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COMPILED BY ROGER HARRISON

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from ETI

Roger Harrison
Editor

As so much of our reader feedback indicates, audio projects are very high in the project popularity stakes! That's probably because so often they serve a very practical purpose — you can build your own audio gear for much less than you can buy the commercial equivalent. And building your own audio projects can provide an immense amount of satisfaction.

The projects gathered together for this compilation provide for a wide range of interests in the audio sphere — from hi-fi sound, through stage and "pro" sound applications, to communications. Some are complete, stand-alone devices, other projects have been designed as an adjunct or add-on to some sort of sound system.

The projects included here range from the simple and immensely popular "Digi-125" Audio Power Amp Module, through several hi-fi loudspeakers using the high-quality European Vifa range of drivers, to guitar sound projects. So, whatever your interest in audio may be, doubtless you'll find something of interest and of use in these pages.

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ETI-1417: VIFA SA-100 SPEAKERS

Build these high performance speakers at a low budget price

Tom Manning

THIS IS THE GOLDEN age for the Hi-Fi enthusiast. The recent introduction of sophisticated musical recording and playback technology makes it possible for modern budget equipment to outperform the very best of front ends available just ten years ago. Digital recording techniques, the compact disc, Domestic Digital Audio Tape (DAT) recorders plus an extremely high standard of analogue (via L.P.) reproduction make a first class audio front end available to many.

However, this performance per dollar factor simply does not, and never has, extended to the loudspeaker — the quality of this electro-mechanical system has always been price proportionate. Today, awareness of the inherent shortcomings of (particularly) low cost speakers is prompting many enthusiasts to seek alternative ways to achieve high quality sound for a reasonable outlay. Not surprisingly, home built speakers, (very popular some years ago) built from professional designs are enjoying a healthy renaissance. The Melbourne loudspeaker distributor, Scan Audio, has just released a high quality DIY speaker, locally designed using Danish Vifa drivers. This system, the SA-100, loosely based on a much earlier and very popular design using similar components, compliments the SA-70 bookshelf system described last month in ETI.

First Design Considerations

To compliment both the Vifa SA-70 and SA-50 designs (the latter a high quality true miniature speaker) substantial increments in SPL (Sound Pressure Level), bass end performance, and overall efficiency were seen to be both desirable and logical. To achieve a worthwhile increase in bass capability, a proportionate increase in cabinet volume is mandatory — after predicting the performance of many drivers in various enclosures one combination seemed ideal. The Vifa P-21 bass-midrange unit, housed in a 42 litre vented enclosure would give us the desired bass end capability with excellent transient performance and high power handling.

Alignment

It's worthwhile looking at the steps used in determining an "alignment" (suitability of matching a bass driver with a cabinet) — to appreciate this an explanation of some basic parameters will help. The key points are:

1. The free air resonance (Fo), which is the frequency at which the moving system of the driver exhibits the maximum output for the minimum electrical signal.
2. The total Q factor of the driver (Qt), which is the quality factor at resonance, and which indicates the

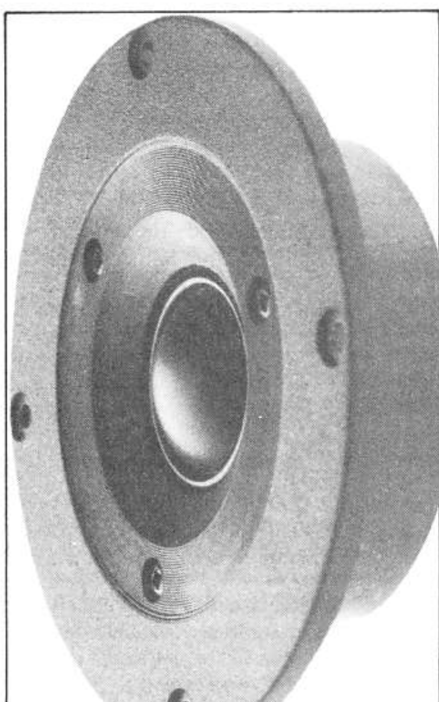
combined effects of the driver's electrical and mechanical damping of the moving system at its free air resonance.
3. The VAS, which is the volume of air in litres need to provide the same restoring force to the cone as does the suspension. Expressed more generally, an indication of the "springiness" of the suspension system.

We can now predict:

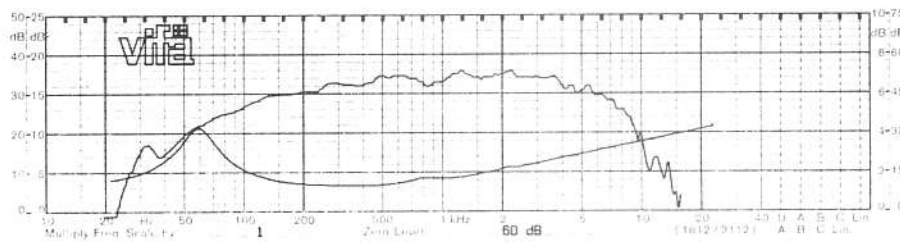
1. Vb, the optimum cabinet volume for any driver,
2. Fb, the frequency at which the reflex port exhibits the maximum output — also known as the box resonance.
3. F3, the −3dB point (half output) of the bass response.

Consultation of Thiele's alignment charts indicates that a cabinet volume of 42 litres with a cabinet resonance of 48 Hz and a 3 dB of 43 Hz will be ideal.

These computations, now used worldwide, are the result of some pioneering research by two Australian engineers, Neville Thiele and Richard Small (now of KEF fame). Begun in the early 1970's by Thiele and later expanded by Small, their work provides designers with a solid grounding on which to model low frequency speaker behaviour.



The D25 tweeter frequency response and input impedance curves.



Frequency response and input impedance of the P21 woofer.

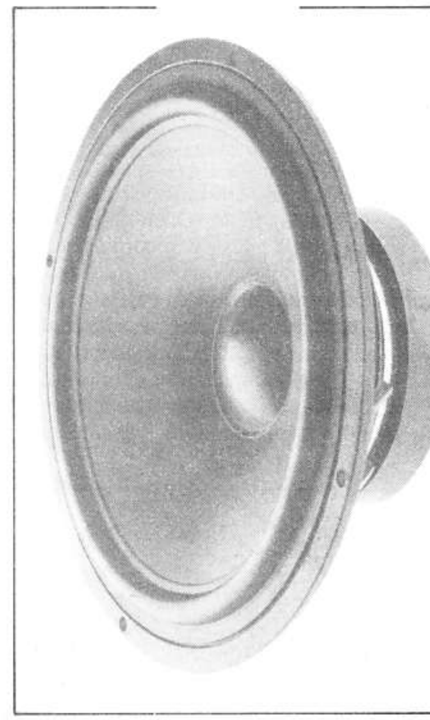
Midrange Musings

Numbers aside — the bass driver in a two way system such as this is required to reproduce the lion's share of the midrange, and its behaviour in this region will more than anything else determine the overall sonic result. The P-21 features a non-resonant alloy frame, rubber roll outer suspension for good linearity at high excursions, and a thick, rigid polypropylene cone. The latter is particularly important for midrange resolution since the driver must work in a frequency region where the mass of the cone prevents it from moving in unision with the voice coil, causing "wave modes", which are then propagated through the cone and (hopefully!) terminated by the edge suspension. Both the type of material and its constructional consistency determine how well this task is accomplished. It is now widely accepted that thick polypropylene is a very suitable material for the purpose.

As the frequency increases, the cone becomes acoustically heavier resulting in a substantial reduction of output. Looking at the frequency response graph of the P-21, there's nothing much useful above the 3 kHz mark, and at this region it is necessary to consider a treble driver capable of covering the remainder of the frequency spectrum.

Tweeter Thoughts

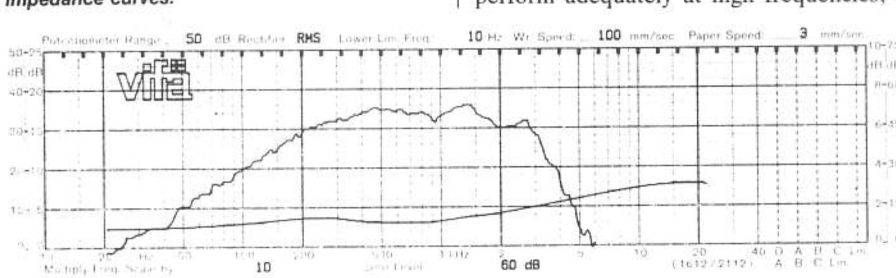
Frequency range is largely a function of size. Just as a 210 mm bass driver cannot perform adequately at high frequencies, a



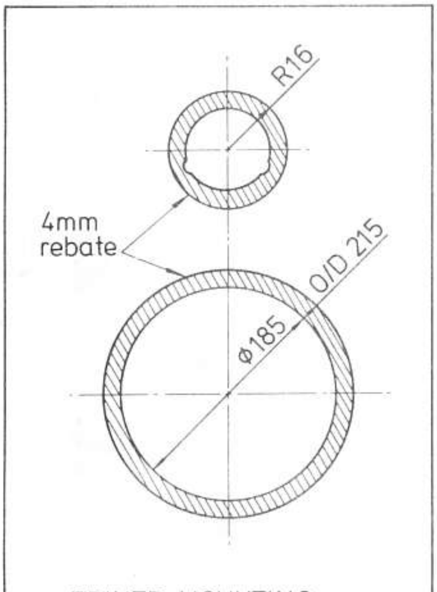
small tweeter has neither the radiating area nor the excursion capabilities to reproduce low frequency energy. Certain compromises must be made to satisfy, partly, conflicting requirements. Choosing a tweeter which will both operate at the 2 to 3 kHz region and extend out to the upper limits of the frequency range while still maintaining good power handling can be problematic, but fortunately Vifa make such a device. The D-25-TG tweeter uses a roll suspension 25 mm polyamide dome, ferrofluid voice coil cooling and an oversized magnetic structure. The low mass diaphragm gives excellent high frequency extension, the ferrofluid filled voice coil gap ensures high power handling and the "suspension" is sufficiently compliant to ensure reliable operation at the 2 to 3 kHz region. In short it is an ideal choice.

Dividing Decisions

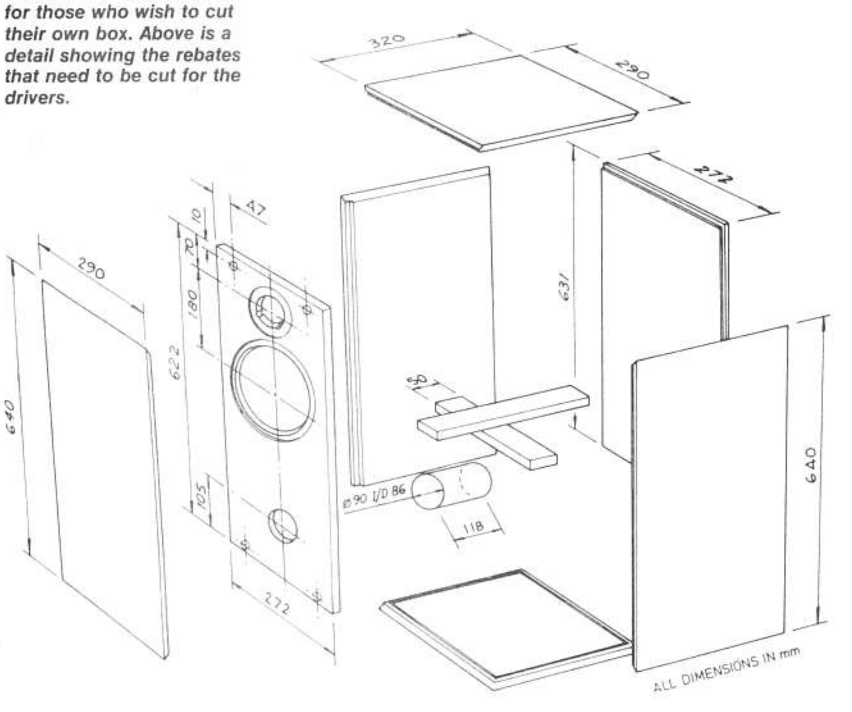
With drivers decided, we were now ready to tackle the crossover — usually the most interesting and always the most complex aspect of speaker design. It is good to start analysing the individual response of



each driver (Figure 1 and 2). A study of the on axis response of the P-21 shows reasonably flat response out to 2 kHz. Shortly after this a fairly fast roll off takes place. This natural roll off, coupled with a first order electrical filter (more on this later) will provide adequate signal attenuation. The treble circuit, however requires a different approach. Just as the woofer has a natural resonant frequency (discussed earlier) so too does the tweeter — but here it becomes important since operating the unit at or near this frequency will cause exessive excursion, resulting in definite distortion and possible damage. Since the resonance of the D-25 lies at 1.5 kHz and the crossover frequency at 2.5 kHz, it is essential that the voltage applied to the tweeter at its resonance be substantially lower than that in its operating band. This necessitates a second order slope of 12 dB per octave. C2 and C3 in parallel form the first reactive component (6.9 μF), with the .28 mH coil, L2, the second. With each of these components contributing 6 dB of attenuation, the result is a rapid electrical roll off ensuring reliable operation. Obviously this is not a problem for the woofer, but a symmetrical crossover function is desirable to achieve a balanced response, and as mentioned earlier, the natural roll off coupled with the previously mentioned first order electrical filter provides us with an overall acoustical attenuation of around 18 dB per octave. The resulting phase error at the crossover frequency is now less than 60 degrees, explaining the in phase connection of the drivers, which, when considered purely in electrical terms would not be logical, since a phase error of 135 degrees (the cumulative result of three reactive components each contributing 45 degrees) would necessitate reversing the tweeter's connections to negate a large acoustical cancellation at the crossover point. The remaining components in the trebel section, R2 and R3 constitute an attenuation pad, reducing the treble level about 2 dB, providing an overall flat response. C1 and R1, connected in series across the woofer, serve as an equalizer, exhibiting an impedance decrease with rising frequency. This compensates for the rise in impedance of the woofer caused by the large self inductance of the four layer voice coil and oversized magnet/pole piece assembly.



DRIVER MOUNTING



Dimensions of the cabinet for those who wish to cut their own box. Above is a detail showing the rebates that need to be cut for the drivers.

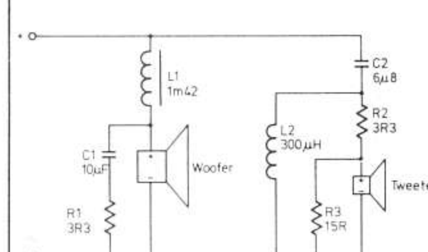
Performance

The subjective performance of the SA-100 is extremely good. The bass seems to extend even deeper than the computer alignment (Figure 3) predicts. This is not surprising, given that the proximity of the floor and walls provide a sizeable increase in output in the low registers. Midrange and treble performance, too is excellent, the speaker having the ability to place various voice and instruments in the sonic soundstage with depth and precision. The speakers are also capable of high sound pressure levels — enough at least to satisfy most afficionados of loud rock music!

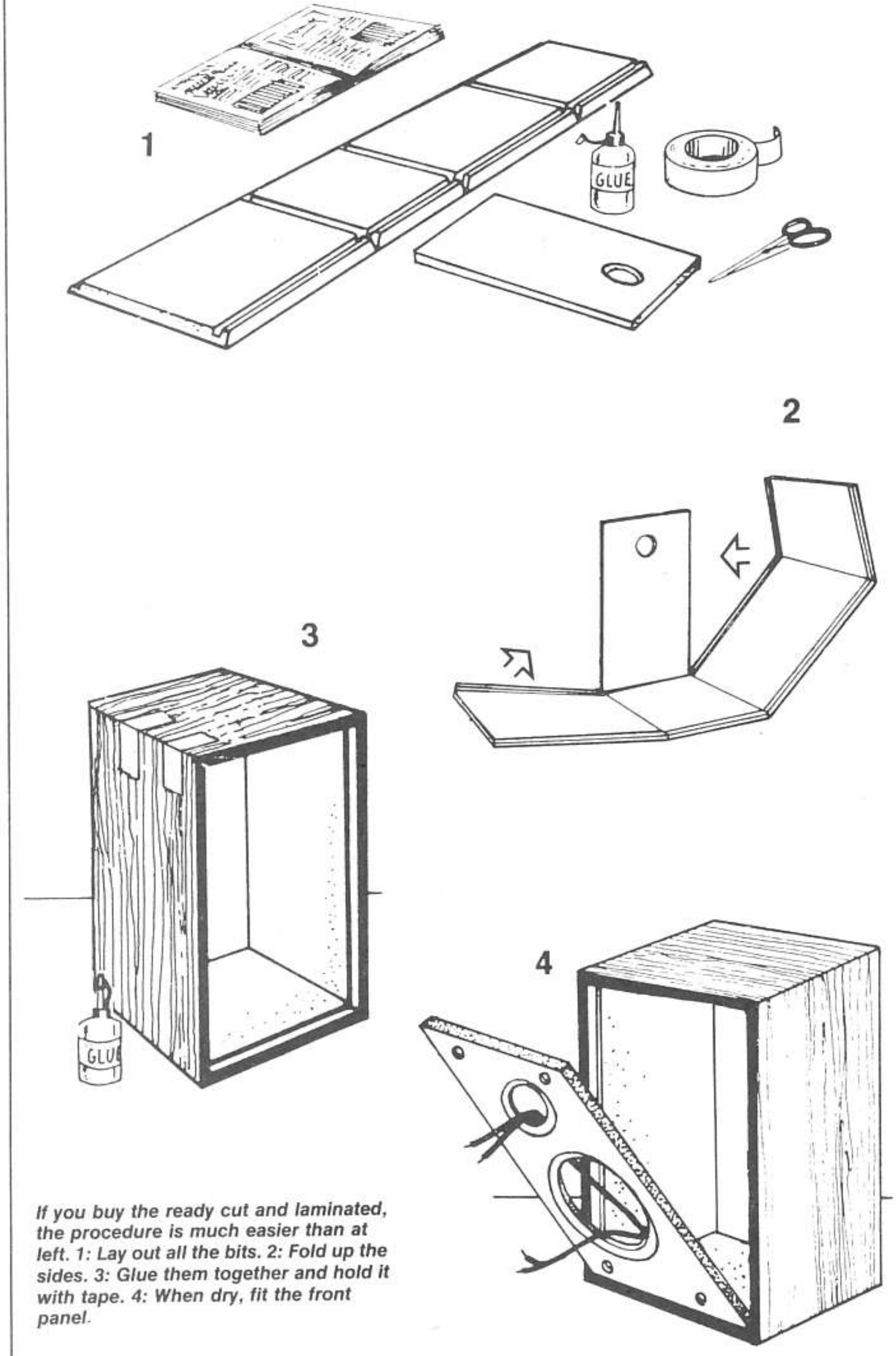
Construction

Scan Audio will be supplying complete kits for the SA-100, however you could save substantially by buying just the drivers and constructing the cabinet yourself. Detailed dimensions are given in the diagram.

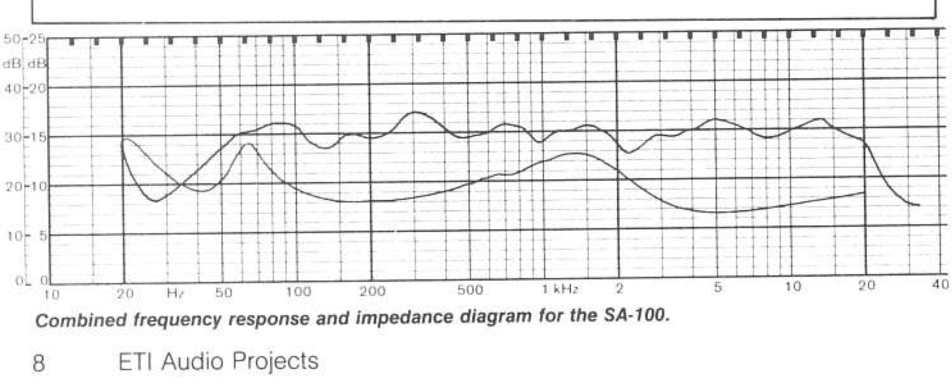
Vifa SA-100



2 Cabinets — see text.
2 Vifa P-21-WO-12 woofers.
2 Vifa D-25-TG tweeters.
2 90mm x 118mm PVC bass reflex ports.
2 Connecting terminals — see photograph.
8 Male, 8 female grill clips.
2 Pieces of grill cloth 75cm x 35cm.
Quantity of self-tapping Philips head screws.
Medium density polyurethane foam, 890 x 270 x 25mm.
Additionally, you will require some simple tools, a can of adhesive spray glue, a roll of 50mm wide P.V.C. tape, some P.V.A. wood glue and a can of matt black spray paint.



If you buy the ready cut and laminated, the procedure is much easier than at left. 1: Lay out all the bits. 2: Fold up the sides. 3: Glue them together and hold it with tape. 4: When dry, fit the front panel.



Combined frequency response and impedance diagram for the SA-100.

Construction

If you're assembling the speakers from a supplied kit, begin by unpacking the contents of the cabinet pack. This must be done carefully because the top, sides and bottom of each enclosure are in a wrap-around piece and held together by the decorative veneer "hinges" which allow them to be folded. If you are not careful in handling the enclosure in this form you could tear the veneer and spoil the finished result. You should also remove the contents of the box containing the individual drivers, the crossover networks and the other components, and check them against the parts list.

Before starting assembly, it is wise to drill pilot holes for the self-tapping screws, ensuring that they penetrate in a straight line, a 2 mm drill gives a suitable pilot hole. Use the drivers as a template when marking the holes, but take care to avoid damaging them.

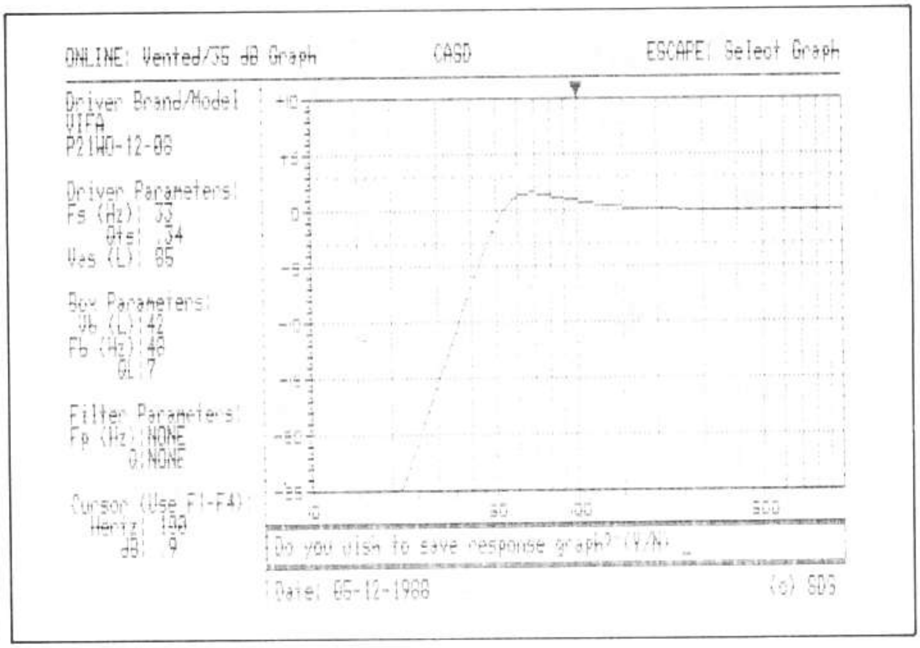
The nickel plated terminal blocks are mounted on the back panel. Each requires four mounting screws for which the pilot holes should also now be drilled.

With all the drilling completed, you can proceed with glueing the enclosures. The wraparound member actually folds around the back panel and has a machined rebate to hold it in place. This gives a rigid structure, even before the glue sets.

The procedure is quite simple. Lay out the continuous side piece on a flat surface such as a floor or large table. The three fold joints should be flexed as little as possible, as noted above. Then run a line of PVA glue into each of the V-cuts for the three fold joints and into the rebate channel. The back panel can now be fitted into the channel of what will become the base panel. Make sure that the terminal block hole is at the bottom; i.e: it corresponds to the join in the veneer which should also be at the bottom. Then it is a matter of carefully wrapping the sides around the back panel, making sure that no stress is placed on any of the three corner joints.

That done, the final corner is held together with strips of packaging tape. Don't worry if a little glue oozes out onto the veneer, it peels away from the plastic quite easily once it is dry. The bass reflex ports should be fitted now. Once they are painted and dry they can be glued from behind with hot melt glue. Leave the assembly for at least 2 hours to allow the glue to set.

Each crossover is preassembled on a piece of chipboard, so that it can be screwed in position on the back of the enclosures. Make a note of the connections for the respective terminals on the crossover; ie, input, bass and treble. Once this is done, you can fit the damping material to the back and bottom of the enclosure, stapling or glueing it in place. Do not attempt to line the sides or top of the cabinet. Meanwhile, the grills can be prepared. 'A framework is supplied over which the screen cloth must be stretched. As each side of the cloth is stretched and folded into position, it can be retained with spray glue and staples.



A computer generated prediction of the performance of the ETI 1417.

When the grill cloths are fixed into place, you should trim off the excess. Be sure to uncover the grill mounting holes. There is one in each corner — they are 12 mm in diameter. A special plastic clip is inserted into each, the mating half of each clip being mounted in the front panel of the enclosure. They can be inserted now with a gentle tap from a hammer.

Now that the glue is completely dry, fit the terminal blocks and solder them to the crossovers. This done, you can complete the remainder of the cabinet assembly. The front panel can now be fitted, rather more simply than the back panel — It just slides into the rebated front of the box. Run a bead of glue around the perimeter of the box first, using a generous squirt of glue, enough to give the front panel an airtight seal. Leave the whole enclosure for another hour or so, to let all the glue set.

After the requisite drying time has passed, the drivers can be mounted. Solder their terminals first, paying particular attention to polarity and making sure that you do not transpose the woofer and tweeter connections — if you make a mistake here you will ruin the tweeter.

Then it is a matter of fitting the grills onto the enclosures — just push them on — and you are finished. Connect them up to your amplifier, select some music and settle back to enjoy the sound. ●

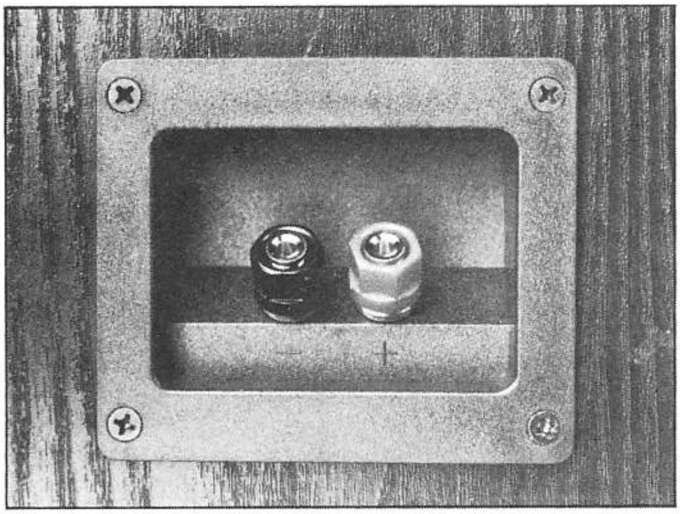
Tom Manning is with Scan Audio in Melbourne.

Price List

Components for this project are available from Scan Audio

D25 tweeter	**$74**
P21 woofer	**$188**
SA100 kit (no cabinet)	**$629**
SA100 kit complete	**$799**

MODEL:	VIFA SA-100
SYSTEM:	2-way bassreflex
WOOFER:	197 mm (8") VIFA P21W0-12-08
TWEETER:	25 mm dome f.fl. VIFA D25TG-55-06
RATED POWER:	100 Watt peak
IEC POWER:	70 Watt RMS
SENSITIVITY (1W/1m):	91 dB
FREQUENCY RESPONSE:	35-20,000 Hz
CROSS-OVER FREQ'CY:	2200 Hz
TUNING FREQUENCY:	48 Hz
IMPEDANCE:	8 ohms
INTERNAL VOLUME:	42 litres
DIMENSIONS (H x W x D):	64 x 29 x 32 cm
WEIGHT:	15 kgs (approx)
SUITING SPKR. STAND:	VIFA SA-100
PRICE EXCL. CABINETS	$629.00 pair
PRICE INCL. CABINETS	$799.00 pair



The terminal block fits on the bottom of the back panel.

WHERE TO GET THE VIFA SA-100 KIT

ACT

Brashs
168 Melrose Drive
Phillip 2606
(062) 81 5255

NSW

Jaycar Electronics
Cnr Carlingford & Pennant Hills Rds
Carlingford 2118
(02) 872 4444
115-117 Parramatta Rd
Concord 2137
(02) 747 2022
188 Pacific Hwy
Gore Hill 2065
(02) 439 4799
121 Forest Rd
Hurstville 2220
(02) 570 7000
117 York St
Sydney 2000
(02) 267 1614

NT

Sound Spectrum
51 Stuart Hwy
Darwin 5790
(089) 1 5060

QLD

Queensland Stereovisual
Shop 4, 50 Lutwyche Rd
Lutwyche 4030
(07) 357 7433

Jaycar Electronics
144 Logan Rd
Buranda 4102
(07) 393 0777

SA

International Sound
11 Carrington St
Adelaide 5000
(08) 212 5006
Miltronix Hi-Fi
125 Payneham Rd
St Peters 5069
(08) 42 3781

Eagle Electronics
54 Unley Rd
Unley 5061
(08) 271 2885

Aslin Hi-Fi
41 Commercial Street West
Mount Gambier 5290
(087) 25 6000

VIC

Radio Parts
1097 Dandenong Rd
East Malvern 3145
(03) 211 8122

Radio Parts
562 Spencer St
West Melbourne 3003
(03) 32 9788

Kitronics
4 A'Beckett St
Melbourne 3000
(03) 663 6151

Ritronics
425 High St
Northcote 3070
(03) 489 8866

The Contemporary Sound Centre,
87-89 Riversdale Road
Hawthorn 3122
(03) 818 5585

Jaycar Electronics
45 A'Beckett St
Melbourne 3000
(03) 663 2030

Jaycar Electronics
887 Springvale Rd
Mulgrave 3170
(03) 547 1022

WA

Altronics
174 Roe St
Perth 6000
(09) 328 2199

Alberts Hi-Fi
396 Murray St
Perth 6000
(09) 322 4409

Alberts Hi-Fi
642 Albany Highway
Victoria Park 6100
(09) 361 5422

Alberts Hi-Fi
177 High Street
Fremantle, 6160
(09) 335 7644

ETI-1415 The Vifa SA-70

In September 1986 the Melbourne Hi-Fi distributors, Scan Audio, released the SA-60 speaker kit based on drive units from the Danish manufacturer, Vifa. Since most things good can usually be improved, the availability of some interesting new drivers and some recent research into crossover behaviour prompted us to design a second generation speaker. This article deals with the design and construction of such, the SA-70.

Tom Manning

TO MAKE SUCH a change truly worthwhile, we sought to make several significant improvements whilst keeping the modifications retrofittable so owners of old SA-60s could upgrade at minimal cost.

A word or two on kit speakers — imported systems attract several unfortunate cost hiking factors such as freight, distribution, duty and of course labour charges. By using imported drivers and locally sourced cabinetry and crossovers, substantial savings are available compared with imported speakers of equal quality, many of which employ identical or similar drive units.

Driver Considerations

A new woofer from Vifa sparked the birth of the SA-70 — it's the C20-WG-09 8" driver with some interesting constructional features. A bass-midrange unit must be capable of performing in the dome excursion at the tweeter's resonant frequency. This allows some flexibility with both crossover slope and frequency which would otherwise need to be higher than that which we have chosen.

Crossover Calculations

The wide bandwidth of both drivers allows some manouverability with crossover considerations, and after carefully measuring the impedance and frequency characteristics of both drivers, a second order function at 29 MHz was seen to be ideal.

This is low enough to avoid the erratic response of the woofer at frequencies above this, yet high enough to eliminate potential problems with operating the tweeter close to its natural resonance (17 MHz). In order to ensure that the voltage of both drivers would be quite a few dB down in these troublesome regions, we have used a second order (12 dB/octave) network for both the high and low pass sections. Note the out of phase connections on the drivers — this is done because each reactive component contributes 45 degrees of phase error, therefore the treble leads, and the bass lags the input signal by 90 degrees. The cumulative effect is a 180 degree phase cancellation at the crossover point — reversing the connections on one driver is the cure and theoretically a perfect response should result.

An examination of the data for any modern woofer will show a set of numbers which critically determine the cabinet volume and tuning for which it is most suited. These are:

1. *The free air resonance* (Fo), which is the frequency at which the moving system of the driver exhibits the maximum output for the minimum electrical signal.
2. *The total Q factor of the driver* (Qt), which is the quality factor at resonance and which indicates the combined effects of the driver's electrical and mechanical damping of the moving system at its free air resonance.
3. The VAS, which is the volume of air in litres needed to provide the same restoring force to the cone as does the suspension. Expressed more generally, an indication of the "springiness" of the suspension system.

In the case of the C20-WG-09, the high Q factor of .78 dictates that a sealed box must be used. Here, the enclosed air is acting as a "spring" on the cone, providing a high degree of mechanical damping, part of which would otherwise be provided electrically by a low Q driver more suited to a vented box. To reduce the total system Q (damping effect) to the ideal value of .8, a small amount of Dacron damping material is stuffed lightly inside the cabinet.

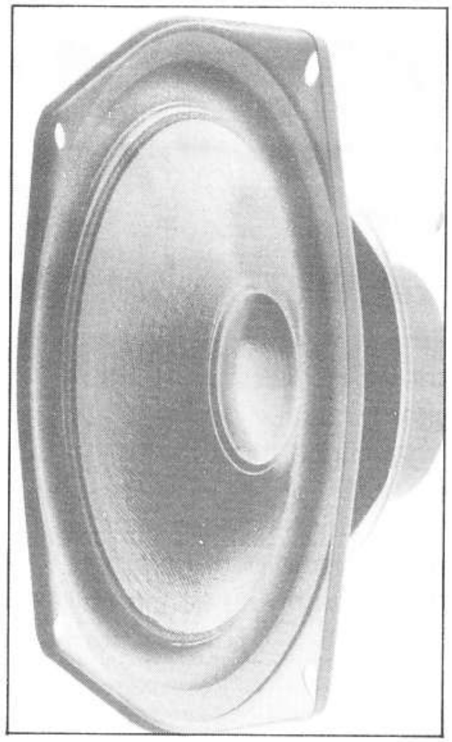
Construction

Only the most basic of tools is required to assemble these loudspeakers. You don't need special clamps or jigs and all timberwork has been precisely machined. You will need to use a soldering iron though, to connect the drivers to the crossover network.

On the other hand, if you are experienced in carpentry, you could make your own enclosures and purchase just the drivers and crossover networks; doing it this way you stand to save quite a bit of money. However this should be balanced against the high quality finish that these pre-cut enclosures will give. They are finished in a very good-looking synthetic black veneer which is a good match for most modern decors.

The baffleboard (on which the speakers are mounted) is also finished in a subdued grey vinyl.

Not supplied in the kit, but nevertheless essential to construction are: (1) a tube of PVA woodworking glue; and (2) a roll of



adhesive foam tape (eg, Engels No. 5 draught exclusion tape). The tape is needed to make airtight gaskets for mounting the drivers on the front baffle.

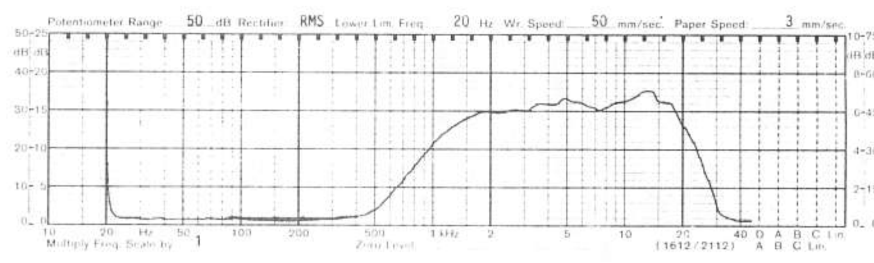
Begin by emptying the contents of the long flat box. This must be done very carefully because the top, sides and bottom of each enclosure are in a wrap-around piece and held together only by the decorative veneer "hinges" which allow them to be folded. If you are not careful in handling the enclosure in this form you could tear the veneer and spoil the finished result.

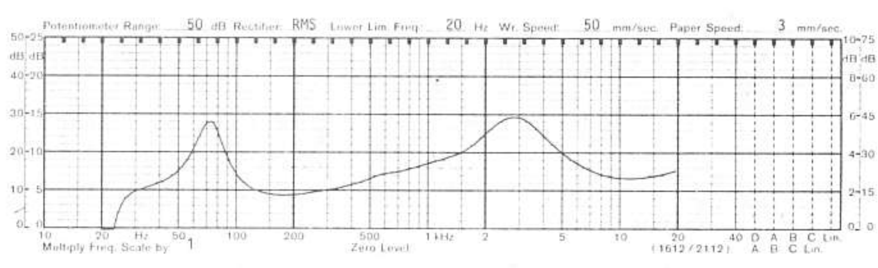
You should also empty the box containing the individual drivers, the crossover networks and the other components, to check that all have been supplied and are in good condition.

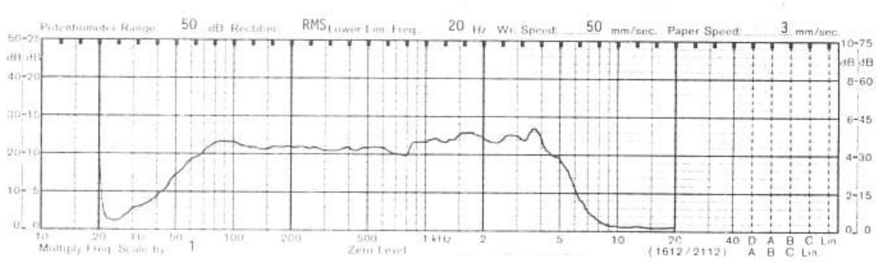
Before starting assembly, it is wise to drill pilot holes for the self-tapping screws. This ensures that they penetrate in a straight line and also obviates the possibility of splitting the timber. The screws are supplied and all are the same size; a 2mm drill gives a suitable pilot hole. Use the drivers as a template when marking the holes, but take care to avoid any damage.

Recessed terminal blocks are mounted on the back panel. Each requires four mounting screws for which the pilot holes should also now be drilled.

With all the drilling completed, you can proceed with glueing the enclosures. The wraparound member actually folds around the back panel and has a machined rebate







to hold it in place. This gives a rigid structure, even before the glue sets.

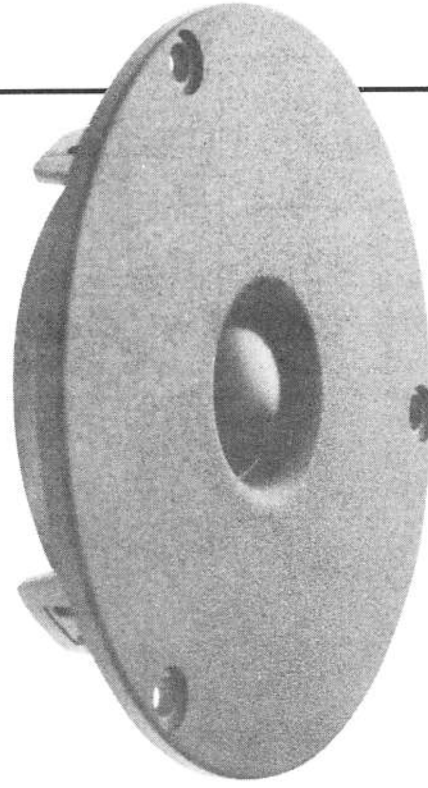
The procedure is quite simple. Lay out the continuous side piece on a flat surface such as the floor or a large table. The three fold joints should be flexed as little as possible, as noted above. Then run a line of PVA glue into each of the V-cuts for the three fold joints and into the rebate channel.

The backpanel can now be fitted in to the channel of what will become the base panel. Make sure that the terminal block hole is at the bottom; ie, it corresponds to the join in the veneer which should also be at the bottom. Then it's a matter of carefully wrapping the sides around the back panel, making sure that no stress is placed on any of the three corner joints.

That done, the final corner is held together with strips of packaging tape. Don't worry if a little glue oozes out onto the veneer, it peels away from the plastic quite easily once it's dry.

Leave the assembly for at least 30 minutes to allow the glue to set and cure.

Each crossover is preassembled on a piece of chipboard, so that they can be glued in position on the bottom of the enclosures. This may as well be done now so that the glue can be drying at the same time as the back panel. Take a note of the connections for the respective terminals on the crossover: ie, input, bass and treble.



Meanwhile, the grills can be prepared. A framework is supplied over which the screen cloth must be stretched. As each side of the cloth is stretched and folded into position, it can be retained with staples.

When the grill cloths are fixed in place, you should trim off the excess. Be sure to uncover the grill mounting holes. There is one in each corner — they are 12mm in diameter. A special plastic clip is inserted into each, the mating half of each clip being mounted in the front panel of the enclosure. They can be inserted now with a gentle tap from a hammer.

Now that the cabinet has been sitting for half an hour or so, fit the terminal blocks and solder them to the crossovers. Dacron filling material has been supplied, enough to half-fill each cabinet. This can now be installed.

The front panel can now be fitted, rather more simply than the back panel. — it just slides into the rebated front of the box. Run a bead of glue around the perimeter of the box first, using a generous squirt of glue, because it has to give the front panel an airtight seal. Leave the whole enclosure for another half hour or so, to let all the glue set.

After the requisite drying time has passed, the drives can be mounted. Solder their terminals first, paying particular attention to polarity and making sure that you don't transpose the woofer and tweeter connections — if you make a mistake here you will ruin the tweeter.

Then it is a matter of fitting the grills onto the enclosures — just push them on — and you are finished. Connect them up to your amplifier, select your program and settle back to enjoy the sound. ●

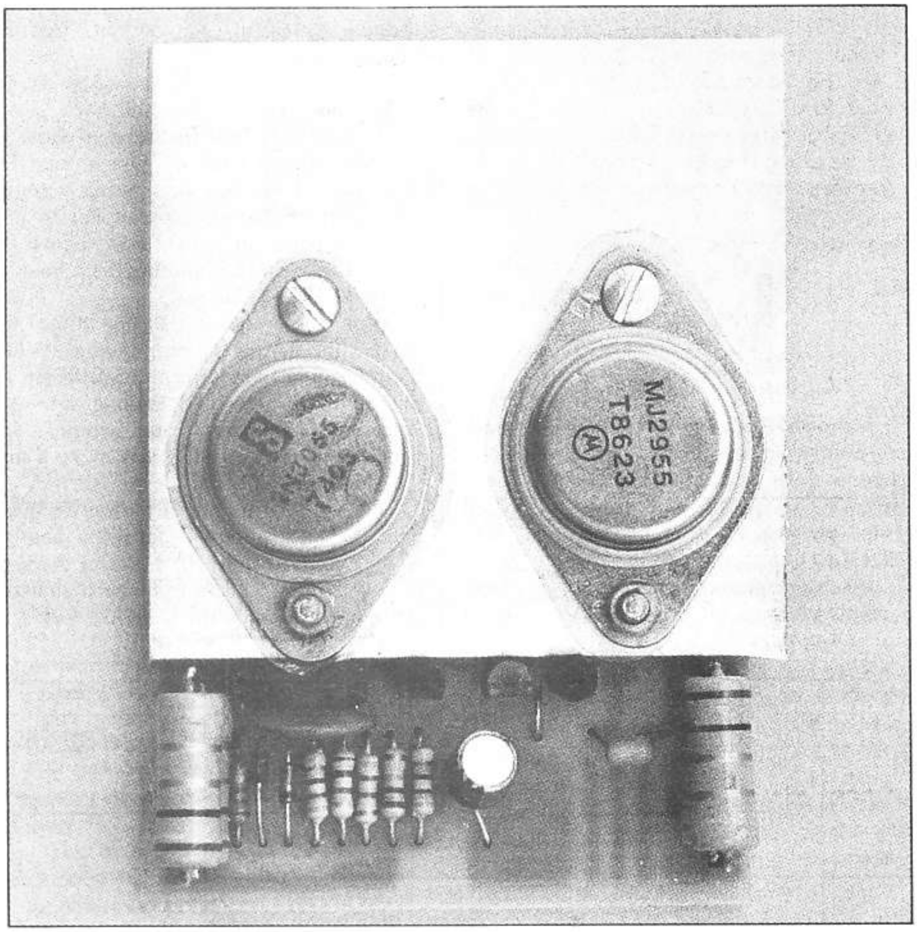
MODEL:	VIFA SA-50	VIFA SA-70	VIFA SA-100	VIFA SA-130	VIFA SW-1
SYSTEM:	2-way bassreflex	2-way sealed	2-way bassreflex	3-way bassreflex	Stereo Subwoofer
WOOFER:	125 mm (5") VIFA C13WG-08-08	194 mm (8") VIFA C20WG-09-08	197 mm (8") VIFA P21WO-12-08	245 mm (10") VIFA P25WO-00-08	2 x 245 mm (10") VIFA P25WO-00-08
MIDRANGE:				75 mm textile dome VIFA D75MX-30-08	
TWEETER:	19 mm dome f.fl. VIFA D19TD-05-08	19 mm dome f.fl. VIFA D19TD-05-08	25 mm dome f.fl. VIFA D25TG-55-06	19 mm dome f.fl. VIFA D19TD-05-08	
RATED POWER:	50 watt Peak	70 watt Peak	100 watt Peak	130 watt Peak	130 watt Peak
IEC POWER:	30 watt RMS	50 watt RMS	70 watt RMS	90 watt RMS	80 watt RMS
SENSITIVITY (1W/1m):	87 dB	89 dB	91 dB	90 dB	85-88 dB*
FREQUENCY RESPONSE:	60-20,000 Hz	50-20,000 Hz	35-20,000 Hz	28-20,000 Hz	25-88 Hz
CROSS-OVER FREQ'CY:	3500 Hz	2800 Hz	2200 Hz	800 / 3000 Hz	100-150 Hz
SYSTEM RESONANCE:		78 Hz			33 Hz
TUNING REQUENCY:	68 Hz		48 Hz	35 Hz	48 Hz
IMPEDANCE:	8 ohms	8 ohms	8 ohms	8 ohms	8 ohms
INTERNAL VOLUME:	5.3 liters	21 liters	42 liters	60 liters	62 liters
DIMENSIONS (H x W x D):	26 x 17 x 19.5 cm	49 x 26 x 23.5 cm	67 x 29 x 29.5 cm	94 x 30 x 29.5 cm	30.5 x 60 x 41 cm
WEIGHT:	4 kgs	8 kgs	15 kgs	24 kgs	20 kgs
SUITING SPKR.STAND:		SAS-1	VIFA 6102		
PRICE EXCL. CABINETS	$ 319.00 pair	$ 379.00 pair	$ 629.00 pair	$ 929.00 pair	$ 449.00
PRICE INCL. CABINETS	$ 399.00 pair	$ 499.00 pair	$ 799.00 pair	$ 1199.00 pair	$ 699.00**

*depends on positioninc
**fully built

DIGI-125 AUDIO POWER AMP MODULE

A 75/125 watt audio power amp module for around $30? Unbelievable! But here it is, by Graham Dicker.

ELECTRONICS
ETI - 1430



The ETI-1430 75/125 watt audio power amp module prototype; up to 125 watts output for $30 or less! And it fits in the palm of your hand.

With the advent of modern digital audio devices such as the compact disc, digital audio tape (DAT), pulse-code modulation (PCM) and FM stereo sound on hi-fi VCRs, the quality of the audio source of music has improved dramatically. With this improvement in signal sources came the use of vented enclosures with typical efficiencies around 0.2%. This has placed a strain on the average domestic stereo amplifier to be able to provide enough headroom with adequate sound pressure levels. There are two solutions to the problem – spend megabucks on a commercial amplifier, or build your own.

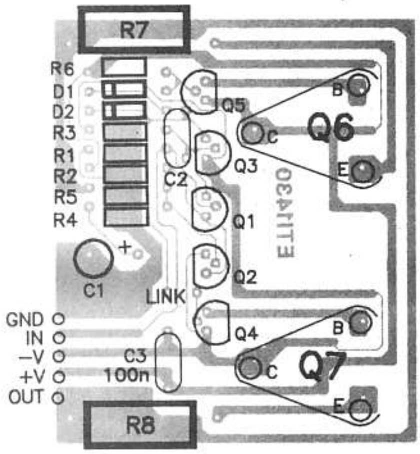
Over the years there have been many power amplifier designs published for home construction; some have been complex, others difficult to build and set up and designs using MOSFETs, against all the theory and promises, have, in general, not been as reliable as some earlier bipolar designs. Some designs have been very good but there are virtues to a bipolar design, if nothing else the ready availability and low cost of output devices.

Without doubt, the most popular power amplifier module ever described and built worldwide is the original ETI-480, using 2N3055/MJ2955 output transistors, with an estimated 100,000 being constructed in one form or another. The 480 had only a few problems for constructors, and with the new DIGI-125 power amplifier these problems have been addressed and overcome.

The new design

The new DIGI-125, so named from the common use of digital sources and its 125 watts RMS capability, uses new techniques and old, well proven technology, some a little unusual.

When Sir Clive Sinclair designed his series of amplifier modules in the 1960s, he worked on the premise that small is good and his modules were very popular the world over. This project is smaller than Sir Clive's earlier efforts as I have used CAD (computer-aided



Component overlay, showing placement of the components on the printed circuit board and the supply, input and output connections.

draughting) for the pc board design; this makes the best use of board area while allowing simple construction. The whole module uses only 22 components and, unlike any other amplifier, this one has been specially designed for the easiest home construction and setup ever. The circuit in Figure 1 shows just how simple it is.

All the resistors and capacitors are assembled in a row so that there is no need to go looking to ensure the right component is in the right hole.

The board has been designed to mount to an aluminium bracket, secured by bolts holding the output devices, the base and emitter pins of which solder directly into the board. The bracket then mounts to a suitable heatsink.

All the low power transistors are placed in a single line and the pc board has been designed so that only one link is necessary. The board measures just 55 x 60 mm and has only 49 holes. The average time taken to assemble a complete amplifier is around 10 minutes and for a production run of 40 modules completed shortly before writing this article, it took an average of 3.5 minutes per module.

The design uses no supply rail fuses, and has no presets for quiescent current as in other designs, and as such is ready to go as soon as it is assembled. The project can be assembled in two versions, one with a pair of MJ802/MJ4502 transistors which will deliver 125 watts RMS into a 4 Ohm load from a +/- 35 volt regulated supply, or using the popular 2N3055/2N2955 pair the project will deliver 50 watts RMS into 8 Ohms or 75 watts RMS into 4 Ohms from a +/- 35 volt unregulated supply. The first system can be bridged for up to 250 watts RMS output into 8 Ohms. The module should be used with a heatsink with a thermal resistance of better than 10°C/watt, if it is required to run at full power continuously.

Assembly

First, check the printed circuit board for any shorts or open tracks, using a small magnifying glass if necessary. Of necessity, this board has some fine tracks and closely-spaced pads. When assembling it, you will need to use a fine-tipped iron, preferably featuring temperature regulation of the tip. In addition, use a narrow gauge (say, 20 gauge) resin-cored solder for best results.

The bracket is prepared from a 60 mm length of 50 x 50 mm by 3 mm thick right angle aluminium extrusion. You can use the pc board as a template to mark out three of the four hole positions for each transistor, then use the insulating washer to mark the position of the remaining holes. Centre-punch or otherwise mark the hole centre before drilling. Hole sizes are determined from the insulating washer and plastic bushes supplied with the transistor insulating kits. Don't forget to mark out and drill suitable holes for attaching the bracket to your heatsink.

> *'Fewer than 25 components on a 55 x 60 mm pc board and no setting up required'*

Start assembling the board by soldering the low power resistors, the two diodes, the wire link and the three capacitors in place. Note the polarity of C1 and the two diodes. The wire link goes between the two holes adjacent to Q2 and Q4. All components must be seated right against the pc board. Next insert the plastic low power driver transistors, ensuring that the right device is in the right hole. Note that Q5 and Q4 face the opposite way to the other three transistors. WARNING: some manufacturers' devices have differing pinouts, so it's wise to check the emitter-base-collector pinout of each of these transistors before placing them in the board.

It is easiest to solder the R7 and R8 resistors in last. Unless 2 W or 5 W 0.33 Ohm resistors are used here, you should first solder one 0.68 Ohm 1 W resistor in each place, positioning it about 2 mm above the board to allow some air flow. Then place the other 0.68 Ohm resistor on top, trimming the leads to a suitable length and soldering them to the leads of the first resistors.

The last stage is to assemble the output transistors and the bracket, as shown in the assembly diagram given here. Note that Q6 and Q7 are first mounted to the bracket and secured by one bolt and nut each, then the board is placed in position and the second bolts and nuts are secured. Then solder their base and emitter pins.

Power supply

The power supply circuit shown here is a universal design to provide +/- ("split") rails. The same supply is suitable for a 100 or 50 watt version of the project. If two 100 watt power amps are required it is suggested that two transformers be used, each with separate rectifiers and filters. This will improve the crosstalk and peak music power capability of the modules.

