Paul Stenning ushers us into the cinematic world of surround sound at home, with his complete decoder.

Get Surround,

Like many owners of stereo satellite receivers or NICAM stereo equipment, I have connected the audio output to my hi-fi system. This gives a vast improvement in realism, particularly with films. However, by comparison to the cinema, there is still something missing! One thing that really makes a visit to the cinema memorable is the Dolby Stereo surround sound system.

Nearly all new films are made with Dolby Stereo sound. The surround sound information will often remain encoded into the stereo soundtrack when the film is transmitted by satellite or in NICAM stereo, or released on video. The Sky Television film channels have a "Dolby Surround" caption at the beginning of these films. These are also indicated in some television listing guides by the letters "SS", but this does not seem to be too accurate in the magazine I use.

There are some excellent Dolby licensed decoders available for home use, however these tend to be rather expensive. The unit presented here is a "sound-alike" circuit which, while not performing quite as well as the real thing, gives superb results for a more affordable price. An 8-watt power amplifier is included to drive the rear speakers.

Construction is straightforward, probably the most difficult part is drilling and preparing the case. The completed unit needs no setting up or alignment.

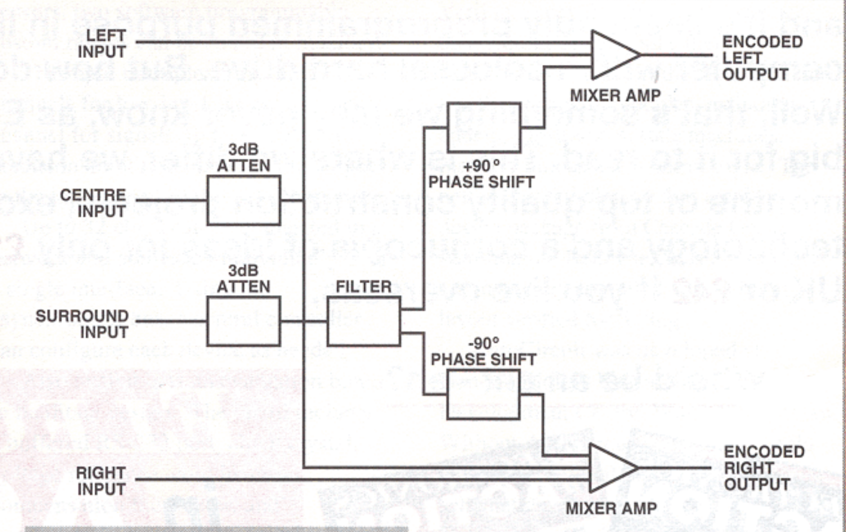


Fig.1 Surround sound encoding

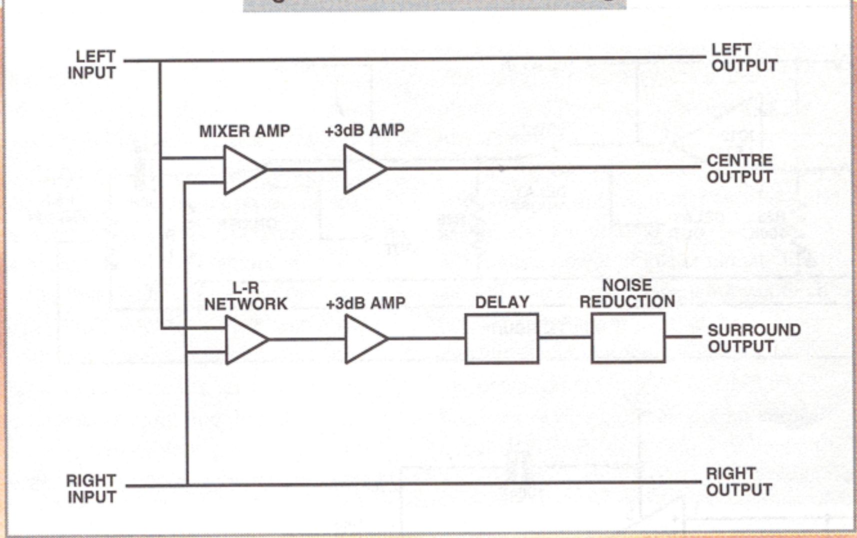
Surround Sound Operation

The professional system used in cinemas has four channels, Left, Right, Centre and Surround. The Centre channel is necessary in a cinema, to ensure that people sitting to one side of the auditorium do not miss the effect. It also fills the gap caused by the Left and Right speakers being so far apart. A Centre channel has not been included on this more basic home system, since the effect does not have to cover such a wide area.

When the system was developed, the most common cinema film format was 35mm, which has provision for only two sound channels (the 70mm format now commonly used in larger cinemas has four sound channels). This presented Dolby with the problem of encoding four channels into two, in a way that could be replayed successfully on a two-channel stereo system without a suitable decoder.

A basic form of the encoding system is shown in Figure 1. The Centre channel is attenuated by 3dB and added equally to both the Left and Right channels. The Surround channel is attenuated by 3dB before passing through a filter circuit having a pass band of 100Hz to 8kHz. The signal then passes through a pair of phase shift circuits, before being added to

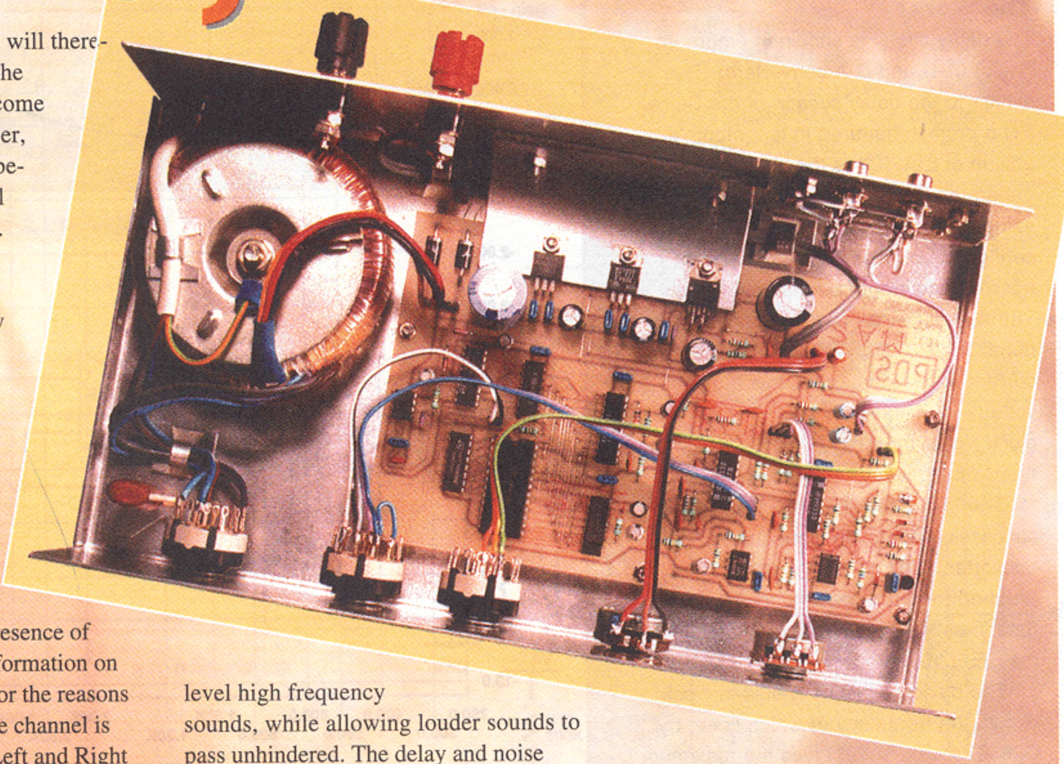
Fig.2 Surround sound decoding



round, round

the Left and Right channels, as shown.

