Contents

SURROUND SOUND PROCESSOR

Based on a design by D. Laues

The processor described in this article expands the sound of an existing stereo TV receiver or audio installation with a centre channel and a surround channel. It does not use special Dolby ICs. Additional output amplifiers are not needed, because they are provided in the processor.

Creating surround sound in a domestic room can be approached in two ways: by a processor that generates the four signals required: left-hand, right-hand, centre and surround, or by one that adds the two missing channels, that is, centre and surround, to the existing stereo sound. The first is the most elegant, but also the most complicated and most expensive. Moreover, it requires an additional line to return the left-hand and right-hand signals to the input of the TV/audio equipment. The second way is much more straightforward and has proved in practice to give an excellent spatial effect. Moreover, it can be accomplished in a compact and fairly inexpensive unit. The design in this article is of the second kind.

The basic setup in a domestic room is shown in Fig. 1. The left-hand and righthand channels are reproduced as before, that is, via the loudspeakers in the TV receiver, as shown, or by those of the audio installation to which the TV receiver is connected. The extra items are the processor and three loudspeakers. The inputs of the processor are linked, possibly via the SCART connector, with the line out terminal of the TV receiver, or audio amplifier, while the extra loudspeakers are connected to the outputs of the processor. The processor contains two integral amplifiers each of which provides 20 W output into 4 Ω: quite sufficient for the centre and surround loudspeakers.

As briefly discussed in last month's article on Dolby surround sound, the additional loudspeakers need be no more than compact (bookcase type) hi-fi types that are not too expensive. If possible, however, choose types whose efficiency is about the same as that of the main loudspeakers: this affords rather more freedom when the system is set up as a whole.

Basic design

A block diagram of the basic design is given in Fig. 2. The design of the processor is similar to that of the active decoder discussed in last month's article on Dolby surround sound. There are some differences between the two, since the present processor not only functions as a decoder, but also provides the signals for the centre and surround channels and contains two output amplifiers.

As explained last month, the two additional channels are processed (whence the name of the unit) from the sum and difference of the two stereo channels. The centre channel is the simpler to produce, because it suffices to add the left-hand and right-hand channels together and apply the sum to a suitable output amplifier via a voltage-controlled amplifier (VCA).

To produce the surround signal, the right-hand signal is deducted from the left-hand signal (or vice versa) and the resulting signal is applied to a delay network via a low-pass filter. The delay can be preset between 10 ms and 30 ms. The signal is then applied to an expander via a band-pass filter. The expander is essential because the surround signal is compressed during recording. The output of the expander is applied to a second output amplifier via a VCA.

The dynamic compensating network, in conjunction with the VCAs, reflects the difference between an active and a passive (matrix) design. In this network, the correlation between the two stereo channels is analysed continuously. The results of the analysis are converted into control signals for the VCAs which constantly adjust the levels of the centre and surround signals. This arrangement ensures a much larger channel separation than possible with a passive design.

Centre channel

From the inputs of the left-hand and right-hand channels, K_1 and K_2 in Fig. 5, the signals are taken via buffer amplifiers IC_{1a} and IC_{1b} to R_{51} and R_{52}, and then summed in preset P_6. From there, the signal is applied to pin 5 of IC_9. This IC contains four electronic potentiometer circuits (of which only two are used) which function as VCAs; their amplification is governed by a control voltage at pins 9 and 10. The output of one of the circuits is available at pin 7, from where it is applied to output amplifier IC_{10}. This circuit provides an output of up to 20 W into 4 Ω.

The output of IC_{10} is applied to the centre channel loudspeaker via relay contact Re_{1b}. The relay is controlled by a simple delay circuit, T_1, and obviates any clicks and plops in the speaker caused by the switching on and off of the processor.

Surround channel

The signals at the outputs of IC_{1a} and IC_{1b} are also applied to the inverting and non-inverting inputs of IC_{2b} respectively, so that the output of the op amp is the difference of the two stereo signals (L–R). The difference signal is applied to a 4th-order low-pass filter based on IC_{2a}, which limits its upper bandwidth to 7 kHz. This anti-aliasing filtering serves to obviate the formation of spurious mixing products of the signal and the clock of the following delay line, which is based on IC_4. This IC is a 2048-stage bucket brigade device. The rate at which the internal electronic switches are operated is