The design uses a conventional full wave bridge with centre tapped transformer to charge suitable reservoir capacitors on

ABBREVIATED DATA SHEET
2N3055/MJ2955, MJ802/MJ4502

The output devices employed in the ETI-1430 have been chosen for their availability, low cost and ruggedness, as much as for the specifications required for the job. Complementary pairs of silicon NPN/PNP devices are selected: the 2N3055/MJ2955 pair for the lower power version; the higer-rated MJ802/MJ4502 pair for the higher power version.

The devices are all housed in the common TO3 package, the pinout for which is given here.

DEVICE TYPE

NPN : C	PNP :	Ic max.	Vceo	hFE at Ic	Pd at 25 deg.
2N3055	MJ2955	15A	60V	20-70 at 4.0 A	115 W
MJ802	MJ4502	40A	100V	25-100 at 7.5 A	200W

Notes: Ic max. is the maximum continuous collector current permissible; Vceo is the minimum breakdown voltage between collector and emitter (base open-circuit); hFE is the dc current gain at a given collector current – maximums and minimuns are given; Pd is the maximum permissible power dissipation of the device.

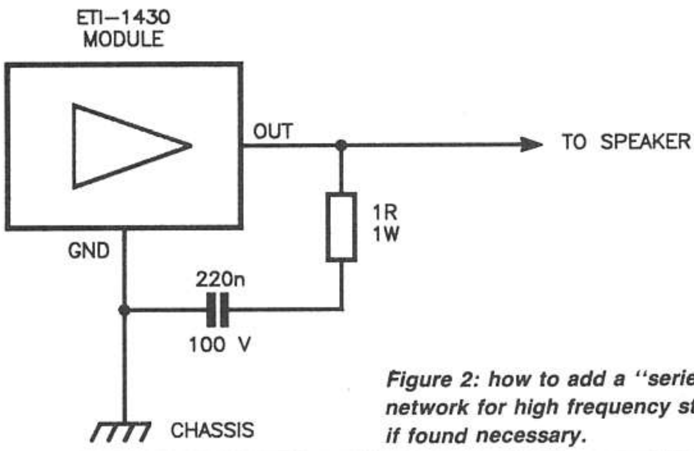


Figure 2: how to add a "series step" network for high frequency stability, if found necessary.

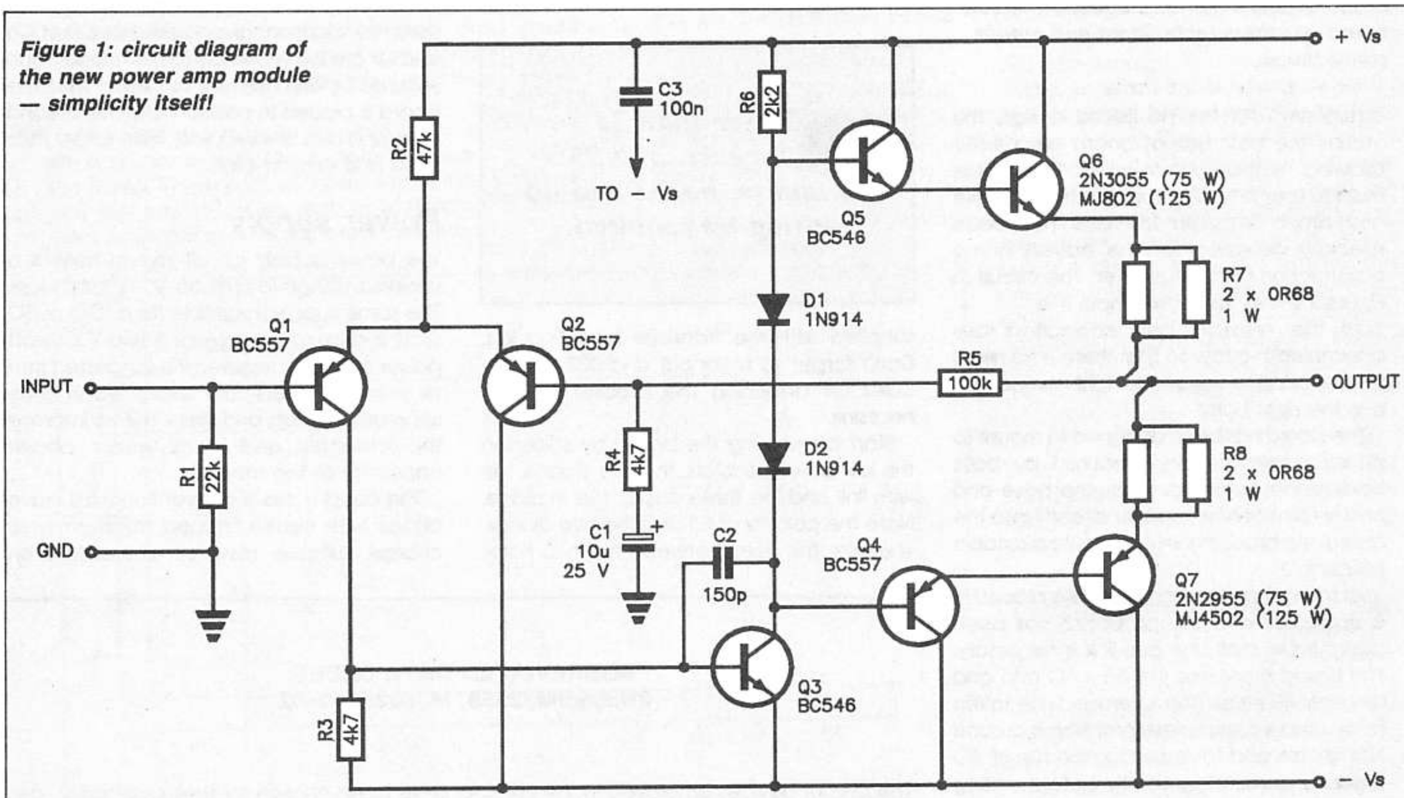
alternate cycles from each half of the transformer winding. Alternatively, two transformers may be used with their secondaries connected in series. The peak voltage appearing across the capacitors is given by:

VRMS x 1.42

which gives +/- 38 volts when using a 28-O-28 V secondary transformer, such as the DSEO144 from Dick Smith Electronics and legions of similar trannies, or +/- 32 volts from a 24-O-24 V transformer. These voltages are within the specified working voltage of the electrolytics. Under load, these voltages will reduce as a result of the dynamic regulation of the design (owing to the internal resistance of the transformer and diodes).

The wiring diagram here shows how to wire

Figure 1: circuit diagram of the new power amp module — simplicity itself!



How it works

The circuit is shown in Figure 1. The input stage consists of two BC557 PNP transistors, Q1 and Q2, connected as a differential pair with Q1's base being ground referenced. This results in a slight output offset voltage of about 300 mV, but does not adversely affect performance or reliability; it does however allow for dc coupling to the audio source. It should be noted tha many of the new CD players have frequency responses that extend down to a few Hertz (almost dc!), and it has been argued in the past about the benefits and disadvantages of a response this low. Needless to say, if it's there it can be reproduced.

The output of Q1's collector goes to the base of Q3, which is the main gain stage for the amplifier. This stage has a voltage gain of approximately 200 and unilateralisation and compensation is provided by C2. The two diodes D1 and D2 provide forward bias for the output stage to enable a quiescent current in the order of 40 mA to flow, to reduce crossover distortion. Resistor R6 provides current from the positive supply rail when Q3 is not conducting. Bootstrapping was initially tested in the prototype but the added components did not warrant the extra power available.

The output stage consists of a super-gain pair of output transistors in the familiar complementary symmetry configuration. The 50 watt version uses 2N3055/MJ2955 output devices which have a Hfe of 20 at 1.5 amps collector current, whereas the MJ802/MJ4502 pair have a Hfe of 25 at 4.5 amps collector current. The salient characteristics of the devices are detailed in the accompanying data sheets.

It is here that economy of design can be made over the original EIT-480 design. Instead of using two pairs of devices for 100 watts and increasing the drive requirements, we actually decrease the drive requirements over the 50 watt design by using devices with higher gain at the normal operating conditions. Secondly, as the output stage of up the power supply and module. While can-type (chassis-mounting) electrolytics are shown, pigtail types (either RB or axial) may be used and wired directly to the tagstrip. However, they should be mechanically secured to prevent fracture of the leads sometime in the future. The leads between the diodes and the electrolytics should be heavy duty hookup wire (at least 32 x 0.2 ? mm), and also the supply and ground leads to the module and speaker. Keep the supply and ground leads as short as possible, preferably shorter than 150 mm.

To improve the regulation, larger electrolytics and a transformer with a lower winding resistance (higher current rating) may be used. As a rule, the value of the electrolytic capacitor in such circuits is approximately 4000 ?F per amp, and here a 1.5 amp +/- 38 V supply was required, hence the 5600 ?F capacitors. With audio it is wise to go by the adage "bigger is better", so you may wish to use some of the many secondhand computer grade capacitors about.

Should you be using two separate transformers in series instead of a single centre-tapped transformer, ensure that the windings are in phase (adding).

Testing

Check with an Ohm meter that the output transistors are insulated from the heatsink. If all is well, apply the supply voltage without an input and without a load. Measure the output voltage; if it is close to zero volts, then all is well. You may now connect a load (loudspeaker or "dummy" 8 Ohm resistor load) and apply a sine wave input.

Check the output at a wide range of levels; if you can beg, borrow or steal an oscilloscope, view the output waveform. You should see a clean, well formed sine wave with no traces of high frequency oscillation at any output level. Try turning up the output until the waveform is "clipped"; the waveform should neatly square off top and bottom. If you are happy now, hook up a source of music and blow the cobwebs out of your speakers!

Load stability

The project has proved stable driving a wide variety of loads, including a 100 volt PA line output transformer. However, you may encounter some situations where the load causes high frequency instability, in which case either a HF damping network or a Zobel network (or both) connected to the module's output terminals will be called for.

Figure 2 shows how to connect a "series step" resistor-capacitor network to provide a high frequency load on the output. The resistor should be rated at 1 W and the capacitor should be a low inductance metallised polyester (MKT) or polypropylene

the new design has a voltage gain of 1 by using emitter followers (super alpha pairs) the output stage simply becomes a large current amplifier, whereas the ETI-480 design used common emitter configuration drivers and provided a voltage gain of the order of times 4. This is all very well but it results in two problems:

(1) nonlinearity in base drive impedances and currents in the pre-drivers which can result in increased distortion (try removing the ac feedback and observe the unequal voltage gains on positive and negative peaks), and (2) the driver stage is inefficient due to the load sharing and local negative feedback resistances in the circuit. This can result in a problem of obtaining sufficient drive for the output devices.

After years of experience in designing equipment for broadcast applications, one thing I have learnt is that symmetry is the best way of solving problems before they occur. If the device is full of worms to begin with, negative feedback will only hide the symptoms not cure the problem.

It is interesting to note that, in the development stages, all ac negative feedback was removed from the prototype amplifier and the distortion rose to only 0.12% and the resulting output was stable and symmetrical. In this case, the feedback applied is to stabilise the closed loop gain of the amplifier to 20 times, to compensate for parametric spreads in Q3, not to reduce distortion or other ills.

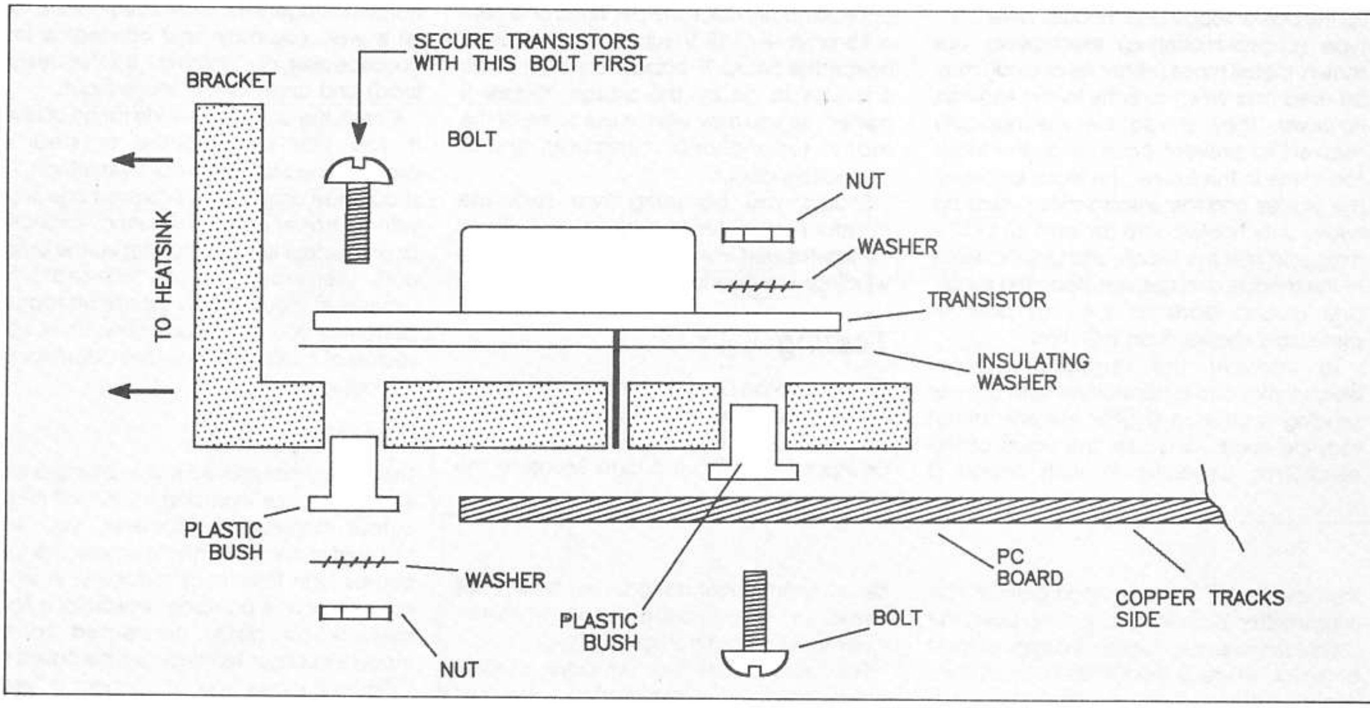
It is interesting to look at common current designs of amplifiers and note that very few designs use complete symmetry in all audio stages. Logically, if one does not use all fully symmetrical stages then local or overall negative feedback must be used within a design to overcome the problems caused by parametric spreads in components. It is my design philosophy that all stages, regardless of the signal levels involved, should be of symmetrical design. This, in turn, will assist in the reduction of transient intermodulation distortion (TIM).

For those purists that still have a valve power/preamplifier combination, you should note that most valve amplifiers were of symmetrical designs and few had local feedback around stages to compensate for spreads in components. In most cases there was only 6-12 dB of feedback used overall in power output stages, mainly to improve the distortion figures and frequency response of the output transformers. This is negligible to an average stereo amplifier of today, with 60 dB of feedback in the magnetic cartridge preamp, 30-45 dB feedback in the preamp & bass/treble circuits, 40-60 dB feedback in the power amplifiers, and if opamps are used with open loop gains of around 110 dB for each opamp, then the overall feedback from three stages is staggeringly high. On listening tests, it is interesting to compare the sound quality of the DIGI-125 and other top-of-the-line commercial amplifiers.

Because of the reasons explained above, the BC546/BC556 drivers have a greater reserve of drive current in the 100 watt version than the BD139/BD140 devices used in the ETI-480.

Note that the output stage bias diodes are not mechanically coupled to the heatsink bracket. I found this mechanically cumbersome and, in practice, unnecessary – in defiance of conventional wisdom. With dozens of these modules installed and operating in differing applications, no thermal problems have been experienced; the bias does climb with increasing temperature, but it does stabilise and in any case, more bias improves performance, albeit slightly!

No supply rail fuses were incorporated as, from my experience, the balance of expediency tips in favour of not having them. Often, I found transistors failed and protected the fuses!



Assembly diagram showing how the transistors, heatsink bracket and pc board go together.

SPECIFICATIONS

Output devices	Supply voltage	Load	Output Power
2N3055/2N2955	+/- 35 V	8 ohms	50 W RMS
2N3055/2N2955	+/- 35 V	4 ohms	78 W RMS
MJ802/MJ4502	+/- 37.5 V	8 ohms	62 W RMS
MJ802/MJ4502	+/- 37.5 V	4 ohms	112 W RMS
MJ802/MJ4502	+/-35 V Reg.	4 ohms	131 W RMS

Total harmonic distortion at full rated output	0.35%
Total harmonic distortion at 10 watts RMS	0.0015%
Signal-to-noise ratio wrt full output	-115 dBm
Power bandwidth (-3 dB)	100 kHz
Frequency response (1 watt RMS)	2 Hz-110 kHz, +/- 1 dB
	1 Hz-120 kHz, +/- dB
Damping factor (8 ohm load)	90

PARTS LIST ETI-1430

AMPLIFIER

Resistors

All 1/4 W, 5% unless noted

R1 .... 22K
R2 .... 47k
R3, R4 .... 47k
R5 .... 100k
R6 .... 2k2
R7, R8 .... 0R33, 2 W (2x 0R68 paralleled)

SEMICONDUCTORS

D1, D2 .... 1N914, 1N4148
Q1,Q2 .... BC557
Q3, Q5 .... BC546
Q4 .... BC556
Q6,Q7 .... MJ802/MJ4502 or 2N3055/2N2955

CAPACITORS

C1 .... 10 μF/25 V RB electro
C2 .... 150 pF/100 V ceramic
C3 .... 100 nF/100 V ceramic

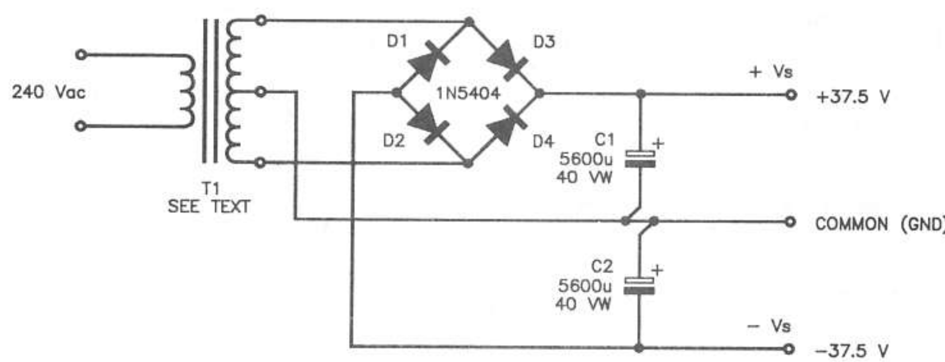
MISCELLANEOUS

ETI 1430 pc board; two TO3 transistor insulating kits; heatsink (each module) – eg. Rod Irving Electronics H10549 (100 W), H10535 (50 W), Dick Smith Electronics H-3426 (100 W).

Approx cost: $20-27

POWER SUPPLY

D1-D4 – 1N5624, 1N5404; two 5600 uF/40 VW electrolytics; T1 (mono 100 W, or stereo 50 W) – Rod Irving Electronics M21092 or Dick Smith DSE0144, two of these per channel for 100 W stereo setup.



Power supply circuit.

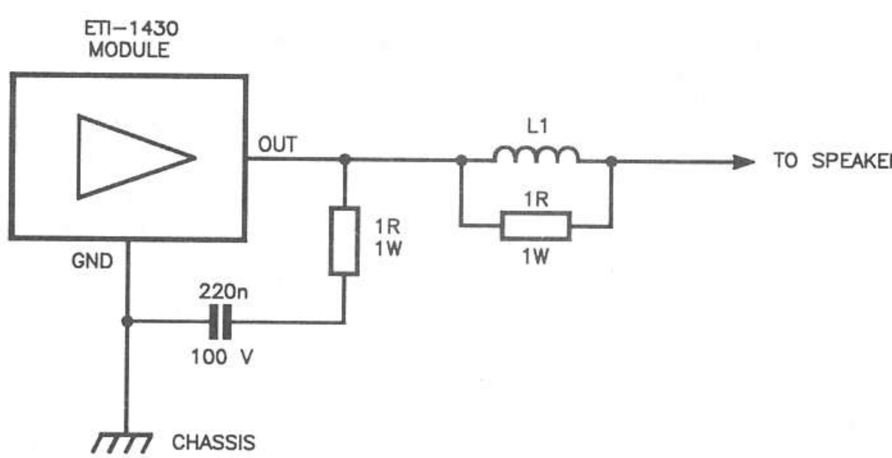
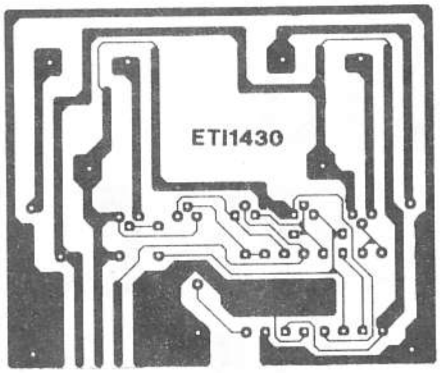


Figure 3: the addition of a Zobel (LR) network, in conjunction with a series step RC network, can solve stability problems with particularly "difficult" loudspeaker loads.

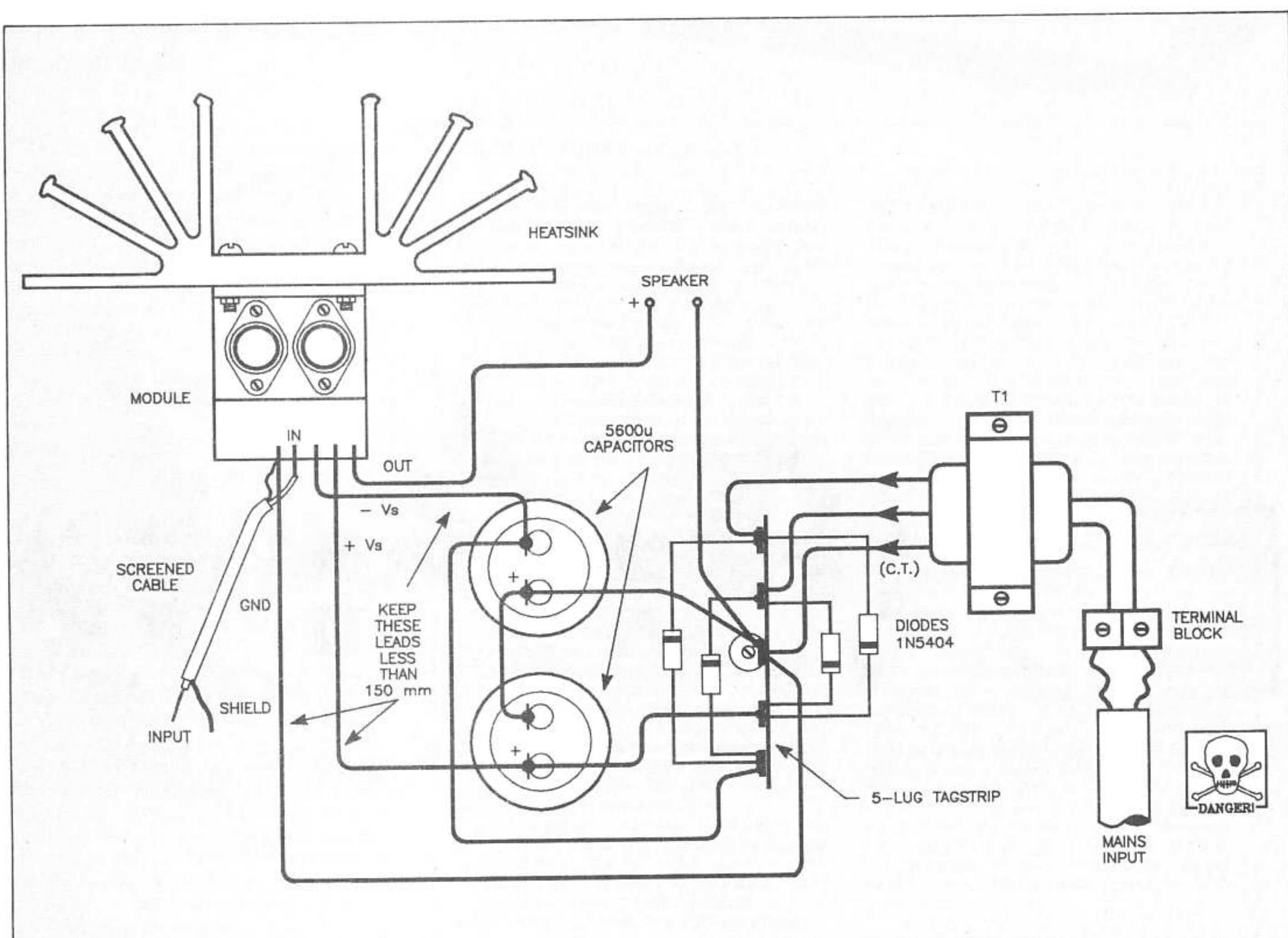


Full size artwork for the ETI-1430 pc board.

(MKP) type. Keep the leads as short as possible and solder the components directly to the tracks on the copper side of the pc board, the junction between the resistor and capacitor sitting in mid-air. The resistor may be any convenient value between 1 Ohm and 4.7 Ohms, while the capacitor may be any convenient value between 100nF (0.2 F) and 220n (0.22 F). It should be rated at 100 V minimum.

A Zobel network may also be added (useful with "difficult" speakers), comprising a coil with a parallel-connected capacitor, as shown in Figure 3. The coil can be wound with 1.0 mm diameter enamelled copper wire, using two 10-turn layers on a P24 potcore bobbin or a short length of 12 mm diameter wooden dowel. The network should be mounted close to the module to keep the lead between the module and the network short. eti

The author has retained copyright on the pc board so that constructors wishing to make boards for their own use may use this artwork for that purpose. Ready-made boards may be obtained from Graham Dicker at PC Computers, 36 Regent St, Kensington S.A. 5068, (08) 332 6513. Boards may be purchased singly or in small quantities, wholesale prices being available for larger quantities.

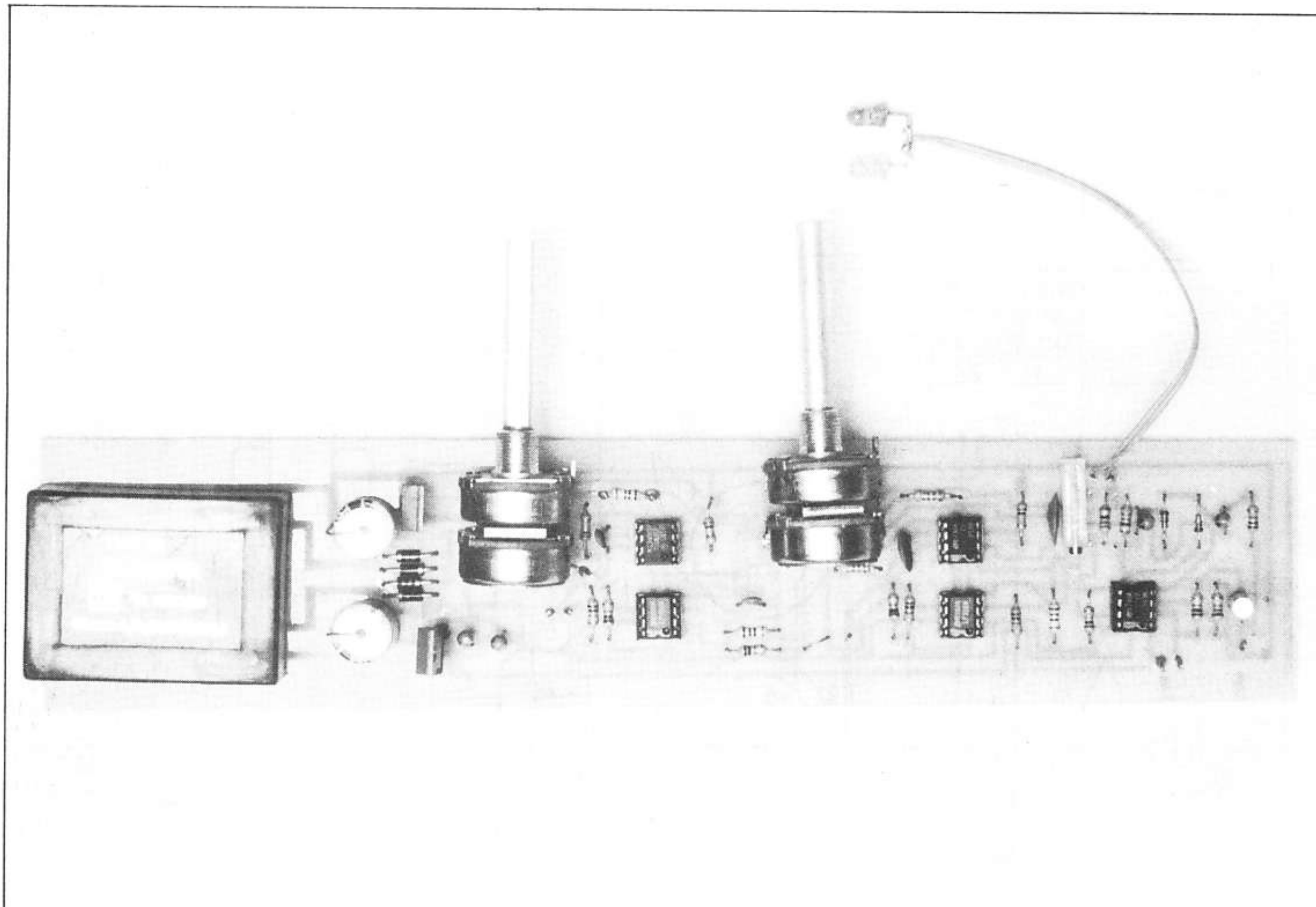


Wiring diagram for a single module and power supply.

ETI-1413 ELECTRONIC CROSSOVER

In the Past there have been many Circuits for Audio Electronic crossovers but the one presented here is based on an unusual circuit, is simple, versatile and above all easy to build and use.

Simon Leadley



ELECTRONIC CROSSOVERS are used to eliminate the need for the inefficient and heavy units that are used in conventional speaker crossovers. They are used to split the audio spectrum into 2,3 and sometimes 4 ranges, which are then fed to a separate amplifier for each speaker. The advantages lie in the fact that the drive to each of the speakers can be adjusted so that the different sensitivities of the drivers match. In an ordinary passive crossover this matching is done by attenuating the drive to the speakers with large wattage resistors which waste power. The passive crossover must be designed with the particular drivers in mind so that it will work properly. Putting a particular passive crossover in a speaker system for which it wasn't designed will result in poor sonic quality unless the drivers are very similar. With an electronic crossover, however, both the crossover frequency and the level of drive to each speaker can be varied, allowing it to compensate for a great range of driver specifications.

Construction:

The whole unit is built on a single board that can be configured 2 ways. As a mono 3-way crossover or as a stereo 2-way. Begin by deciding which version you wish to build and then select the values from the table for the crossover frequency sweep that you desire. Check the circuit board for shorts between tracks and then place all the passive components (take care to orientate the capacitors as in the diagram) Next I suggest that you use IC sockets for all the op amps, it will make testing easier and replacement in case of component failure quicker in the future. (the extra cost is well worth it!) Then place the power supply components, taking care to put the regulators in the correct spot. The power transformer also fits on the board so it may be necessary to drill out the mounting holes to suit. The pots for the frequency adjustment also fit onto the board. Solder short lengths of wire to the six terminals of the pot and then pass these through the board at the correct points and solder them to the board. The two LEDS are soldered together with the Anodes of each connected to the Cathodes of the other. Two flying leads are then taken to the board. (Note: the anode is the longer leg of the LED).

TESTING:

I found it best if the IC's wern't fitted at this point so that you can test if the power supply is functioning correctly. Connect 240v to the transformer input and then use a multimeter to see if the power rails are present at the correct pins of each of the IC sockets. If not, power down (remove the 240 V from the wall socket) and check that the diodes and the regulators are properly orientated. If all is well, power down and insert the IC's (check orientation!) Power up again and feed an audio signal into the input. Using an amplifier you should hear filtered output from each of the outputs. Turning the pots on each channel should sweep the frequency up and down. If not power down and check all connections. Because the two filter blocks are identical, at least one should operate so you can compare them.

Setting the LED threshold will depend on the usage that you have for the unit. Besides providing power on indication, the LED circuitry can be set to indicate that a predetermined input level limit has been reached. In a PA system this might be the 0dB level while in a home Hi-Fi it may indicate onset of clipping in the preamplifier. The multiturn trim pot is adjusted to give the required point.

Packaging:

If you intend using the unit with a PA system you will probably want to mount the unit into a 19" rack mounting case. The frequency adjust pot will be best mounted on the rear of the unit as they will usually be set once and forgotton, also it is best that they are mounted far from prying hands. The level controls are best mounted on the front panel so that the system can easily be tuned in different rooms. Note that the Level control pots are connected to the outputs with flying leads so that the unit has flexibility in construction.

If, on the other hand the unit is being used with a home Hi-Fi unit then the construction can be more modest. A small instrument case is ideal, since the signal levels are quite high and a fully shielded case therefore unnecessary. Mount the power switch and power cord away from input circuitry to minimise hum pickup.

Filter values

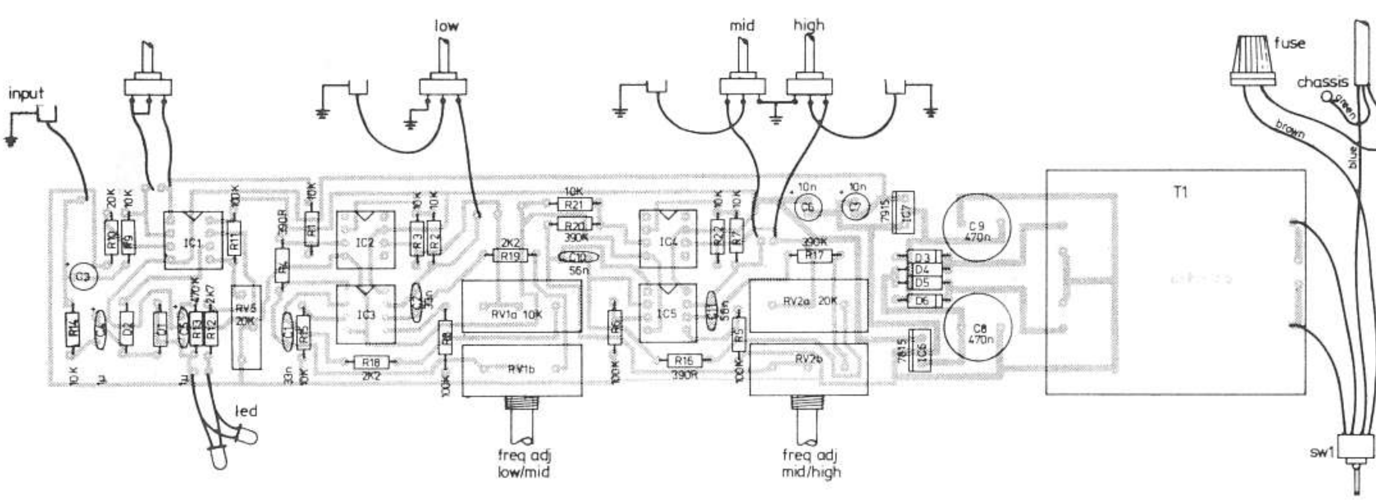
To set the cross-over frequencies, you need to select values for the resistors and capacitors that surround the filter elements. Around IC3a, for instance, we need values for RV1a, R18 and C1. To do this we apply the formula:

$$F = \frac{1}{2 \times 3.14 \times C1 \times (RV1a + R18)}$$

Assuming F for the low to Medium transition is 400 Hz, and we set C1=56n, we discover that (RV1a + R18) equals:

$$\frac{1}{2 \times 3.14 \times 56 \times 10^{-9} \times 400} = 12062\ R$$

which can satisfactorily be realised with a 10 k pot and a 2.2 k resistor in series.

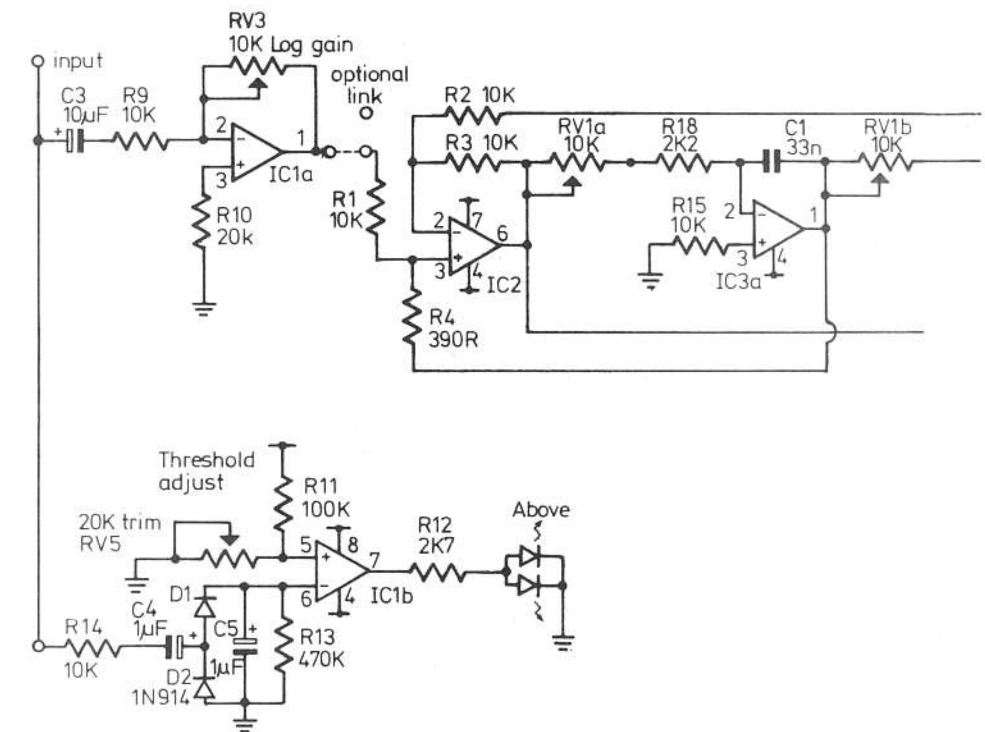


How It Works ETI-1413

The basic design is based on the State Variable Filter technique. The filter used three op-amps (IC2, IC3a and IC3b) to provide high pass and low pass outputs. Centre frequency and Q are independently variable. The circuit is often termed a *Universal Active Filter*. By varying the RC networks (and simplifying the design by allowing C1=C2, C10=C11, & Ra=Rb) then the filter becomes easily turnable.

The input buffer (IC1a) simply provides some gain to offset the loss in filters. If the gain is insufficient merely, change the value of RV3. The threshold indicator (IC1b) is simply a comparator that shows when the rectified input voltage exceeds a certain value. The 1uF cap across the diodes (C5) provides a hold function so that peaks can be detected. The trim pot allows the threshold level to be adjusted over a wide range of values.

To calculate the values for the Cross-over Freq use the above formulas or refer to the values that we have chosen.



The components around IC3b have the same values. Using the components in the circuit gives cross-overs at about 400 Hz and 7 kHz, which seems to be quite satisfactory. However, do not be afraid to select different values to suit your set up, or indeed, the type of music you like to listen too.

In the event that you decide to configure the network as two two-way crossovers, you will need to make sure that both speakers cross over at the same frequency, so all the components will have the same values. In this configuration, the input buffer and the over range LED circuit can be removed.

ETI-1413 — PARTS LIST

Resistors — all resistors ¼ watt, 1% metal film unless otherwise specified. All values in ohms.

- R1, 2, 3, 7, 14, 15, 21 .... 10k
- R4, 16, 17, 10 .... 390K
- R6, 5, 8, 11 .... 100R
- R10 .... 20K
- R12, ... 2K7820R
- R13 .... 470K
- R18 .... 2K2
- R19 .... 2K2

Semiconductors

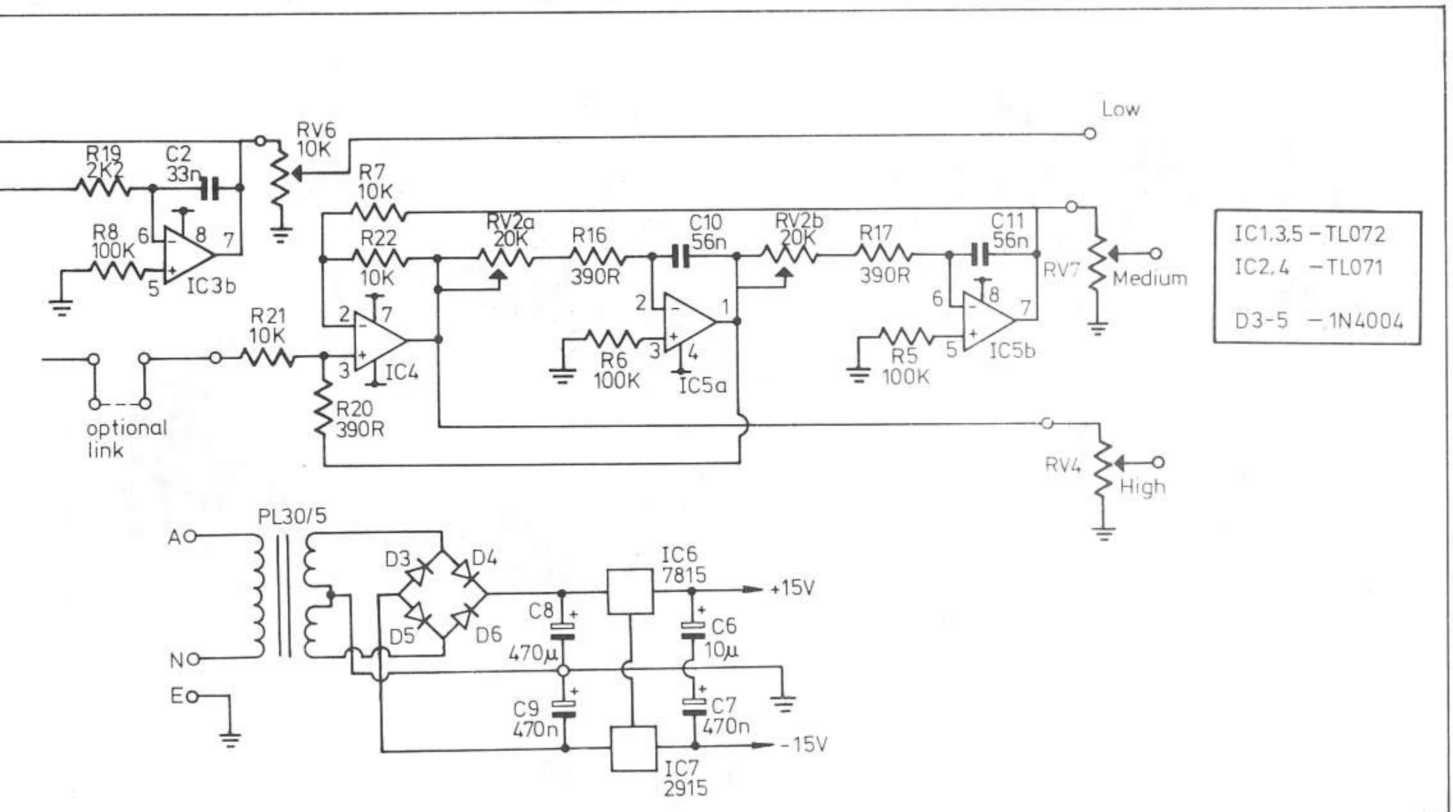
- IC2, 4 .... TL071
- IC3, 5, 1 .... TL081
- IC6 7815
- IC7 7915
- D1, 2, 3, 4, 5 .... IN4001
- D6 .... 1N4004

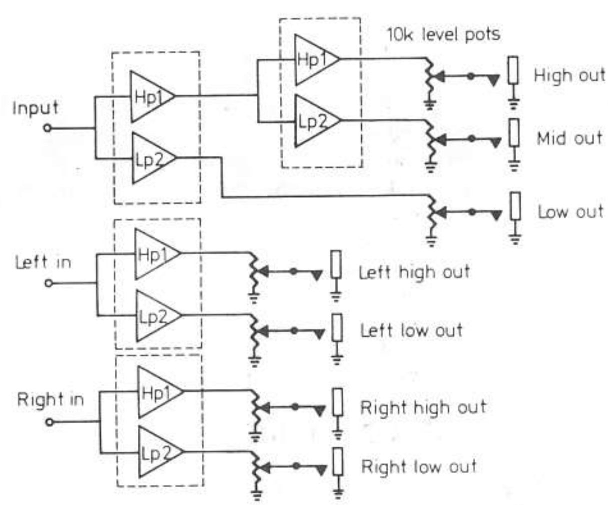
Capacitors

- C1 .... 33n
- C2 .... 33n
- C3 .... 10µ
- C4 .... 1µ
- C5 .... 1µ
- C6 .... 10µ
- C7 .... 10µ
- C8 .... 470µ
- C9 .... 470µ
- C10 .... 56n
- C11 .... 56n

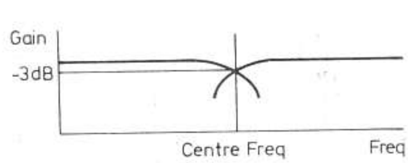
Potentiometers

- RV1 .... 10k dual
- RV2 .... 20k dual
- RV3 .... 10k log
- RV4, 6, 7 .... 10k

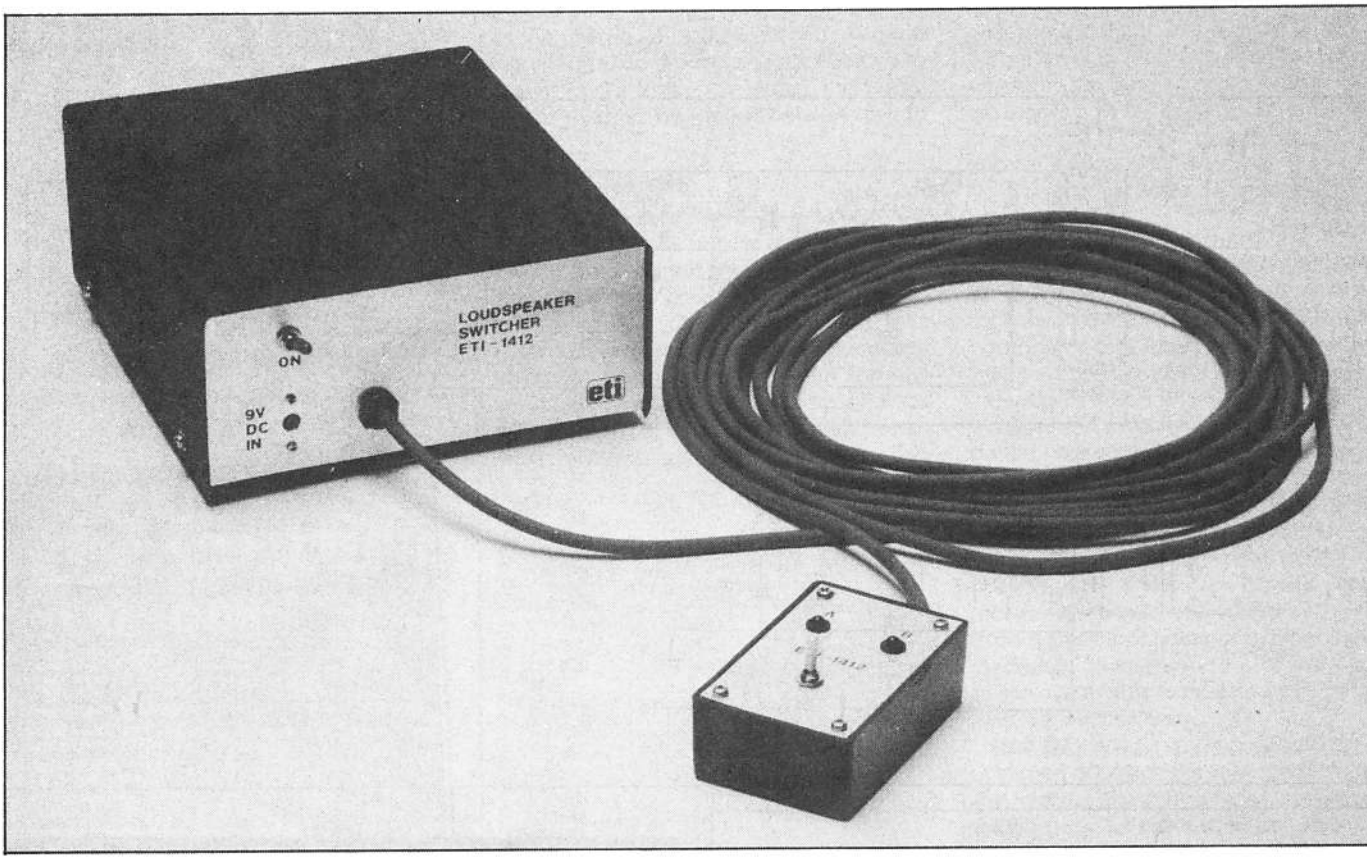




The unit can be configured in two basic ways, to give either a 3 ways or two 2-way systems.



The filters roll off about the centre frequency in the mains shown above.



STEREO LOUDSPEAKER SWITCHER

Terry Kee

Have you ever wanted to switch between two pairs of loudspeakers to confirm that your favourite pair is simply the best? If so then read on. This ultra-useful project includes an anti-thump circuit and also finds an application in the recording or home studio.

A LOUDSPEAKER SWITCHER is extremely simple in concept yet it finds a wide range of applications for the audiophile and in the recording studio. The most obvious application would be to use it as an A/B loudspeaker switcher. Subjective listening tests on the sound quality of two stereo speaker systems can be made while the switcher is used to quickly switch between the two pairs under test.

In a recording studio environment it is often required to use at least two pairs of monitor speakers whilst doing a recording or during a mixdown. A high quality pair is used to monitor the sound quality of the recording, the other pair is used to give the engineer an idea of the way it would sound in a typical hi-fi set-up in the home. The loudspeaker switcher would also be an economical approach to switching between the two pairs using a common amplifier. Do I hear whoops of delight from those who have been denied music because the amplifier in the lounge has walked into the studio in the spare room?

Another feature of the ETI-1412 is its anti-thump circuit. At switch-on in an audio set-up the first sign of life is normally heard as a thump from the speakers. In a system where many audio processors are interconnected, eg, in a studio, where powering up via a single mains switch is common practice, the thump can be quite a problem. In extreme cases it can actually damage the speakers. Some amplifiers produce loud thumps on their own at switch-on. The anti-thump circuit is incorporated to reduce these problems.

Description

A toggle switch is used to select loudspeaker A or B and two LEDs indicate which pair is selected. The switch and LEDs are housed in a hand-held remote control box connected via cable to the main unit.

In the early stages of designing the anti-thump circuit, it was decided to avoid having any form of circuitry connected across the loudspeakers as it would inevitably detract from the speaker units under test.

Furthermore deriving a trigger signal for the anti-thump circuit and power source from the amplifier was also avoided as it would mean dismantling and groping deep into the depths of your prized amplifier.

