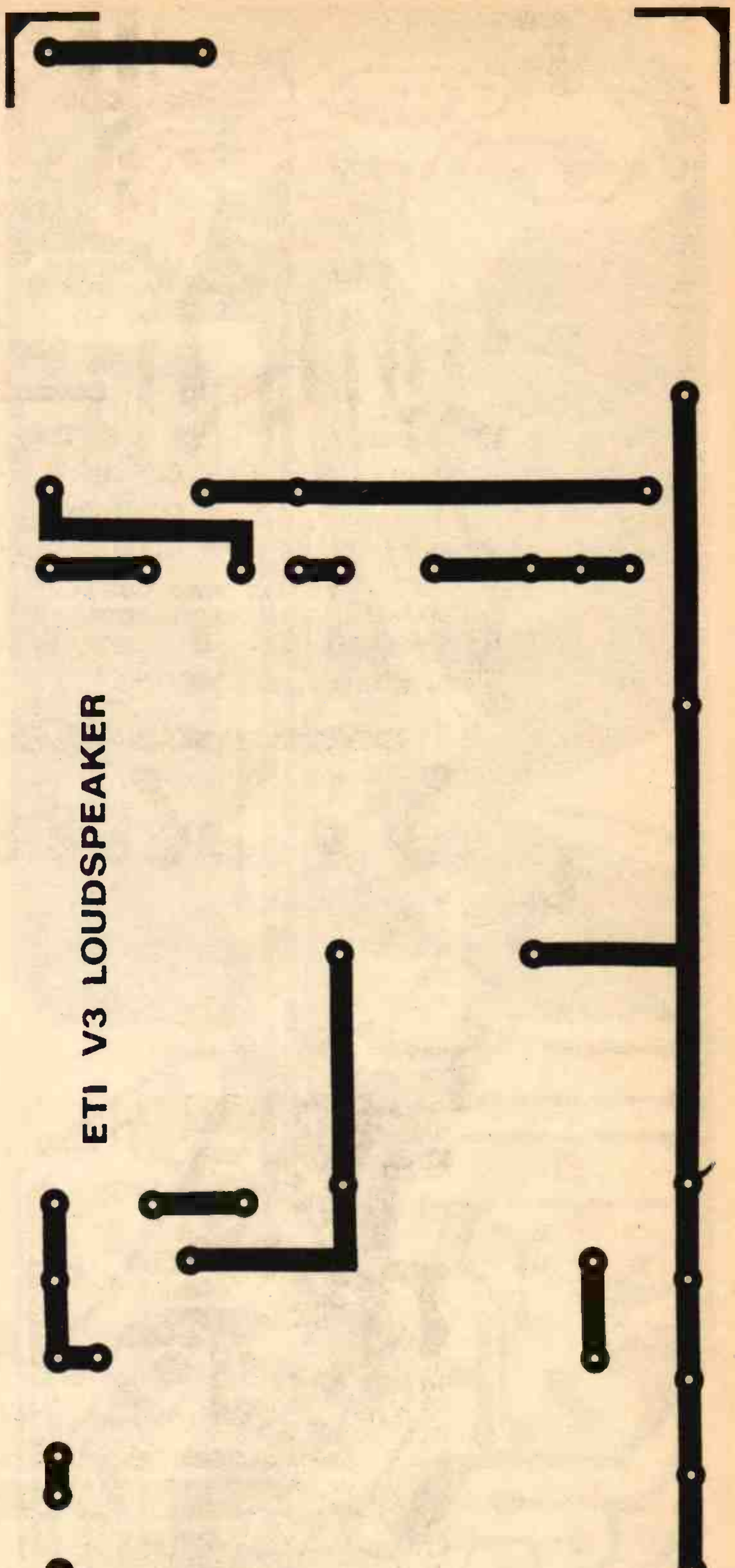


Above: the System A main pre-amplifier board.