If this is replayed on a stereo system without a decoder (like the cinema in the author's home town, until last year), it will sound fine. The Centre channel sounds will come from the Left and Right speakers equally, and will therefore appear in the Centre. The Surround signals will also come from both speakers. However, the 180 degree phase shift between the two channels will give this a very wide effect. A basic decoding system is shown in Figure 2. This is based on the original Dolby Surround system, used in older cinemas and this project. Most modern cinemas, and the more expensive commercial home decoders, use the Dolby Pro-Logic system.



The Left and Right signals pass unaffected through the decoder. The presence of the Centre and Surround information on these causes no problems, for the reasons described above. The Centre channel is extracted by summing the Left and Right signals. The resulting signal is amplified by 3dB to counter the attenuation in the decoding system.

The Surround channel is extracted by a subtraction (Left minus Right) network. In this design an op-amp circuit is used. The resulting signal is then amplified by 3dB. The signal then passes through an audio delay circuit, with a delay of between 10 and 30ms. On this unit the delay may be switched to either 12 or 24ms. If sounds come from both the front and surround speakers, those from the rear will be regarded by the brain as echoes and ignored. The delay also gives the sound field greater depth. Following this is a noise reduction circuit. This will reduce any stray low level high frequency sounds, while allowing louder sounds to pass unhindered. The delay and noise reduction circuits may also contain filtering components, to limit the bandwidth to the 100Hz to 8KHz specification.

This system is relatively simple, and the limitations are apparent. The delay and noise reduction systems are present to remove stray signals from the Surround channel, caused by slight imbalances and phase shifts in the replay system. The more advanced Dolby Pro-Logic system uses active signal channelling technology to ensure that any particular sound comes from one speaker only. This uses DSP (Digital Signal Processing) or similar technology, which is inevitably protected by copyrights and patents. This cannot be simulated in a form that is cost effective for home construction.

Other Uses

In this design the noise reduction and delay circuits may be independently switched out of the signal path. This may be useful for appreciating the effect and limitations of these sections.

With both sections switched out, the unit may be used to add surround effects to music. The sound is similar to that obtained from 1970s "quadraphonic" music centres, with the rear speakers connected in series between the positive terminals of the front speakers. This effect may not be to everyone's taste!

The unit can even be used to record Karaoke tapes, by adjusting the balance to cancel out the vocals. Again the delay and noise reduction should be switched off. Some bass may also be lost, but this probably won't matter too much if everyone is getting drunk!

PCB Construction

The circuit is constructed on a single sided PCB, which is available from Electronics in Action. The component overlay is shown in figure 6.