A simple solution used in the switcher is to have the anti-thump detector circuit across the power amplifier output only for the switch-on period. The trigger signal is actually derived from the audio output of the left power amplifier. When power is applied to the switcher, the relay contacts are configured so that the speakers are disconnected from the amplifier. At this point the anti-thump circuit is connected across the amplifier output and waits for a signal. When a change of dc level or an audio signal exceeds the threshold level of ▶

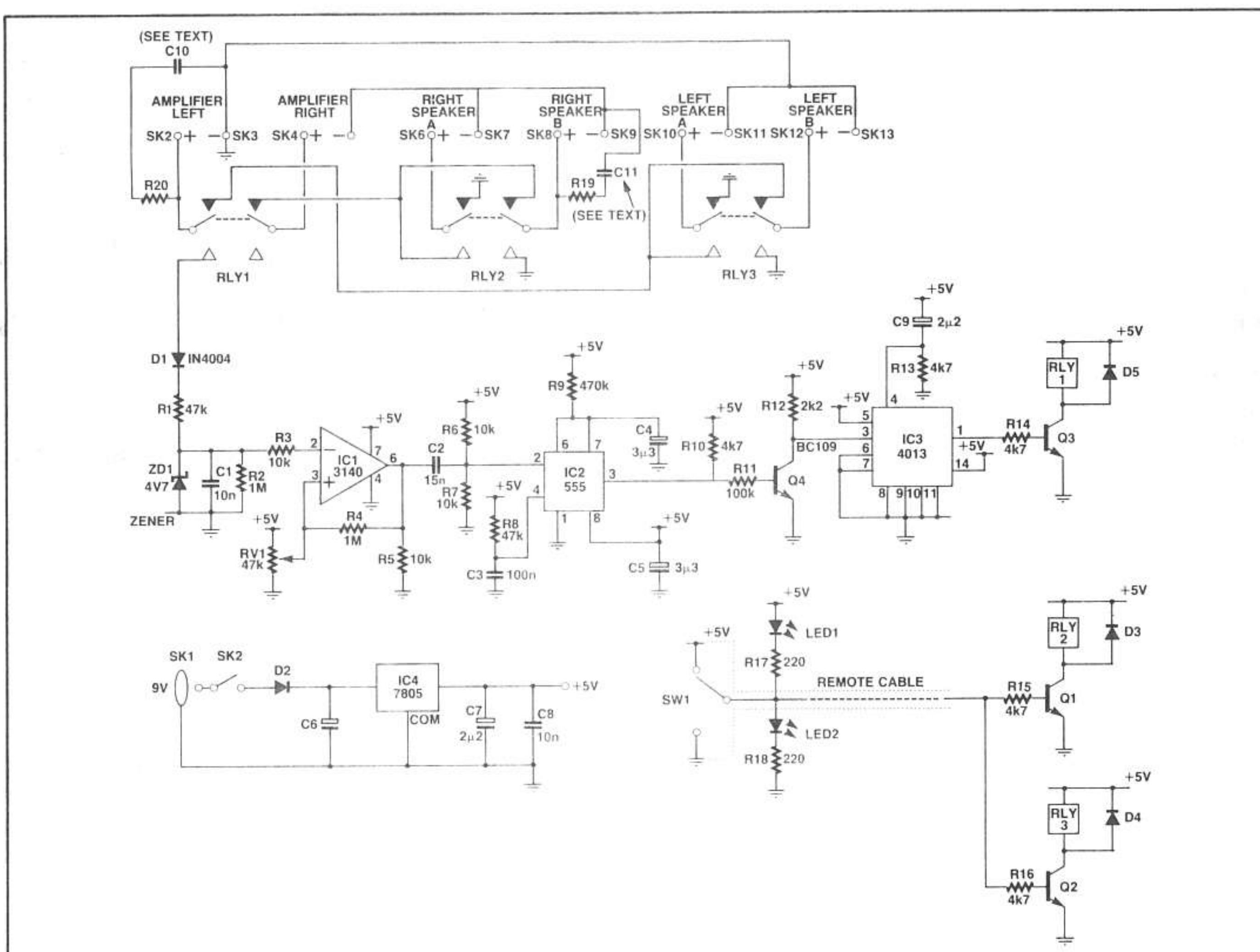
ETI — 1412 — HOW IT WORKS

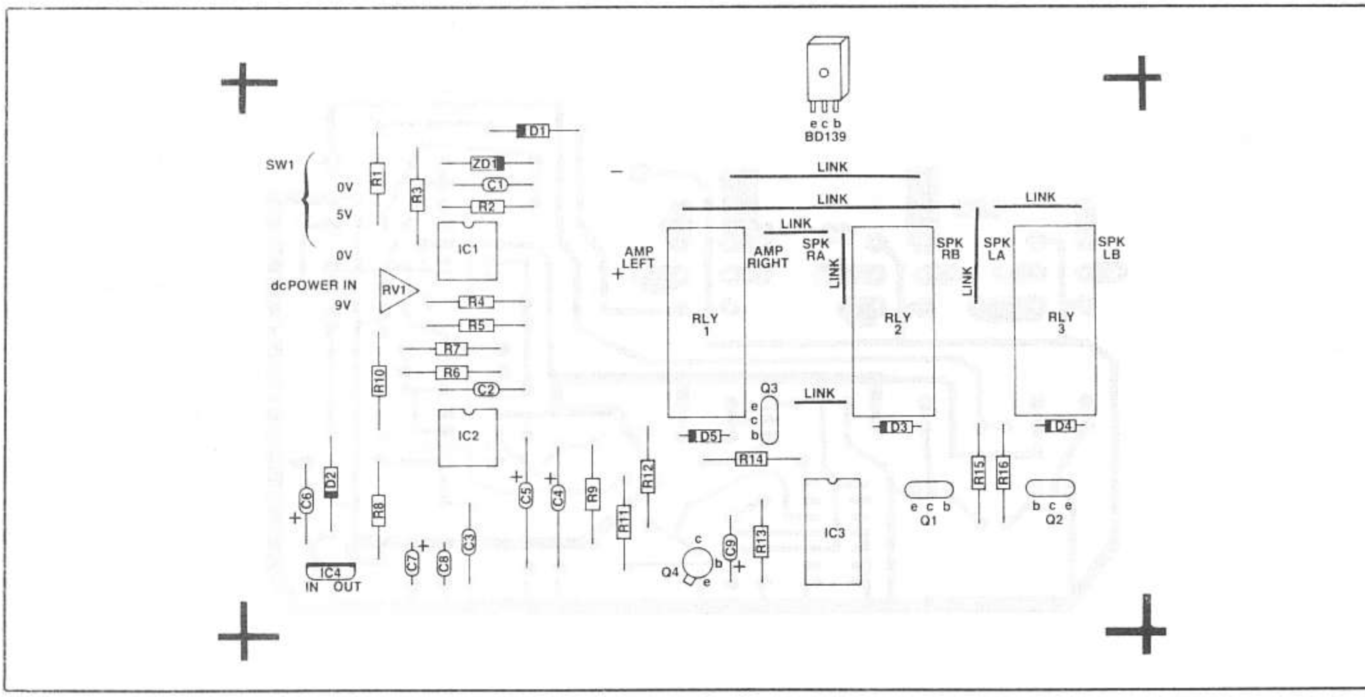
Relays RLY2 and RLY3 actuate the speaker switching from pair A to pair B and vice versa. The relays are configured to ground the speaker pair that is not selected to minimize crosstalk, eg, if pair A is selected then pair B is grounded. Q1 and Q2 are the transistor coil drivers of RLY2 and RLY3 respectively. The switching voltage is derived from SW1 in the remote control.

D1 is a half-wave rectifier and with C1 derives a dc signal from the output of the left amplifier to trigger the anti-thump. ZD1 clamps the voltage to 4.7 V for large voltage excursions. The rectified voltage is fed to IC1 which is configured as a Schmitt trigger.

The non-inverting input of IC1 has its threshold voltage varied by RV1. When the threshold level is exceeded the output of IC1 is driven low and triggers the 555 timer. When triggered, the timer will drive its output (pin 3) to a high for the timing period of about 2 seconds in this case. The on period cannot be re-triggered until it times itself out and is determined by R9 and C4 (see text). On powering up C3 and R8 ensure that the 555 is reset. IC3 is a D-type flipflop and latches its output to a high when the timing period of the 555 expires. This in turn will drive Q2 on and switch over the relay contacts of RLY1. Once the flipflop is latched, the only means of resetting it would be to turn off the dc supply. On powering up the flipflop is reset by C9 and R13.

The dc supply is derived from a 9 V power pack and IC4 provides a 5 V regulated voltage rail. The switcher draws a maximum of around 210 mA and occurs when RLY1, RLY2 and RLY3 are driven on. D2 protects the circuit from power supply reversal.





ETI-1412 — PARTS LIST

Resistors ....all 1/4 W, 5%, unless noted
- R1, 8 ....47k
- R2, 4 ....1M
- R3, 5, 6, 7 ....10k
- R9 ....470k
- R10, 13, 14, 15, 16 ......4k7
- R11 ....100k
- R12 ....2k2
- R17, 18 220
- R19, 20 ....8R2 5 W (see text)
- RV1 ....47k vertical preset

Capacitors
- C1 ....10n greencap
- C2 ....15n greencap
- C3 ....100n greencap
- C4, 5 ....3μ3 electrolytic 25 V
- C6 ....220μ electrolytic 25 V
- C7, 9 ....2μ2 electrolytic 25 V
- C8 ....10n ceramic 25 V
- C10, 11 ....220n greencap

Semiconductors
- LED1, 2 ....5 mm LED
- D1, 2 ....1N4004
- ZD1 ....4V7 Zener 1 W
- D3, 4, 5 ....1N4148
- Q1, Q2, Q3 ....BD139
- Q4 ....BC109 or equiv
- IC1 ....3140 FET input op-amp
- IC2 ....555 timer
- IC3 ....4013B dual D flipflop
- IC4 ....7805 5 V regulator

Miscellaneous
- RLY1, 2, 3 ....pcb mounting DPDT 6 V, 5 A 72 ohm coil resistance (Fujitsu 621D006)
- SW1 ....miniature SPDT toggle
- SW2 ....miniature on-off toggle
- SK1 ....dc power socket (2.1 mm)
- SK2-SK13 ....banana sockets or sockets of your choice: 8-pin DIL IC socket, 2-off; 14-pin DIL IC socket, 1-off

ETI-1412 pcb; metal cabinet (185 x 70 x 160 mm); Zippy box (28 x 54 x 83 mm); 9 V plugpack 300 mA; loudspeaker cable; 6 m twin shielded cable; regulator clip-on; heatsink.

Price estimate: $80
(excluding plugpack)

a Schmitt trigger things start to happen. A 555 timer will be triggered for around 2 seconds and when the delay period expires, the relay contacts will close and connect the amplifier output to the speakers. This should allow sufficient time for the various power supplies to complete charging their smoothing capacitors.

The anti-thump circuit is now set and can only be reset by turning off the power supply. In normal use this function is quite adequate and resetting is only required when the audio system is switched off.

Construction

Building the switcher should not present any problems as most of the components, except for the remote control, are mounted on the pcb. Start by checking the board for bridged or broken tracks. Once you are satisfied that the board is acceptable, drill all the necessary holes, including those for mounting.

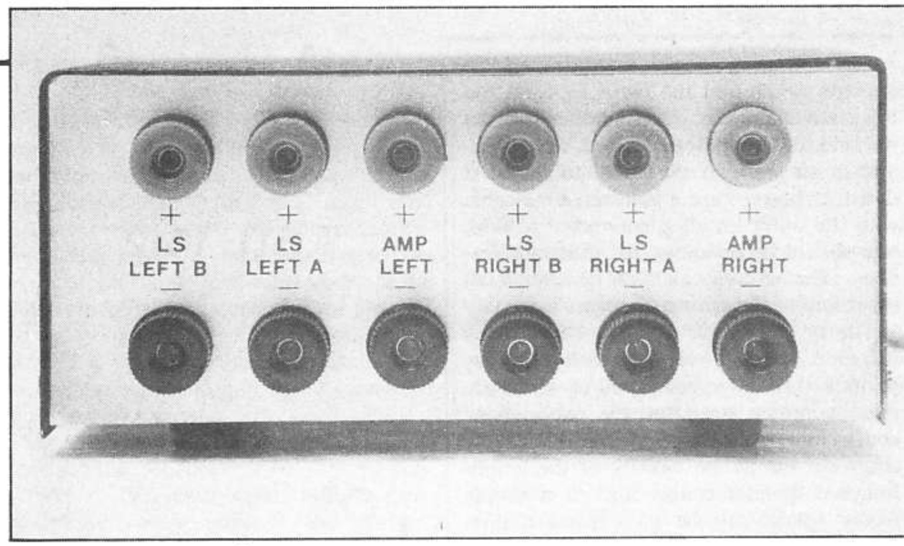
Next insert and solder the links, resistors and capacitors, taking care to get the correct orientation of the electrolytics. The IC sockets come next. Make sure that they are pressed down properly on the board when soldering. Do not insert the ICs themsleves until later on. Next proceed by mounting and soldering in the relays, RV1 and finally the diodes, transistors and regulators. Ensure that the polarity of the diodes is correct and that the transistors and the regulator are oriented the correct way. Refer to the component overlay.

Tin the copper tracks of the relay connections to the amplifier and speaker as they will be carrying high currents under load.

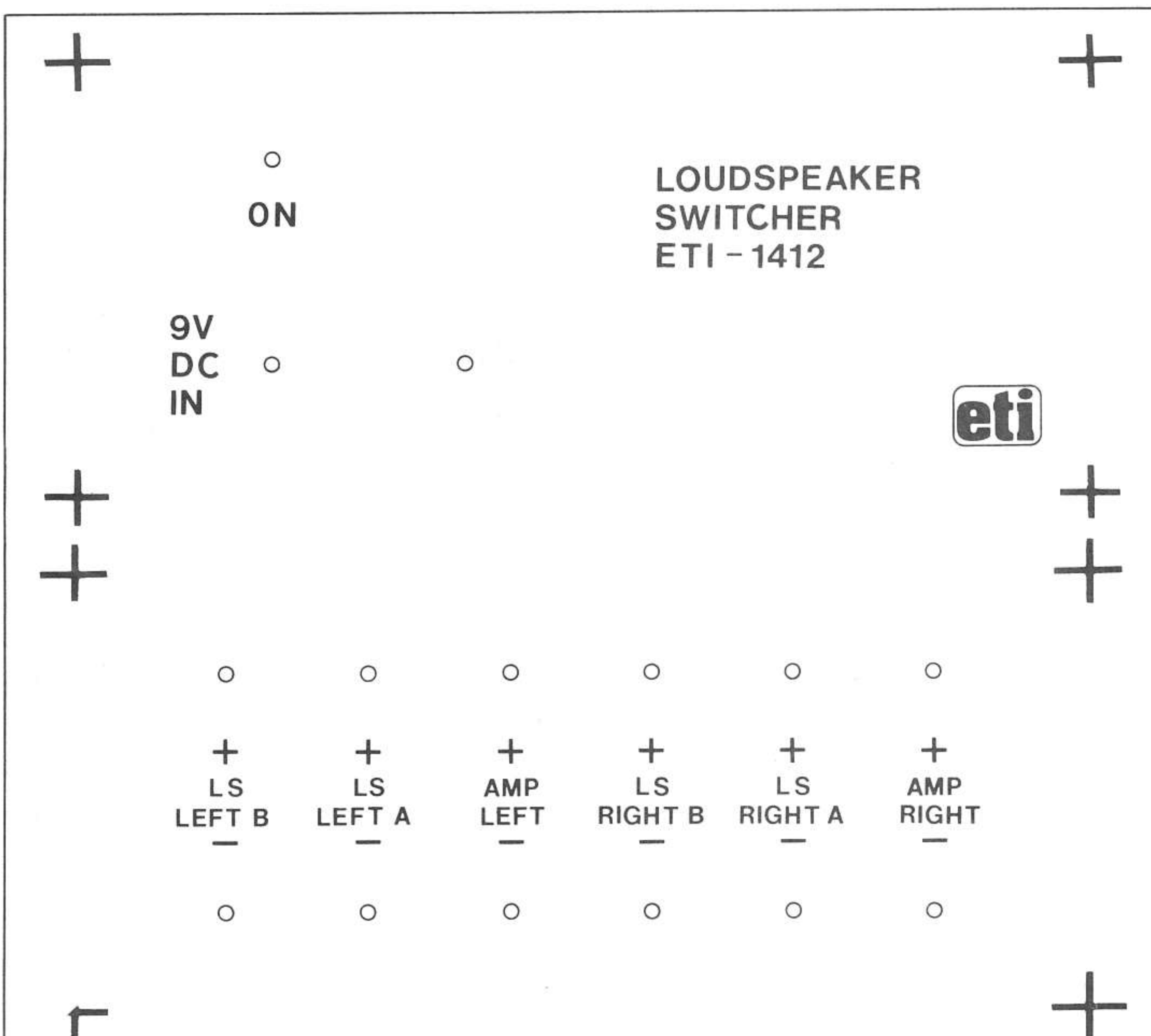
Before the wiring to the pcb can be completed, the switch and sockets need to be mounted on the front and back panel of the box. I have opted to use banana sockets for the speaker and amplifier connections, however other socket types may be used. Starting with the back panel, drill the holes for the banana sockets or sockets of your choice. Then proceed with the holes for the on-off switch, dc power switch and remote cable on the front panel. Once that is done mount the sockets and switches into position. A clamping rubber grommet should be used to fasten the remote cable where it leaves the box.

Use tinned copper wire (22 swg) to connect the negative terminal of the left amplifier to the negative end of the left A and B loudspeaker at the sockets on the back panel. (Refer to the wiring diagram.) Similarly do the same for the right amplifier and right A and B speakers. It is important to ensure that the negative terminal of the left and right channels is not connected as unwanted hum-loops can develop. Also make sure that the correct polarities are observed. Use suitable loudspeaker cable for the speaker and amplifier to socket connections. Note that only the negative terminal of the left amplifier is connected to 0 V on the pcb. The on-off switch and plugpack dc socket can now be wired in.

The construction of the remote control can now be started. Drill the holes for the ▶



Sockets on the rear panel.



selector switch and the two LEDS on the top plate of the box. Drill another hole on the side for the remote cable. Use a grommet or tie knot in the cable to fasten it down. I chose to use a twin screened cable with the outer braiding connected to 0 V; 6 m should be adequate for most applications. The writing can now be completed according to the wiring diagram.

The next exercise is to go back to the pcb and check it very carefully for dry joints and solder splashes, all of which are quite common even for the experienced constructor. It is important to check for short circuits in the vicinity of the amplifier and speaker connections as a simple solder splash can be an expensive one. Once you are convinced that all is well, insert the ICs into their sockets making sure of their correct orientation. Take the usual precautions with handling the CMOS devices.

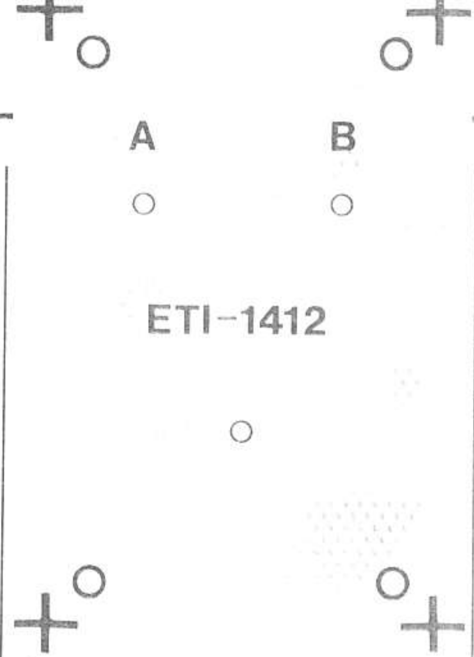
Testing

Connect the 9 V plugpack and switch on the dc power switch. If the measured voltage rail is not within 800 mV of 5 V, disconnect the plugpack and re-check the circuit in the vicinity of the supply rail. If all is well, one LED on the remote will light and toggling the selector switch will toggle RLY2 and RLY3. The relay switching can be heard as the contacts change over.

Next, measure the dc voltage at pin 3 of IC1 and adjust RV1 for a reading of 600 mV on the multimeter. This is the maximum sensitivity for triggering the anti-thump circuitry. Connect the amplifier and speakers to the unit and once that is done, power up the amplifier with the volume control turned right down. Gradually increase the volume control, with a sound source connected to the amplifier, of course. When the signal level exceeds the threshold level, IC1 will trigger and start the delay time thereafter connecting the speakers. The sensitivity can be decreased by adjusting RV1 for a higher threshold voltage. If you want the time delay changed then it is a simple matter of changing the value of R9 or C4. The time delay is given by the equation: $td = 1.1 \cdot RxC$.

Once the circuit is triggered it is time to use the switcher for switching. Toggling the switch on the remote should switch-over your speakers from pair A to pair B or vice versa. If not, then check your wiring to the back panel of the unit and then the amplifier and speaker connections.

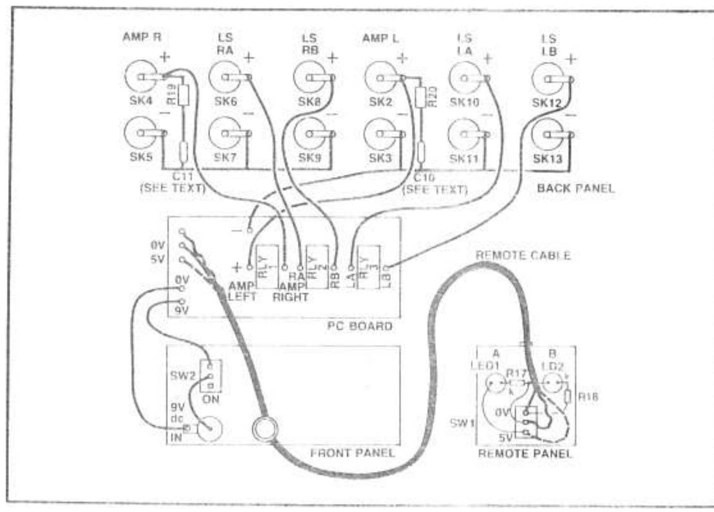
The 5V regulator (IC4) will get warm as the circuit will draw around 210 mA when all the relays are driven on. A small heat-sink clip mounted on the regulator will help dissipate some of the heat.



In use

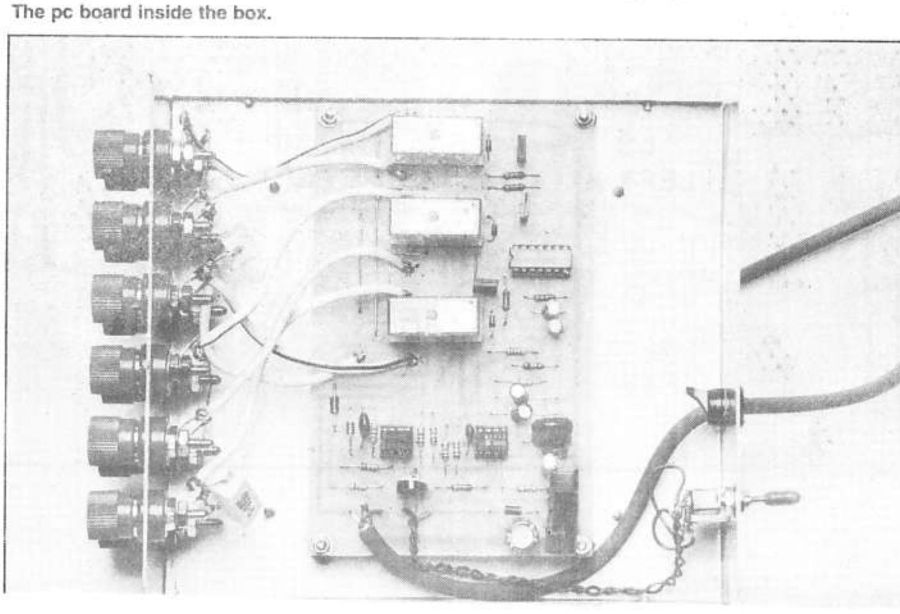
The switcher will cope with a maximum power rating of 100 W into 8 ohms per channel with the stated relays. We recommend you do not switch near or at full power due to the problems of switching highly inductive loads. It's far better to be safe than sorry. Furthermore the life expectancy of the relays tends to decrease when switching high currents.

Some switching noise will be apparent due to the switching of inductive loads and open-circuiting the amplifier outputs for the change-over period. For these reasons it is imperative to use an amplifier that remains stable under these conditions. Certain valve amplifiers, for example, will protest quite strongly by exterminating themselves under no load conditions. Also, avoid using ac coupled output amplifiers, remember them? The switching noise heralds itself as a click and is dependent on signal level, amplifier, crossover and speaker type. As an option, connecting in R19, C11 and R20, C10 across the right and left amplifier respectively should help reduce the switching noise a little by introducing a load at high frequencies. These components can be mounted at the sockets on the back panel (see the wiring diagram). My colleagues and I felt quite strongly about having minimal additional circuitry in the audio path and at average listening levels the transient noise is negligible. ●



Wiring diagram.

The pc board inside the box.

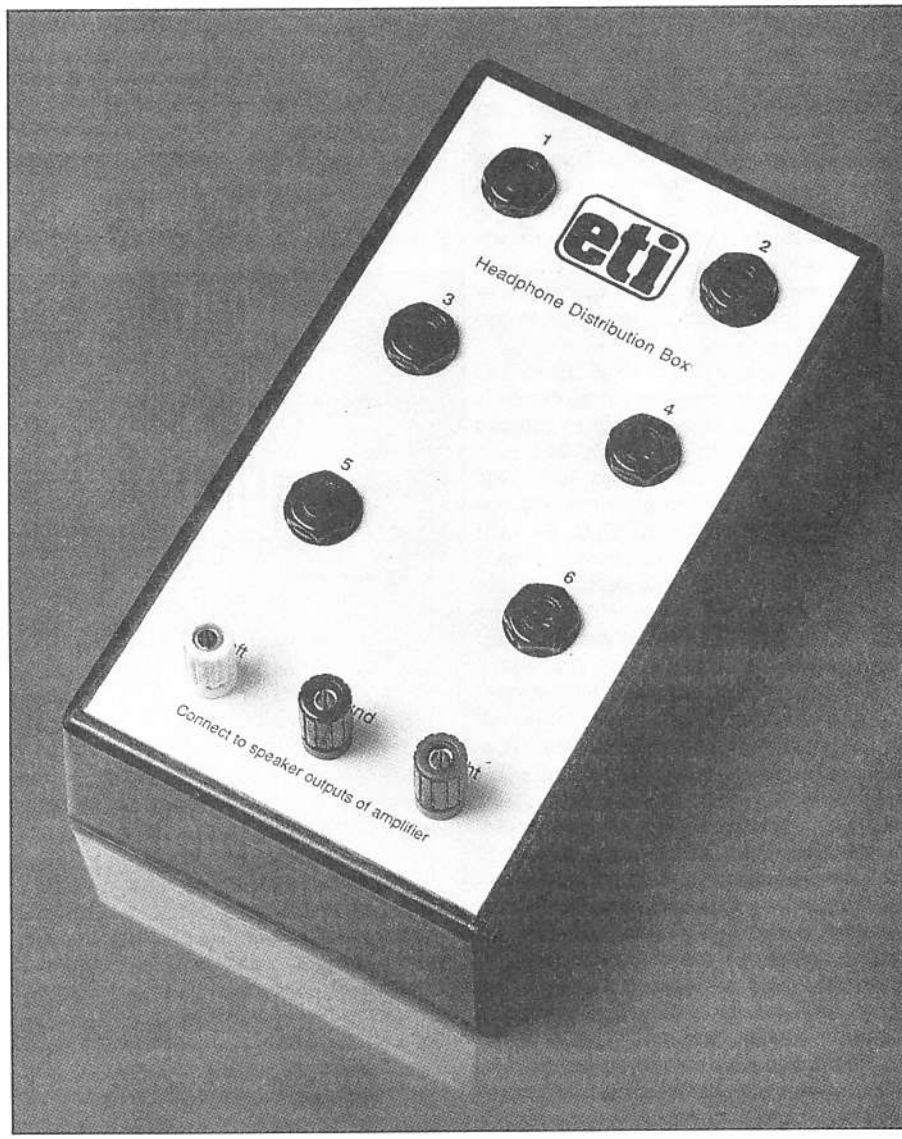


ELECTRONICS
ETI - 1411

HEADPHONE DISTRIBUTION BOX

A project for the home recordist

Professional studios use headphone distribution boxes — you can make one to suit your home studio needs. Greg Simmons tells how.



There comes a time in the career of every ambitious home recordist when the need for more than one pair of headphones arises. Many home studios use domestic hi-fi amplifiers for driving the monitor speakers, which, typically, allows two pairs of headphones (one pair connected to the mixer headphone output, and another pair connected to the headphone socket of the amplifier). Anyone with an electronics background will be able to wire more headphones into the system using a combination of series and parallel wiring. Unfortunately, this is often messy, time consuming and generally not very satisfactory.

Professional studios use headphone distribution boxes which allow up to, say, six pairs of headphones to be connected to the speaker output of an amplifier. If the headphones used are all the same, each pair will receive the same level. If more than six pairs of headphones are needed, another distribution box can be connected to the amplifier.

Many studios use professional 50 to 100 watt amplifiers to drive the headphones(!). The average pair of headphones requires less than 1 watt for high listening levels, so you may wonder why studios use such high powered amps. The reasons for this are similar to the reasons why 20 to 30 watt reference monitors (Auratones, NS10s, etc.) are connected to high powered amplifiers.

Firstly, the headphone amplifier, like the monitor speaker amplifiers, is usually turned on at the start of the day's work and remains on until the studio is shut down for the night (or early morning!) Obviously, the amplifier must be able to withstand this long workload. A 50 watt amplifier delivering a few watts of power to headphones is not going to be working very hard, and there is less risk of it overheating and malfunctioning. In addition to this, there is little risk of amplifier clipping, since the amplifier is running well within its capabilities. Amplifier clipping is possibly the biggest cause of tweeter and headphone voice coil burn-out. It also makes things sound BAD.

Most home studios don't demand this much from their amplifiers, nor can they afford to buy a 50 or 100 watt amplifier for the headphone system.

The Sonics headphone distribution box allows six pairs of high impedance headphones (40 to 600 ohms) to be driven by a 20 to 40 watt amplifier, such as most domestic hi-fi amplifiers. It can be assembled in a few hours using parts that are readily available from electronic component suppliers including Jaycar, Dick Smith Electronics, David Reid Electronics, Tandy, etc. There is no printed circuit board needed, and the whole thing costs approximately $35.00.

Circuit details

The circuit diagram is shown in Figure 1. The circuit is designed to drive up to six pairs of stereo headphones with impedances of 40 ohms or more. For best results use headphones of the same or similar impedance, ideally the same models.

The resistors labelled Rh form voltage dividers with the headphones, and protect the amplifier from being damaged. Even if all six headphones go short circuit, the amplifier never has to drive a load lower than 3 ohms per channel.

The resistors labelled RL ensure that with no headphones plugged in, the amplifier is still connected to a load. Many amplifiers do not appreciate being turned on without a load connected to their outputs – I believe this can sometimes lead to amplifier damage, but I don't intend to prove it.

The circuit diagram shows the stereo

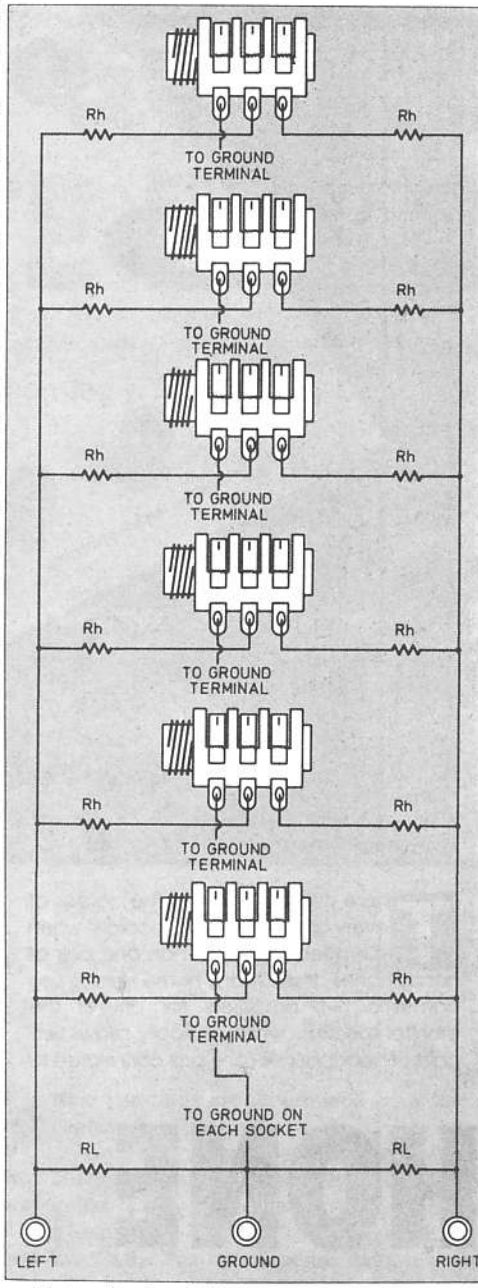


Figure 1: circuit diagram.

sockets as they appear in real life, simplifying the construction and (hopefully) ensuring that everything gets soldered in the right place.

Construction

The unit is housed in a plastic 'zippy' box approximately 11 cm x 20 cm x 6 cm. Since a headphone distribution box spends most of its time on the floor, you may want to build it into a tougher metal or diecast box. If so, make sure the sockets, terminals and wiring are electrically insulated from the box, or else you may create a short circuit and damage the amplifier.

All sockets and terminals in the prototype were mounted on the bottom of the box, as this is much more rigid than the lid. Start by marking out and drilling all the holes. The template provided in this article will make the job easier, and also serves as a front panel. Just photocopy it, cut it out and fix it to the box with a thin smear of glue (I used a 'UHU' glue stick). Make sure the entire template is glued down, not just the edges, or else the holes will not be accurate. Drill small pilot holes first, and don't push the drill too hard or else it will damage the edges of the holes when it comes through the other side, and may also tear the template (you did use a photocopy, didn't you?) After the holes are drilled, clean up the edges and put a layer of clear self-adhesive contact onto the front panel. This will protect the template and provide a durable finish. Use a scalpel or blade to cut the appropriate holes in the contact for the sockets and terminals to go through.

The prototype uses stereo 6.35 mm sockets, available from Jaycar. They are made from black or coloured plastic, with three metal strips across them that serve as contacts. These particular sockets make it easy to see what contact connects to which part of the headphone plug. Being plastic, they can also be used in metal boxes with no insulation problems.

Figure 2 shows the bottom of one of these connectors. Underneath, you will find six terminals, a row of three on each side. Make sure you use the row that connects to the long strips across the top of the connectors. (The terminals that connect to the short strips are used for switching only, and do not connect to anything when the plug is in the socket.) If you use a different type of stereo 6.35 mm socket, you may have to use a multimeter to find out which terminals connect to the left, right and ground connections of the plug. Sometimes you can work this out by carefully studying the construction of the socket.

The next step is to solder the components onto the connectors. The easiest way to do this is to mount the stereo sockets upside down, so that the sockets are on the outside of the box. You can now solder all the components in place with relative ease (don't make any connections to the three 'speaker' terminals yet). Solder the components to the contacts on the stereo sockets exactly as shown in the circuit diagram.

Once everything is soldered in place, unscrew the 6.35 mm sockets and carefully mount the completed circuit inside the box.

The prototype uses three 4 mm 'banana'

terminals for the connections from the amplifier, each one a different colour – white for left, black for ground and red for right. Put the three terminals in place and make the final three connections to them. (You may want to use a 3-pin XLR connector, or a stereo 6.35 mm socket instead.) Double-check your workmanship, screw the lid on, and you're ready to test it.

Testing

Connect the headphone distribution box to a 20 to 40 watt amplifier with a length of 3-core power lead, or two lengths of 2-core speaker wire (avoid using shielded signal cable). Make sure the amplifier is turned off while doing this.

Most amplifiers have four terminals for speaker connections. Two of these will be for ground, or negative, usually marked with a '–' symbol. To connect the distribution box, you must link the two ground terminals together on the back of the amplifier, as shown in Figure 3. These terminals are often linked together inside the amplifier. There are four terminals because each speaker has its own speaker lead, and each lead must have a ground connection.

With headphones, there is only one lead from the amplifier, and since the ground is common to both channels, only three connections are used. (Some amplifiers have totally separate power supplies for each channel, and linking the grounds may degrade performance a little. Fortunately, these amplifiers are generally high powered and unlikely to find themselves being used for headphone amplifiers in home studios. If you're uncertain, consult the owner's manual or a service centre.)

With the connections made, turn the amplifier volume down to minimum and switch it on. Plug some headphones into the box, apply a stereo signal to the amplifier, and slowly bring up the volume until you get a decent monitoring level with no signs of distortion and a good stereo balance. If not, you may have to check the wiring. If anything starts to smoke you have certainly made a wrong connection somewhere! You have my sympathy.

If you cannot get enough level, and all the wiring and soldering proves to be correct, you are probably using very low impedance headphones, less than 40 ohms. Although the distribution box is not designed for this, using a higher powered amplifier might solve the problem. Be careful when you first try it out, though . . .

In use

For best performance use headphones of similar impedance. The circuit is designed so that plugging in an additional set of headphones should not noticeably affect the level of those already connected. Using different impedance headphones may result in different levels, and it will be necessary to find a compromise amplifier volume setting. Try giving the louder headphones to the drummer and bass player – that should keep everyone happy.

Most home studio mixers provide a mono foldback/headphone signal. If so, be sure to connect the appropriate mixer output to both left and right inputs of the amplifier, otherwise you will only have signal in one side of the headphones. Alternatively, if the amplifier has a mono switch, use this instead. Hopefully, all has gone well and you are now the proud owner of a sonics headphone distribution box. Happy headphone monitoring!

PARTS LIST — ETI-1411

- 12 x 18 ohm, 5 watt resistors (Rh)
- 2 x 100 ohm, 5 watt resistors (RL)
- 6 x stereo 6.35mm sockets (see text)
- 3 x 4mm 'banana' sockets (see text)
- 1 x 'Zippy' box – 11cm x 20cm x 6cm
- 1 x 20cm length white hookup wire
- 1 x 20cm length black hookup wire
- 1 x 20cm length red hookup wire
- 1 x 3m length 3-core power lead

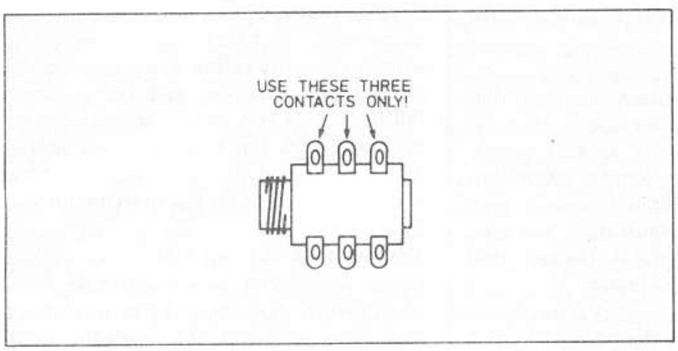


Figure 2: view of stereo socket from underneath.

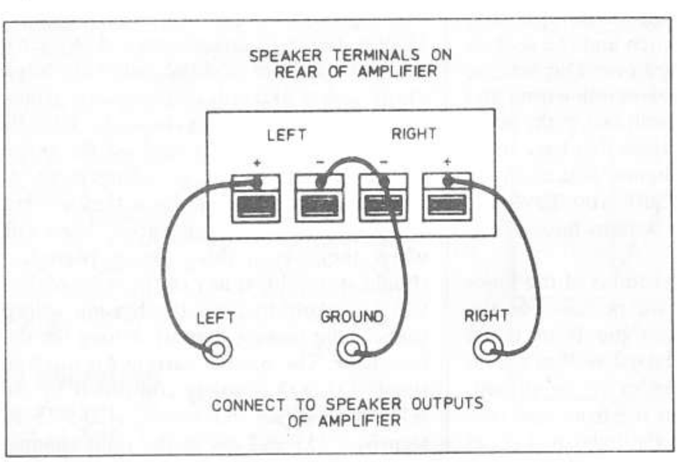
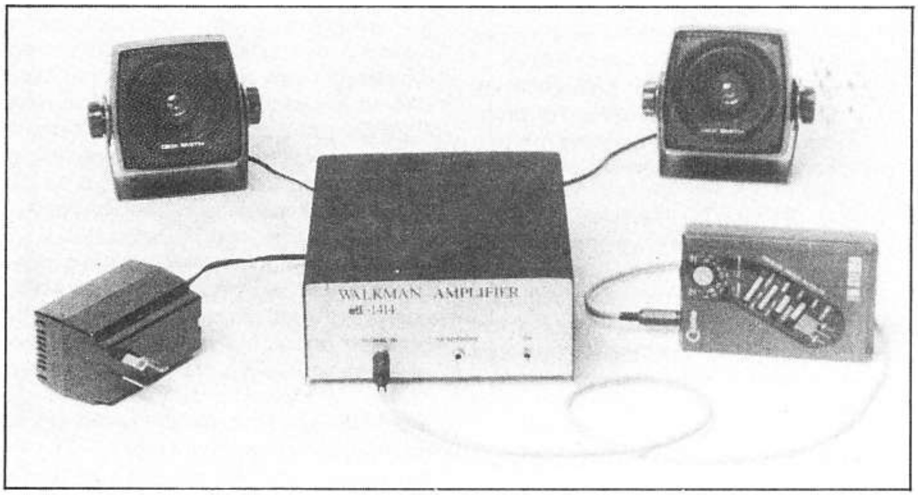


Figure 3: connections from stereo amplifier to headphone box. (Note the link between the negative terminals of the amplifier).

WALKMAN AMPLIFIER

S K Hui



Speaker connectors and the plug pack input sockets are mounted on the rear panel.

THE WALKMAN TYPE radio and tape player is one of the most popular electronic gadgets of modern times. Surprisingly, their fairly high price tag has survived simply because of one unique feature — portability.

Their biggest disadvantage is that the output is available to only one person. The aim of this project is to change that. One solution would be to put the line-out of the walkman directly into a big amplifier. If you experiment with this you will find that the walkman doesn't sound nearly as good as it used to. This is because it uses the headphone cord as part of its antenna system. In fact, the antenna is coupled into the audio output section of the walkman via a pair of capacitors. A second problem is that the essential portability of the walkman system is lost because the amplifier will require mains power.

So the requirements of a walkman amplifier are that it should be able to match the antenna correctly in its output, be small, drive a set of small speakers to a reasonable loudness level and be powered by batteries.

The ETI-1414 walkman amplifier has been designed deliberately to tackle this requirement. Every walkman owner should build one or your unit is just like coffee without coffeemate.

The circuit is housed in a tough mild steel box. It can be powered from either an internal battery pack, a 12 V dc plug pack for use as a bookshelf amplifier or directly from the cigarette lighter socket in your car. The spec' for this little beauty is shown in the table.

Construction

The emphasis of this project is on portability and simplicity. Construction of this project should take no more than a couple of enjoyable hours on a Sunday afternoon. By the evening you should have a fully working amp on your bookshelf. No time consuming tuning is required. The only tool used in testing is a multi-meter.

The pc board itself sits on four plastic standoffs. One edge of the pc board has a switch (SW1), and two 3.5 mm phono sockets (SK1, SK2). This side of the board has to be flush against the metal front panel of the box to allow the switch and the sockets to be accessed from the front. This scheme reduces a lot of time consuming wiring and mistakes. The only difficult task is the accurate positioning of the holes that have to be drilled on the box. So before you assemble the pc board, make sure you have the mounting holes on the bottom floor of the box for the standoffs.

In order to get the positions of the holes drilled correctly, place the pc board on the floor of the box against the front metal panel. By using the pc board itself as a template mark the four holes to be drilled. Next, drill the holes on the front and rear panels of the box. Use the published panel artwork in the article as a drilling template if you like, but don't touch your plastic Scotchcal panels at this stage. Save them for last, as they are easily damaged. To ensure the holes drilled have the right sizes, use the actual components to try them out. A round needle file or a small reamer may be needed. Always start with a small drill bit and gradually enlarge it to the size required with a file or a reamer. It takes longer but you get a perfect hole.

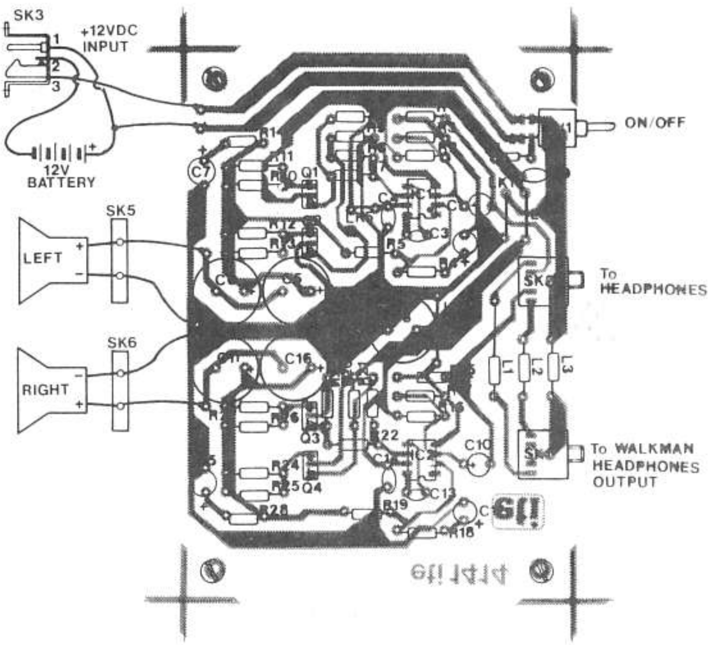
Apart from getting the right sizes, it is also important to get the right positions. Especially the three holes drilled on the front panel for SW1, SK1 and SK2 as they are rigidly soldered onto the pc board. Solder the three components on the pc board and see if you can mount the board on the standoffs. Usually, a bit of enlargement of the holes and re-adjustment of the board is needed.

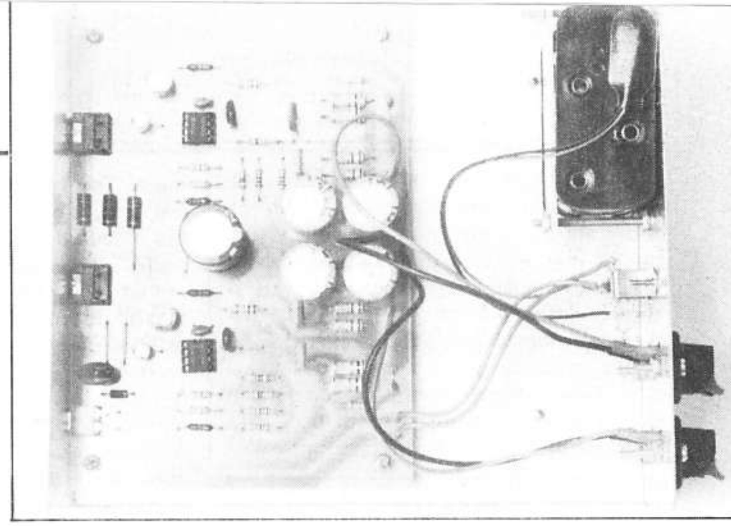
When you assemble the pc board, it doesn't matter which component you start with. If you have all the components available, the assembling should not take more than 45 minutes. To ensure success, watch out with the polarity of the transistors, diodes, electrolytic capacitors and the op-amps. Putting a heat sink on the transistor is not necessary with this type of power output. You can do so if you really wish but make sure the heat sinks do not short circuit each other.

Testing and Setting Up

Before you flick the power switch on, check again that the polarity of the battery or plug pack input is correct. Although the amplifier circuit is protected by diode D1 against any reverse power input, the testing procedure cannot proceed with the wrong input. Unplug the cord connecting the Walkman and the amplifier and the switching action of the socket SK1 will automatically ground the left(L) and right(R) channels. With the speakers disconnected, turn on the power switch (SW1) and put your fingers on the transistors for 30 sec to see if they get hot. Under normal circumstances, especially when there is nothing being played, it should stay cold. If any of them do get hot, you are likely to have the biasing voltage (hence, the biasing current) wrong for that transistor. The biasing current for the transistors Q1, Q2 is solely controlled by R6, R7, R8 and R9. Likewise, R20-R23 for transistor Q3 and Q4 in the right channel. Check that you have the right resistors in the right places.

Next check the output voltages of the cir-

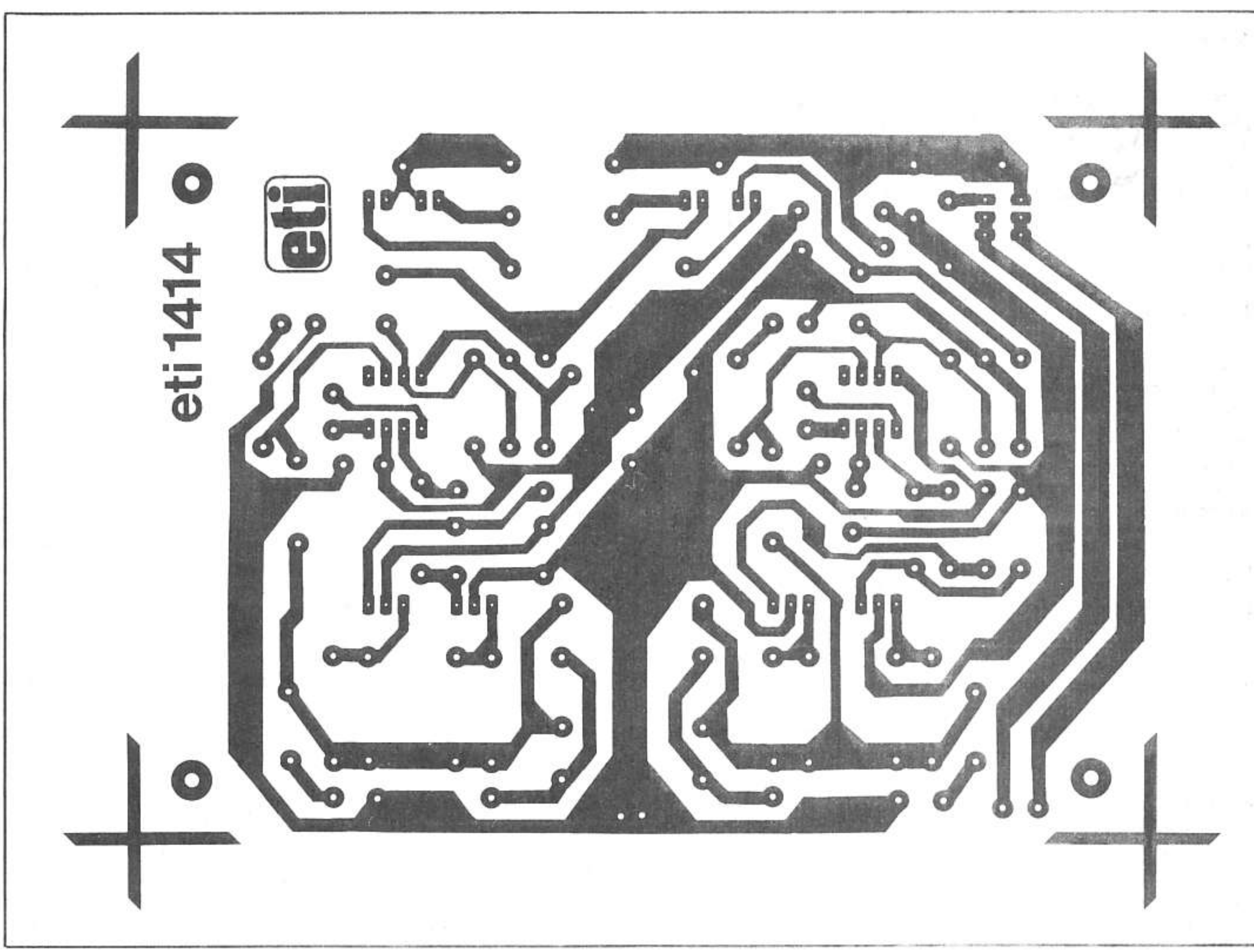




Top view of the pc board and the battery holder inside the box.

Specifications

Test Conditions: 4 ohm load, +12V supply, 400mV ptp input at 1kHz
Frequency Response (−3dB point): 40 Hz to above 100kHz
Sensitivity: about 0.6V
Total Distortion: between 0.1 and 0.3%
Output Power (rms(: 0.8W per channel
Cross-talk: 400mV at 1kHz injected in one channel with signal amplitude measured at 80dB on another channel.



cuit with a multi-meter. The node where the four 1 ohm resistors are joined should be sitting at 6V dc above ground for a 12V supply. This allows a maximum swing in both directions above and below 6V. If it is slightly different from 6V, it is likely that the power supply voltage is not exactly 12V. In the circuit diagram, several vital points have their voltages given so a comparison can be made on yours. Check them with your multi-meter's dc voltage measurement. The values you obtain on yours do not have to be identical to mine, usually, a few hundred millivolts discrepancy is expected. The final check is on the speaker output connector SK5 and SK6. Make sure that no dc voltage (0V) exists at that point, or your speaker will be permanently damaged.

The last thing to make is a cable connecting the walkman headphone output into the 'signal-in' socket on your amplifier. Preferably, use a double-core (stereo) screened microphone cable. A standard 3.5 mm stereo phono jack is soldered on each end of the cable. Apart from serving as an audio link, the cable acts as an antenna for your walkman radio (see How it Works). Obviously, for better reception, the cable should be a generous length.

Connect the speakers the connectors SK5, SK6. Plug the cable link into the walkman and the amplifier and you are ready to go! Don't forget if you are using batteries as a power source, use new ones. ●

We would like to thank Dick Smith Electronics for supplying the walkman radio and speakers used in the development of this project.

Parts List

Resistors (all 0.25W, 1% metal film unless stated otherwise)

R1, R4, R15, R18	1k5
R2, R3, R16, R17	120k
R5, R19	10k
R7, R8, R21, R22	330R
R6, R9, R20, R23	2k7
R10, R11, R12, R13, R24, R25, R26, R27	1R (0.5W, tolerance 5%)
R14, R28	2R7

Capacitators

C1, C10	10uF/25V (Elec)
C2, C12	22uF/25V (Elec)
C3, C13	220pF (Disc ceramic)
C4, C14	33nF (Green)
C5, C6, C16, C11, C8	470uF/25V (Elec)
C7, C15	1uF/35V (Elec)
C9	120nF (Green)

RF chokes

L1, L2, L3	470uH (From Dick Smith Electronics Cat L-1811)

Semiconductors

IC1, IC2	NE5534
Q1, Q4	BD139
Q2, Q3	BD140
D1	1N4002 or iN4004

Miscellaneous

A standard rectangular battery holder for carrying eight 'AA' cells, Dick Smith cat. -6128. A single sided pc board. PC board mounting mini toggle switch (Sw1) for ON/OFF. I could only find this switch from Dick Smith Electronics. Two stereo 3.5mm phono sockets (SK1, SK2) with switching action can be obtained from Jaycar Electronics. A 2.5mm DC power socket with switching action (SK3), Dick Smith cat. p-1665. A metal box about 185 x 70 x 160mm, Dick Smith cat. H-2744. Two stereo clip-on type speaker connector Dick Smith cat. H-6770.

The following parts may or may not be supplied in the kit: The stereo audio cable link between the walkman and the 'Signal In' on to the amplifier. A piece of plastic or bare pc board with two 6BA screws and nuts for clamping the battery holder. For obvious reasons, a pair of book shelf speakers will definitely not be included.