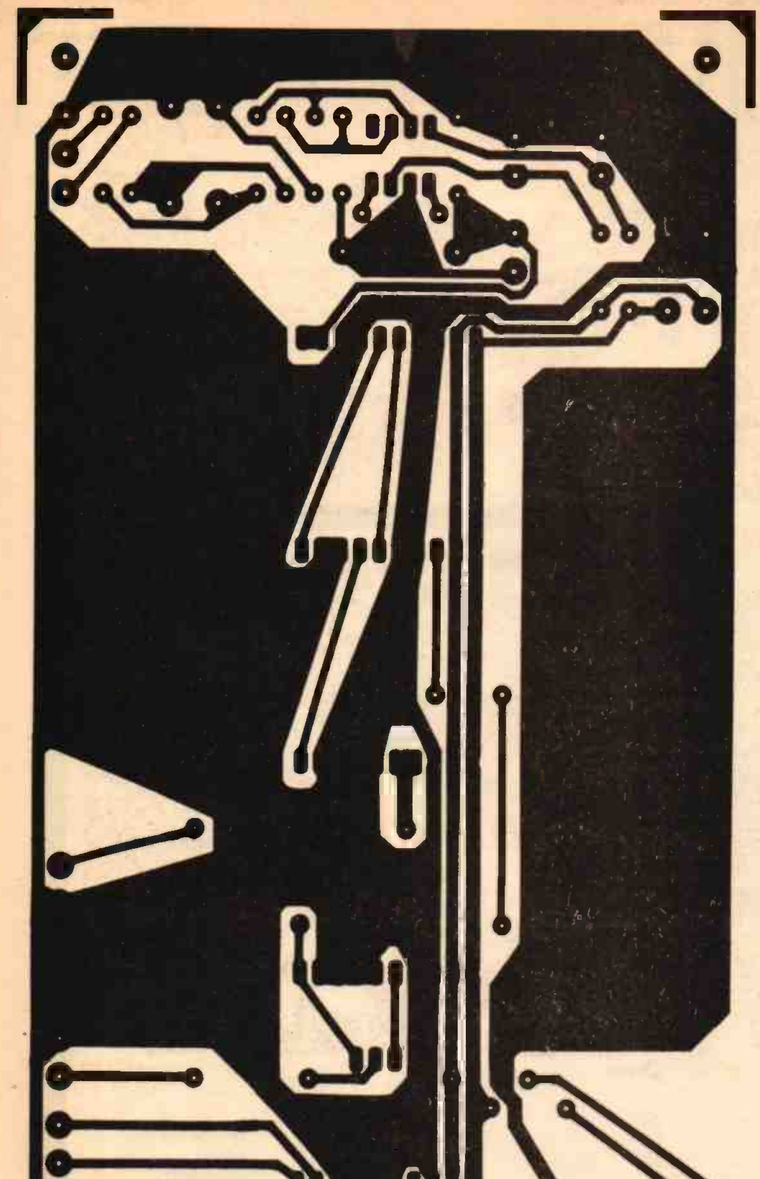


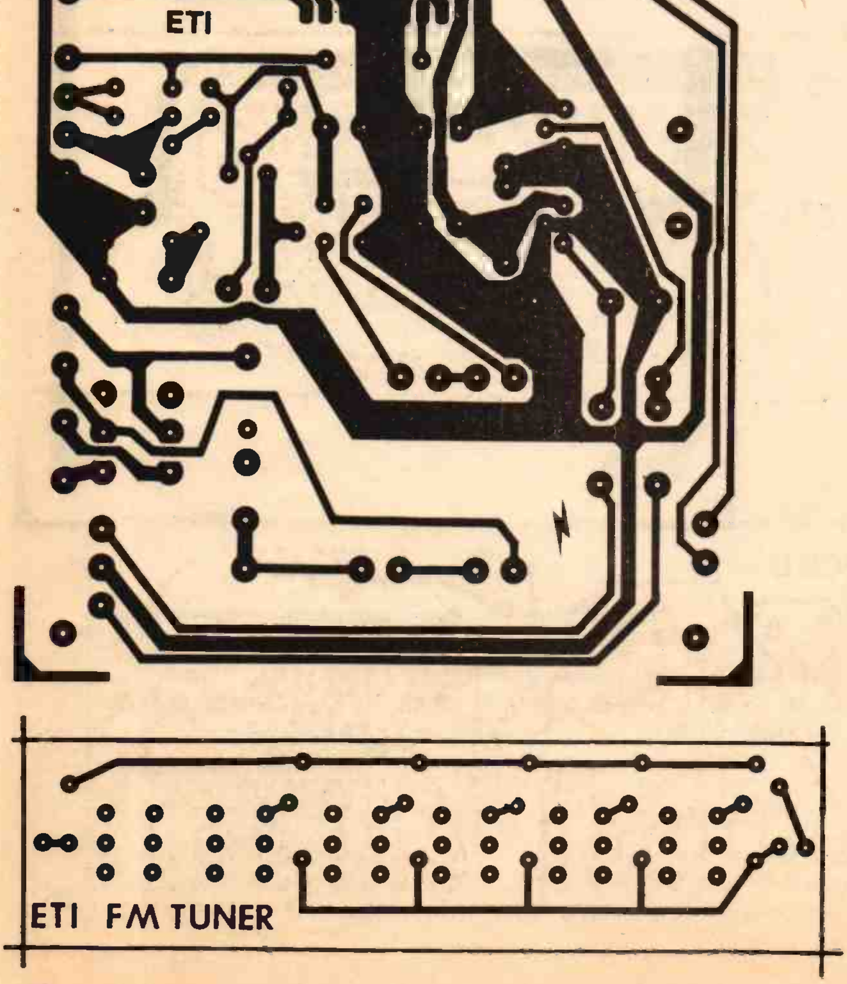
Top: the phono input module.
Below: the System A pre-amplifier power supply foil pattern.

