the LOUD mouth

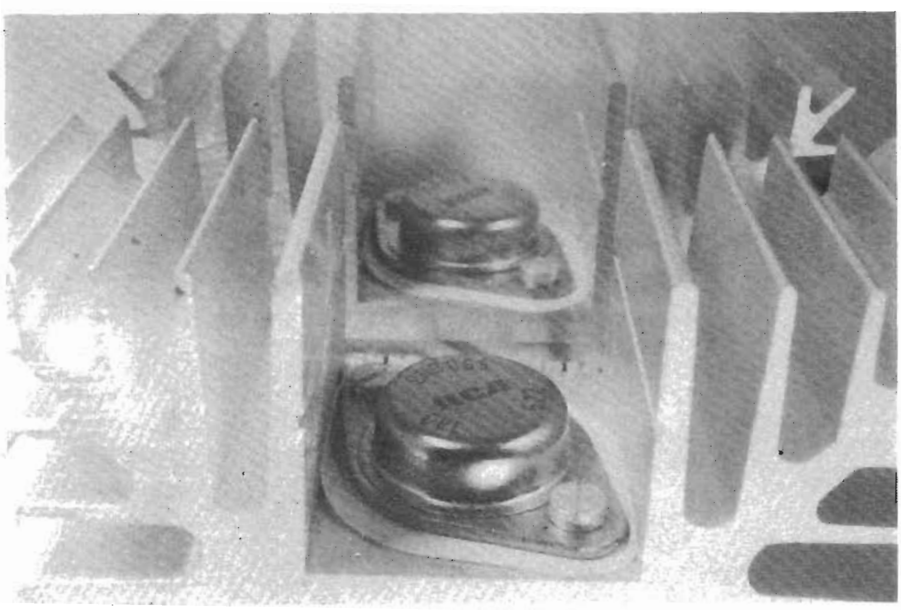
Loud hailers are used wherever people need to make themselves heard over a large distance. The design described here is meant for use in cars and thus derives its power from the car battery. This means that the maximum voltage available is only 12 Volts, so some unconventional circuits are required to produce sufficient audio output power. A symmetrical output stage using a transformer is used to feed a 4 ohm loudspeaker, which results in a 16 to 1 power step-up when compared with a conventional push-pull arrangement. The amplifier operates in class C, which is a bit unusual for audio use. To reduce the effects of (class C) crossover distortion a high frequency bias is superimposed on the input audio signal. The effect of these somewhat unusual features are described in the following paragraphs.

The primary power supply is limited to the 12 volts derived from the car battery. It would, theoretically, be possible to have this voltage stepped up by some sort of converter device which would enable a higher power output level to be obtained with a given loudspeaker impedance, but in practice this leads to poor efficiency and great expense. A more practical approach is required.

First, let us consider the problem in greater detail. The maximum power supplied by a single ended pushpull stage is roughly calculated from the following rule-of-thumb equation:

$$P_{max} = V^2/8R_L$$

in which V stand for the power supply voltage and R_L for the load impedance. This formula completely disregards all losses in the output stage. The actual maximum output power will be even lower. This means that for a 12 volt power supply and a 4 ohm loudspeaker the theoretical maximum output power (not including losses) would be approximately 4.5 W, obviously on the low side for a 'loud' hailing system.



Much more power for a given supply voltage can be delivered by a bridge-type output stage. This consists of two identical output stages driven in anti-phase. In this way the voltage swing is doubled, which results in a 4 fold power increase (P_{max} approximating $V^2/2\,R_L$). Under the same given conditions this arrangement will yield us a maximum output power of about 18 W. In the final design, the power available from the output stage is stepped up by a further factor of 4 by using a 2 to 1 load matching transformer. P_{max} is now approximately $V^2/\frac{1}{2}R_L$, which (theoretically) would give 72 W into a 4 ohm load at 12 V. Allowing for unavoidable losses, the actual output will be about 40 W. The signal delivered by this final stage will not be exactly 'hi-fi'. The main objective with this design was to help a 'soft' speaker to become a 'loud' speaker. High fidelity is of secondary importance. In spite of the relatively poor audio quality, however, good intelligibility is maintained.