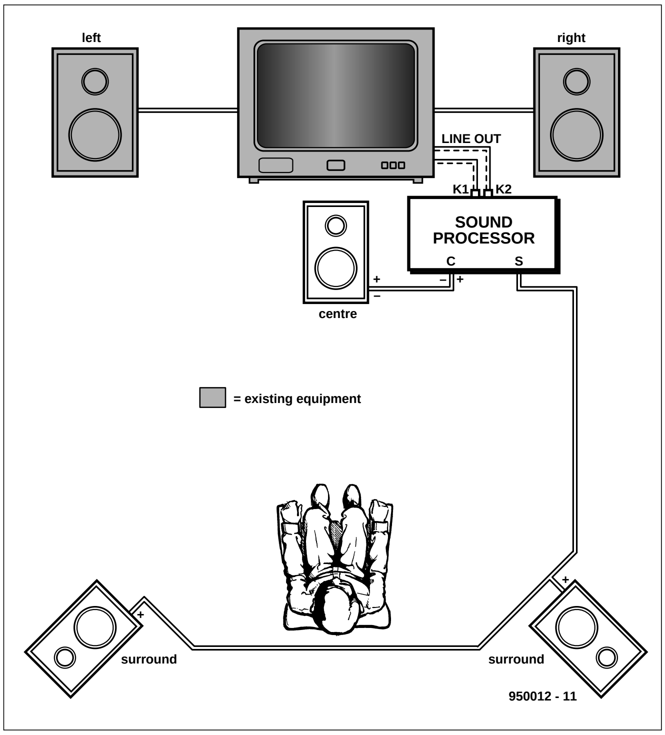


Fig. 1. Basic setup of a surround sound system in a living room.

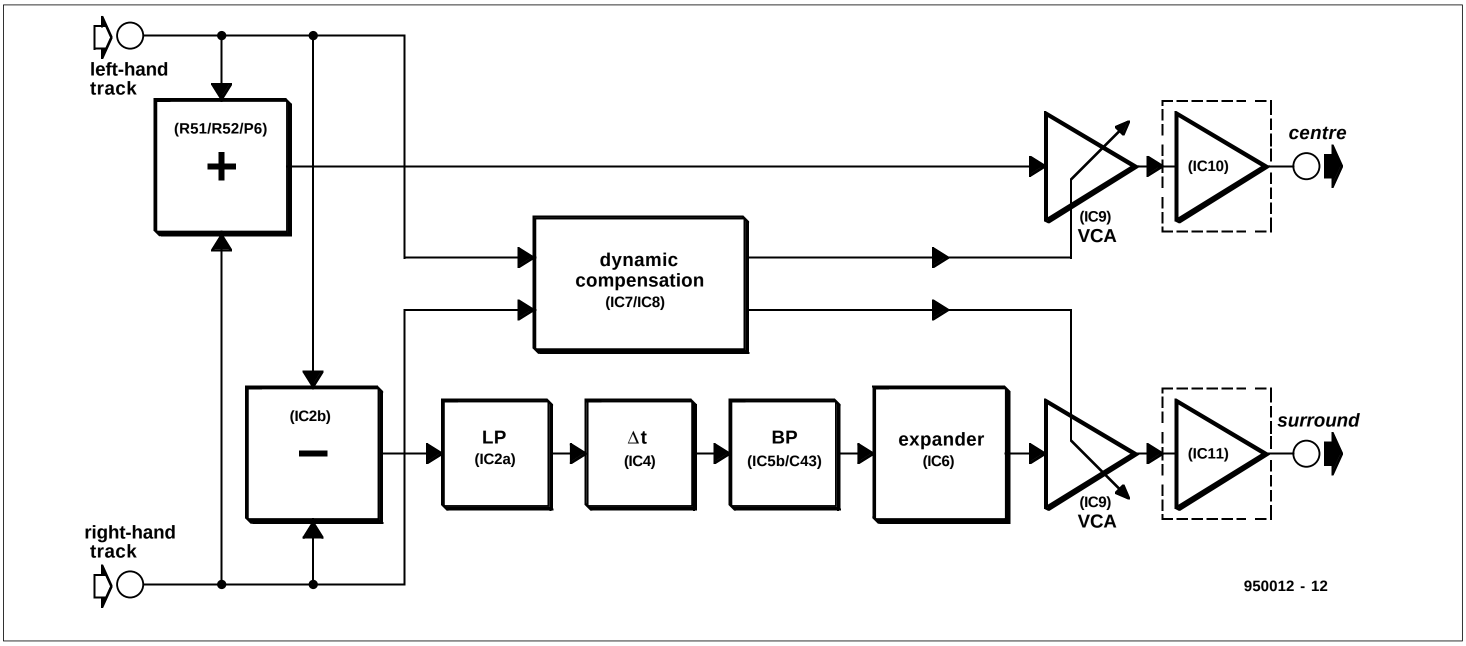


Fig. 2. Block diagram of the surround sound processor.

determined by IC_3. This CMOS-IC is designed especially to generate a low-impedance, double-phase clock. The specified values of its frequency-determining components, R_{15}, R_{16}, C_9 and P_1 allow a delay between 10 ms and 30 ms to be set with P_1.