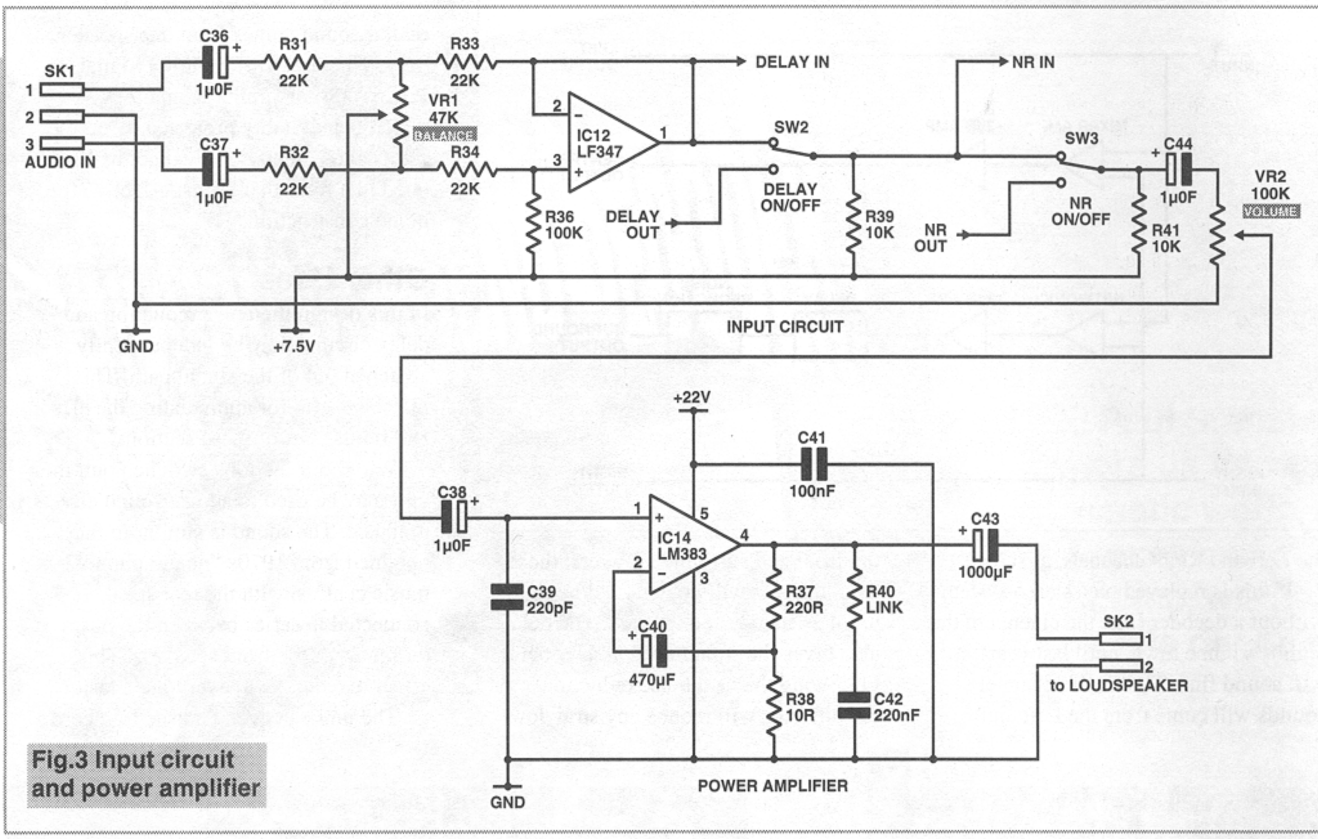


Fig.3 Input circuit and power amplifier

The Works

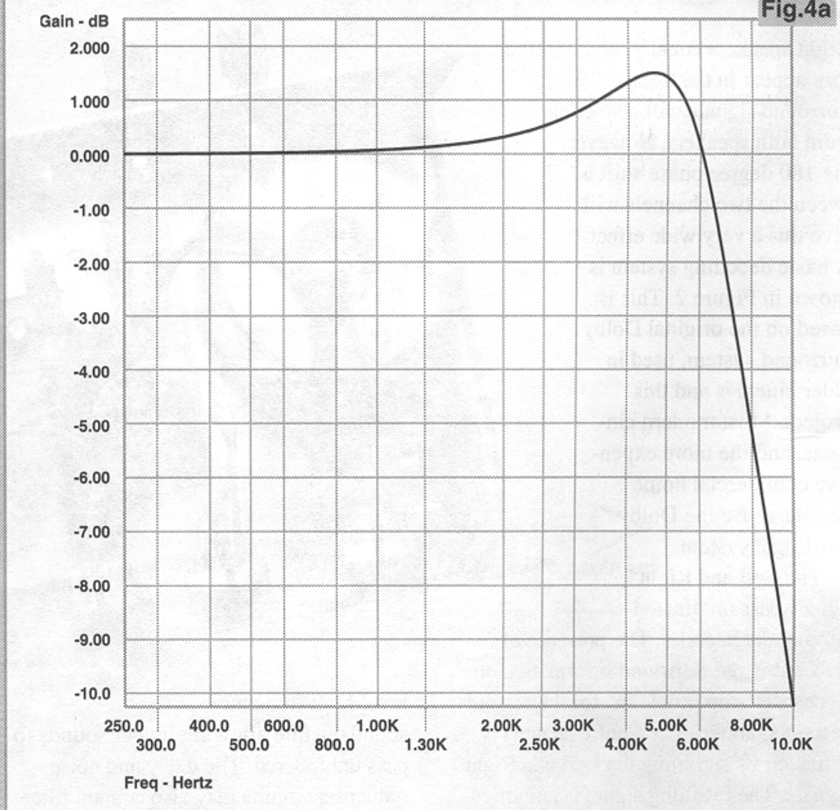
The delay circuit is shown in Figure 4b (logic) (A-D and D-A converters). This is similar to the delay section of the Digital Echo project, featured in last month's edition of EIA. IC1 is the clock generator, which runs at four times the sampling frequency. This unusual configuration gives an output with an approximately equal mark-space ratio. The A-D converter IC10, requires a negative bias on pin five. Since the current required in small, this negative voltage is obtained by rectifying the clock signal from IC1, giving approximately -4V.

The ZN448E (IC10) is an 8 bit successive approximation A-D converter. Nine clock cycles are required for each conversion. The eight data lines are tri-state, and controlled by the OE- pin.

The device contains a built-in clock circuit, the frequency of which is set by C26. The data sheet gives the maximum clock frequency as 1MHz, and a graph shows that a capacitor value of 100pF will give this. However tests with several devices showed that 100pF gave a frequency of only about 500KHz. To achieve 1MHz a 47pF capacitor is required. It is possible that some ZN448E devices will fail to operate with a 47pF clock capacitor. In which case, replace C26 with a 100pF component, or the lowest value that gives correct operation. R12 is the biasing resistor for the internal voltage reference, which is decoupled by C25. The ZN428E (IC11) is an 8 bit D-A converter, which uses an R-2R resistor network driven by an accurate voltage reference. This reference is biased by R15 and decoupled by C27. A latch circuit on the data lines is controlled by the EN- pin. When EN- is inactive, the previous output voltage remains.

The audio signals to the A-D converter and from the D-A converter pass through identical low pass filter circuits. These have a slope of 18dB per octave and a -3dB point at about 8KHz. The frequency response of these filter circuits is shown in figure 4a. The 1dB

Fig.4a



The Works

The audio section of the circuit diagram is shown in Figure 3. Input signals should be within the range 0.5 to 1.0 volts RMS, this is the usual level from the audio output (NOT Speaker) connectors on audio equipment.

The first section of IC12 is the Left minus Right amplifier circuit, with a gain in differential mode of about five. This gain will overcome the loss caused by the balance control and the 3dB encoding attenuation, and will give the correct level to drive subsequent stages.

The Balance control (VR1) is used to offset any difference in the level of the two channels.

This section is followed by the Delay and Noise Reduction switches (SW2 and SW3), and the Volume control (VR2). IC14 is the power amplifier circuit. The LM383 was chosen because it will drive two 4R speakers in parallel. The minimum load impedance in 1R6, and the power output is 8 Watts RMS into 8R. This device also has output short circuit and over temperature protection.

Fit the 22 link wires first, using thin (approx. 26SWG) tinned copper wire. The components may then be fitted in the usual size order. The non-polarised capacitors should have a lead pitch of 0.2", to match the holes in the PCB. Low cost ceramic or polyester devices are ideal. Use terminal pins for the off-board connections, since this will simplify the interwiring.

IC sockets may be used for the DIL devices if you wish. Opinion on this issue is varied, I feel sockets are unnecessary on single sided PCB's, because soldered IC's can be readily removed with a decent de-soldering tool. However, I would always use sockets in a PCB with plated through holes.

The voltage regulators and power amplifier IC's will become warm in operation, and should be mounted on a suitable heatsink or bracket. An 85mm length of aluminium angle was used on the prototype, as shown in the photographs, to transfer the heat to the rear panel of the case. The closest readily available bracket is the Maplin "15W Amplifier Heatsink Bracket" (order code YQ36P), although you will need to drill two extra holes.

The mounting tabs of all three devices are internally connected to 0V, so insulation washers are unnecessary. This should be the only point where the 0V rail connects to mains earth.