Main amplifier

The main amplifier circuit is shown in figure 1. There are two inputs, one for speech, one for music, with mixing controls P1 and P2. The input signal is passed on to a phase splitter that produces the in-phase signal (at the splitter emitter) and the anti-phase signal (at the collector) required for the symmetrical pushpull stage.

To increase the output power even further, an output transformer is used.

The function of this output transformer can be explained as follows. If the amplifier is fully driven, at one 'end' of the output swing T5 will be fully conductive and T6 will be cut off. At that instant, the current flowing through

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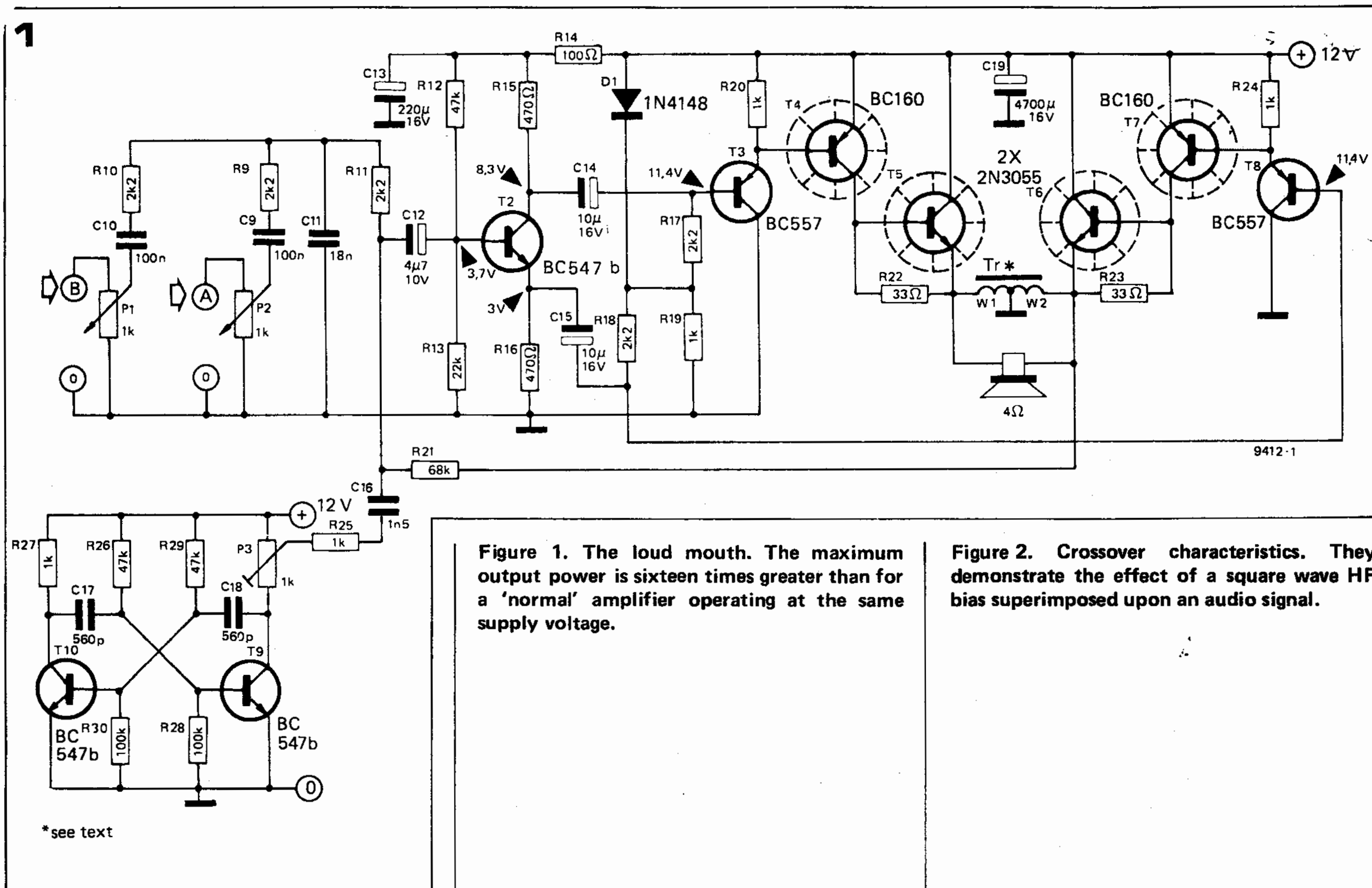