The outputs of IC_4 are applied via buffer IC_{5a} to a low-pass filter based on IC_{5b} (identical to that based on IC_{2a}) which filters out any residue of the clock signal. The cut-off frequency is 7 kHz. The signal is subsequently fed to compander IC_6, whose input network contains a high-pass filter, R_{55}-C_{43}, the specified values of which give a lower cut-off frequency of about 50 Hz. The overall effect of the low-pass and high-pass filters is, of course, that of a band-pass filter as shown in Fig. 2.

The compander IC contains two circuits each consisting of a rectifier, a variable gain cell and an op amp. In the present processor only one of these circuit is used and that as an expander. The values of external components R_{27}–R_{32} and C_{23}–C_{29} allow for an expansion factor of 1:1.3.

The surround signal is then applied to the second electronic potentiometer circuit in IC_9, whose output is available at pin 17. From there, the signal is fed to output amplifier IC_{11}, whose amplification is identical to that of IC_{10}. The output of IC_{11} is applied to the surround loudspeaker(s) via a second contact on Re_1.

Dynamic compensation

The outputs of buffers IC_{1a} and IC_{1b} are also applied to twin comparators IC_{7b} and IC_{7c} via C_{31} and C_{32}. The output of each of these comparators is a rectangular voltage the frequency of which is a measure of the variation in the relevant input signal. Both outputs are applied to XOR gate IC_{8c}. (Remember that an XOR gate has an output only when its inputs are dissimilar). Integration of the output pulses of the gate by R_{37}-C_{36} results in a direct voltage whose amplitude is a measure of the phase difference between the two stereo signals.

This direct voltage is applied via IC_{7a} (inverted) and IC_{7d} (non-inverted) to the control inputs (pins 9 and 10) of IC_9. This arrangement ensures that when a mono signal is present at the inputs (no or hardly any phase difference), the amplification of the VCA controlling the centre channel is raised. Conversely, when a surround signal is present (large phase difference), the amplification of the VCA controlling the surround channel is increased.

The degree to which the amplification of the VCAs is influenced by the control signals is preset by P_2 and P_3. When the wipers of these controls are at earth potential, the amplification is fixed; when they are at the opposite end of their travel, control is maximum.

Presets P_4 and P_5 serve to shift the operating point of the VCAs to some extent. They thus make the preset range wider and, in fact, support the operation of P_6 and P_7. If, for instance,

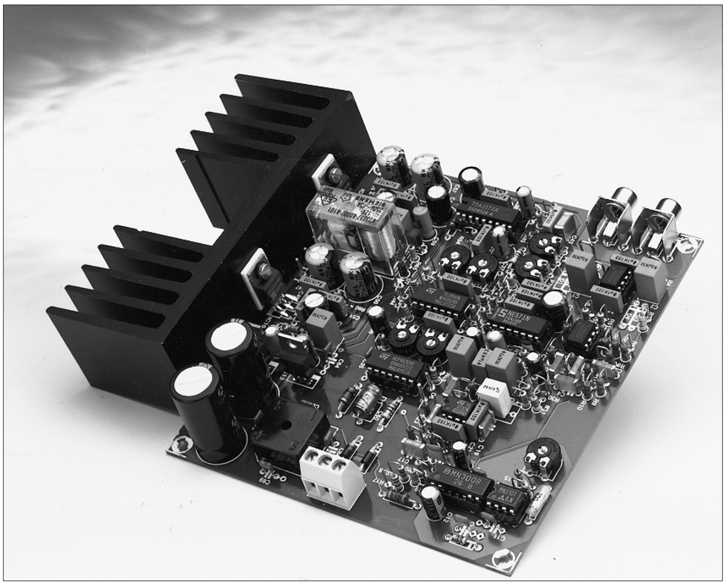


Fig. 3. Completed prototype board.

Fig. 4. Completed prototype processor.