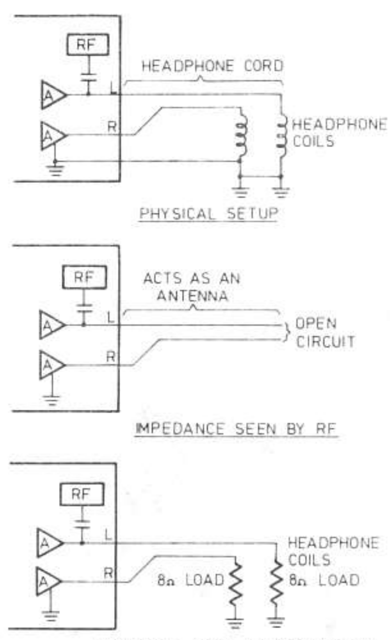


Figure 1: The walkman antenna system.

How It Works

Input section

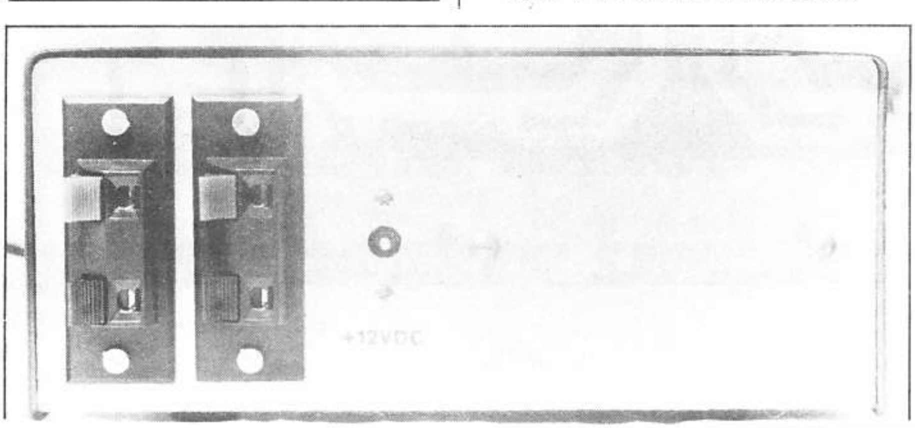
The input section begins with three RF chokes in series with the signal. Their function is to stop the amplifier input impedance from loading the Walkman antenna. To get good radio reception, a Walkman radio uses a headphone cord that serves as an antenna. The headphone coils inside the headphone have such a high impedance at RF frequencies that it is nearly an open circuit as far as RF frequencies are concerned. For audio frequencies, the Walkman sees a dc load of about 8 ohm (for an 8 ohm impedance headphone). Without the coils, the cable (antenna) will be loaded by the input impedance of the amplifier of about 1.5 k. Figure 1 shows the effective impedance as seen by the audio and the RF frequencies.

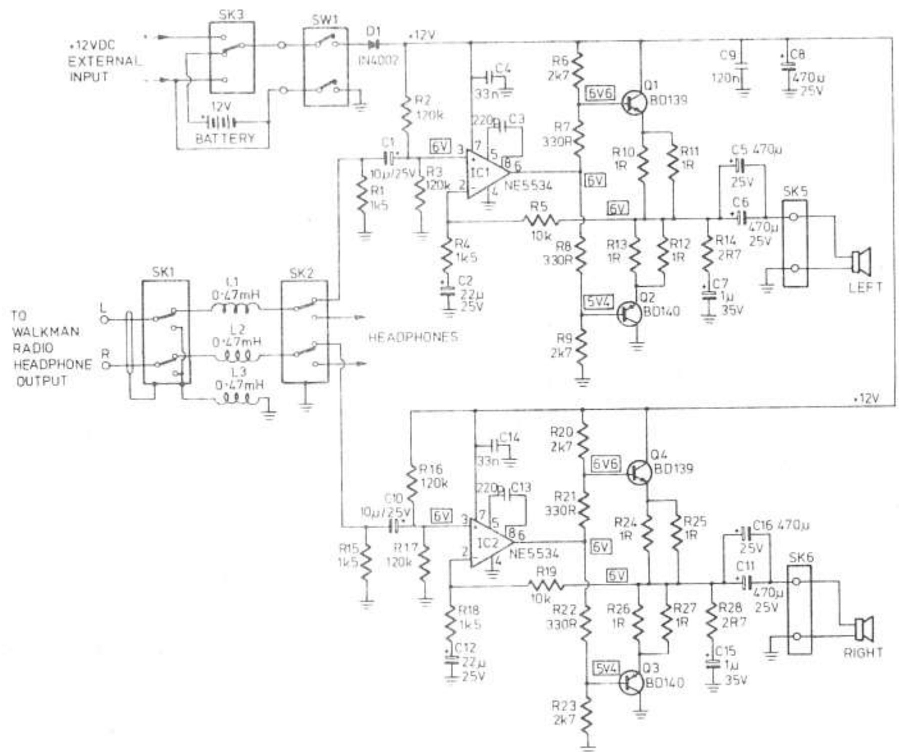
The audio input impedance of the amplifier is simply equal to the parallel resistance of R1, R2 and R3 (R16, R15 and R17 for the right channel). It works out to be around 1.5 k, which is a compromised value after some iterative testing. Lower impedance is possible but will start to load the Walkman output, resulting in a poorer transient and bass response. Any higher impedance and the coils (L1, L3) will start to pick up strong signals nearby like a generator or a alternator, etc. The cutoff, frequency at the input is determined by an emperical equation,

$$\text{Low Frequency Cutoff} = \frac{1}{2\pi\, C1 \times 1.5\,K} \quad 11\ \text{HZ}$$

The amplifier

The amplifier is basically configurated as a high current output op-amp by buffering the output of the op-amp to drive a low impedance speaker. Since the supply is only fixed at +12 V, it is desirable to have the ouput sitting at a dc level of +6 V for maximum swing in both directions. To achieve that input level, pin 3 of the op-amp is fixed at +6 V with a potential divider consisting of R2 and R3. Using the op-amp as a linear amplifying device (as opposed to a comparator), the inverting input of the op-amp (pin 2) must carry the same dc voltage (neglecting input offset voltages). Hence the dc voltage on the output of the op-amp is also fixed at +6 V. If the bases of the npn and pnp transistors connect directly to it, a standard class B amplifier is obtained. The usual cross-over distortion is fairly bad for class B operation, so the transistors have to be biased slightly on during no signal conditions. For the output (where R10-13 joins) sitting at +6 V, the base of Q1, Q2 must be sitting at roughly 0.6 V higher/lower than +6 V respectively. This requires that resistors R6 and R7,R8 and R9 be a fixed ratio. To calculate that, assume we have a +12 V supply and the op-amp output is sitting at exactly 6V. We want the





base of the pnp transistor (Q2) sitting at $6 - 0.6\ V = 5.4\ V$, so we have

$$\frac{6\ V}{(R8 + R9)} \times R9 = 5.4\ V, \text{ hence } R9 = 9(R8)$$

The above equation only gives an indication of their ratio but the actual value depends solely on the gain of the output transistors. After some trial-and-error, I came up with 330R and 2k7 for R8 and R9. The output emitter resistors R10-R13 form a current feedback to stablize the quiescent current due to fluctuation supply voltage. The higher the emitter resistance, the more the stablizing effect. But at the same time, the output voltage swing is more limited due to the drop in the resistors and results in smaller output power. Hence a good compromise is about 0.22 R, but I have checked around with a few major kit supplier only to find 1R in a 0.5 W package. In my design, I have used two commonly available 1 ohm resistors connected in parallel to achieve a 0.5 ohm resistance.

Feedback

The network is controlled by R5, R4 and C2. In the design, the dc voltage feedback from R5 is one. Since dc voltage can't go past capacitor C2, the +6 V dc on the output of the transistor is fully fed back directly onto pin 2 of the op-amp. This overall dc feedback is essential to keep the quiescent state of the circuit unchange. Any drift to voltage on the output of the circuit will cause an opposing change on the op-amp output to suppress the drift.

On the other hand, ac signals are fed back with a ratio determined by R4/ (R4 + R5). A five second mathematical manipulation shows that the gain for the ac signals is equal to;

$$\text{AC Voltage Gain} = 1 + \frac{R5}{R4} = 7.6$$

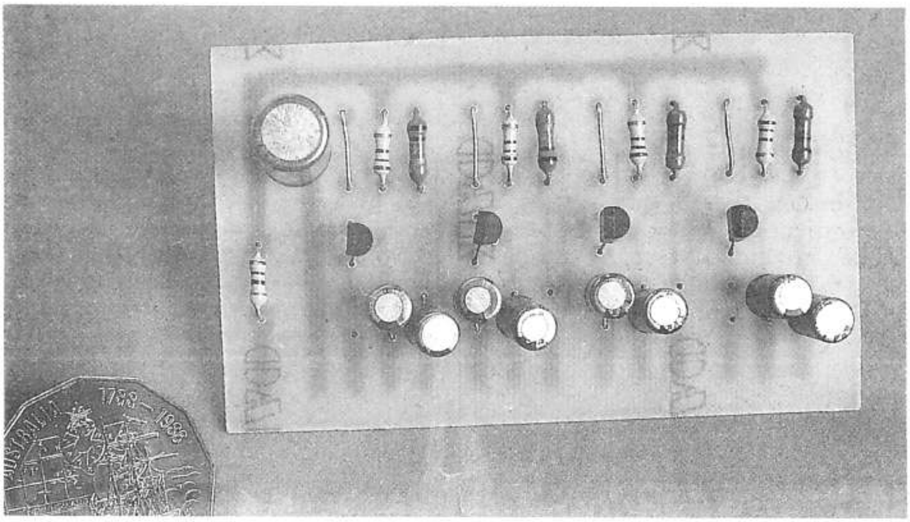
Bear in mind that this gain figure is frequency dependent. As frequencies drop to below 100 Hz, the gain will be largely determined by C2 and R4. The high frequency gain roll off at a point controlled by the compensating capacitor C3.

Wiring

Sockets SK1 and SK3 are 3.5 mm stereo phono sockets with in built switching action. As indicated in the circuit diagram, with the cable link unplugged from the amplifier, the coils L1 and L3 are grounded to avoid picking up any spurious signals. SK2 will divert the input signals from the Walkman to the headphone if it is plugged into the socket and hence, the speakers will be quiet. Socket SK3 is a dc plug pack type socket with a single pole switching action. If nothing is plugged in, the internal battery pack is connected to the circuit board. If an external power source is plugged in, the battery pack is switched out and power is drawn directly from the external source.

THE AUDIO TOOLKIT

Here's a project, or rather a series of adaptable projects, to solve those little audio problems or requirements that arise from time to time. By Graham Dicker.



Over the years, thousands of audio projects have been published and many constructors have assembled them with a great deal of success - only to find that they wish to modify or add some facilities to "personalise" the project. This usually requires that a breadboard-type circuit be constructed to provide the facility missing and if a pc board is not designed and made, then the mechanical reliability suffers. This is where the Audio Toolkit comes in.

A standard printed circuit pattern has been designed to provide a number of different single stage building blocks. They can be assembled in any number of modules up to a maximum of four or eight per board. There are five basic designs in the series, each with a different set of parameters and uses. All have been arranged with simplicity of design and construction foremost while offering staggering performance where required, even to full broadcast standard used by radio and television stations.

Let's take a look at the five circuit combinations.

Balanced input dynamic mic preamp

Circuit A is a high quality, very low noise, differential input, balanced line mic preamplifier. The circuit consists of a single stage BC549C transistor which has the inverting input from the Cannon (XLR) connector pin 2 to the base via a 4u7 tantalum capacitor. The use of good quality tantalum capacitors here is paramount to obtaining the noise figures as a standard electrolytic has far too much inherent noise and leakage. Should you wish to improve the noise figure, polycarbonate capacitors could be used if mounted on end.

This input is configured with the amplifier in common emitter with the stage gain determined primarily by the transistor h_{fe} and the collector resistor, R_C. The non inverting input comes from the pin 3 of the Cannon connector via another 4u7 tantalum to the emitter of Q1 in effectively a grounded base configuration. As the input signal is balanced, the effective base-emitter input voltage of Q1 is the total of the differential voltage from pins 2 and 3 of the Cannon socket.

While the common mode noise rejection ratio (CMRR) is not as good as most op-amps, this ciruit provides 60 dB (1000:1 ratio) CMRR which is more than adequate for the application and without the additional noise of a second transistor used in a more conventional two-transistor emitter-coupled differential amp design.

For those who are new to balanced line techniques, the principle in simple terms is that a professional microphone provides either a floating output from the insert or a balanced output with respect to ground with the phase of the microphone output voltage being 180 degrees between pins 2-3 of the Cannon socket. Provided that this phase relationship exists, then the amplifier will provide an output.

Should the microphone or cable have any noise induced (ie, from dimmers or nearby mains wiring) the amplitude of noise will be the same, and in the same phase, on both legs of the cable. If a differential amplifier or transformer is used at the other end, the common mode noise (interference) will algebraically cancel.

This technique is frequently used by commercial broadcasters to provide remote control and telemetry to and from transmitter sites without additional landlines. It is also frequently used during outside broadcasts to provide talkback, program back and cueing facilities.

E L E C T R O N I C S
E T I - 1 4 3 2

SPECIFICATIONS, as measured on prototype

Frequency response +/- 3 dB	**1 Hz — 125 kHz**
Maximum output level	**+14 dBm (bridging)**
Normal output level	**+8 dBm (bridging)**
Headroom	**6 dB**
Distortion (THD) @ +14 dBm	**0.015%**
Distortion (THD) @ -10 dBm	**0.00125%**
Distortion (TIM)	**unmeasurable**
Signal/noise ratio wrt +8 dBm	**-118 dBm**

TABLE 1. SAMPLE GAIN SPREADS OF TESTED TRANSISTORS.

	GAIN									
	< 400	400-450	450-500	500-550	550-600	600-650	650-700	700-750	750-900	>
BC549C	53	65	283	260	100	65	40	25	9	0
BC109C	0	3	11	285	220	221	142	85	16	17

What happens is this: a common mode signal is transmitted down the line between the single pair cable and the return signal formed by the ground return at both ends. This is commonly called a Hybrid or Kylo circuit as depicted in Figure 1. It is also possible to send dc levels down the line as well as the ac Kylo signals or to split the audio spectrum up for different purposes. One AM broadcast station used this technique for some months in place of two landlines when it first converted to stereo.

An interesting hint for those in the PA business: a number of my colleagues have in the past used figure eight (8) lighting flex with balancing transformers at both ends of the line in place of very long screened microphone leads, with considerable success.

The dc bias for the preamplifier is provided by a self-biasing 2M2 resistor between base and collector. This was selected to set the transistor's operating point to centre the collector voltage at approximately 12 V.

The BC459 transistors were chosen because of three main factors:

1) high gain, typically greater than 500,
2) low noise,
3) their h_{fe} provides pretty well an optimal input terminating impedance for a wide range of microphones, giving optimum power transfer.

Phantom-fed balanced mic preamp

This circuit, Circuit B, is identical in operation to Circuit A, with the exception of the addition of two 22k resistors going to the +24 V rail, and the reversal of the tantalum input capacitors.

Modern condenser (capacitor) microphones and some broadcast quality electret microphones are all externally powered by a nominal 24V supply; some early microphones require +48 V. In this case dc is fed to the microphone via both leads, bypassed internally within the microphone to provide its operating voltage. This is called phantom powering.

Audio output is sent 180 degrees out of phase on the two leads from the microphone. The dc is blocked by the tantalum capacitors and the audio coupled in the same way as in Circuit A.

Unbalanced mic or GP amp

A general purpose gain stage which can be used for a myriad of uses is shown in Circuit C. Applications include Baxendall-type bass and treble controls, compressors, expanders, limiters, etc. The voltage gain is set by the ratio of R_C to R_E. In the example given, the voltage gain will be approximately 100 (being 10k/100). Should you, for example, wish to have a voltage gain of 80 times, the required emitter resistor would be given by the collector resistor R_C divided by the voltage gain (Av):

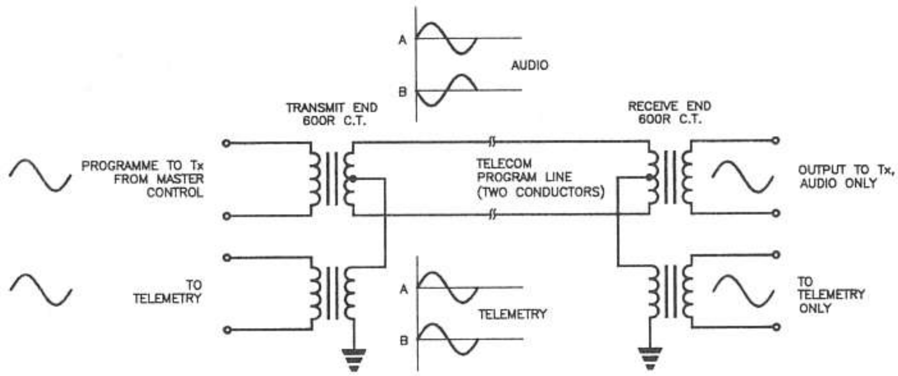


Figure 1: hybrid or Kylo circuit.

$$R_C/Av$$
$$\text{or } 10{,}000/80 = 125 \text{ ohms.}$$

Depending on the peak-to-peak output voltage required, the base bias resistor (R_B) may need some adjustment.

The input impedance of this stage is equal to approximately

$$h_{FE} \times R_E$$
or in the case of the example
$$500 \times 100 = 50\text{k ohms.}$$

It is worth noting that under most circumstances the value of the base bias resistor is usually larger than the transistor input impedance and as such may be ignored for the calculation.

Emitter follower

The emitter follower, shown in Circuit D, has many applications. This circuit has the same input impedance as Circuit C (calculated in the same way), and provides slightly less than unity voltage gain but substantial current gain as the output impedance is approximately $R_E/2$

The collector is bypassed to audio frequencies and the retention of the collector resistor limits the maximum peak-to-peak output voltage. For most applications in low level audio where the peak-to-peak voltages rarely exceed 2 Vp-p, the circuit will be quite useful. Should you wish to construct a conventional emitter follower the additional base resistor required between base and ground can be placed on the bottom of the pc board (the collector resistor can be bridged out with a link).

Simple headphone amplifier

This circuit operates in the same manner as Circuit C with the exception of a change of transistor type, no emitter resistor and a lower value of collector resistor. This stage has an output impedance of approximately 150 ohms and will produce a peak-to-peak voltage across a single 8 hm headphone of 1.2 V. This corresponds to a power level of 2 mW which will provide a 3 dB headroom above the maximum power rating of most headphones.

As the stage runs in pure class "A" it represents a good quality headphone amplifier for the digital era and is suitable for use with CD players, DAT recorders and as a studio or recording headphone monitor amp.

For home recording with the new generation of multitrack recorders available, a simple multiple output mixdown/foldback system could be constructed to enable each artist to receive a separate mix in their headphones.

Assembly hints

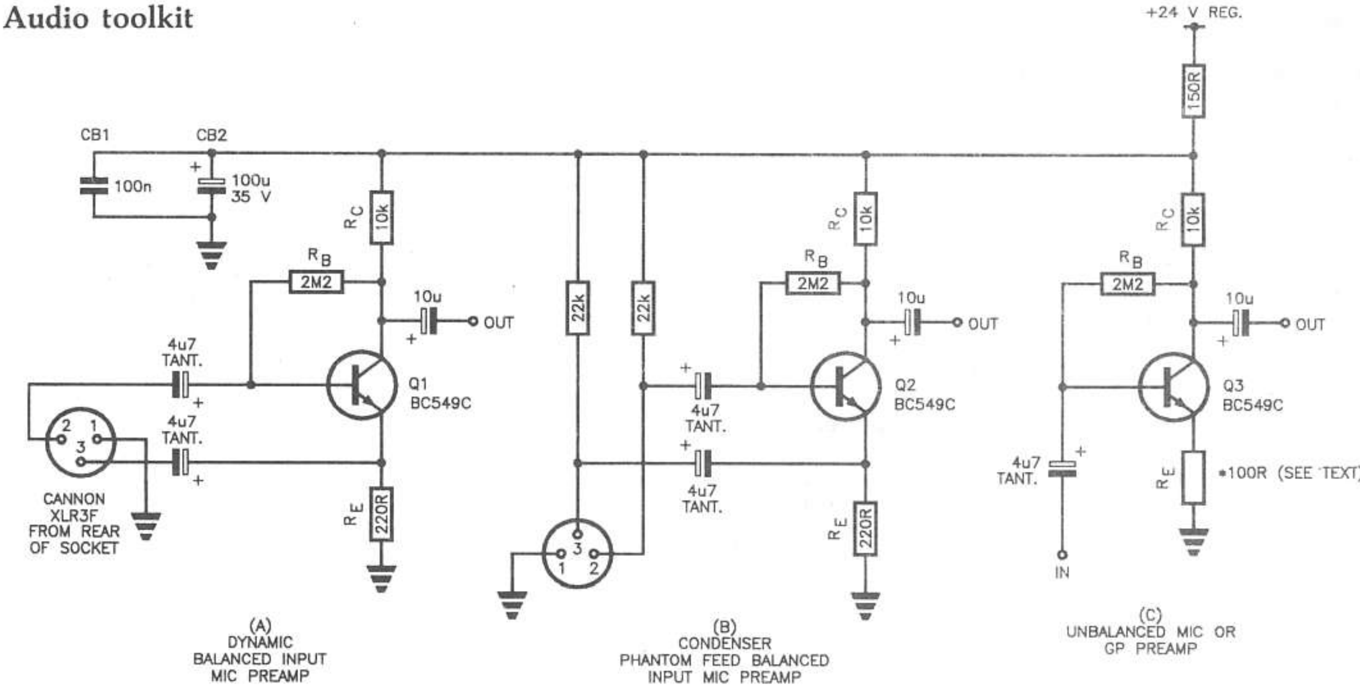
Before any modules are built, it is worth taking special note of the power supply. I have specified that this be derived from an external 24 V regulated source. This is necessary for noise-free operation of all low level stages as any ripple or noise on the supply would be readily amplified.

Two bypass capacitors are installed across the supply on the pc board, CB1 and CB2. CB1, a metallised polyester type, is used to effectively bypass any high frequency noise while CB2, an electrolytic, bypasses low frequency noise and prevents mutual coupling between stages which is the cause of motorboating. If the length of the leads from the power supply of the pc board exceeds 200 mm, the inductance of the cable becomes a factor to be considered, and the use of both capacitors is strongly recommended.

Because the basic board design is open, with well spaced component layout, construction is quite straightforward. Whether you make your own printed circuit board or purchase one ready-made, it is always good practice to check it thoroughly first. All the holes should be drilled out and of the right diameter to accept the component leads. See that no tracks are over-etched and possibly broken.

When mounting components, you can solder them in place in any order; just take care with the orientation of polarised components, such as the tantalum and electrolytic capacitors.

Orientation of the transistors should be quite clear. However you should check their pinout to see that the emitter base and collector leads coincide with the board layout. A quick foray with your multimeter in the ohms range or "diode check" will confirm which are the emitter and base leads, the remaining lead being the collector, of course.



Circuit A. Balanced input dynamic mic preamp.
Circuit B. Phantom feed balanced input condenser mic preamp.
Circuit C. Unbalanced mic or general purpose preamp.

In general, components should be mounted flush to the surface of the printed circuit board to minimise lead length. This provides mechanical stability and reduces stray coupling between circuits and possible pickup of R_F or extraneous hum or noise. Where an emitter resistor is not called for, as in Circuit E, simply bridge the two pc board holes with a short link of tinned copper wire. Just as you checked your pc board before assembly, it is always wise to check it afterwards, too. Check that components are in the right place and that polarised components are correctly orientated.

When you're satisfied that all is hunky-dory, you can connect a power supply and try out your circuit or circuits with real signals!

Design of a low noise preamp

Before this project was taken on, years of experience were taken into account for the optimisation of the design of the balanced line mic preamp. It may make interesting reading for those who build the projects to have a little more insight into the design stages that were undertaken and some of the unique methods that can be applied to design broadcast standard audio equipment.

Low noise mic preamps have long been an area for manufacturers of mixing desks to invest huge amounts of R&D. While the performance of a single channel may not pose a problem, as in the case of a simple hi-fi preamp, by the time you add the combined output of 16-32 channels the resultant noise figures can bring tears to an engineer's eyes.

It is because of this this that, generally, two areas are used as a compromise in designs. These are headroom and signal-to-noise ratio. If you have good noise figures, invariably the headroom is poor and vice versa.

Passive components. All passive components, particularly resistors, will generate noise so to optimise for minimum noise the highest quality resistors must be used, not necessarily for tolerance but for minimum generation of thermal noise. It has been found that Beyschlag 0.5% tolerance metal film resistors, or similar, offer advantages over cracked

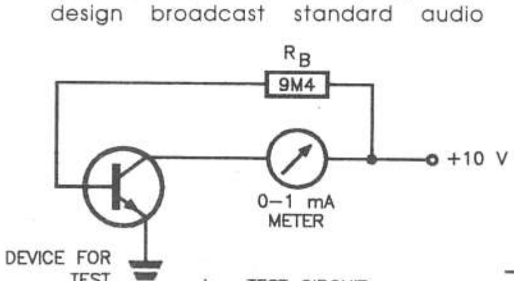
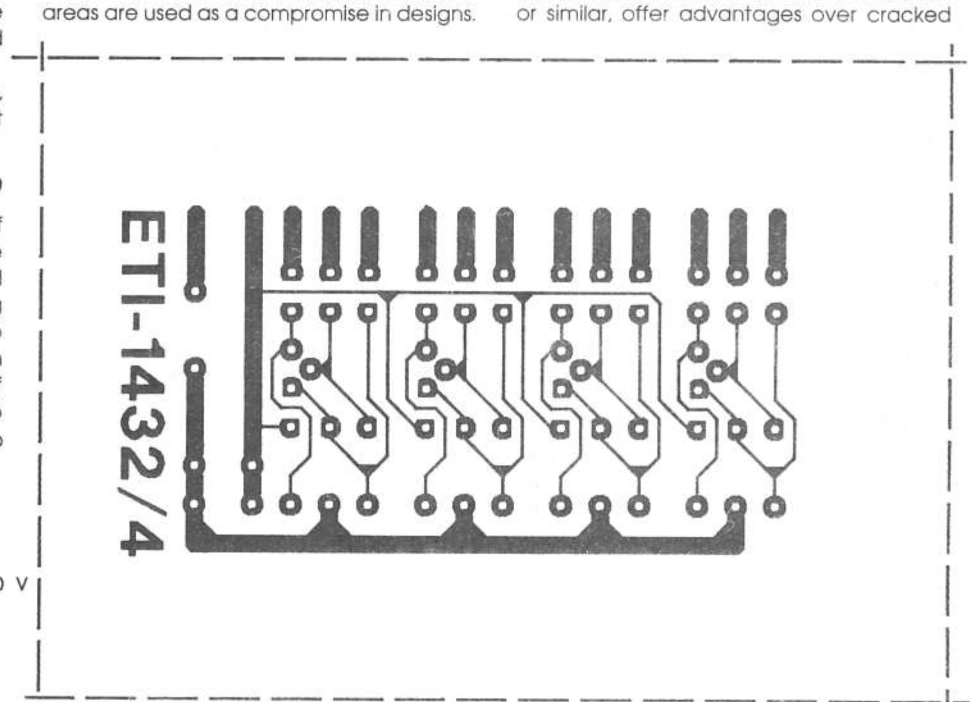


Figure 2: circuit to check h_{FE} of transistors.



Four-circuit printed circuit board (ETI-1432/4), full size.

carbon or composition resistors, with wire wound devices being a definite no-no because of their inductance and construction.

Capacitors are, again, another story, with electrolytics to be avoided because of their inherently high leakage and increasing impedance at higher audio frequencies. It is commonplace to find where electrolytics have been used in broadcast applications they are used with polycarbonate and/or disk ceramics connected in parallel to ensure a good impedance vs frequency performance.

Where large capacitances are required, for example interstage coupling, tantalum capacitors may be used, but these devices tend to break down and become noisy if for any reason they are reverse polarised. If the input impedance of the amplifier can be raised, then the use of high quality polycarbonate capacitors may be used as the capacitance values required are lower. This has been the choice of many designers today with the availability of BiFET low noise op-amps allowing the use of very high impedance circuits.

This may be very well, but the designer then must be careful of stray capacitances causing unwanted stability or bandwidth problems.

IC sockets should not be used as the least amount of contact resistance or dissimilar metals can cause rectification or noisy joints. The best solution is to directly solder devices to a good quality epoxy glass printed cirucit board, preferably double sided with the topside acting as a ground plane, and being constructed from 2 oz. or 4 oz. copper laminate, gold plated. (You also need an understanding bank manager!)

The use of laminated pc board with more copper can affect the noise performance of the design, especially at elevated temperature, and the thicker copper tracking will result in a lower resistance per square centimetre of track. Connectors of any sort are to be avoided, regardless of construction, for the same reasons listed above; the best solution is to solder everything.

Input transformers. Most high quality preamps use an input transformer to match the impedance of the microphone to that of the amplifier to ensure correct termination of an inductive source, for a flat frequency response, to provide a balanced input to the amplifier for good common mode noise rejection, and to provide an amount of voltage gain to improve the signal-to-noise ratio of the device.

This is a good design practice; however a good transformer will cost between $18 and $200 depending on the quality, and regardless of the expensive mu-metal shield incorporated, it will still have stray magnetic and electrostatic fields induced into its output. This is apart from the mounting and weight problems. It is also worth noting that all low noise preamps of any note have been frequently enclosed within mu-metal cases.

Earthing. The simplest and best designs can often be degraded by lack of care in design of the grounding systems employed for both modules and ancillary hardware.

Single point earthing is recommended for constructors with the common point being taken to ground separately with a short length of wire. In many systems, isolating the ground point then experimenting with a short jumper lead to find the best ground point can make a difference of up to 12 dB in noise figures. Also make sure that all input and output sockets are insulated from ground, and beware of one of my pet hates – chassis-mount RCA sockets! (If you must use them, put a grommet in the chassis hole to insulate them by other means.)

Potentiometers. It has amazed me how reputable designers of hi-fi equipment still continue to use lousy quality carbon track pots in their designs when the broadcast industry has for years been using quality conductive plastic faders, or even the antiquated but functional stud-type stepped faders.

Apart from the noise generated by the carbon pots, the mistracking between pots on a concentric shaft (stereo controls) is completely unacceptable for today's digital standards of 100 dB average head room. Surely the costs in quantity would be minimum.

On testing a batch of pots on my personal computer (I have written a simple program to display graphically the tracking of a pot's rotational position against resistance using the games port), of 100 pots only seven were to be found to be within 2% of each other, with the remainder all over the place. The only consolation was that the resistance from end to end was within 5%.

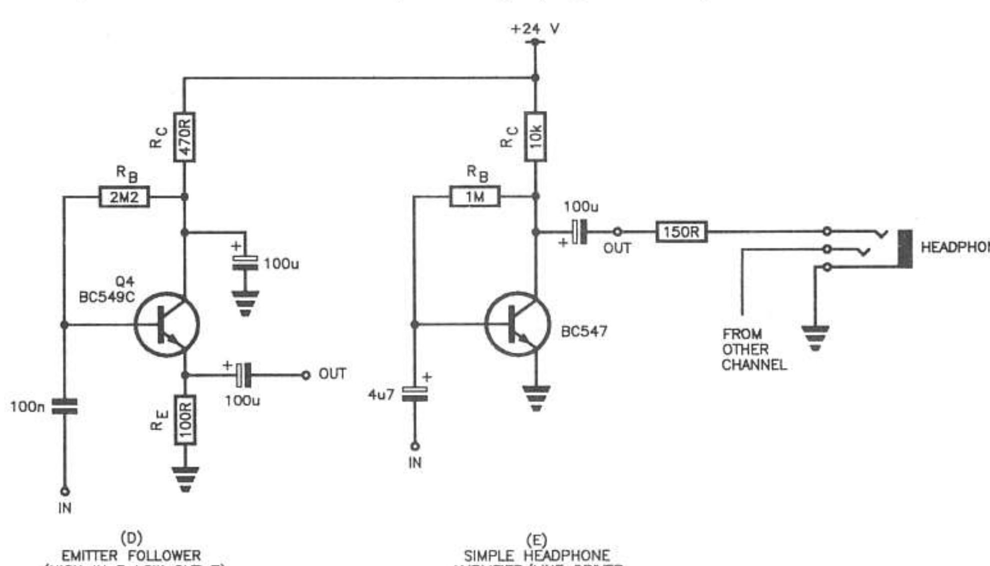
One could argue that, as our ears are logarithmic, who cares about trackability better than 10% as the difference is less than 1 dB in audio terms (1.41:1 ratio = 3 dB).

The problem is that, with the advent of digital equipment, the music headroom of 100 dB, plus the ambient noise level of 45 dBa, results in 145 dB range plus the scaling factor determined by the amplifier gain, all in 270 degrees of rotation of a gain control. While this may have been entirely suitable back in the days when vinyl disks reigned supreme, with CD and DAT now present this requires some rethinking.

Power supplies. The power supply rail (or rails) must be well regulated and filtered and if three-terminal regulators are used, they must be free from HF noise. The use of split supplies offers an additional advantage in that if any mains spikes do get their way through, then generally the spikes are equal and opposite in amplitude thus resulting in little or no change to the biasing of active devices and hence fewer clicks and plops in the output.

It is also vital that the mains transformers be well shielded to prevent stray electromagnetic radiation, toroids and "C" cores with copper shorting straps are recommended. If possible, mount the power supply in a different case away from the preamplifier stages. One of the best designs I ever tested used NiCads, which were charged when the unit was switched off.

Active devices. Op-amps are commonly used in preamp designs but suffer from two problems – noise and transient intermodulation distortion (TIM). With these designs it can be argued that you can't have both. Owing to the number of devices in the op-amp the rise time and transition time (propagation delay) through the devices



Circuit D. Emitter follower (high input impedance, low output impedance).
Circuit E. Simple headphone amplifier or line driver.

causes a delay before any feedback can be made effective. This results in the now well known TIM distortion.

For this same reason, the more devices in a circuit under feedback, the greater the noise introduced by each device and the greater the TIM. Among other factors, valve amplifiers usually have less than 12 dB negative feedback (due to limited open loop gain) and invariably had good TIM distortion figures. One point to remember – the noise figures will be degraded by every additional device put in circuit so the old rule of "keep it simple" applies.

Transistors have been around for years, and good ones are readily available at reasonable prices. The device selected for this series is the BC549, which offers good noise figures with high gain and excellent gain-bandwidth product figures. The noise figures do not vary greatly from device to device but the h_{FE} of the individual devices does.

In the units built for prototyping, 2000 transistors were purchased at an average cost of 6 cents each, then their gains tested with 1 mA of collector current. A typical breakdown of gains is listed in Table 1.

The first row shows BC549C devices (plastic pack), the second row BC109C (Philips metal can package). Of the 100 BC109s tested one had a gain of 1350 and, yes, it was functional.

To test devices a simple test circuit, as in Figure 2, can be constructed. The 9M4 base resistor can be made by connecting an 8M2 and a 1M2 resistor in series. The theory is that if a constant base current of 1 uA is supplied, a 0-1 mA meter in the collector circuit will directly read h_{FE} times 1000. If the h_{FE} is 1000, then exactly 1 mA of collector current will flow. If the h_{FE} is 500, then 500 uA (half a milliamp) will flow.

As mentioned earlier, the noise figures remain substantially the same between transistors, but the h_{FE} varies within a batch and in our prototypes we used the devices with gains of 750 or more.

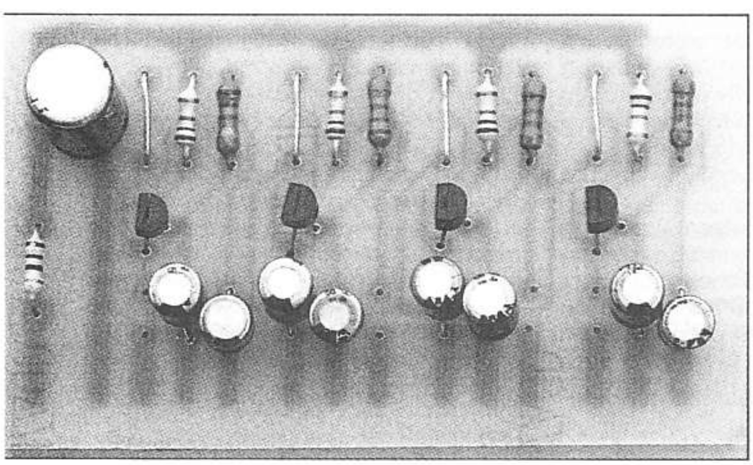
To design the preamp, the collector current must be noise optimised and from Figures 3 and 4 it can be seen that the noise figure changes with the collector current and source impedance. Here we are attempting to directly match a 500 ohm source impedance microphone with h_{ie}, the input admittance of the transistor (effectively the base-emitter input impedance), to get the maximum power transfer and to ensure correct termination of the microphone source impedance.

It will be seen that the noise performance is best in Figure 4 (at 10 kHz) and the Q point (quiescent, or dc, operating point) chosen is with a 600 ohm source. The collector current noise is optimised at 1 mA.

Figure 5 shows that at 1 mA collector current, the f_T (cutoff frequency of the device) is 125 MHz; in this case at least 1000 times the highest frequency required (100 kHz). Figure 6 shows that the h_{FE} at 1 mA should be typically 500 for an untested device.

The devices selected had an actual h_{FE} of nearly double this. It can also be shown from Figure 4 that the noise figure for these operations should be about 1.75 dB, which translates to approximately -120 dBm with an input voltage of 10 mV.

When you consider that an $18,000 NEVE mixing desk only quotes figures in the mid 90s, the performance obtained sets new levels and is suitable as a reference preamplifier. Once the collector current has been selected the rest is Ohm's law. **eti**



View of the printed circuit board of one prototype. This is a four channel headphone preamp.

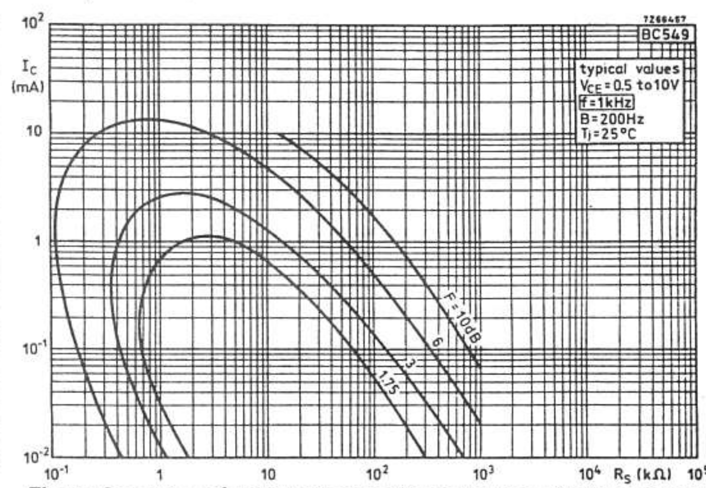


Figure 3: curves of constant noise figure for the BC549 at 1 kHz.

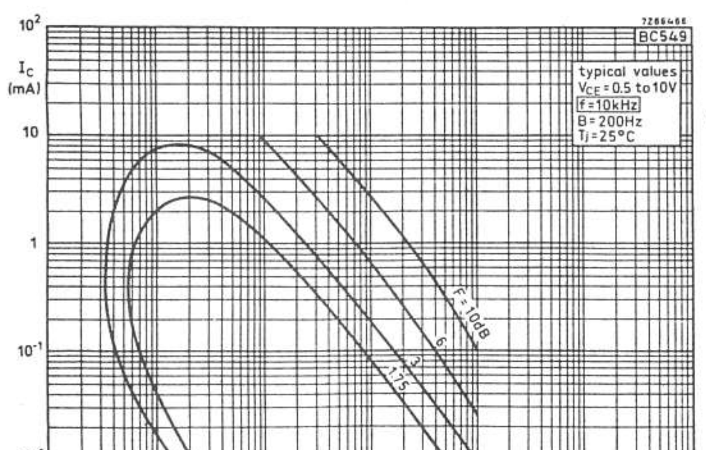


Figure 4: curves of constant noise figure for the BC549 at 10 kHz.

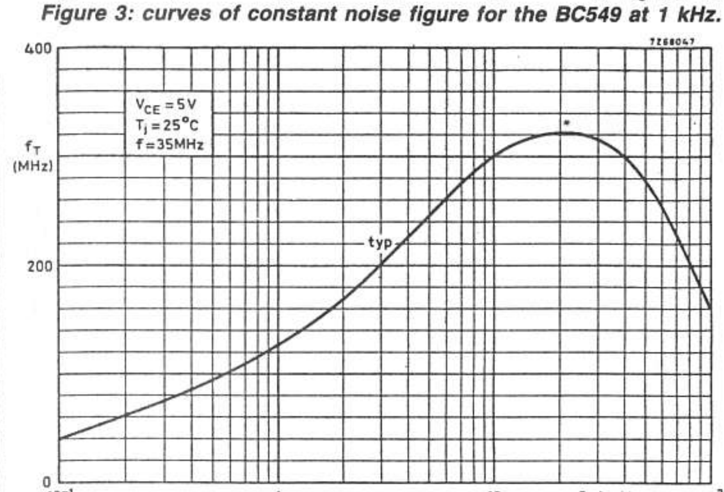


Figure 5: how f_T (cutoff frequency) of the BC549 varies with collector current.

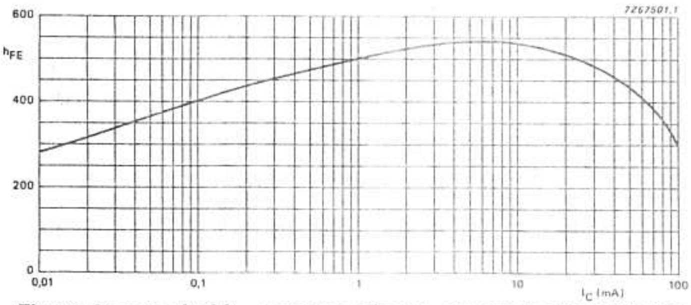
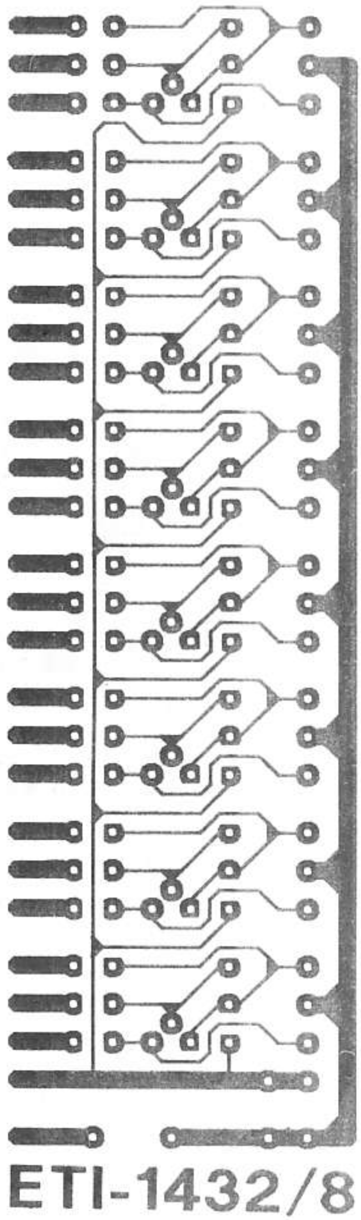
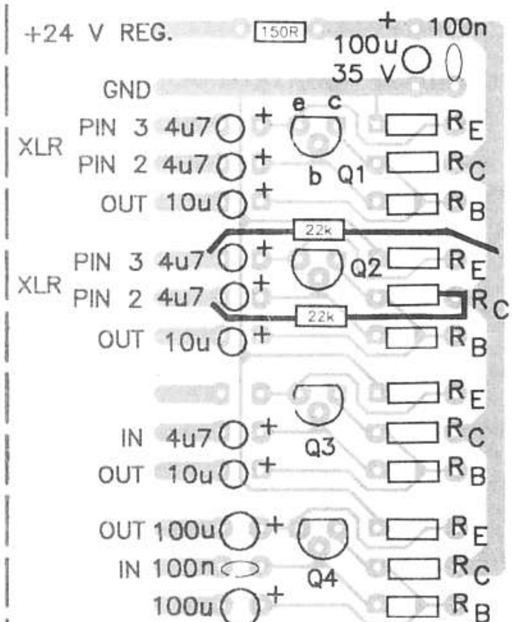


Figure 6: spread of h_{FE} versus collector current for the BC549C.



Eight-circuit printed circuit board (ETI-1432/8), full size.



PARTS LISTS — ETI-1432

CIRCUIT A
1 × BC549C or BC109C
1 × 2M2, 1/2 W, 5% (see text)
1 × 10k, 1/2 W, 5% (see text)
1 × 220R, 1/2 W, 5% (see text)
1 × 4u7/16 V tantalums
1 × 10u/25 V tantalum

CIRCUIT B
1 × BC549C or BC109C
2 × 22k, 1/2 W, 5% (see text)
1 × 2M2, 1/2 W, 5% (see text)
1 × 10k, 1/2 W, 5% (see text)
1 × 220R, 1/2 W, 5% (see text)
2 × 4u7/16 V tantalums
1 × 10u/25 V tantalum

CIRCUIT C
1 × BC549C or BC109C
1 × 2M2, 1/2 W, 5% (see text)
1 × 10k, 1/2 W, 5% (see text)
1 × 4u7/16 V tantalum
1 × 10u/25 V tantalum
1 × 100R selected for gain required (see text)

CIRCUIT D
1 × BC549C or BC109C
1 × 2M2, 1/2 W, 5% (see text)
1 × 470R, 1/2 W, 5% (see text)
1 × 100R, 1/2 W, 5% (see text)
1 × 100n/100 V polyester
2 × 100u/25 V electrolytics

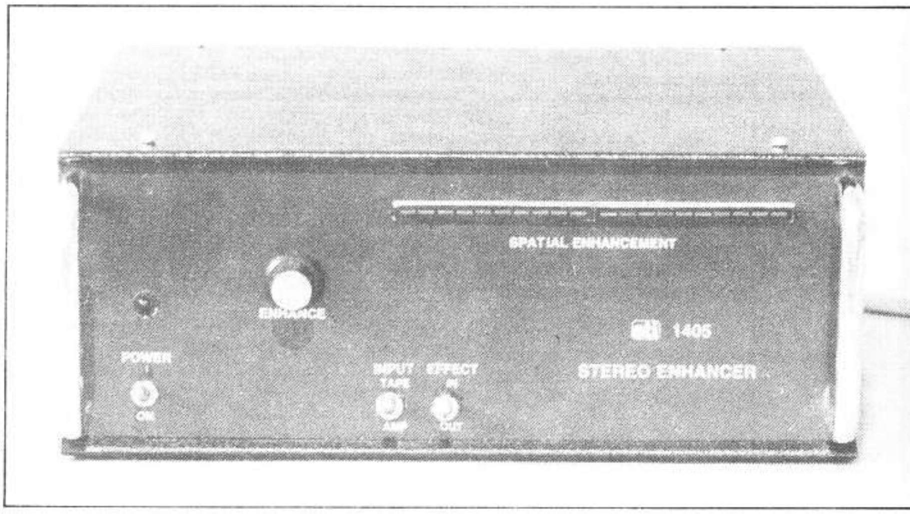
CIRCUIT E
1 × BC547
1 × 1M, 1/2 W, 5% (see text)
1 × 10k, 1/2 W, 5% (see text)
1 × 150R, 1/2 W, 5%
1 × 4u7/16 V tantalum
1 × 100u/25 V electrolytic

MISCELLANEOUS
1 × 100n/100 V polyester
1 × 100u/35 V electrolytic
1 × 150R, 1/2 W, 5%
ETI-1432/4 or ETI-1432/8 pc board as required; input connectors as required; output connectors as required; tinned copper wire as required.

STEREO ENHANCER

The best thing about stereo is that it sounds good! The greatest stereo hi-fi system loses its magnificence if the effect is so narrow you can't hear it. This project lets you cheat on being cheated and creates an 'enhanced stereo effect' with a small unit which attaches to your amp.

Robert Irwin



WHEN IS A STEREO not a stereo? Answer: When you try to fit your 60 cubic foot, six way monitors into your new, one bedroom flatette and find that the only place you can put them is side by side where they double as a dining table. Not the ideal situation to get that wonderful stereo separation that causes steam trains to thunder in one side of your living room, roar across the coffee table and exit through the window on the opposite side.

The ETI-1405 is designed to 'widen' out the stereo image from your amp and allow you to maintain that stereo feel even when you have to put your speakers close together.

The unit is designed to plug into the cassette input and output on the back of a standard stereo amp and can be switched in and out of circuit by using the TAPE/SOURCE switch. An alternate set of cassette in/out terminals is provided on the back of the enhancer to plug your deck into.

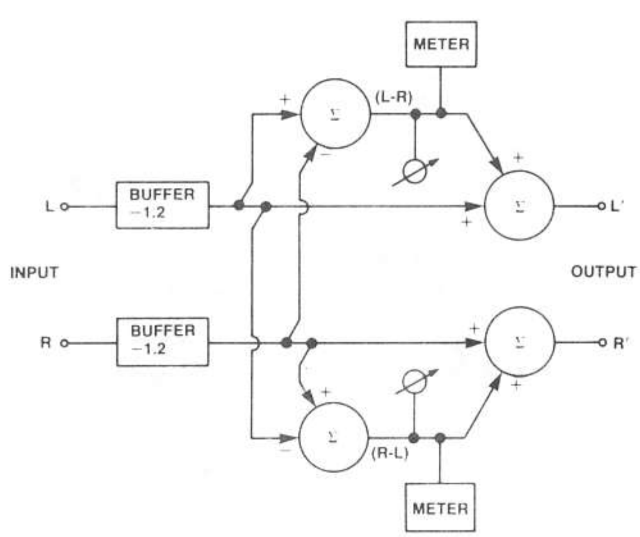


Figure 1. Block diagram of the enhancer circuit.

Design details

The idea behind the circuit is very simple and has been used in small portable stereo tape decks for quite a while. Figure 1 shows a block diagram of the circuit of the enhancer. The principle used is basically to obtain the difference between the left and right channel and then to subtract this from the original left and right channel signals. This is explained more fully in the "How it Works" section.

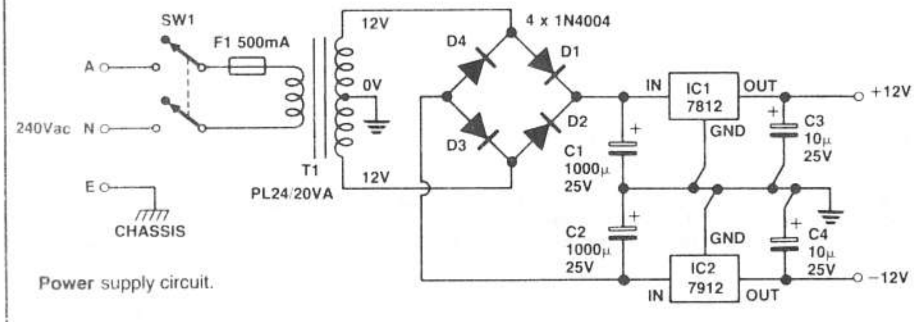
The end result is to obtain a 'super stereo' signal which has components for the left channel of $L+A(L-R)$ and for the right channel of $R+A(R-L)$ where R is the original right channel signal, L is the original left channel signal and A is a proportion between 0 and 1 which is set by the level control.

One unusual feature of the design came about in the metering provided. It was firstly thought that a measure of the signal levels at the output, along the lines of VU metering, should be provided.

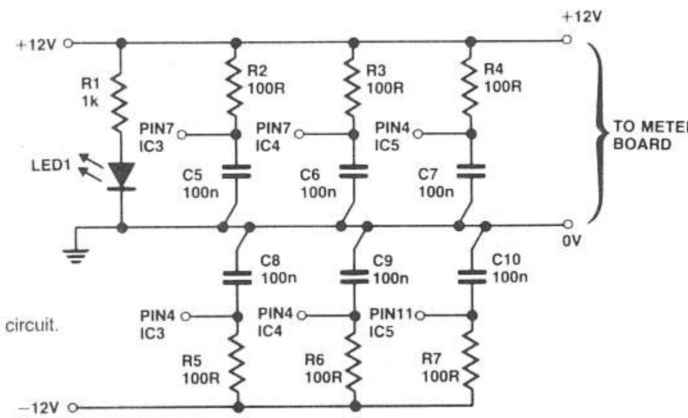
After a quick rethink though, it was decided that this wouldn't really be of much benefit to anyone as the signal levels here weren't really interesting. A much more relevant meter would be one that in some way indicated the amount of enhancement taking place. LED level meters were, therefore, placed at the output of the level control pot in the difference amp section of each channel. This gave a direct and dynamic representation of the enhancement. To add to this, the displays for each channel were mounted back to back giving a centre-zero, bar graph display which visually shows the widening of the stereo image.