Figure 1. The loud mouth. The maximum output power is sixteen times greater than for a 'normal' amplifier operating at the same supply voltage.

transformer winding W1 will set up a voltage across this winding of 12 V. The emf induced in the opposite transformer winding will also be 12 V, which results in 24 volts across the entire transformer and, therefore, across the loudspeaker terminals. This is the peak swing for a half-period of the output signal. The r.m.s. value is 0.7 x 24 V, which is approximately 17 V. The corresponding maximum output power will then be approximately $P_{max} = \frac{U^2}{R_L} = \frac{17^2}{4} \approx 72$ W. As stated earlier, the inevitable losses will reduce the maximum output power in practice to about 40 W.

To cut down distortion, resistor R21 provides some degree of negative feedback.

The HF bias

The output stage operates in class C, so the drive signal must reach a certa[illegible] level before the output responds. [illegible] advantage of class C operation is the relative immunity to temperature fluctuations. However, class C also has a major drawback: if no special precautions are taken, the 'dead zone'

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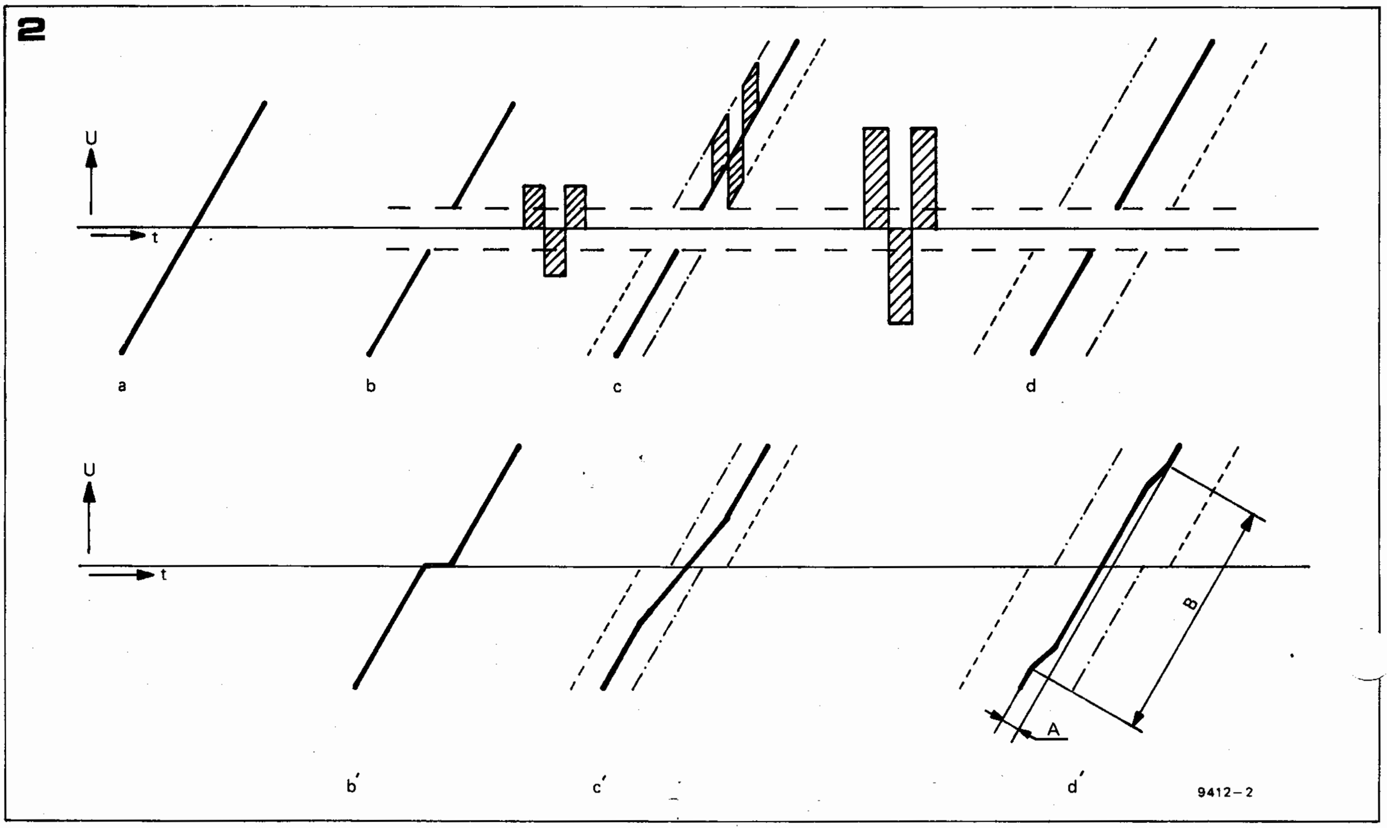
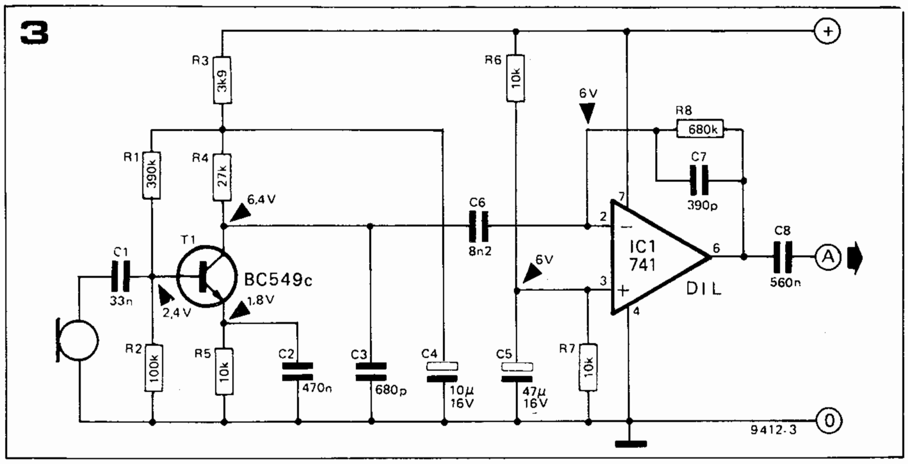
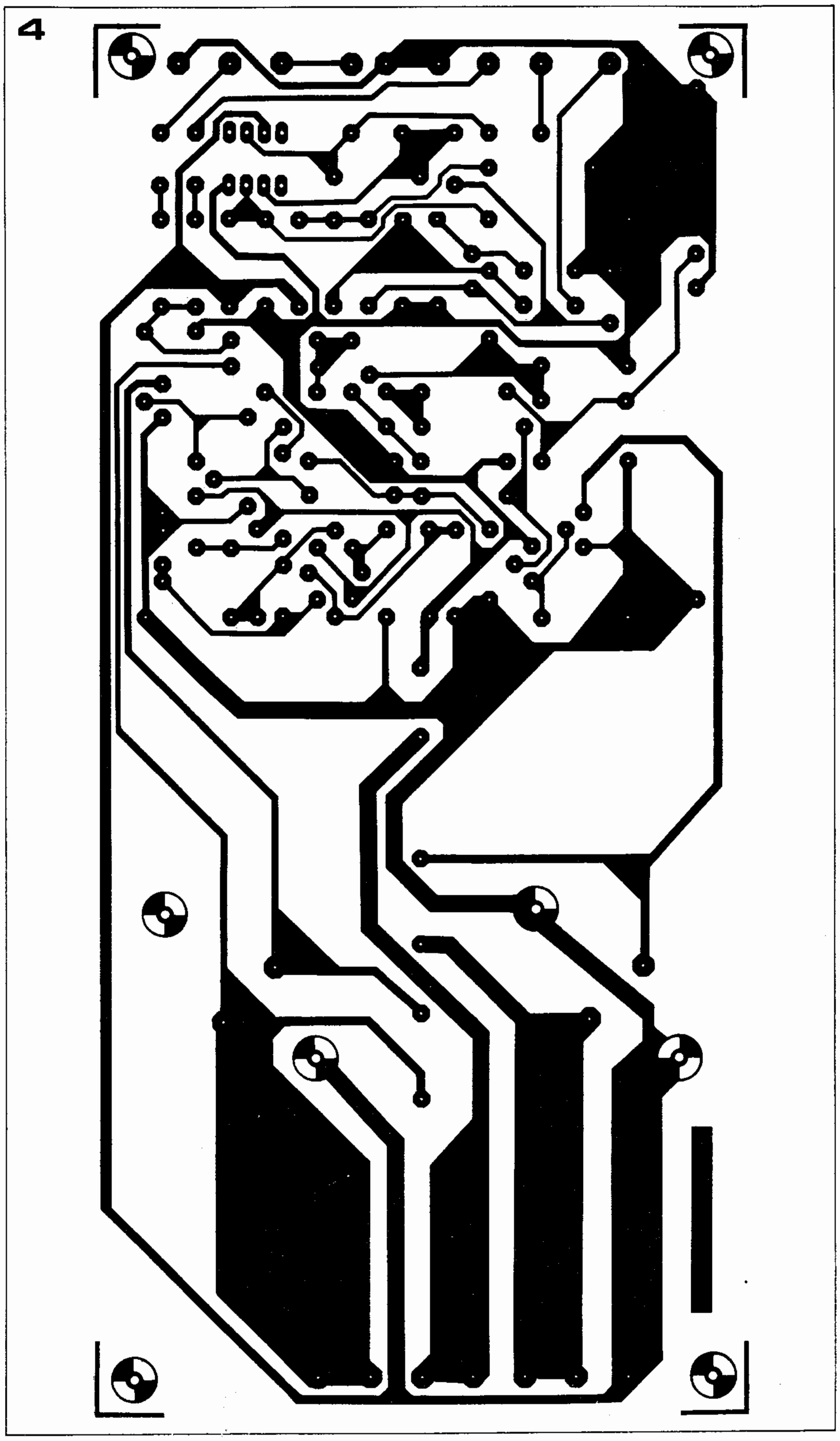