Contents

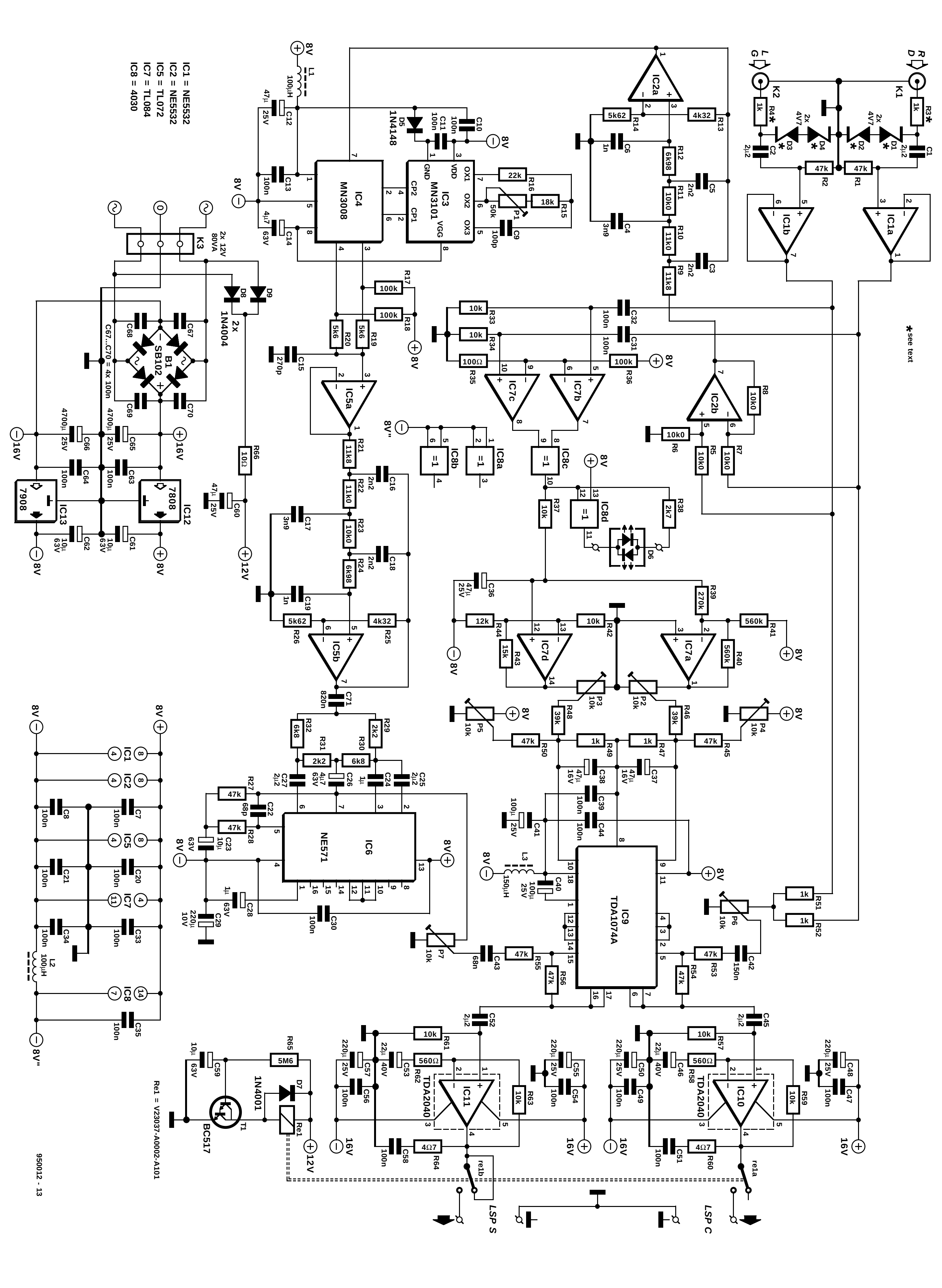


Fig. 5. Circuit diagram of the surround sound processor.

