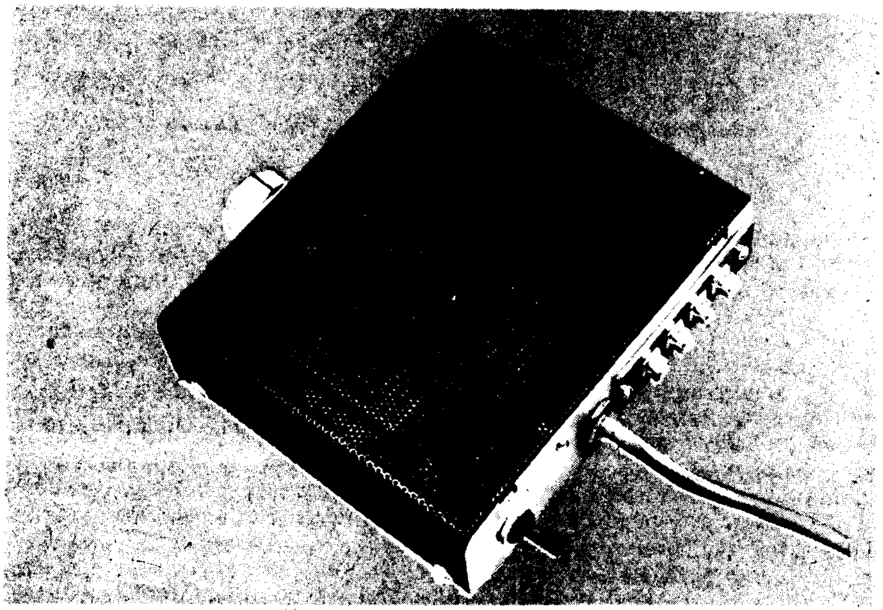
SUPER–



ETI PROJECT 410

SELECT THE WIDTH OF YOUR STEREO'S EFFECTIVE IMAGE – FROM A POINT SOURCE TO A SPREAD MUCH GREATER THAN NORMAL.

When stereo reproduction was a novelty, many recordings were made with grossly exaggerated stereo 'image'. So much so that on some orchestral recordings the second violins appeared to be playing somewhere to the left of the gentlemen's toilet.

Now, some record companies have swung the other way, and music lovers complain that a number of recordings – especially of symphonic music – have *insufficient* spread, and the apparent stage is restricted to a small area either side of the centre line of the speaker enclosures.

To some extent this can be remedied by increasing the spacing between speakers – but only if room dimensions permit.

This is a problem that has attracted the attention of Mullard Ltd, and they have developed a 'sound-source width control' that enables the stereo 'image' to be adjusted so that at one extreme both stereo channels are spatially combined so that the sound apparently emanates from a point half way between the two speakers, whilst at the other extreme, the effective