Figure 2. Crossover characteristics. They demonstrate the effect of a square wave HF bias superimposed upon an audio signal.

Figure 3. Microphone preamplifier. The value of R8 can be modified if higher or lower input sensitivity is required.

Figure 4. Printed circuit board and component layout (EPS 9412).





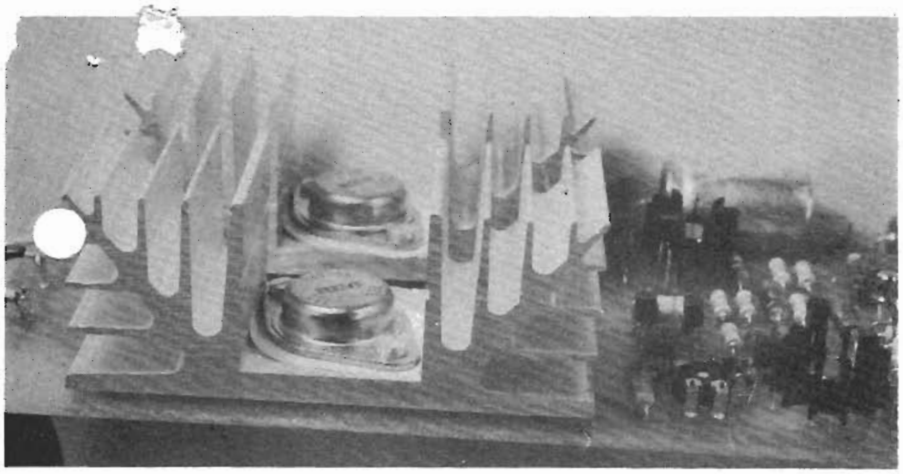