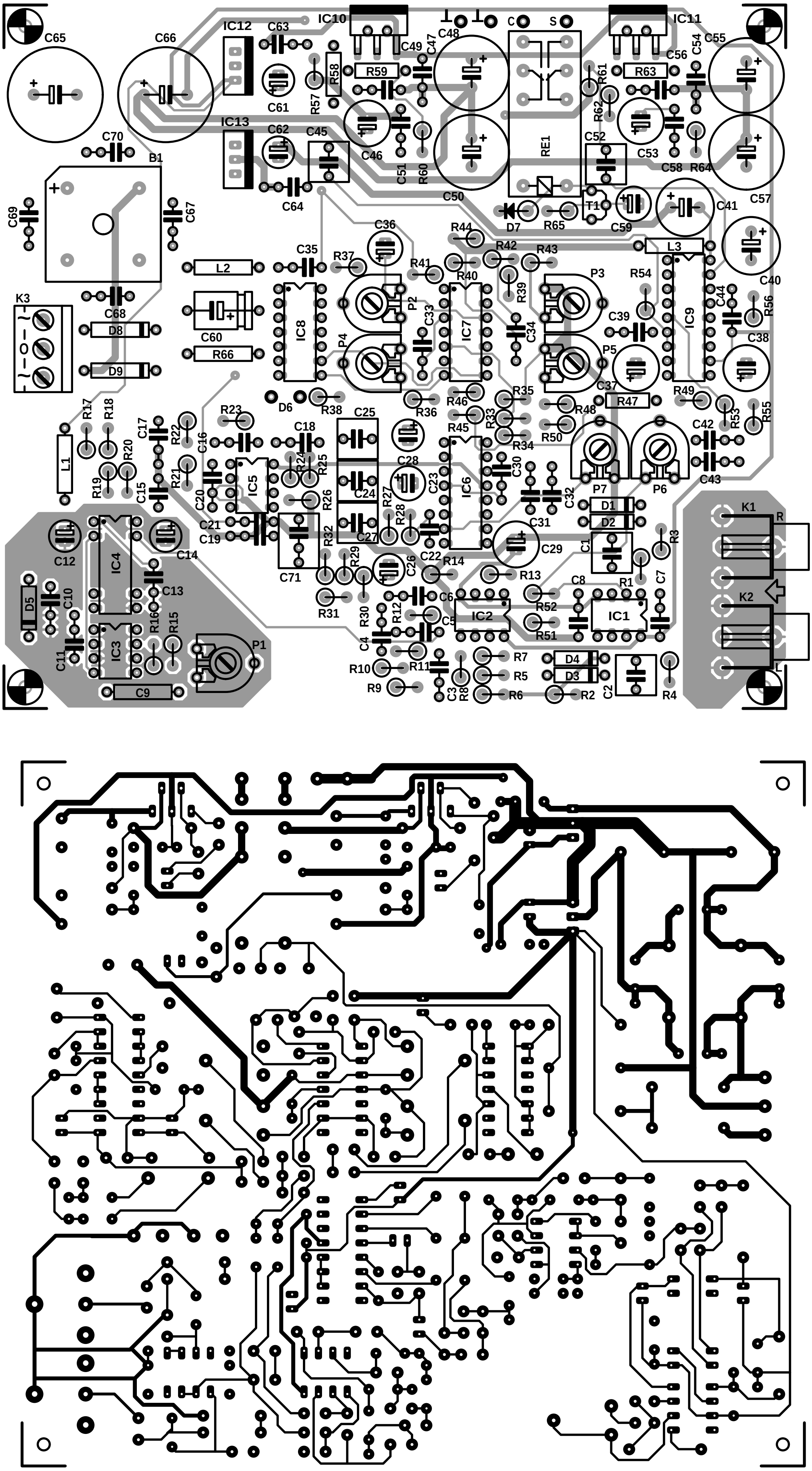


Fig. 6. Printed-circuit board for the surround sound processor. (See also next page)

P_7 has already set the surround level to maximum, P_5 enables this to be increased slightly. The same applies to P_4 insofar as the level of the centre channel preset with P_6 is concerned.

The currents through R_{45} and R_{46} and those through R_{48} and R_{50} are simply added together: there is, therefore, no interaction between P_2 and P_4 nor between P_3 and P_5.

Further circuit details

Resistors R_3 and R_4 and diodes D_1–D_4 limit the level of the input signal to a safe value and are imperative if the stereo signals are taken from the loudspeaker outputs of the TV receiver. Note that even line out terminals sometimes provide a signal at a level well above 1 V. If it is absolutely certain that the line output level is 1 V, and this is the only input, the resistors can be replaced by a wire bridge and the diodes may be omitted.

Bi-colour LED D_6 functions as a kind of signal monitor that shows the change from surround channel to centre channel and vice versa. In the case of a surround signal, the output of IC_{8c} is high. Since one output of IC_{8d} is at +8 V, both inputs of this XOR are then high, so that its output is low. This results in the red segment of D_6 lighting. In the case of a centre signal, pin 12 of IC_{8d} is low, so that its output is high, resulting in the green segment of D_6 lighting. In practice, the changes between the signal are so rapid that the LED shows a fluent transition from red to green and back to red again.

The power supply provides three different voltages. The secondary of the mains transformer is connected to K_3. The 12 V input, after rectification and smoothing, results in a symmetrical supply of ±16 V, which is used to power output amplifiers IC_{10} and IC_{11}.

From the ±16 V lines, voltage regulators IC_{12} and IC_{13} derive a supply of ±8 V, which is used to power the remainder of the circuit.

The 12 V line for the relay is taken directly from K_3 and rectified by D_8-D_9.

Construction

The processor is best constructed on the PCB illustrated in Fig. 6. Since this board is double sided and through-plated, it is not possible to make it without special tools and equipment.