Construction

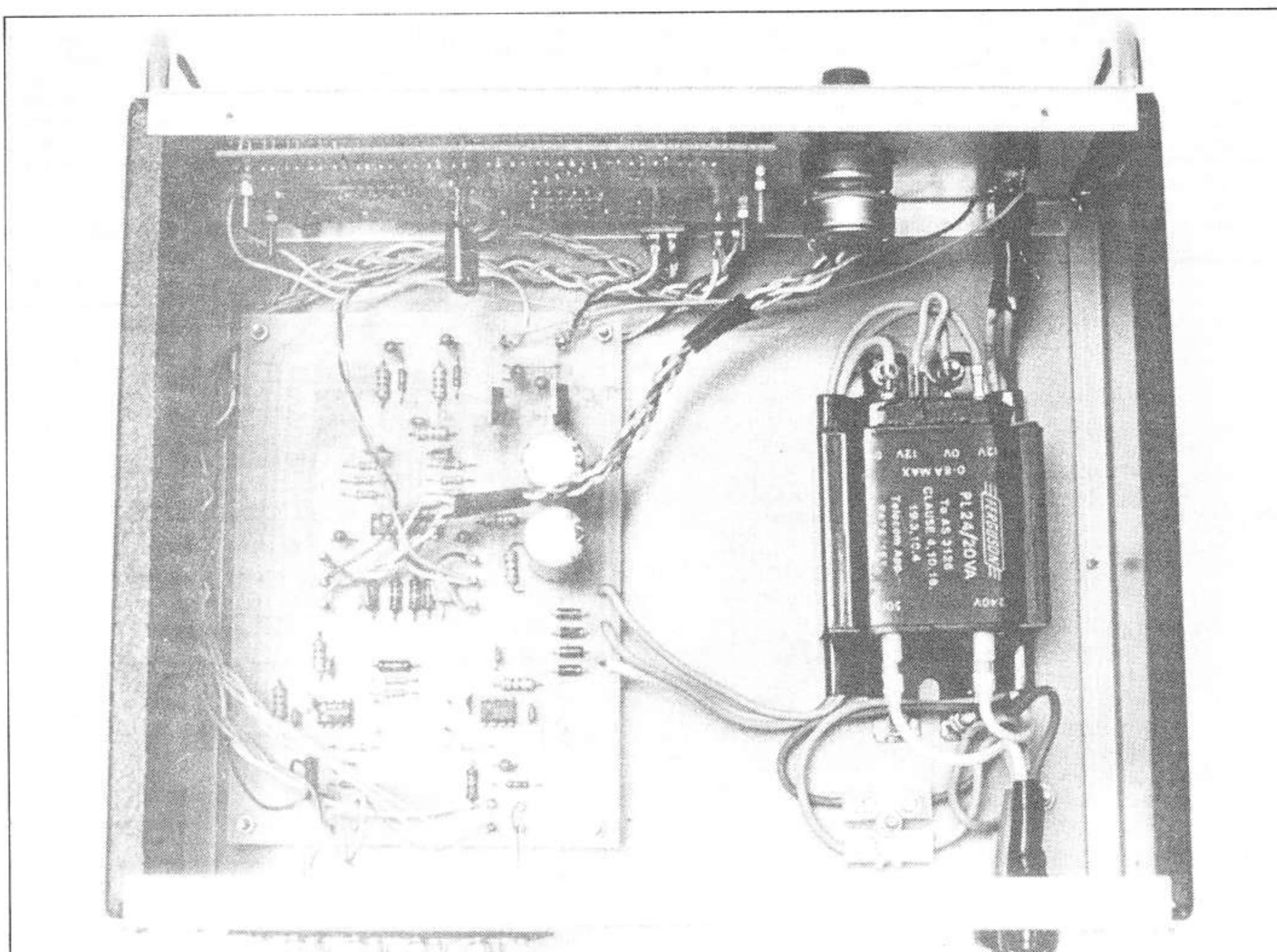
It is best to begin construction with the case. The prototype was mounted in a Horwood instrument case. If you are using an



Power supply circuit.



Noise decoupling circuit.



HOW IT WORKS — ETI-1405

The idea behind the enhancer is to generate two signals, R-L and L-R, where L is the left channel signal and R is the right channel signal. These signals are then mixed back with the appropriate original signals (R is mixed with R-L and L is mixed with L-R). The final composite signals are R+a(R-L) and L+a(L-R) respectively where a is the proportion of the difference signal. If a is zero then the output will be just the original left and right channel signals. When a is one, the outputs will be 2L-R and 2R-L. This creates a 'super stereo' image because signals which are the same in both channels (that is centered in the stereo field) will be left unchanged but signals which are panned to one side will be of a different amplitude in each channel and the enhancer will increase this difference and thus give a feeling of a wider stereo separation.

The circuitry of the enhancer is relatively straightforward. Referring to the circuit diagram, IC3 and IC4 are NE5534 low noise op-amps configured as inverting buffer stages with a gain of −1.2 set by the ratio R12/R10 and R13/R11. The inputs are ac coupled via C11 and C12. C13 and C14 provide unity gain compensation for the NE5534s. The difference signals are created by IC5a and IC5b which are configured as unity gain differential amplifiers. The output of IC5a is the L-R signal and the output of IC5b is the R-L signal. C15 and C16 prevent any high frequency instabilities.

The outputs from the diff amps are fed to a dual pot, RV1, which controls the amount of difference signal being fed to the final summing amplifier stages formed by IC5c and IC5d and associated resistors. This stage sums the original and difference signals. The summing amps are virtual earth, inverting summers with the gain of both inputs set to unity by the ratio of the feedback resistors R26 and R27 and the input resistors R22, R23, R24 and R25. Once again C19 and C20 ensure there are no high frequency instabilities. R28 and R29 supply some isolation from capacitive loading on the output and C21 and C22 ac couple the output.

The power supply circuitry is fairly standard with the 24 V centre tap secondary of the transformer being full wave rectified by D1, D2, D3 and D4. The output is smoothed by C1 and C2 and this is then fed to LM7812 and LM7912 12 V regulator ICs (IC1 and IC2). The output from the regulators provides accurate + and − 12 V supply rails. C3 and C4 ensure that the regulators remain stable. R2, 3, 4, 5, 6, 7 and C5, 6, 7, 8, 9, 10 provide decoupling for the op-amp supplies. R1 limits the current through the LED1, the power on indicator.

THE METERING CIRCUIT

The metering circuit is based around the LM3914 LED driver IC. Both the left and right meter are the same so I will only refer to the left channel.

The input to the meter is ac coupled via C24 and R32 and fed to the input pin (5) of the LM3914. D5 limits the reverse voltage swing to −0.6 volts and C26 slows down the response time of the meter. Pin 7 of the LM3914 is internally referenced to 1.2 V. This is divided down to about 0.2 V by R36 and R34. This voltage is then fed to pin 6 of the IC and sets the full scale of the meter to 0.2 V. If less sensitivity is required then R34 can be increased. The sum of R34 and R36 should be kept to less than about 1 K otherwise the brightness of the display may decrease.

PARTS LIST — ETI-1405

Resistors ....all ¼ W, 5%
- R1 .... 1k
- R2, 3, 4, 5, 6, 7, 28, 29, 34, 35 .... 100R
- R8, 9, 30, 31, 32, 33 .... 100k
- R10, 11, 14, 15, 16, 17, 18, 19, 20, 21, 22, 23, 24, 25, 26, 27 .... 10k
- R12, 13 .... 12k
- R36,37 .... 560R
- RV1 .... 100k lin. dual ganged

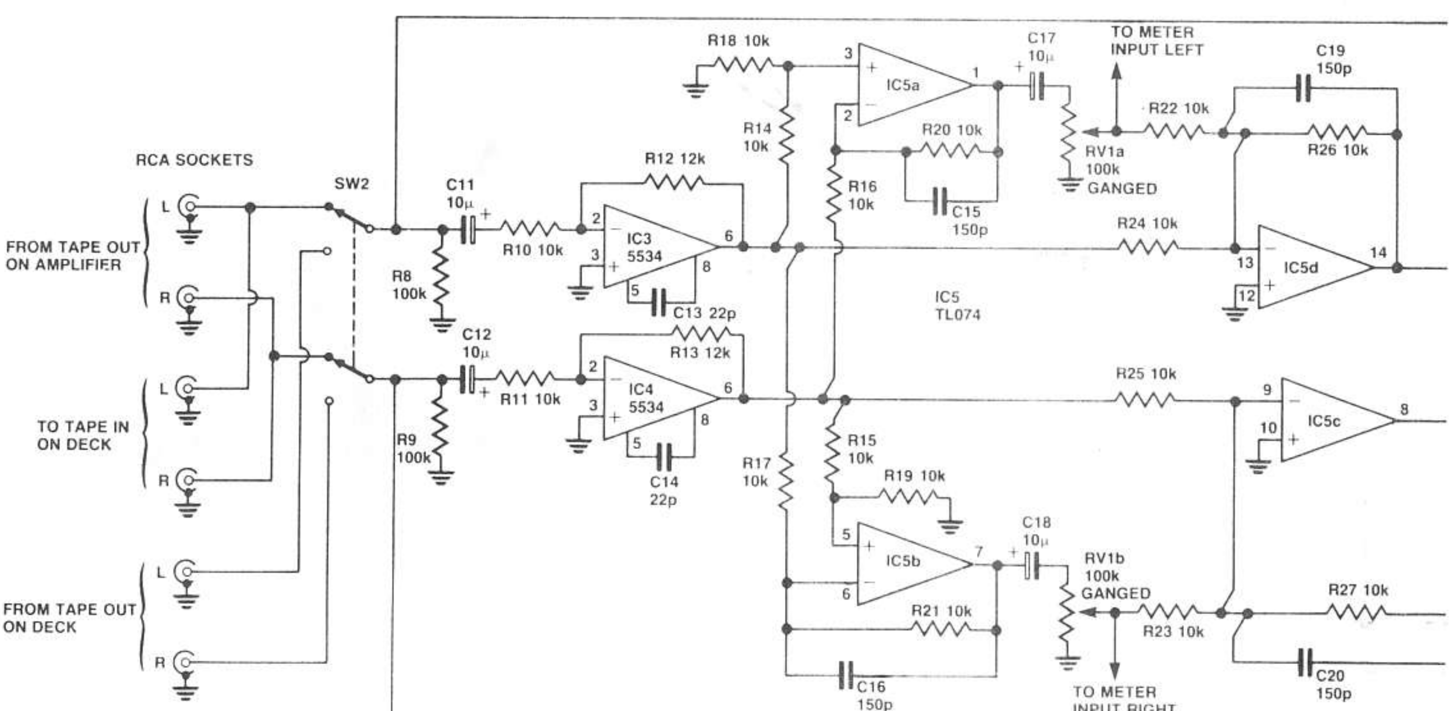
Capacitors
- C1, 2 .... 1000µ 25 V RB electro.
- C3, 4, 11, 12, 17, 18, 21, 22, 24, 25 .... 10µ 25 V electro.
- C5, 6, 7, 8, 9, 10 .... 100n ceramic bypass
- C13, 14 .... 22p ceramic
- C15, 16, 19, 20 ...... 150p ceramic
- C23, 28 .... 2µ2 25 V RB electro.
- C26, 27 .... 220n greencap

Semiconductors
- IC1 .... LM7812 pos. regulator
- IC2 .... LM7912 neg. regulator
- IC3, IC4 .... NE5534 op-amp
- IC5 .... TL074 quad op-amp
- IC6, IC7 .... LM3914 LED driver
- D1, 2, 3, 4 .... 1N4004 rectifier diodes
- D5, 6 .... 1N914 small signal diode
- LED1 .... 5mm red LED
- two 10 LED arrays

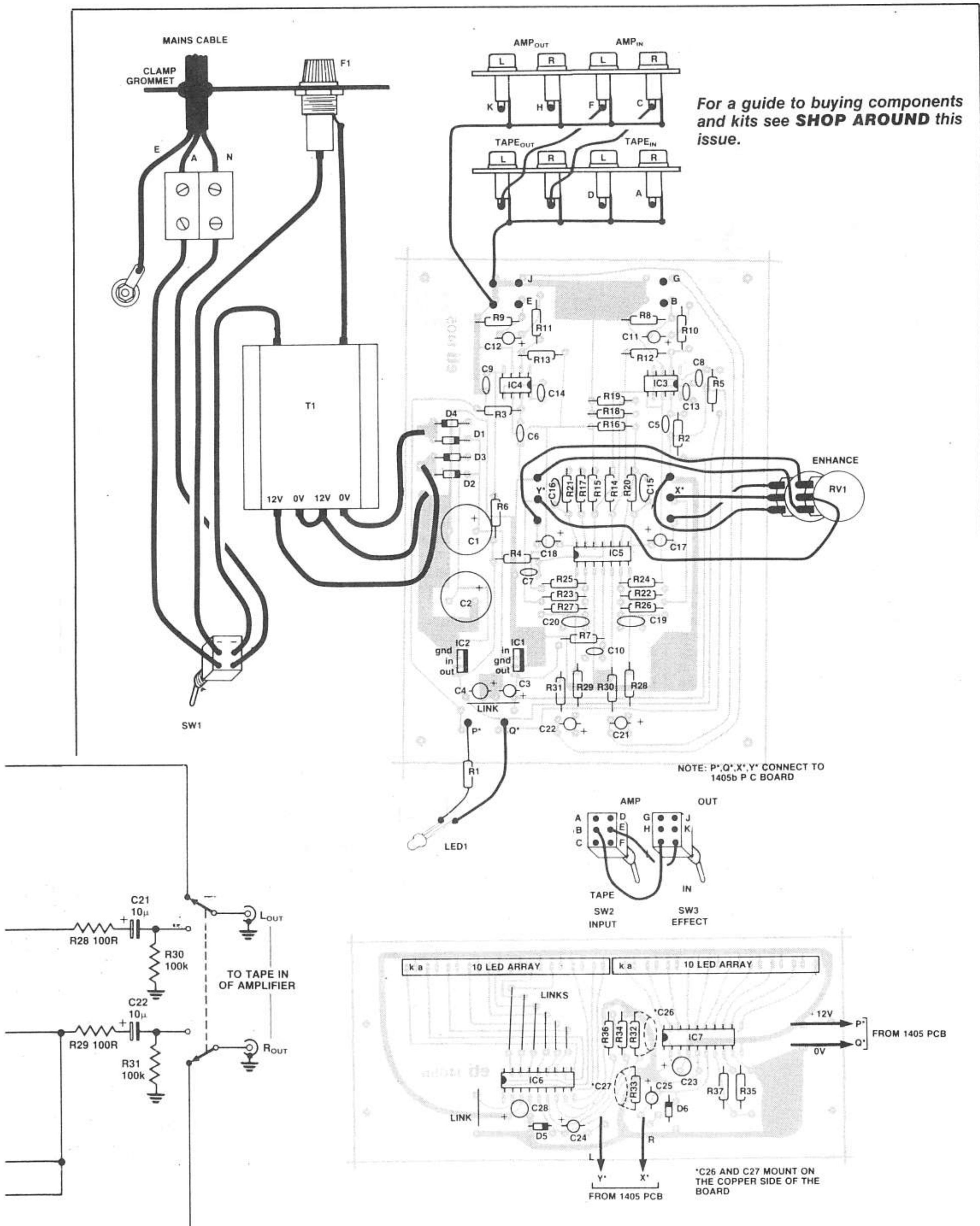
Miscellaneous
- SW1, 2, 3 .... DPDT toggle
- T1 .... Transformer Ferg. type PL24/20VA or similar

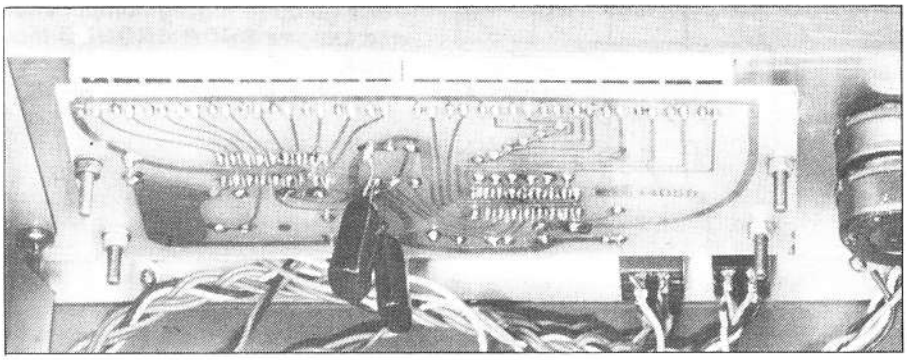
ETI-1405 and 1405b pc boards; Horwood case type 84/10/V; two four way RCA socket arrays; Scotchcal front and back labels; mains flex and plug; mains grommet and clamp; four way terminal block; 2AG fuse holder and 500 mA fuse; hookup wire; nuts and bolts; knob for rotary pot; four rubber feet.

Price estimate: $70-$80



For a guide to buying components and kits see ***SHOP AROUND*** *this issue.*





The back of the front! This shows the meter board mounting. Note the caps mounted on the rear of the board.

identical case then you can follow the mounting details exactly. If not, then you will have to make any changes appropriate to the case you are using.

The Horwood case comes assembled except that the top and bottom are only stuck on with masking tape. The first thing to do is to take the top and bottom off and find the little packet of self-tapping screws lurking inside.

The next step is to mark out the positions of the holes on the top and bottom plates so that these screws can hold the top and bottom plates on. The top and bottom should then be stuck back in position with masking tape and the small holes for the self-tappers drilled. Note that you should drill through the mounting lips on the sides of the case at the same time as you are drilling through the top and bottom so that the holes line up accurately. Don't use too big a drill otherwise the self-tapper will not have enough metal to grip.

Take the top off and then, with a marking pen, mark the top, bottom, front, back, left side, right side etc, so that you will have no trouble re-assembling the case the same way. If you don't you will have all sorts of problems getting all the holes to line up.

Before disassembling the box you can mark out the hole positions on the bottom plate. Take a close look at the picture of the inside of the prototype and lay out your transformer and pc board in the same relative positions. Looking from the front, the transformer is mounted on the left hand side of the box about half way back. Once this is located mark out the positions of the four mounting holes. Just behind the transformer mark a hole for the mains terminal block and earth lug.

The pc board can now be located on the right hand side of the box once again about half way back. Make sure you orientate the board so that the edge with the transformer input connections is adjacent to the transformer itself. Mark out the pc board mounting holes. The bottom can then be removed and drilled.

The front panel can be marked out next. Disassemble the rest of the case. The front panel can be marked out using the drilling diagram or the front panel artwork can be used as a template. Drill the front panel.

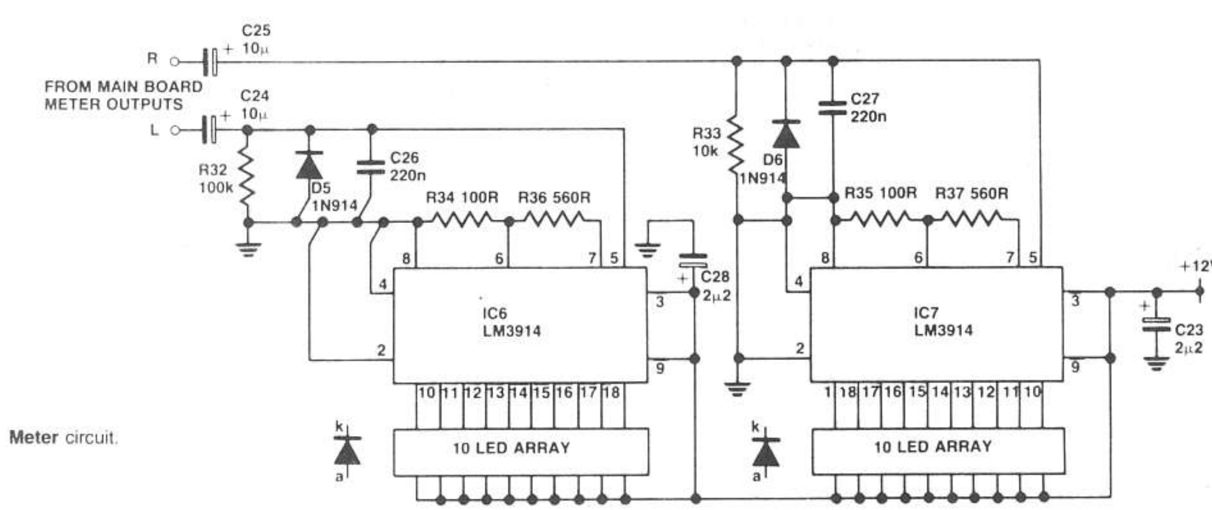
The slot for the LED arrays can be made using the old drill, nibble and file method (or fill, dribble and nile if you prefer!). Firstly, drill a series of holes along the centre line of the slot. The drill used should be slightly smaller than the width of the slot. Keep the holes as close together as you can so there is a minimal amount of metal left. A small nibbling tool (hacksaw blade, chisel, sidecutters or whatever!) can then be used to cut out the remaining metal between holes. The slot can then be carefully filed out to size with a small flat file.

Once the front panel is finished you can mark and drill the rear panel. The mains cord and fuse holder sit directly behind the transformer and the RCA sockets mount behind the pc board. To mount the rear panel RCA sockets you will have to position and drill the holes for the lugs and then file out the holes so that there is no metal to metal contact between the RCA sockets and the chassis.

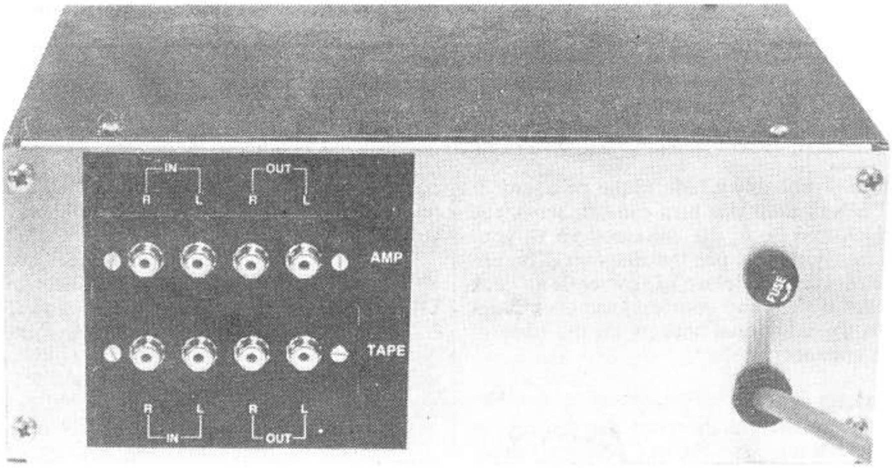
On the prototype I actually cut out a slot so that the lugs would have good clearance. Once the back panel is finished you can put away your drill press and get out your soldering iron for the next step.

Begin the mounting of the components with the display pc board (ETI-1405b). First, check the pc board thoroughly for broken or shorted tracks. Locate and solder in the seven wire links. The resistors and capacitors can be mounted next followed by the diodes. It is advisable to use IC sockets for the LM3914s and these should be mounted next and the ICs put in. Make sure you get them round the right way.

The LED arrays should now be mounted. You may have a little trouble getting them in at first but just persevere. Once you get them in the holes then push them down until they are standing about 5 mm above the pc board. Make sure they are sitting straight and level and then solder them in.



Meter circuit.



View of the rear panel showing the two RCA arrays.

Take care that you get the displays the right way round as desoldering them can be a real pain in the enhancer.

The two capacitors which mount on the copper side of the board can now be soldered on. The only thing left to be done on this board is to solder about 100 mm lengths of hookup wire to the input and power supply pads.

Turn your attention now to the main pc board. This should be constructed in a similar way to the display board. Resistors and caps first followed by diodes and ICs. Once again take care to get all the polarized components the right way round, especially the big electrolytic caps on the power supply.

Do not use IC sockets with the ICs on this board as the stray resistances and capacitances can degrade the performance of the circuit. Solder pegs should be used on all the flying lead connections on this board to make the wiring up easier.

The next step in construction is to attach the Scotchcal artwork to the front panel. Firstly drill small pilot holes in the centres of all the holes and at the end points of the slot on the Scotchcal. Peel off the backing paper on the Scotchcal and thoroughly wet the back of the Scotchcal and the front panel (the water will allow you to easily peel off the label again if you don't line it up correctly the first time).

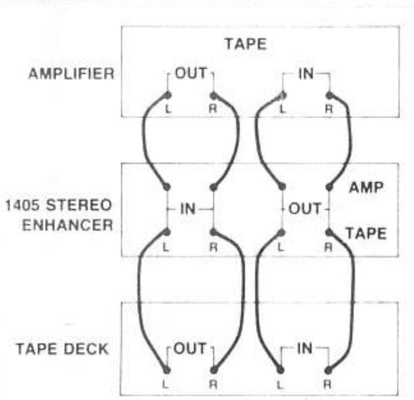


Figure 2. Connecting the stereo enhancer to your amp and tape deck.

Line up the holes on the Scotchcal with the holes drilled in the front panel and then press the Scotchcal firmly in place. Gently rub out the excess water with a soft cloth and set the front panel aside to thoroughly dry. The same treatment can be given to the Scotchcal back panel.

When both the panels are completely dry the holes can be carefully cut out and trimmed using a very sharp knife or scalpel. Try not to tear the Scotchcal or cut off a finger when doing this.

Before re-assembling the case you should attach the mounting bolts for the display board. Sit the display board on the back of the front panel so that the LED displays fit through their slot. Mark out the positions of the mounting holes. Four 20 mm long 6BA flat headed bolts should then be glued to the back of the front panel with Araldite or something similar. Allow the glue to completely harden. The case should now be reassembled with the exception of the lid.

Now comes the wiring up. Start with the mains wiring. Firstly, securely mount the transformer, terminal block and fuse holder. Mount a length of mains flex using either a clamp type mains grommet or a grommet and separate mains clamp.

Carefully follow the wiring diagram and wire up the mains switch, fuse holder and primary of the transformer with heavy duty hookup wire. Make sure that you insulate any exposed joins or terminals with heatshrink or insulating tape. REMEMBER: mains voltages are lethal so double and triple check your wiring and make sure that you can't accidentally touch any exposed terminals. The mains switch can now be securely mounted to the front panel.

Next mount the display board. The board should be mounted so that the LED arrays protrude about 2 mm through the front panel. The main board can be mounted next using 12 mm spacers.

Using the wiring diagram, wire the secondary of the transformer to the main board. The flying leads from the display board can also be attached to the main board at the appropriate points. The power indicator LED can then be mounted and wired up. The two RCA socket arrays should then be mounted on the rear panel. The input and output wiring, including the two switches, can then be done.

Finally, 250 mm lengths of hookup wire can be attached to the dual 'enhancement' pot and then attached to the pc board. The pot can then be mounted on the front panel and the knob attached.

Testing and setting up

The only real way of testing the enhancer is to turn it on so, with nothing attached to the input or output, plug it into the mains and switch on for a few seconds. The power LED should light and all the display LEDs should remain off. Other than that nothing should happen. If anything else happens (smoke, fire, brimstone, nasty hissing or popping noises, etc) switch off immediately and check everything.

If all is OK then switch on again and try rotating the enhancement pot. The display LEDs should stay off. If any come on then there is probably a fault in the pc board. If all is well then you turn can off, screw the lid on and hook the enhancer up to your stereo system as per the diagram. The enhancer takes the place of your cassette deck in the system and your cassette deck plugs into the additional sockets on the back of the enhancer.

Using it

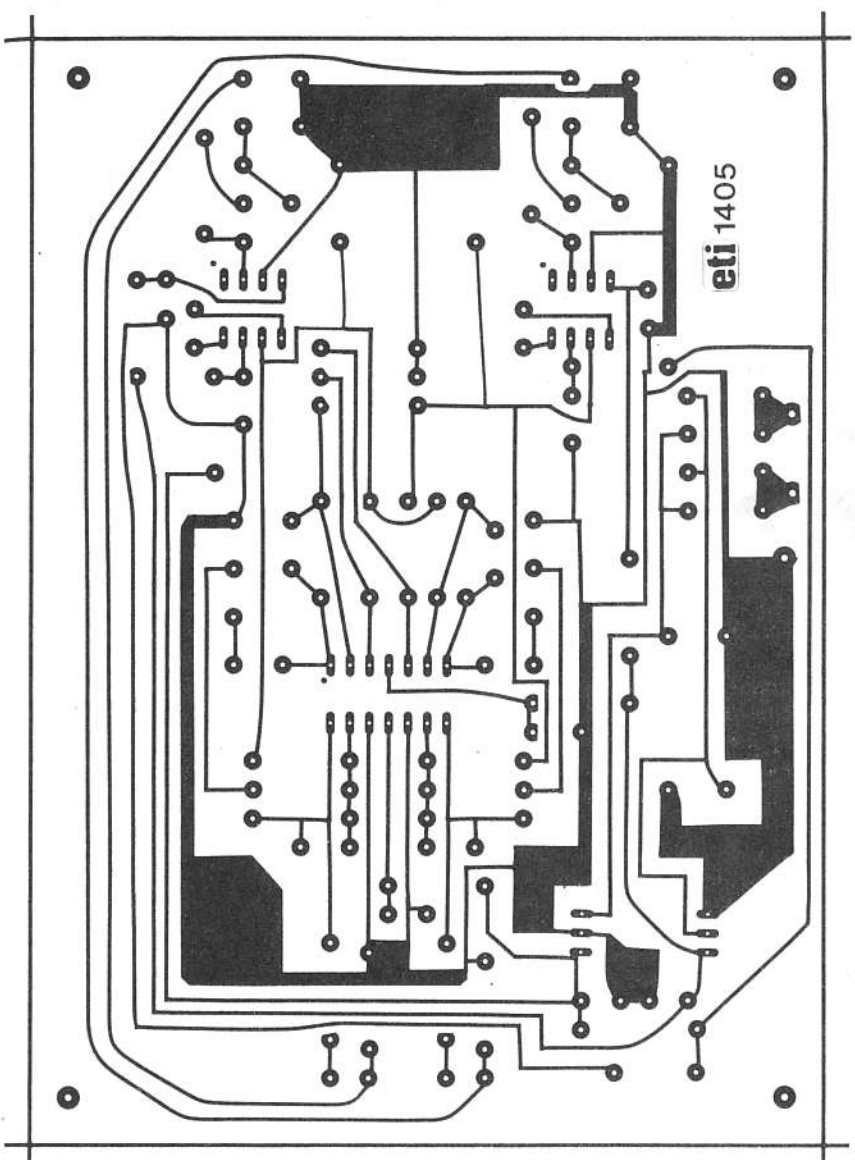
To use the enhancer set the controls of your stereo as follows. Select either PHONO or TUNER on your amplifier source switch. Set the effect switch on the enhancer to IN and the input switch to AMP.

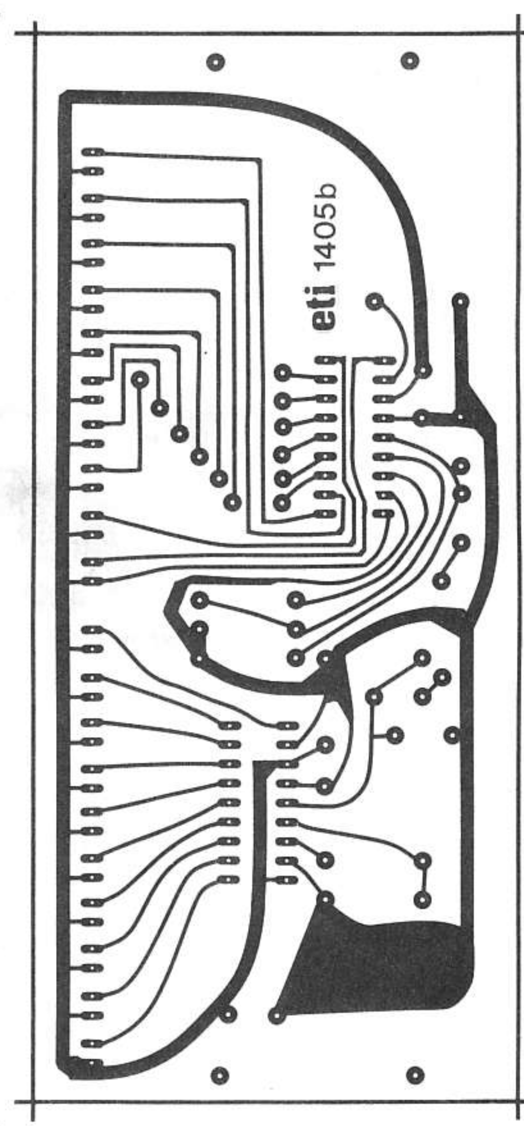
Now all you need do is select TAPE on the TAPE/SOURCE button on your amp to cut the effect in. With the TAPE/SOURCE button on SOURCE the effect will be bypassed. To play a cassette through the enhancer once again select TAPE on the amp and switch the input switch on the enhancer to TAPE.

To play a cassette without the effect just switch the effect switch on the enhancer to OUT. Don't worry, you'll get the hang of it!

Turning the enhancement control clockwise increases the effect and this will be echoed in the LED display. As the enhancement is increased the bargraph should get wider and wider from the centre outwards.

It should be noted that the amount of 'stereoness' is fairly subjective and some tracks may appear to be affected more than others. At first the effect may sound a little false but it is a bit like 3D pictures in that you must let your brain deceive your senses. When you get over trying to convince yourself that your speakers aren't far enough apart to get good stereo then the effect will be more convincing. ●



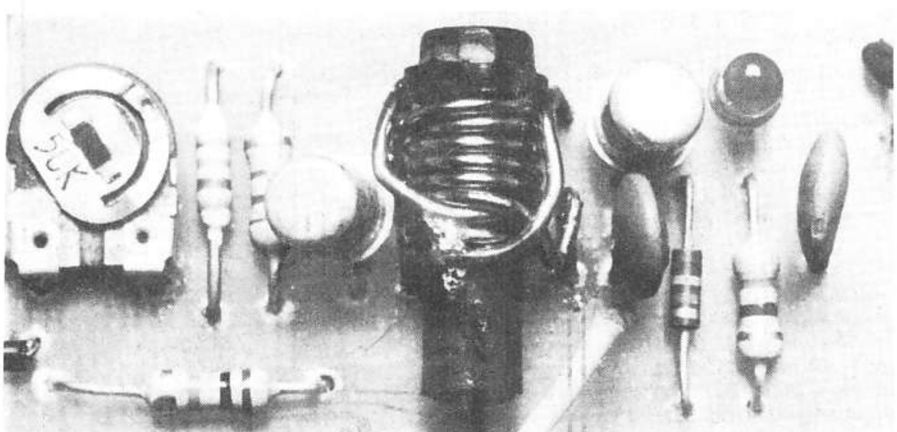


MINIATURE FM TRANSMITTER

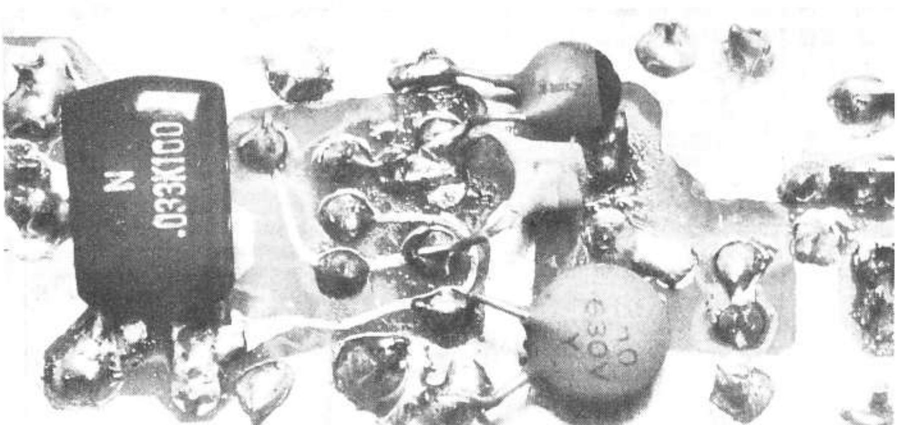
S.K. Hui

Jon Fairall

ETI continues its quest for smallness almost to vanishing point. The biggest part of this ultra tiny device is the battery. A fun project for Christmas.



Both sides of the board. Note the capacitors soldered to the bottom.



WE HAVE PUBLISHED several extremely compact projects recently. These have included the transmitter section of the optical car alarm switch (September '85) and the ETI-741 radio mic (January '85). For one reason or another both these projects were extremely complex and building them required considerable experience and a good deal of patience.

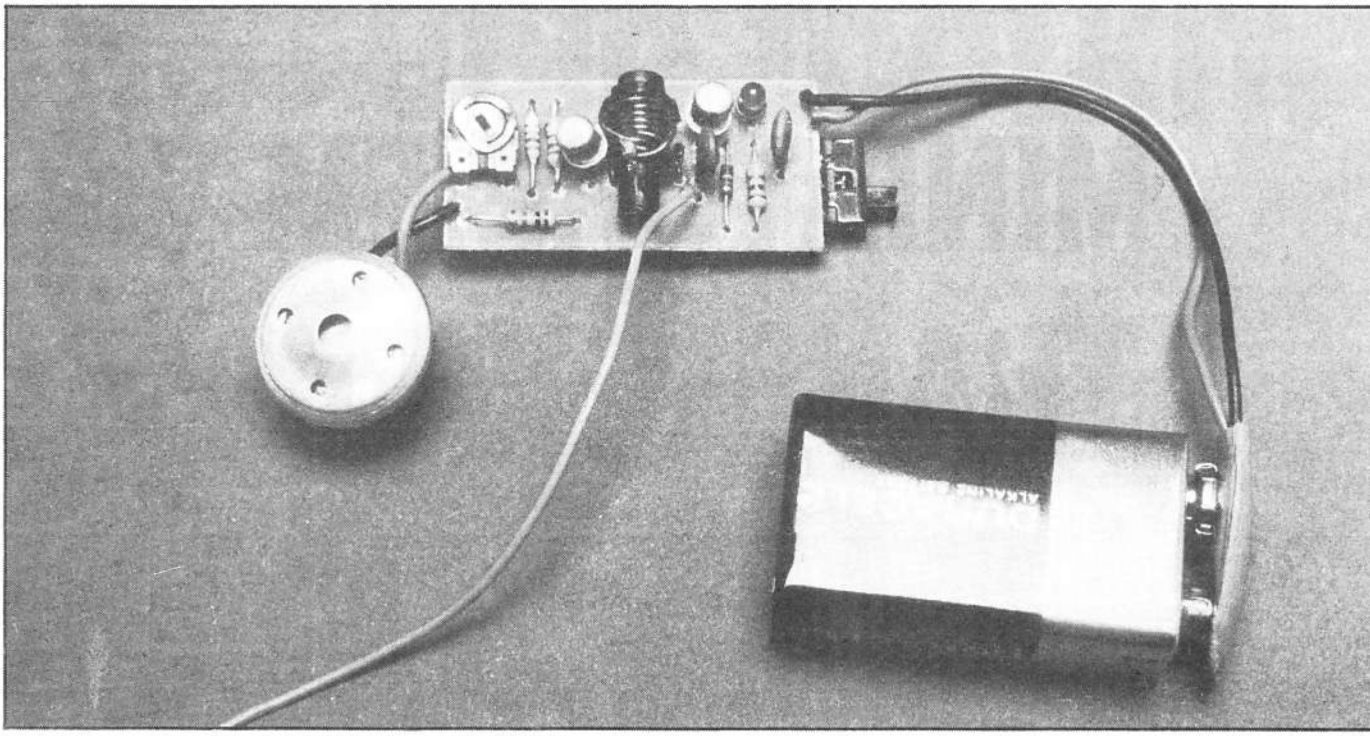
Here we publish something that's ultra small and easy to build. It's a minimum component radio transmitter operating at the bottom of the VHF radio band, somewhere around 88-90 MHz. We make no great claims for its fidelity, (strictly lo-fi) but it is perfectly adequate for transmitting speech over distances of 15 to 20 metres indoors, and further distances outside.

The design emphasises simplicity to the exclusion of almost every other consideration. The audio frequency from the microphone modulates a tuned circuit formed by the coil and some capacitors to derive an FM signal in the text book manner. Output from this is buffered and amplified by a single transistor amplifier and then fed to an aerial.

Construction

Construction is quite straightforward. First check the circuit board. Notice that it is rather small, so check that the etchant hasn't eaten right through the tracks anywhere, and also that no bridges remain where they shouldn't. It's a good idea to clean the board with some abrasive cleaner and plenty of water at this stage — that way you get to look at it closely.

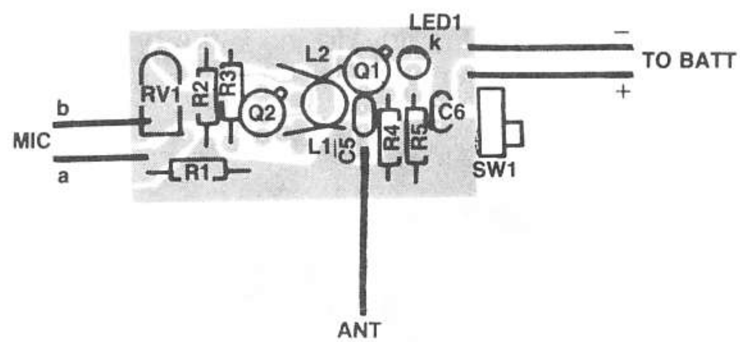
If the board hasn't already been drilled, do it now. Note that some of the components are soldered on to the back, so some of the pads don't have holes in the middle of them. Don't drill these out.

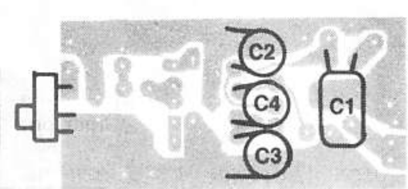


Now begin to solder the components on to the board. Begin with the resistors, then solder the two microphone leads into place. Then place the variable resistor. Do it in this order because one of the mic leads goes under the resistor, and it's rather tricky trying to get the wire into place if the resistors are already there. When you've done this, solder the two capacitors that fit on the component side into place, then the two transistors and the LED. Check the orientation of these components.

Now turn the board over and place the switch and four capacitors on the back. The easiest way of doing the capacitors is to first bend the legs at right angles about one mm from the component body. Cut the leg about one mm from the corner. Now drop some solder on to the pads, rest the capacitor on top of the fillet and apply some heat to the leg. This will heat the solder below and the component will sink into it. Apart from anything else, this method means you never need more than two hands for the operation. (Have you ever noticed how often in electronics you need one hand to hold the board, one hand for the component, one for the solder and one for the iron? If God had meant us to be electronic technicians we would all look like a multi armed Buddhist statue.) When you've finished this, fold over the 33n greencap, but leave the others standing upright for the time being.

Now comes the fun part — winding the coil. First wind on two turns around the former, cut the ends off and tin them. This forms L2. Next do the same for L1 making eight turns, and solder the four ends on to the board. Finally, solder the battery leads and the aerial in place, and go back over your work, searching for misplaced components, solder bridges or splatter.





PARTS LIST — ETI-751

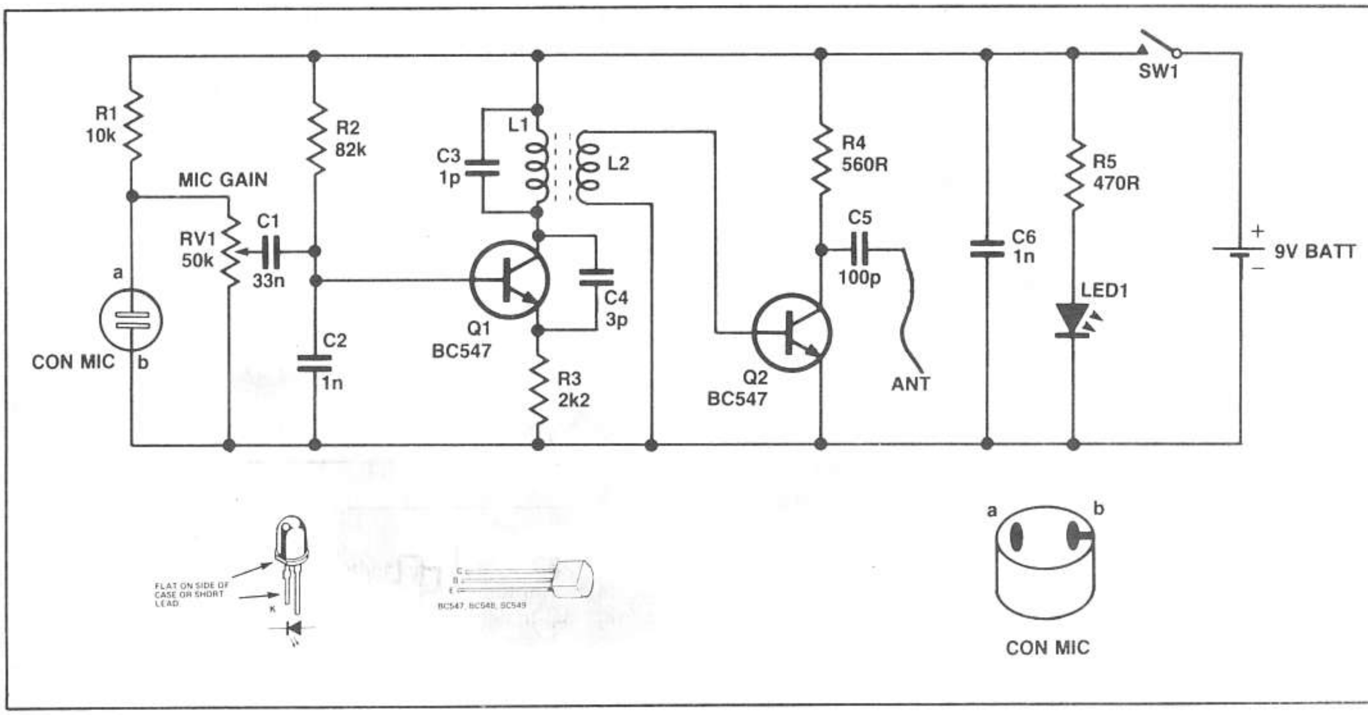
Resistors .... all ¼ W, 5%
R1 .... 10k
R2 .... 82k
R3 .... 2k2
R4 .... 560R
R5 .... 470R
RV1 .... 50k

Capacitors
C1 .... 33n greencap
C2, 6 .... 1n ceramic
C3 .... 1p ceramic
C4 .... 3p ceramic
C5 .... 100p ceramic

Semiconductors
Q1, 2 .... BC547, '549 or equivalent
LED1 .... 3 mm round LED

Miscellaneous
1 x 5 mm coil former, DSE Cat L-1010, together with a ferrite core like DSE Cat L-1302 and some enamelled 0.8 mm winding wire like DSE Cat W-3120; one ultra miniature SPDT slide switch (DSE Cat S-1015). The microphone should be an electret type — we used the DSE C-1160. A battery type 216 snap connector and 1300 mm of thin hook up wire to serve as the antenna.

Price estimate: $8.50



Setting up

At this stage, you need some other bits and pieces. An FM receiver is essential, a function generator and CRO desirable. Firstly, connect the battery, operate the switch and check that the LED comes on. If it doesn't, switch off straight away. Either the battery is a dud, the LED is in the wrong way around or you have the battery leads swopped over.

With the LED on, switch on the receiver and sweep through the FM band to see if you can find the 751's carrier. You should detect this as a sudden silencing of the FM noise. It helps if you can connect the function generator to the mic leads. Feed in a sine wave of about 1 kHz, and then you should hear tone from the receiver when you're tuned in.

More than likely you will find nothing on this first attempt. Tune the receiver to about 88 MHz, right at the bottom of the FM band, and screw the slug through the barrel. Still no luck? Carefully remove one side of L1 and take a turning off the coil, and then repeat the whole process. The frequency of the coil is quite sensitive to the number of turns and the position of the slug, so you may have to do this a number of times. We eventually got ours set up at about six turns.

It's a good idea to put on more turns than you need. That way you know that if it's not tuning properly you must remove coils. Also, if you need to increase the number of turns, you don't have to start right from the beginning again with a new piece of wire. Be careful to use the minimum amount of heat possible during all this resoldering, to avoid damaging the tracks.

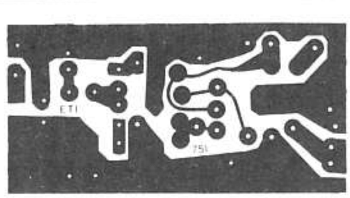
If you have trouble during this stage of the operation, a CRO is very helpful. If you probe the collector of Q1 you should see the oscillator waveform, which will tell you what is happening. If you don't have access to a CRO that will run about 90 MHz, a multimeter across the base emitter of Q1 will at least prove that part of the circuit, if it reads approximately 0.6 volts.

When you get it set up remove the function generator, solder in the mic and adjust the pot for maximum volume and minimum distortion. Fold over the remaining capacitors on the back of the board and glue the windings, slug and former together so nothing can move.

Using

The next problem is mounting the device. This is entirely up to your own ingenuity and imagination. A few points to notice. If you want to use it to monitor a particular position, you may not need the switch. In that case remove it and bridge across the switch pads with a bit of wire. The LED then serves very little purpose, so you may wish to consider removing it together with its associated resistor. As a bonus this will also reduce battery drain.

However you do it, make sure the mic is securely mounted. If you leave it to dangle off the leads it won't last long. If you intend mounting the device in a case things will take care of themselves. If not, consider using a short piece of one of the component legs instead of wire for the connection. That should lead to a sufficiently stiff mounting. ●



HOW IT WORKS — ETI-751

The principle of the circuit is obvious but the exact solutions for characteristics of the oscillator are hard to calculate. Making rough approximations on the way, we found that the inductor L1 would be somewhere around 500 nH. A working sample of the inductor was actually measured under the L-C-R bridge. We obtained a value which is fairly near to the calculated one.

The signals picked up by the condenser microphone are ac coupled to the transistor Q1. RV1 can be manually set to attenuate this coupling. Varying this pot will affect the sensitivity and the distortion of the circuit.

The ac signal from the microphone will vary the ac emitter current of transistor Q1. This changing current changes the effectiveness of the capacitor C4 as seen by the tuned circuit formed by L1, R3, C3 and C4. The apparent changing capacitance of C4 shifts the oscillation frequency of the tuned circuits.

Capacitor, C4, also plays an important feedback role keeping the oscillator oscillating. The free running frequency of the tuned circuit with no input microphone signal is around 88 MHz. This is the carrier frequency. The modulation of this frequency by the microphone signal generates the FM signal which can be picked up by a normal radio.

To stop the antenna from loading the tuned circuit directly, a secondary wiring is used to couple the signal to transistor Q2. This also introduces a bit more gain to the circuit. If the output resistance of Q1 is Ro and the input resistance of Q2 is Ri, the turns ratio N of the transformer formed by L1 and L2 is given as:

$$N = k \frac{Ro}{Ri}$$

where k is the coupling between the windings. Its value depends solely on the material of the slug. The LED and R5 form a circuit ON indicator when the battery is connected to the circuit.

E T I - 1 4 2 6
E L E C T R O N I C S

In the true tradition of our search for new technology, ETI's John Dix presents a novel way of enhancing the low frequency response of small speaker units.

'The enclosure has been designed to make construction as simple as possible'

QUARTER WAVE LOADING

Utilising the space within and under your speaker stands

In these enlightened days of CD, DAT and all that is silent at the source, there is proportionately less in the music budget to be spent on the loudspeakers. The loudspeaker systems manufacturers have concentrated on smaller enclosure designs to achieve a cost reduction with the minimum possible sacrifice in performance.

Although it is more difficult to maintain the low frequency response with a small enclosure, reducing the dimensions has a number of advantages. A significant increase in structural stiffness reduces unwanted radiation from the cabinet walls. The narrow frontal area also improves the sound distribution.

Larger loudspeaker systems have to be complex because a mid-range unit is needed, with careful integration of responses to cover the whole frequency range. In a smaller unit, a single bass/mid-range unit provides seamless coverage beyond the critical mid-frequency range, easing crossover design and producing a radiation pattern conducive to a natural spread of sound and a useful wide stereo sound stage.

However, if a small enclosure results in an abrupt roll off of bass level below 100 Hz, the bass lightness becomes readily apparent and there is, therefore, a limit to the economy feasible if a unit is to provide the reasonably long throw cone excursions necessary for adequately low frequency radiation.

Close proximity to a wall can give rise

to interfering standing wave patterns which deteriorate the stereo image. Having the speakers away from the wall, stably in space at a height such that the high frequencies are not absorbed by the sofa, gives an obvious improvement in depth and image precision.

Bearing all these points in mind it seems logical to consider whether the space within and under the speaker stands could not be used to enhance the low frequency response while maintaining a low cost, freestanding unit.

Design

A freestanding loudspeaker enclosure with similar dimensions to that of a small speaker on a stand, if of conventional design and construction, presents a difficult acoustic problem to the designer because of the long narrow parallel walls. These will tend to vibrate and resonate giving a resonant pipe-like colouration to the low frequency sound which is difficult to control and eliminate.

Another approach (satisfying from an acoustic engineering point of view) is to deliberately exploit the characteristics of a resonant pipe in such a way that the loudspeaker unit is correctly loaded and terminated at the low frequencies, while adequately suppressing unwanted pipe resonant modes.

The low frequency efficiency of such an arrangement is somewhat between that of a horn and a bass reflex enclosure and, therefore, reduces the demands made on the low frequency excursions of the small diaphragm bass speaker unit.

The principle involved utilises the properties of a closed at one end quarter wavelength pipe as originally proposed by Voight in his patent No 447749 and subsequently adapted and described by R. West and R. Baldock in their designs. The design produced by R. West was intended for a corner position with the speaker unit firing into the corner to spread the high frequency sound by reflection from the walls, and R. Baldock's designs were intended to either a semi-omnidirectional sound distribution or a wall reflected distribution.

Present day practice favours loudspeaker operation away from corners and walls, firing directly at the listeners.

The enclosure

The construction of the design is depicted at left. The bass enclosure consists of a quarter wavelength rectangular section pipe with a linear taper, resonant at about 50 Hz.