Complete parts list

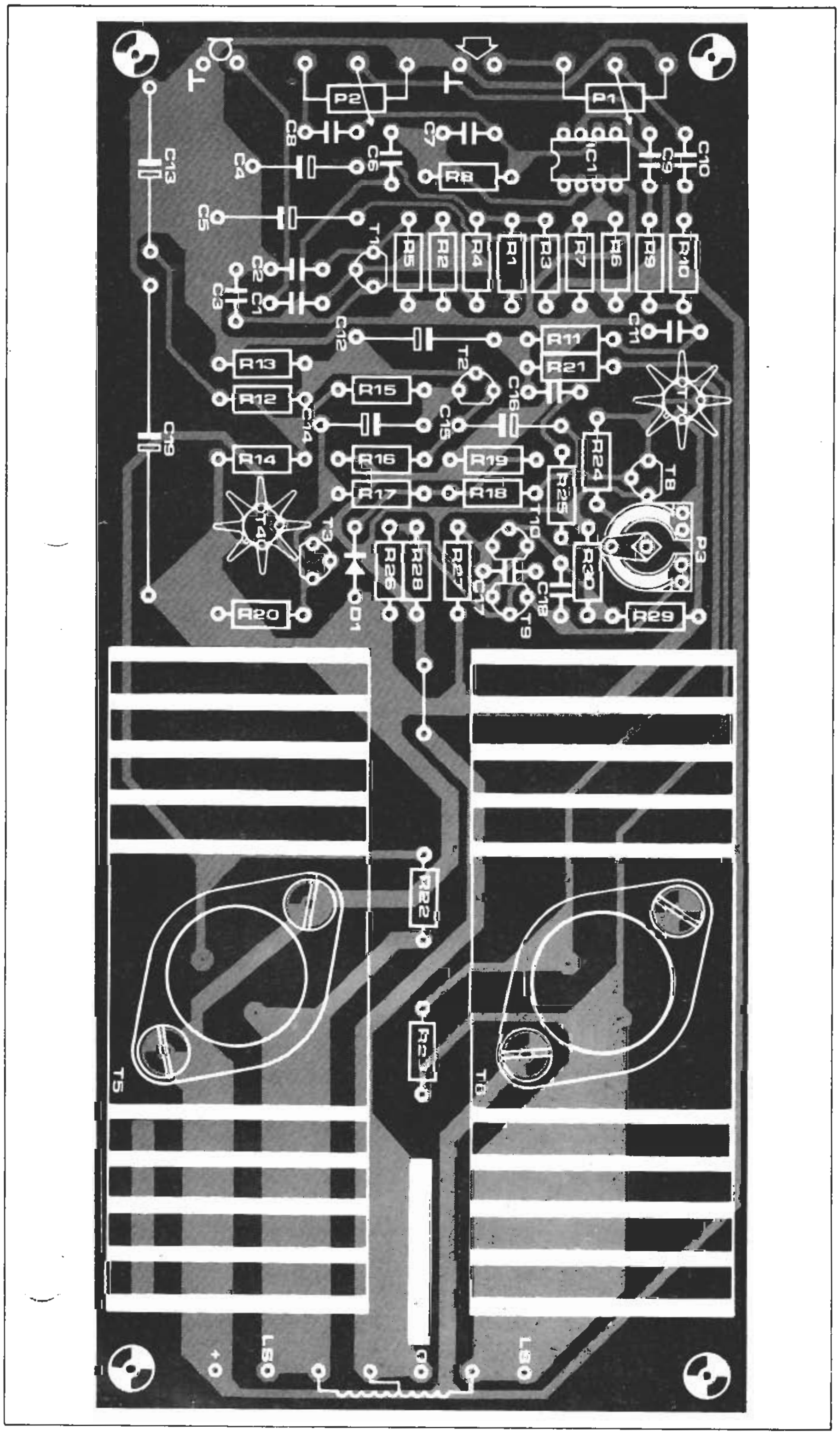
Resistors:
R1 = 390 k
R2,R28,R30 = 100 k
R3 = 3k9
R4 = 27 k
R5,R6,R7 = 10 k
R8 = 680 k (see text)
R9,R10,R11,R17,R18 = 2k2
R12,R26,R29 = 47 k
R13 = 22 k
R14 = 100 Ω
R15,R16 = 470 Ω
R19,R20,R24,R25,R27 = 1 k
R21 = 68 k
R22,R23 = 33 Ω
P1,P2 = 1 k log. pot.
P3 = 1 k preset pot.