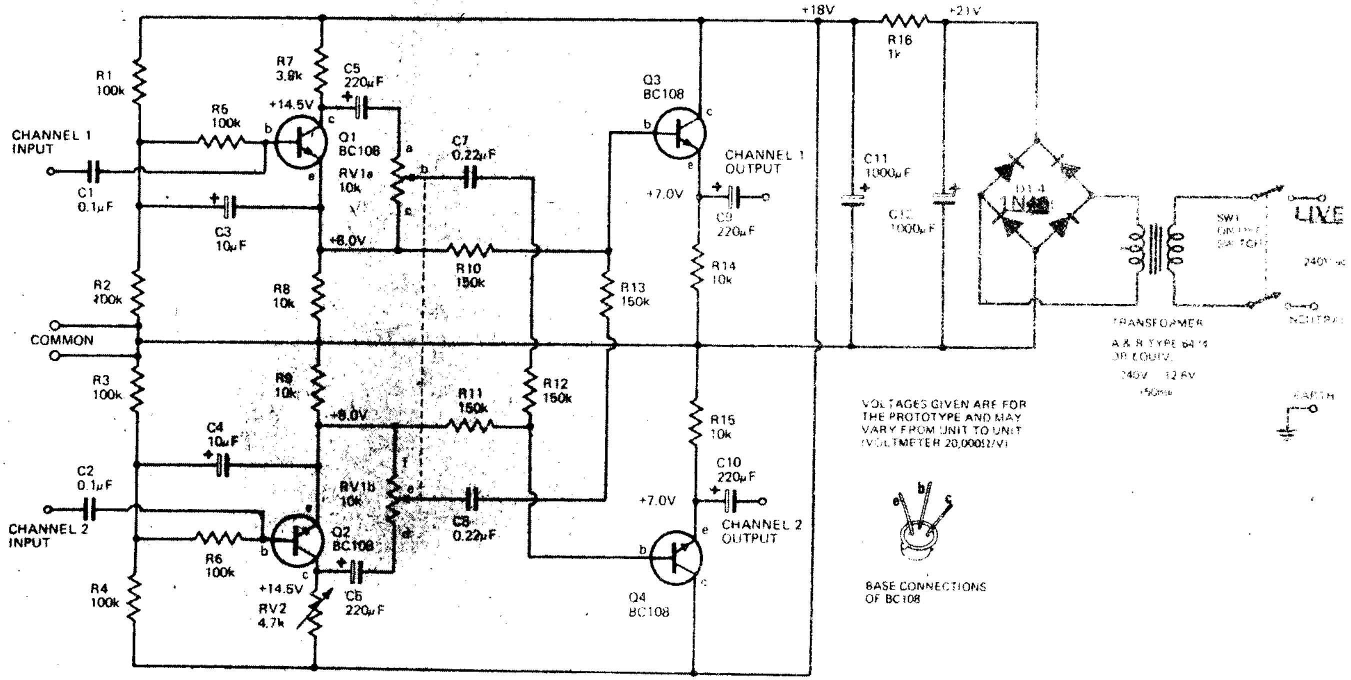


Fig. 1. Circuit diagram of complete unit.

STEREO

stereo image is increased quite considerably.

The circuit operates by adding part of the signal in one channel to the signal in the second channel. This is done with both signals in phase (to produce mono effects), or with the signals out of phase (to produce stereo-width enhancement).

Care has been taken to ensure that the unit does not introduce hum or distortion.

When we first assembled the unit we found that the range of adjustment provided by the width control was not really sufficient to cater for all programme material and the circuit described here has been modified to provide continuous adjustment, from mono, through the normal stereo image, to an apparent stereo spread approximately 40% greater than normal.

INTER-UNIT CONNECTIONS

This unit is designed to accept high level signals – exceeding 100 mV, and is intended for connection between a pre-amplifier and main amplifier.

It is not suitable for handling signals directly from a low level (less than 100 mV) magnetic pick-up. This is because internal noise generated by the unit will degrade the low level signals.

Crystal or ceramic pick-ups have sufficient output successfully to drive the unit and it may of course also be used between a pre-amplifier and tape recorder, tape recorder (reel-to-reel or cassette) and amplifier, or between two tape recorders.

Many modern amplifiers are of course built with the pre-amplifier and main amplifier combined. With these it is generally possible to connect the unit's input to the 'tape-out' connections and the unit's output to the 'tape-input' sockets on the stereo amplifier. (This approach is also used by the Bose company – their active equaliser is interconnected in the same way).

TESTING

At this stage connect input 1 and output 2 only, input 2 and output 1

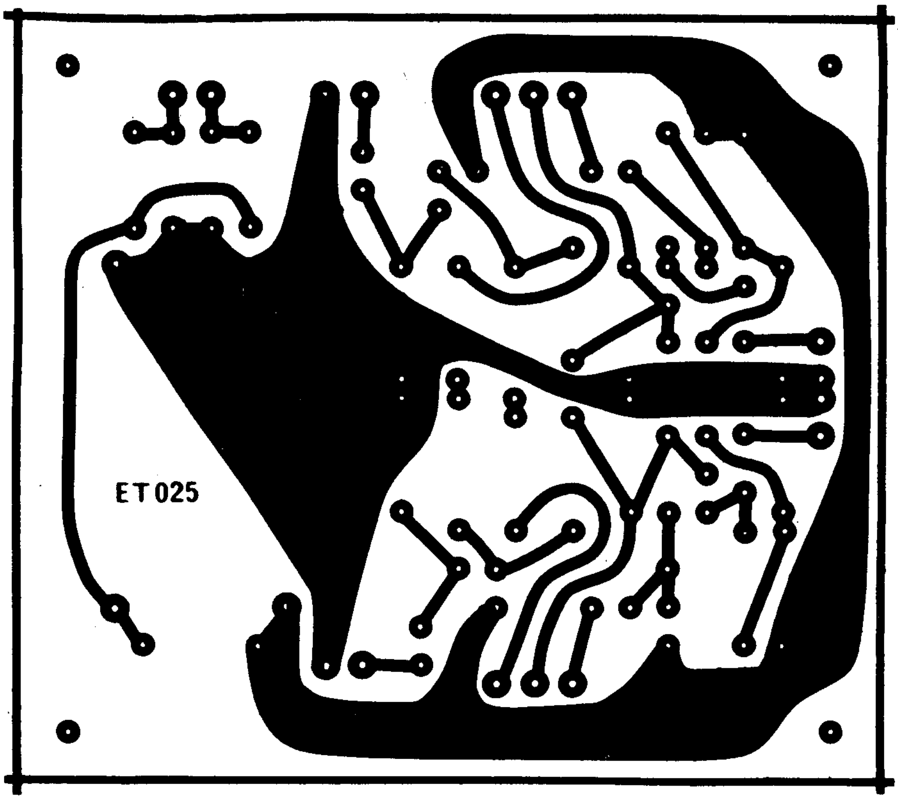


Fig. 2. Foil pattern of printed circuit board – full size.