The bass loudspeaker unit is situated at approximately halfway along the acoustic axis in the best position to suppress higher order resonant modes. At resonance the acoustic pressure is high at the tapered end and still reasonably high at the loudspeaker unit. This ensures that effective acoustic loading is presented to the loudspeaker cone and small excursions of the cone at high pressure are manifested as much larger low pressure movements of air out of the port at the bottom of the enclosure.

Such a process, similar to horn loading, contributes to efficient bass frequency operation with low distortion up to a frequency of 200 Hz, where direct radiation from the cone takes over. The enhanced bass response produced by this method of loading compared with that from the same unit in a 10 litre sealed enclosure is shown in Figure 2, where the curves were obtained under identical measurement conditions.

This enclosure not only satisfies the requirements of being free-standing with the drive units at a convenient height but also provides an enhanced bass response, using the space that would otherwise have been taken up by a stand. Furthermore, only small cone excursions are required in the bass loaded region and this places the minimum of demands on linearity of the cone suspension and the magnetic field in the voice coil gap, allowing reasonably low priced drive units to be employed.

Continuing the quest for a low price design, it is tempting to consider a wide range twin cone unit for use in this enclosure. Figure 3 shows the high frequency response of a 165 mm diameter paper cone bass unit used in this position with considerable ripple in the response due to cone "break-up" modes. Unfortunately, when a small tweeter cone is added to the main cone to widen the frequency range, any improvement in frequency response is accompanied by main cone "break-up" ripple as shown in Figure 4.

A much smoother performer is the 165 mm polypropylene cone bass unit with a frequency response as shown in Figure 5 and this type is recommended for use in the quarter wave enclosure.

Because of the unsatisfactory response of twin cone units, space is provided in the top of the quarter wave enclosure to house a suitable tweeter.

Construction

The enclosure has been designed to make the construction as simple as possible and if the various pieces (see Figure 6) are cut accurately square than there should no difficulties in assembly.

Referring to Figure 7 it can be seen that there are only two angle cuts to be made, those at the top of both the long front and back panels. All the rest are simple 90° butt joints, and it is left to the individual constructor to decide whether to attempt the angle joints or simply butt the joints and fill the wedge shape gaps with whatever technique and material is convenient.

The dimensions quoted are not critical provided everything is checked to fit as shown in the diagrams so that airtight joints are obtained, particularly in the high acoustic pressure areas in the tapered wedge and around the speaker unit.

The front, back, bottom, top and internal partition members are all made of nominally 12 mm thick chipboard and should all be matched to the same width of 175 mm. The two side panels are made of nominally 6 mm thick plywood and it is recommended that one of the panels is marked out indicating where the 12 mm thick panels are located.

As the assembly progresses check it for squareness and, if necessary, secure

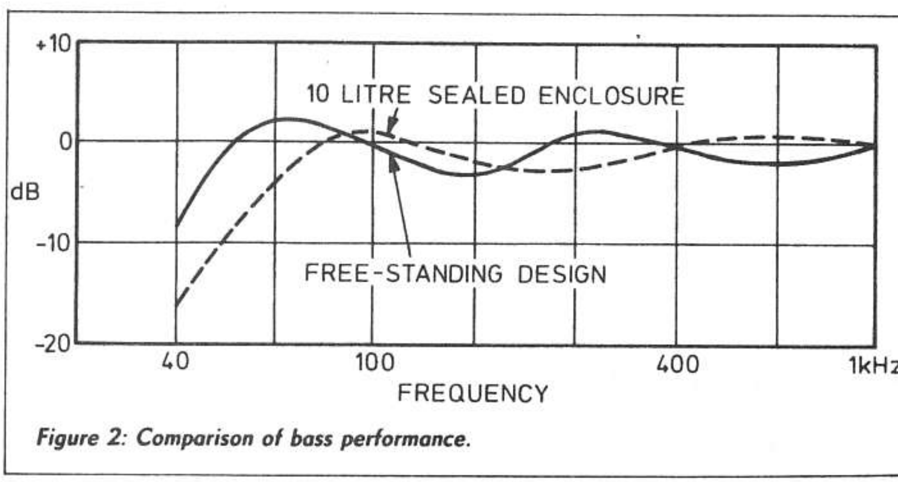


Figure 2: Comparison of bass performance.

one or two cross pieces of plywood with pins driven a little way in to hold the assembly square while the glue sets.

Being reasonably liberal with the glue should ensure airtight joints but pay particular attention to the pointed end of the wedge section and, if necessary, run a fillet of glue along this particular joint.

Finally complete the assembly by glueing and pinning the second 6 mm thick plywood panel into place. It will be noted that the enclosure is reasonably light and stiff and this minimises the energy storage in the enclosure walls. Tapping the sides of the enclosure produces different notes at different positions indicating that the internal bracing and asymmetry is working to minimise undesirable reflections and panel resonances.

Finishing tasks involve punching the pins home, filling and sanding prior to painting or covering with material or an iron-on veneer.

After several years experimenting with various drive units, the best solution — both in terms of cost and performance — seems to be the simplest of crossover arrangements with a direct connection to the bass unit and a capacitor feed to the tweeter.

The need for attentuation is avoided by choosing a sensitivity for the high frequency unit just below the bass unit. The speaker baffle is as small as possible for rigidity and minimum frontal area. The sloping of the baffle time-aligns the outputs from the two units, improves the coupling of the bass unit to the air column in the enclosure. It also exploits an improved smoothness in frequency response of the bass/mid frequency unit observed at this angle off its central axis rather than complicating the crossover.

The response

Figure 8 shows the combined anechoic response of the two units as derived from the manufacturer's quoted responses as a dotted line, with the in-room frequency response as a solid line (in-room measured using third octave noise with a calibrated mic at 0.9 m height). The responses show good integration and smoothness. A bonus of the simple crossover and small sloping baffle is an excellent off-axis response.

The modulus of the installed bass unit's impedance against frequency as shown in Figure 9. A resonance was detected at about 250 Hz but became inaudible with the insertion of damping material into the open end of the closed tapered section as indicated in Figure 1.

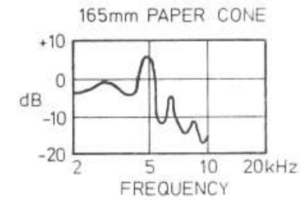


Figure 3: Frequency response of paper cone unit.

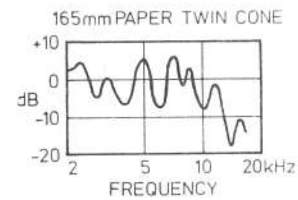


Figure 4: Frequency response of twin cone unit.

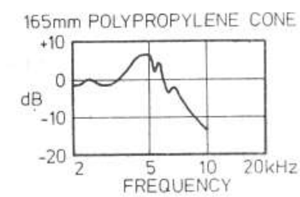


Figure 5: Frequency response of polypropylene cone unit.

The effect of the damping is also shown in Figure 9.

The damping material is a square metre of terylene wadding (from addressmaker). It should weigh about 100 grams and is cut in two — a piece for each enclosure. Each piece is folded lengthwise in two and the resulting strip is folded again twice to form a 25 cm square, ready for insertion.

The loudspeaker units are mounted from the outside of the enclosure and the bass unit needs a sealing gasket cut out of a thin sheet of plastic foam or paper, depending on the surface finish of the baffle. Use chipboard screws and do not over tighten.

Electrical connections may be made to a connector block fastened just above the port. The bass speaker lead is simply passed down through the bass enclosure and out through the port, while the tweeter lead is secured by clips down the back of the enclosure. The series capacitor supplied with the tweeter is a non-polarised elecrolytic and readers may wish to upgrade the performance by replacing this component by a better quality version. Readers may also wish to experiment with the provision of steel or plastic spikes in the base of the enclosure.

Performance

The present design is the result of many

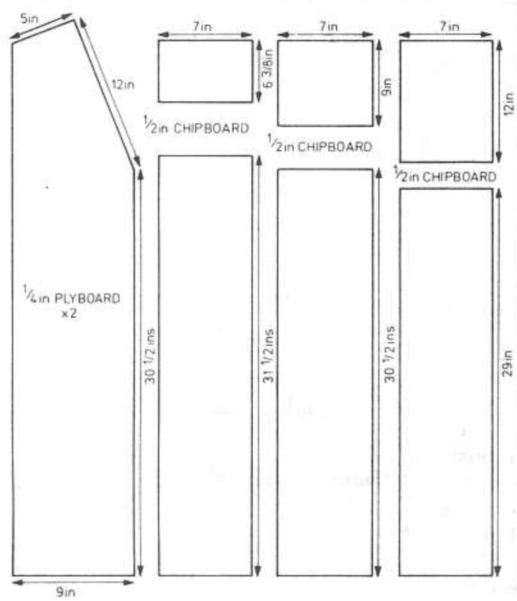


Figure 6: The cutting diagram for a single QWL loudspeaker (the baffle plate is also needed and is shown in Figure 7).

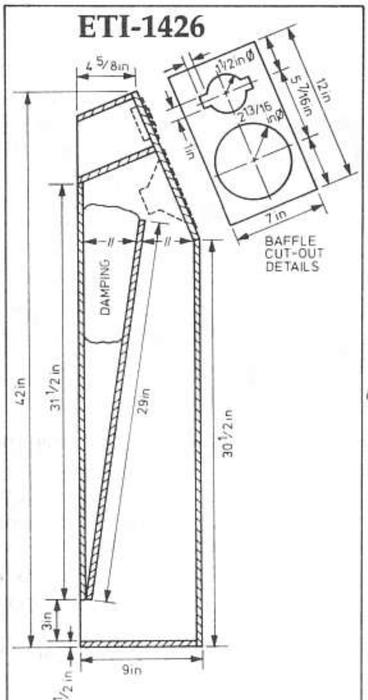


Figure 7: Enclosure dimensions and construction.

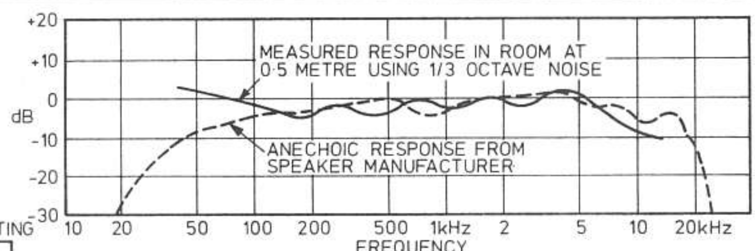


Figure 8: Frequency response of complete loudspeaker systems.

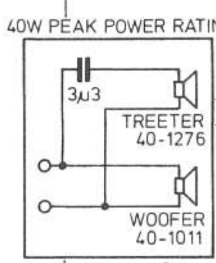


Figure 9: Off-axis response.

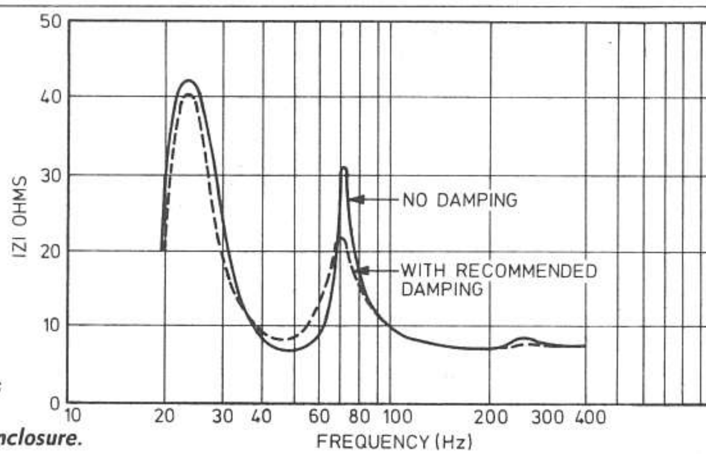


Figure 10: Bass loudspeaker impedance in enclosure.

hours of measurement and listening.

This QWL design is relatively cheap and easy to build but achieves a combination of good measured frequency response, stereo imaging, sound quality and efficiency. They occupy very little floor space, are easily moved and are the correct height to preclude the need for stands.

Happy listening.

The drivers used in this project are the Tandy 40-1011 and 40-1276.

ERRATA

Some dimensions were missing from Figure 7. The bass driver should be 3¾ in above the base of the baffle panel. The notches in the side of the tweeter cut-out are ½ in wide. The top plate is missing from the cut-out diagram (Figure 6). This is 7 x 4 5/8 in.

ETI-1425 GUITAR NOTE EXTENDER

If you want your guitar notes to keep on hanging on cleanly, then the ETI-1425 may be just for you.

Terry Kee

THE INFAMOUS "fuzz box" or distortion box has long been the electric guitar player's most used effect pedal. One of the reasons responsible for this is that the distortion allows the player to sustain the notes much longer than normal and the other is due to the harmonic change generated by the clipping or bending of the guitar signal. It is often difficult to achieve a guitar sound with heaps of sustain and without it being buried in distorted grunge.

The ETI-1425 solves the problem and provides the sustain without the distortion. Using the Note Extender as the first pedal in the effects chain, and there are usually many, can accentuate the effect of other ones, as an example, using it with a distortion unit set to give just a hint of "edge" can fire that guitar solo into life.

The 1425 has a Bypass LED indicator which tells us when the effect is switched in or out. The Input Drive knob is used to set the input gain so as not to overload the unit and it also determines the amount of sustain required. The output level can be varied by the volume knob. The unit is powered by a 9V battery. There is also a dc socket to accept a 9V battery eliminator. The power to the unit is switched on when a jack plug is inserted into the output socket, thus eliminating an on/off switch.

Design Considerations

The Note Extender works on the same principles as the familiar Automatic Level Control circuit found in radio receivers. The heart of such a device is a Voltage Control Amplifier (VCA) and a rectifier that converts the ac signal into an appropriate dc level to control the VCA and keep the level constant. The time constant of the rectifier filter plays an important role in determining the overall performance of the unit. The attack time has to be fast enough to bring down the gain quickly to stop the transient portion of the guitar signal from being amplified excessively.

Remember that once the guitar note has decayed the system gain will be at its maximum thus making the speed of the attack time even more accute. The decay time has to be fairly long to smooth out the low frequency that can effectively modulate the VCA and manifests itself as distortion at these frequencies. Furthermore the decay time has to be short enough to follow the envelope of the signal. In a simple RC filter network the fast attack and slow decay requirements are conflicting as the time constants are determined by a single capacitor.

After some experimentation with different time constants and plucking endless guitar notes, I came to the conclusion that a dual time constant rectifier circuit was essential. The compromise between low distortion and a fast response became too critical. Furthermore the results that were obtained were too dependent on guitar playing styles.

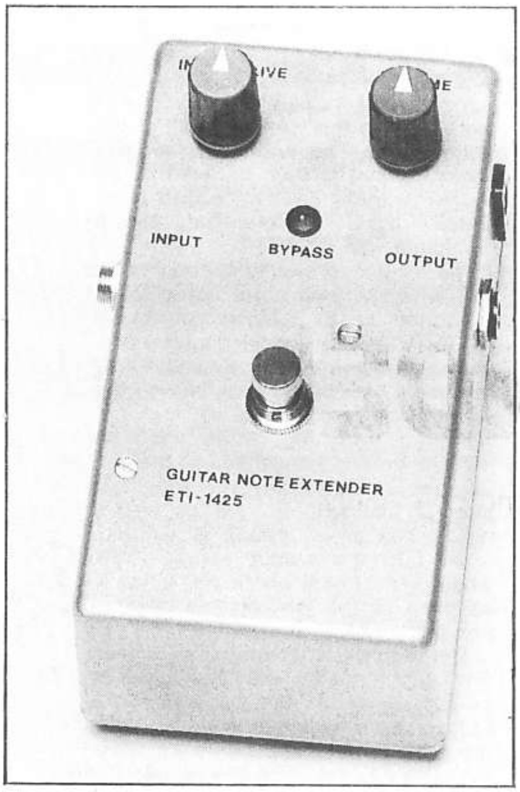
Occasionally plucking a note hard and fast allowed a nasty click through to the output as the sytem was too slow to respond to it. My attention was then drawn to the 572 companding chip which has a separate attack and decay time constant built into the rectifier filter network. The 572 can be configured as a compressor or an expander circuit hence the name, companding. In the Note Extender the 572 is configured as an Automatic Level Controller (ALC) with the signal input connected to the rectifier input rather than the signal output, which is the compressor configuration. This makes the gain inversely proportional to the input level so a decaying signal level will produce a proportional increase in gain to keep the level constant.

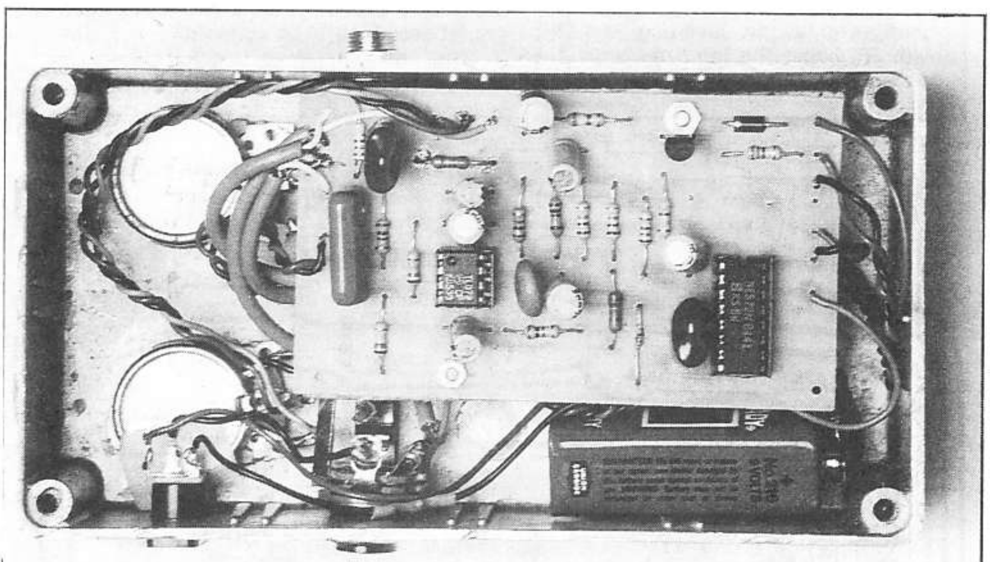
The amplification of the lower levels of the guitar signal does present us with a problem as the level becomes comparable to the guitar pick-up noise and that dreaded hum!

Unfortunately the envelope of a plucked guitar string decays fairly quickly so low level amplification is important to the amount of sustain perceived by the ear. However, there has to be a compromise between the amount of sustain and acceptable noise levels. This compromise is also a function of the type of guitar pick-up to be used with the 1425. The single-coil pick-up as found on Fender Strats are renowned for their ability to pick up hum and to generate noise of their own.

The humbucker is a twin coiled pick-up which cancels out any picked up hum (hence the name). These pick-ups have their own characteristic sound and both are widely used.

Furthermore some guitars are not properly screened which makes them a perfect aerial for hum! To accommodate different types of pick-ups and to obtain the best performance with your guitar a Select On Test resistor is included in the design. A measured SNR of −80dB was obtained





with the input short circuited so most of the noise will be generated by the pick-ups.

Circuit Description

The input amplifier IC1a serves to provide a high input impedance of around 90k ohms so as not to load the guitar pick-ups and to provide amplification for those lack-lustre pick-ups. The gain is made variable via the Input Drive pot from 0 dB to a maximum of 20 dBs.

A bass cut is also built into the input amplifier to attenuate low frequency noise and hum. The mid to high frequencies are thus accentuated by the ALC action and this makes the sustain effect more noticeable. The signal is then applied to the heart of the Note Extender which is the gain control section consisting of the 572 Companding ic. The gain cell contained in the 572 is placed in the feedback loop of an op-amp with the input signal connected to the rectifier input.

The 572 thus acts as a varying impedance in the feedback loop that is controlled by the input signal and in turn, varies the voltage gain of the op-amp. When the input amplitude decreases below the cross-over point which is around −30 dBm, the overall system gain increases proportionally to hold the output at a constant amplitude of −30 dBm.

As the input level increases above the cross-over point, the system gain decreases proportionally again holding the output amplitude constant. The amount of ALC action is thus level dependent and the Input Drive control ensures that the input level can be varied to an amplitude comparable to the cross-over point.

The rectifier in the 572 consists of a full-wave circuit and a buffer amplifier that implements the separate attack and decay filter network. Capacitor C6 and a 10k internal resistor determines the attack time.

In the Note Extender an attack time of 1.5mS was found to be suitable only after extensive plucking of guitar notes and listening carefully to the transient response at the output. The decay time is determined by C7 and another internal 10k resistor.

A time constant of 68 mS was suitable and produced about 1% distortion at 100 Hz. Note that the 572 is a dual channel device, however only one channel is used.

To reduce the gain at very low levels, R15 supplies an extra current to the rectifier which raises the voltage applied to the gain cell and effectively reduces compression at low levels. The ALC action is further reduced at low input levels by R5 which limits the maximum gain of the circuit.

Unfortunately limiting the low level gain by these methods also affects the amount of gain availble at the cross-over point. Hence the trade-off is acceptable noise generated by your particular guitar pick-ups and sustain. I used a Stratocaster guitar with single-coil pick-ups to set up the values of R15 and R5 experimentally.

The procedure to set up the 1425 to work best for your individual guitar is described in the Testing section.

The input jack socket is wired to short out the input when no plug is inserted. The output socket has a switched connection that disconnects the ground connection from the battery or dc power source. The footswitch allows the output to be switched between the output of the Note Extender or the input signal. A LED is also switched on when the unit is in use.

The unit can accept a maximum input level of 10 dBm with the Input Drive set to minimum before clipping commences at an output level of −6 dBm.

Construction

The circuit board is built on a single-sided pc board measuring 90 x 53 mm. Once the pc board is etched and the holes are drilled then start by mounting and soldering in the resistors, capacitors, link and ic sockets (if you have decided to use them). Check the polarity orientation of the electrolytic capacitors by referring to the overlay. Note that C10 is a 470µ/25V capacitor and if you find that the component height is too high to fit in the box, then solder the cap on the solder side. Solder in the diode and transistor, noting again their correct orientation.

The next step before the wiring can commence, is to decide on the type of box you want to use. Since the unit is to be floor mounted, it needs to be rugged and made of metal for screening purposes. I used an aluminium diecast box measuring 150 x 50 x 80 mm, commonly available from electronics stores. The holes on the box need to be marked and drilled out. I did not use a Scotchcal panel as they tend to scratch quite easily and would not survive all that foot pounding! Measure out the holes by dropping in the hardware and checking that everything fits. Do not forget about the 9V battery! Ensure that the marks are neatly aligned before the holes are drilled. I managed to fit everything in the diecast box although it was a tight squeeze.

The pc board was mounted on top of the footswitch and input jack socket and tightened down to the top panel via 2 bolts and spacers. The footswitch contacts were bent down so that the pc board could fit inside the box.

Next comes the spray painting, if you want the unit to look really professional. I coated the box with a metallic blue spray, available from your local car accessory store. Smooth the surface of the box with a small piece of fine wet and dry sandpaper. Wipe it clean and allow it to dry. Spray the box with three light coats of paint, allowing each coat to dry fully. Use some Letraset or something similar to letter the knobs and sockets. Spray the lettering with a clear protective coat to avoid the lettering from being rubbed off.

Once that is done it is time to commence with the wiring. Use screened audio cable for the connections to the jack sockets and footswitch. Note that the connections to the footswitch have the screen cut at the switch end but soldered to ground on the pc board. Sleeve the connection at the switch end to prevent any short circuits to ground. Use hook-up wire for the rest of the wiring keeping the connections to the two pots as short as possible to minimise stray pick-up. Insert the ics into their respective sockets if you have not soldered them in already. Take note of their polarity.

The final stage is to cut a piece of cardboard to insulate the solder side of the pc board from the footswitch and input jack socket. Tighten the nut of the input socket SK1 firmly as it also acts as the ground connection to the metal box.

HOW IT WORKS — ETI 1425

IC1a is configured as a non-inverting amplifier with the gain set by RV1 and R4 to a maximum of 20 dB. R2 and R1 establish the biasing point 4.5V for the op-amp to operate with a 9V supply and sets the input impedance to 90k. The signal is then ac coupled via C3 and C4 to IC1b and the rectifier input of the 572 (IC2). C6 determines the attack time with an internal 10k resistor and sets it to 1.5 mS; C7 sets the decay time to 68mS.

The resistor R5 limits the maximum gain of the circuit to reduce the ALC action at very low input levels. Reducing the value of this resistor reduces the gain and increasing it, increases the ALC action. R15 is connected between pin 2 to 9 V to reduce the gain even further at very low levels.

Resistor R9 is used to bias the output halfway between the supply and ground to obtain the maximum headroom. A value of 22k sets pin 7 of IC1b to 4.5 Vdc feedback is provided by R10 and R11 and C8 ensures that no ac signal is present in this path.

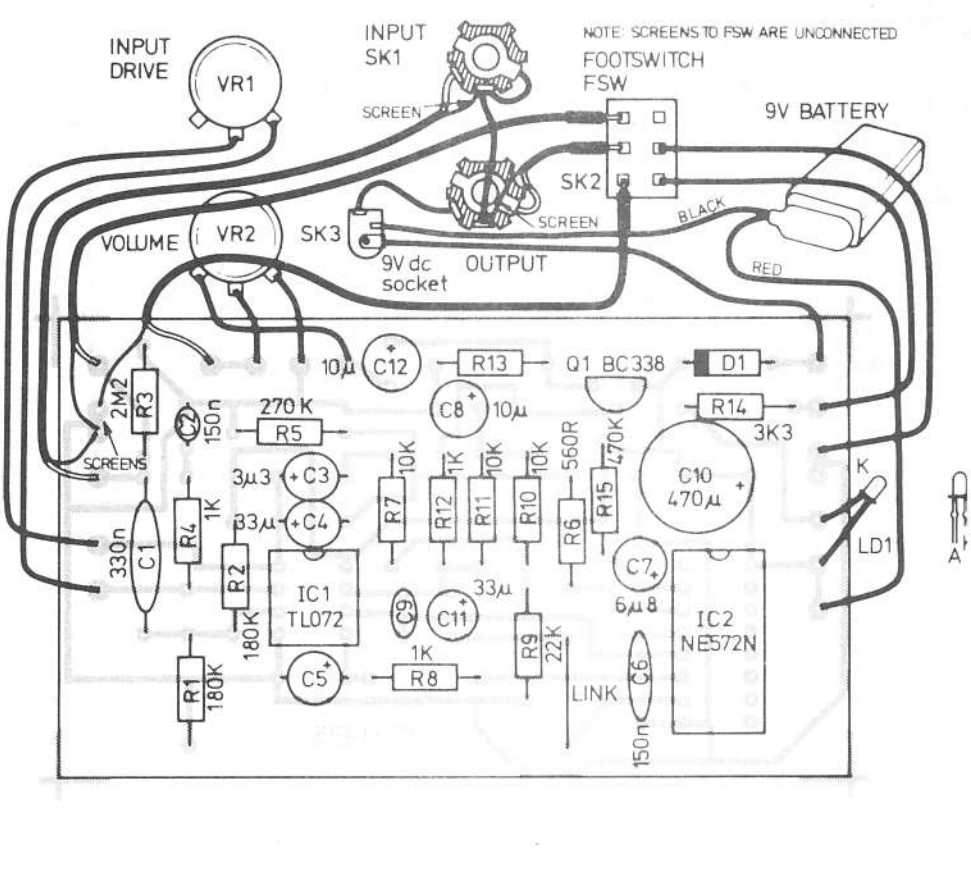
The 572 is placed in the negative feedback loop of IC1b with the output ac coupled via C11 into the input of the gain cell (pin 7). The gain circuit is configured as an inverting amplifier with R7 being the input resistor. The non-inverting input of the op-amp (IC1b) derives its dc bias from pin 6 of the 572. Note that only one half of the 572 is used. The output of the gain control circuit is then buffered by an emitter follower Q1. The output is ac coupled by C12 before being fed to the volume pot (VR2).

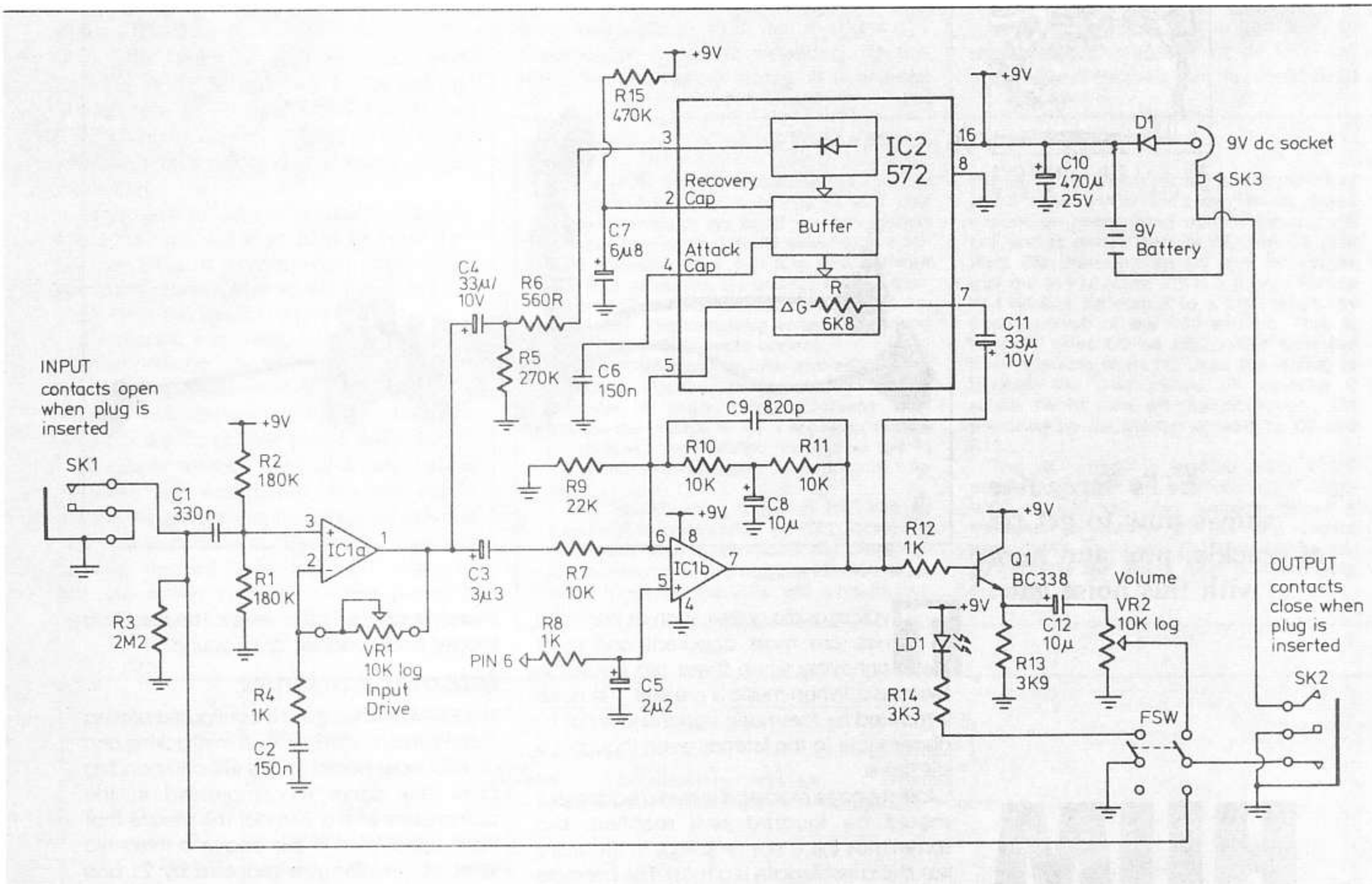
The 9V rail is heavily decoupled by C10 which smooths out the glitches generated by the gain control circuitry. The diode D1 provides circuit protection for incorrect supply polarity by becoming reversed bias when the wrong polarity is applied.

Battery switching is arranged so that the ground connection of the battery becomes disconnected when a dc plug is inserted into the dc socket (SK3). The on/off switch is incorporated into the output socket (SK2) where the ground continuity is made as soon as the jack plug is inserted into the socket.

The footswitch is configured to feed the output socket with the input or output signal. At the same time the LED circuit is switched in via the limiting resistor R14 and lights up when the output is selected. The resistor R3 discharges the voltage on any capacitors in preceeding units and enables quiet switching to be achieved.

The unit consumes about 9 mA dc current when the unit is switched in.





Testing

Check the pc board for solder bridges across pads and track for broken tracks, before it is fastened down to the box.

Connect in a 9V battery and insert a jack plug into the output socket to switch on the power. Measure the dc voltage across the power rails with a multimeter and check that it is indeed around 9V. If the voltage falls to 0V then disconnect the battery and check your soldering for a short circuit across the power rails. If you are using one of the commercial 9V battery eliminator then ensure that the polarity of the dc power plug has the positive terminal at the centre of the plug. Note that the unit needs a regulated 9 Vdc supply so it will not operate satisfactorily with a plug pack. Switching the footswitch should light up the bypass LED.

Once that has been ascertained, plug in your guitar and connect the output to an amplifier. Rotate the input and output controls to maximum and listen to the output. Compare the difference by plucking a note, the open G string is a good one to use. Switch in the unit and you should hear the note hanging on much longer than the straight through version. If the output is distorted then the input level is too large. Back off the Input Drive controls until the distortion disappears. Note that decreasing the input drive will also reduce any extraneous pick-up noise.

If your guitar pick-ups are excessively noisy then experimenting with the value of R5 should help enormously. Reducing the value will decrease the gain; a value of 220k would be a good starting point. Conversely if your pick-ups are particularly clean then increasing the value of R5 will extend the sustain. Happy Plucking. ●

ETI-1425 PARTS LIST

Resistors ....all ¼W, 5%

R1, R2	180k
R3	2M2
R4, R8, R12	1k
R5	270k
R6	560R
R7, R10, R11	10k
R9	22k
R13	3k9
R14	3k3
R15	470k
RV1, RV2	10k log pot

Capacitors

C1	330n greencap
C2, C6	150n greencap
C3	3µ3/25V pc mount electro
C4, C11	33µ/10V pc mount electro
C5	2µ2/25V pc mount electro
C7	6µ8/25V pc mount electro
C8, C12	10µ/25V pc mount electro
C9	820p ceramic
C10	470µ/25V pc mount electro

Semiconductors

D1	1N4001
Q1	BC338
IC1	TL072
IC2	NE572
LD1	Yellow LED

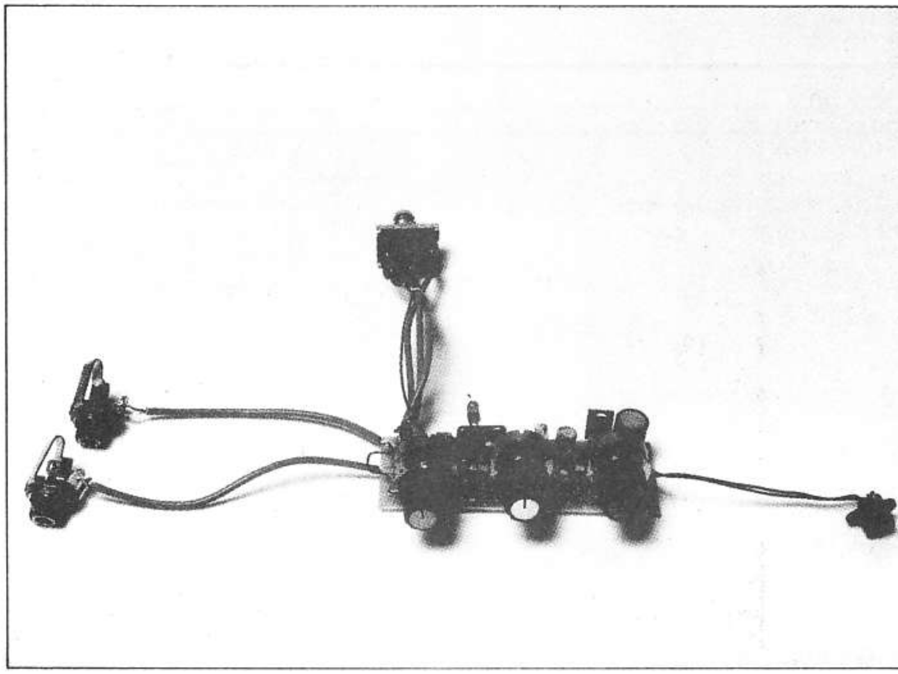
Sockets and Switches

SK1	6.5mm Switched Jack Socket (contacts open when plug is inserted).
SK2	6.5mm Insulated Mono Jack Socket (contacts close when plug is inserted).
SK3	Insulated Switched dc socket (contacts open when plug is inserted)
FSW	Heavy Duty Footswitch DPDT

Miscellaneous

9V battery holder, 1 off 8 pin ic socket, 1 off 16 pin ic socket, diecast box (150 x 50 x 80 mm), 2 knobs, automotive paint spray

ELECTRONICS
ETI - 1429



ETI's Terry Kee shows how to get rid of crackle, pop and hum with this noise gate.

HUM AND HISS NO MORE

Eliminate annoying audio noises

Typical audio noises, such as hum and hiss, are most apparent and most annoying when there are pauses in the music. When music is present, the noise is masked by the music signal and is not as discernable to the listener, even though it is still there.

Ideally, noise problems in audio equipment should be located and rectified, but sometimes this is not practical. In situations like this, a noise gate is a must. The principle is quite simple. Any noise signal will generally be at a lower level than the average value of the signal, hence if the noise gate can be set to allow signals at high levels through but shut down at lower levels then the dynamics of the signal will largely be retained, but the hum will be reduced. When signal is not present the noise is effectively gated out.

The applications of the noise gate can be used to good effect in many situations. Musical instruments that use magnetic pick-ups are prone to pick up hum, particularly if they are located close to a magnetic field like a power supply in an amplifier, which is quite likely to occur in a live situation. The noise gate can also improve the quality of live recordings where audience noise, or sounds from instruments, spill over into adjacent microphones. Connecting a noise gate in each of the mic channels cleans up the sound by gating out the spill-over. Another use of the noise gate, common in recording studios, is to reduce the reverberation times of instruments, notably drum sounds, which already have added reverberation. In this context the noise gate acts as an envelope generator which shapes the envelope of the signal. The ETI-1429 can produce some envelope shaping by varying the threshold where the gating starts, so that lower signal levels are attenuated more sharply than the higher levels. The result is a shorter and punchier drum sound.

Internal workings

The ETI-1429 noise gate is configured around a compressor with low level mistracking and an expander based on the 572 companding chip. The signal is compressed in the compressor with a 2:1 ratio. This means that input signal level in dBs above a threshold level of −18 dBm are reduced by 2:1 and levels below the threshold are increased by 2:1 (refer to the graph plotted in table 1). If the threshold level is referenced at 0 dB then an input level of −20 dB is increased to −10 dB and a 12 dB input level is reduced to 6 dBs.

> *'Another use is to reduce the reverberation times of instruments'*

The output of the compressor is fed into an expander circuit which provides a complementary action to the compressor. It expands signal levels above and reduces signals below the threshold with a 1:2 ratio. The result is a unity gain signal i.e. a 1:1 ratio. A graph of the companding system levels of the 572 is shown in Table 1. The noise gate action is introduced at the compressor by mistracking the compression ratio at low levels. This means that at high levels the signal undergoes a 2:1 compression whereas low levels tends towards a 1:1 ratio.

At the output of the expander the applied signal undergoes the nominal 1:2 expansion, which reconstructs the top half of the signal so that it follows the input signal. But due to the low level mistracking of the compressor the bottom half signal is expanded down by a factor of 2.

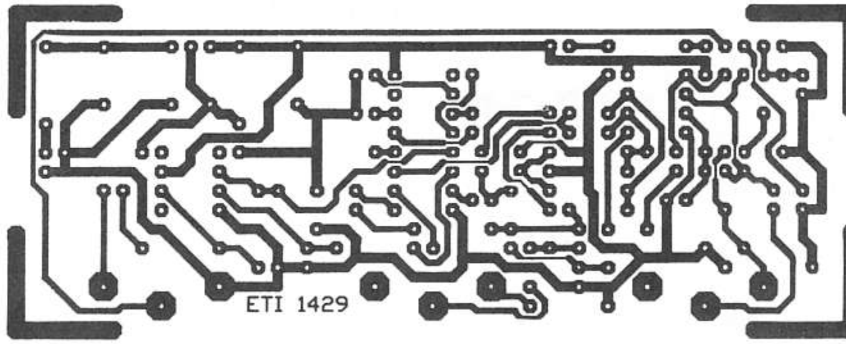
The threshold where low level mistracking of the compressor occurs is made variable. The graph plotted in Table 2 shows the effect of varying the threshold. The response of the noise gate has to be compared to the 1:1 graph where the input equals the output. With the threshold set to minimum, signal levels below about −50 dBm start to detract from the nominal 1:1 curve. When the threshold is set to maximum it is raised to around 5 dBm. The thresholds of −40 dB, −30 dB and −20 dB are also shown in table 2.

NE 572 companding chip

The 572 is a dual channel gain control circuit where one channel is configured as the compressor and the other as the expander. Each channel has a full-wave rectifier that detects the average value of the applied signal, a variable gain cell and a time constant buffer. This buffer allows separate control of the attack and recovery time constants. This feature reduces the low frequency ripple that controls the gain cell and reduces the distortion at these frequencies. The attack time of the noise gate is set fairly fast to 3.3 mS via C12 and C16 and the release time by C11 and C15 to 47 mS (see the circuit diagram). These time constants work well with sharp percussive drum sounds and guitar. Note that C12 must equal C16 in value (ditto for C11 and C15) if you want to experiment with different time constants. Reducing the release time will increase the low frequency ripple and thus worsen the low frequency distortion.

Performance

With the values shown in the circuit a THD of



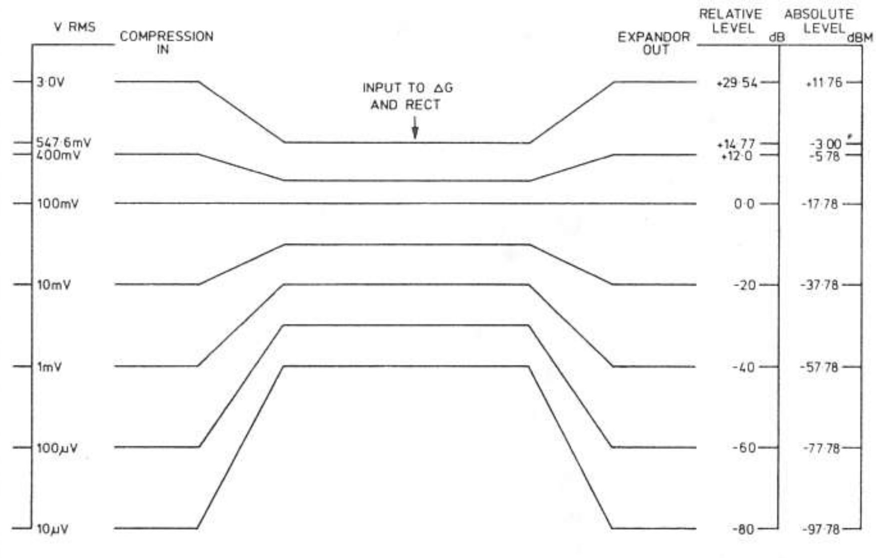
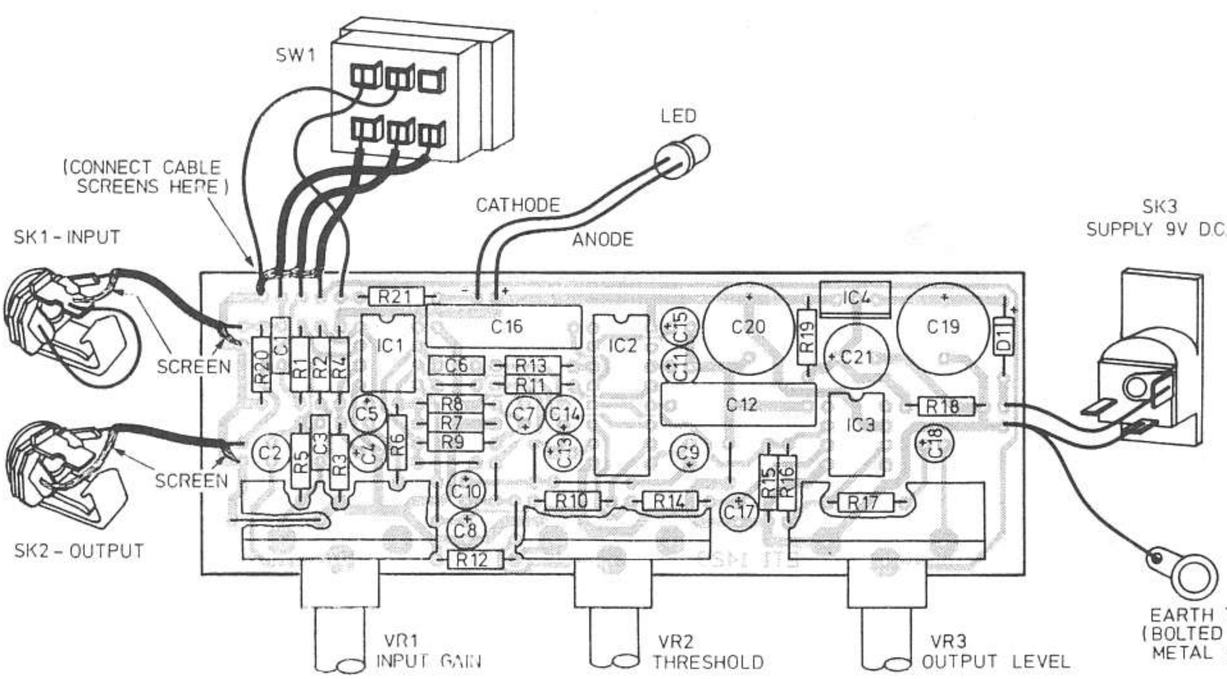


TABLE 1. NE572 SYSTEM LEVEL



Noise gate

0.6% at 100 Hz and 0.08% at 1 kHz was measured at a 0 dBm level. The noise floor of the unit sits at about −84 dBm with the input shorted to ground and signals starts to clip at about 8 dBm (1.9 V rms). The bandwidth (−3 dB points) occurs between 8 Hz to 40 kHz.

Signal levels

Due to the companding process, the operation of the noise gate is dependent on the applied signal level, threshold level setting and input gain. Optimum signal levels are between −10 dBm (245 mV rms) and 0 dBm (775 mV rms) which are typical line levels so a good place to insert the unit would be between the pre-amp and the power amp or at the effects send and return of a mixing stage. The noise gate also incorporates an input amplifier that boosts signals over a variable 0 to 18 dB range to give some control on the input levels. The output level control can then be used to attenuate the output to compensate for the additional gain set at the input amplifier. In practice this is done by switching over the bypass switch and adjusting the output control until the levels are equal.

Parts List — ETI-1429

Resistors: – all 1/4 W, 5% unless stated otherwise.

R1, R2	100k
R3, R4	100k 1%
R5	4M7
R6, R12	18k
R7	22k
R8, R9	10k
R10, R14	1k
R11, R21	3k3
R13	2k2
R15	15k
R16	12k
R17	82R
R18	150R
R19	220R
R20	2M7
VR1, VR2	1M log pc mount pot, 24 mm diameter
VR3	5k log pc mount pot, 24 mm diameter

Capacitors

C1	82n greencap
C2	10 u/25V bipolar electro
C3	100p ceramic
C4, C7, C13, C14, C5, C8, C9, C10,	10 u/25V electro
C17, C18	2u2/25V electro
C11, C15	4u7/25V electro
C6	390p ceramic
C12, C16	330n greencap
C19	1000u/25V electro
C20	100u/25V electro
C21	33u/25V electro

Semiconductors

IC1	TL072 dual op-amp
IC2	NE 572 companding ic.
IC3	TL071 op-amp
IC4	7805 5V regulator
D1	1N4004
LED	any 5 mm led will suffice

Miscellaneous

SK1, SK2	6.5 mm Jack socket (contacts open when plug is inserted).
SK3	dc plug pack socket
SW1	DPDT switch (can be toggle or footswitch type). ETI-1429 pc board, 9V dc plugpack at 200mA, suitable metal box, 3 off knobs to fit ports.

Construction

The construction is fairly straightforward as all the electronic components, except for the sockets and switch, are contained on the pc board with dimensions 110 mm × 40 mm. The pc board is designed to accept pc mount pots which simplifies the wiring and the board can be mounted directly to the panel of the box with the pot nuts bolting the pc board down. The choice of box is left to the reader so no enclosure details are given here. However a metal one is preferred for screening purposes. If the noise gate is to be floor mounted then a diecast box is recommended. The unit will have to endure quite a bit of foot bashing, so a heavy duty footswitch should be used for the bypass switch. Otherwise a standard DPDT toggle switch should suffice.

Start by inspecting the pc board for any unetched or broken tracks as some are quite thin and run close together. A finely tipped soldering iron will make life easier. Mount and solder in the links (use some hook-up wire) resistors and capacitors. Take note of the polarity of the electrolytics. Refer to the component overlay.

Next, insert and solder the pc mount pots making sure they sit parallel to the board. Solder in the op-amps, 572, D1 and the 5 V regulator, taking careful note of their correct orientation. The pc board soldering is now finished and all that remains to be done is to wire in the switch, LED and sockets.

Drop the board inside the box and measure and drill out the holes for the 3 pots, sockets, switch and LED. Refer to the overlay and commence with the wiring. It would be easier to mount all the hardware into the box so that the length of the wires can be measured out accurately. Use screened audio cable for the input, output and switch connections and hook-up wire for the dc socket and LED connections. Note that the ground screen is cut off at the switch (SW1) end but joined together and soldered to ground at the pc board. The metal chassis of the box has to be connected to ground and this can be done by bolting a tag to the chassis and soldering a ground connection to the tag. Insulate any exposed contacts if there is any possibility of shorting.

Testing

Before you apply power, sit back and carefully check the pc board for solder splashes across tracks and bad solder joints.

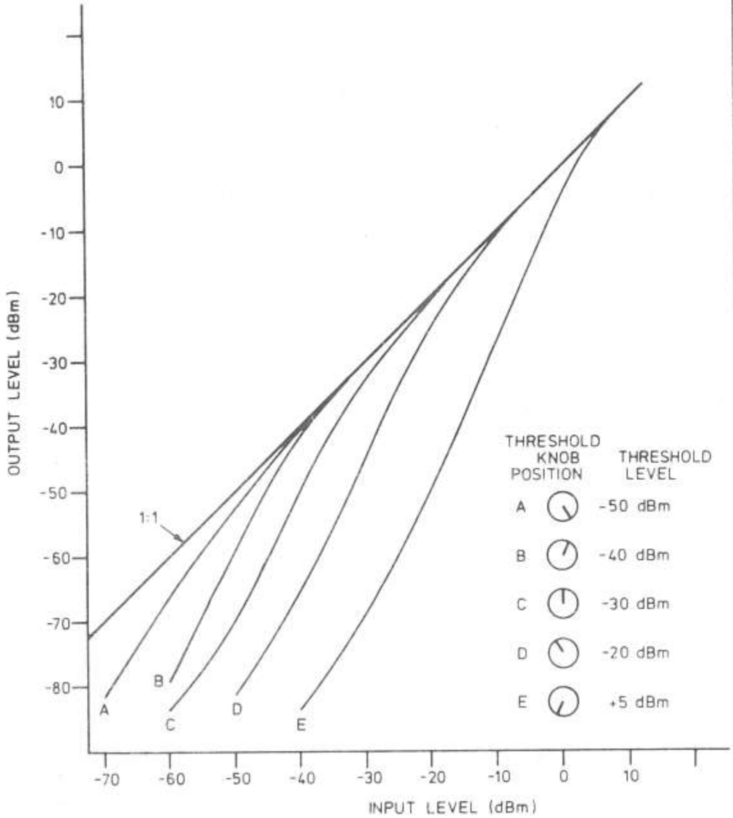
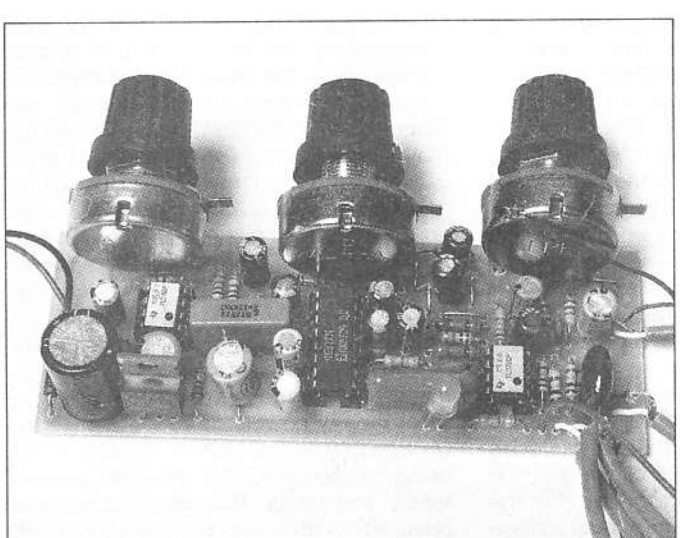


TABLE 2 RESPONSE WITH VARYING THRESHOLDS

Check the wiring as well. Once you are satisfied with your handywork connect in the 9 Vdc plugpack and check that 9 Vdc is present on the power rails with a multimeter. If not then disconnect the power and check that the polarity of the plugpack plug corresponds to the socket. Once the 9 V has been established check that it appears on pin 8 of IC1, pin 16 of IC2 and pin 7 of IC3.