Capacitors:
C1 = 33 n
C2 = 470 n
C3 = 680 p
C4,C14,C15 = 10 μ/16 V
C5 = 47 μ/16 V
C6 = 8n2
C7 = 390 p
C8 = 560 n
C9,C10 = 100 n
C11 = 18 n
C12 = 4μ7/10 V
C13 = 220 μ/16 V
C16 = 1n5
C17,C18 = 560 p
C19 = 4700 μ/16 V

Semiconductors:
IC1 = 741
T1 = BC 549c, BC 109
T2,T9,T10 = BC 547b, BC 107
T3,T8 = BC 557, BC 177
T4,T7 = BC 160
T5,T6 = 2N3055
D1 = DUS

Miscellaneous:
Tr = transformer 2 x 12 V/3 A





inherent to this type of operation will cause objectionable crossover distortion. In this design the distortion was reduced by superimposing an HF bias signal onto the input audio. This HF bias consists of nothing more than a 30 kHz square wave signal. It is generated by an astable (T9, T10). The effects of this HF bias can be better explained from the graphs shown in figure 2.

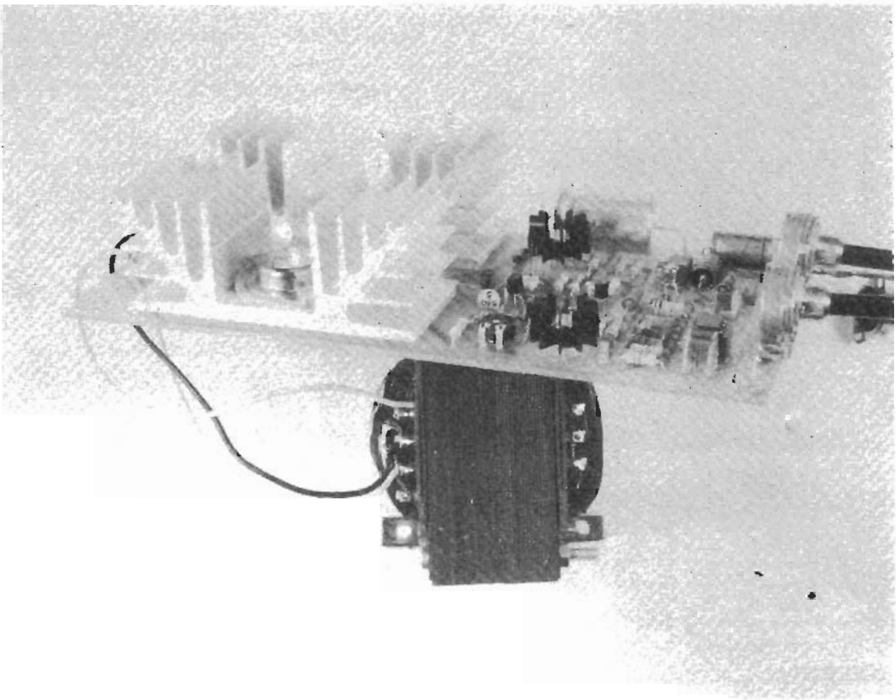
Figure 2a shows a given input audio signal. The central dead zone in which there is no output response is shown in figure 2b, the resultant output according to 2b′ clearly indicating severe crossover distortion. Figure 2c shows the effect of a square wave superimposed upon the original input signal. The output signal, which also has this square wave superimposed on it, would fill the area between the dashed lines in figure c′. However, after filtering, the output signal becomes equal to the average value, which is shown as a bold line in figure c′. This curve demonstrates a more continuous transition over the dead zone, resulting in a much lower distortion.