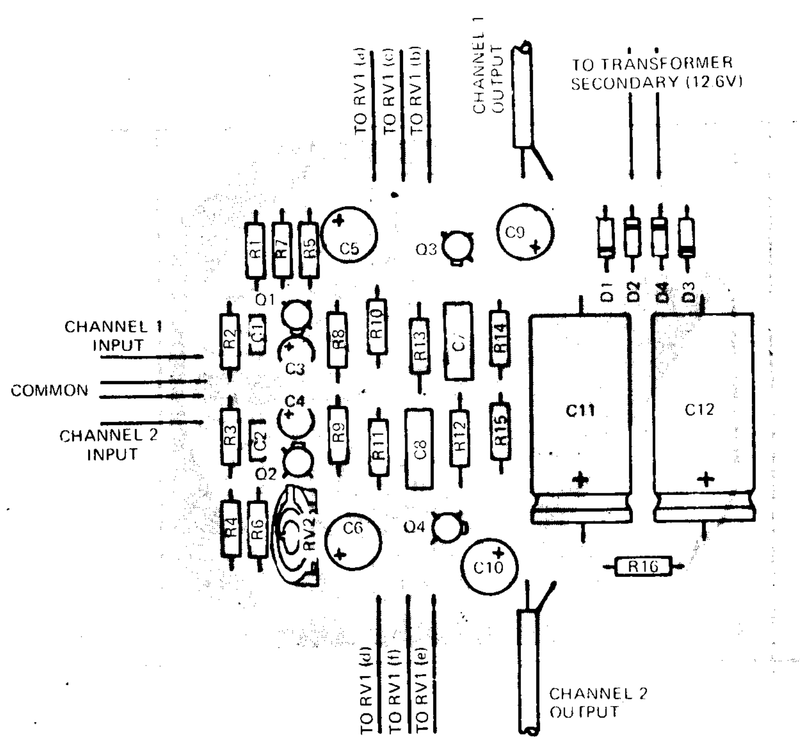
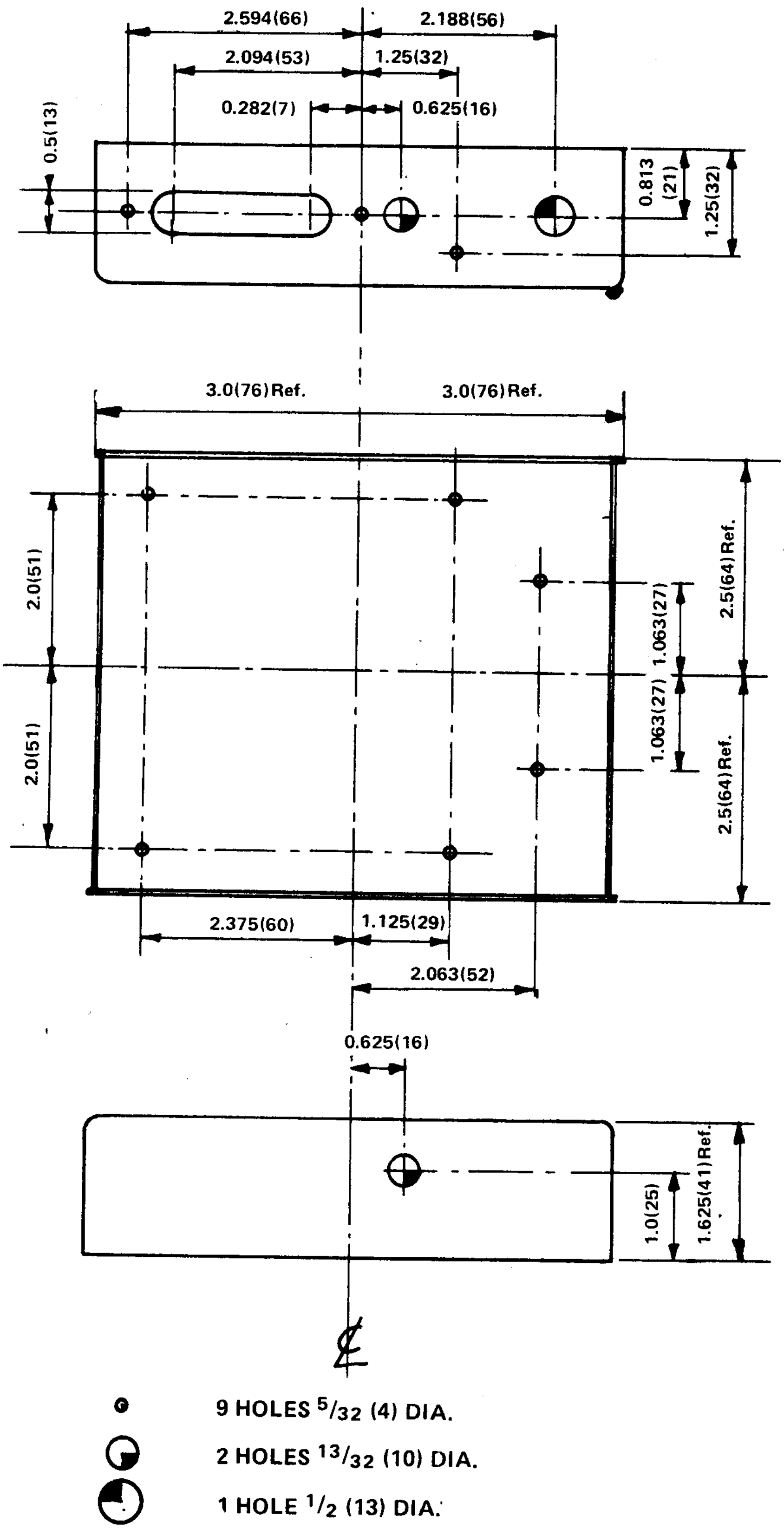