Case

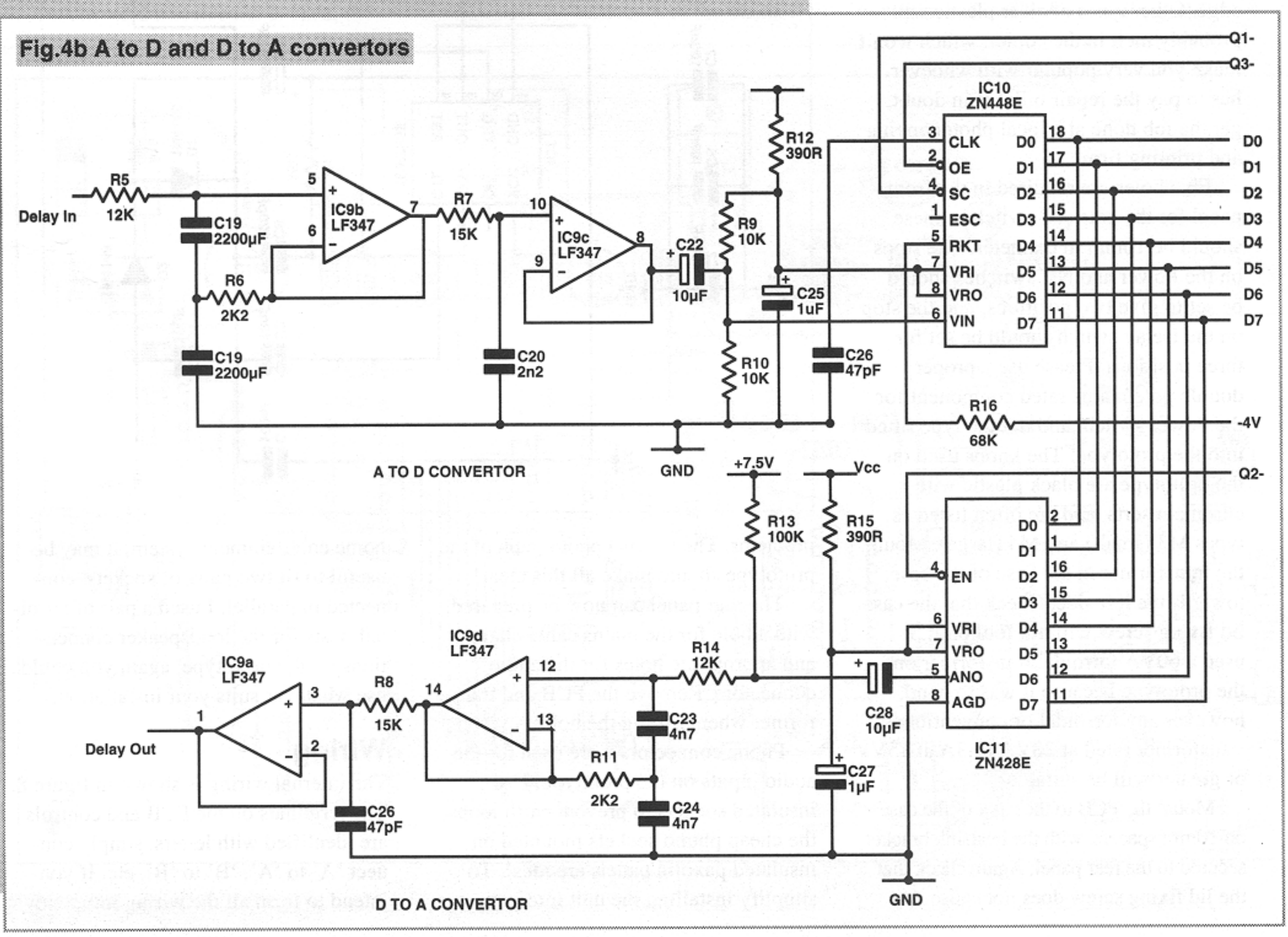
The prototype was constructed in an aluminium case, 280mm × 155mm × 75mm. This is often listed as type WB5. Before removing the protective film, check the panels are not scratched (Maplin, please take note). Once you have removed this film, you may not be able to return a damaged case for replacement.

A suitable front panel overlay is shown in figure 9. This may be photocopied (enlarge to 278mm × 75mm) onto transparent film, and fixed to the front of the box with clear self adhesive vinyl. A second paper copy can be used as a drilling template.

peak before the cut-off point is not important in this application.

The filter on the input to the A-D prevents frequencies too high to be accurately sampled, from reaching the converter. Allowing these signals through would result in severe aliasing distortion, leading to a poor quality output. The filter on the D-A output removes noise due to the sampling frequency, and smoothes the stepped output.

Fig.4b A to D and D to A convertors



The Works

IC2 produces the four timing pulses required, three of which are inverted by gates in IC3. When Q1 goes high, the A-D converter IC10 starts a conversion. When Q2 is high, the data in the RAM IC7 is sent to the D-A converter IC11, which produces the appropriate voltage. When Q3 is high, the data from the A-D is stored in RAM. Finally when Q4 pulses high, the address counter IC4 is incremented.

SW1 sets the amount of the RAM chip to be used to either 512 or 1024 bytes. This sets the delay to either 12mS or 24mS. I chose a 2K RAM chip because it is the cheapest readily available device. Observant readers may notice that the address line functions on IC7 are different to those shown in the 6116 data sheet. This was done to ease the PCB track layout. Static RAM address and data lines may normally be connected in any order, providing they are the same in the read and write circuits.