When the HF bias amplitude is increased, as shown in figure 2d, the smoothed output signal will be as indicated by the bold line in 2d′. Evidently, crossover conditions are determined by the HF bias amplitude and the offset 'A' which in turn depends on the width of the dead zone. This width may be affected by temperature changes, but this has little effect on the audible distortion. The transition zone 'B' depends directly on the HF bias amplitude, which in turn depends on the power supply voltage. However, 2d′ indicates that fluctuations in the supply voltage and, consequently, fluctuations in the width of zone 'B' will not audibly affect the distortion in the output signal. P3 is used to set the HF bias amplitude.

The microphone preamplifier

The sensitivity (gain) of the main amplifier, shown in figure 1, is not very high. For this reason a preamplifier stage must be added if a dynamic microphone is to be used. The circuit for a suitable preamp is shown in figure 3. It is included on the printed circuit board.

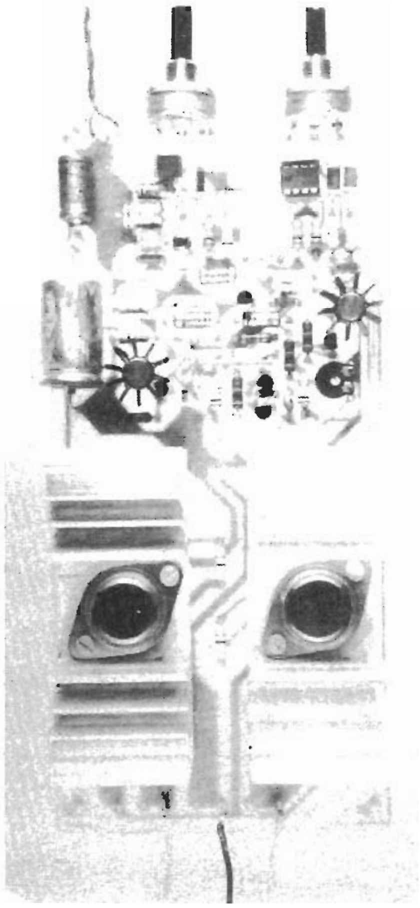
The discrete transistor stage (T1) acts both as an impedance matching device and a low noise amplifier. Since the noise characteristics of the preamplifier output stage, a 741 opamp, are not all too favorable, a satisfactory signal-to-noise figure could not be obtained without the help of this additional transistor. The overall gain now becomes so high that it will readily lead to clipping of the preamplifier output signal. In practice, this is usually not so serious for speech: the intelligibility of spoken messages is still quite good, and it has the advantage that the average output power of the system is greatly increased. However, if the distortion is too objectionable the value of R8 can be decreased, reducing the sensitivity to a suitable level. On the other hand, if the gain is too low then R8 can be increased



to a suitable value.
The preamplifier is meant for 50 kΩ microphones, but also works quite well with 500 ohm mikes.

Some practical advice

Figure 4 shows the printed circuit board and component layout. The HF bias is set with P3. Connect an ammeter in the power supply line and adjust P3 so that under no-signal conditions the current is about 300 to 400 mA.
To prevent any mishaps when the equipment is being connected to the car battery it is recommended to insert a fuse of approximately 5 A in the power supply line.



The power transistors T5, T6 and the drivers T4, T7 require heat sinks. Cooling fins will suffice for T4 and T7; T5 and T6 must be mounted on a heat sink with a thermal resistance of 3°C/W or less. There is sufficient room on the p.c.b. for all of these sinks.
T_r can be a standard mains transformer with two 12 V, 3 A secondaries. The primary is not used, but since it will develop an uncomfortably high voltage its terminals should be well insulated.
The circuit is designed for 4 ohm loudspeakers. However, if 40411 transistors are used for T5 and T6, a 2 ohm load is permissable. In that case the maximum output power will increase to some 80 W.