Fig. 3. How the components are located on the printed circuit board.



DIMENSIONS IN BRACKETS ARE IN MILLIMETERS

Fig. 4. Constructional and drilling details of metal case.

PARTS LIST ET410

R1	— resistor	100k	½ W	5%
R2	— "	100k	"	"
R3	— "	100k	"	"
R4	— "	100k	"	"
R5	— "	100k	"	"
R6	— "	100k	"	"
R7	— "	3.9k	"	"
R8	— "	10k	"	"
R9	— "	10k	"	"
R10	— "	150k	"	"
R11	— "	150k	"	"
R12	— "	150k	"	"
R13	— "	150k	"	"
R14	— "	10k	"	"
R15	— "	10k	"	"
R16	— "	1k	"	"
RV1	— 2 gang 10k Lin. Pot			
RV2	— Trimpot 4.7k (Large Type)			
C1	capacitor	0.1 uF 100V		
C2	"	0.1 uF 100V		
C3	"	10 uF 25V electrolytic		
C4	"	10 uF 25V "		
C5	"	220 uF 16V "		
C6	"	220 uF 16V "		
C7	"	0.22 uF 100V		
C8	"	0.22 uF 100V		
C9	"	220 uF 16V electrolytic		
C10	"	220 uF 16V "		
C11	"	1000 uF 25V "		
C12	"	1000 uF 25V "		
Q1	Transistor	BC108 or equivalent		
Q2	"	BC108 "		
Q3	"	BC108 "		
Q4	"	BC108 "		

D1-D4 Diode 1N4001 or equivalent
Transformer 240/12.6V 150mA

PC board ET025.
Metal box

4 way coaxial sockets.
Double pole 240V switch MSP 625 or similar.
Rubber grommets.
3 core flex and plug.
Cable clamp (for mains cord).
Knob for pot.
Nuts and bolts.
4 spacers ¼" long for PC board.
Coaxial cable wire etc.

must be left disconnected.

Switch on all units, and play a stereo recording through the system. Adjust RV1 (front panel control) for minimum output in speaker channel 2. Leave RV1 in this position for the time being. Mark this position on the case – it represents the normal stereo setting.

Now connect input 2 and output 1, and disconnect input 1 and output 2. Again play the record through the system but this time adjust RV1 to give minimum output in speaker channel 1.

Reconnect input 1 and output 2. The unit is now ready for use.

It should be noted that the volume level will drop as RV1 is turned towards the 'super-stereo' position, this should be corrected by adjusting the volume control.

CONSTRUCTION

The circuit diagram of the complete device is shown in Fig. 1.

The simplest way to build the unit is to assemble the components on the printed circuit board – the foil pattern of which is shown full-size in Fig. 2.

Figure 3 shows how the components are assembled on the printed circuit board. Ensure that transistors, diodes and electrolytic capacitors are correctly orientated. Trimming potentiometer RV2 should be bent over slightly to allow ease of adjustment.

The assembled printed circuit board, together with the mains transformer and potentiometer RV1, should then be fitted into the metal case.

For our prototype unit, we used the chassis shown in Fig.4. This drawing shows sufficient details for those who wish to construct their own case.

Leads carrying audio signals must be screened if they are longer than an inch or so. However if the unit is assembled as shown in Fig. 5, only the output leads require screening. Co-axial cable or standard screened lead is suitable for this purpose.

When wiring up the power supply note that the transformer centre tap is not used.