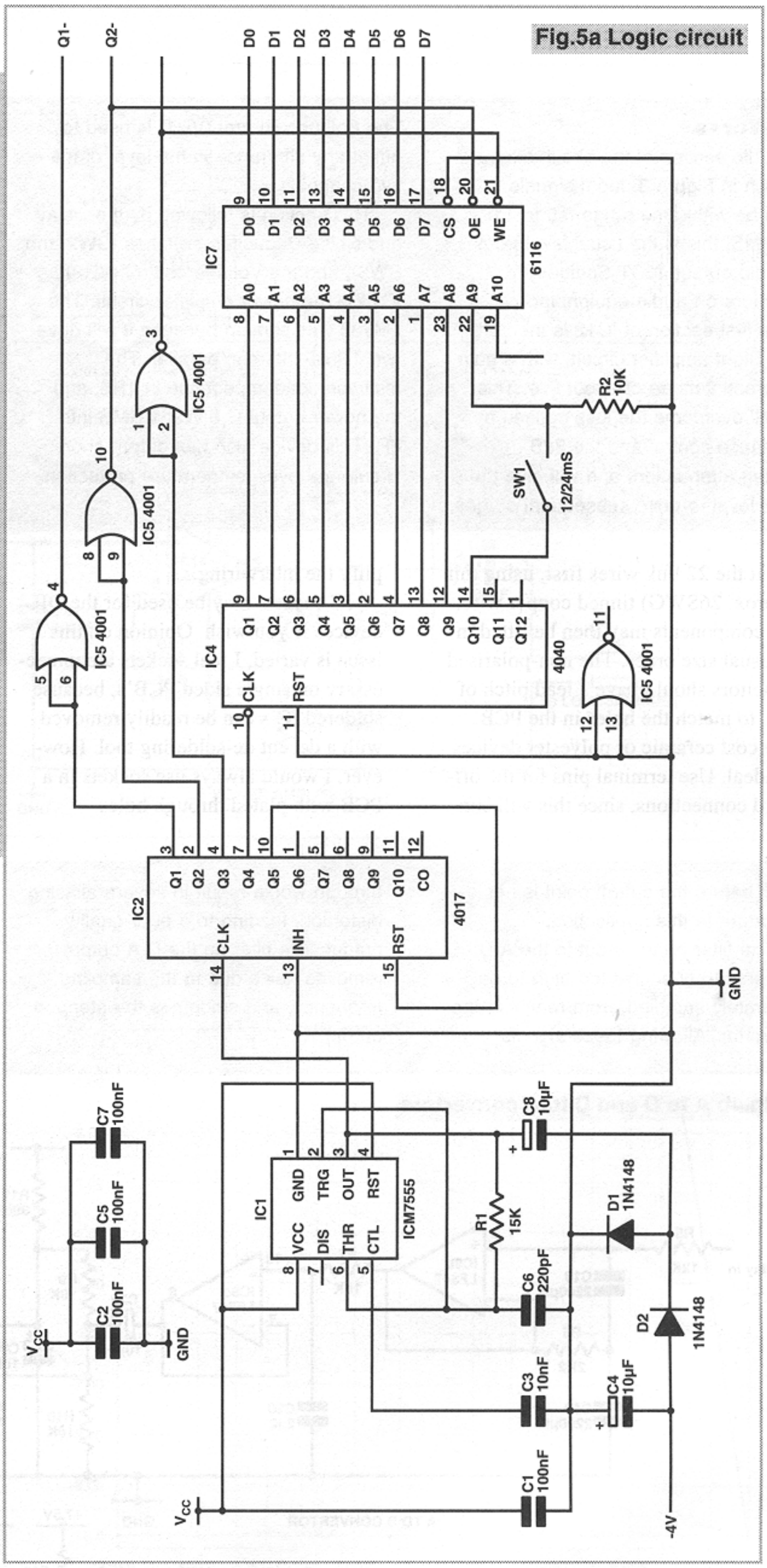


Fig.5a Logic circuit

When copying onto clear film, please make sure the film you use is suitable for photocopiers. The correct material is normally sold as photocopier overhead transparency film, and has a sheet of thin paper behind it, attached by one edge. Other types of clear plastic will probably melt in the copier, which won't make you very popular with whoever has to pay the repair bills! If in doubt, get the job done at a local photocopying and printing firm.

Five holes are required in the front panel for the pots and switches, these should be 10mm in diameter. The stops on the Power and NR switches should be set to give two positions, and the stop on the Delay switch should be set for three positions. Please use a proper double pole mains rated component for the Power switch, and not the type fitted into the prototype! The knobs used on the prototype are black plastic with chrome inserts, and are often listed as types M3 (small) and M4 (large). Mount the transformer in the base of the case, towards the left side. Check that the case lid fixing screw will not foul on it. I used a 60VA torroidal transformer on the prototype because it was to hand, however any torroidal or conventional transformer rated at 20VA (1.3A at 15V) or greater will be suitable.

Mount the PCB to the right of the case on 10mm spacers, with the heatsink bracket secured to the rear panel. Again check that the lid fixing screw does not cause any problems. The internal photograph of the prototype should make all this clear!

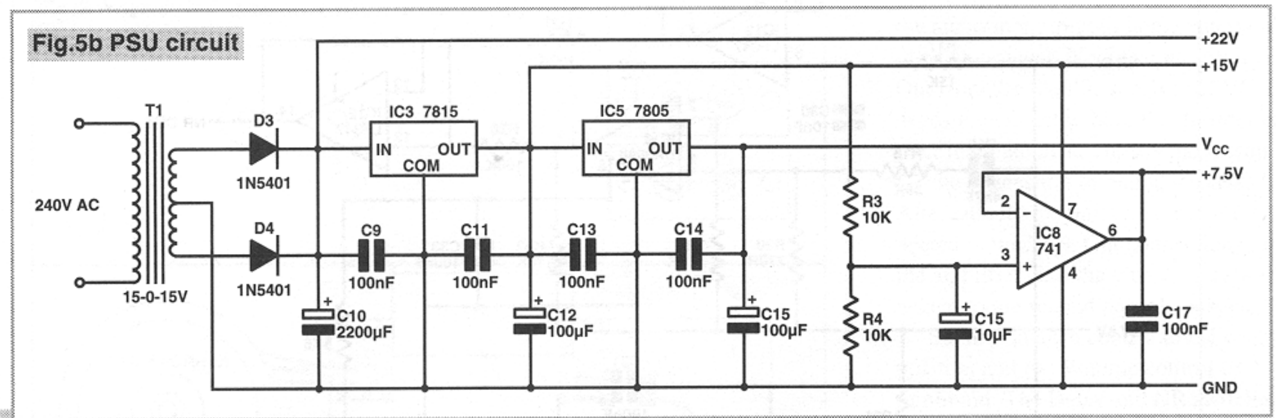
The rear panel can now be prepared, with a hole for the mains cable clamp, and appropriate holes for the audio connectors. Remove the PCB and transformer when drilling the holes.

Phone connectors were used for the audio inputs on the prototype. Use insulated sockets to prevent earth loops, the cheap phono sockets mounted on insulated paxolin panels are ideal. To simplify installing the unit into your home entertainment system, it may be useful to fit two pairs of sockets, connected in parallel. I used a pair of terminal posts for the loudspeaker connections on the prototype, again you could use whatever suits your installation.

Wiring

The internal wiring is shown in figure 8. The terminals on the PCB and controls are identified with letters, simply connect 'A' to 'A', 'B' to 'B', etc. If you intend to form all the wiring into a tidy

Fig.5b PSU circuit



The Works

The power supply circuit is shown in figure 5a. An unregulated 22V rail is used to drive the power amplifier, while regulated 15V and 5V rails power the remainder of the circuit. IC8 gives a mid-rail for the op-amp circuits. The transformer should have a rating of at least 20VA. A 10nF Class X capacitor is connected across the primary on the prototype, to remove the crackle sound from the speakers when the unit is being switched off.

loom, you may need to use screened cable for the signal connections, due to the extra length and the nearness of the cables. No problems were experienced using ribbon cable for the short point to point on the prototype.

If a torroidal transformer is used, the two primary windings must be wired in series for 240V operation. The mains input should be connected between the brown and blue transformer tails, and the grey and mauve wires should be connected together (check the label on the transformer first). On the prototype this was done on a pin of an unused section of the power switch. However since you will be using the correct type of switch (please), you will need to solder the two tails together and cover the joint with insulation sleeves.

C45 must be a Class X rated component, suitable for direct connection across the mains. If you use any other type of capacitor, it will probably explode sooner rather than later. Mount the capacitor on the switch terminals, in parallel with the transformer primary.

Connect the earth core of the mains flex securely to the metal case. No mains fuse was used on the prototype, a 3A fuse was fitted in the 13A mains plug. There is space on the rear panel to fit a fuse holder you wish. I would suggest a 250mA anti-surge fuse (T-250mA), since the initial current surge of a torroidal transformer will blow a fast-blow fuse.

The transformer used in the prototype had four tails for the secondary windings. Connect both the red and orange wires to the centre pin on the PCB (0V). The yellow wire is then connected to one of the outer pins, and the black wire to the other. Again, check the label on the transformer first.