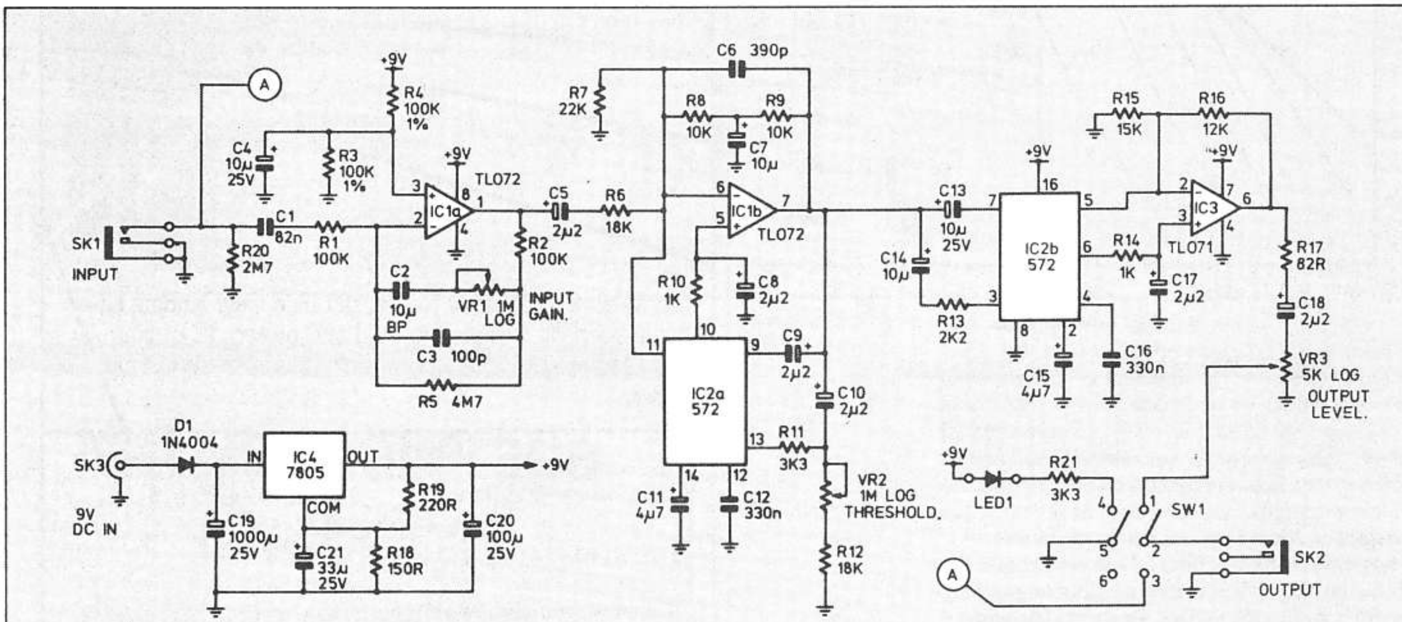
Switch over the bypass switch and the LED should light up in one position. If not, then the legs of the LED are possibly reversed. If the circuit is tested outside the box, connect the metal case of the input control pot and the footswitch to ground, to avoid inducing hum into the audio circuits.

Setting up

The performance of the unit depends on the level of the applied input signal and the settings of the input gain, threshold and output level controls. Because the three controls interact with one another it would be best to follow a sequence to set it up. Start the procedure with the input gain control turned fully anti-clockwise (i.e. unity gain), the threshold turned fully clockwise (i.e. maximum threshold) and the output level control set fully clockwise (i.e. maximum volume). Use this setting as the starting reference point because the output of the noise gate closely follows the input except at input levels below about −50 dBm. The response follows curve A in the graph plotted in table 2. Apply a signal to the input and connect the output to an amplifier. Activate the bypass switch and the levels should be similar; incidentally the LED lights up when the output of the noise gate is selected.

The threshold level should be reduced by rotating the knob slowly anti-clockwise until the hum or hiss is gated out. Test that the main signal is still present and adjust the input and output controls for equal level if it is required. The unit is designed to operate at levels higher than about −10 dBm (245 mV rms) so connect the noise gate at the output of a pre-amplifier or at the end of a chain of effect pedals if the input level is insufficient.

Increasing the input gain should be complemented by reducing the output level control, because the signal including the hum and hiss is boosted by the input amplifier. Be careful with the gain control set to boost input signals and switching between input and output as there can be heaps of gain difference between them; certainly enough to give one's ears quite a fright not to mention overloading power amps and speakers! Avoid just twiddling the knobs and hoping for the best. Always start from the reference setting and work from there. For envelope shaping the threshold level needs to be set fairly high to produce maximum attenuation and thus the most noticeable effect. Happy gating.



How it works

IC1a is connected as an inverting amplifier with a gain range of 0 dB to 18 dB, made variable by VR1. The gain resistors R2 and VR1 are ac coupled by C2 to reduce pot noise due to dc currents in the feedback path. Resistor R5 provides the necessary dc feedback. Input impedance is set to 100 k by R1 and is ac coupled by C1. R4 and R5 biases the non-inverting input to 4.5 Volts.

IC1b and IC2a comprise the compressor with the variable gain cell of the 572 placed in the feedback path of the op-amp. Pin 7 of the op-amp also feeds the output of the compressor to the rectifier input of the 572 via pin 13. The resistor network VR2 and R12 sets the threshold where the compressor deviates to a 1:1 compression ratio. The attack and release time is determined by C12 and C11 together with internal 10 k resistors and are set to 3.3 mS and 47 mS respectively. Resistors R8 and R9 introduce dc feedback and C7 provides a low impedance path for ac. The non-inverting input (pin 5) of IC1b is biased to 2.5 V which is derived from an internal reference voltage. R7 sets the output dc level of the op-amp to 4.5 V for maximum swing.

The expander circuit consists of IC2b and IC3. The variable gain cell of the 572 is placed in the inverting input of the op-amp and acts as a variable input resistor which varies the gain depending on the input signal. The rectifier provides the control signal and derives its input from the output of the compressor via C14 and R13. The resistor R16 and the op-amp IC3 converts the output current of the gain cell to the output voltage. The non-inverting input (pin 3) of the op-amp (IC3) is biased at the internal reference voltage of 2.5 V via pin 6 of the 572. The resistor R15 sets the dc output level of the pre-amp to 4.5V to allow maximum voltage swing.

The attack and release time of the expander is set to 3.3 mS and 47 mS respectively by C16 and C15 and internal 10 k resistors. The output is attenuated by the output level control (RV3) before it is fed to the bypass switch. The bypass LED and associated current limiting resistor R21 is switched into circuit and lights up when the noise gate is selected. R20 discharges any floating voltages on C1 and reduces switching noise when the bypass switch is activated.

A 9V regulated power rail is derived from the 7805 regulator (IC4) and R19 and R18 biases the common terminal to 4 V which raises the output voltage to 9 V. The diode D1 protects the circuit from incorrect polarity being applied and C19 provides power supply smoothing. The circuit consumes about 10 mA dc current plus the LED current.

OP-AMP POWER SUPPLY

Robert Irwin

An ideal supply for op-amp experimenters and those with solderless breadboards. The ETI-251 provides ±12 V rails at 1 A and solves those 'split rail blues'.

A DUAL RAIL supply is a handy piece of equipment for anyone who is even thinking of playing around with analogue ICs. The ETI-251 is a simple, easy-to-build, low cost supply that will be ideal for breadboarding up circuits which require single or split 12 V rails. The ETI-251 provides regulated positive and negative 12 V rails and can supply up to 0.5 A from each. An overload LED on each rail gives a visual indication when you try to draw too much current from the supply. All the components used are very common and most could probably be found in the average hobbyist's 'bits-and-pieces' draw. The supply is relatively easy to build and should be suitable for even inexperienced constructors, although not recommended as a very first project. The construction section has been made very detailed to accommodate any beginners who wish to build this supply.

Design details

The circuit is designed around the very widely used LM7XXX series of three terminal regulators. The LM7812 and LM7912 provide +12 V and −12 V respectively. Both ICs have built in short circuit foldback current limiting and thermal protection and are therefore very hardy devices. As well as the internal protection built in to the regulator ICs, several external protection diodes are included in the circuit to guard against any accidental load faults that may otherwise destroy the regulators.

The transformer used is a widely avail-

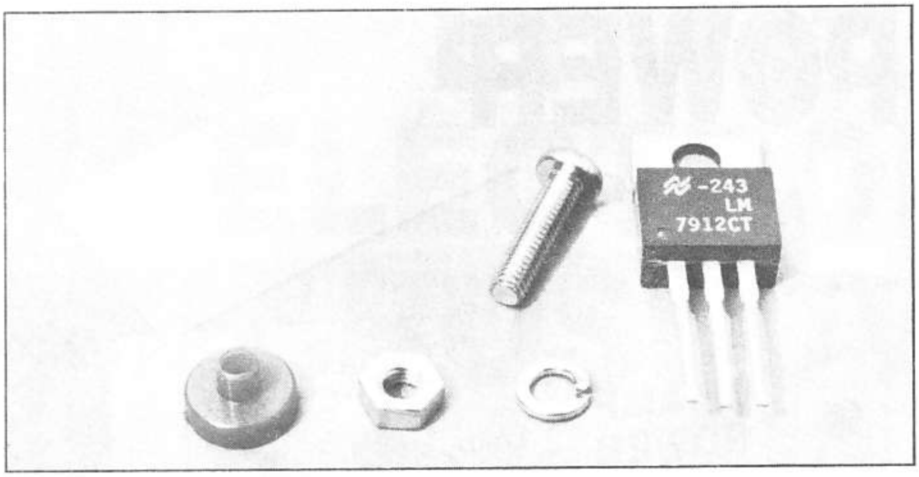
HOW IT WORKS — ETI-251

The circuit is very simple as the three terminal regulators are basically autonomous, requiring no external circuitry to make them work. Protection from unusual load conditions is needed though. Referring to the circuit diagram, the transformer output is 30 Vac with a centre tap. This gives two 15 Vac signals which are rectified by a D1, 2, 3, 4 which form a full wave bridge rectifier. This produces both a positive and a negative rectified output with reference to the centre tap. The rectified signals are smoothed by C1 and C2 which, with the values chosen, will give a peak output of 21 V and a ripple of about 4 V p-p at 1 A output.

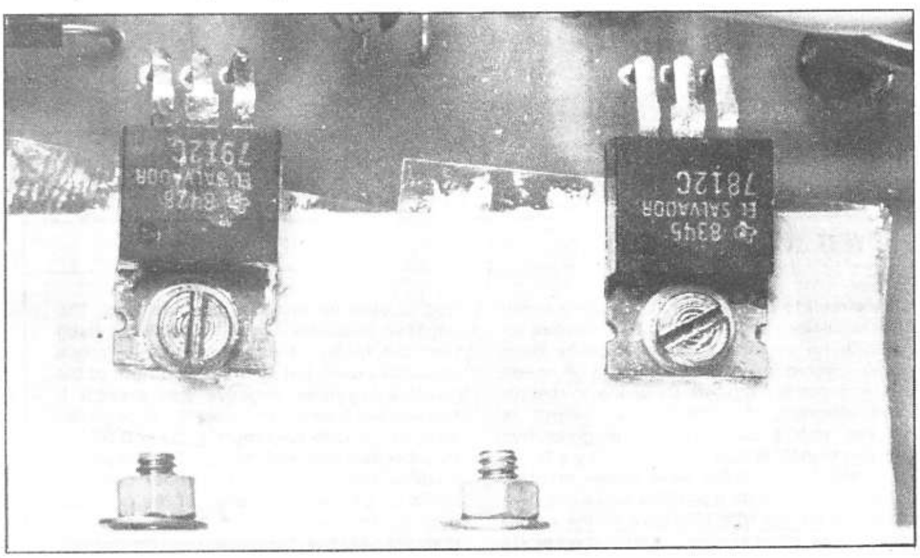
Before going to the regulators, the current is monitored by the overcurrent circuitry. Both the positive and negative circuits are identical (except for the direction of current flow) so we will just look at the positive overcurrent circuit. R1 is in series with the supply current and will develop a voltage across it which is given by Ohm's law, V=IR. The emitter of Q1 is connected to the supply side of R1 and the base is connected to the load side. When the load current reaches 0.6 amps, 0.6 V is developed between the base and emitter of Q1 and it begins to turn on. This will allow current to flow through LED2 which will cause it to light and indicate an overcurrent condition. R5 limits the current through the LED to about 15 mA.

IC1 is a positive regulator which takes the unregulated input and gives a regulated +12 V output. IC2 does the same for the negative side and gives a regulated −12 V output. One problem that sometimes occurs when using regulators in a split rail supply is that of start up under a common load. The negative regulator tends to establish itself first and, under a shared load (particularly a capacitive one), it may drag the output of the positive regulator negative and prevent it starting up. It may also cause the IC to be destroyed. To help prevent this, R3 and D7 are incorporated into the design. This helps the positive regulator start up under common loads by providing isolation of the common pins on the two ICs and, if the output is dragged negative, the current can be shunted by D7 and allow the positive regulator to establish itself. R4 is included in the circuit to maintain a voltage balance between IC1 and IC2. Without R4, the quiescent current in the common terminal (about 6 mA) would raise the output voltage on the positive regulator slightly.

As an added precaution against destruction of the regulators, D8 and D9 prevent any reverse polarity voltages from developing on the outputs of the regulators. D5 and D6 will protect the regulators from any overvoltages on the outputs which may occur when reactive loads are being driven. C6 and C7 are not crucial but provide some filtering to the input of the regulators. C3 and C4 improve the transient response of the regulators and prevent high frequency instabilities. C5 ac couples the power supply earth to the chassis which allows the metal case to act as an electrostatic shield and prevent any rf inteference in the supply. There should be no dc connection between chassis and power supply earth. LED1 and R7 provide a power on indication.



Above: IC mounting hardware. This pic shows the various bits and pieces you will need to mount each regulator. The mica washer is at the top left, and the insulating washer is at the bottom left.
Below: The regulators mounted on the heatsink bracket. The 'messy white stuff' all over the bracket is the thermal grease used to provide good heat transfer.



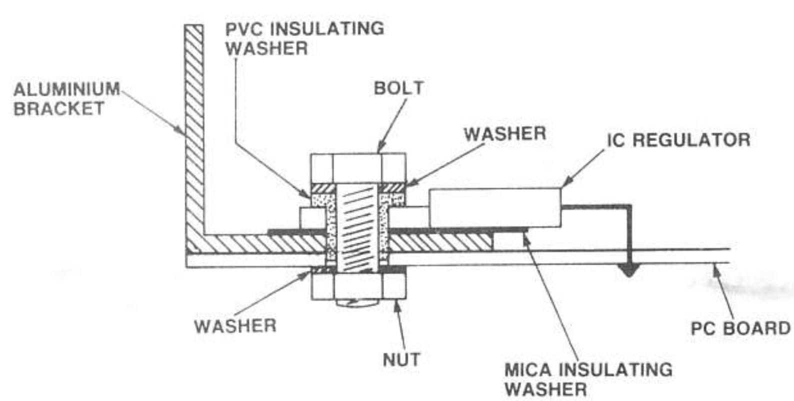


Figure 1. When mounting the IC regulators to the heatsink bracket, make sure the tabs on the regulators are electrically insulated from one another. This is done by placing a PVC insulating washer around the bolt and a mica insulating washer between the IC and the bracket.

able multi-tapped secondary type which provides 15 V and 30 V taps and is rated at 30 VA maximum. The main reason for the choice of this transformer is that it is cheap and easy to get. It should be noted however, that under a direct short circuit between the positive and negative terminals, the output current will be just over 2 A. The transformer will handle this sort of overload for quite a few minutes without damage but may heat up if the short is left for long periods. This will not be a problem in normal operation but if the supply were, for example, to be used to power a circuit which was to be left running overnight, it would be a good idea to use a transformer with a higher output current rating so that any sustained short that may occur will not thermally stress the transformer. A short from either the positive or negative to ground will only cause about an amp to flow and can be handled indefinitely by the specified transformer. A PL30/60 VA is an ideal substitute but is quite expensive. For most applications, though, the specified transformer will be more than adequate.

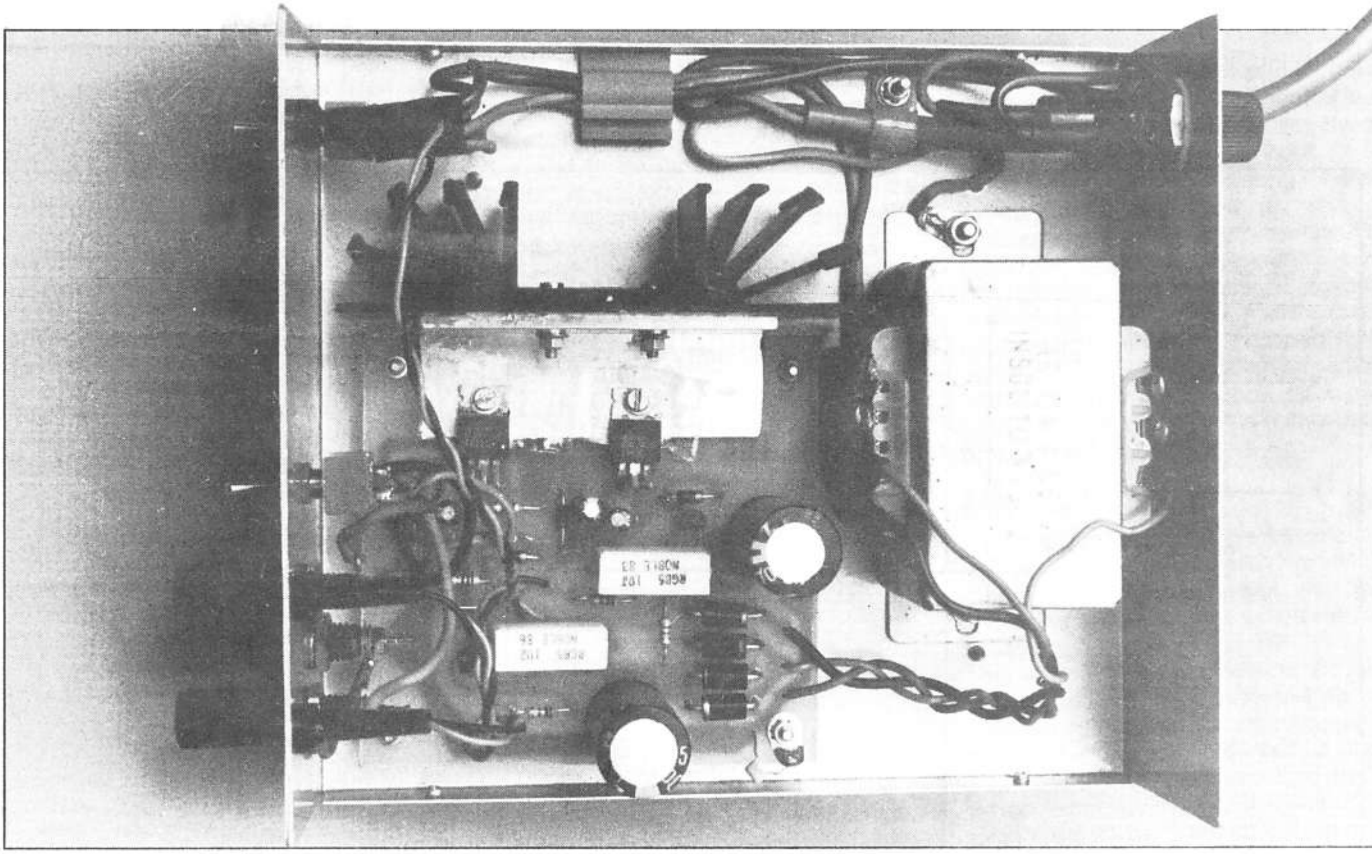
Incorporated into the design is an overload indication for each rail. This is set to indicate that a current of 0.6 A or more is being drawn from the rail. This overload in no way damages the supply but is there to indicate excessive current drain which may be cause to investigate the circuit you are powering for shorts.

Construction

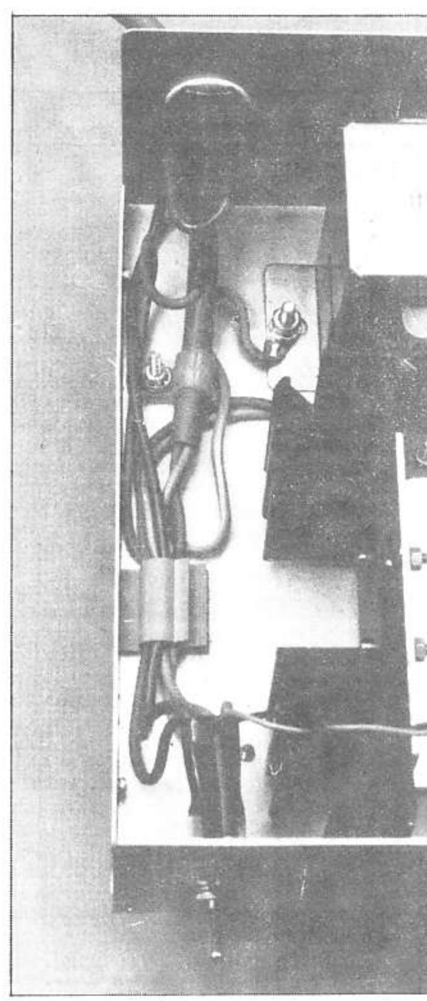
The construction of the power supply presents no real problems. However, I will describe the construction in detail for those who may not be as familiar with a soldering iron as others.

It is recommended that the circuit be constructed on the ETI pc board. If you wish (and you know what you're doing) you can, of course, use Veroboard or the like but using the pc board will greatly simplify construction and minimize the chance of a wiring error. Having said that, once you get a pc board check it very carefully for faults on the copper side. The most common faults are broken or shorted tracks caused by problems in the etching stage. If a track is thin, then over-etching or faults in the resist can cause the track to be etched away in parts and thus be open circuit, so check all thin tracks for breaks. Where two tracks come close together, under-etching or dust on the negative can cause the copper between the tracks to remain and thus cause a short so check all points where tracks come close together. Finally, check that all the holes have been drilled. Once you are satisfied that the pc board is in good shape, you can move on to the soldering in of the components.

Referring to the overlay diagram, locate the position of the wire link. This link



Above: The general layout inside the case. Make sure that the transformer and heatsink are clear of one another.



Right: The mains wiring is run down the left hand side of the box. Try to keep it neat and insulate all connectors on the fuse and switch so that you can't accidentally brush against a live terminal.

should be made on the pc board with a piece of tinned copper wire (a discarded piece of component lead is ideal). Once this is in position locate and solder in all the resistors. R1 and R2 are high power resistors and in the course of normal operation may get quite hot. To help cooling and to prevent scorching the board these two resistors should be mounted so that they stand off the board by about a millimetre or so. The parts list specifies 1 ohm, 5 watt resistors for R1 and R2 but two 2.2 ohm, 1 watt resistors in parallel can be substituted for each resistor if you wish. There are extra holes on the board to allow for this.

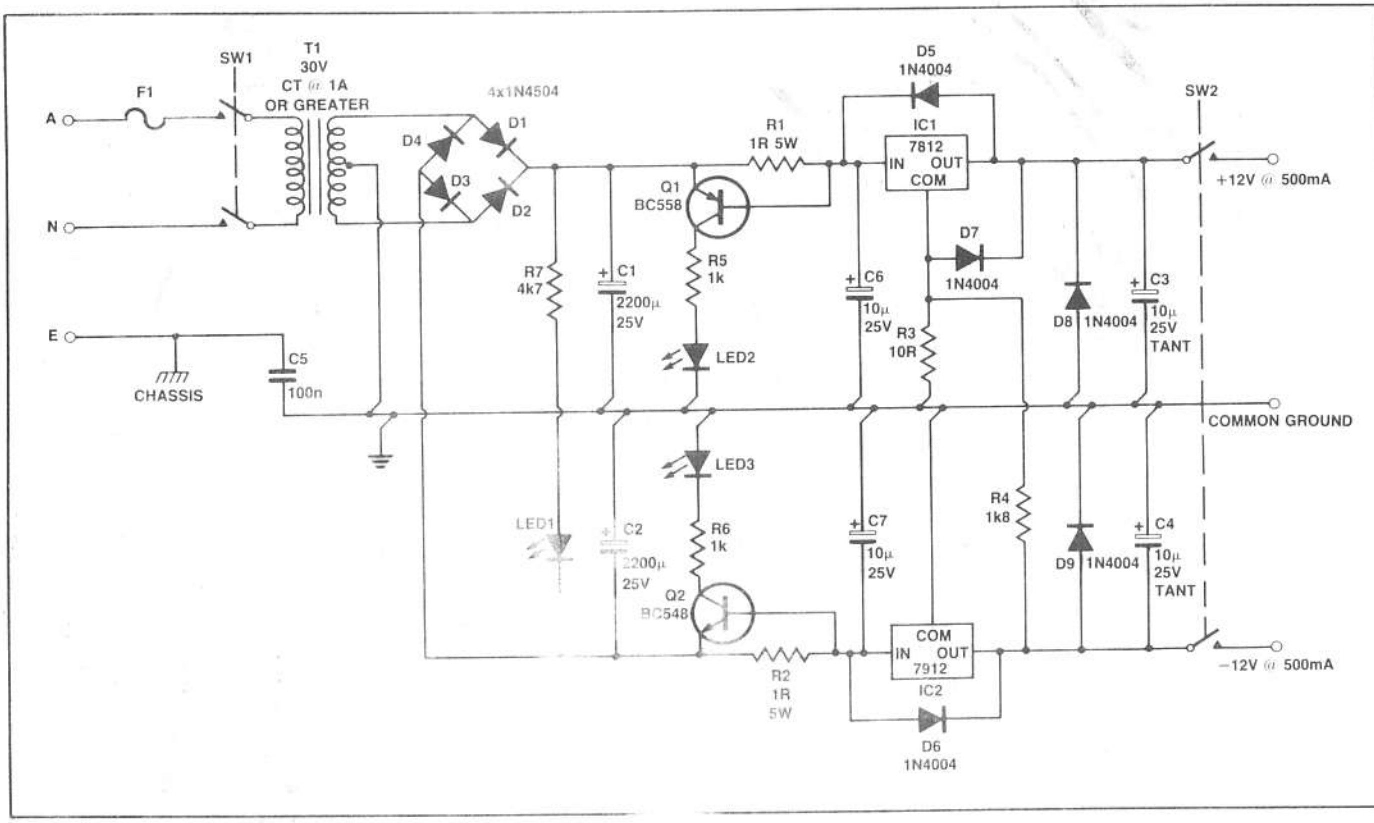
Next, locate and solder in all the capacitors. Take very special note of the way these are put in as they are polarized and may be destroyed rather spectacularly if they are put in the wrong way round. Note that C5 is only soldered to the board at one end. The other lead will be bolted to the pc board mounting bolt at a later stage. For the moment just leave it dangling over the side.

The semiconductors can be soldered next. Start with the diodes. Mount the large rectifier diodes off the board as you did with the power resistors as these too may get hot. Once again pay attention to the way the diodes go in as they are also polarized components.

The two transistors can now be soldered in. The only remaining components to mount are the IC regulators. These will be a bit of a problem in that they mount on an aluminium heatsinking bracket but must be electrically isolated from one another. The first thing to do is to prepare the bracket.

Cut a 70 mm length of 1 inch aluminium angle. Study the photographs and drawings and position the angle at the edge of the board so that it sits between the pc board mounting holes on the component side. The side should point upwards from the board. Use a felt pen or scribe to mark the position of the two mounting holes for the ICs. These holes should be drilled large enough to fit the IC mounting bolts. The heatsink (a 35 mm length of radial fin type) mounts vertically on the side of the aluminium angle. This should be positioned against the angle and the centres for two holes to mount the heatsink should be marked. It is best to drill the holes in the heatsink first and then use these to mark and drill the holes in the angle. This way they'll line up. All the holes in both the bracket and heatsink should be carefully de-burred and the edges made smooth. Thermal conduction to the heatsink is dependent on how well contact is made between the two surfaces. If the edge of the hole is rough it may prevent the two surfaces from contacting each other properly.

To mount the two ICs examine Figure 1 carefully. Position the bracket and lay the ICs on it. Bend the legs of the ICs in the appropriate place and push them through their mounting holes. Take a TO220 package mica mounting washer and thinly coat it on both sides with a layer of thermally conduc-



PARTS LIST — ETI-251

Resistors..................all ¼ W, 5% unless noted
- R1, 2......................1R0, 5 watt
- R3..........................10R
- R4..........................1k8
- R5, 6......................1k
- R7..........................4k7

Capacitors
- C1, 2......................2200µ, 25 V, RB electro
- C3, 4, 6, 7..............10µ, 25 V tantalum
- C5..........................100n ceramic bypass

Semiconductors
- IC1.........................LM7812
- IC2.........................LM7912
- D1, 2, 3, 4..............1N5404
- D5, 6, 7, 8, 9..........1N4004
- LED1, 2, 3..............5 mm red LED

Miscellaneous
- T1..........................30 V CT, 1 A sec (Arista type A-6672 or similar)
- SW1, 2...................DPDT toggle
- F1..........................500 mA 2AG fuse

Fuse holder; 185x70x160 mm metal cabinet (DS-H-2744 or similar); mains flex and plug; mains clamping grommet; 3 x 5 mm LED mounting grommets; ETI-251 pc board; Scotchcal front panel; 70 mm length of 1 inch aluminium angle; 2 x TO220 mounting kits (mica washer, insulating washer and nuts and bolts); 3 x 4 mm banana socket binding posts (red, black and green); 35 mm length of radial fin heatsink; heavy duty hookup wire; 4 x 6 mm stand-off pc board spacers; 4 x pc board mounting nuts and bolts; 4 x 15 mm 4BA nuts and bolts; 2 x solder lugs; silicon thermal grease.

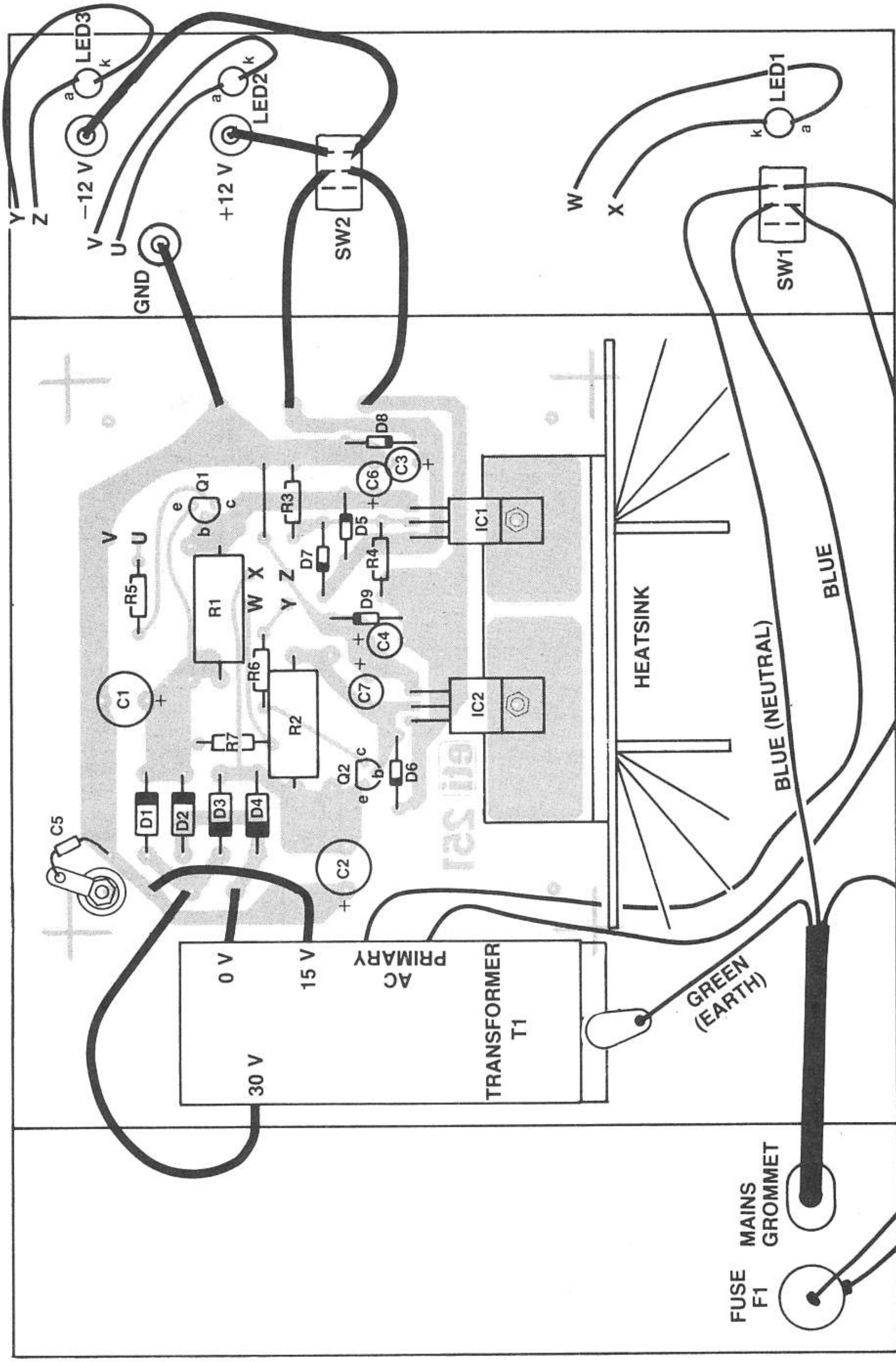
Price estimate: $45-$50

tive silicon grease. This will ensure good contact from the IC to the bracket. Put the mica washer in place over one of the mounting holes on the bracket and position the hole in the IC over the mounting hole. The bolt and washers can then be put through the hole and tightened up. Note that the bolt is isolated from the metal tab on the IC by a PVC insulating washer specially made for the TO220 package. Repeat the procedure for the other regulator.

To test whether the regulators are isolated from the heatsink, measure the resistance from the centre leg of one IC to the center leg of the other with a multimeter. You should get a high resistance (open circuit) reading. If not then take the ICs off and try again with fresh mica washers. After the ICs have been mounted correctly you can solder the legs to the pc board. The heatsink can now be mounted to the bracket. Use silicon grease to ensure good contact and screw the bolts up fairly tightly.

The next step is to solder flying leads to the input, output and LED mounting points on the pc board. These should be colour coded so you can easily identify the positive, negative or ground when you come to wire the LEDs or terminals. Light hookup wire is adequate for the LEDs but heavier wire (5 amp or more) should be used on the ac input and dc output points. If the pc board is placed temporarily in position in the case, you can get an idea of the length of wire needed. Always allow a bit more length than you think you'll need as the wire can be trimmed later. Once all the wires are attached the pc board is completed and you can turn your attention to preparing the case.

The prototype was housed in an inexpensive aluminium instrument case (see parts list for details). Take the case apart and don't lose the screws. If you examine the photographs you can get an idea of the general layout of the inside of the case. Looking from the front, the pc board is mounted on the right-hand side at the front. The transformer mounts directly behind this. The mains cord and wiring is all on the left-hand side. Position the transformer and pc board in the box and, after you have ensured that there will be no possible shorting between transformer and the case or the transformer and pc board, mark the positions of the mounting holes for the transformer and board, and drill the holes to fit a 6BA bolt. Holes for the mains cord grommet and the fuse should be marked and drilled in the back panel. If necessary, drill a hole in the floor of the box for a mains cord clamp. To mark out the front panel you can either use the drilling diagram or you can use the front panel artwork as a template and centrepunch the holes from this. Drill the front panel holes to the appropriate sizes. Remove any burrs from the holes and smooth any rough edges.

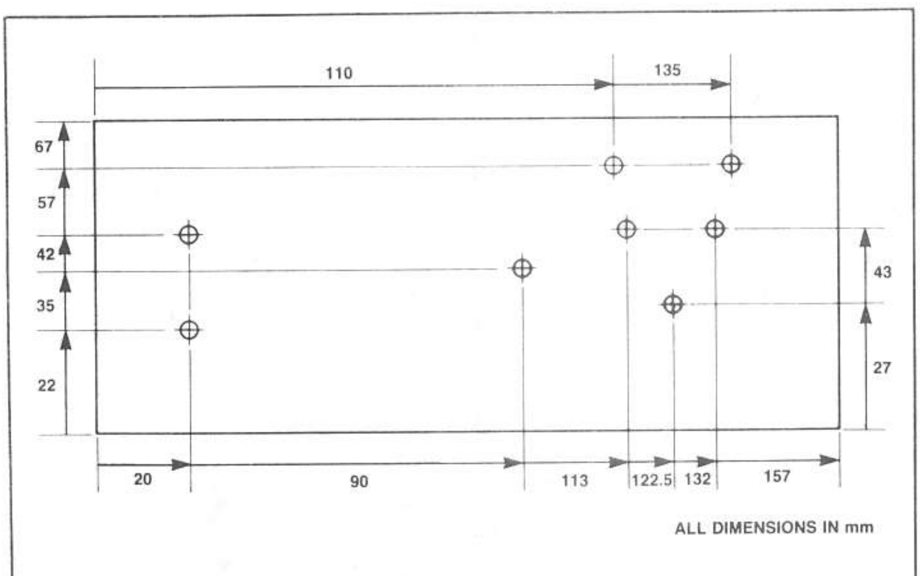

LED3
LED2
LED1
−12 V
+12 V
GND
SW2
SW1
Y
Z
V
U
W
X
a
k
HEATSINK
IC1
IC2
Q1
Q2
R1
R2
R3
R4
R5
R6
R7
C1
C2
C3
C4
C5
C6
C7
D1
D2
D3
D4
D5
D6
D7
D8
D9
TRANSFORMER
T1
0 V
15 V
30 V
AC
PRIMARY
GREEN
(EARTH)
BLUE
BROWN
BLUE (NEUTRAL)
BROWN
(ACTIVE)
MAINS
GROMMET
FUSE
F1

The next step is to attach the Scotchcal label to the front panel if it is needed. Firstly drill small pilot holes at the centres of all the mounting holes. Do a trial fit to see how the pilot holes line up with the holes in the front panel. If they are OK then peel off the backing paper and place the Scotchcal sticky side up on the bench. The front panel can now be carefully positioned above the Scotchcal. Be careful not to touch the Scotchcal as it will stick fast and is hard to get off again. When you are satisfied that the front panel is lined up, carefully lower it onto the Scotchcal and press down. The Scotchcal should now be stuck to the front panel. The holes can be cut out with a sharp knife or scalpel and the edges trimmed. Be careful not to tear the Scotchcal when doing this. Screw on the four rubber feet and the case is now ready.

The next step is to do the mains wiring. Be *very* careful with this as mains voltages are *lethal*. Try to keep all mains wiring neat and tidy and out of the way of everything else. Strip 200 mm of the outer insulation off a length of mains flex. Thread this through the hole in the back panel and clamp it with a clamping grommet. Make sure it is secure and will not pull out. If necessary use a screw-on mains cord clamp as well. Mount the fuse holder on the back panel and the mains switch on the front panel. Following the wiring diagram carefully, wire up the switch and fuse. All exposed terminals and joins in the mains wiring should be well insulated so that they cannot be accidentally touched. This is best done with heatshrink tubing which is placed over the wire before soldering. When the wire is soldered in place the heatshrink is pushed up over the join till it covers the exposed area. The heatshrink is then heated with a hairdryer or soldering iron and it will shrink to form a tight seal over the join. This should be done for all the connections on the mains switch and fuse holder. As added protection, insulating tape can be wrapped round the entire switch and fuse assemblies.

The transformer should now be mounted using 4BA nuts and bolts. Note that the earth wire of the mains bolts to the transformer mounting bolt. The earth wire should be long enough to ensure that if the mains cord is pulled out of the box the earth will be the last to break. A solder lug should be soldered on to the end of the earth to ensure that it mounts securely. The transformer primary can now be wired up. Once again heatshrink should be used to insulate the terminals. On the transformer specified the primary connections are at the bottom. This can be a bit awkward to get at so be careful and make very sure that the terminal joins do not short on the floor of the box and are well insulated. This, then, completes the mains wiring. Double check that everything is correct and that no connections are exposed.

The pc board should be mounted next. The mounting bolts should be put through the floor of the box and secured with a nut (see Figure 2). The board then mounts on 6 mm standoff spacers and is bolted down. Solder a lug on the remaining lead on the 100n bypass cap and bolt it to the nearest pc board mounting bolt. The switches, terminal posts and LEDs should now be mounted on the front panel. These can be wired up according to the wiring diagram. When wiring the LEDs be sure to get the wires the correct way round. Finally, the transformer secondary can be wired to the board. A 500 mA fuse should be fitted to the fuse holder and a mains plug (if not already fitted) should be wired to the mains flex (be sure to get the connections right). You are now ready to test the supply out.



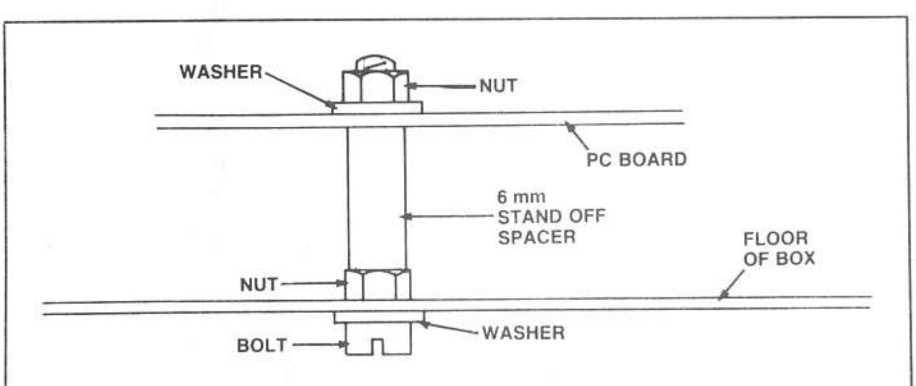
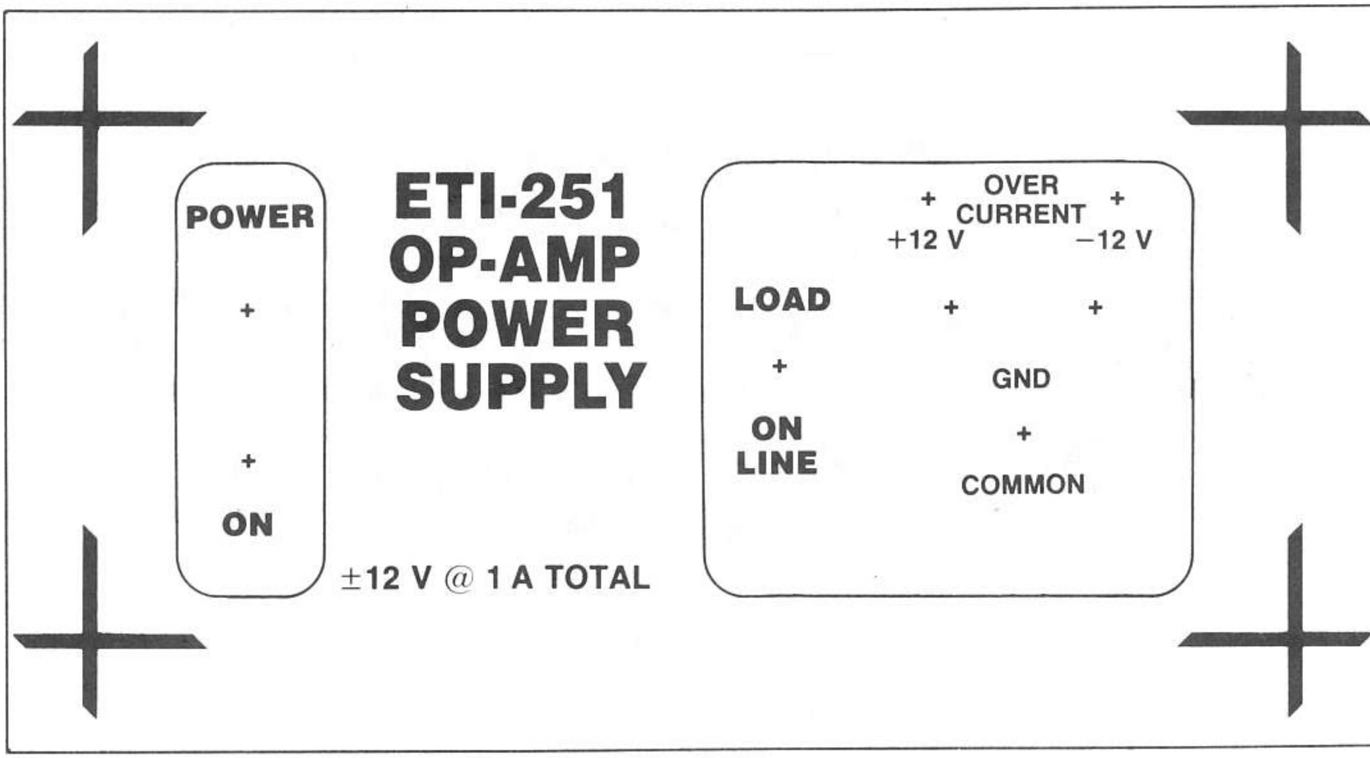


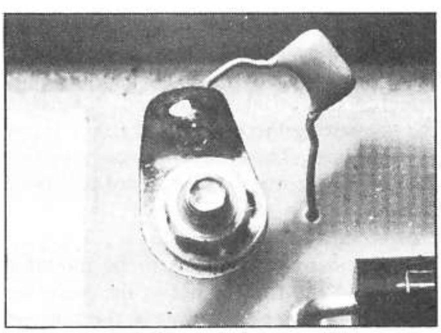
Figure 2. Mounting the pc board.

Testing and using it

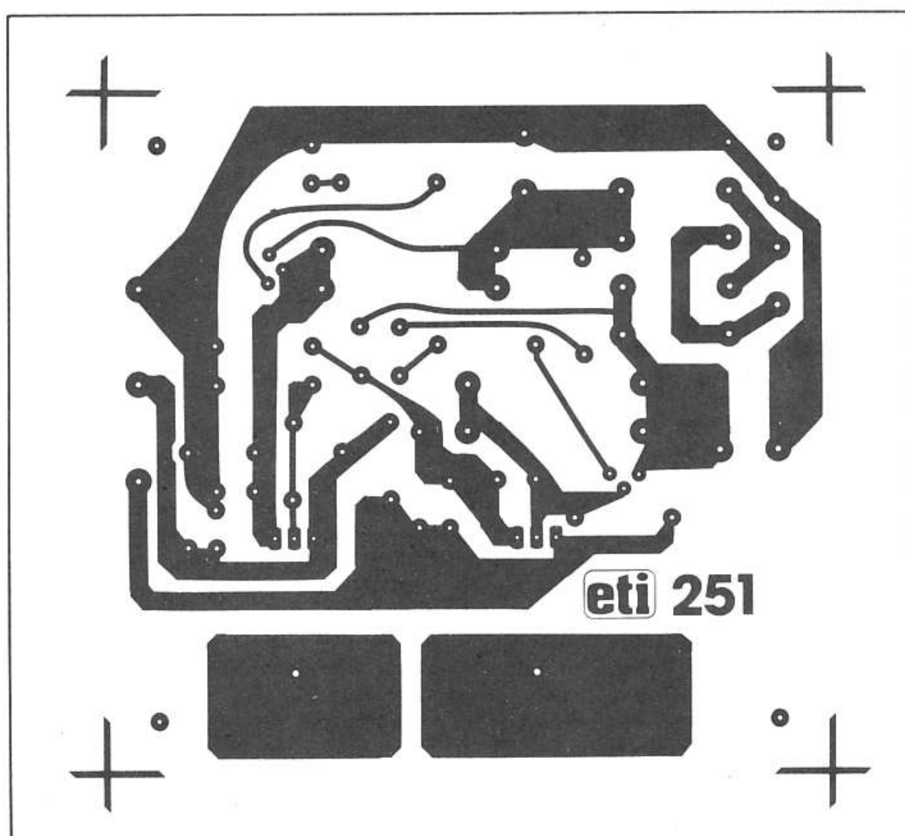
Plug the supply into a mains socket. With the LOAD switch off switch the power on. The power indicator LED should glow and nothing else should happen. With nothing connected to the output terminals, switch the LOAD switch to the ON LINE position. Measure the voltage between the ground and positive terminals. It should read +12 Vdc. Do the same between ground and the negative terminal. It should read −12 Vdc. If this is not the case then unplug, and recheck the wiring and pc board.

If all is well then you can apply a load to the output. Ideally, an 18 ohm, 10 watt power resistor should be wired between the positive terminal and ground. This will draw about 700 mA from the supply and when the supply is turned on with the LOAD switch in the ON LINE position the overcurrent LED above the +12 V terminal should light. If the load is wired across the negative and ground the LED above the −12 V terminal should light. You can accurately determine the current that the LEDs switch at if you have a high power variable resistor or rheostat. This can be





A solder lug should be used to bolt the 100n bypass cap to the chassis. This allows the case to act as an rf shield but still provides for a floating supply.



varied until the LED just lights. The resistance can then be measured and the current determined.

If the supply tests out OK so far, then connect a piece of heavy duty hookup wire between the positive and ground, and switch on. The supply should power up with the +12 V overcurrent LED lit. Do the same for the negative terminal. Finally, do the same thing with a piece of wire between the positive and negative terminals. Both LEDs should light this time. If all is well then switch off, unplug and put the lid back on. You are now ready to use the supply.

The only thing on the front panel that may require a few words is the LOAD switch. This merely disconnects the output terminals from the supply. This was done so that on power up and power down the load can be disconnected from the supply and thus be unaffected by any transients when the regulators power up or down. This is particularly useful when making changes to a breadboard for instance. If the main power switch were used to turn the supply off then you would have to wait a couple of seconds for the capacitors to discharge and the voltage to go to zero before altering the circuit. Using the LOAD switch, the circuit is instantly disconnected from the supply.

If you are powering a circuit and the overload LED comes on you should switch off. Although the supply can maintain overloads for quite long periods it is wise not to run the supply continuously in this mode as thermal stressing of the components can occur which may eventually lead to failure.

ETI 1424 VERSATILE GUITAR PRE-AMPLIFIER

Guitar players always place over the top demands on their gear. This guitar pre-amp certainly delivers over the top specifications.

Terry Kee

COMMERCIAL GUITAR AMPLIFIERS do not appear to be getting any cheaper so a good low cost alternative is to build your own. Power amplifier modules are commonly available from your local kit suppliers at very reasonable prices with excellent performance, particularly those published in this very magazine.

The ETI-1424 is intended to provide high quality pre-amplification especially tailored for the electric guitar. The equalizer sections are optimised for the frequency range where the guitar needs them most.

The main features of the pre-amp include a top boost and normal input for guitars, two pre-eq line inputs, bass and treble controls, effects send and return, a sweep eq section, four post-eq line inputs and a master level control. No level control is provided for the line inputs as typical inputs would be drum machines and synthesisers that have their own individual volume controls. This set-up is designed for the all too common situation where there are insufficient amplifier inputs for all the instruments. More often than not, this happens in a rehearsal situation.

The bass and treble are designed to provide maximum cut and boost of frequencies at 100 Hz and 8 kHz to obtain a wide tonal range for an electric guitar. To give a harder edge to sounds, a top boost input is available whereby frequencies above 1 kHz are amplified; at around 10 kHz there is a massive 20 dB of boost! The normal input has a flat response and is excellent for those mellow rhythm chords. A bass cut is built into the input amplifiers IC1a and b as in a live set-up, very low frequencies combined with speaker cabinet resonances tend to muddy the overall guitar sounds, not to mention setting off the cymbols at some resonant frequency! The guitar inputs have a fairly high input impedance of 220 k to ensure that the pick-ups are not loaded and thus obtain maximum sustain. Due to the high input impedance hum pick-up can be a problem, so the jack sockets are wired in such a way that any unused inputs are shorted to ground. No casing details will be described here as it is likely that the pre-amp will be built into the box that houses the power amplifier. A metal box is recommended to minimise hum pick-up.

Many of the facilities of the 1424 can be tailored to your requirements. If you decide that you do not want any pre or post eq line inputs, as an example, then it is simply a matter of linking the relevant inputs to ground on the pc board. A similar procedure applies if you do not want an effects send and receive, simply link the effect send out pad to the effect return pad with some hook-up wire.