It is of course perfectly feasible to build the circuit and controls within an existing stereo amplifier – in which case the power supply would not be required. (The unit draws only a few milliamps and within the range of 12 to 18 volts, voltage is not overly critical). ●

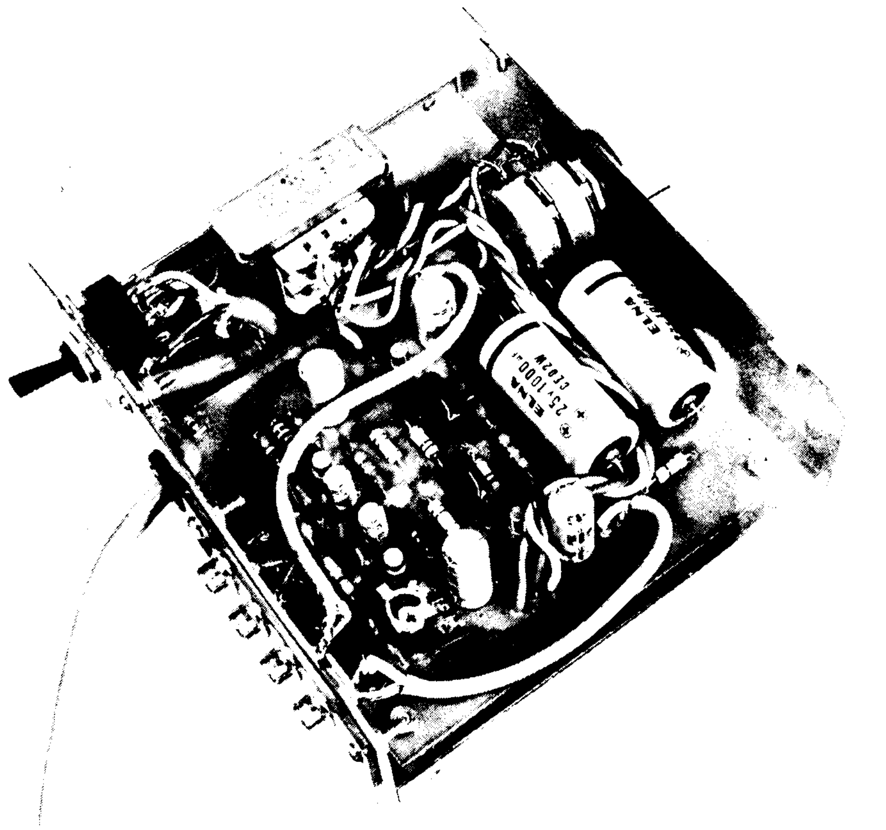
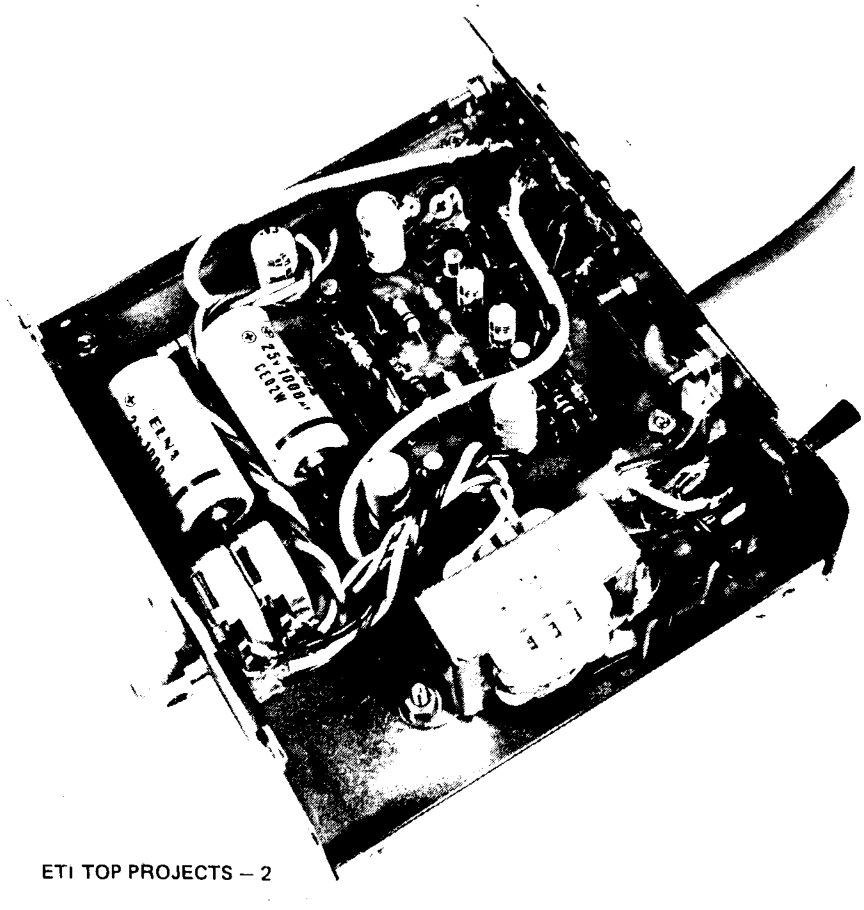


Fig. 5. Note how very short leads are used to connect the output sockets to the printed circuit board. Screened leads have been used only for the longer input connections.