Populating the board is straight-

Contents

forward and should not present undue difficulties. Note that when the specified enclosure is used, capacitors C_{65} and C_{66} must be not higher than 38–40 mm.

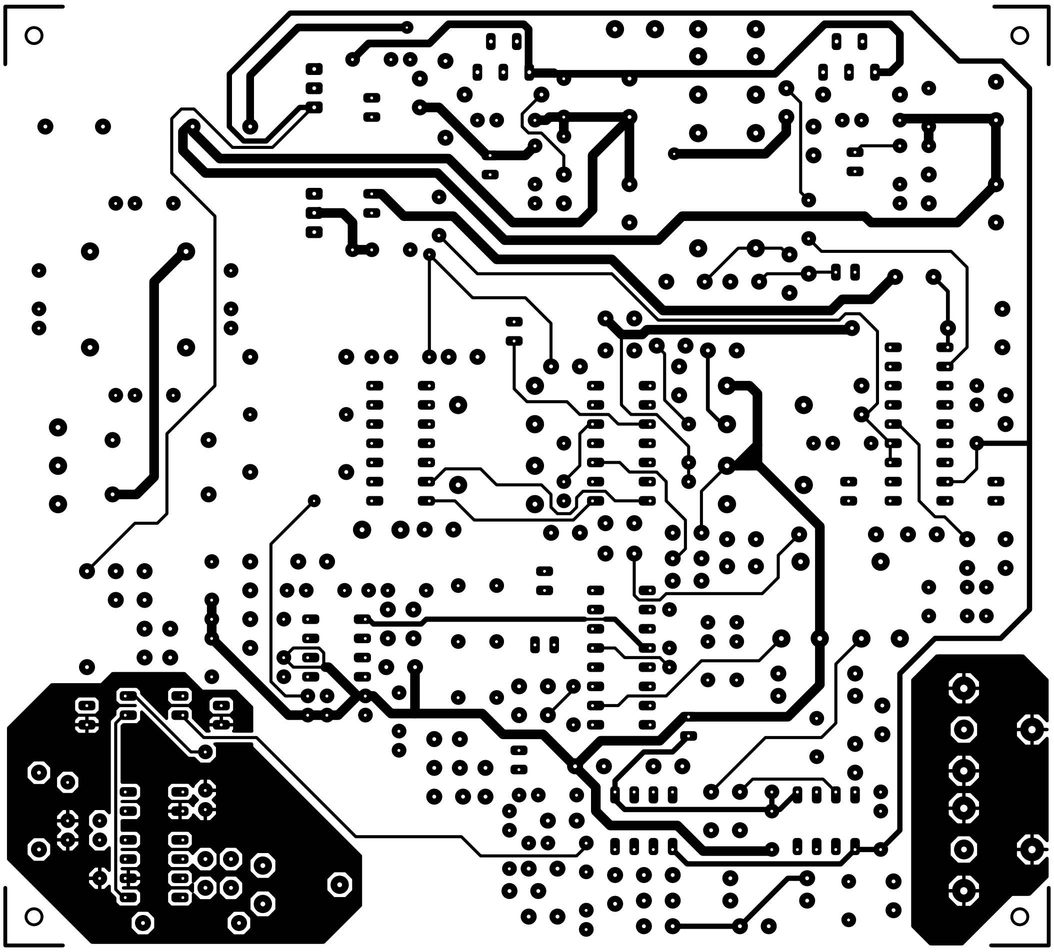
Use gold-plated connectors for K_1 and K_2 to ensure good, lasting connections. The positions for these connectors are at the edge of the board so that all that is necessary when fitting the board into the enclosure is drilling a few holes in the back panel through which these sockets can protrude. Note that they must not touch the enclosure.

Output amplifiers IC_{10} and IC_{11} are located at the edge of the board so that they can be fitted readily to a heat sink. The ICs must be electrically isolated from the heat sink by ceramic washers and heat conducting paste.

The photograph in Fig. 3 shows the completed prototype board.

The main requirements of the enclosure are that it is made of metal and that it provides adequate space for the finished board.

Apart from K_1 and K_2, fit suitable sockets or spring-loaded terminals at the back of the enclosure for connecting the centre and surround loudspeakers. Link these terminals with heavy-duty insulated wire to points 'C' and 'S' and the adjacent earthing points on the board. The specified mains entry with integral fuse holder should also be fitted at the back of the enclosure.



The mains on/off switch and D_6 should be fitted at the front panel (for which a ready-made foil is not available). The diode should be connected to the relevant points on the board via lengths of flexible stranded wire.

Finally, link the centre pin of K_3 to the mass of the enclosure with the aid of a solder tag.

The completed prototype is shown in the photograph of Fig. 4.