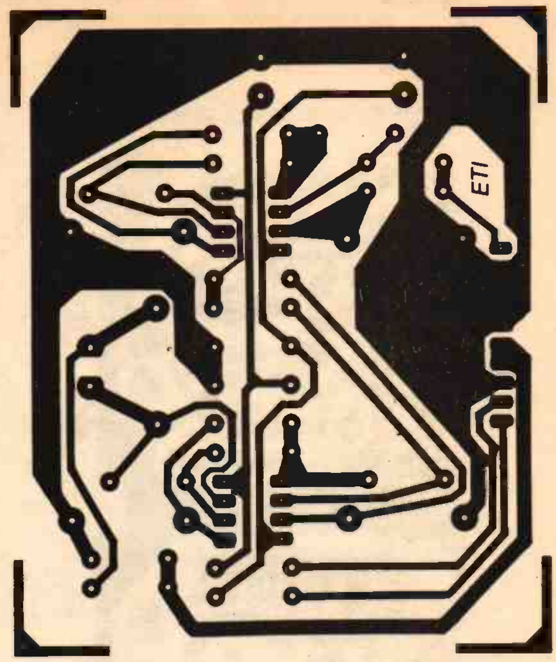




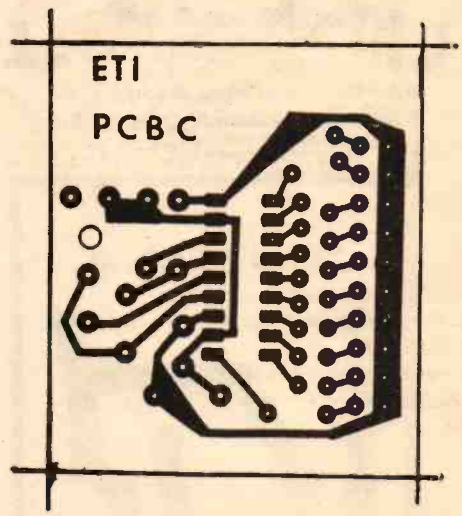
Above: the V3 loudspeaker system crossover PCB.