Initial Testing

If you have a test meter, it would be a good idea to check the following voltages, before finally installing and

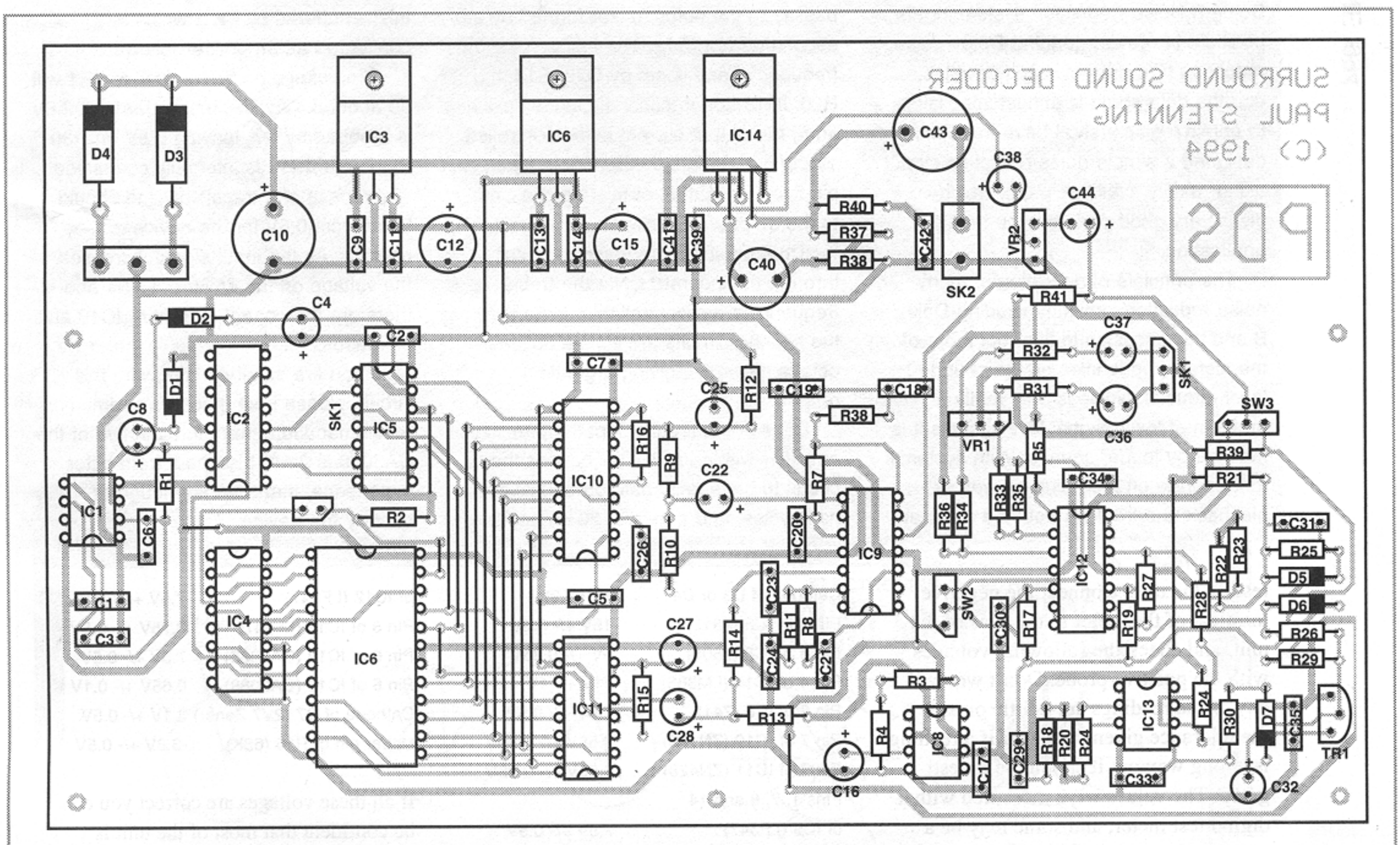


Fig.6 Component positioning

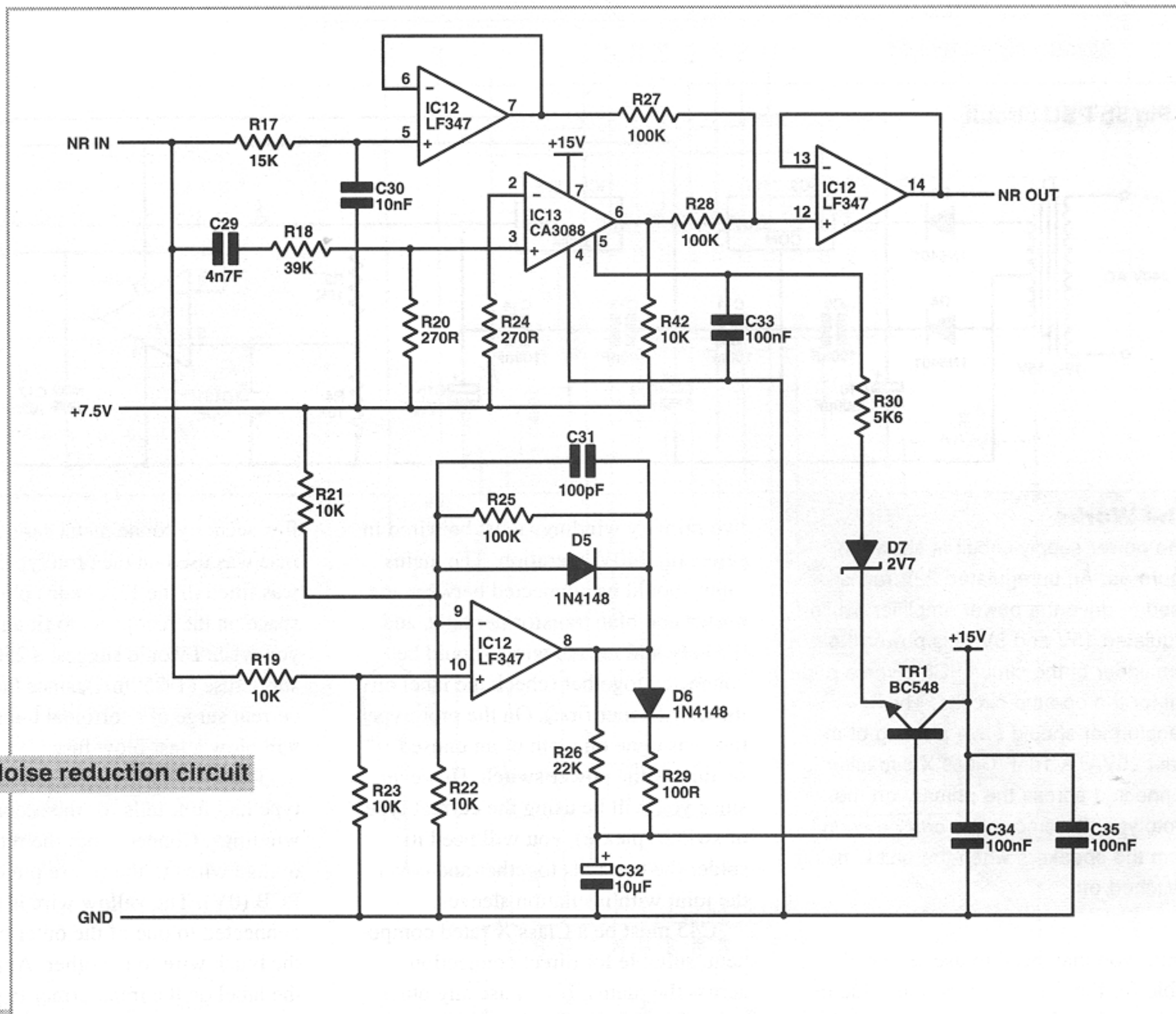


Fig.7 Noise reduction circuit

The Works

Figure 7 is the circuit diagram of the noise reduction system. A Dolby licensed decoder would use a modified Dolby B Noise Reduction system in this position. However, genuine Dolby noise reduction IC's are not readily available, and the data sheet is almost impossible to obtain legitimately. I have therefore designed a simple noise reduction circuit using readily available parts, which gives very good performance in this application.