The output amplifiers are suitable for operation with load impedances ≥ 4 Ω, but not lower ones. Thus, for the surround channel, two 8 Ω loudspeakers may be connected in parallel only if it is absolutely certain that the impedance is 8 Ω. If it is not, connect the speakers in series: this is safer. It is essential that the two speakers are in phase: the + terminals must go to the same terminal on the board: whether this is earth or 'S' does not matter.

The centre loudspeaker must be in phase with the main speakers. Since the relevant VCA functions as an inverter, the + terminal of this speaker must be connected to the earth point on the board; the –terminal to point 'C'.

Calibration

Start by setting presets P_2–P_7 to the centre of their travel, and P_1 to maximum (fully clockwise).

Inject a (mono) speech signal and adjust P_6 until the sound appears to come from the centre speaker. Wait for a surround signal (indicated by the red segment of D_6 lighting) and turn P_7 till sound emanates from the surround speakers. Do not set the level too high, because this leads quickly to an exaggerated effect. If, however, it is felt that the desired level can not be obtained with P_6 or P_7, as the case may be, adjust P_4 or P_5, or both, as required.

Next, create a spatial effect by slowly turning P_2 and P_3 until the centre channel and the

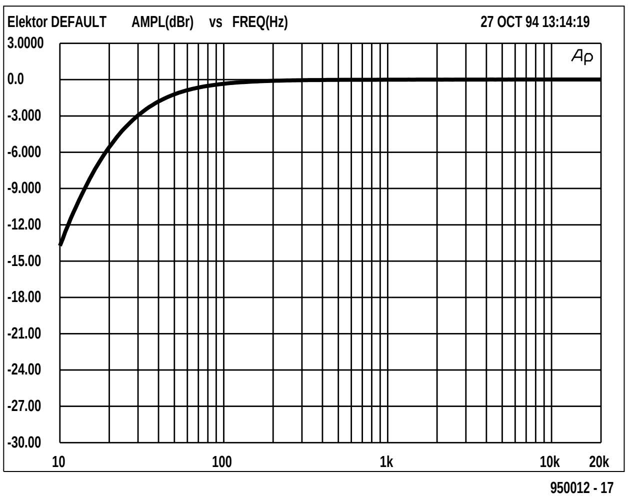


Fig. 7. Frequency response of the centre channel.

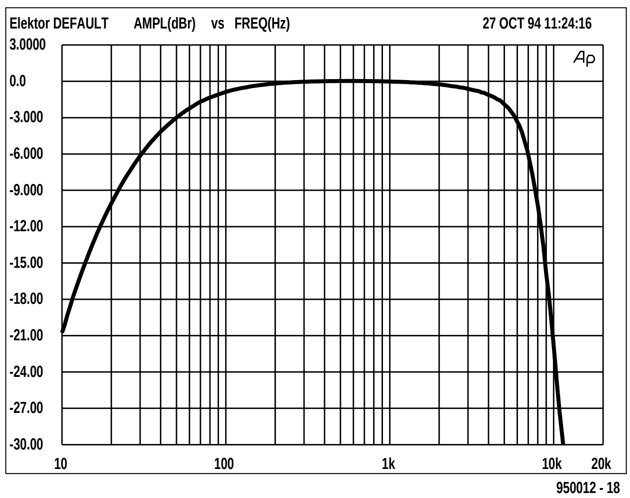


Fig. 8. Frequency response of the surround channel.

surround channel seem well ‘separated’. It is more than likely that P_6 and/or P_7 must then be readjusted. Note that these controls give an instinctive ‘wrong’ feel: turning them clockwise *reduces* the level.

If the surround speakers give exaggerated reverberation, reduce the delay with P_1. In the average living room, a delay of 25 ms appears correct: this corresponds to P_1 being almost at its maximum setting.

Do not be surprised if after watching and listening to a number of films, some readjustment of the controls are found desirable.

Characteristics

The curves in Fig. 7 and 8 show the amplitude vs frequency characteristics of the centre channel and surround channel respectively. The curves were obtained with an audio analyser. It is evident that they correspond closely with the descriptions. The –3 dB point of the centre channel is at 30 Hz. The –3 dB points of the surround channel are at 50 Hz and 6 kHz; the –6 dB bandwidth is roughly 30 Hz to 7 kHz.

Figure 8 shows that it does not make sense to use tweeters with a linear characteristic up to 20 kHz for the surround channels.

Both curves make it clear that a subwoofer must be connected to the main channels and not to the centre channel or surround channel.