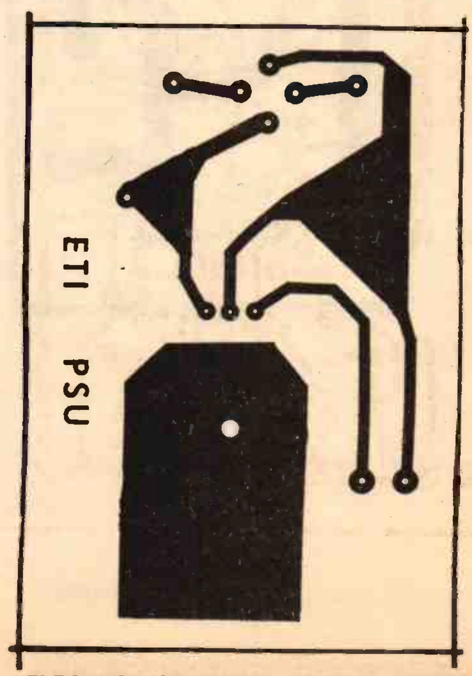


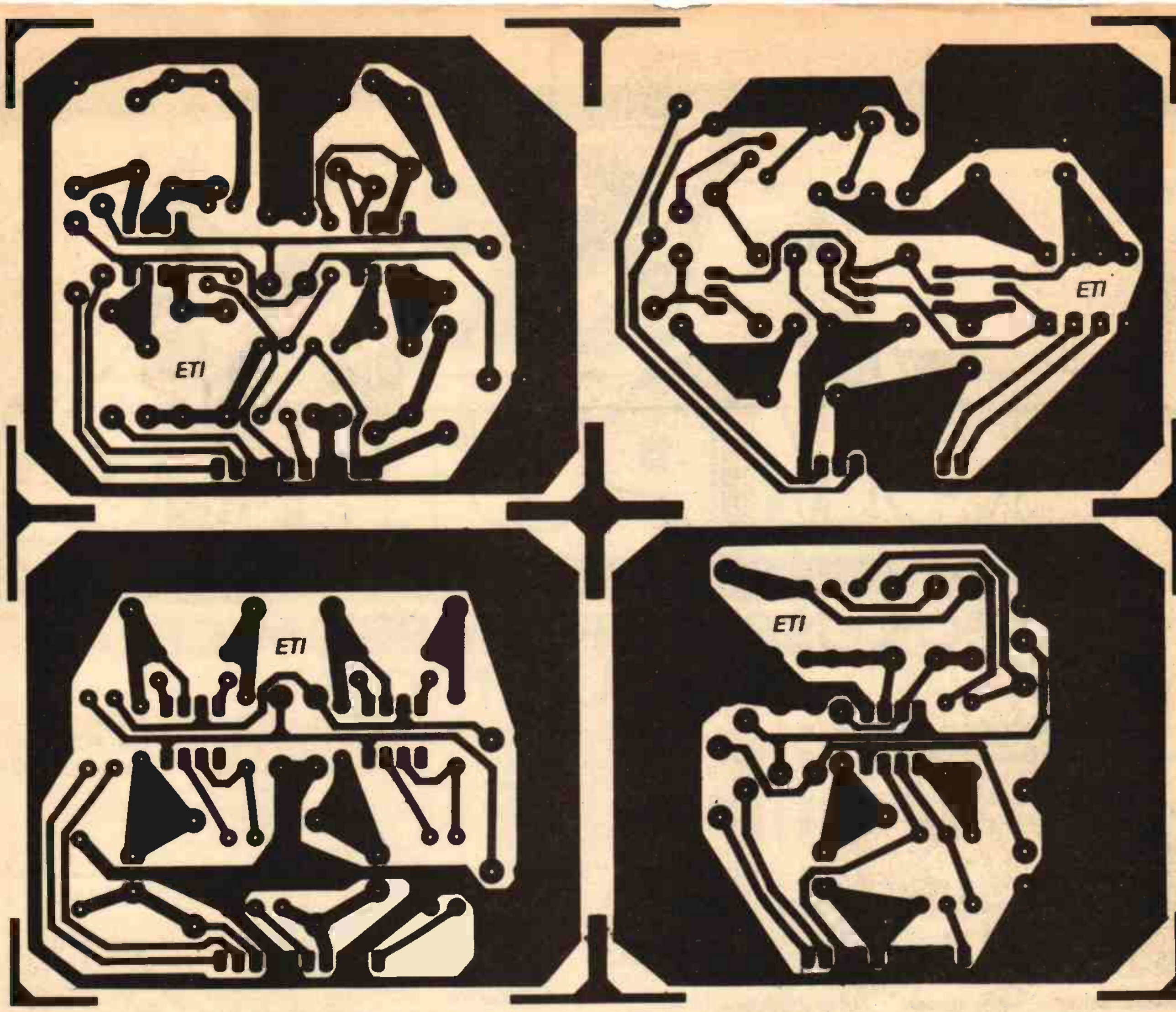




Top left: the Active 8 motherboard.
Above: the Active 8 delay unit PCB.
Below left: the Audiophile tuner switch board.







Above: the four plug-in modules for the Active 8 loudspeaker.
Left: the FM tuner 'C' board and (below left) the PSU.
The remaining boards on this page are the Audiophile FM tuner 'A', 'B', and 'D' PCBs.