HOW IT WORKS

Basically the circuit consists of two practically identical channels. This text describes the operation of channel 1.

The circuit consists of a buffer stage that provides both a unity gain 'in-phase' output, and an attenuated (39%) 'out-of-phase' output. Potentiometer RV1 enables the output of the buffer stage to be varied between either the 'in-phase' or 'out-of-phase' condition. This output is fed to mixer transistor Q3.

The output of Q3 consists of the channel 1 input plus a proportion of channel 2. The amount of channel 2 signal that is mixed with channel 1 depends upon the setting of RV1 and can vary from the full 'in-phase' channel 2 signal (mono) through zero input from channel 2 (normal stereo), through to 39% 'out-of-phase' (super-stereo).

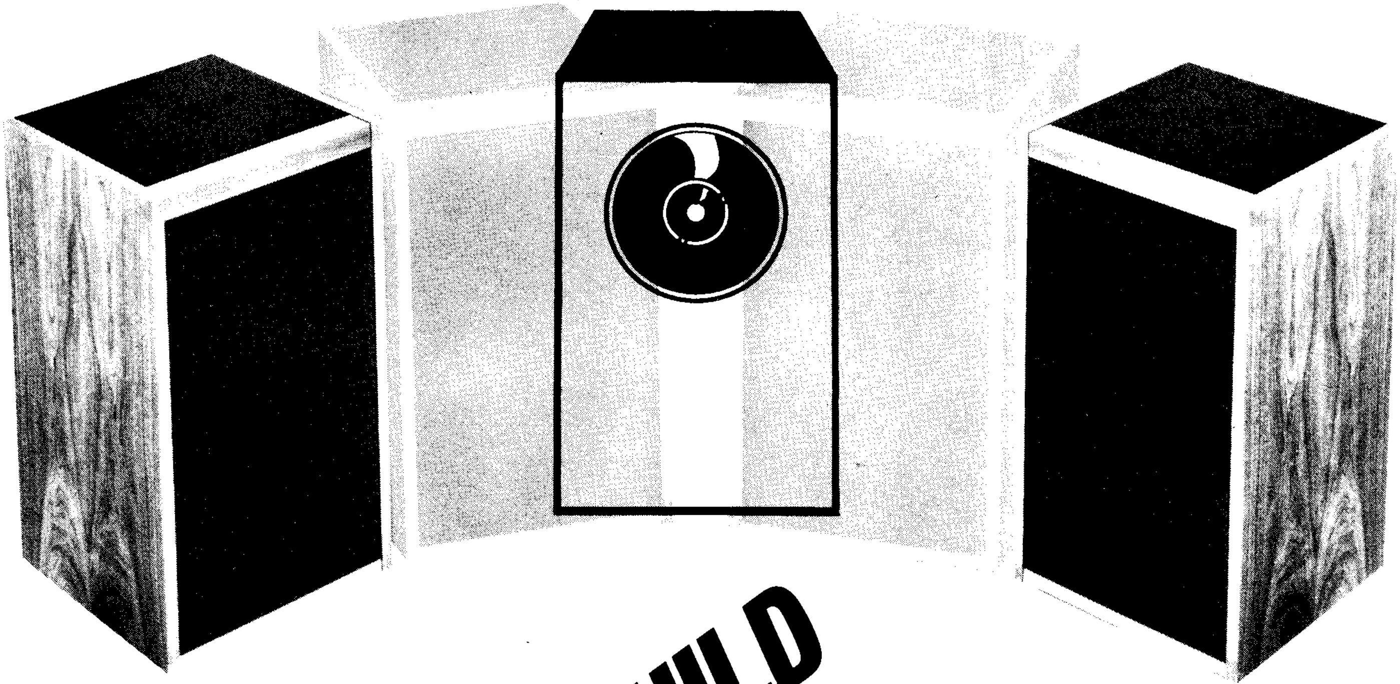
The operation of channel 2 is similar to that described above.

The mixing process either amplifies signals common to both channels – or attenuates the common signal (super stereo).

Buffer transistor Q1 is biased by R1, R2 and R5. Capacitor C3 provides some positive feedback to increase the input impedance (bootstrapping). Q1 has unity gain at the emitter (in-phase), and an inverted signal of 39% of the input voltage at its collector. RV1 provides the variable output. The bias for mixer Q3 is derived from the output of Q1 via R10.

The power supply consists of a 12.6V transformer, a bridge rectifier, and filter capacitor C12. Further filtering is provided by R16 and C11. Due to the large amount of filtering capacitance – and the low current drawn by the unit – the unit will continue to operate for about 45 seconds after it is switched off.

BY PAUL E. MILLER



BUILD A STEREO BLEND CONTROL

Vary channel separation to suit your taste with this inexpensive circuit.