Parts list

Resistors:
R_1, R_2, R_{27}, R_{28}, R_{45}, R_{50}, R_{53}–R_{56} = 47 kΩ
R_3, R_4 = 1 kΩ (see text)
R_5–R_8, R_{11}, R_{23} = 10.0 kΩ, 1%
R_9, R_{21} = 11.8 kΩ, 1%
R_{10}, R_{22} = 11.0 kΩ, 1%
R_{12}, R_{24} = 6.98 kΩ, 1%
R_{13}, R_{25} = 4.32 kΩ, 1%
R_{14}, R_{26} = 5.62 kΩ, 1%
R_{15} = 18 kΩ
R_{16} = 22 kΩ
R_{17}, R_{18}, R_{36} = 100 kΩ
R_{19}, R_{20} = 5.6 kΩ
R_{29}, R_{31} = 2.2 kΩ
R_{30}, R_{32} = 6.8 kΩ
R_{33}, R_{34}, R_{37}, R_{42}, R_{57}, R_{59}, R_{61}, R_{63} = 10 kΩ
R_{35} = 100 Ω
R_{38} = 2.7 kΩ
R_{39} = 270 kΩ
R_{40}, R_{41} = 560 kΩ
R_{43} = 15 kΩ
R_{44} = 12 kΩ
R_{46}, R_{48} = 39 kΩ
R_{47}, R_{49}, R_{51}, R_{52} = 1 kΩ
R_{58}, R_{62} = 560 Ω
R_{60}, R_{64} = 4.7 Ω
R_{65} = 5.6 MΩ
R_{66} = 10 Ω
P_1 = 50 kΩ preset
P_2–P_7 = 10 kΩ preset

Capacitors:
C_1, C_2, C_{25}, C_{27}, C_{45}, C_{52} = 2.2 µF, polypropylene, pitch 5 mm
C_3, C_5, C_{16}, C_{18} = 2.2 nF
C_4, C_{17} = 3.9 nF
C_6, C_{19} = 1 nF
C_7, C_8, C_{20}, C_{21}, C_{30}–C_{34}, C_{39}, C_{44}, C_{47}, C_{49}, C_{51}, C_{54}, C_{56}, C_{58} = 100 nF
C_9 = 100 pF polystyrene, axial
C_{10}, C_{11}, C_{13}, C_{35}, C_{63}, C_{64}, C_{67}–C_{70} = 100 nF, ceramic
C_{12}, C_{36}, C_{60} = 47 µF, 25 V, radial
C_{14}, C_{26} = 4.7 µF, 63 V, radial
C_{15} = 270 pF
C_{22} = 68 pF
C_{23}, C_{59}, C_{61}, C_{62} = 10 µF, 63 V, radial
C_{24} = 1 µF, polypropylene, pitch 5 mm
C_{28} = 1 µF, 63 V, radial
C_{29} = 220 µF, 10 V, radial
C_{37}, C_{38} = 47 µF, 16 V, radial
C_{40}, C_{41} = 100 µF, 25 V, radial
C_{42} = 150 nF
C_{43} = 68 nF
C_{46}, C_{53} = 22 µF, 40 V, radial
C_{48}, C_{50}, C_{55}, C_{57} = 220 µF, 25 V, radial
C_{65}, C_{66} = 4700 µF, 25 V, radial
C_{71} = 820 nF

Inductors:
L_1, L_2 = 100 µH
L_3 = 150 µH

Semiconductors:
D_1–D_4 = zener diode, 4.7 V
D_5 = 1N4148
D_6 = bi-colour LED (green/red)
D_8, D_9 = 1N4004
B_1 = SB102, 10 A, 100 V, for PCB mounting
T_1 = BC517

Integrated circuits:
IC_1, IC_2 = NE5532
IC_3 = MN3101
IC_4 = MN3008
IC_5 = TL072
IC_6 = NE571
IC_7 = TL084
IC_8 = 4030
IC_9 = TDA1074A
IC_{10}, IC_{11} = TDA2040
IC_{12} = 7808
IC_{13} = 7908

Miscellaneous:
K_1, K_2 = audio socket for PCB mounting
K_3 = 3-way terminal block, pitch 5 mm
Re_1 = relay 12 V, 5 A, 270 Ω
Heat sink SK57, 37.5 mm high*
Ceramic washers Type AOS220*
Enclosure 300×45×210 mm (W×H×D) ($11^7/_8$×$1^3/_4$×$8^1/_4$ in)
Mains transformer with 2×12 V, 80 VA secondary
Mains entry with integral fuse holder and 500 mA slow fuse
Loudspeaker terminals (spring loaded) or suitable audio sockets
Mains on/off switch
PCB Order No. 950012-1

[950012]

* Available from Dau (UK) Ltd, 70–75 Barnham Road, Barnham PO22 0ES, telephone (01243) 553 031