The principle of operation of many noise reduction circuits, including Dolby B and this circuit, is to filter out more of the higher frequencies as the sound level reduces. Noise is only really a problem at lower signal levels, since it is masked by louder sounds. With systems that operate on recording as well as playback, the treble is boosted at lower levels for recording. The input signal is split into two bands. R17 and C30 set the lower frequency band. This is buffered by one part of IC12, and then passes to the output mixer formed by a second part of IC12. The higher frequency band is set by C29, R18 and R20. IC13 is a transconductance op-amp, the output current being controlled by the current into the control input on pin five. The output of this also passes to the output mixer circuit.

Thus the lower frequencies pass through unhindered, while the treble frequencies are controllable. Note that the filtering circuits are simple 6dB per octave networks, giving a gentle response.

D5 is in the feedback path of an amplifier within IC12. This causes the circuit to have unity gain on negative half cycles, and a gain of 20 on positive half cycles. Thus, the circuit acts as a half wave rectifier. The biasing gives an output voltage of about 4V with no signal. C32 stores the peak output level, this is buffered by TR1, which is configured as an emitter follower.

The voltage on the emitter of TR1 will be about 3.4V with no signal. About 2.6V is dropped by D7, leaving 0.8V on R30. Pin five of IC13 is internally connected to the base of a transistor, and should be at least 0.6V for the device to operate. As the input signal increases, the voltage on the emitter of TR1 and therefore the control current to IC13 also increases.

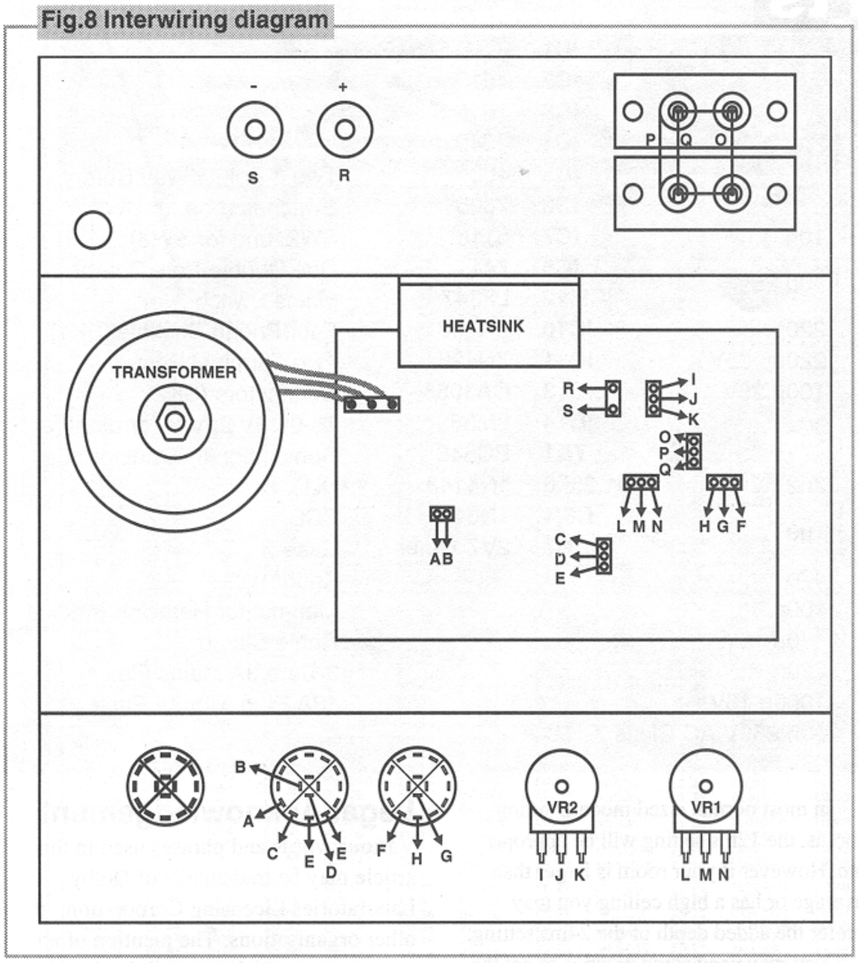
If you are experimenting with this circuit, please note that the maximum recommended current into pin five of the CA3088 is 2mA. I learned, from bitter experience, that 5mA is enough to destroy the device.

testing the unit. Connect the negative terminal of the meter to the case of the unit, and check the following voltages with the positive probe. Don't worry if any of the readings are a little outside the tolerance given, however if a reading is a long way out it should be investigated. The values were measured with a digital test meter, and some may be a little lower if an analogue meter is used.

Cathode of D3 or D4	22V +/- 2V
Pin 3 of IC3 (7815)	15V +/- 0.5V
Pin 3 of IC6 (7805)	5V +/- 0.25V
Pin 4 of IC14 (LM383)	11V +/- 1V
Pin 6 of IC8 (741)	7.5V +/- 0.5V
Pin 7 of IC10 (ZN448)	2.56V +/- 0.05V
Pin 7 of IC11 (ZN428)	2.56V +/- 0.05V
Pins 1, 7, 8 and 14 of IC9 (LF347)	7.5V +/- 0.5V
Pins 1, 7 and 14 of IC12 (LF347)	7.5V +/- 0.5V
Pin 8 of IC12 (LF347)	3.75V +/- 0.5V
Pin 6 of IC13 (CA3088)	7.5V +/- 0.5V
Pin 5 of IC13 (CA3088)	0.65V +/- 0.1V
Cathode of D7 (2V7 Zener)	3.1V +/- 0.5V
Upper pin of R16 (68K)	-3.8V +/- 0.5V

If all these voltages are correct you can be confident that most of the unit is working correctly. The digital section

Fig.8 Interwiring diagram



cannot be checked easily, but you will know soon enough if any of this is faulty.

Check that the heatsink is not getting too hot - it should be barely warm. A hot LM383 is a sign of instability - this is very unlikely but worth checking. If it is unstable, replace C41 and C42 with polyester capacitors, and thicken the tracks between C39, C41, C42 and the LM383 by soldering tinned copper wire along them.