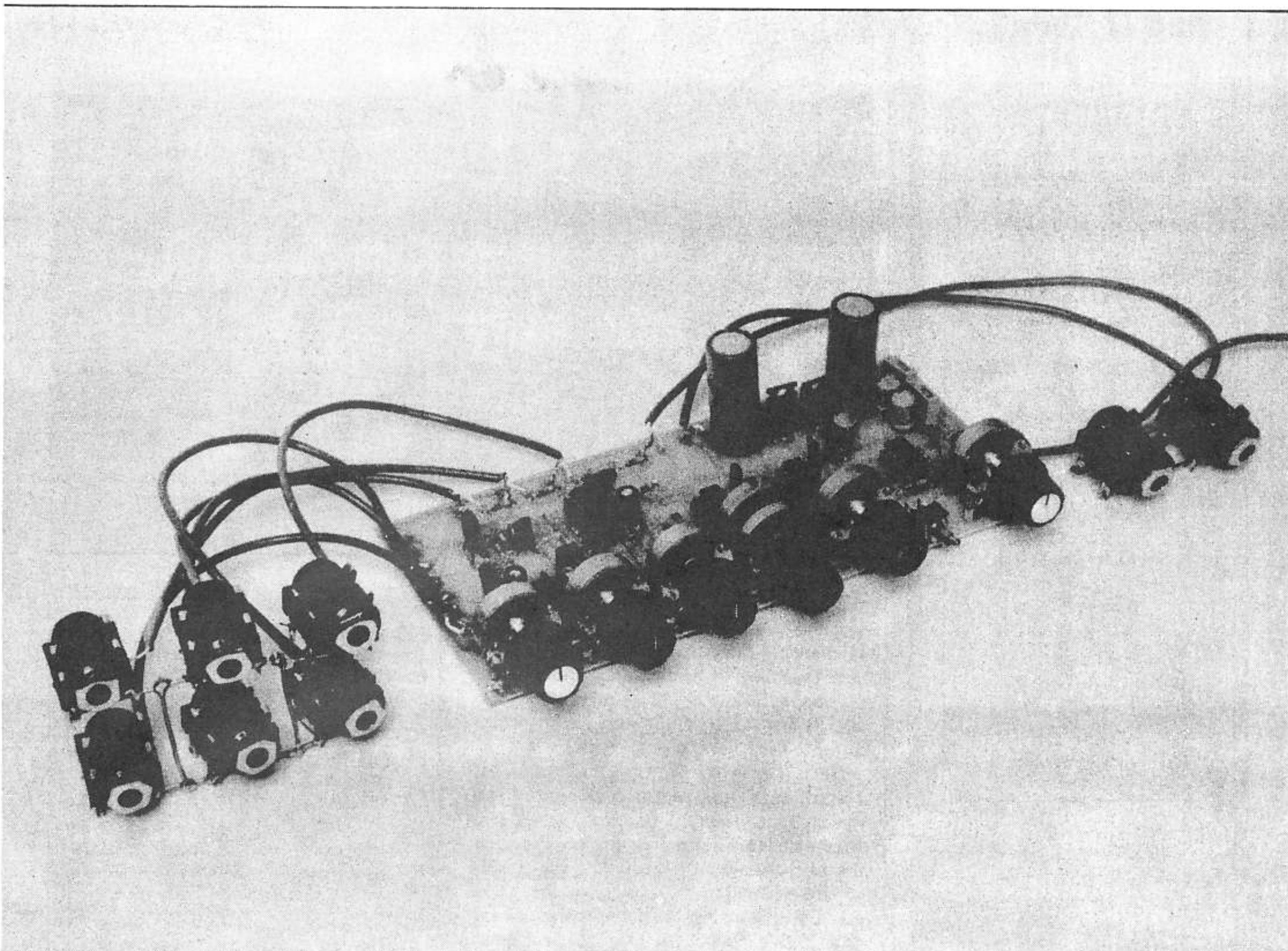
The effects return is fed to a sweep equaliser that has an adjustable frequency and gain control. The circuit is a modified version of the parametric equaliser that appeared in an earlier ETI. A bandpass or bandstop type of response is exhibited with the resonant frequency made adjustable over an extremely wide range of 200 Hz to 8.5 kHz using a single control. The sweep eq supplies a massive 18 dB of boost and −22 dB of cut at the resonant frequency and can be adjusted by the gain control. The Q-factor is fixed and gave good tonal variations with a guitar signal. A bypass switch is included to switch the effect in or out when required. It's useful for pre-setting the eq for that funky toppy rhythm lick! The 1424 derives it's +/− 12 Vdc power from it's on board regulated power supply.

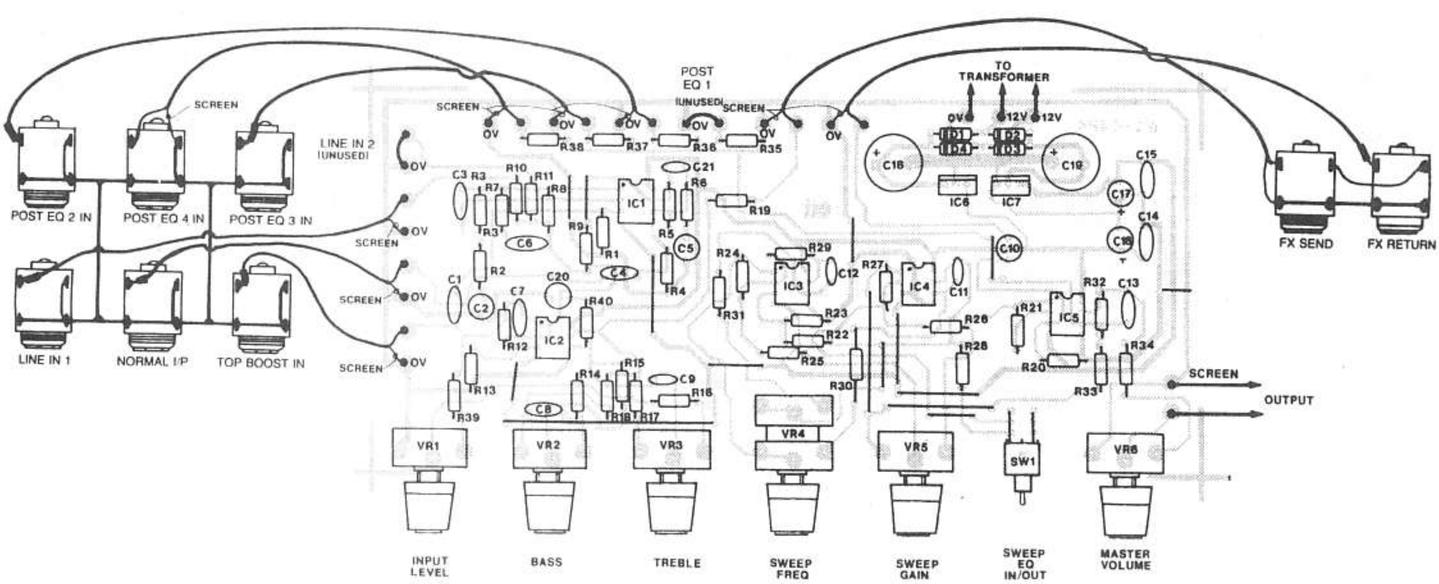
Construction

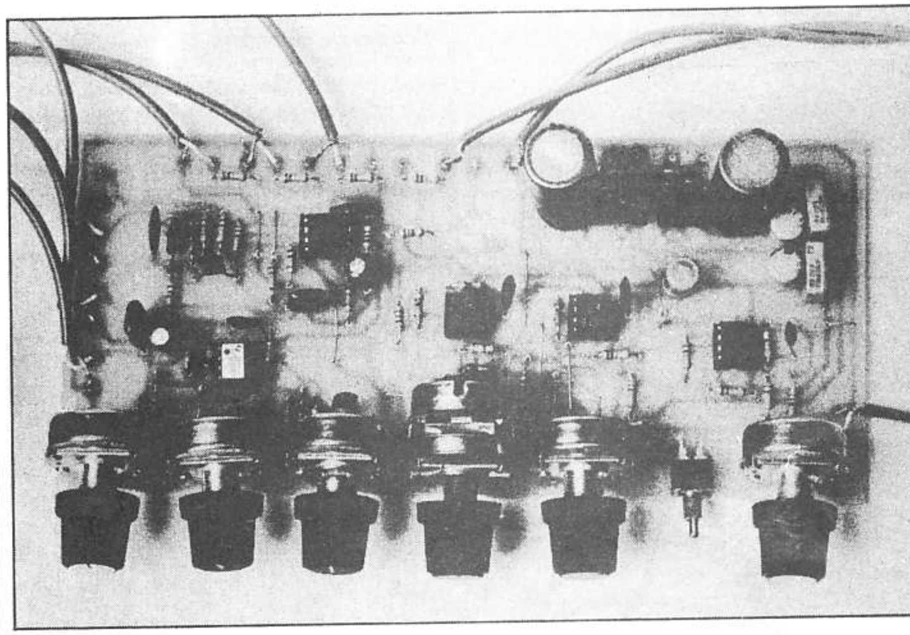
The pc board is designed to be mounted directly to the front panel of the amplifier box and is fastened down via the nuts on the pots. If you do not want to use pc mount pots and pc mount switches then use the shortest length of hook-up wire to make the connections to these components to minimise hum and pick-up. You will also have to drill some mounting holes on the pc board. Make sure that the holes do not break any tracks.

Building up the pc board should not present any problems as the entire circuit is contained on one single-sided board. The first task is to check the board for track breakage and shorts. It's a good idea to go through this process even though you may have obtained the board from a kit supplier. No-one is perfect! Faults will be much easier to spot now than when the board is populated with components. Once you are satisfied it's time to drill the component holes, if it is an undrilled board. Make sure that the holes for the pc mount pots are large enough, a 2 mm drill bit should be adequate.

Construction can start by inserting the links, resistors, capacitors, and ic sockets and soldering them in. Take note of the polarity of the electrolytic caps, refer to




SCREEN
POST EQ 2 IN
POST EQ 4 IN
POST EQ 3 IN
LINE IN 1
NORMAL I/P
TOP BOOST IN
LINE IN 2 (UNUSED)
POST EQ 1 (UNUSED)
TO TRANSFORMER
0V
12V
12V
FX SEND
FX RETURN
OUTPUT
VR1
VR2
VR3
VR4
VR5
SW1
VR6
INPUT LEVEL
BASS
TREBLE
SWEEP FREQ
SWEEP GAIN
SWEEP EQ IN/OUT
MASTER VOLUME



the component overlay. Do not as yet insert the IC's themselves. Insert and solder in the pc mount pots and switch, making sure that they sit firmly and parallel to the pc board. The +/− 12 V regulators (IC6 and 7) are the final components to solder in, note the orientation on the overlay.

Next comes the wiring of the inputs, output, send and return sockets and the mains transformer, if required. After you have decided what inputs you require then it's just a matter of measuring the length of shielded cable to connect to the sockets. Do not forget to link any unused inputs to 0 V on the pc board. It is a good idea to mount the input sockets as far away from the mains transformer as possible to minimise hum pick-up. Also mount the inputs away from the outputs to minimise cross-talk. Use insulated 6.5 mm jack sockets to avoid hum loops. The hot end of the input sockets need to be grounded when no plugs are inserted hence sockets with closed contact are required. The input sockets should have their earth connected together at the sockets with some tinned copper wire. A single connection from the braiding of one of the input screening cable is all that is required to connect the socket earths to the pc board. Refer to the wiring diagram on the overlay. The earth screen of the other inputs needs to be soldered to the 0 V but cut off at the socket end. Make sure that the open ended screen does not short any of the inputs, use some sleeving or insulated tape if you are unsure of your wiring.

The same procedure applies to the effects send and return sockets. Do not forget to link the send out to the return in on

ETI 1424 Parts List

Resistors

All ¼ W Metal film, 5% tolerance

R1,4 .... 220k
R2,6,30,31 .... 2k2
R7,13,40 .... 1k
R8,9,10,11,12,18,20,21,26,28,32,33,35,36,37, 38 .... 10k
R3,5,14,15,27 .... 22k
R16,17 .... 4k7
R19,34,39 .... 56R
R22,25 .... 56k
R23 .... 3k9
R24,29 .... 100k
VR1,6 .... 10k log pc mount pot
VR2,3 .... 100k lin pc mount pot
VR4 .... 100k lin pc mount dual-gang pot
VR5 .... 10k lin pc mount pot

Capacitors

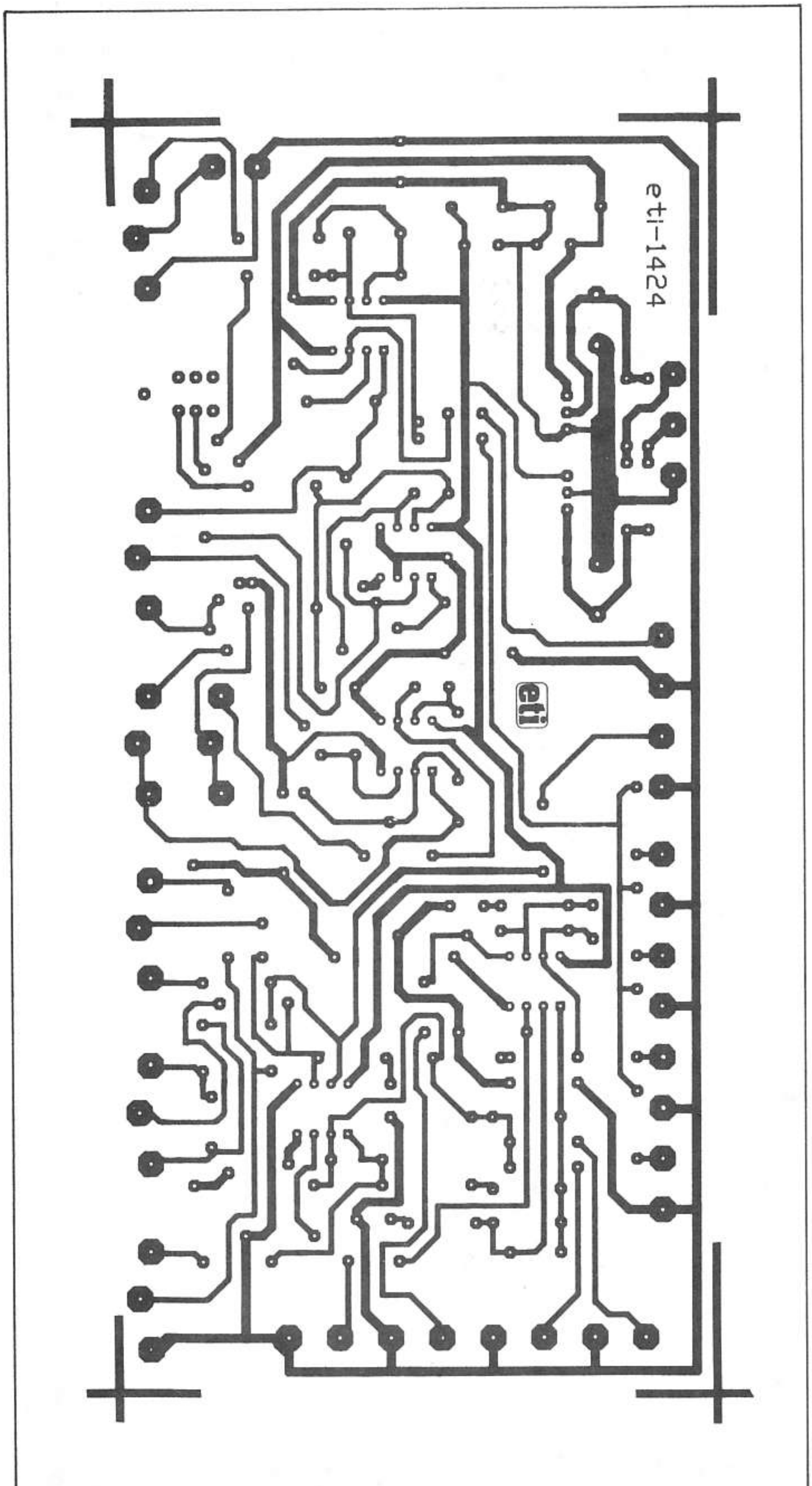
C1,4 .... 56n greencap
C2,5 .... 1u/35V bipolar electro
C3,21 .... 82p ceramic
C6 .... 10n greencap
C7,13 .... 560p ceramic
C8 .... 22n greencap
C9,11,12,20 .... 8n2 greencap
C10 .... 3u3/35V bipolar electro
C14,15 .... 100n greencap
C16,17 .... 33u/35V axial electro
C18,19 .... 1000u/35V axial electro

Semiconductors

IC1,2,3,4,5 .... TL072
IC6 7812
IC7 7912
D1,2,3,4 .... 1N4001

Miscellaneous

SK1 to 11 ....
.. 6.5mm Mono Insulated Jack Sockets (contacts closed when plug is not inserted)
SW1 .... PC Mount Miniature DPDT toggle
T1 .... 12V-0-12V at 500 mA Mains Transformer
Shielded Audio Cable
5 off 8 pin dil IC sockets


eti-1424
eti

the pc board, if you are not using this facility. These sockets are wired in such a way that the signal path is broken when a plug is inserted otherwise continuity is made, refer to the wiring diagram. Use a length of screened cable to connect the output of the pre-amp to the power amplifier.

The mains transformer for your power amplifier may have 12V-0-12 Vac tappings that can power the 1424 directly, otherwise a separate transformer will be required. Ensure that you use mains cable for all connections that carry mains voltage and that the metal chassis of the box is securely earthed. Be extra careful with any mains wiring.

Testing

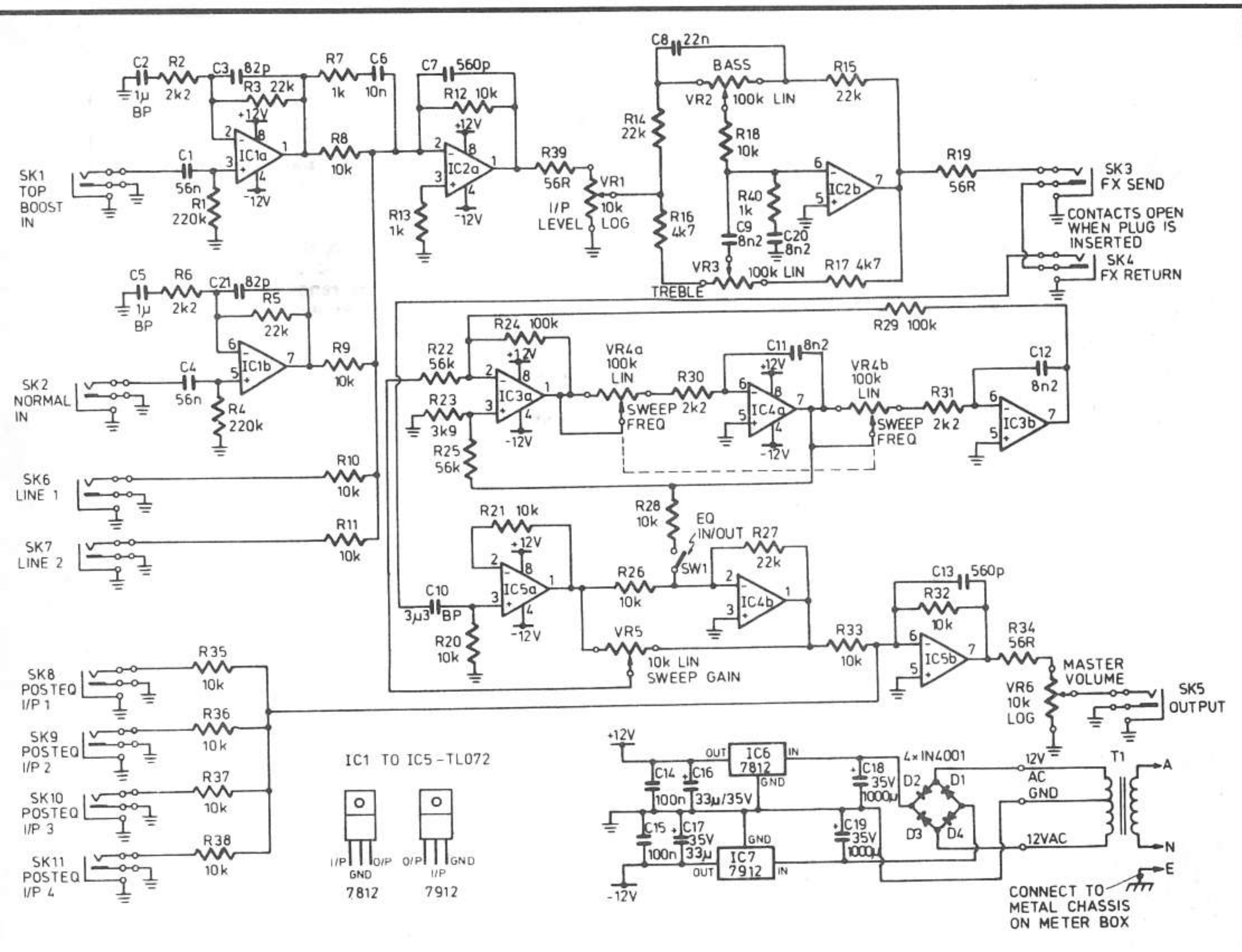
Once you are satisifed that the pc board is free of solder splashes and dry joints, then it's time to fire up the circuit. Without any of the ic's inserted, apply the power and check that there are +12 Vdc at pins eight and −12 V at pins 4 of the IC's. When it has been established, switch off the supply and insert the IC's. Make sure that they are orientated the correct way, refer to the overlay. Power up again and re-check the dc rails. It would be a good idea to disconnect the output to the power amp for this test. Set the bass and treble knob to midway and the sweep eq to out. Plug in a guitar and check that each input produces an output signal at the output of the pre-amp. Twiddle the tone controls individually and you should hear the difference! Switch in the sweep eq and turn the gain control clockwise to give a boost. Twiddle the frequency knob and you should hear the peak being swept over the frequency range. ●

ETI 1424 How it works

IC1a and b are connected as non-inverting amplifiers with a 20 dB fixed gain set by R2, R3 and R5, R6 for each input amplifier. The sensitivity of these amplifiers can be altered quite easily to suit the different levels of various pick-ups and to match the input sensitivity of different power amplifiers. The voltage gain equation is given by Av1a=1+R3/R2 and Av1b=1+R5/R6, hence to reduce the gain R3 and R5 need to be reduced. These amplifiers are ac coupled via C1 and C4 and the input impedance is set to 220 k for both the normal and top boost guitar input. The output of IC1a is fed to an equalizer network consisting of R7, R8 and C6 that boosts signal frequencies above about 1 KHz and will have a massive 20 dB boost at 10 kHz. The outputs of IC1a and b are mixed in IC2a which is connected as a adder, the two pre-eq line inputs are also summed at this point via R10 and R11. The level can be adjusted via VR1 before it is fed to the bass and treble tone controls built around IC2b. It is an active tone filter with the frequency selective components in the feedback path of IC2b. The bass control has a +/− 12 dB of cut and boost at 100 Hz and the treble control has a +/− 20 dB at 8 kHz. With the tone controls set to midway a flat response within 3 dB can be expected. R40 and C20 provides attenuation at high frequency for stability. The output of IC2b is buffered by a 56R resistor R19 to aid stability when driving capacitive loads.

The effects return is ac coupled via C10 and is buffered by IC5a and provides an input impedance of 10 k set by R20. The sweep eq is based on a standard active state variable filter, built around IC3a, 3b, 4a and b. The centre frequency is determined by VR4a and b, R30, R31 and the capacitors C11 and C12. An extremely wide frequency range of 200 Hz to 8.5 kHz can be obtained using a single pot control, VR4. The Q-factor is set to 4 by R25 and R22 and VR5 allows a variation over the stated frequency range. SW1 allows the sweep eq to be bypassed with an overall unity signal level except at the boost or cut frequency. The eq output is then mixed with the four post-eq line inputs in the summer, IC5b, VR6 controls the main output level of the pre-amp.

The 12 V-0-12 Vac is rectified by D1 to D4 which are connected as a full wave rectifier and smoothed by capacitors C18 and C19. A regulated +/−12 Vdc supply is derived from the 7812, (IC6) and 7912, (IC7) regulators. The circuit draws a dc current of around +/− 21 mA.



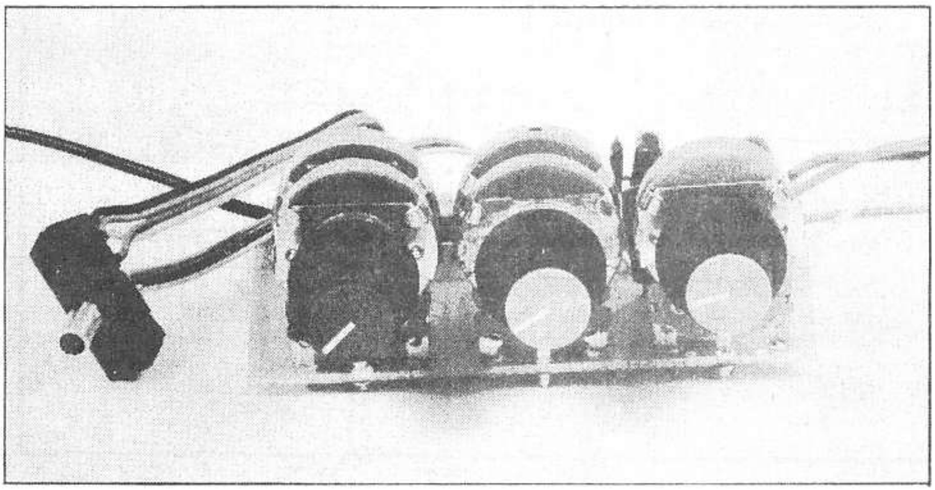
C2 R2 C3 82p R3 22k R7 C6 1k 10n
1μ BP 2k2
+12V
IC1a
-12V
SK1 TOP BOOST IN
C1 56n R1 220k
R8 10k
C7 560p R12 10k
IC2a
R39 56R
VR1 10k LOG
I/P LEVEL
R13 1k
C8 22n
BASS
VR2 100k LIN
R15 22k
R14 22k
R18 10k
R40 1k
C9 8n2
C20 8n2
R16 4k7
IC2b
R19 56R
SK3 FX SEND
CONTACTS OPEN WHEN PLUG IS INSERTED
SK4 FX RETURN
VR3 100k LIN
R17 4k7
TREBLE
C5 R6 C21 82p R5 22k
1μ BP 2k2
IC1b
SK2 NORMAL IN
C4 56n R4 220k
R9 10k
SK6 LINE 1
R10 10k
SK7 LINE 2
R11 10k
R24 100k
R29 100k
R22 56k
R23 3k9
IC3a
R25 56k
VR4a 100k LIN
SWEEP FREQ
R30 2k2
C11 8n2
IC4a
VR4b 100k LIN
SWEEP FREQ
R31 2k2
C12 8n2
IC3b
R28 10k
EQ IN/OUT
SW1
R27 22k
R21 10k
IC5a
R26 10k
IC4b
C10 3μ3 BP
R20 10k
VR5 10k LIN SWEEP GAIN
C13 560p
R32 10k
R33 10k
IC5b
R34 56R
MASTER VOLUME
VR6 10k LOG
SK5 OUTPUT
SK8 POSTEQ I/P 1
R35 10k
SK9 POSTEQ I/P 2
R36 10k
SK10 POSTEQ I/P 3
R37 10k
SK11 POSTEQ I/P 4
R38 10k
IC1 TO IC5 - TL072
I/P GND O/P
7812
O/P I/P GND
7912
+12V
IC6 7812
OUT IN GND
C14 100n
C16 33μ/35V
C18 35V 1000μ
4×IN4001
D1 D2 D3 D4
12V AC GND
12VAC
T1
A N E
C15 100n
C17 35V 33μ
C19 35V 1000μ
IC7 7912
-12V
CONNECT TO METAL CHASSIS ON METER BOX

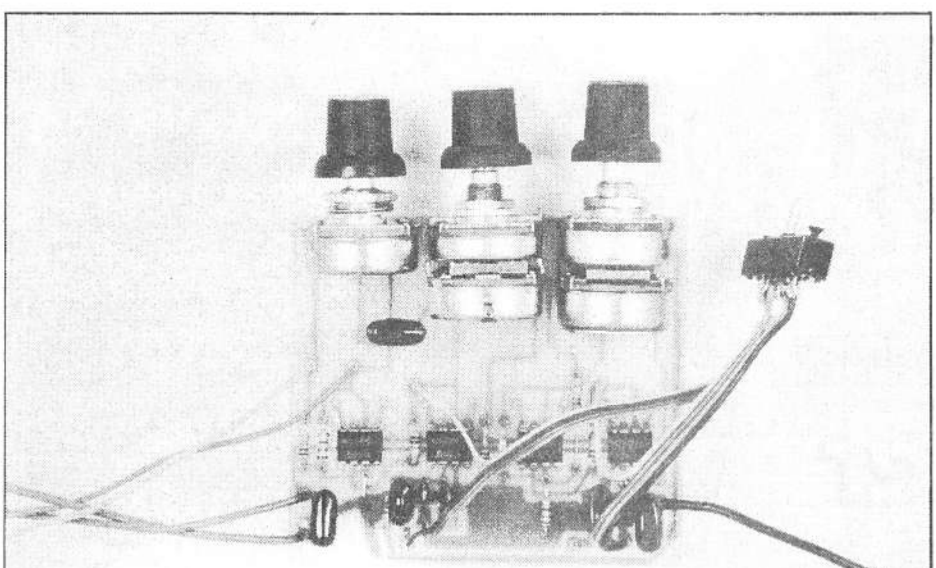
Project 1406

PARAMETRIC EQUALISER

Does your music system want a new frequency response? Does your guitar or keyboard need some equalisation to brighten the sound? Well, here is a module which can be used by itself on individual instruments or ganged to equalise your music system.

Neale Hancock





Top view of the module.

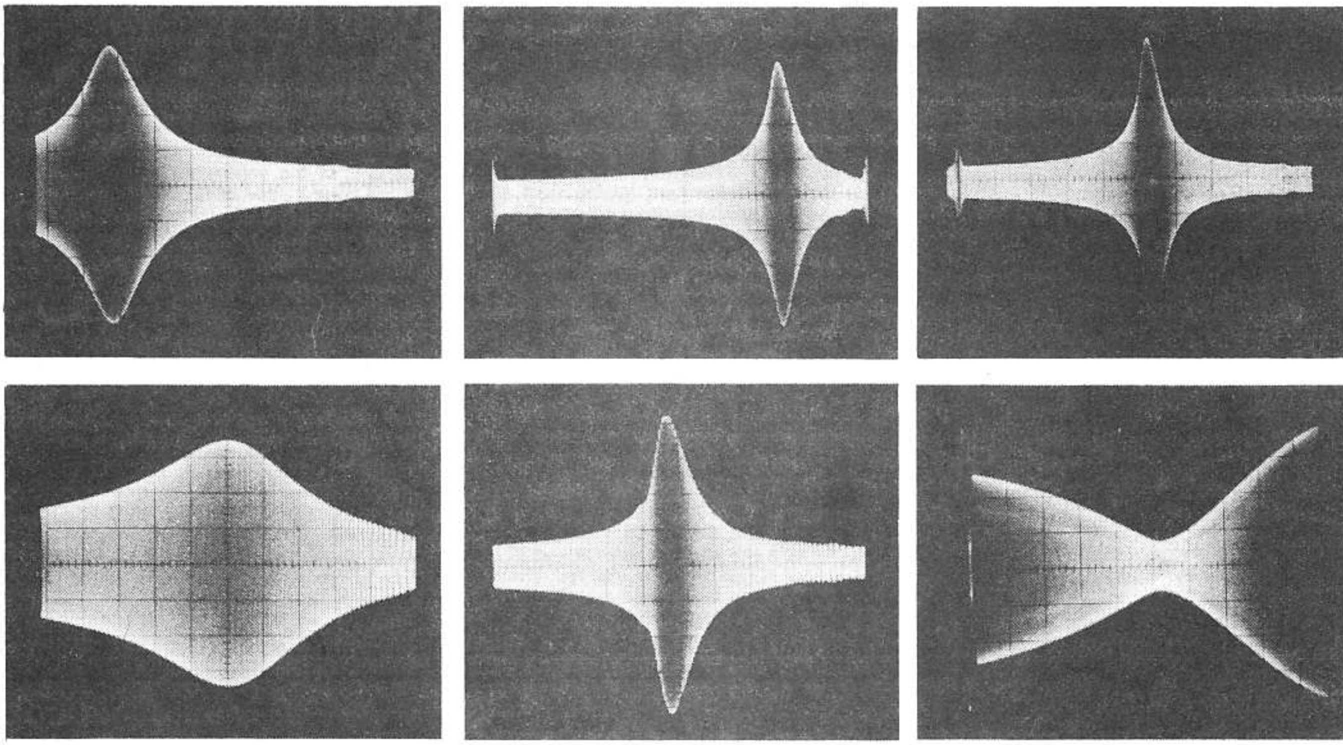
GRAPHIC EQUALISERS are widely used and accepted by audio enthusiasts as a means of correcting the acoustic deficiencies of a listening space. They are also used (probably by the majority of us) to make a stereo system sound better, by making the bass thump and adding more sparkle to the treble. The graphic equaliser is one way of optimising the frequency response of a music system to give our ears what they want.

Equalisation is not a process solely used in hi-fi applications. It's also used in public address systems as a way of eliminating feedback and in recording studios as a way to give an instrument a desired sound. In recording situations the parametric equaliser is a very versatile instrument, because it can be cascaded to give overall equalisation of a recording or used individually on separate instruments.

Both graphic and parametric equalisers use active bandpass filters to achieve equalisation. But whereas graphic equalisers use a number of preset bandpass filters called 'gyrators' (one for each slider on the front panel), parametric equalisers are tunable bandpass filters. This use of tunable bandpass filters in parametric equalisers allows each band to be more effective, thus reducing the number of bands required. And the increased efficiency allows a multiband parametric equaliser to be modular in design, with one bandpass filter in each module, making it more versatile than the graphic equaliser in the studio.

However, the ETI-1406 parametric equaliser module is designed to be used in any application where equalisation is required. These modules can be used independently or connected in series to form a multiband equaliser. The unit requires a signal of 100 mVrms line level or greater (up to 700 mV) to drive it, and runs from a ±15 volt supply. Casing details are left up to the user.

FREQUENCY RESPONSES (see text).



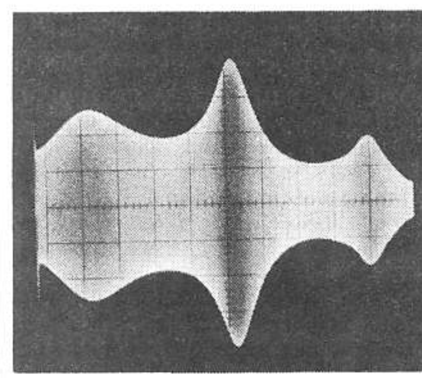


TABLE 1. PERFORMANCE OF ETI-1406

Range of centre frequencies	20 Hz to 19 kHz	Continuously adjustable
Range of Q	1 to 10	
Cut	−23 dB	
Boost	+18 dB	
Roll off	15 dB/octave	
Dynamic range	100 dB	measured at maximum gain and maximum Q
S/N ratio	90 dB	measured at maximum gain and maximum Q
Distortion	0.005%	measured at maximum gain and maximum Q

Measured at 1 kHz and a signal level of 500 Vrms.

Circuit concepts

As I mentioned earlier, the parametric equaliser modules consist of tunable active bandpass filters. To make a bandpass filter tunable, parameters such as centre frequency, bandwidth and the amount of cut and boost are made adjustable. Graphic equalisers have the centre frequency and width of the band preset, with only the cut and boost variable.

The accompanying series of photographs shows the effect on the frequency response of changing the tunable parameters. All the photographs show the same range of frequencies being swept, with the lower frequency on the left (2.6 kHz) and the higher frequency on the right (7.5 kHz).

Figures 1 and 2 show the effect of shifting the centre frequency of the parametric equaliser. Figure 1 shows the parametric equaliser set on a high frequency which would result in a boost of the frequencies around the peak. Figure 2 shows it set to a lower frequency.

Figures 3 and 4 illustrate the effect of increasing or decreasing the width of the passband. This is also referred to as the Q. In Figure 3 the circuit has a high Q, thus a narrow range of frequencies around the peak is emphasised. Figure 4 has a low Q and shows a broad range of frequencies being emphasised.

Figures 5 and 6 show boost and cut of the range of frequencies. In Figure 6 the signal was amplified vertically so that the cut could be more closely observed.

The circuit for the parametric equaliser is based on a state variable filter circuit. This type of filter features low pass, high pass and bandpass outputs. They are also capable of providing a high Q and they can be readily tuned.

To convert a state variable filter into a parametric equaliser, the circuit has to be modified to so that its Q, centre frequency and gain are all variable. Q can be varied by replacing the pair of resistors used to set it with a dual-ganging potentiometer. The centre frequency can be tuned by using a switch to select the range and a dual-ganging potentiometer to tune the centre frequency of the filter within that range. The range is selected by switching in different capacitor values and the dual-ganging potentiometer replaces the resistor pair used to set the centre frequency.

To enable the filter to have variable gain or attenuation (boost or cut) at the centre frequency, a gain stage is added to the state variable filter circuit. This gain stage allows the filter to have bandpass or band reject characteristics.

The circuit has been designed using high performance op-amps, such as the NE-5534AN and the TL-071. Of these two op-amps the NE-5534AN gives the best results for noise and distortion, but at a higher component cost. The specifications for the ▶

circuit using NE-5534AN op-amps are listed in Table 1. The circuit can also use the good old 741 op-amp however higher levels of noise and distortion can be expected when using this device.

Construction

Commence construction by examining the pc board for broken tracks, and bridges between tracks. The first components to be mounted are the resistors, capacitors and the link. Next mount the ICs, but first check their orientation with the overlay. To supply the voltage rails of the modules, the +15 and −15 volt power supply is used. The ETI-581 dual power supply would be ideal in this case.

To keep the number of flying leads in the project to a minimum I have used pcb-mounting pots. The only hassle involved in using these pots is that you may need to drill 2 mm mounting holes in the pc board to accommodate their pins. After the mounting holes have been drilled the pots can be mounted on the board.

The triple-throw toggle switch can now be connected to the pc board. The best way of connecting the switch is to use ribbon cable as it makes the wiring neater and simpler. Try to obtain the thicker gauge cable as it is easier to work with in this case.

The wires connecting the input and output sockets and the power supply to the pc board can now be connected. It is best to leave these connections until you have decided what case to use.

The type of case used to house the parametric equaliser depends largely on how you want to use it. For instance, if you use a single module as an independent unit, it should be housed in a case by itself. However, if you are constructing a multiband parametric equaliser or integrating the modules into a music system, you will have to drill out your case to suit.

The component overlay shows the purpose of each pot, and what frequency range is selected by each position on the triple-throw switch. The overlay also shows how to connect the modules in series, using switched 3.5 mm or 6.5 mm phono jacks, to create a flexible multiband parametric equaliser. The use of switched jacks between each module allows them to be used independently

Using it

When using individual parametric equaliser modules to modify the sound of a musical instrument, first set the Q control to the centre position the frequency range selection switch to its centre position and the cut/boost control either fully clockwise or fully anticlockwise

Play a sustained note (preferably middle C) and turn the centre frequency control until you hear the sound of the note change. If the cut/boost control is turned fully anti-

HOW IT WORKS — ETI-1406

As mentioned in the text, the circuit is based around an active state variable filter. The op-amps in this filter are ICs 1, 2 and 3. The output from IC1 is high pass, the output from IC3 is low pass and the output from IC2 is bandpass. The output from IC2 goes into the gain stage, which consists of IC4, R6, R9 and RV3.

The centre frequency of the parametric equaliser is determined by resistors RV2, R5 and R10 and capacitors C2 and C7. The frequency range is selected by switching in a pair of capacitors using a dual-pole triple-throw switch, SW1. The frequencies within the range are determined by the resistor pair R5 and R10 with the dual-ganging potentiometer RV2. The equation which sets the centre frequency, fc, is:

$$fc = 1/2\,(RV2 + R5)Cx$$

where Cx can be C2, C3 or C4.

This equation could also be written using R10, C5, C6 and C7 since all these components are the matching pairs of those in the equation.

C2 and C5 select the frequency range of 2 kHz to 20 kHz, C3 and C6 select the range 200 Hz to 2 kHz and the pair C4 and C7 selects the range 20 Hz to 200 Hz. The frequencies within the ranges are obtained by using a dual-ganging potentiometer RV2.

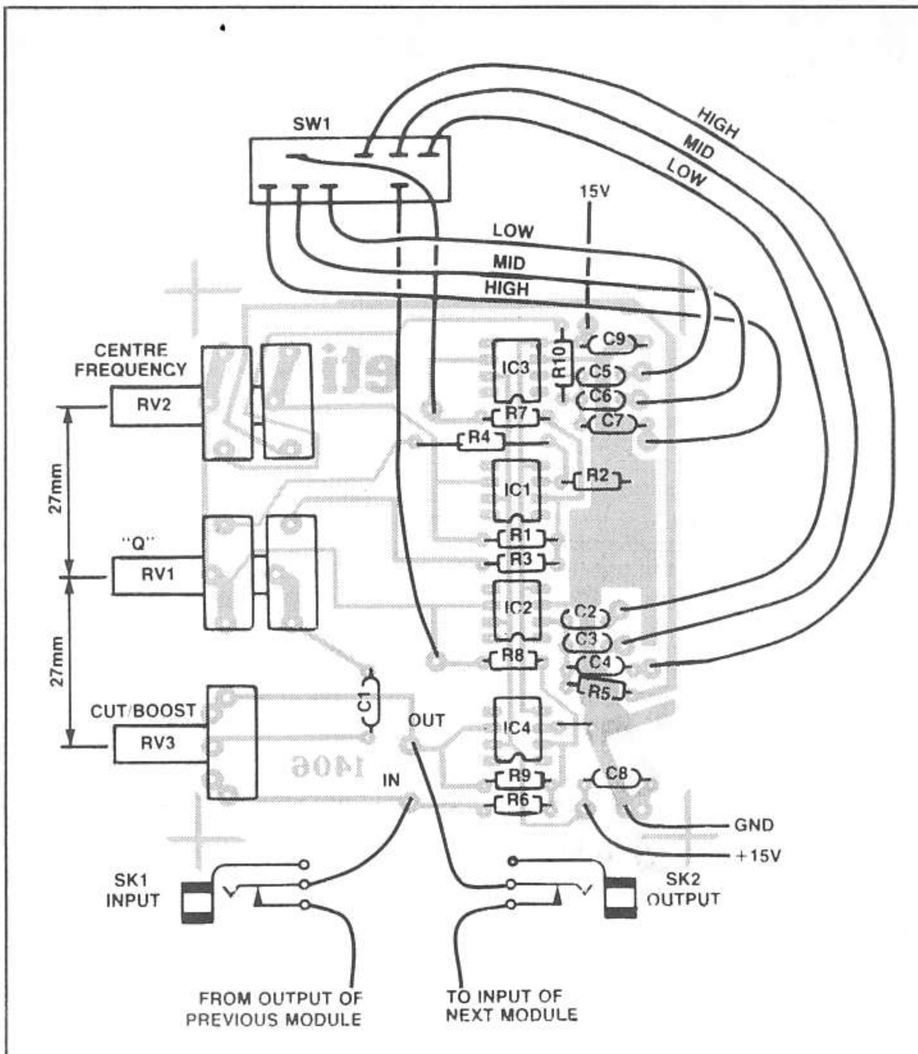
The highest frequency in the range is obtained when RV2 is turned fully clockwise making RV2 equal to zero in the equation above Therefore, R5 and R10 set the high end of the frequency range. When RV2 is turned fully anti-clockwise its value is equal to 100k in the equation above, thus setting the low end of the frequency range.

The Q of the parametric equaliser is set by the combination of R2, R4 and RV1. The equation used to determine the Q of a state variable filter is as follows:

$$(R4 + RV1)/R2 = 3Q-1$$
$$1 + (R4 + RV1)/R2 = 3Q$$
$$(1 + (R4 + RV1)/R2)/3 = Q$$

By substituting in the values of R2 and R4 as well as the maximum and minimum values for RV1, a maximum value for Q is 10 and a minimum value for Q is 1.

The gain stage gives the parametric equaliser circuit the ability to cut or boost the frequencies to which the filter is tuned. RV3 gives control over the amount of cut or boost while R6 and R9 set the overall gain of this stage. The capacitors C8 and C9 are there to remove high frequency noise from the supply rails.



PARTS LIST — ETI-1406

Resistors .... all ¼W, 5%
- R1, 7 .... 100k
- R2 .... 3k9
- R3-6, 8, 10 .... 10k
- R9 .... 22k
- RV1, 2 .... 100k dual-ganging linear with pc mounting pins
- RV3 .... 10k linear with pcb mounting pins

Capacitors
- C1 .... 220n greencap
- C2, 5 .... 820p ceramic
- C3, 6 .... 8n2 greencap
- C4, 7 .... 82n greencap

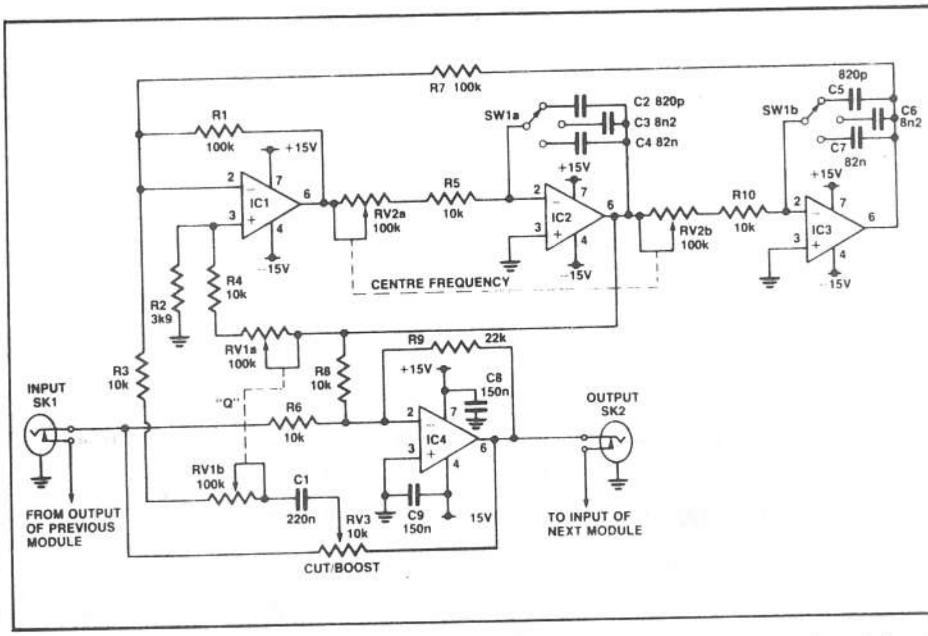
Semiconductors
- IC1-4 .... NE-5534AN or TL-071 (see text)

Miscellaneous

ETI-1406 pc board; 3 x potentiometer knobs; 2 x switched phono sockets; dual-pole triple-throw switch; hookup wire; case to suit.

Price estimate: $22-$33*

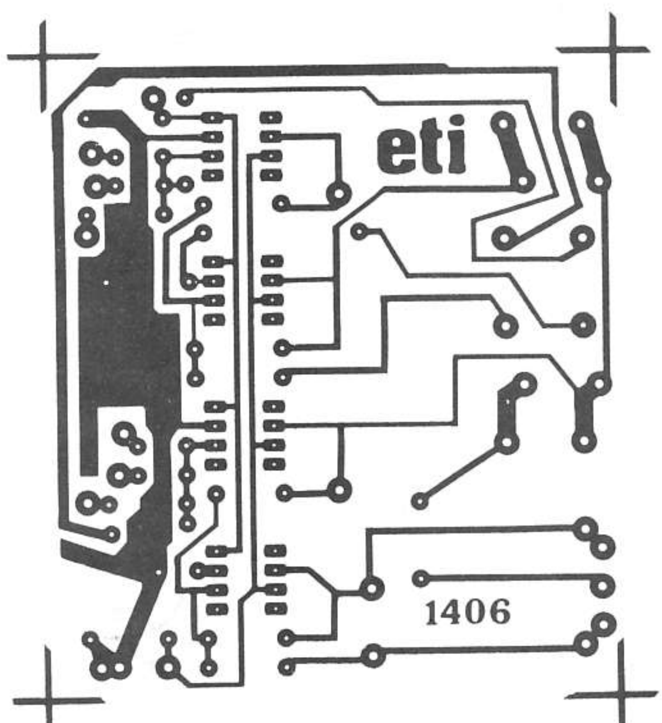
*The lower figure corresponds to the circuit using TL-071s



clockwise, the effect will be a dulling of the note, alternatively the sound will be brighter if the cut/boost control is turned fully clockwise By turning the Q control clockwise the range of notes dulled or brightened will be reduced. When it is turned anticlockwise the range of notes will be increased.

A good multiband parametric equaliser can be created by using three or four modules in series. Using such a multiband equaliser on orchestrated music would require a similar procedure to that outlined above. The only differences are that the input into the equaliser is different and that there are more bands to tune.

When using a number of modules in series each one increases the gain and the Q of the system. Make sure that the first module in the system does not have a high gain or Q; this applies to a lesser extent to each successive module. If the gain or the Q of the system is too large the result will be excessive distortion of the musical signal. ●



SHOPA

This page serves as a guide to constructors on where to source components, printed circuit boards and kits for the projects included in this book. While the information included here was checked just prior to publishing, a variety of things may have changed matters by the time you read this. It pays to check with suppliers first, before seeing them or trying to place an order.

Printed circuit boards
There are three suppliers able to provide printed circuit boards for projects included in this book, with certain noted exceptions. They are, in alphabetical order:

Acetronics PCBs
112 Robertson Road
BASS HILL, NSW 2197
(02) 645-1241

All Electronics Components
118-122 Lonsdale Street
MELBOURNE, VIC 3000
(03) 662-1381

RCS Radio Pty Ltd
651 Forest Road
BEXLEY, NSW 2207
(02) 587-3491

Note that the last two suppliers can also supply front panels for many projects.

When chasing up kits and components to build projects, you should first arm yourself with current catalogues from the major retailers — Altronics, Dick Smith Electronics, Jaycar and Rod Irving Electronics. Where specialised components are required, suppliers are suggested here, but where the exact components may no longer be available, ask their advice as they may be able to suggest a suitable equivalent or replacement. Now read on . . .

ETI-251 Op-amp Power Supply
Another project for the dyed-in-the-wool experimenter or enthusiast, op-amps being so widely used in circuits these days. Components for this project, as with the majority of others, are widely available from electronics retailers.

ETI-1411 Headphone Distribution Box
There should be no problem in obtaining components for this project — there aren't that many of them to start with, and there is nothing out of the ordinary. The enclosure is a standard sized 'zippy box', and all the components should be readily obtainable from any parts supplier.

ETI-1432 Audio Toolkit
Again, there is nothing unusual component-wise here — many readers will probably have most of the components already lying around. The printed circuit board can be obtained from the suppliers listed at the beginning of Shoparound.

ETI-751 Mini FM Transmitter
Few projects come as simple as this — and the parts list notes the Dick Smith Electronics catalogue numbers for critical components; note that many rival retailers stock the same parts.

ETI-1405 Stereo Enhancer
Now this project can develop some startling effects! Yes, it was designed using bog-standard, off-the-shelf components, even the Horwood case is a readily available item.

ETI-1406 Parametric Equaliser
This project will find particular application in sound reinforcement and

ROUND

musical instrument amplifier systems. It's based on the popular high performance NE5534AN op-amp, which is relatively cheap and widely available. Constructors should have no difficulty sourcing components for this project.

ETI-1413 Electronic Crossover
Whilst it's a good performer, this project uses nothing unusual in the way of components, all of which can be sourced from stock-standard parts at your favourite electronics retailer.

ETI-1414 Walkman Amplifier
Heave-off your headphones! As usual, youll find part for this project are readily available from electronics retailers.

ETI-1415 Vifa SA-80 Speaker and ETI 1417 Vifa SA-100 Speaker
These speakers are based on the popular Vifa drivers. Kits for these projects were originated, and are distributed by Scan Audio of Melbourne, who are located at:

52 Crown Street
RICHMOND, VIC 3121
(03) 429-2199
Kits are stocked by a number of electronics retailers, including Rod Irving Electronics and Jaycar.

ETI-1424 Guitar Pre-amp
Like the majority of projects included in this book, this project was designed using widely available components so you should have no difficulty getting your requirements together for this one.

ETI-1425 Guitar Note Extender
Another one for the musicians amongst you. The comments given for the ETI-1429 Noise Gate apply to this project too, as the heart of this design also is the NE572 compandor IC.

ETI-1429 Noise Gate
Here's one for the musicians! Components for this should be readily obtainable from most electronics suppliers, with the exception of the NE572 compandor IC. For this, you should check with those retailers specialising in a wide range of semiconductor products, such as in Melbourne:

All Electronic Components
118-122 Lonsdale Street
MELBOURNE, VIC 3000
(03) 662-1381

The Electronic Component Shop
289 Lonsdale Street
MELBOURNE, VIC 3000

Stewart Electronic Components
PO Box 281
OAKLEIGH, VIC 3166
(03) 543-3733

In Sydney:

Geoff Wood Electronics
PO Box 671
LANE COVE, NSW 2066
(02) 428-4111

ETI-1430 Digi-125 Audio Power Amp Module
A very popular, low-cost, simply to build, yet high performance amplifier modular. Components for this project you'll find as bog-standard, off-the-shelf items from most electronics retailers. The printed circuit board, as noted in the article, is available from Graham Dicker at PC Computers, 36 Regent Street, Kensington, SA 5068, (08) 332-6513. The board costs $4.95 singly, plus $2 post and handling, or $9 for a pair, plus $2 post and handling. The following retailers have indicated they stock the necessary components for this project: in Melbourne — All Electronic Components at 118-122 Lonsdale Street, and The Electronic Component Shop at 289 Latrobe Street (except for the 2 W resistor). Dick Smith Electronics indicate they stock the components in their stores around Australia and New Zealand, too.