AUDIO designers usually try to maximize their products' stereo separation. There are times, however, when a measure of crosstalk between channels is desirable. For example, the disquieting "orchestra in the cranium" effect experienced with stereo headphones can be mitigated by reducing the program material's channel separation. The stereo blender described here allows the user to vary channel separation to suit his taste. Also, the two channels can be transposed with adjustable separation—left input to right output, and vice versa. The blender employs inexpensive components, and can be bypassed at the touch of a switch.

About the Circuit. The schematic diagram of the stereo blender is shown in Fig. 1. The heart of the circuit is contained in two variable voltage dividers, comprising *R1* through *R4* and *R9* for the left channel, and *R5* through *R8* and *R10* for the right channel. Input signals are applied to the voltage dividers via coupling capacitors *C1* and *C2* and voltage followers *IC1A* and *IC1B*.

A dual 10,000-ohm, linear-taper potentiometer is used for *R9* and *R10*. When the potentiometer wipers are at one extremity of their travel, the stereo separation and spatial location of the input signals are preserved. At the other end, there is still no introduction of crosstalk but the channels are transposed. Adjusting the wipers for the center of their travel gives a complete "blend," with both inputs mixed equally and fed to both outputs. Between the center and either extreme, partial blending of the two channels is obtained.

The voltage dividers have an insertion loss of approximately 4.7 dB. This loss is compensated for by the gain introduced by *IC2A* and *IC2B*. To ensure that the voltage divider losses and op amp gains cancel each other, resistance tolerances should be kept fairly close. If this is done, no audible change in volume will occur when the project is switched on or out of the signal path.

Another reason for using close-tolerance resistors lies in an important characteristic of the voltage dividers. That is, the overall output should remain constant regardless of the setting of the dual potentiometer BLEND control. Actually, the signal level at the output will be 3 dB below the input when the BLEND control is at its mid-position. But this loss is compensated for by the fact that the inputs are mixed equally and fed to each output. To maintain this relationship, actual resistances should be close to the components' nominal values.

Signals from the op amps are coupled to the output jacks via capacitors *C3* and *C4*, which also block any dc offsets generated by the gain stages. Fairly large values are required if output impedances are to be kept fairly low. At 20 Hz, a 1-μF capacitor has a reactance of approximately 8000 ohms. Therefore, the circuit should drive a load with a fairly high input impedance—a condition satisfied by most power amplifiers and tape deck record preamplifiers.

The output coupling capacitors must

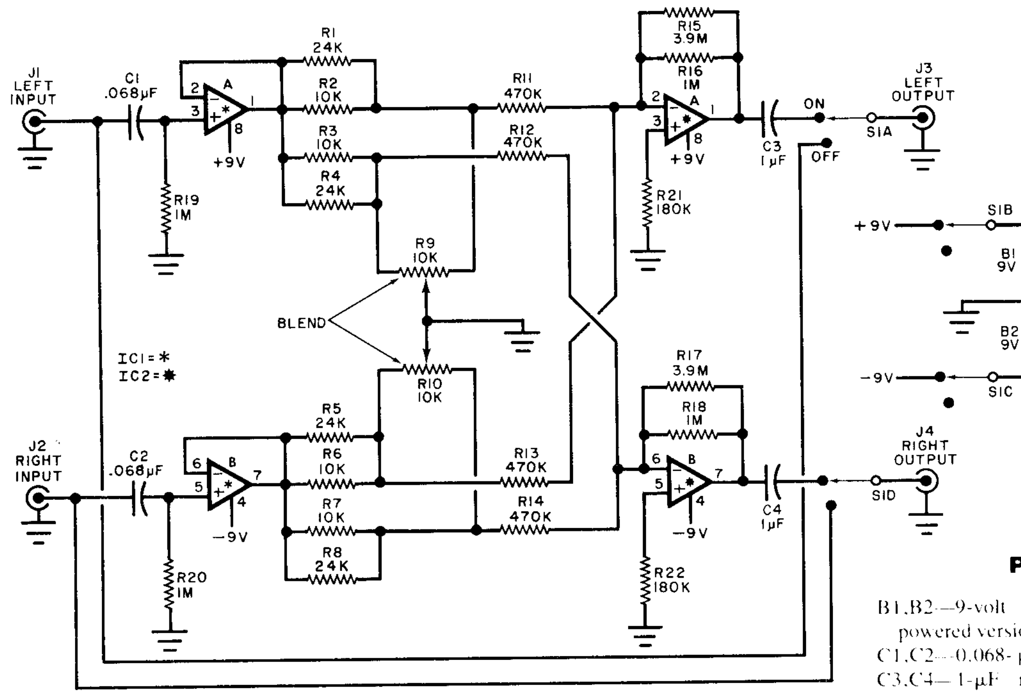


Fig. 1. Schematic diagram of the stereo blend.