You can now install the unit and test it, as detailed below.

Installation and Use

To set-up your viewing area, place one speaker either side of your television and two speakers behind your head, towards the rear corners of the room (if only one speaker is used its position will be heard). The effect may be a little better if the rear speaker(s) are further from the viewer than the front speakers.

The rear speakers do not have to be of hi-fi quality, since the rear channel bandwidth is only 100Hz to 8kHz. On the other hand though, don't use the nasty plastic or hardboard cased offerings that are supplied with many cheaper stereo systems. I am using a pair of solid wooden cased speakers that came with a defunct Ferguson stereo music centre from a local car boot sale, and these sound superb.

A similar comment applies to the cable for the rear speakers. The unit certainly does not need specialist hi-fi speaker cable, costing several pounds per metre. However since the cable run is likely to be fairly long it would be a good idea to choose something heavier than doorbell wire. 5A twin mains flex is a sensible choice, and was used successfully with the prototype. Connect the audio input to your home entertainment equipment. Note that the Audio Out from the satellite receiver or VCR needs to connect to both the decoder and your main amplifier, this is the reason for the suggestion of paired sockets. Alternatively the unit could be connected to the Tape Out connections on the amplifier, and the tape deck connected to the second pair of sockets.

Set the Balance control to the centre position and the Volume control to minimum. The Delay and NR switches should be in the off positions. Set the main amplifier volume to minimum for now, and switch on the surround sound decoder and the satellite receiver or stereo video recorder.

Choose a mono program (if you have satellite try Sky News or CNN), and advance the volume control to about the 10 O'clock position. If you adjust the balance control towards either end, you should hear the sound from the rear speakers. When the balance control is adjusted towards the centre there should be virtually no sound from the rear speakers.

Leave the balance control at this position, and select a stereo film channel (Sky Movies or The Movie Channel). If the film is in Dolby Stereo, you should get a significant amount of sound (but not speech) from the rear channel. If you can't get this to work, check your satellite receiver is set to the main stereo audio channel (AU1 on Amstrad receivers), and check the film is really in Dolby Stereo.

Now switch the NR on. You should notice a significant drop in the treble on quieter sounds, while louder sounds will be unaffected.

Switch the Delay to 12ms. You may notice a slight drop in the treble response, due to the filter circuits. Switching to 24ms should cause no noticeable effects. The full effect of the delay will not be evident until the front amplifier is turned up.

Set the volume on the main amplifier to your usual listening level. Now adjust the level of the rear channel to a level where the rear sounds contribute to the complete sound, without intruding.

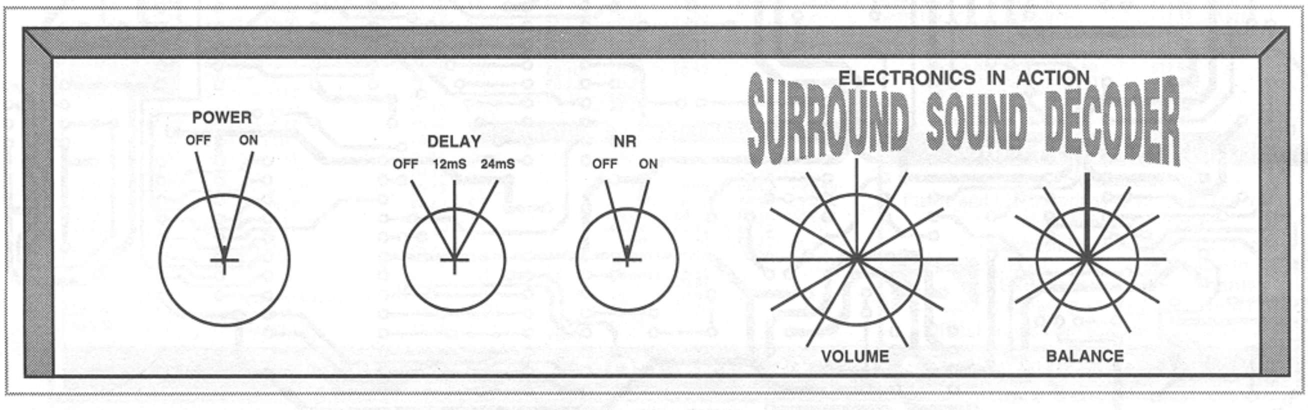


Fig.9 Front panel layout (enlarge this by 166% on a photocopier)

Parts

Resistors

(all 0.25W 5% or better)

R1,7,8,17	15K
R2,3,4,9, R10,19,21, R22,23, R39,41,42	10K
R5,14	12K
R6,11	2K2
R12,15	390R
R13,25,27, R28,35,36	100K
R16	68K
R18	39K
R20,24	270R
R26,31,32, R33,34	22K
R29	100R
R30	5K6
R37	220R
R38	10R
R40	LINK
VR1	47K Lin Pot
VR2	100K Log Pot

Capacitors

C1,2,5,7,9, C11,13,14, C17,33,34, C35,41	100n
C3,30	10n
C4,8,16, C22,28,32	10µ 25V
C6,C39	220p
C10	2200µ 35V
C12,15	100µ 25V
C18,19,23, C24,29	4n7
C20,21	2n2
C25,27,36, C37,38,44	1µ0
C26	47p
C31	100p
C40	470µ 16V
C42	220n
C43	1000µ 16V
C45	10n 250V AC Class X

Semiconductors

IC1	ICM7555 (CMOS 555)
IC2	4017
IC3	7815
IC4	4040
IC5	4001
IC6	7805
IC7	6116
IC8	741
IC9,12	LF347
IC10	ZN448
IC11	ZN428
IC13	CA3088
IC14	LM383
TR1	BC548
D1,2,5,6	1N4148
D3,4	1N5401
D7	2V7 Zener

Additional Items

Two 4 Pole 3 Way Rotary Switches (one for SW1 & SW2, one for SW3)
One Double Pole Rotary Mains Switch
Four Phono Sockets (SK1)
Two Terminal Post Connectors (SK2)
15-0-15V 20VA Torroidal (or Conventional) Transformer (X1)
PCB
Case
Knobs
Material for Heatsink Bracket
Cable Clamp
3 Core 3A Mains Flex
13A Plug with 3A Fuse

Speech should still appear to come from the front, while crowds, music and dramatic sound effects should fill the room.

The effect of the delay switch will now be more apparent. With no delay the sound will seem a bit flat, and the rear channel may be distracting. Switching to 12ms will give the sound more depth, and you may want to turn the surround volume up a bit more! Switching to 24ms will add more depth - this will probably give a slight distracting echo effect unless you have a large lounge.

In most normal sized modern sitting rooms, the 12ms setting will be appropriate. However if your room is larger than average or has a high ceiling you may prefer the added depth of the 24ms setting.

You may need some practice to get the best results from the unit, it's really a matter of trial and error, but it's worth the effort! The results from this unit can be stunning. It really does add a new dimension to home entertainment, the effect on some modern films is dramatic. It almost makes the Sky film channel subscription charges seem good value!

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