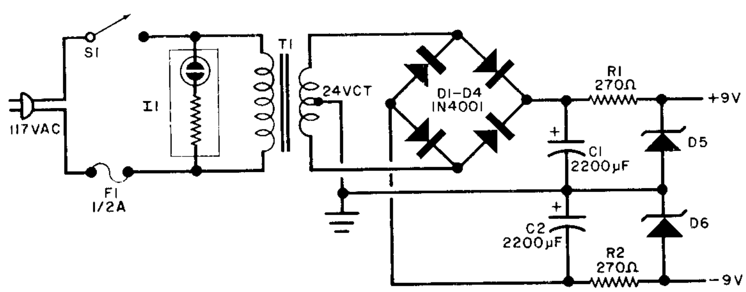


Fig. 2. Ac power supply features zener diode regulation.

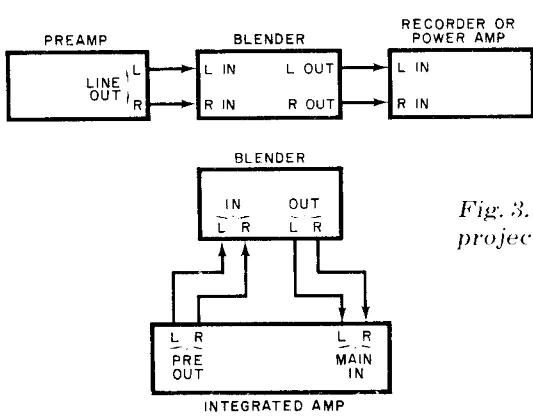


Fig. 3. Connecting the project to your system.

PARTS LIST

B1,B2—9-volt transistor batteries (battery powered version only)
C1,C2—0.068-μF Mylar capacitor
C3,C4—1-μF monolithic or nonpolarized electrolytic
IC1,IC2—MC1458 or 5558 dual op amp
J1 through J4—RCA phono jack
The following are 1/4-watt, 5% (or better) fixed resistors.
R1,R4,R5,R8—24,000 ohms
R2,R3,R6,R7—10,000 ohms
R11 through R14—470,000 ohms
R15,R17—3.9 megohms
R16,R18,R19,R20—1 megohm
R21,R22—180,000 ohms
R9,R10—dual 10,000-ohm, linear-taper potentiometer
S1—4pdt (battery powered version) or dpdt (line powered version) toggle or slide switch
Misc.—IC sockets or Molex Soldercons, printed circuit or perforated board, shielded or coaxial cable, hookup wire, suitable enclosure, battery clips, battery holders, machine hardware, solder, etc.

AC SUPPLY PARTS LIST

C1,C2—2200-μF, 25-volt electrolytic capacitor
D1 through D4—1N4001 rectifier
D5, D6—9.1-volt, 1-watt zener diode
F1—1/2-ampere fuse
I1—Neon indicator assembly with integral current-limiting resistor
R1, R2—270-ohm, 1/2-watt, 10% tolerance carbon composition resistor
S1—spst switch
T1—24-volt center tapped, 85-mA transformer (Stancor No. P8394 or equivalent)
Misc.—Line cord, fuse holder, terminal strips, strain relief, hookup wire, machine hardware, solder, etc.

be nonpolarized because the ac signals are not riding on a large dc level. The author suggests the use of monolythic capacitors because of their high capacitance-to-volume ratio. Other types can be used if space permits. Nonpolarized electrolytics, which are commonly used in speaker crossovers, are readily available in unit quantities.

Much smaller coupling capacitors are used at the project inputs. Although they have fairly high capacitive reactance at audio frequencies, the resistance of *R19* and *R20* and the very high input impedances of the voltage followers prevent significant signal attenuation.

Two 9-volt transistor batteries power the circuit of Fig. 1. Total current drain is fairly low, so fairly long battery life can be expected if the project is used intermittently. However, you might prefer to power the project from the ac line. A suitable regulated bipolar supply is shown schematically in Fig. 2.

In the battery-powered version, *S1* is a 4pdt switch. The circuit is inserted into the signal path and the batteries connected to the op amps when the switch is placed in its ON position. The batteries are disconnected and signals at the input jacks routed directly to the output jacks, effectively removing the project from the signal path, when the switch is placed in the OFF position. In the line-powered version, *S1* becomes dpdt switch and is used only to insert or remove the circuit from the signal path. To keep the line-power ac away from the low-level signal lines, a separate spst switch is used to control the primary of the power supply.

Construction. The circuit can be assembled on either a printed circuit or perforated board. Shielded wire or small diameter (RG-174-U) coax should be used for all signal leads. If the line-powered supply is to be housed in the same enclosure as the signal processing circuitry, the two should be physically isolated as much as possible. A metal utility box should be used to house the project.

Use. The blender should be connected to your audio system as shown in Fig. 3 by means of shielded patch cords terminated with suitable connectors. As mentioned earlier, it can be used to make listening through stereo headphones more enjoyable. The project also allows home recordists to introduce interesting special effects when taping program material. Imaginative users will no doubt find other applications. ◇